

Configuring Interoperability between Avaya IP Office and Avaya Business Communication Manager

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Contents

1.0 Introduction 3

1.1 Supported Features 3

1.2 Network Diagram 6

1.3 Supported Phones 6

1.4 Software Version 7

1.5 Hardware Platforms 7

2.0 Configuring Interoperability 7

2.1 Avaya IP Office Configuration 7

2.1.1 IP Office SIP Trunk Licensing 8

2.1.2 IP Office SIP Line to BCM 8

2.1.3 IP Office Incoming Call Route 12

2.1.4 IP Office Short Code 14

2.1.5 IP Office System Settings 15

2.1.6 IP Office Small Community Networking (SCN) 20

2.1.7 Verify Basic Connectivity 22

2.2 Business Communication Manager Configuration 23

2.2.1 BCM Keycodes 23

2.2.2 BCM Business Name 24

2.2.3 BCM IP Trunks 25

2.2.4 BCM SIP Trunking 26

2.2.5 BCM SIP Trunking - Public trunk 28

2.2.6 Telephony - Dialing Plan 32

2.2.7 BCM MCDN Line 38

3.0 Issues Found 41

1.0 Introduction

This document provides a description of the solution where a network of Avaya Business Communication Manager (BCM) is connected to a network of Avaya IP Offices. In this configuration, a single BCM is connected to a single IP Office using SIP Trunks.

1.1 Supported Features

- **Basic Call** – We verified that:
  - Caller receives Ring-back tone
  - Basic calls can be successfully established between IP Office and BCM with two-way talk path.
  - Proper Media Type is used.
- **Basic Call Completion** – We verified that:
  - Terminated calls are properly cleared from the phones.
  - Involved trunks and resources are released promptly.
- **Handling of busy called party** – We verified that:
  - Busy tone is properly delivered to the caller when a called party is busy.
  - Caller receives the right treatment based on the called party recovery options (Voicemail, forward).
  - Correct behavior where called party has Do Not Disturb active or when all lines were busy.
  - Busy tone is properly delivered when all trunk channels are busy.
- **DTMF** – We verified that:
  - DTMF tones are properly transmitted and processed over SIP trunks.
  - Voicemail systems properly processed the DTMF tones received from the remote side.
- **- Hold and Retrieve** – We verified that:
  - Calls can be successfully placed on hold from both sides
  - Calls can be successfully resumed from both sides.
- **Call Waiting presentation** – We verified that:
  - Waiting calls are successfully presented on the phones.
  - User is able to switch between the calls.
- **Called Number display** – We verified that:

- The Called Number is properly displayed on the calling party's phone.

- **Calling number and name display** – We verified that:
  - The calling number and name are properly displayed on the called party's phone.
- **Abandoned call –** We verified that:
  - Abandoned calls are properly cleared from both parties.
  - Abandoned calls are tracked in the History list.
  - The involved trunks and resources are released promptly when a call is abandoned.
- **Call Redirection**:
  - **Call Forward** – We verified:
    - The behavior of all three types of call forward: Forward All, Forward Busy and Forward No Answer.
    - That the call is successfully redirected to the forward destination.
    - The call is successfully connected with forward destination and there is two way talk-path.
    - That the CLID and Contact Name are updated accordingly.
    - Trunk channels and resources are used correctly
  - **Call Transfer** – We verified:
    - The behavior of both transfer types, blind and consultative.
    - That SIP trunk calls can be successfully transferred to a local extension, to the PSTN or back to the originating site, over a SIP trunk.
    - That local calls can be successfully transferred over a SIP trunk.
    - The CLID and Contact Name information are updated accordingly.
    - The hold and ring-back tones are properly presented to the transferred party.
  - **Transfer from Voicemail system** – We verified that:
    - Voicemail system is able to transfer the local parties over the SIP trunk (available only for IP Office Voicemail).
    - Voicemail system is able to transfer SIP trunk calls to a local extension.
  - **Pick-up** – We verified that:
    - SIP Trunk calls can be successfully picked up.
    - CLID and Contact Name are successfully updated.
  - **Hunt Group Coverage** – We verified that:
    - SIP Trunk calls are successfully redirected based on the Hunt Group Coverage settings (verified only on the IP Office side)

- CLID and Contact Name are successfully updated

- **Follow Me Here/ Follow Me To** – We verified that:

  - The SIP Trunk calls are successfully redirected based on the Follow Me Here and Follow Me To features.
  - CLID and Contact Name are successfully updated.

- **Twinning** – We verified that:

  - - The SIP Trunk calls are successfully redirected based on the Twinning feature (verified only on the IP Office side)

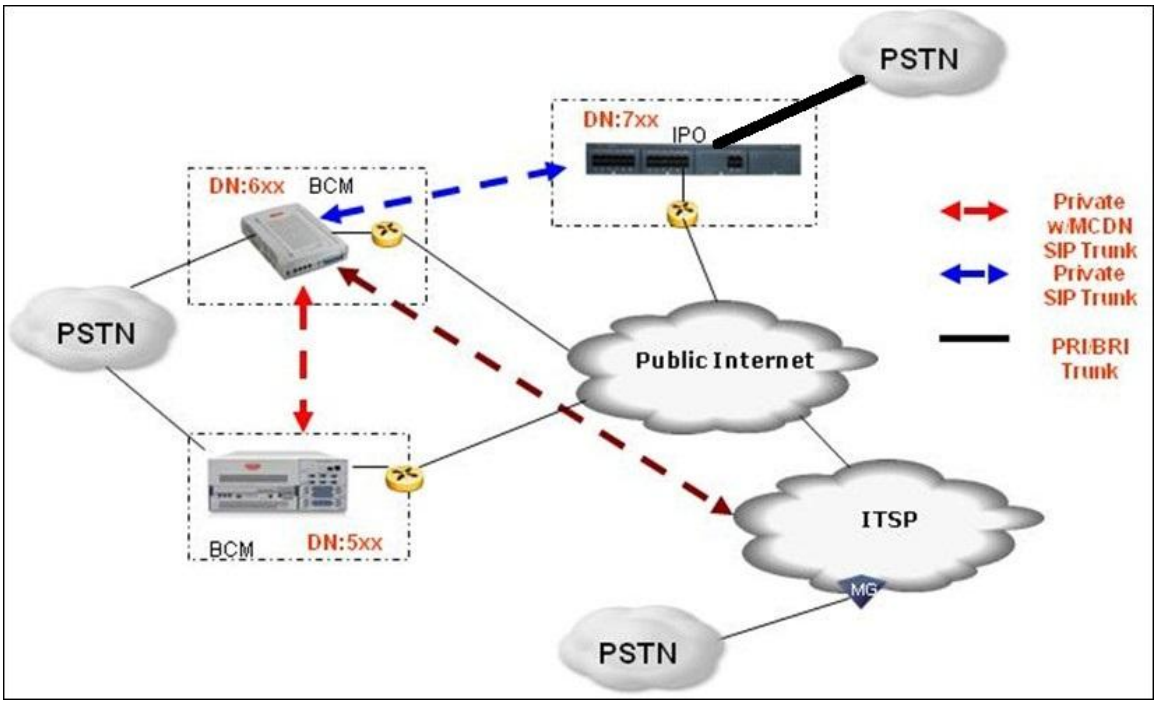
- **Conferencing** – We verified:

  - Both types of conference, Ad hoc and Conference Meet Me.
  - Ad hoc conferences with 3 to 6 parties.
  - That each party has two way speech path.
  - Trunk and resource utilization.
  - Voice quality.

- **PSTN Toll Bypass** – We verified:

  - That PSTN Toll Bypass calls can be successfully established with two-way talk path.
  - That the CLID is properly presented.
  - Involved trunks and resources are properly released when a call is terminated.

1.2 Network Diagram



1.3 Supported Phones

IP Office Phones	
Types	**Series**
DS	1400
	5400
	2400
	9500
	M-series
	T-series
	T3
SIP	1100e
	1200
	Softphone
	A1010/1020/1030/1040
	B179 - Konftel
	3-rd party
H323	1600
	3600
	4600
	5600
	9600

	96x1
	T3
DECT	ASCOM
Analog	B149 - Konftel

BCM Phones	
Type	**Series**
Unistim	2000
	1100e
	1200
	2050 - Softphone
DS	M-Series
	T-series

1.4 Software Version

- IP Office 8.1 running Essential, Preferred or Advanced Editions.
- BCM 6.0 with latest Service Update

1.5 Hardware Platforms

- IP Office 500v2
- IP Office Mid Market
- BCM450
- BCM50

2.0 Configuring Interoperability

2.1 Avaya IP Office Configuration

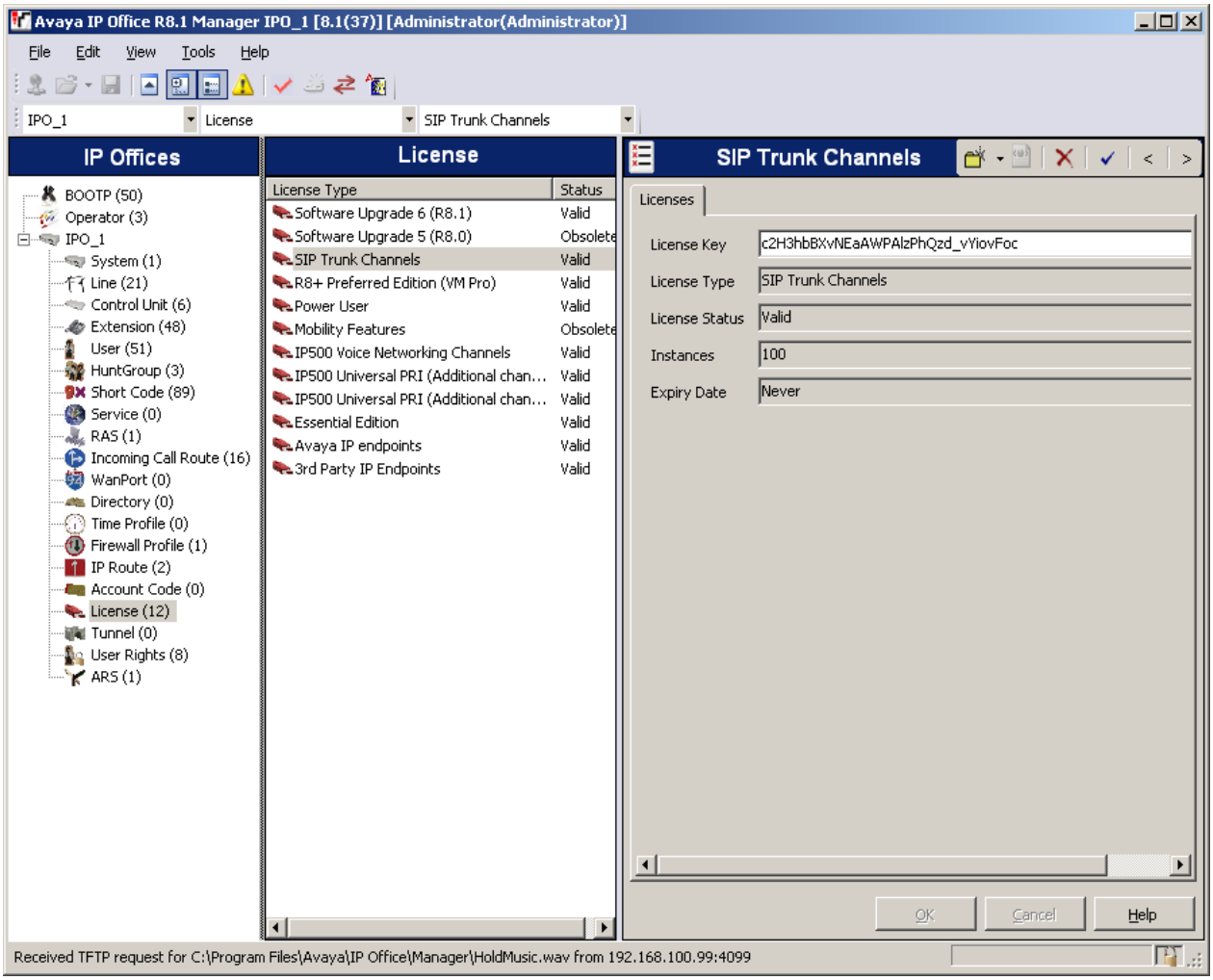
This section provides the procedures for configuring Avaya IP Office using the Avaya IP Office Manager application.

The procedure covers the following areas:

- SIP Trunk Licensing
- SIP Line to BCM
- Incoming Call Route
- Short Code
- System Settings
- Small Community Networking (SCN)
- Verify Basic Connectivity

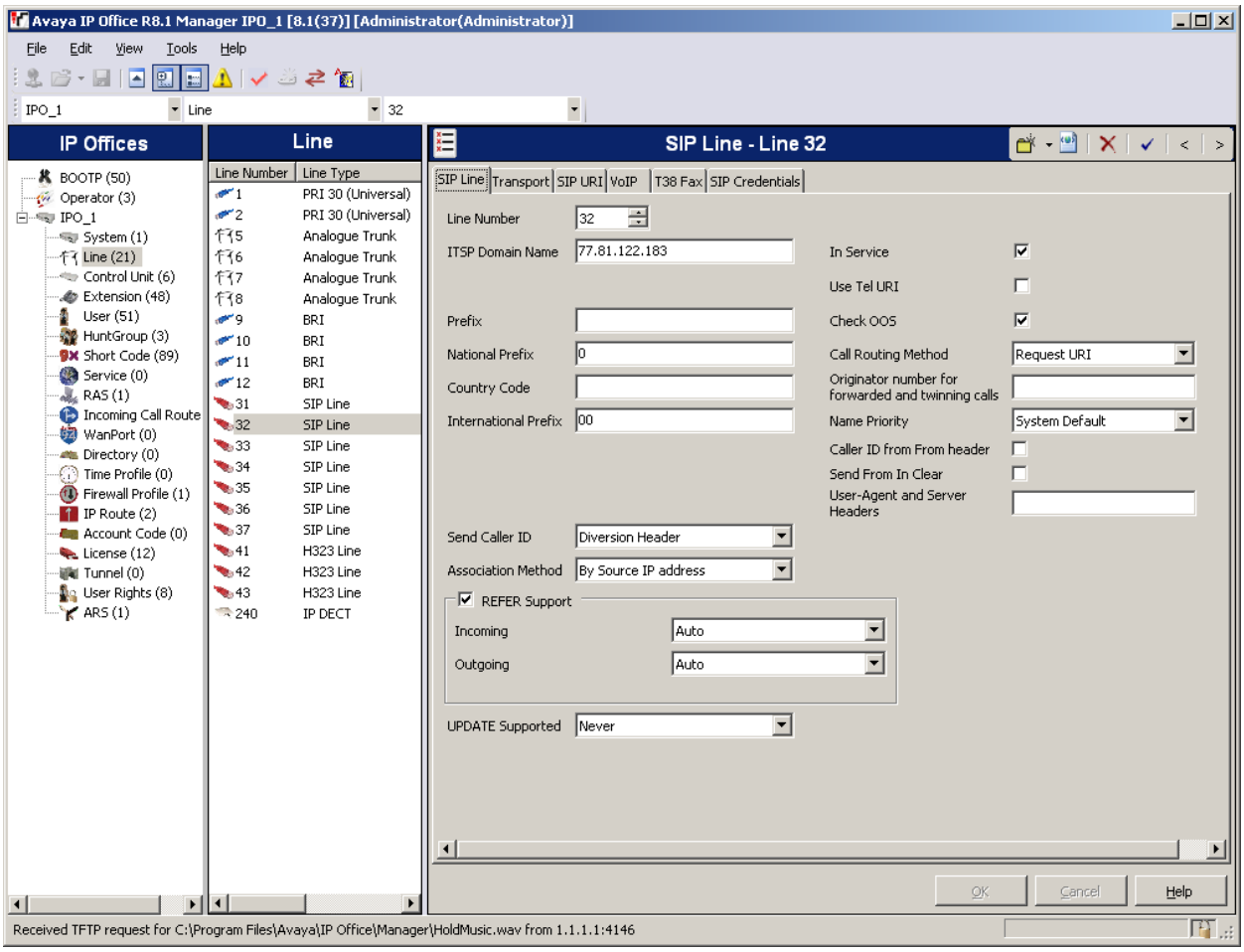
2.1.1 IP Office SIP Trunk Licensing

From the configuration tree in the left pane, select **License > SIP Trunk Channels** to display the **SIP Trunk Channels** screen in the right pane. Verify that the **License Status** is 'Valid' and that the 'Instances' field contains the appropriate number of allowed simultaneous SIP Trunking calls.



