

Configuring Interoperability between Avaya IP Office, Avaya Business Communication Manager, and Avaya Communication Server 1000

Issue 01.01
July 2012

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1.0 Introduction

This document provides a description of the solution where an Avaya Network Routing Service (NRS) is used to connect Avaya IP Office, Avaya Business Communication Manager and Avaya Communication Server 1000 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.

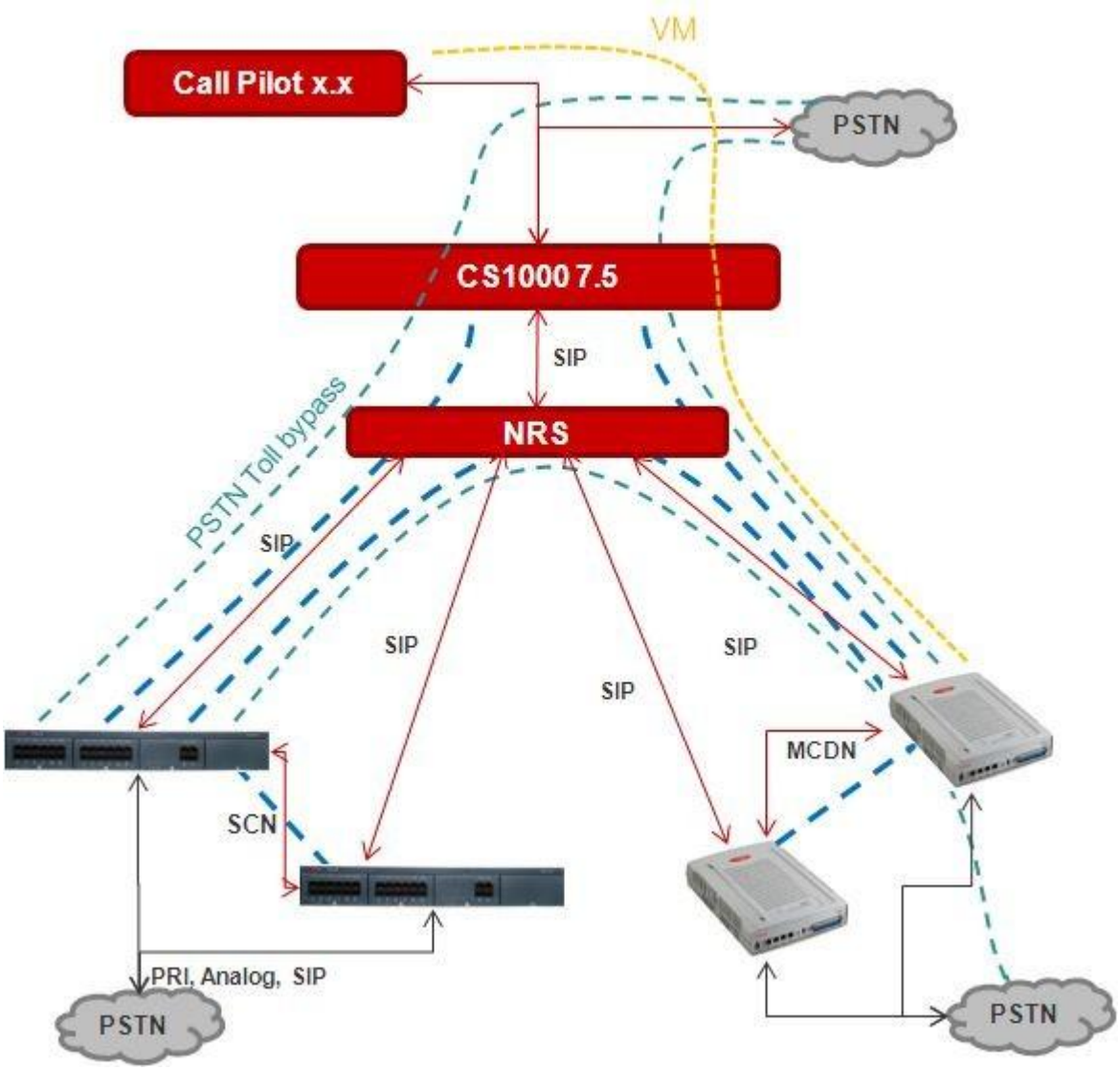
- **Abandoned call –** We verified that:
  - Abandoned calls are properly cleared from both parties.
  - Abandoned calls are tracked in the History list.
  - The involved trunks and resources are released promptly when a call is abandoned.

- **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.

- **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 / CS1K 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 Edition
- BCM 6.0 with latest Service Update
- NRS 7.5 with latest Service Update
- CS 1000 7.5 with latest Service Update

1.5 Hardware Platforms

- IP Office 500v2

- IP Office Mid Market
- BCM450
- BCM50

2. Configuration Guide

2.1 Avaya IP Office Configuration

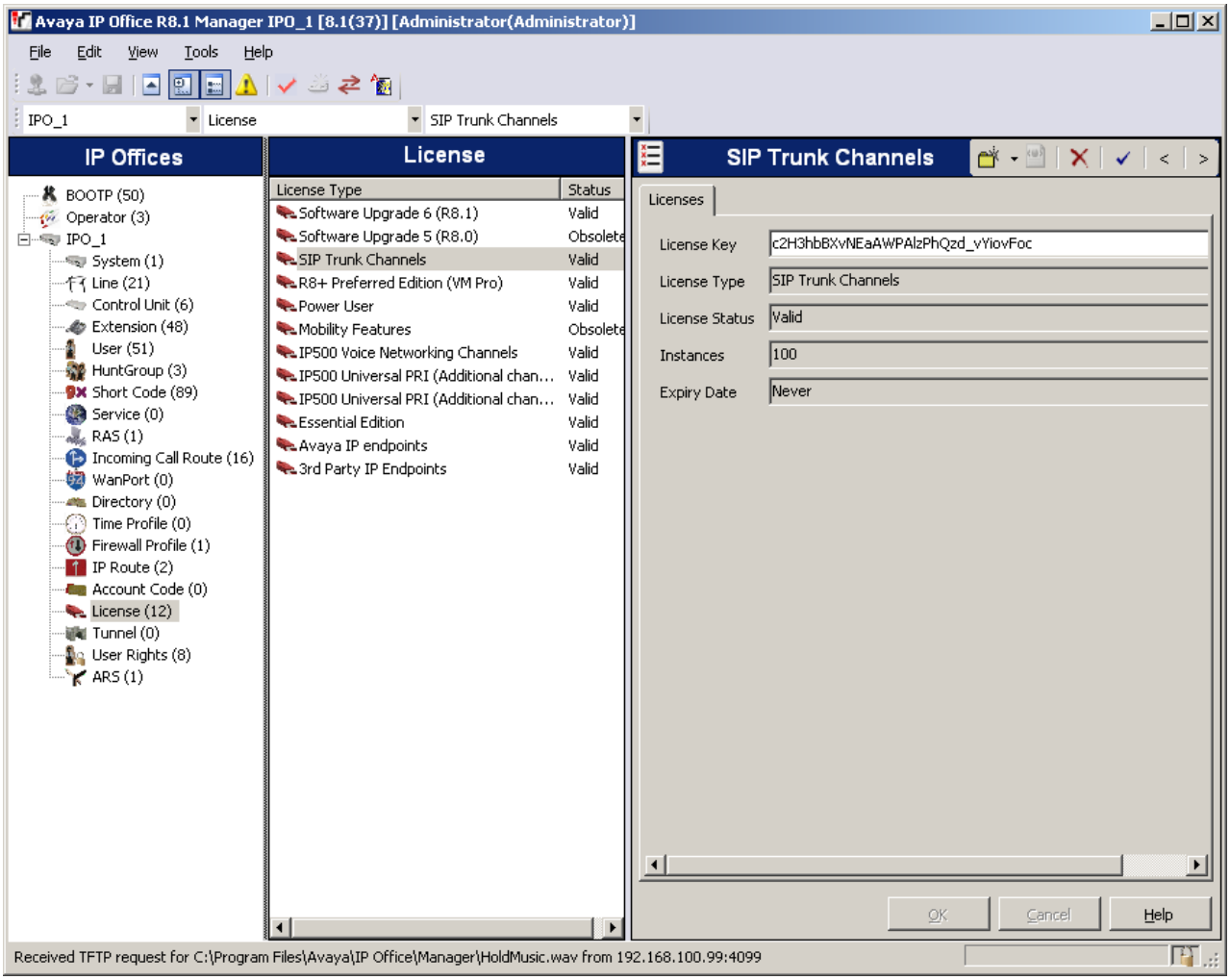
This section provides the procedures for configuring Avaya IP Office using Avaya IP Office Manager.

The procedure covers the following areas:

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

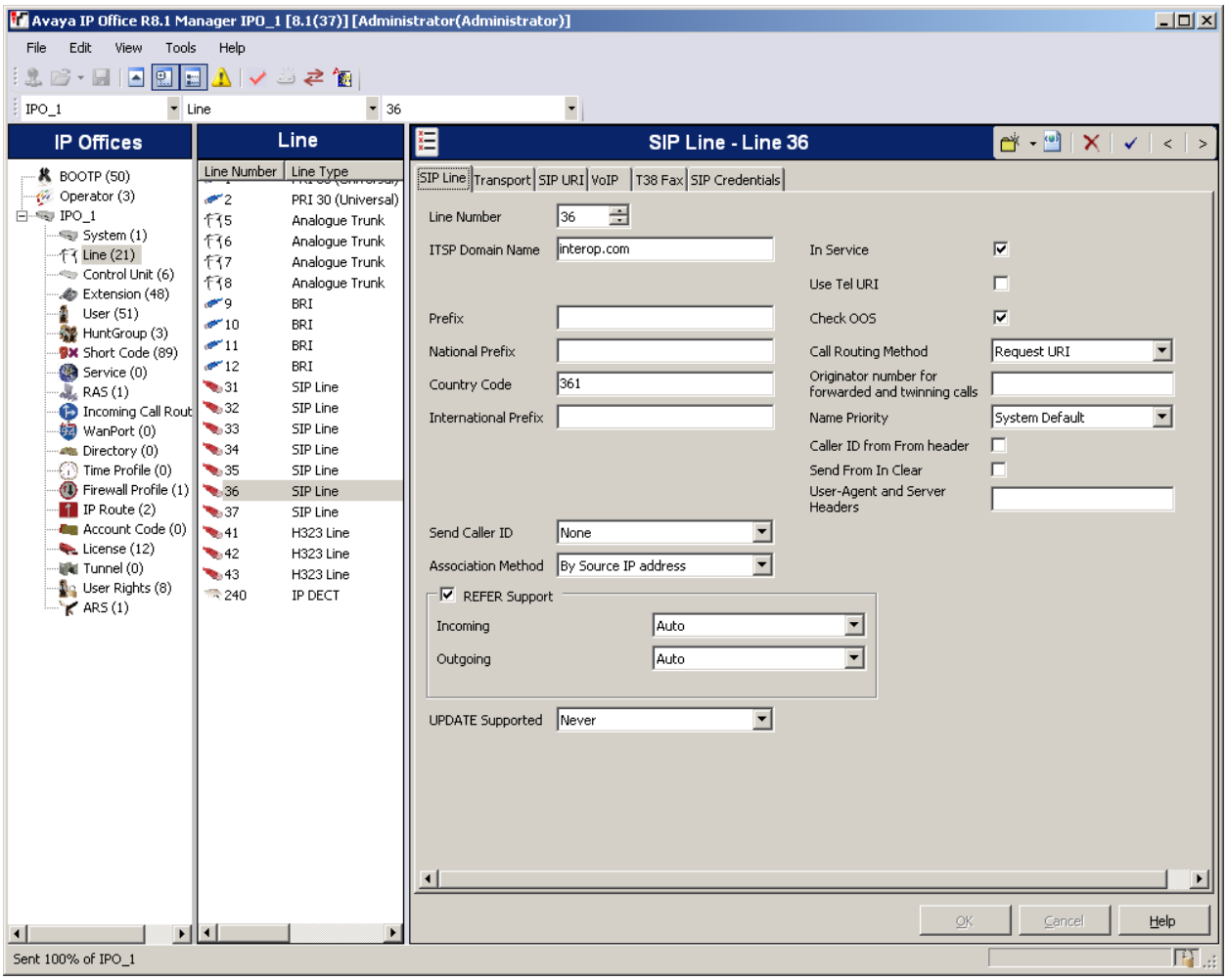
2.1.1 IP Office SIP Trunk Licensing

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 NRS

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 domain name as defined in NRS, **interop.com** in our case.
- Make a note of the **Line Number**.
- Retain default values for all other fields.
- Send Caller ID can be set to **P Asserted ID** for proper display updates.



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

- ITSP Proxy Address - Enter the IP address of **NRS**. This is the SIP Proxy address used for outgoing SIP calls.

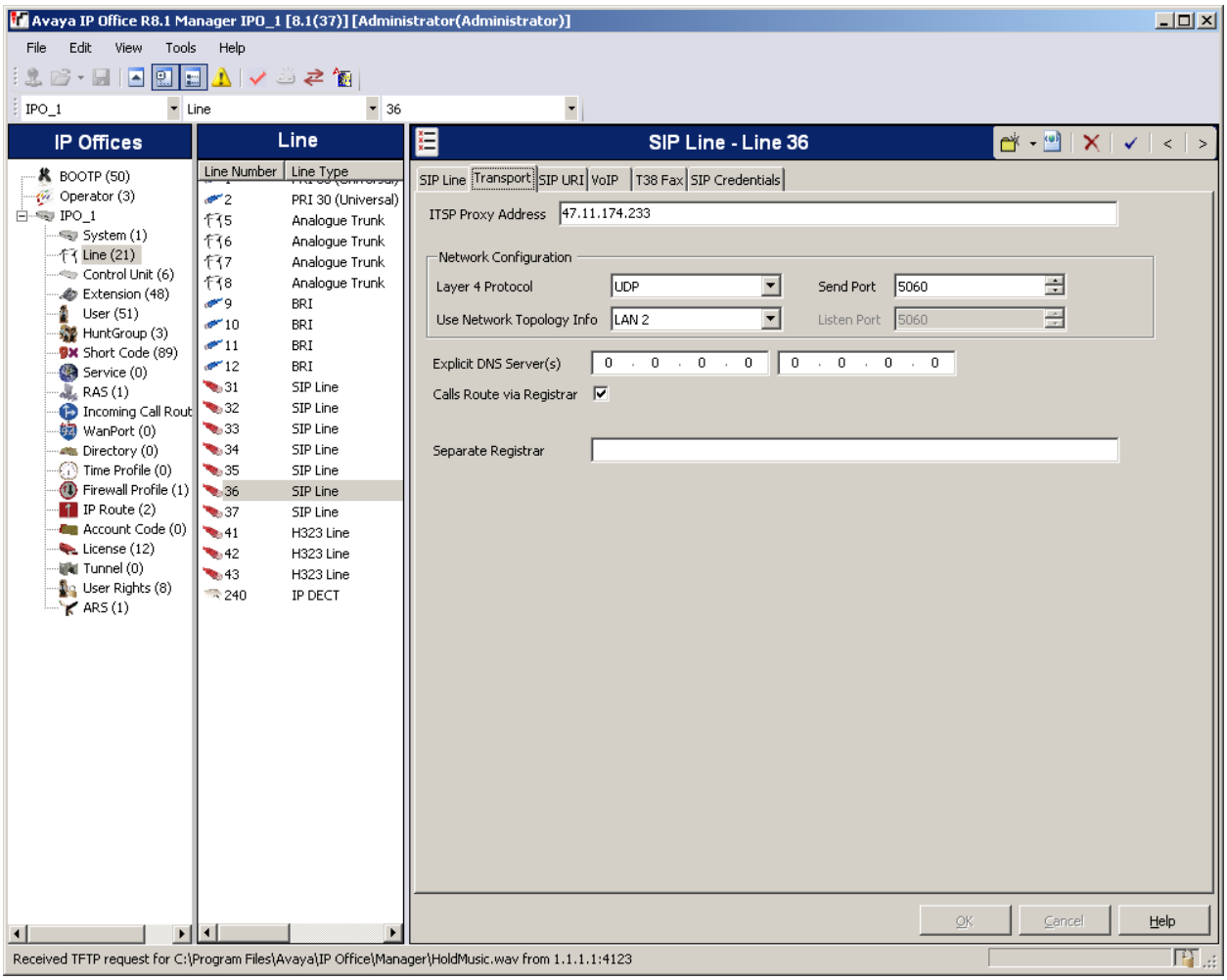
- Layer 4 Protocol - Ensure that **UDP** is selected. This field sets which protocol the line uses for sending and receiving SIP packets.

TCP protocol is not supported for interoperation with NRS/CS1000.

- Send Port - Retain the default value, **5060**. This field sets the port to which the system send 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.

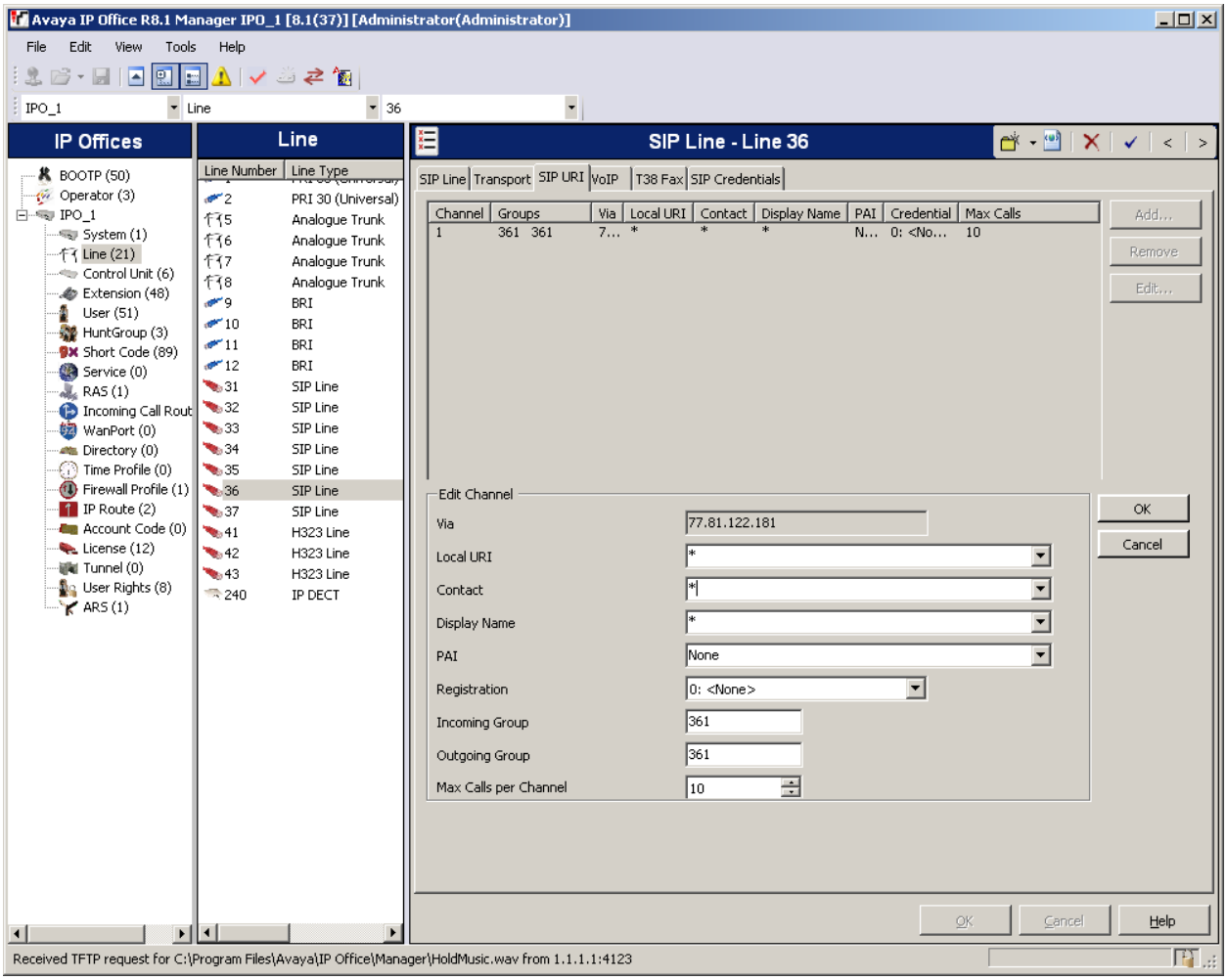


In the **SIP URI** tab, select **Add** and enter the following:
- **Local URI** - Enter "***".** This field sets the 'From' field for outgoing SIP calls using this URI.
- **Contact** - Enter "***".**This field sets the 'Contact' field for SIP calls using this URI.
- **Display Name** - Enter "***".** This field sets the 'Name' value for SIP calls using this URI.
- **PAI** – The default setting should be retained, 'None'. 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.