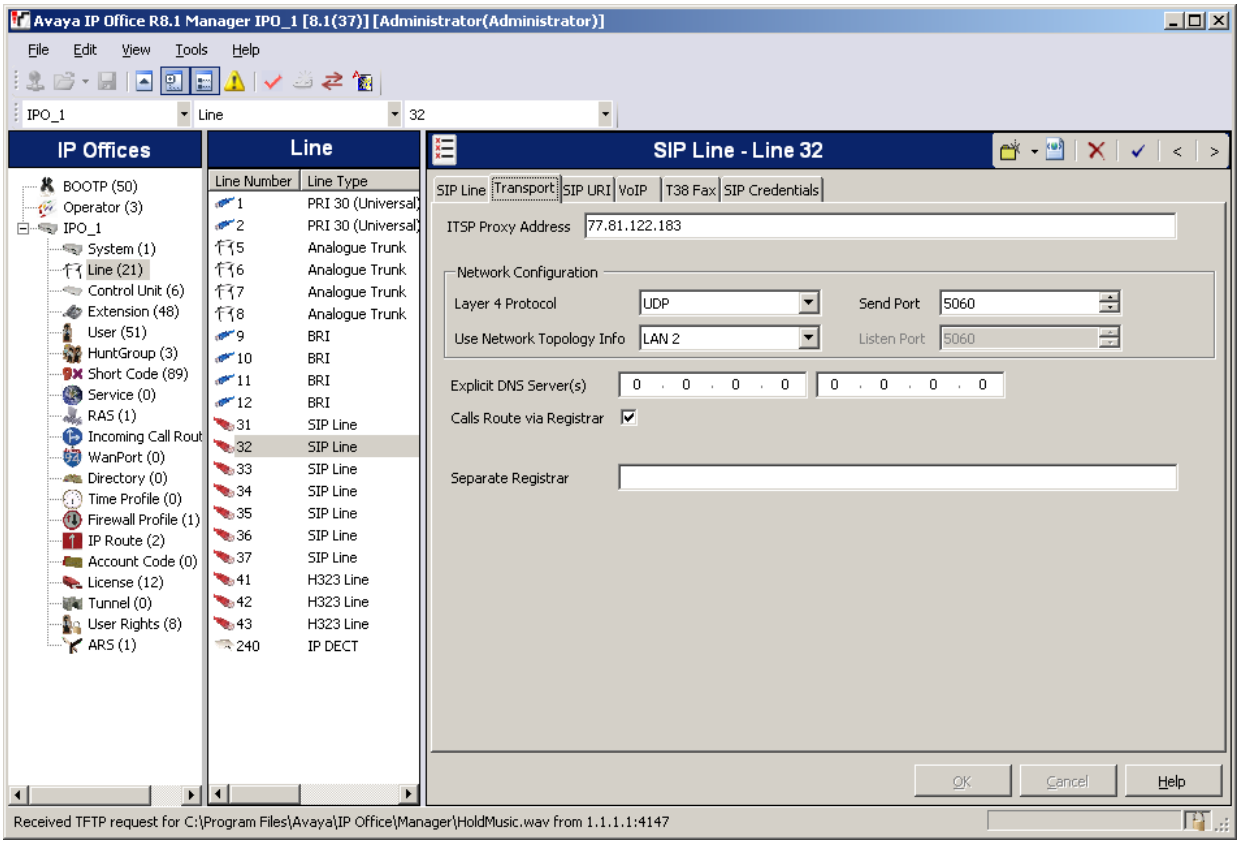
2.1.2 IP Office SIP Line to BCM

From the configuration tree in the left pane, right-click on **Line** and select **New > SIP Line** to add a new SIP Trunk.
In the **SIP Line** tab, enter the following:
- **ITSP Domain Name** - Enter the BCM's IP Address or Domain Name.
- Make a note of the **Line Number**.
- **Send Caller ID** should be set to **Diversion Header**.
- Retain the default values for all other fields.



In the **Transport** tab, enter the following:

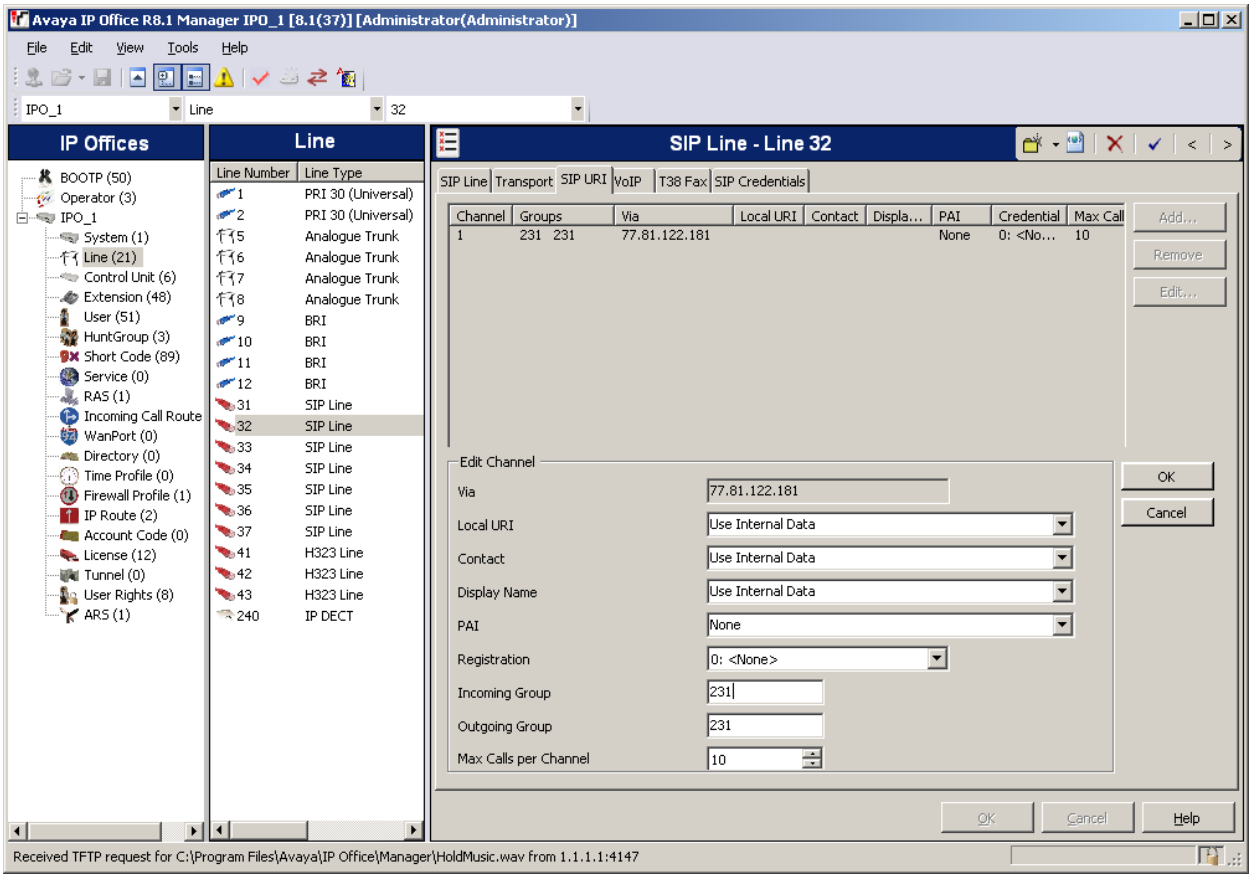
- **ITSP Proxy Address -** Enter the IP address of **BCM**. This is the SIP Proxy address used for outgoing SIP calls.
- **Layer 4 Protocol –** Retain the default value, **UDP**. This field sets which protocol the line uses for sending and receiving SIP packets.
- **Send Port -** Retain the default value, **5060**. This field sets the port to which the system sends outgoing SIP calls.
- **Use Network Topology Info -** Select the appropriate LAN interface. This field associates the SIP line with the **Network Topology** settings.
- Retain Default values for all other fields.



In the **SIP URI** tab, select **Add** button and enter the following:
- **Local URI** - Enter "**Use Internal Data**". Use the SIP Name value from the User > SIP tab of the user making the SIP call. This field sets the 'From' field for outgoing SIP calls using this URI.
- **Contact** - Enter "**Use Internal Data**". Use the Contact value from the User > SIP tab of the user making the SIP call. This field sets the 'Contact' field for SIP calls using this URI.
- **Display Name** - Enter "**Use Internal Data**". Use the SIP Display Name (Alias) value from the User > SIP tab of the user making the SIP call. This field sets the 'Name' value for SIP calls using this URI.
- **PAI** – The default setting "None", should be retained. When 'None' is selected, the P-Preferred-Identity header is used instead of the P-Asserted-Identity, in order to ensure compatibility with legacy networks.
- **Registration** – The default setting should be retained.
- **Incoming Group** - Enter an Incoming Group ID. This ID will be used when defining the corresponding Incoming Call Route. The Incoming Group ID to which a line belongs is used to match it to incoming call routes in the system configuration. The matching incoming call route is then used to route incoming calls.
- **Outgoing Group**: Enter an Outgoing Group ID. This ID will be used when defining the corresponding Short Code.

It is recommended that the Incoming and Outgoing Group ID be appropriate to Line Number, for easy maintenance.

- **Max Calls per Channel** - This field sets the maximum number of simultaneous calls that can use the URI before the system returns busy to any further calls. This number is related to the number of 'Instances' available for 'SIP Trunk Channels' license.



In the **VoIP** tab:

- **Codec Selection** - this field defines the codecs order offered during call setup.

For **System Default**, the codec preference defined under System > Codecs tab will be used. The **Custom** option, available under the drop-down list, allows a specific codec selection to be made.

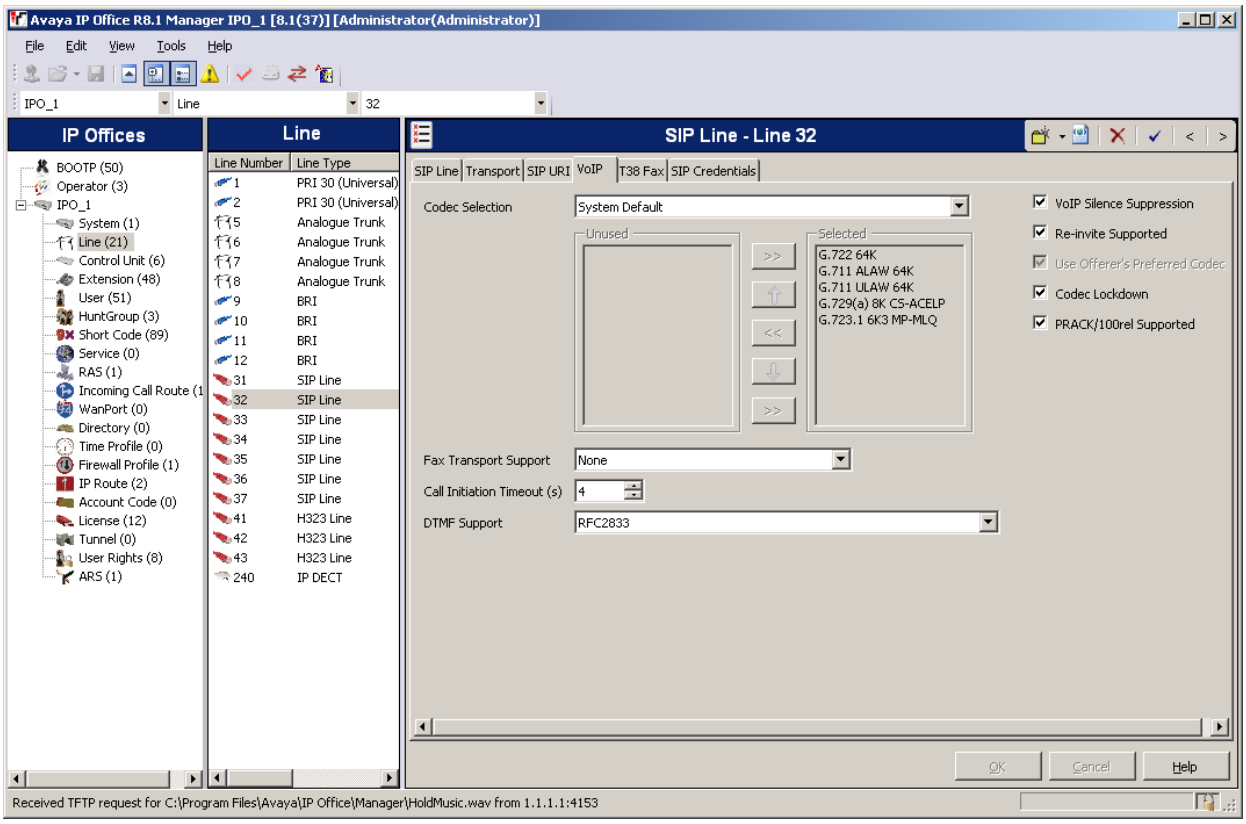
- **VoIP Silence Suppression** -When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This setting should be coordinated with BCM.

- **Re-invite Supported** – Should be enabled. When enabled, re-invite can be used to change the characteristics of the session.

- **Use Offerer's Preferred Codec –** Should be enabled. This option is only used when IP Office sends INVITES.

- **Codec Lockdown –** Should be enabled. If enabled, when the system receives an SDP answer with more than one codec from the list of offered codecs, it sends an extra re-INVITE using just a single codec from the list and resubmits a new SDP offer with just the single chosen codec. RFC3264 is covered if both Re-Invite Supported and Codec Lockdown are enabled.

- **PRACK/100rel Supported –** Should be enabled. When enabled, the SIP Trunk supports Provisional Reliable Acknowledgements, meaning that it ensures that provisional responses, such as announcement messages, have been delivered.
- **Fax Transport Support** – The default setting should be retained, 'None'.
- **Call Initiation Timeout** - The default setting should be retained. This option sets how long the system should wait for a response to its attempt to initiate a call before following the alternate routes set in an ARS form.
- **DTMF Support** - The default settings should be retained, RFC2833.
This setting is used to select the method by which DTMF key presses are signaled to the remote end.



2.1.3 IP Office Incoming Call Route

Incoming call routes are used to match incoming calls with destinations. Routes can be based on the incoming line group, the type of call, incoming digits or the caller's ICLID.
From the configuration tree in the left pane, select **Incoming Call Route,** right-click and select **New** to add a new Incoming Call Route.

In the **Standard** tab:

- **Bearer Capability –** Should retain the default setting, **Any Voice.**
- **Line Group ID –** Incoming Group ID configured for SIP line under URI tab should be used.
- **Incoming Number** – Should retain the default value, blank, to match all calls that do not match other records. Matches the digits presented by the line provider. By default, this is a right-to-left matching

.

- **Incoming Sub Address** - Should retain default value, blank, to match all calls.

It matches any subaddress component sent with the incoming call.

- **Incoming CLI –** Should retain default value, blank to match all.

Enter a number to match the caller's ICLID provided with the call.

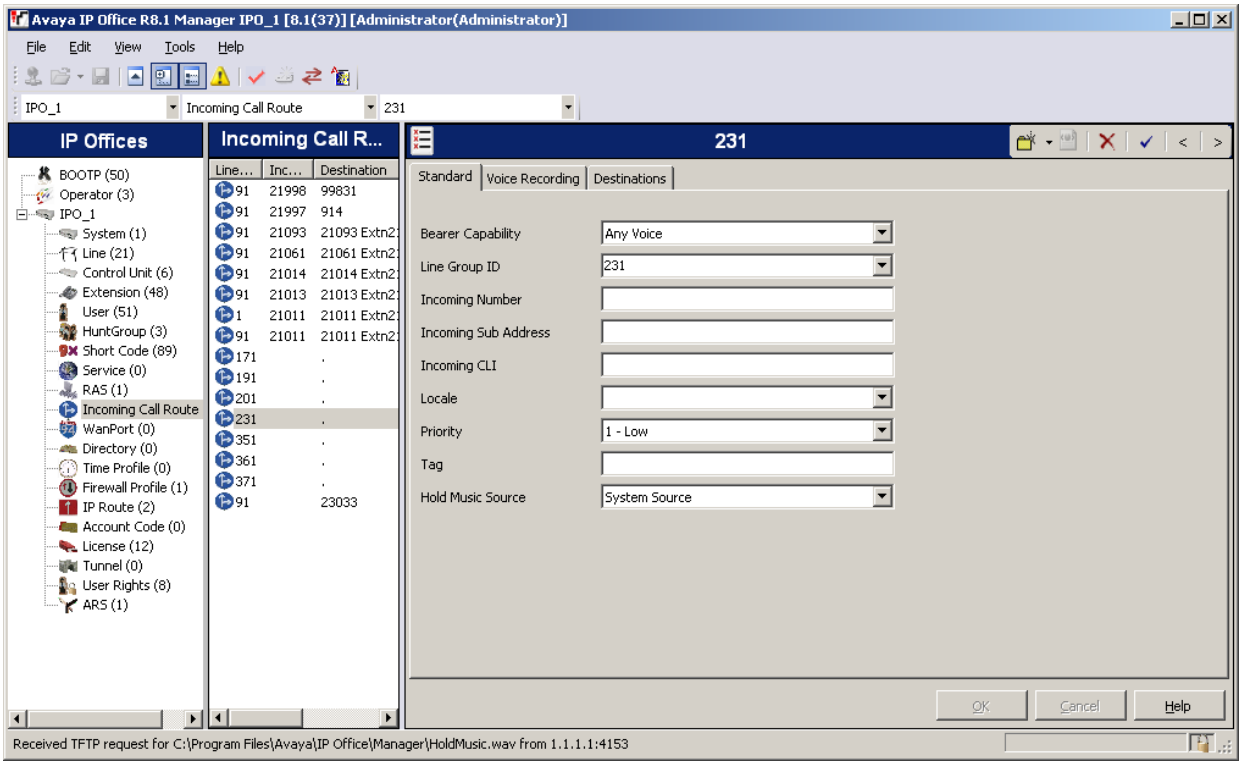
- **Locale** - Should retain the default value, blank, to use System locale.

This option specifies the language prompts, if available, that voicemail should use for the call if it is directed to voicemail.

- **Priority** - Should retain the default value, 1-Low.

This setting allows incoming calls to be assigned a priority used in case of queued calls.

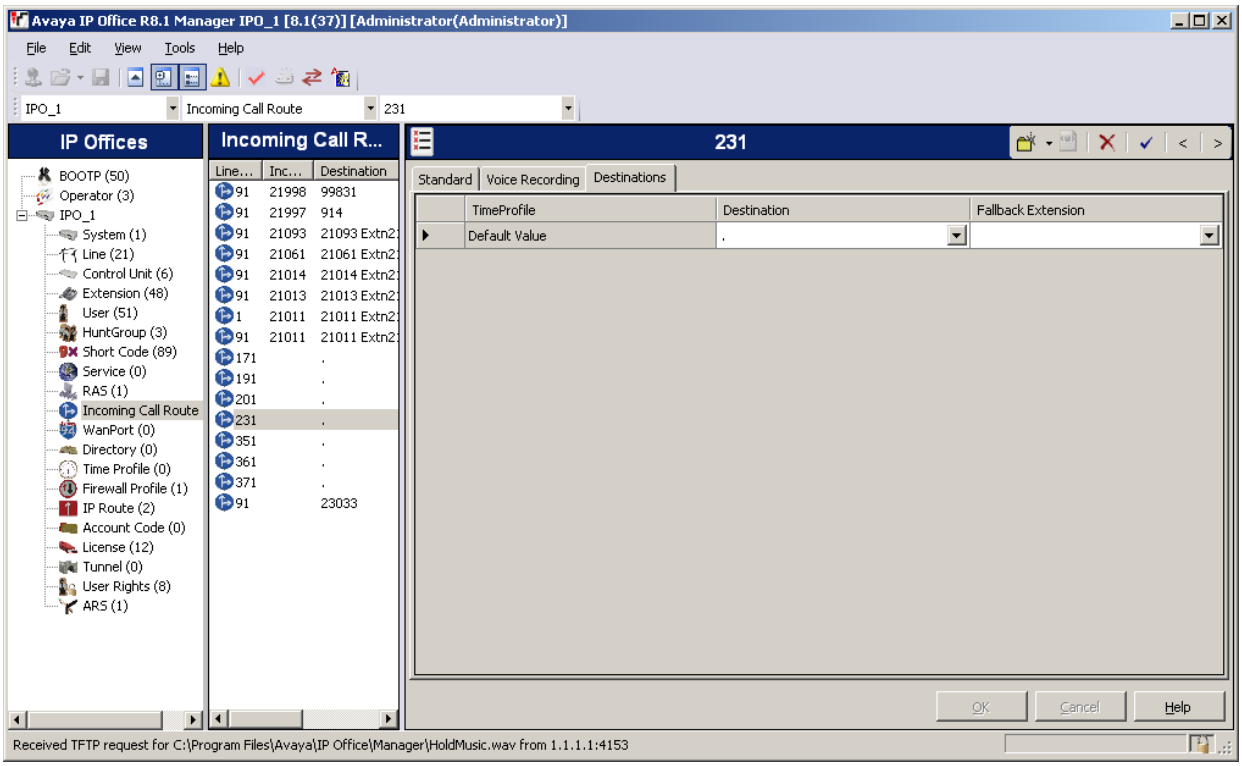
- **Tag –** Should retain the default value, blank. This allows a text tag to be associated with calls routed by this incoming call route.
- **Hold Music Source –** Should retain the default value, System Source.



In the **Destination** tab:

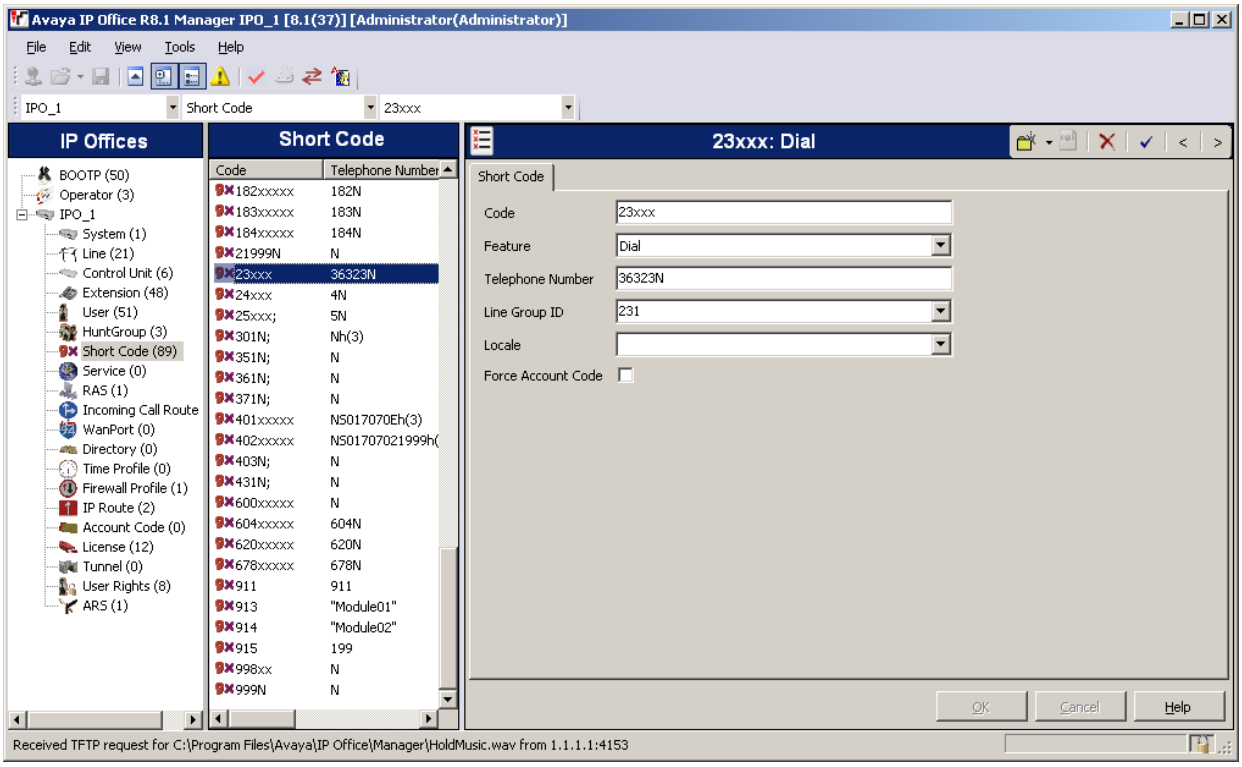
- **Default Value** field - Enter "**.**" into the Destination column to match all the destinations available.

Or you can enter a specific destination using drop-down box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and dialing numbers can be entered manually. Once the incoming call is matched, the call is passed to that destination.
-Fallback Extension column - Should retain default value, blank.
Defines an alternate destination which should be used when the current destination, set in the **Destination**, is unreachable.



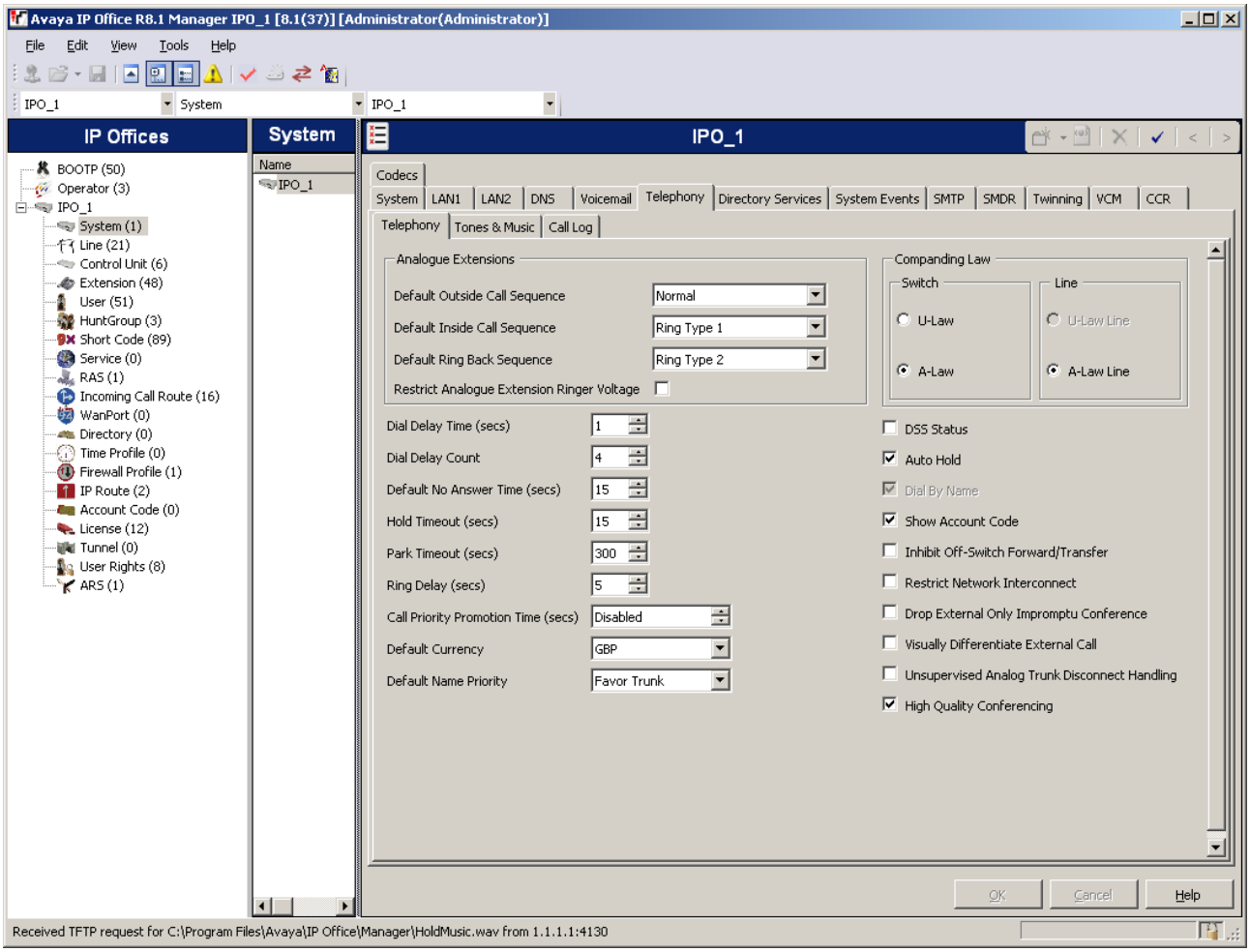
2.1.4 IP Office Short Code

The system uses short codes to access the SIP trunk for outgoing calls.
From the configuration tree in the left pane, select **Short Code,** right-click and select **New** to add a new short code.
Introduce the following:
- Code - the dialing digits used to trigger the short code.
- Feature – Select **Dial** action to be performed by the short code.
- Telephone Number – The number dialed by the short code. The digits sent to SIP Trunk when calling.
- Line Group ID – The **Outgoing Group ID** configured for the SIP line under the URI tab.
- Locale – The default value should be retained, blank.
- Force Account Code – The default value should be retained, Off.
If enabled, the user is prompted to enter a valid account code before the call is allowed to continue.

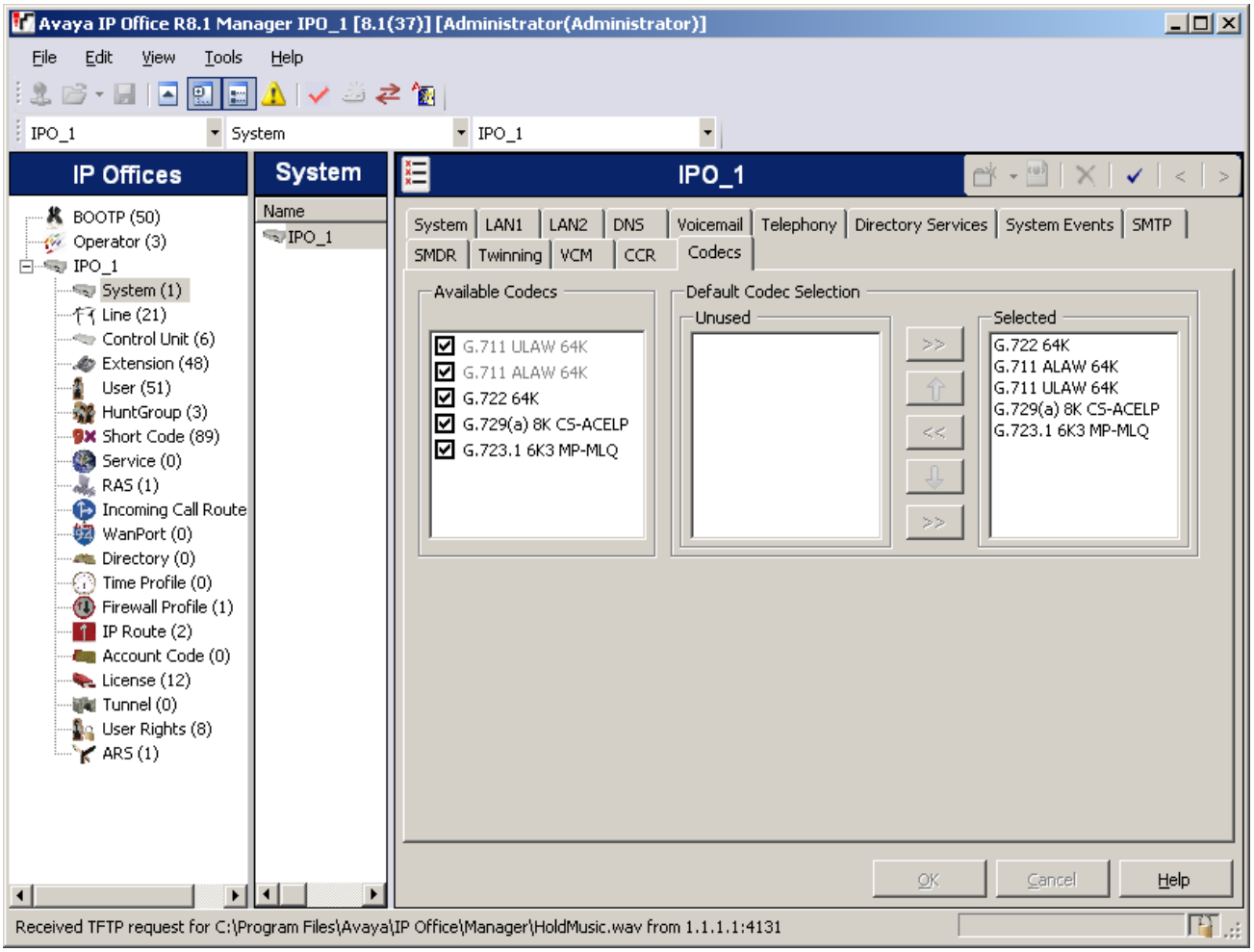


2.1.5 IP Office System Settings

In the **Telephony** tab:
- **Companding Law** - Choose the appropriate Companding Law depending on the region (U-Law for North America and Japan, A-Law for elsewhere).
- For all other fields, the default values should be retained.
- When **Inhibit Off-Switch Forward/Transfer** is enabled, it stops any user from transferring or forwarding calls externally. Default is Off.



In the **Codecs** tab:
- This tab is used to set the codecs available for use with all IP lines and extensions, and specifies the default order of codec preference.
- **Available Codecs** list - This list shows the codecs supported by the system and those selected as usable. Those codecs selected in this list are then available for use in other codec lists shown in the configuration settings.
- **Selected** list - specifies the codec preferences.



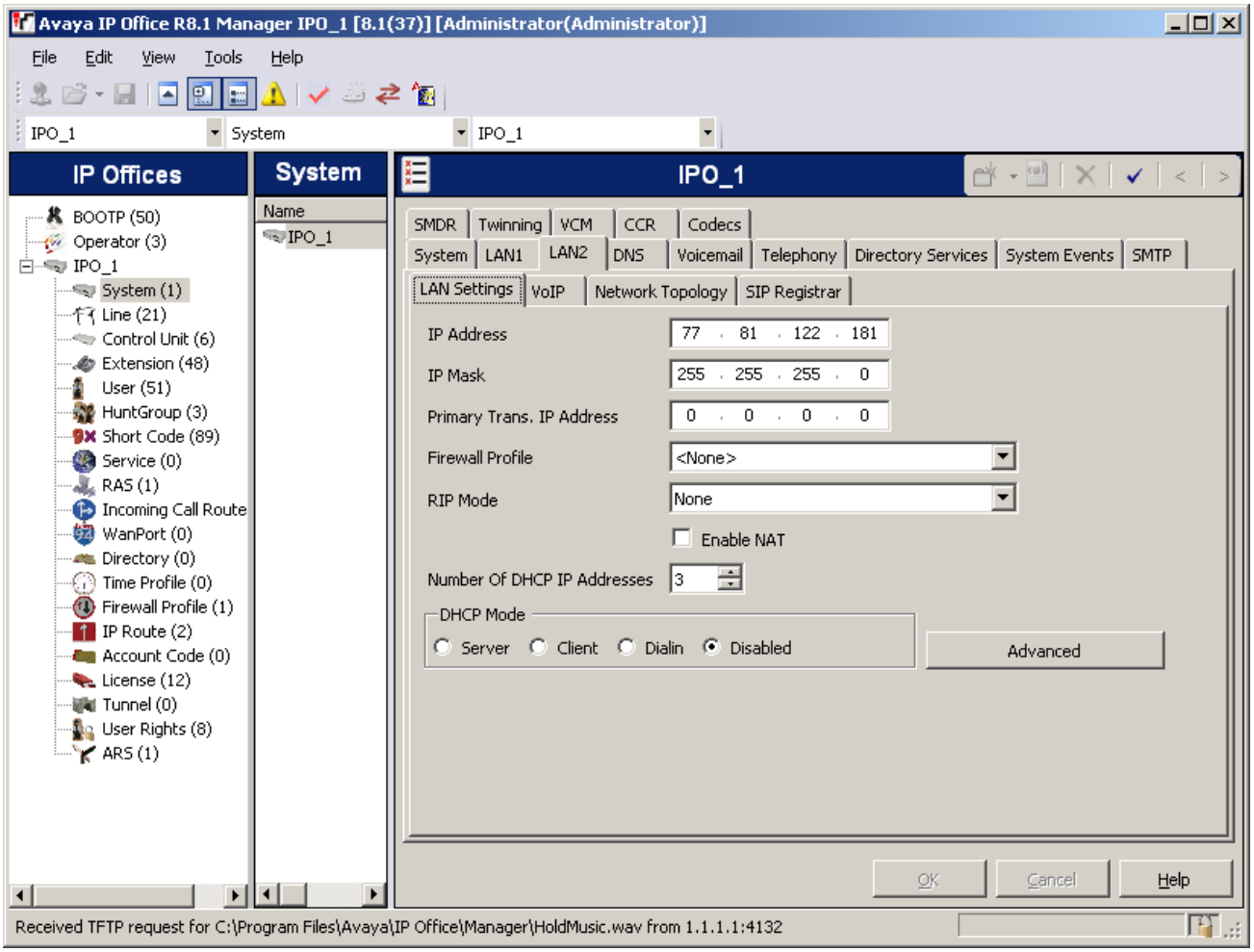
In the **LAN** tab:
Usually LAN1 is used for the local network and LAN2 is used to connect to the BCM, via WAN. LAN2 settings are presented bellow.