For an easy maintenance, the Incoming and Outgoing Group ID should be appropriate to the Line Number.

- 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**, codec preference defined under System > Codecs tab will be used.

Custom option, available under drop-down list, allows a specific codec selection to be made.

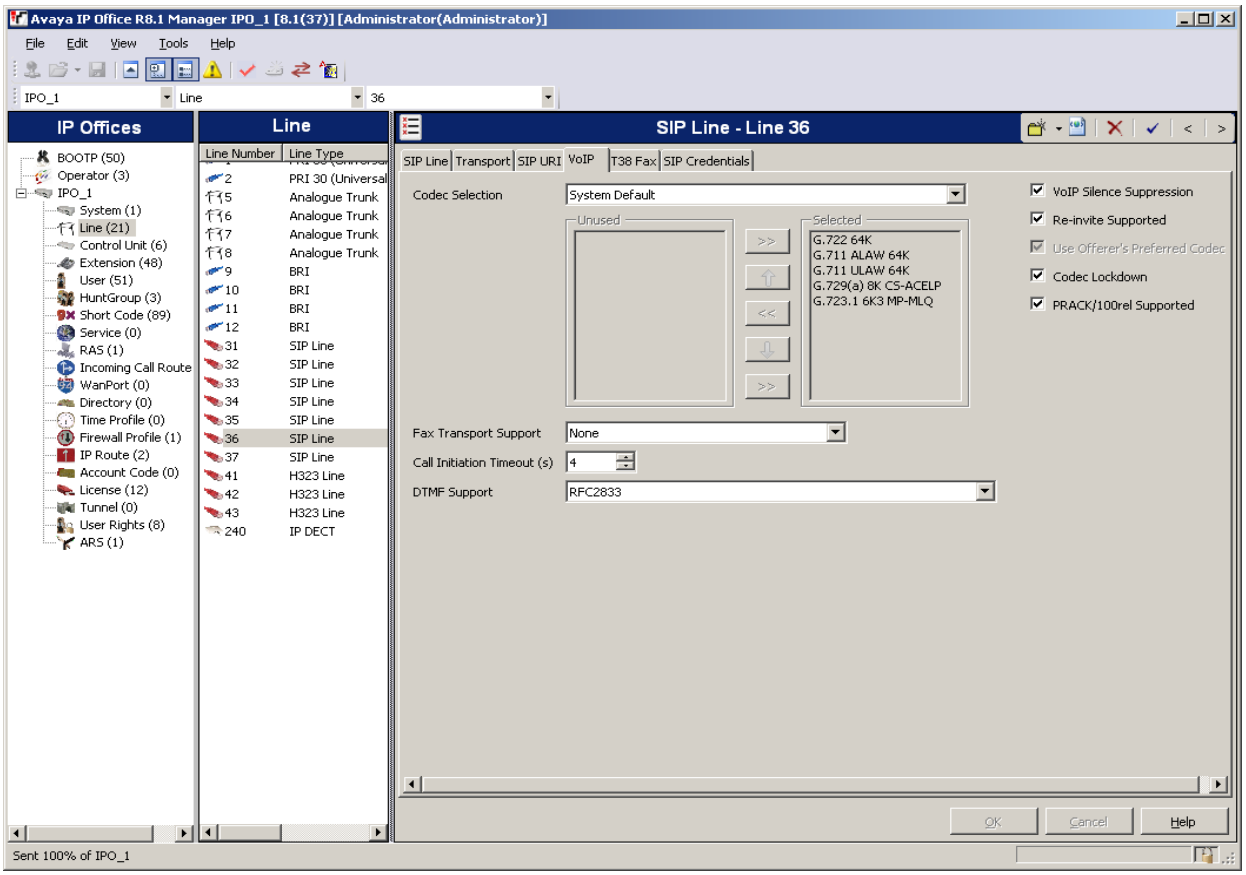
- VoIP Silence Suppression -When selected, this option will detect periods of silence on any call over the line and will not send any data during those silent periods. This setting should be coordinated with BCM and CS1000.

- 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, SIP trunk supports Provisional Reliable Acknowledgement 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.