- LAN Settings tab:

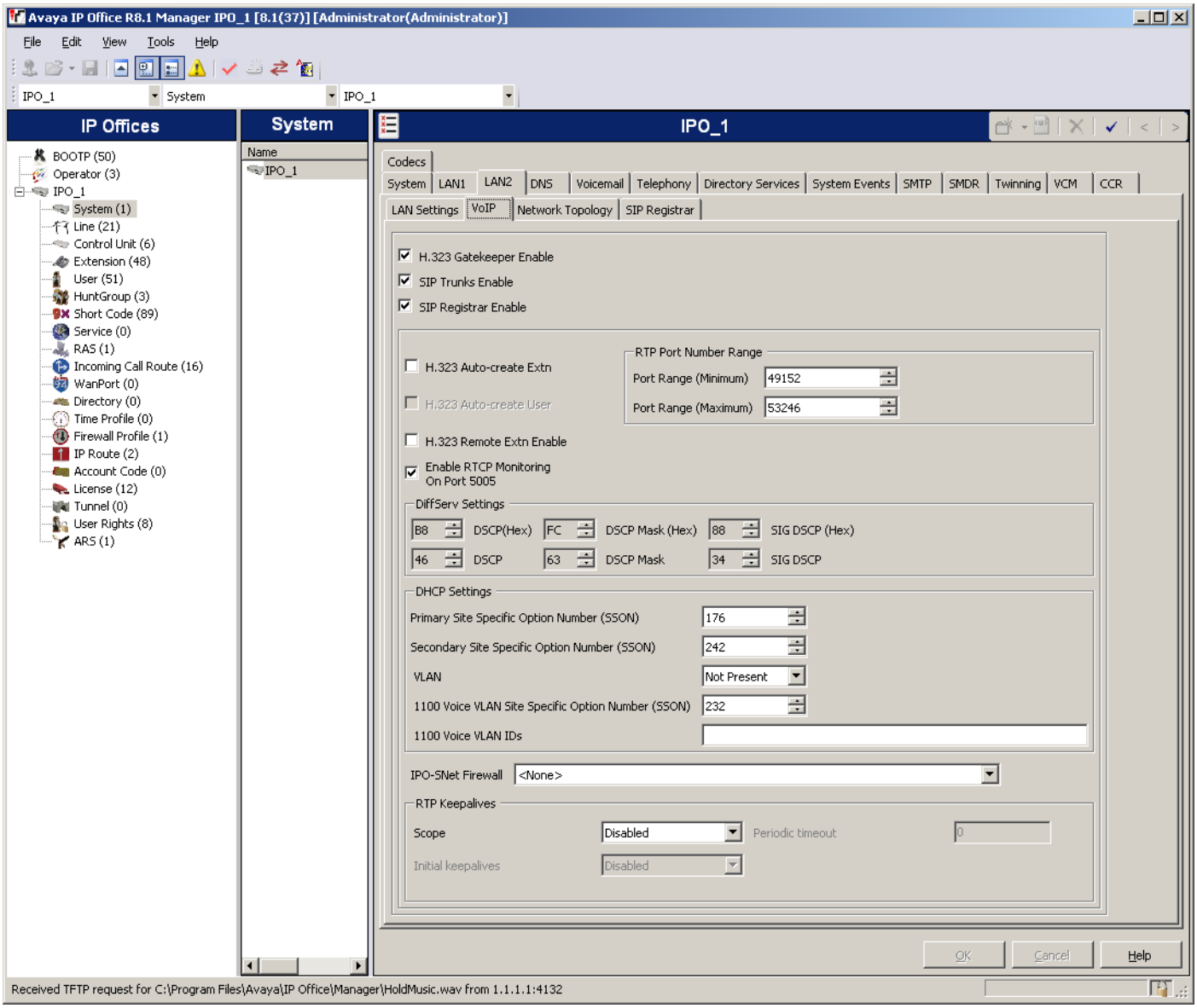
Enter the appropriate IP Address and IP Mask in the corresponding fields.

Retain Default values for all other fields.

If a DHCP client or server is required this can be activated using the DHCP section.



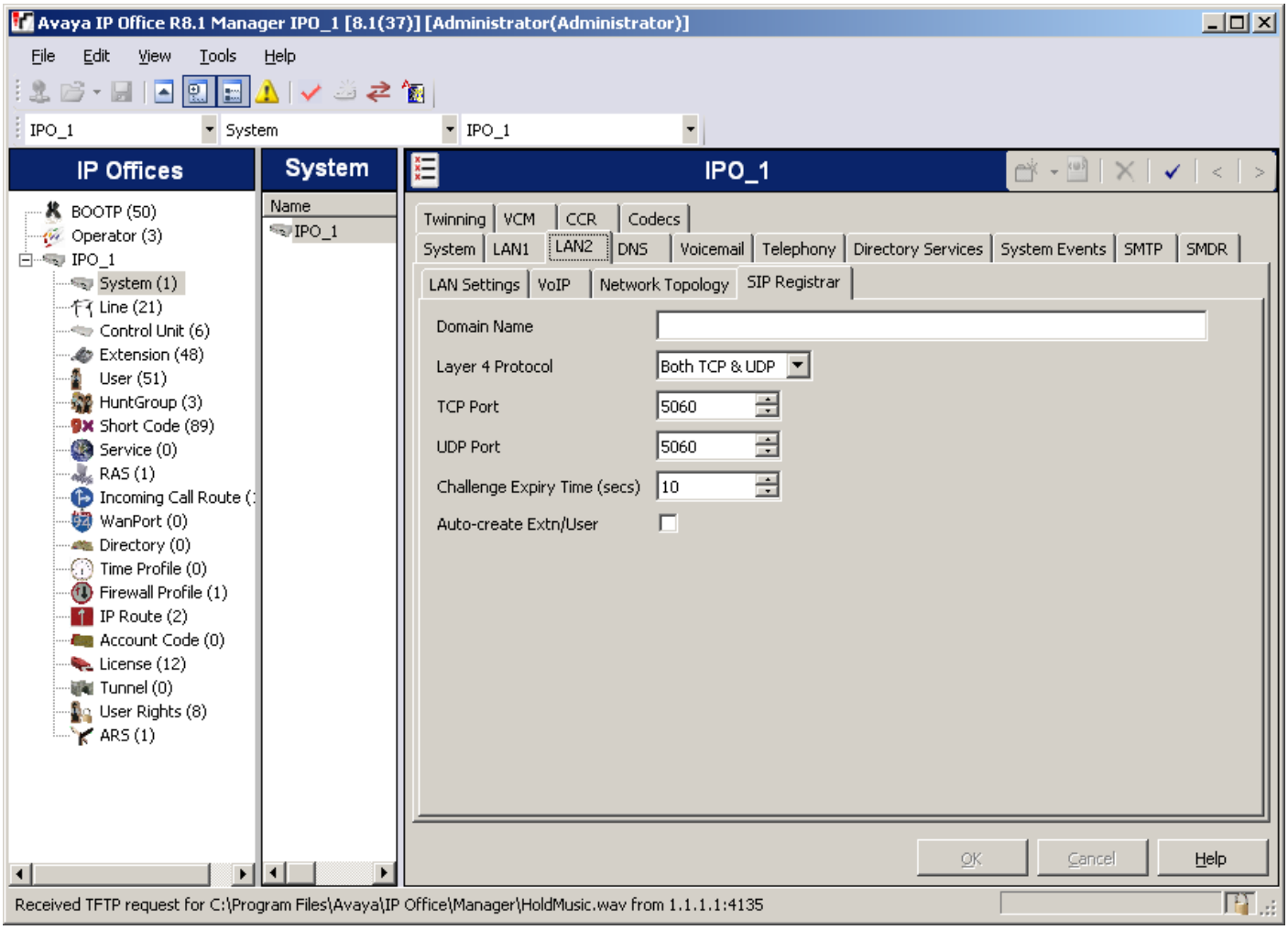
VoIP tab:
- **Default** values should be retained for all fields.
- Ensure that **SIP Trunks Enable** is checked.



SIP Registrar tab:

This tab is used to set the system parameters for the system acting as a SIP Registrar to which SIP endpoint devices can register.

Default values should be retained for all fields.



2.1.6 IP Office Small Community Networking (SCN)

Systems linked by H.323 IP trunks can enable voice networking across those trunks to form a multi-site network. Within a multi-site network, the separate systems automatically learn each other's extension numbers and user names. This allows calls between systems and support for a range of internal call features.

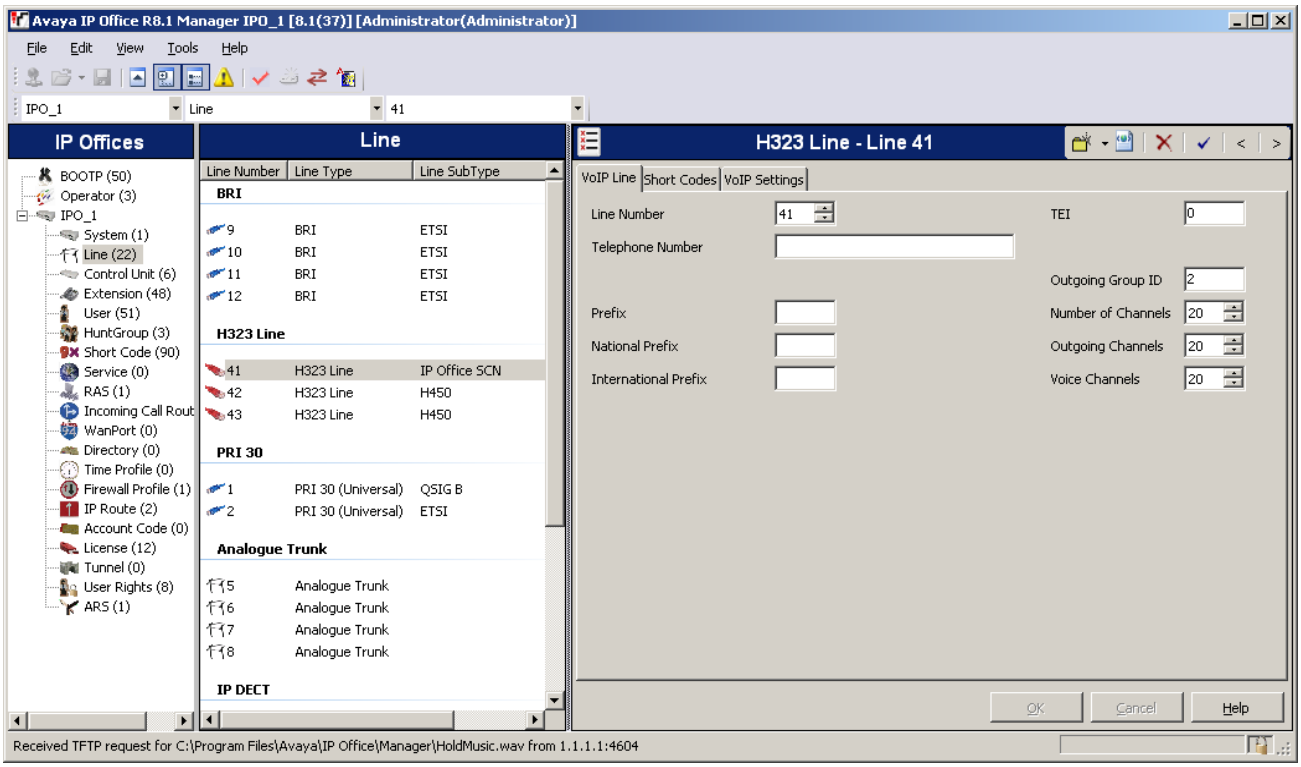
To set up a Small Community Network, the following are required:

- A working H.323 trunk between the systems that has been tested for correct voice and data traffic routing.
- Within a particular network, all SCN trunks should be on the same LAN interface.
- VCM channels are required in all systems.
- The extension, user and group numberings on each system must be unique.
- The user and group names on each system must be unique.
- The Outgoing Group ID on the Small Community Network lines should be changed to a number other than the default , 0 (zero).

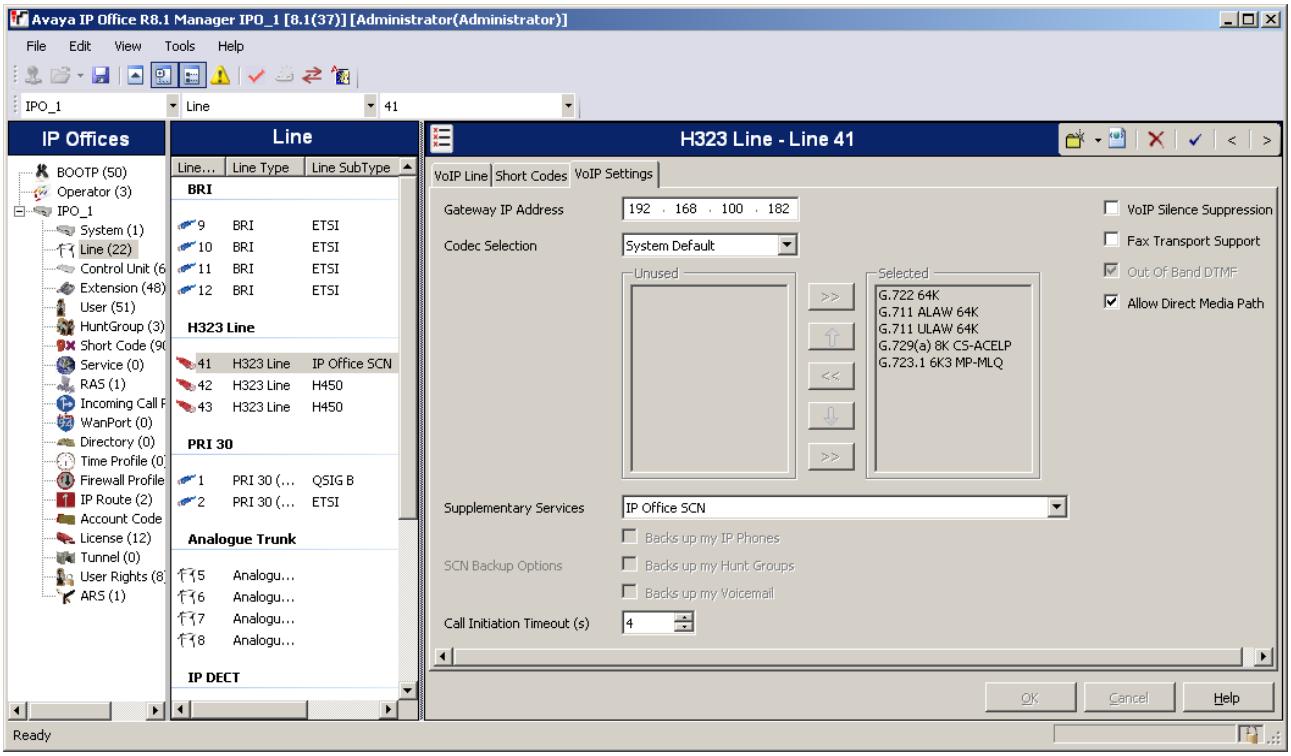
- All systems should use the same set of telephony timers, especially the **Default No Answer Time**.
- Only one system should have its **Voicemail Type** set to **Voicemail Pro/Lite**. All other systems must be set to either **Centralized Voicemail** or **Distributed Voicemail**. No other settings are supported.

From the configuration tree in the left pane, right-click on **Line** and select **New > H323 Line** to add a new H323 Trunk.

In the **VoIP Line** tab:
- Adjust the **Line Number** as you desire, this must be unique.
- Change the default value for Outgoing Group ID to a different number.
- Default values should be retained for all other fields.
- The **Number of Channels**, **Outgoing Channels** and **Voice Channels** can be adjusted depending upon system resources and setup needs.



In the **VoIP Settings** tab:
- **Gateway IP Address -** enter the IP address of the remote IP Office. This address must not be used by any other IP line.
- Retain default settings for all other fields.
- Ensure that **Supplementary Service**s is set to **IP Office SCN**.

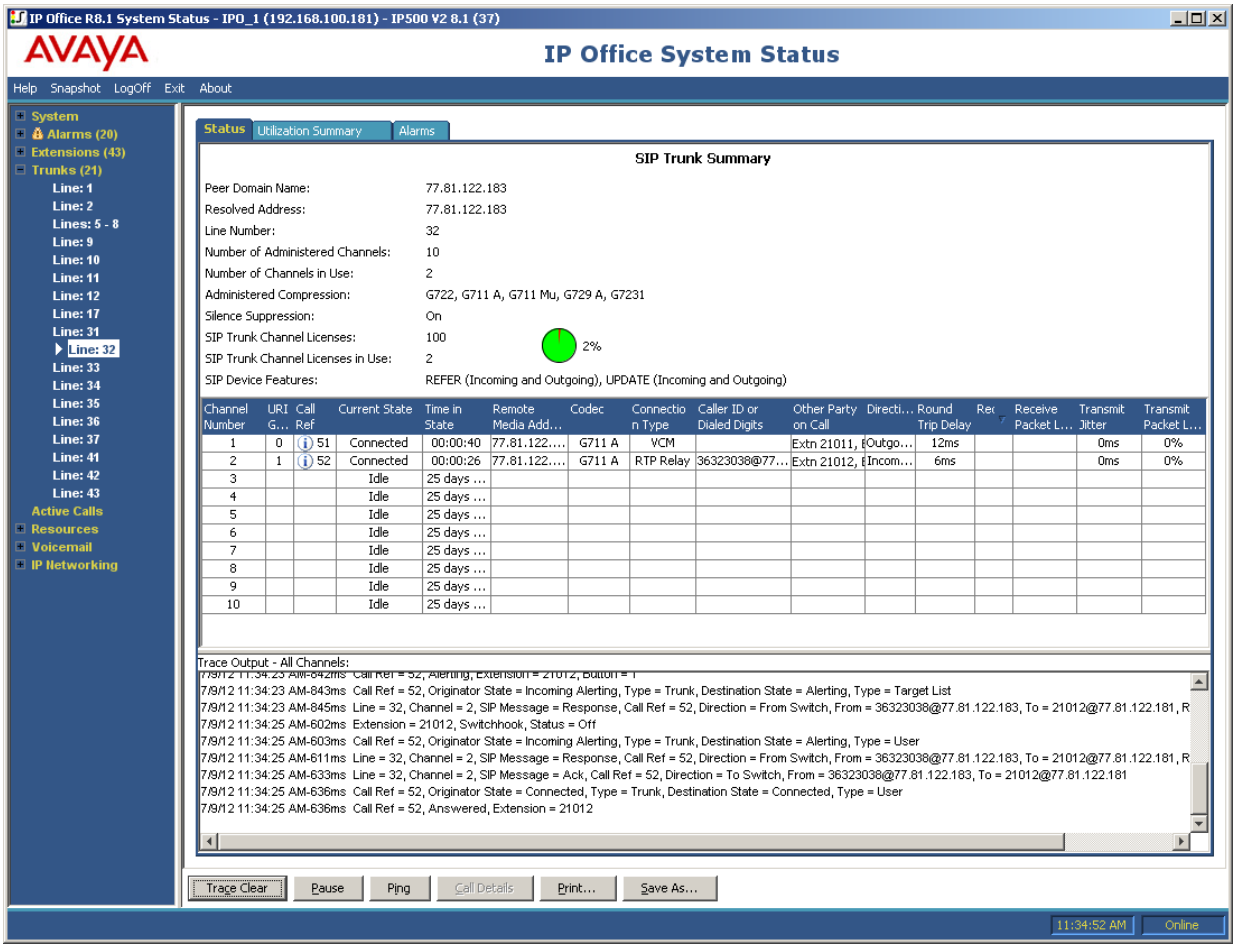


After all IP Office configuration changes are made, save the configurations and send them to the corresponding IP Office unit. Please note that a system reboot will be required for these changes to take effect.

2.1.7 Verify Basic Connectivity

The Avaya IP Office System Status application can be used to check the status of the created trunks.
- To check the trunk state, from the explore tree, go to **Trunks** and click on the configured **Line.**
- The **Status** page for that line is expanded in the right pane.
- Check that the Channels are either **Idle** or in an active state.

The Avaya IP Office System Monitor can be used to debug and track calls.
These applications are installed together via the Admin CD.



2.2 Business Communication Manager Configuration

This section provides the procedures for configuring BCM using the Business Element Manager application.

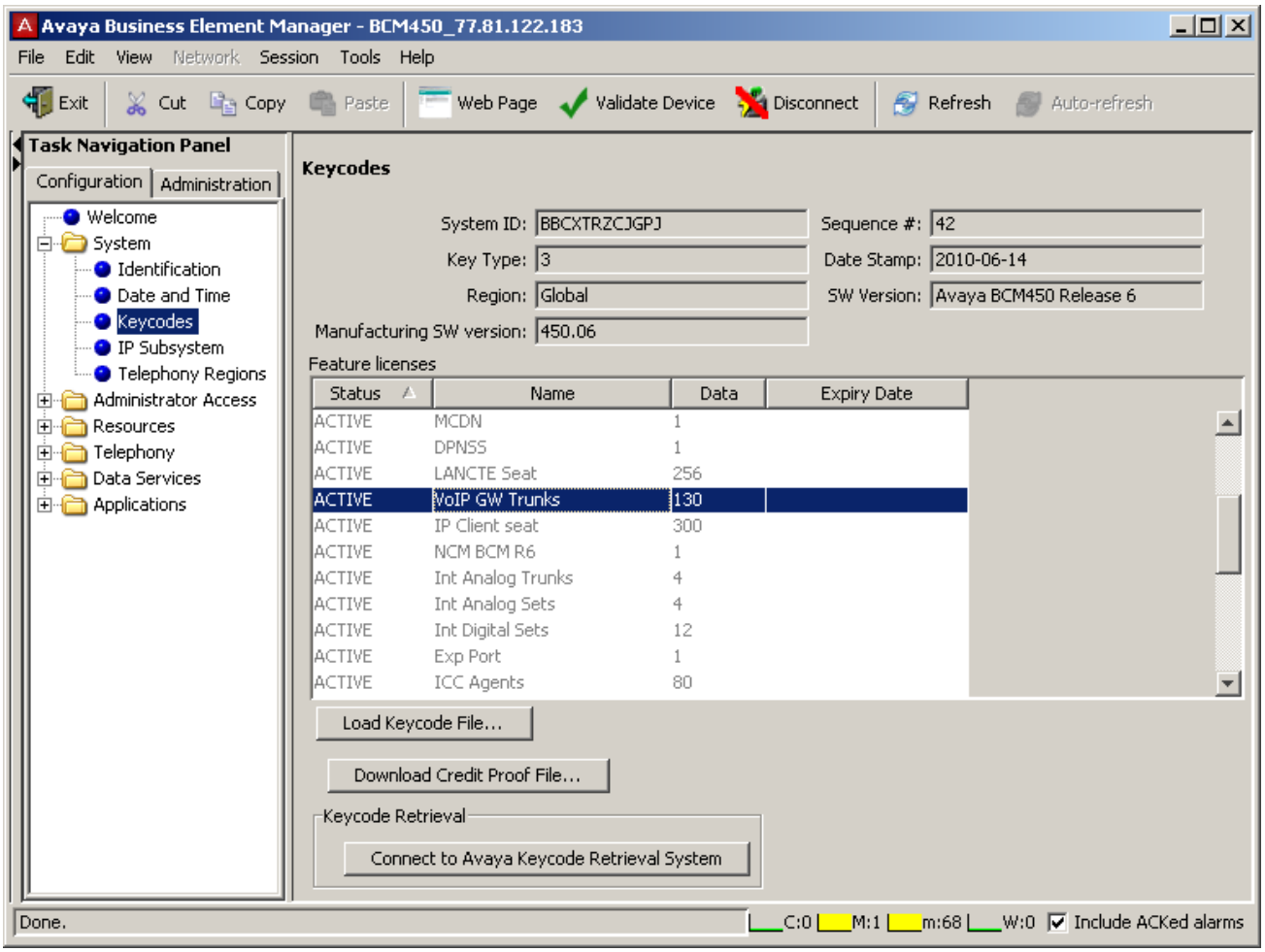
The procedure covers the following areas:

- BCM Keycodes
- BCM Business Name
- IP Trunk
- SIP Trunking
- Dialing Plan
- MCDN Line

2.2.1 BCM Keycodes

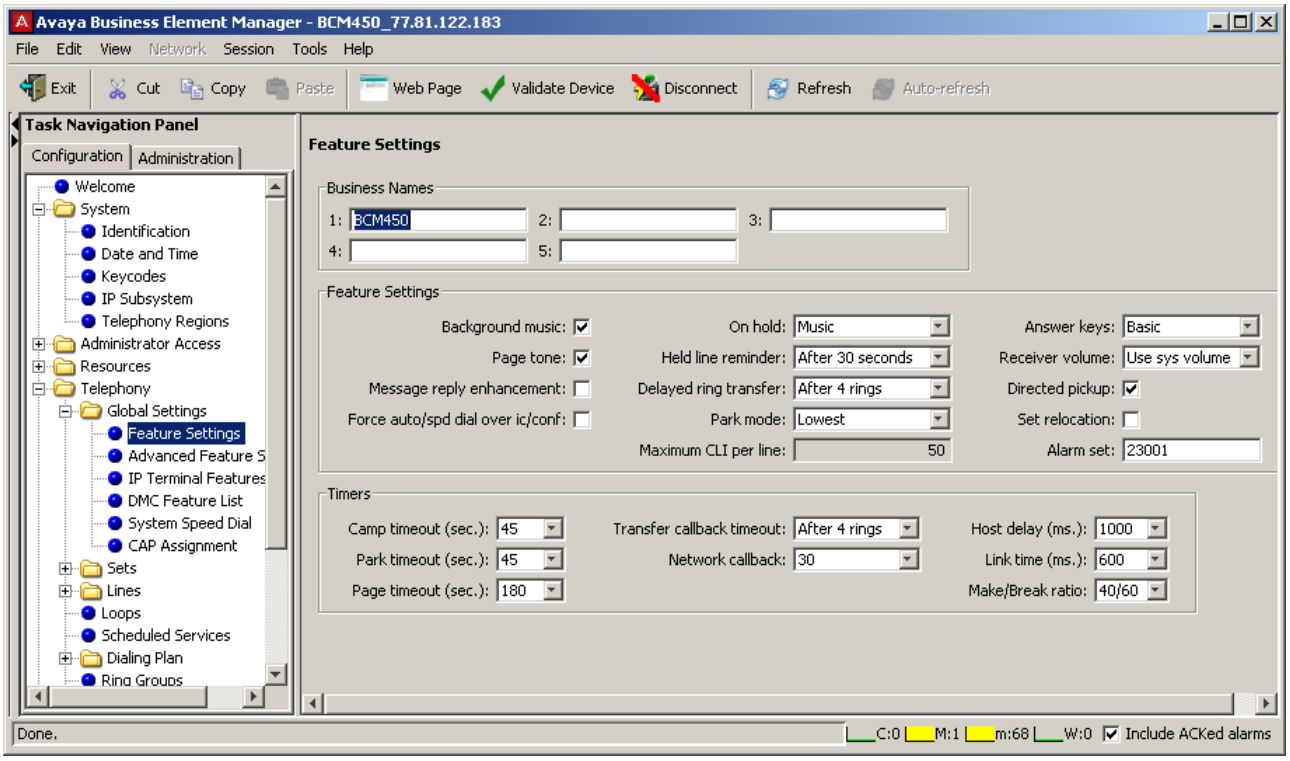
A ‘VoIP GW Trunks’ license is required in order to activate IP trunks (SIP and H323) on BCM.

There is, alternatively, a 'SIP GW Trunks' keycode which can be used to enable only SIP trunks. To check the available keycodes, login to the BCM system using the Business Element Manager, and under the Configuration tab go to System > Keycodes.



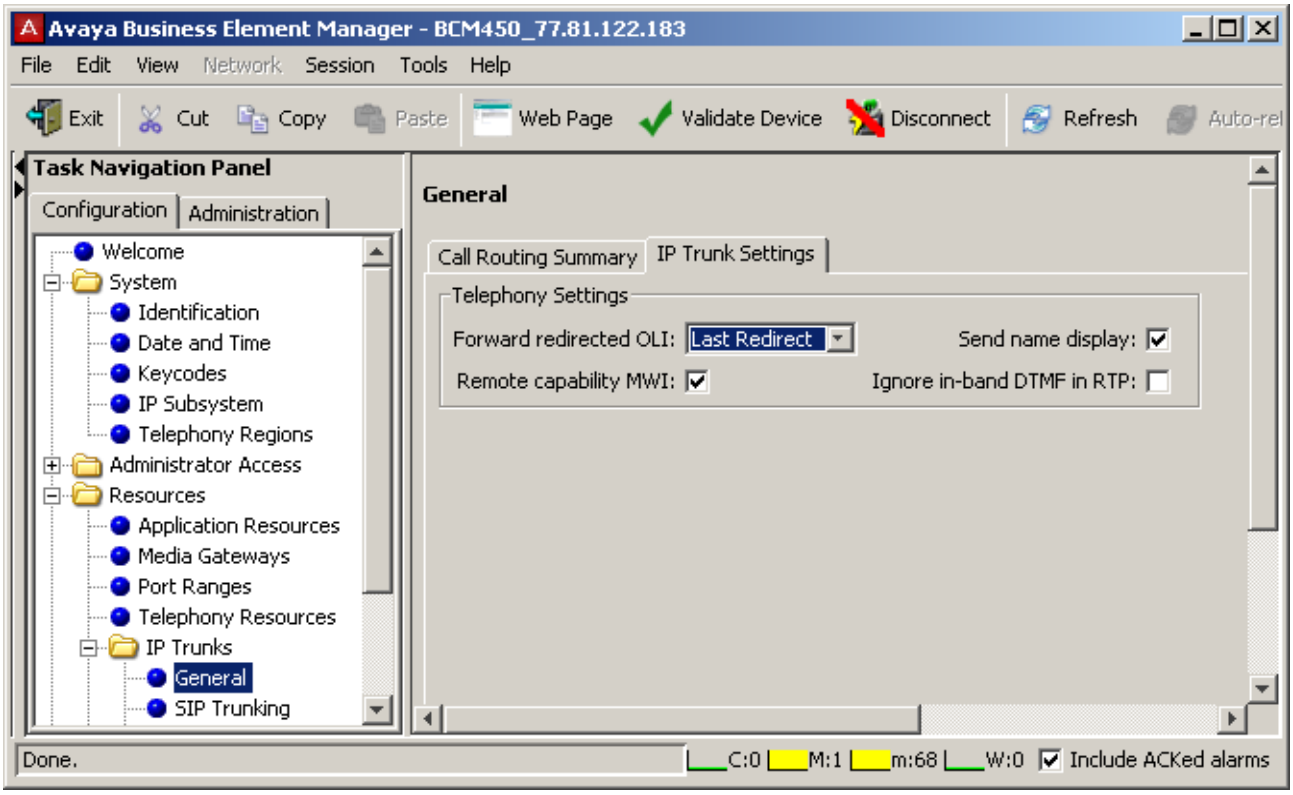
2.2.2 BCM Business Name

Under the Configuration tab, go to **Telephony > Global Settings > Feature Settings,** and enter a Business Name. This allows the BCM to send the CLID. Network Name Display uses the configured Business Name as the prefix and, when applicable, the telephone or hunt group Name. No space is automatically inserted between the combined Business Name and telephone or hunt group name when the Network Name Display is composed. Avaya recommends that you use a space as the last character of the relevant Business Name or Business Names.



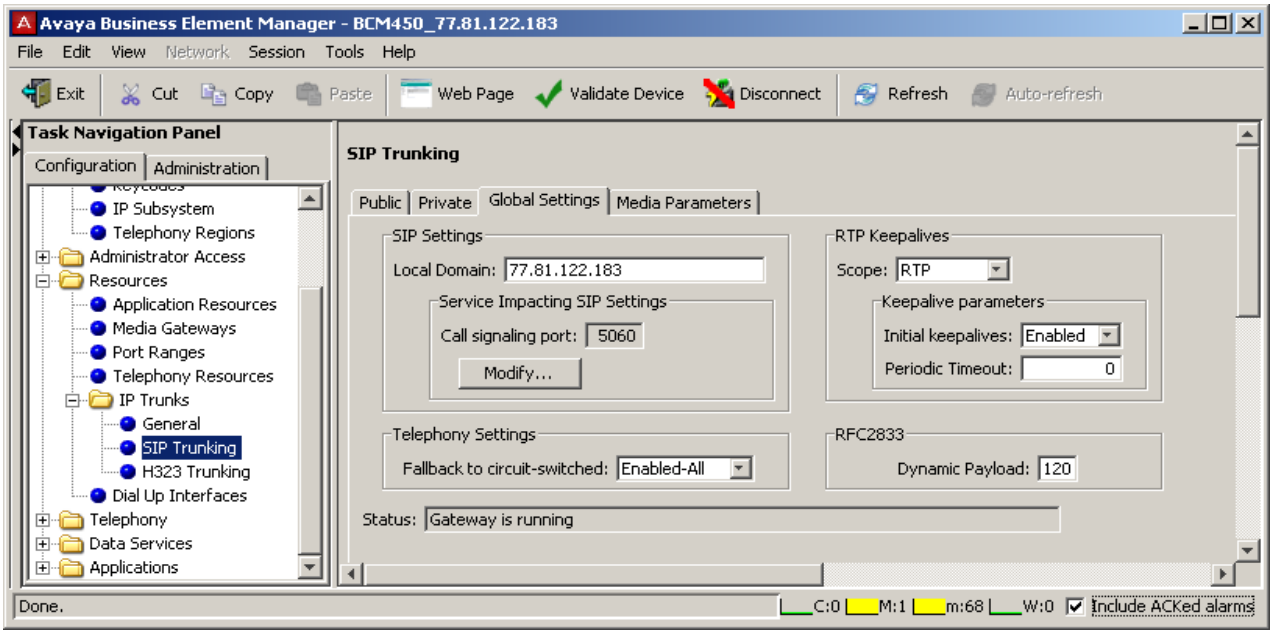
2.2.3 BCM IP Trunks

To administer IP Trunks go to **Resources > IP Trunks**.
Under the **General > IP Trunk Settings** tab:
- Forward Redirect OLI - Should be set to **Last Redirect** or **First Redirect** depending on the expected treatment from voicemail.
- Remote capability MWI - Should be enabled. It indicates that Message Waiting indications will be sent across the SIP trunk if there is a message for a set on the remote switch, though centralized voicemail is not supported between IP Office and BCM. This setting must be coordinated with the functionality of the remote system that hosts the voice mail.
- Send name display - Should be enabled. When enabled, the system sends the telephone name with outgoing calls to the network.
- Ignore in-band DTMF in RTP – Should be disabled. When enabled, the BCM ignores audible in-band DTMF tones received over VoIP trunks after the BCM connects to the remote end of a locally hosted call center or Call Pilot application.



2.2.4 BCM SIP Trunking

In the **Global Settings** tab:
- Local Domain - Enter the local domain of the SIP network, IP address of the BCM, or leave it blank.
- Retain the default values for all other fields.

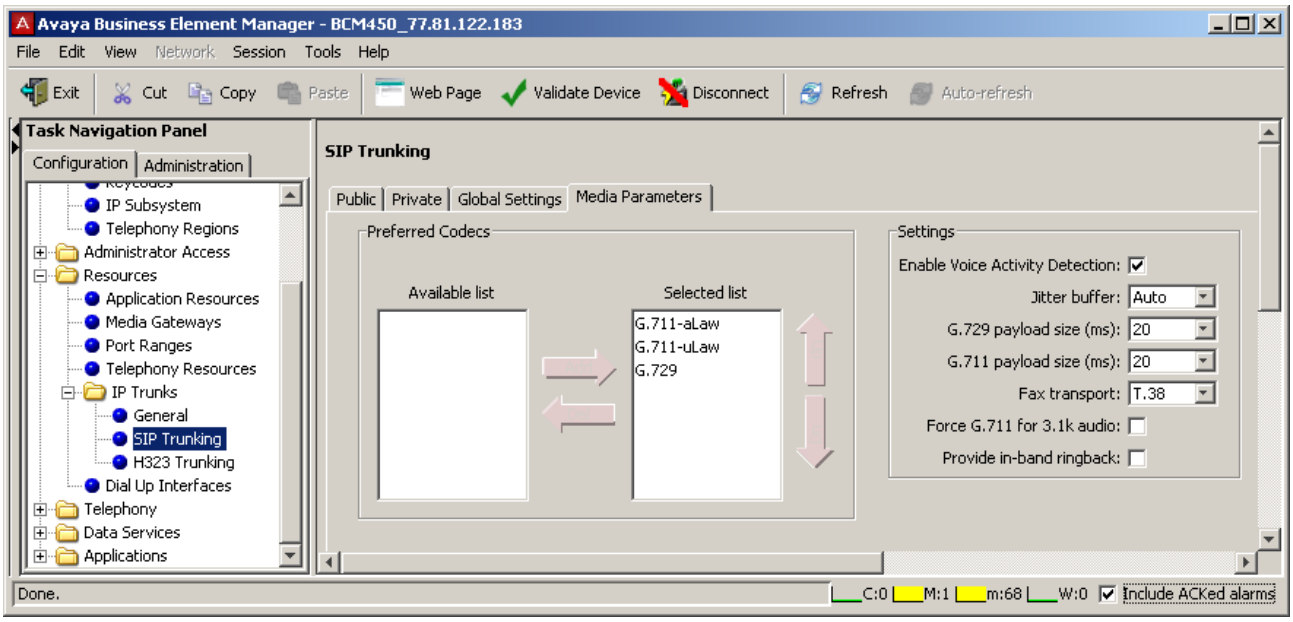


The SIP Media Parameters tab allows you to specify the order in which the SIP trunk selects IP telephony system controls for codecs, jitter buffers, silence suppression, and payload size. Media parameters are common to both public and private SIP trunks.