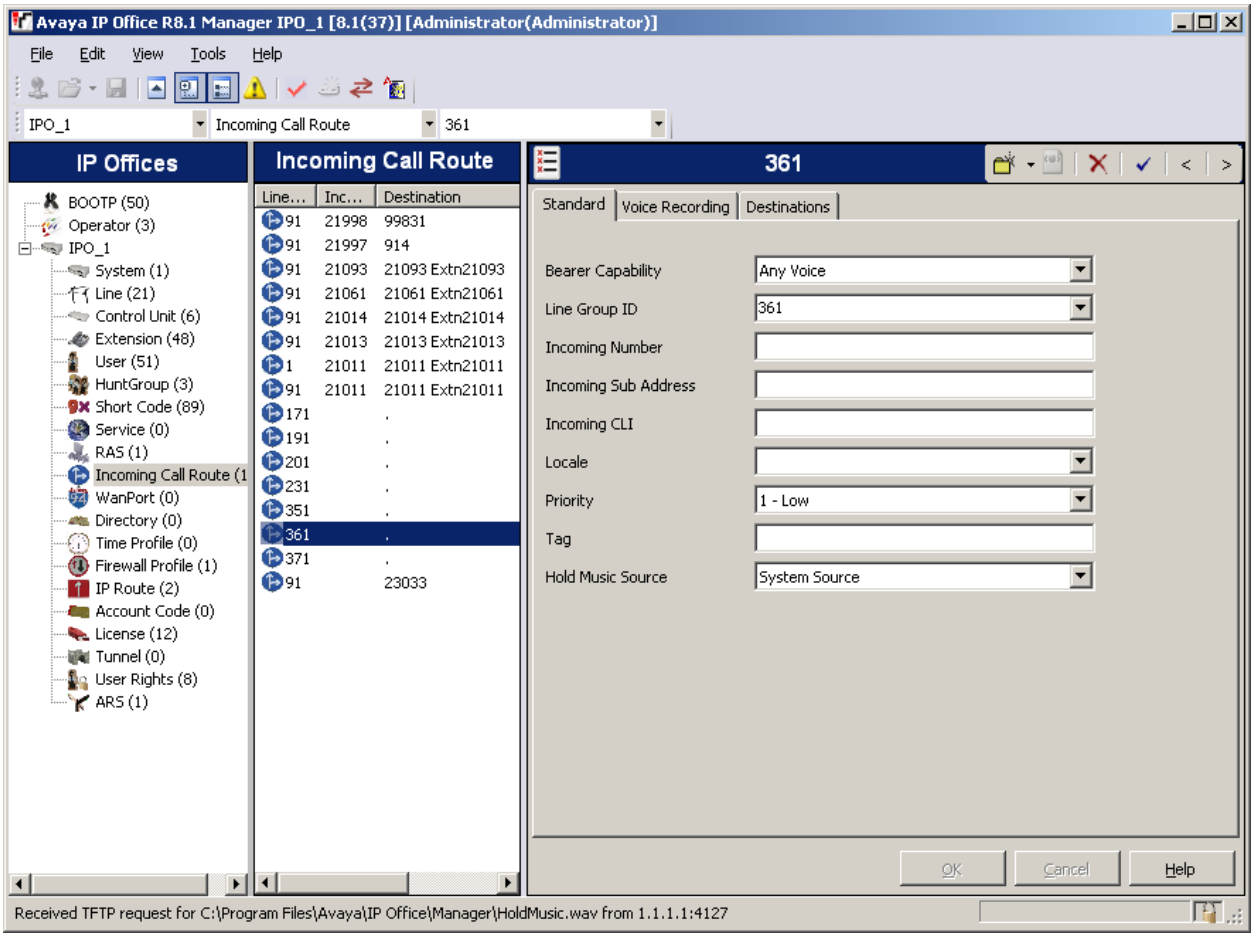


2.1.3 IP Office Incoming Call Route

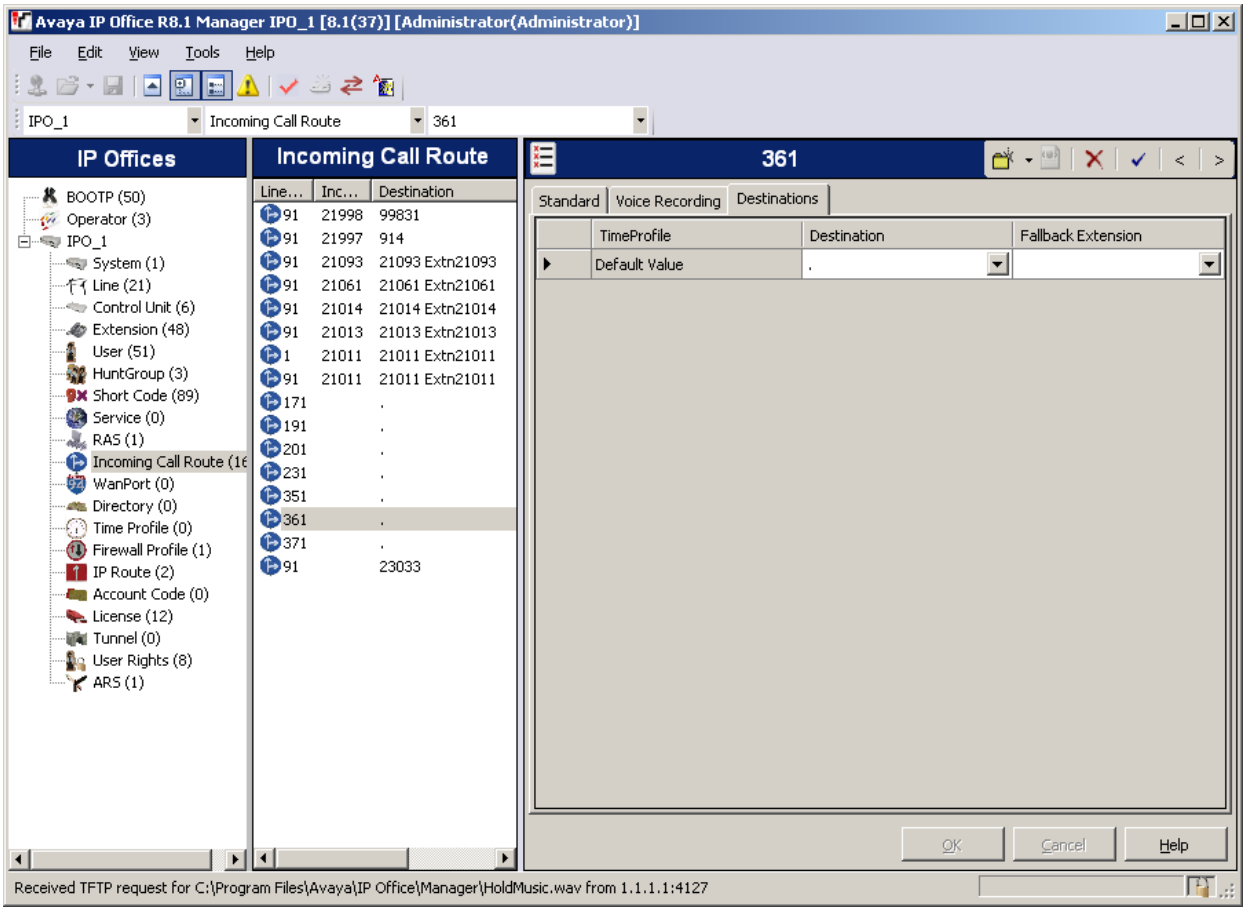
Incoming call routes are used to match incoming calls with destinations. Routes can be based on the incoming line group, the type of call, incoming digits or the caller's ICLID.

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 sub address 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.



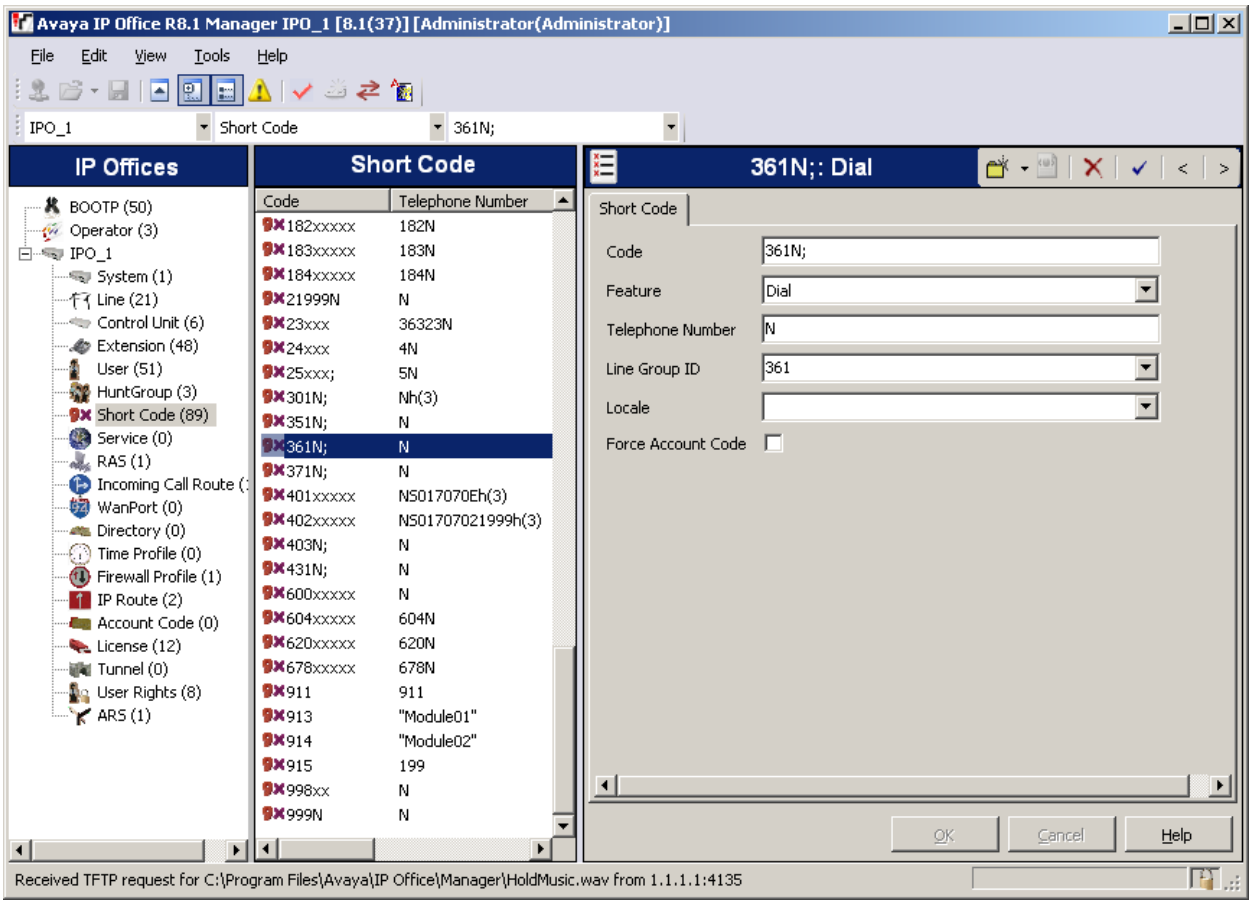
In the **Destination** tab:
- Default Value field - Enter "**.**" into the Destination column to match all the destinations available.
Or you can enter a specific destination using drop-down box which contains all available extensions, users, groups, RAS services and voicemail. System short codes and 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

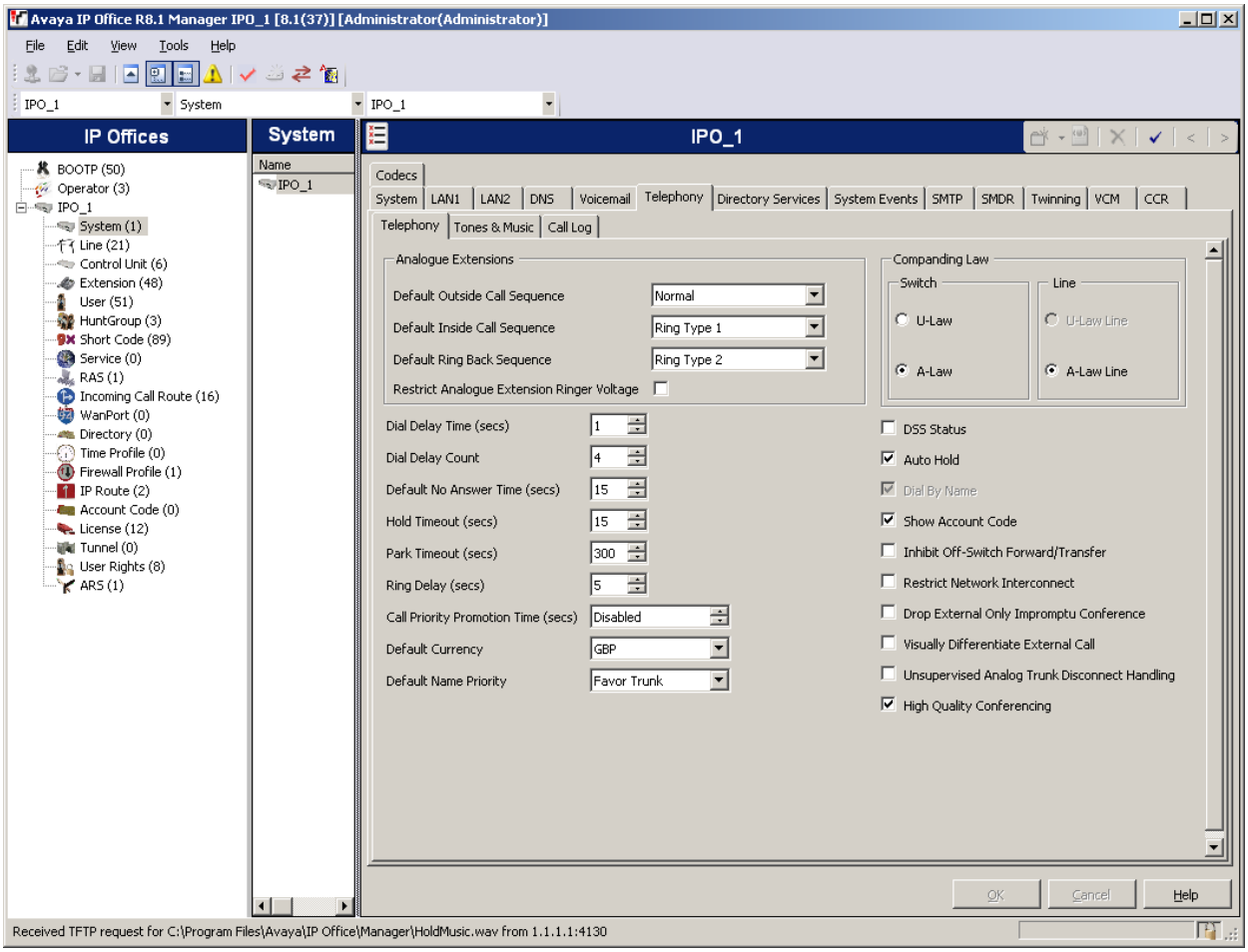
The system uses short codes to access the SIP trunk for outgoing calls.
From the configuration tree in the left pane, select **Short Code,** right-click and select **New** to add a new short code.
Introduce the following:
- **Code -** the dialing digits used to trigger the short code.
- **Feature –** 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 SIP line under 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 being prompted to enter a valid account code before the call is allowed to continue.

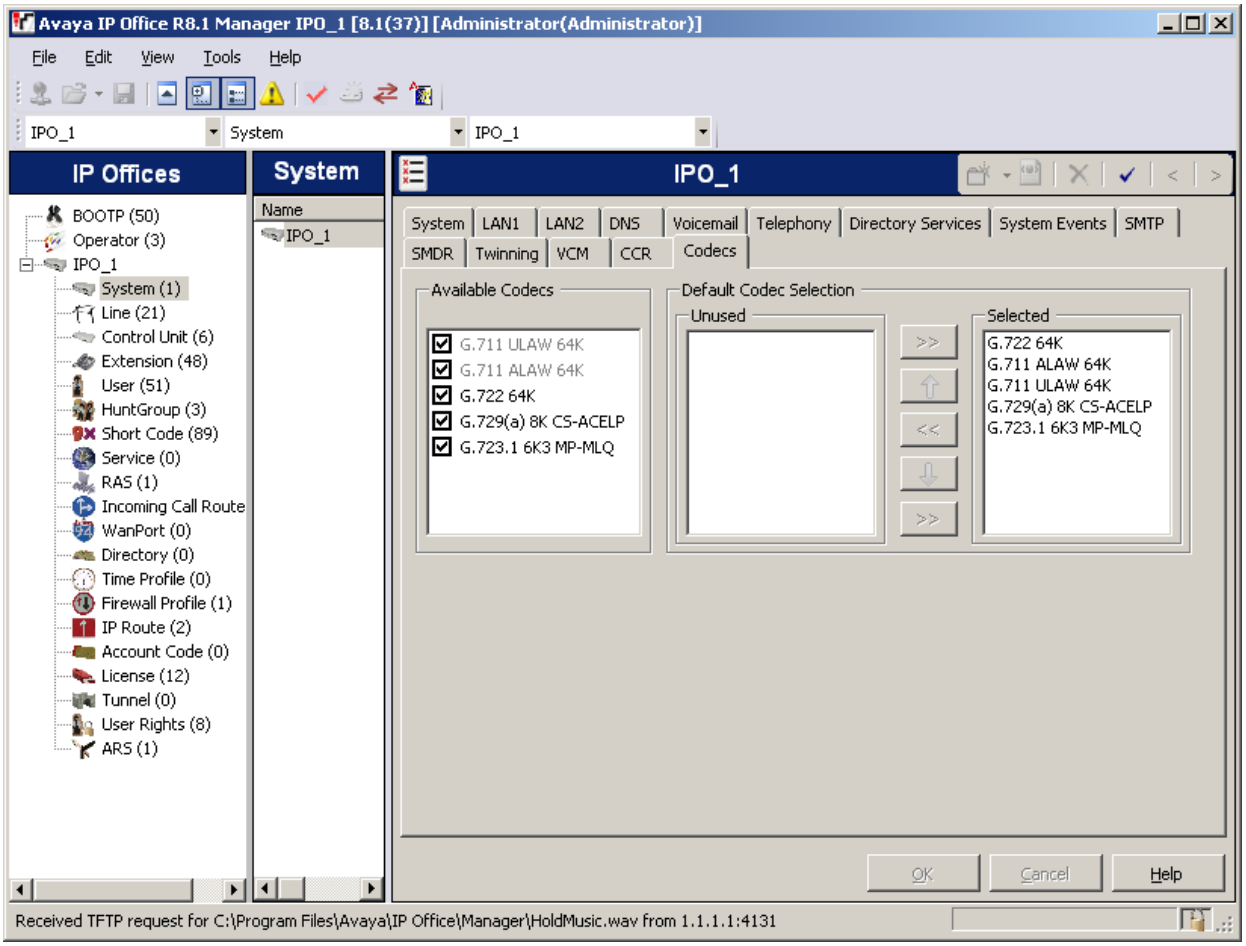


2.1.5 IP Office System Settings

In the **Telephony** tab:
- **Companding Law** - Choose the appropriate Companding Law depending on the region (U-Law for North American 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 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 - sets the codecs preference.



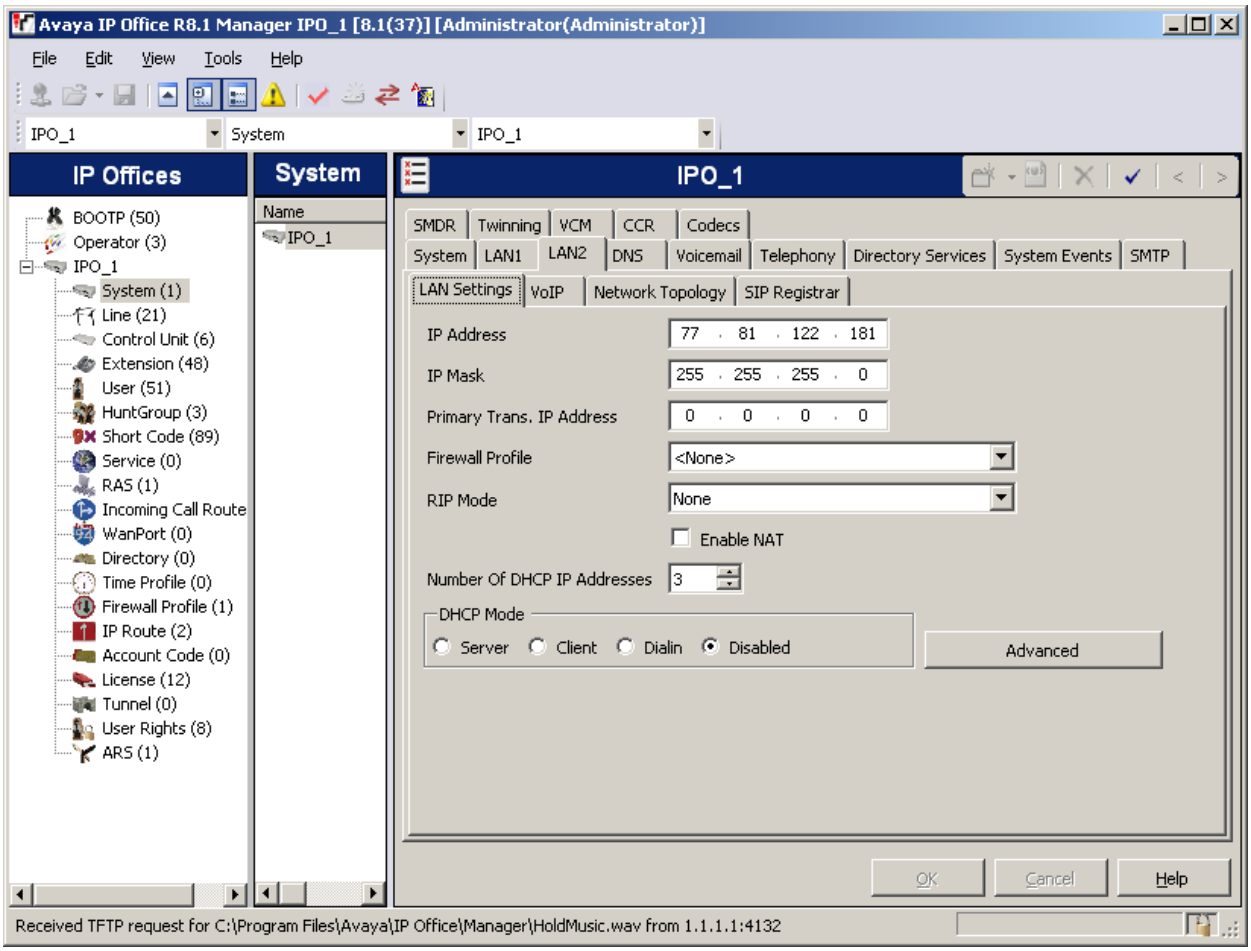
In the **LAN** tab:
Usually LAN1 is used for local network and LAN2 is used to connect to NRS, via WAN. LAN2 settings are presented bellow.