- **Preferred Codecs -** Select the codecs in the order in which you want the system to attempt to use them. This should be coordinated with IP Office.
- **Enable Voice Activity Detection** (Silence Suppression) - Should be coordinated with IP Office.
- **Provide in-band ringback -** Ensure it is not enabled.

This setting affects in-bound SIP trunk calls. When enabled, the BCM attempts to stream ringback, tones, or announcements in-band to the caller using RTP. This setting results in in-band ringback.

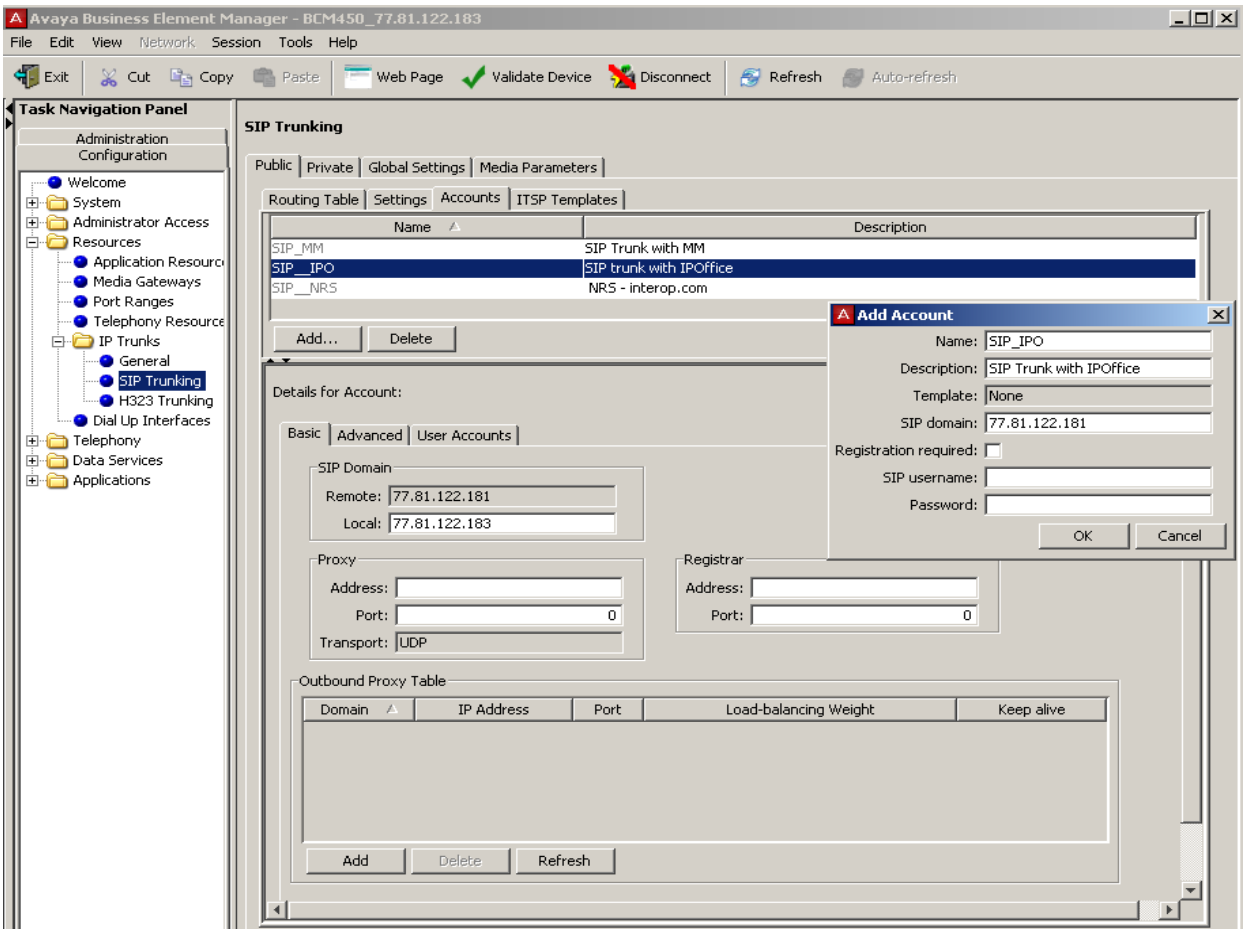
- Default values should be retained for all other fields.



2.2.5 BCM SIP Trunking - Public trunk

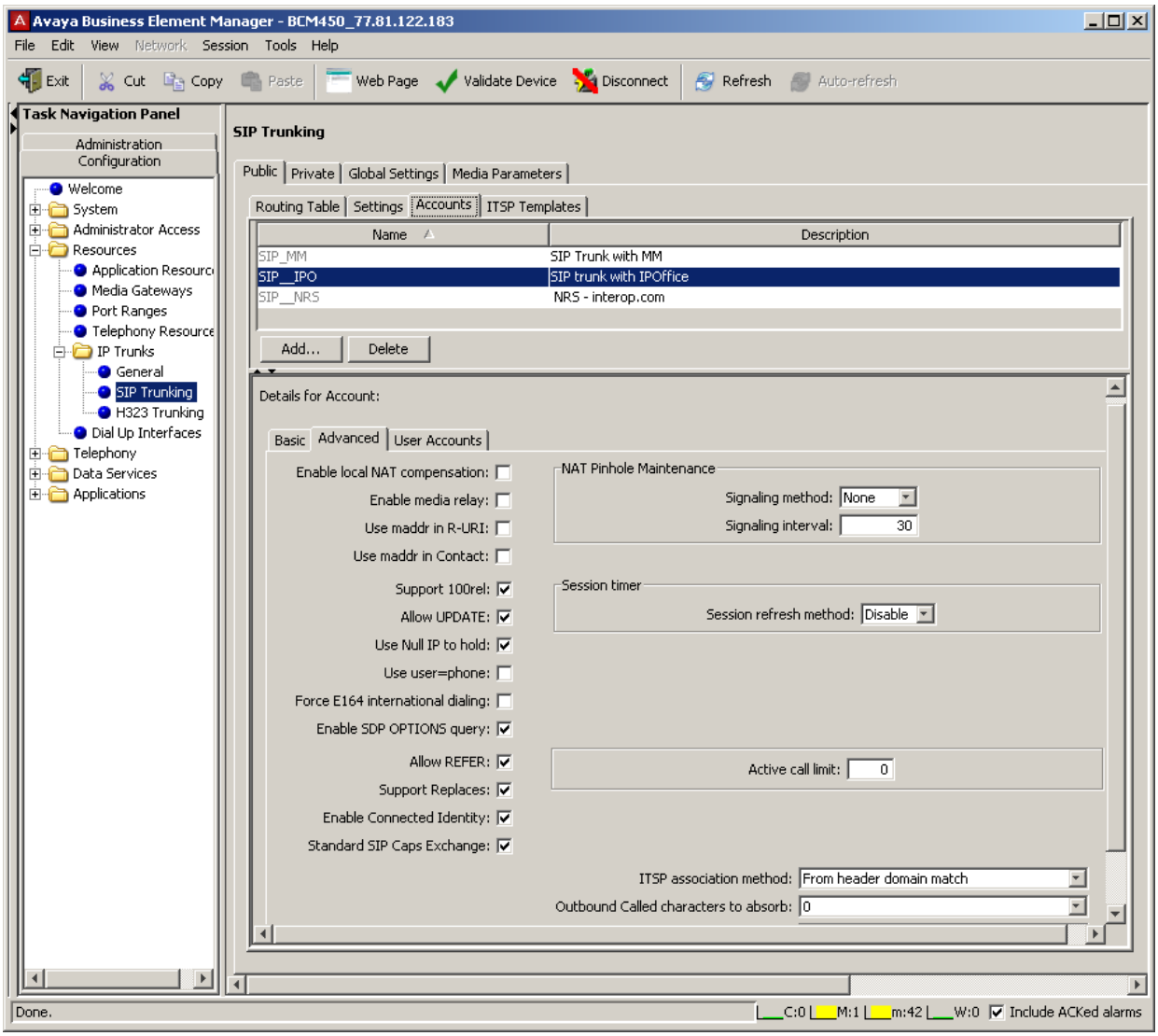
Public Trunk should be used to connect the BCM to IP Office.
Configuring a public SIP trunk on the BCM involves creating an optional ITSP template, account configuration and the configuration of a route.
You must configure an account for IP Office before you can add a route for IP Office.
To configure an account, under **Public > Accounts** tab click **Add** button:
- Name – Introduce the desired name. This will appear in accounts list for selection when defining the route.
- Description – Optional field, used to add a specific description for this account.
- SIP Domain - Remote – Can be either FQDN or an IP address. Enter the IP Address of the IP Office.
- Registration Required – Make sure it is unchecked (not applicable in our case)
- SIP username and **Password -** Leave them blank.
Press **OK** to complete and close the **Add Account** dialog box.
These parameters will be displayed in the **Basic** tab of this account.
In the **Basic** tab:
- SIP Domain Local – Introduce the IP Address of the local domain (BCM IP address) or leave it blank.
- Proxy – Address – Leave this field blank or enter the IP Address of the IP Office in case that IP Office domain name was entered into SIP Domain field.
- Retain default values for all other fields.

Please note that the **Transport** field is grayed out with **UDP**.
This means that BCM doesn't support SIP over TCP.



In the **Accounts – Advanced** tab:
- **Enable Connected Identity** – Should be enabled. It enables delivery of connected identity across the trunk.
- **Enable SDP Options query** – Should be enabled. If disabled, speech path issues can be encountered when transferring calls.
- **Standard SIP caps exchange –** Should be enabled.
- Retain default values for all other fields.

Make sure that the following parameters are enabled:
- **Support 100rel** - Flag indicating whether the BCM advertises support for 100Rel (PRACK) in the Supported header.
- **Allow Update** - Indicates whether the BCM advertises support for UPDATE in the Allow Header.
- **Use Null IP to hold** - Determines whether the BCM uses Null IP address (0.0.0.0) when putting a call on hold. If it is set to true, 0.0.0.0 is used when putting a call on hold. Otherwise, a valid IP address as per RFC3264 is used.
- **Allow Refer** - Enables support for the REFER method being advertised in the Allow header.
- **Support Replaces** - Enables support for the Replaces header being advertised in the Supported header.



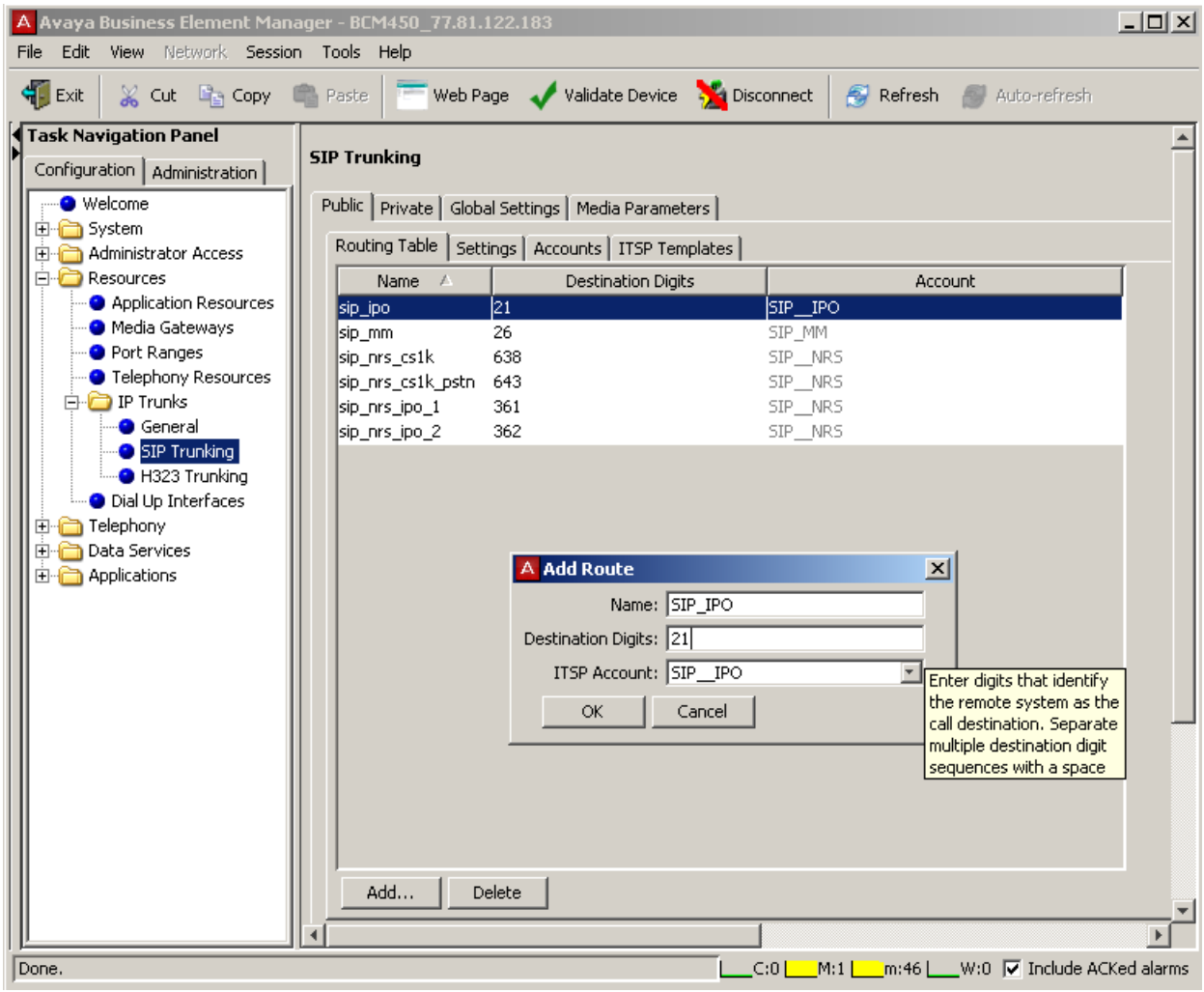
In the **SIP Trunking > Public > Routing Table** tab, click the Add button to add a public route. Enter the required information in the Add Route dialog box:

- Name – The name for the route.

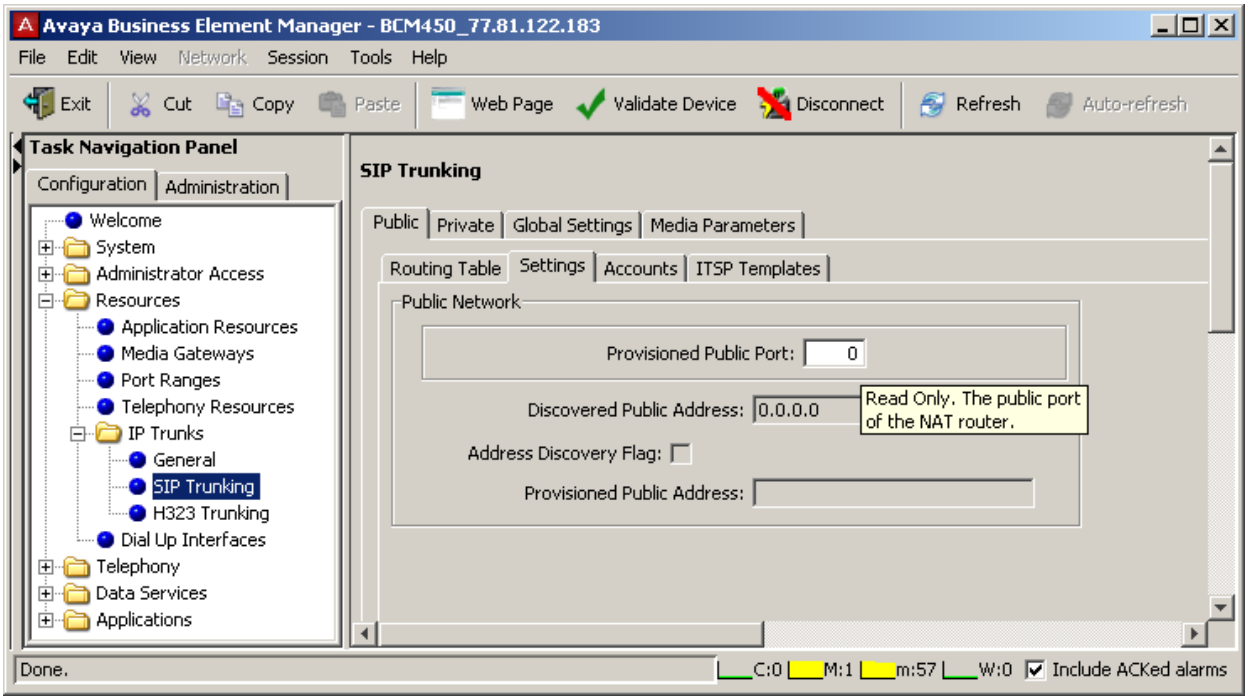
- Destination Digits – The leading digits which callers can dial to route the calls through the remote gateway. It should be the same as the destination code defined on the **Telephony > Dialing Plan >Routing > Destination Codes** tab. Ensure that there are no other remote gateways currently using this combination of destination digits. If multiple leading digits map to the same remote gateway, separate them with a space.

- ITSP Account – Select from the drop-down list the ITSP account created in the previous step (SIP_IPO).

- Click OK to complete and close the Add Route dialog box.