- LAN Settings tab:

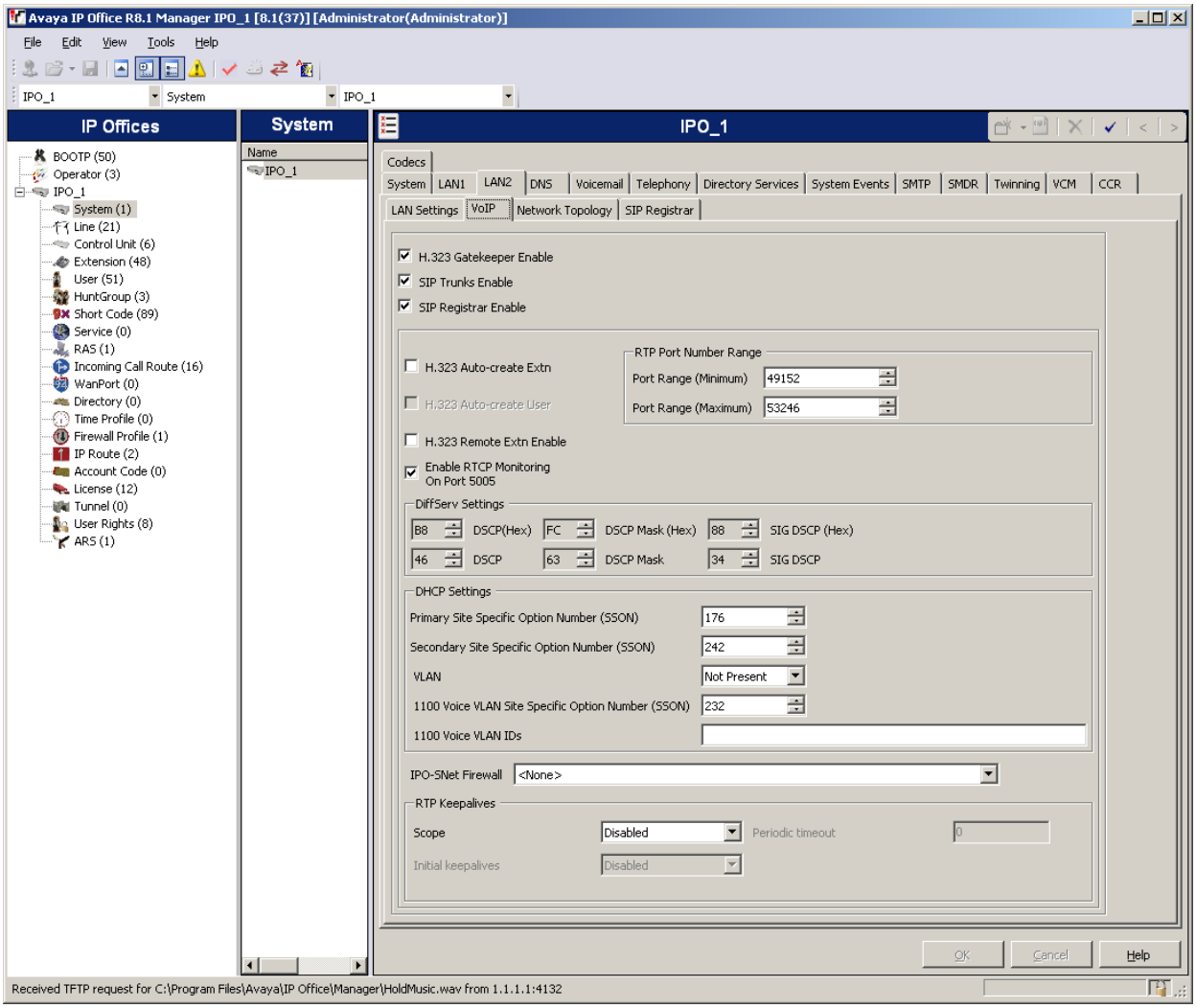
Enter the appropriate IP Address and IP Mask in the corresponding fields.

Retain Default values for all other fields.

In case of DHCP client or server is required this can be activated using DHCP section.



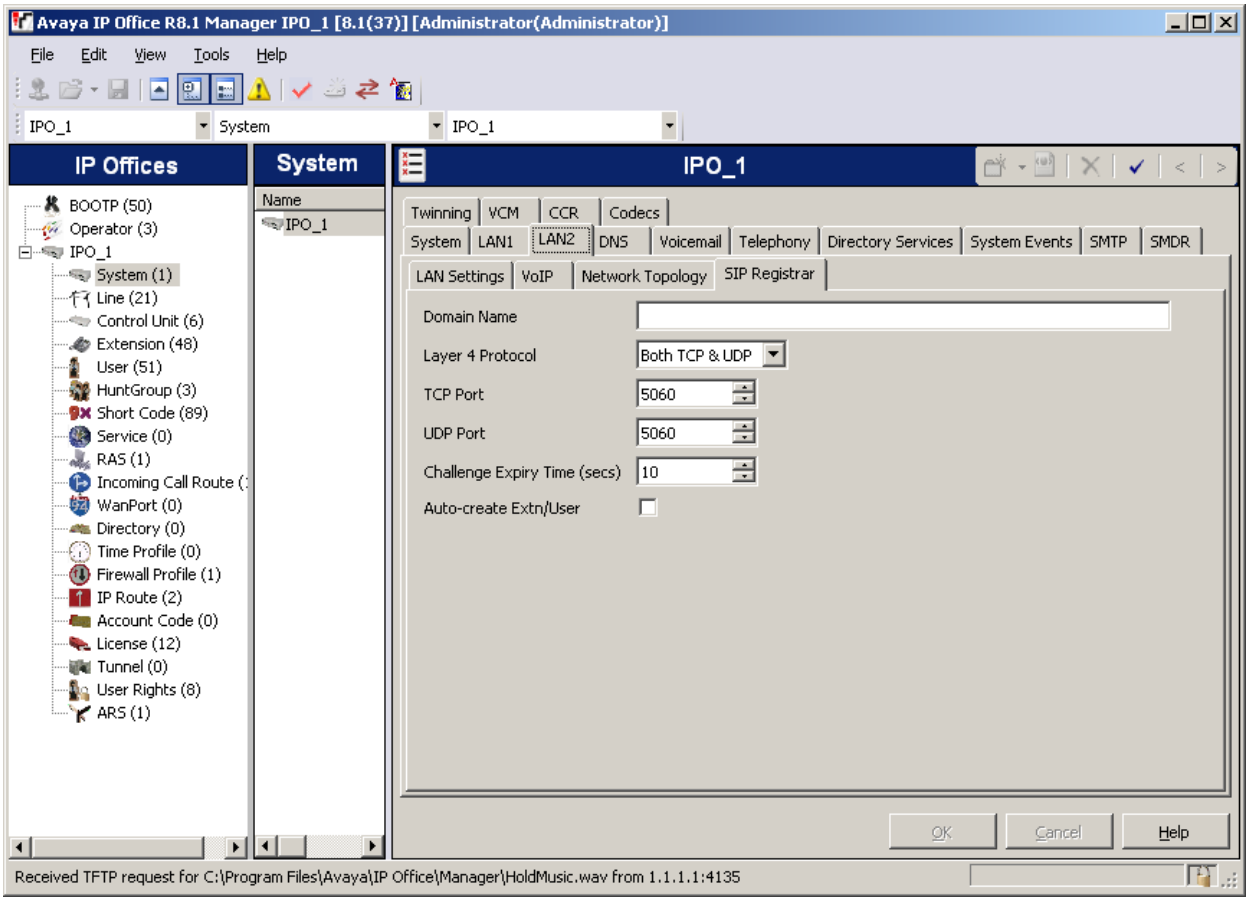
VoIP tab:
- **Default** values should be retained for all fields.
- Ensure that **SIP Trunks Enable** is checked.



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)

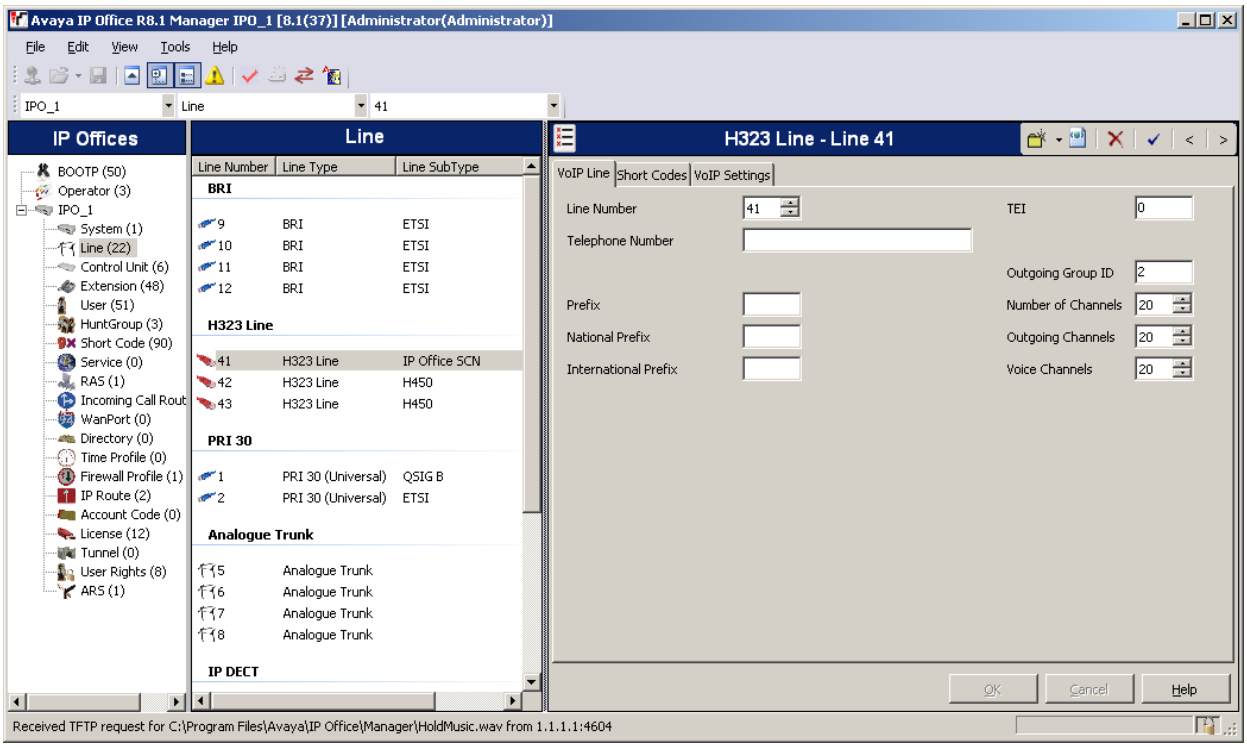
Systems linked by H.323 IP trunks can enable voice networking across those trunks to form a multi-site network. Within a multi-site network, the separate systems automatically learn each other's extension numbers and user names. This allows calls between systems and support for a range of internal call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

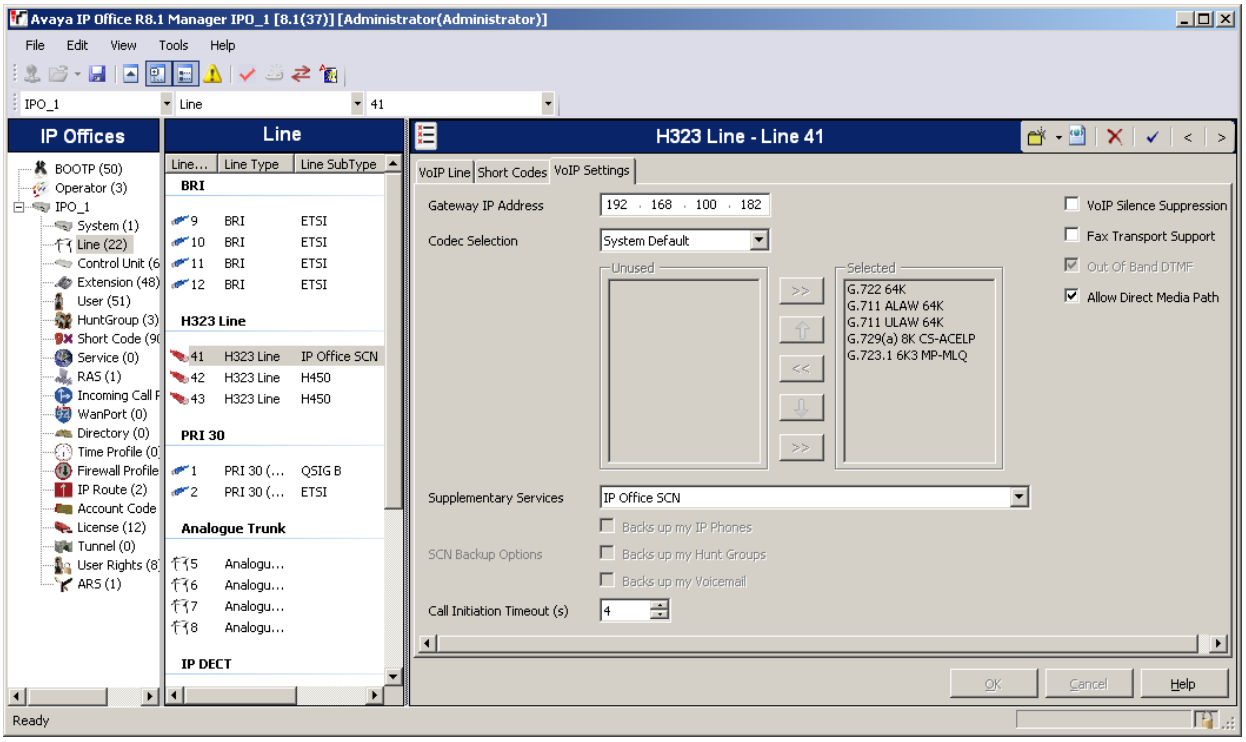
From the configuration tree in the left pane, right-click on **Line** and select **New > H323 Line** to add a new H323 Trunk.

In **VoIP Line** tab:
- Adjust the **Line Number** as you desire, this must be unique.
- Change default value for Outgoing Group ID into 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 **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**.

Incoming Call Route and Shortcode are not required to exchange calls through an IP Office SCN line.

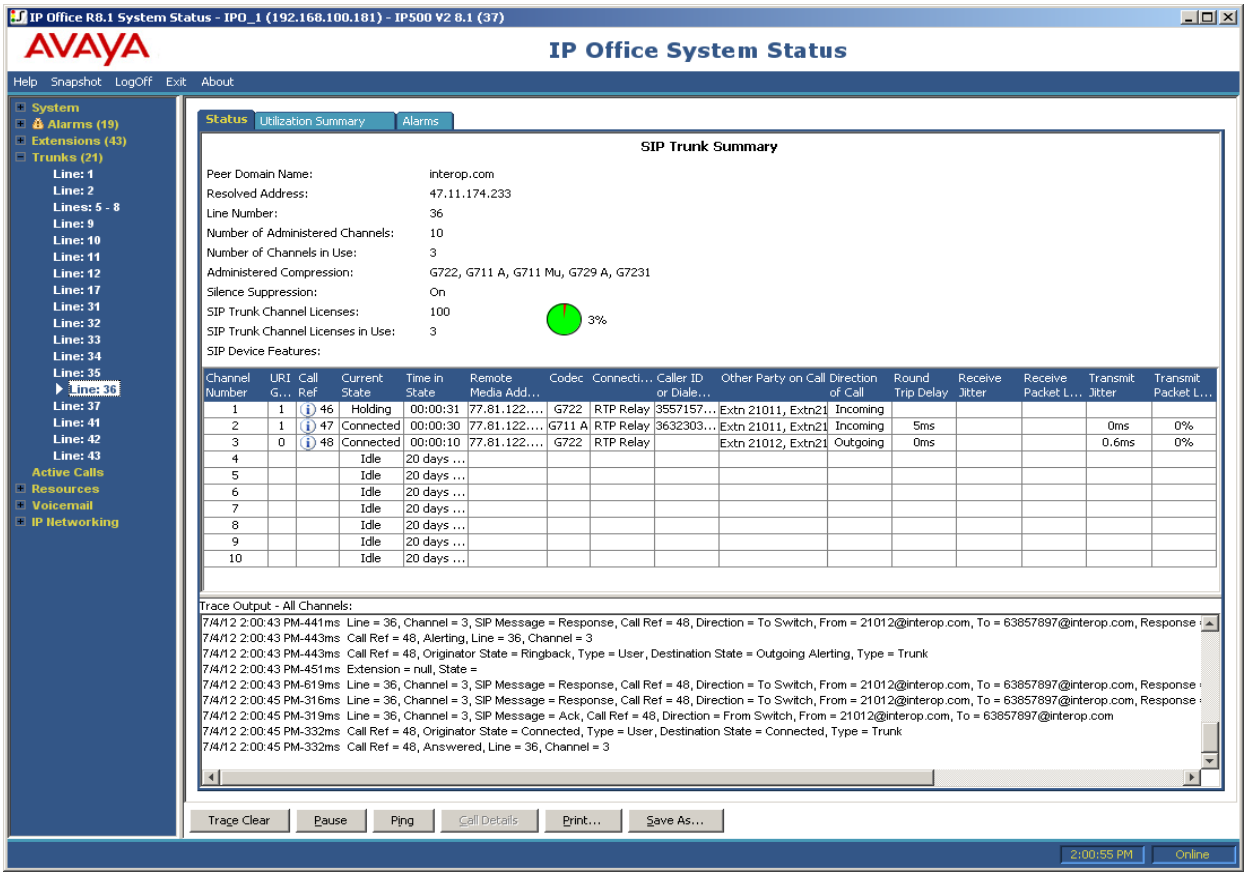


After all IP Office configuration changes are made, save the configurations and send them to the corresponding IP Office system. Note that a system reboot will be required for changes to take effect.

2.1.7 Verify Basic Connectivity

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 in **Idle** or an active state.

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



2.2 Business Communication Manager Configuration

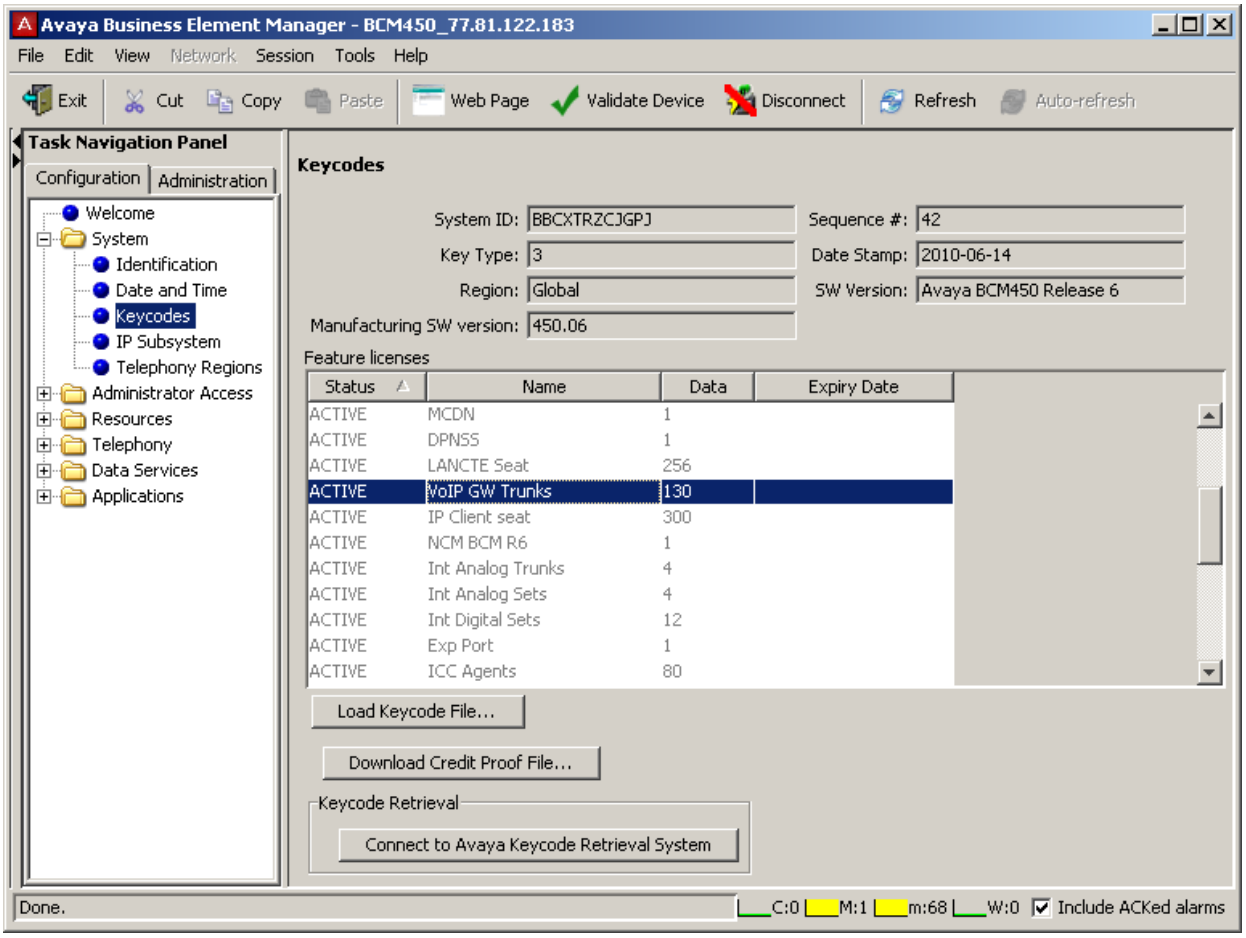
This section provides the procedures for configuring BCM using Business Element Manager.