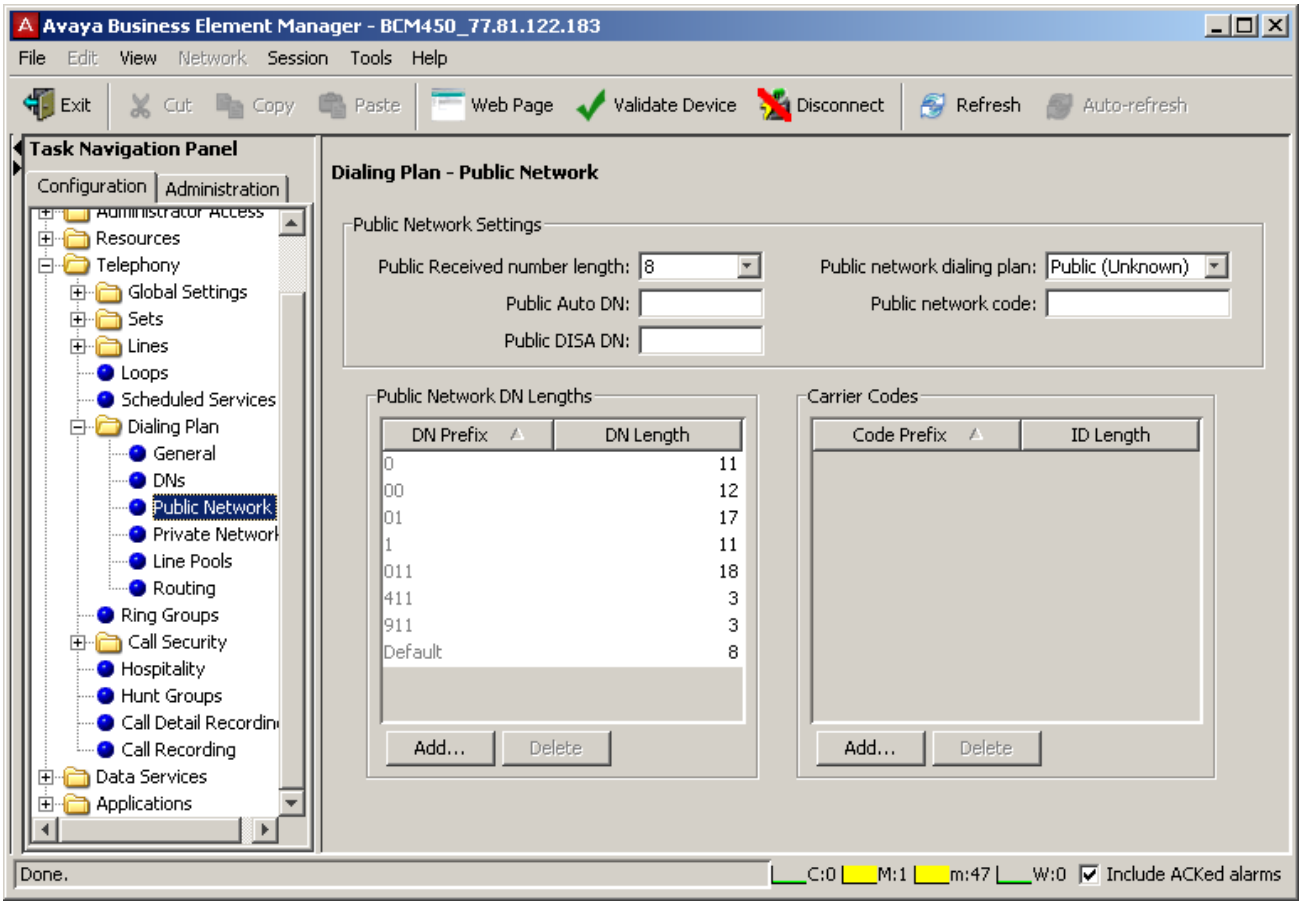
In **SIP Trunking – Public – Settings** tab
- Retain the default values for all fields.



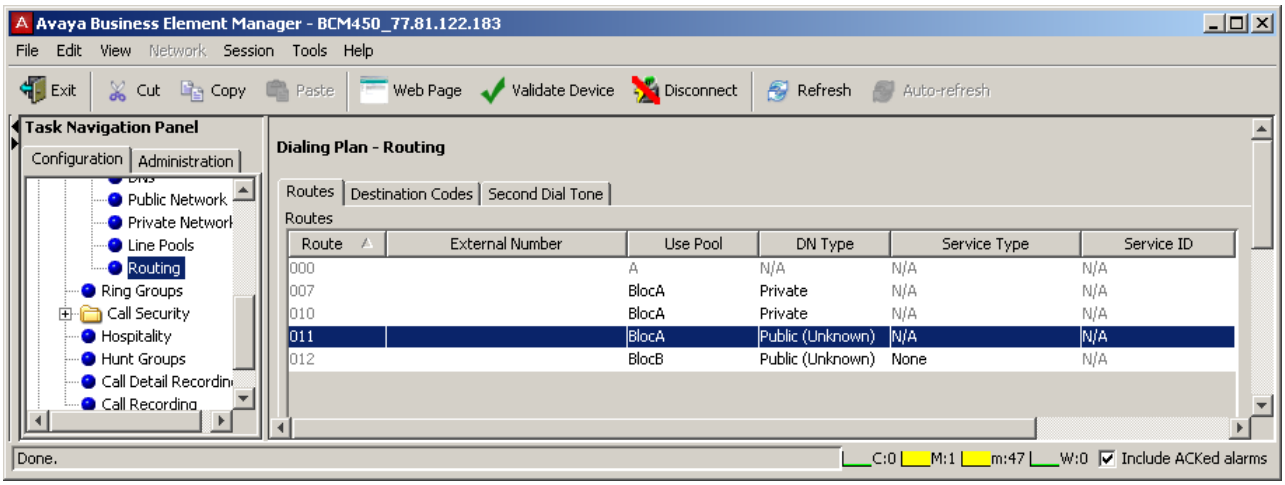
2.2.6 Telephony - Dialing Plan

In the **Public Network** section:
This section provides the settings that allow the system to determine if an incoming call is meant for the local system and how many digits the system needs to receive before sending the dial string over the trunk.
- **Public Received number length** – represents the maximum number of digits that the system uses to determine if an incoming call tagged as public fits the system public DN numbering. This should be adjusted according to the numbering plan.
- **Public Network DN Lengths** – Should be set to Public (Unknown) for a variable length.
- **Public network code** – The number entered here is concatenated with the Public OLI. It can be set according to the numbering plan or left blank.
- **Public Network DN Lengths** – Entries should be added to this list based on the Destination Code and the number of digits dialed within the interop nodes. The Public network DN length tells the system how long dialing strings are when entering the network. If the values for Public Network DN length are set too short, digits are stripped from the dialing string. Conversely, if the values are set too large, the dialing takes longer to process.

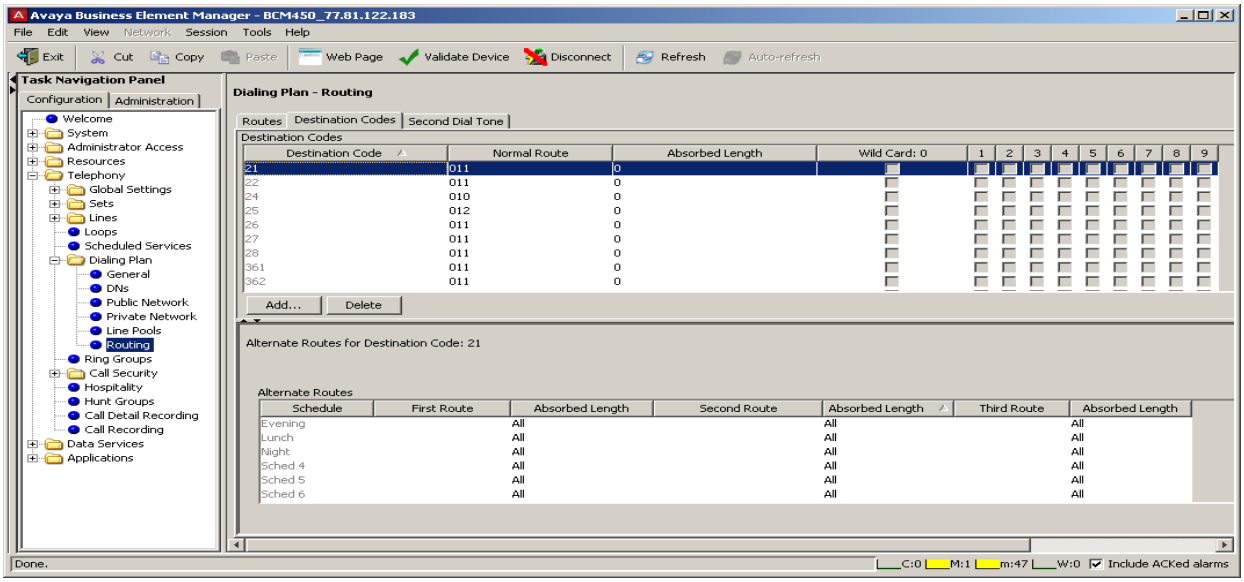
DN Prefix – This is the number (destination code) that must precede a dial string exiting the system to the public network.
DN Length - This number indicates how many numbers, starting from the front of the dial string, the system waits before sending to the public network.



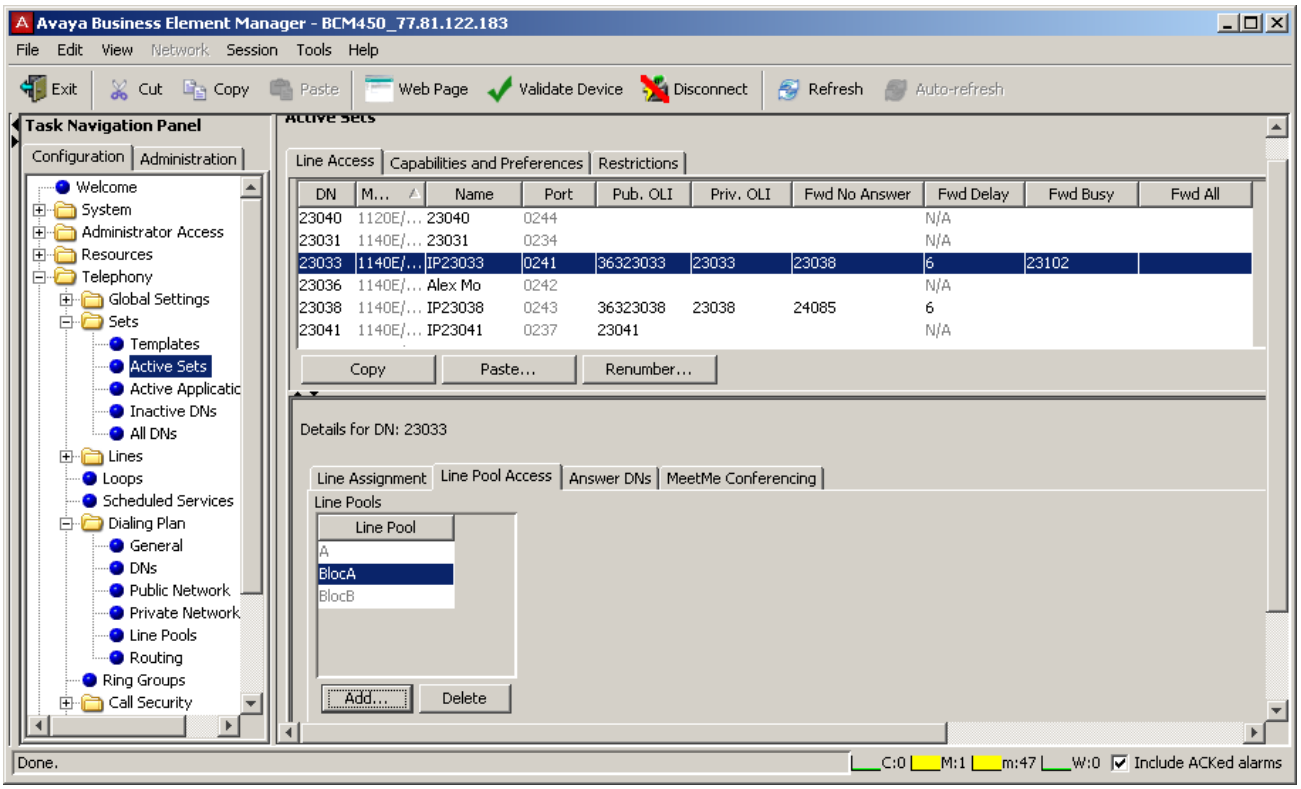
In **Telephony > Dialing Plan > Routing > Routes** tab.
This section gives the option to configure the lines and loops to allow users to dial out of the system.
A route can be used with more than one destination code, but a line pool should only be used with one route.
Click the Add button to add a new Route and fill in the required fields:
- **Route –** This is the route number, it is unique. Chose a number between 001 – 999.
- **External Number –** Is a digit or group of digits that get inserted in front of dialed digits. If all the required numbers are covered by a destination code, this box can be left blank.
- **Use pool** – Enter the Bloc pool which is assigned to VoIP trunks. This can be verified from Telephony > Lines > Active VoIP Lines.
- **DN Type** – This field should be set to **Public** (**Unknown**). This setting tells the system what type of line protocol the route uses to process the dial string.



In the **Telephony > Dialing Plan > Routing > Destination Codes** tab:
The destination code allows users to access routes. A route can be used for more than one destination code.
Click on the Add button to add a new destination code:
- Destination Code – Enter the destination code number. It can have up to 12 digits. This should be the same as the **Destination Digits** defined for Public Routes under Resources > IP Trunks > SIP Trunking > Public.
This number precedes a telephone number to tell the system where the call needs to be routed. The **A** code is a wildcard.
Click OK to close the dialog box. The new destination code appears in the destination codes list with the default parameters; these need to be modified.
- Normal Route – Enter the route number created in the previous step. This is the route that the system uses when a destination code is added to the dial string.
- Absorbed Length – This indicates how much of the destination code gets removed before the system sends the dial string to the network. Available values are **All, None, 1** through **(X-1).** This should be set according to the IP Office configuration, meaning that BCM should send the digits that IP Office is expecting to receive from BCM side in order to route the calls to desired destinations.
- Wild Card 0-9: This applies if wild card **A** is used at the end of the destination code.

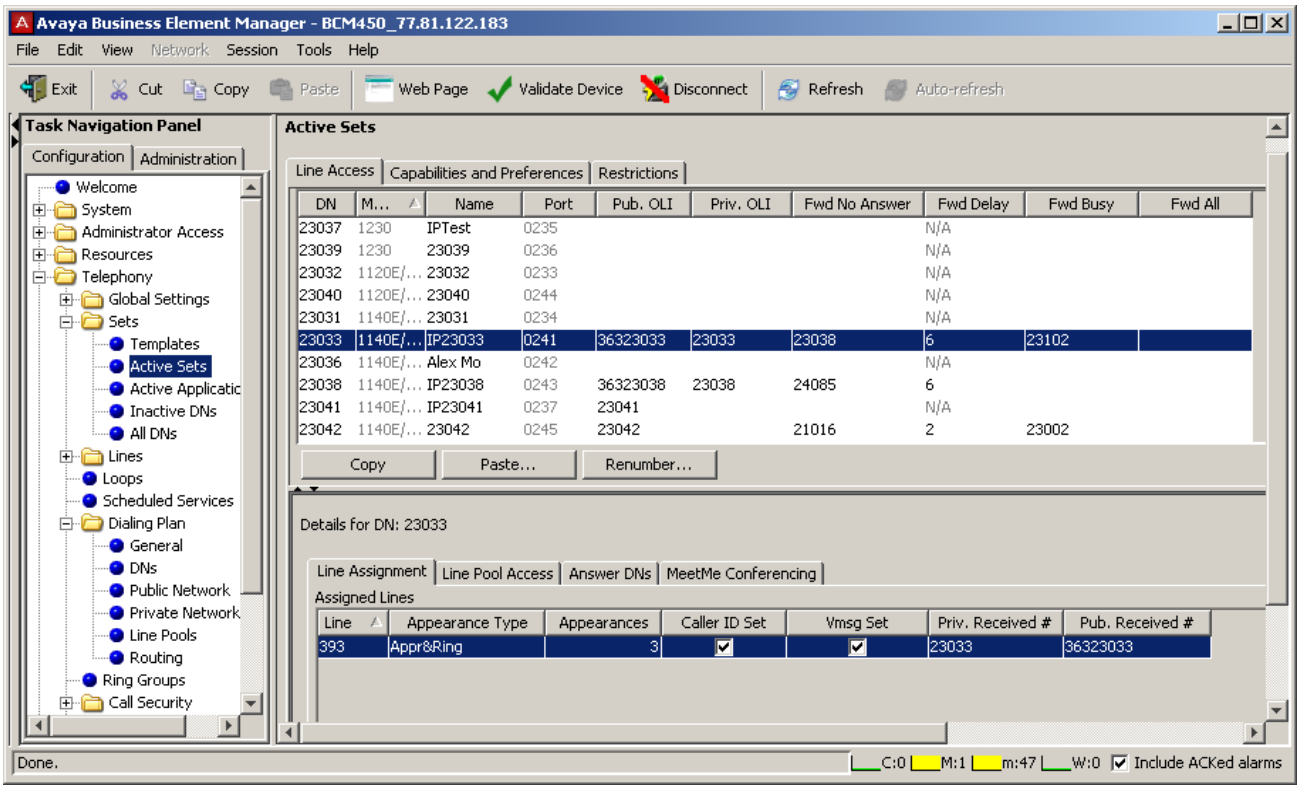


Phone configurations required to originate calls via SIP Trunk.
In the **Telephony > Sets > Active Sets** section**,** highlight a phone from the list and go to the **Line Access** tab:
- Line Pools Access – In order for a phone to be able to originate calls using VoIP lines, it must have access to the Line Pool on which VoIP lines belongs to. By Default, VoIP lines are mapped to **BlocA.** Using the **Add** button, add to the Line Pools list the appropriate Line Pool (BlocA in our case).
This mapping can be verified or configured from Telephony > Lines > Active VoIP Lines.
- Pub OLI (Originating Line Identification**) –** Is used to provide the CLID for Outgoing calls over public networks. If the calling telephone has no appropriate OLI configured, then no outgoing CLID number (or name) is sent. The maximum number of digits for this field is coordinated with configurations made under Telephony > Dial Plan > Public Network.
- Priv OLI – In the same manner, Priv OLI is used to construct the CLID for Outgoing Calls over private networks.



Phone configurations required to receive calls via SIP Trunk.
In the **Telephony > Sets > Active Sets** section**,** highlight a phone from the list and go to the **Line Access** tab:
- Line Assignment: Using the Add button, assign a **Target** line to the selected phone in order for the BCM system to be able to route incoming calls to this phone. The entire list of Target lines can be found in the Telephony > Lines > Target Lines section.
Target lines are virtual communication paths between trunks and telephones on the BCM system. They are incoming lines only, and cannot be selected for outgoing calls or networking applications. With target lines, you can concentrate incoming calls on fewer trunks. This type of concentration is an advantage of DID lines. Avaya BCM target lines allow you to direct each DID number to one or more telephones.
After the line is assigned, we have to configure its parameters:
- Appearance type – Appr&Ring should be selected. In this case the line is displayed on the phone and it rings when a call is presented.
- Appearances - Target lines can have more than one appearance, so that multiple calls can be accommodated.
- Caller ID Set – It should be enabled. When enabled it displays caller ID for calls coming in over the target line.
- VMsg set - When activated, an indicator on the telephone appears when a message from a remote voice-mailsystem is waiting.
- Priv. Received # - Enter the private received number (DID) that the system will recognize as the target telephone. This is usually the same as the DN.

- Pub. Received # - Enter the public received number (DID) that the system will recognize as the target telephone. If the received digits match this number then the BCM system will route the call to the corresponding target line and the call will be presented to the mapped phone.

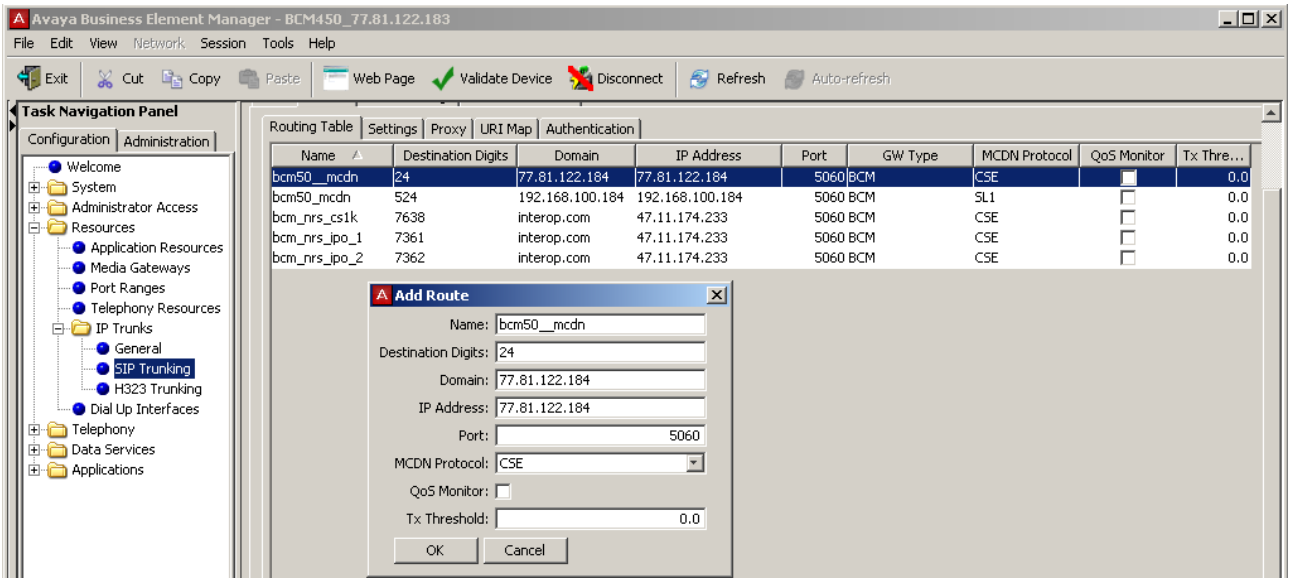


A brief description on how external calls are processed by the BCM system.
Originating calls:
- Based on the Destination Code entered, BCM gets the Route number (Telephony > Dialing Plan > Routing).
- Using the Route number, BCM decides which Pool is involved and if the calling phone has access to that Pool. (Telephony > Dialing Plan > Routing).
- Based on the mappings made between the Pool and Lines, BCM decides what kind of resources are going to be used, VoIP or Physical line (Telephony - Lines).
- If VoIP lines are involved, as in the current setup, the BCM looks at the IP Trunks Routing tables to find a match between dialed Destination Code and Destination Digits programmed for each route (Resources > IP Trunks).
- After a match occurs, BCM will fallow the Route parameters to process the call.
Receiving calls:
- When receiving an external call, BCM is looking in the Target Line list for a match between received digits and **Publ. Received #** or **Priv. Received #** fields.
- If a match occurs, BCM will route the call to the corresponding phone based on the Target Line selection.

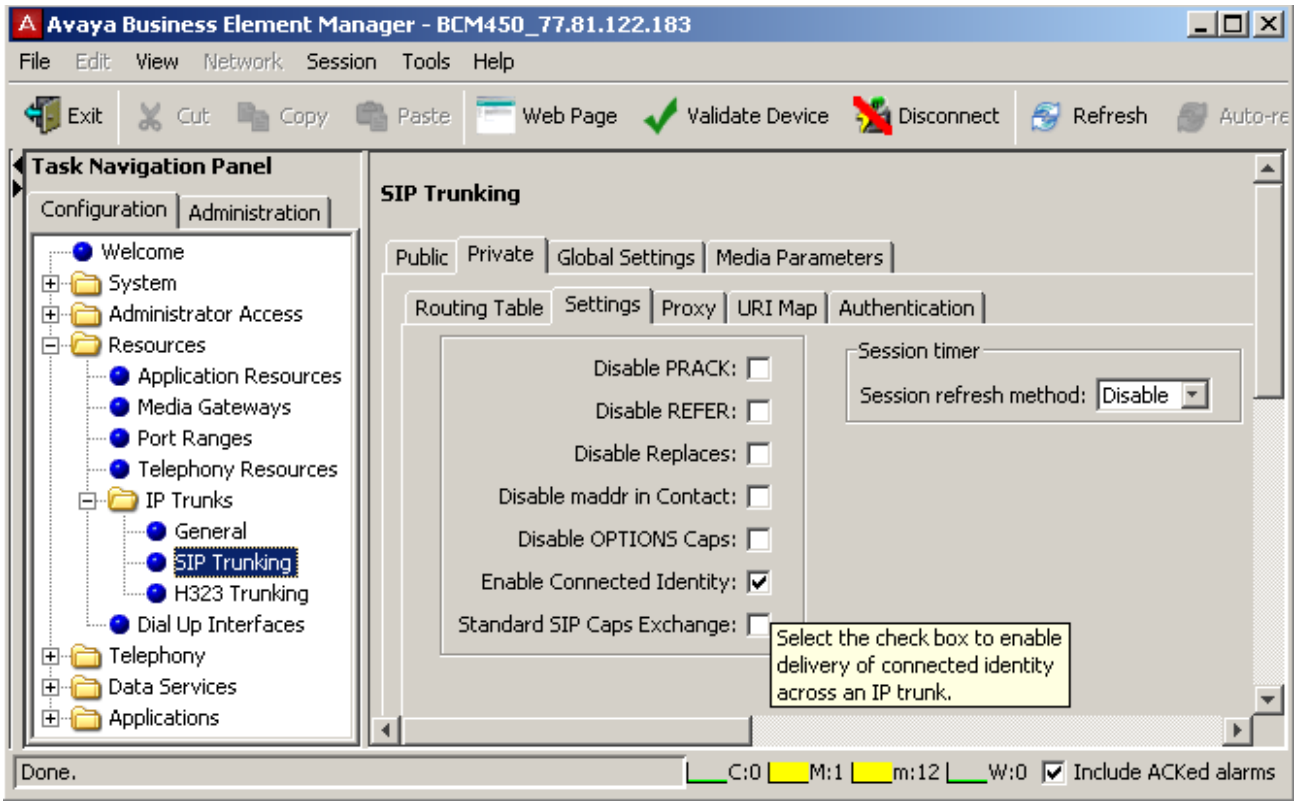
2.2.7 BCM MCDN Line

MCDN is a supplementary service available for private trunks.
An MCDN keycode is required to activate this capability.
MCDN is supported for PRI and VoIP(H323 and SIP) trunks .
When a PRI trunk is used the **SL-1** Protocol should be selected.
If connected with other BCMs through a VoIP trunk then **CSE** protocol should be selected.
In our setup, a private trunk between the BCM systems is configured over SIP.
To create a private SIP trunk go to **Configuration >Resources > IP Trunks > SIP Trunking > Private > Routing Table** tab. Click the **Add** button and complete the fields of the Add Route dialog box:

- **NAME** – Enter a unique name for this route, usually the name of the remote system is chosen.
- **Destination Digits** – Enter the destination digits for the remote system. It should be the same as Destination Code defined under the Telephony > Dialing Plan > Routing > Destination Code tab.
- **Domain** – Enter the domain name or IP Address for the remote BCM.
- **IP Address** – Enter the IP Address of the remote BCM.
- **Port** - The port number should be set to 5060.
- **MCDN Protocol** – **CSE** protocol should be selected.
- Retain the default values for all other fields.



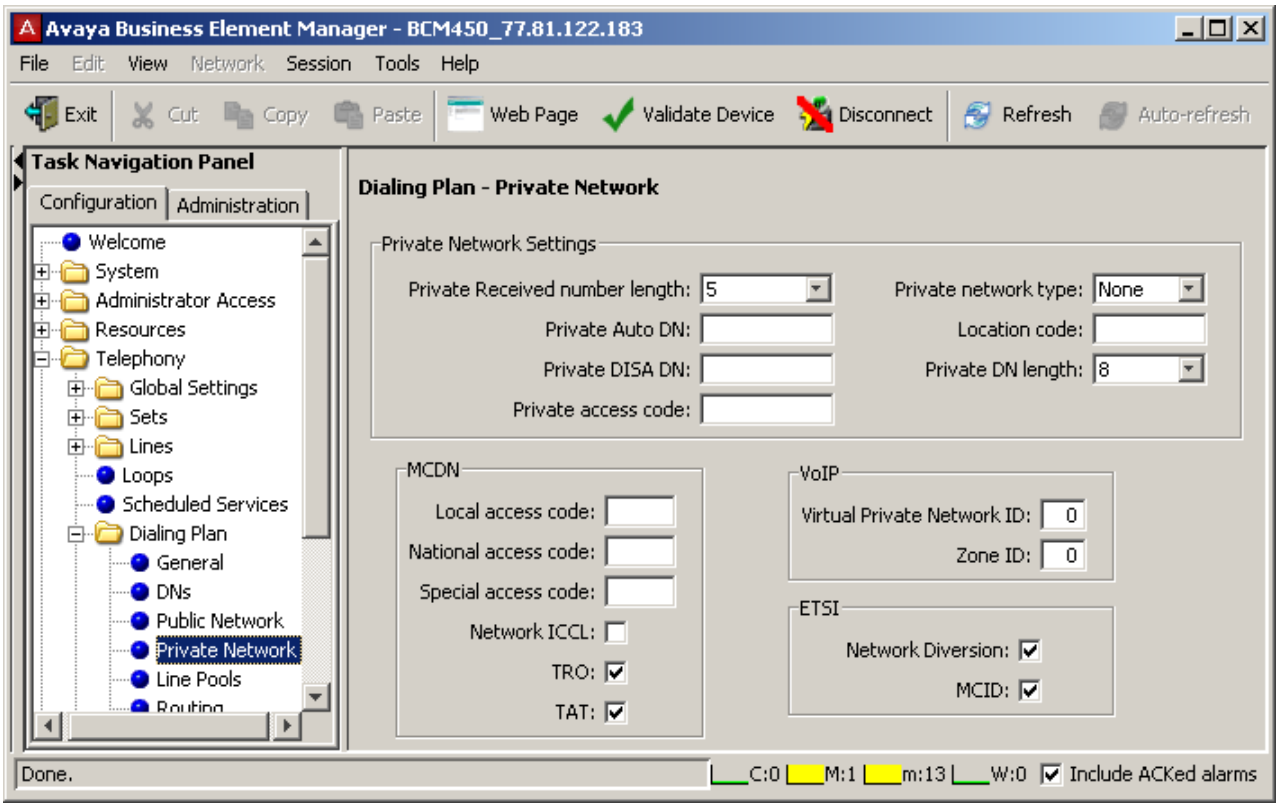
For SIP Private Trunk settings go to **Configuration > Resources > IP Trunks > SIP Trunking > Private > Settings** tab.
- **Enable Connected Identity** - Should be enabled to activate the capability to send the connected identity.
- Retain default values for all other parameters.



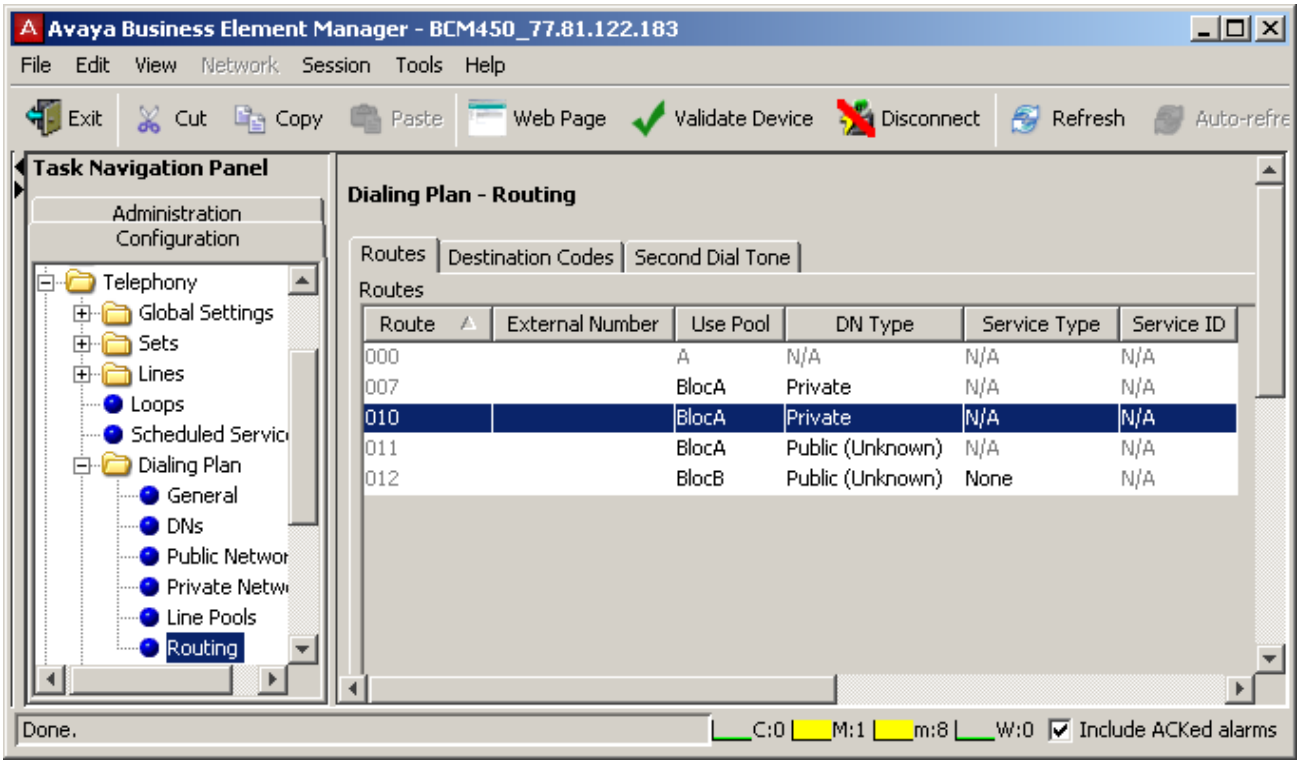
Go to **Telephony > Dialing Plan > Private Network** to configure the Dial Plan for Private Networks.
These settings should be coordinated with the remote system.
- Private Received Number Length – It should be the same as DN length. It represents the number of digits of an incoming dial string that the system uses to determine if a call tagged as Private fits the system private DN numbering. The length can be between 2 to 7 digits.
- Private access code - This code identifies this system to the private network. The Private access code is part of the dialing plan but is not a part of the numbering plan. It is used in MCDN UDP private networks.
- Private network type – You can specify whether your Private network uses a coordinated dialing plan (CDP) or a universal dialing plan (UDP). If you choose None, the private networking supplementary services are not available.
- Location code - This code identifies this particular system for calls within the network for a UDP dialing plan. This number must be unique.
- Private DN length - The Private DN length parameter specifies the length of a dial string that the system uses to determine that the call is a private network call, when the route uses DN Type: Private. The length can be between 3 to 14 digits.
MCDN network values:

Private networking also provides access to tandem calling and toll bypass functionality to users calling into systems. In this way, calls are routed as private over the private network and then flagged as public to go out to the end node PSTN.

- **Local access code** – This number is prepended to an incoming E.164/ Local call.
- **National access code** - This number is prepended to an incoming E.164/ National call.
- **Special access code** - This number is prepended to an incoming E.164/ International or Private/Special call.
- **Network ICCL** - ISDN Call Connection Limitation is part of the call initiation request. This feature acts as a check at transit PBX points to prevent misconfigured routes or calls with errors from blocking channels**.**
- **TRO –** Trunk Route Optimization occurs during the call setup. This feature finds the most direct route through the network to send a call between nodes. It should be enabled.
- **TAT** - Trunk anti-tromboning works during an active call to find the optimum routing.
- Retain default values for VoIP and ETSI settings.



Go to Telephony > Dialing Plan > Routing to program the routing for private network. The configuration is made in the same manner as for Public lines, but with the difference that the DN Type needs to be set as **Private**.



BCM Monitor can be used to monitor the system status and activity.
BCM Monitor can be accessed from Administration > Utilities.

3.0 Issues Found

IPOFFICE-25429 – BCM fails to supervise transfer an IPOffice call back to IPOffice.
IPOFFICE-26148 - No Talk Path when IPOffice blind transfers a BCM call back to BCM - Direct Media is enabled.
IPOFFICE-23530 – IPOffice doesn't reply to an '487 – Request Terminated' message that comes from BCM
IPOFFICE-24415 – IPOffice phone doesn't show the BCM endpoint number after blind transfer is completed.