The procedure covers the following areas:

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

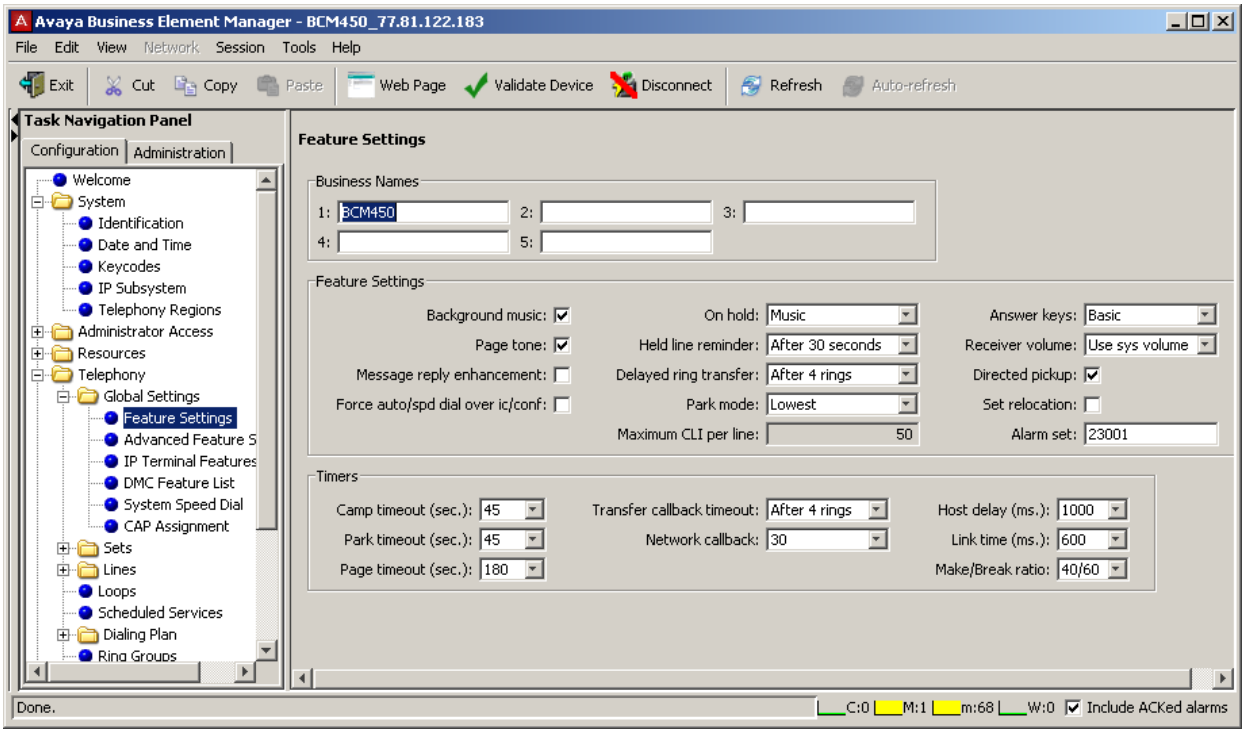
2.2.1 BCM Keycodes

'VoIP GW Trunks' license is required in order to activate IP trunks (SIP and H323) on BCM. Alternatively, the 'SIP GW Trunks' keycode can be used to enable only SIP trunks.
To check the keycodes available, login to BCM system using Business Element Manager and under Configuration tab go to **System** > **Keycodes**.



2.2.2BCM Business Name

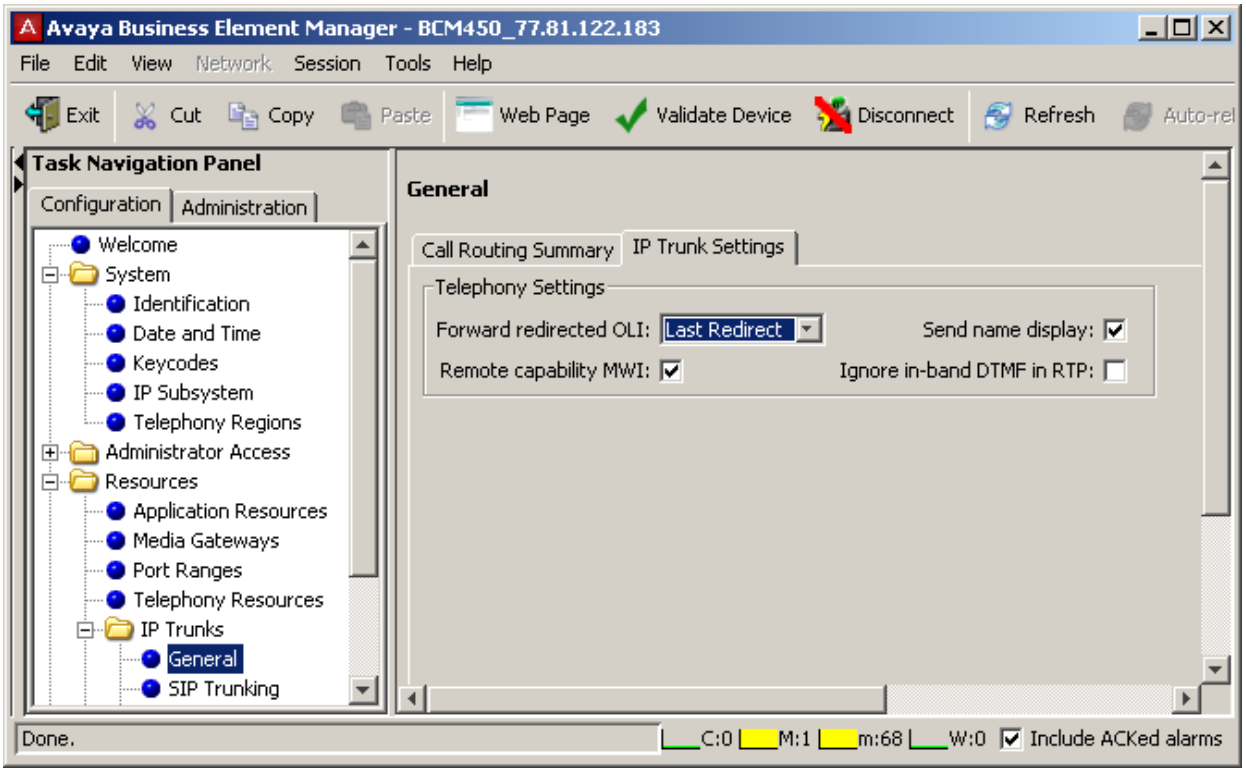
Under the Configuration tab, go to **Telephony > Global Settings > Feature Settings** and enter a Business Name. This allows the BCM to send the CLID. Network Name Display uses the configured Business Name as the prefix and, when applicable, the telephone or hunt group Name.
No space is automatically 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

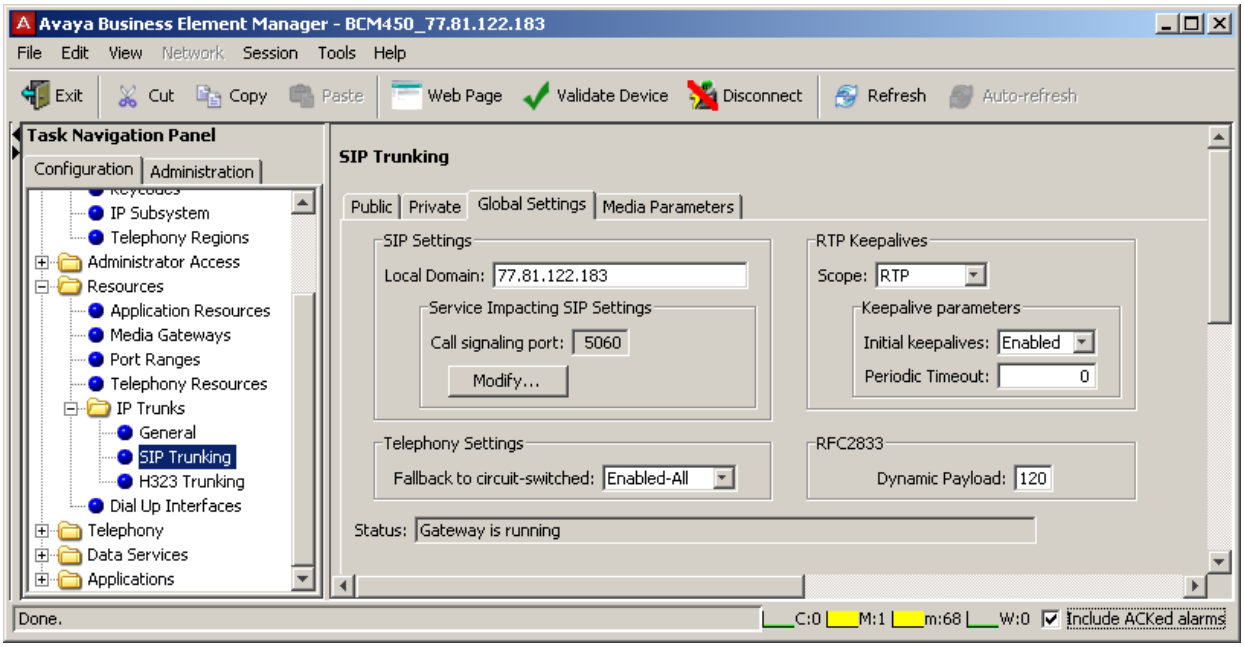
To administer IP Trunks go to **Resources > IP Trunks**.

Under **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.

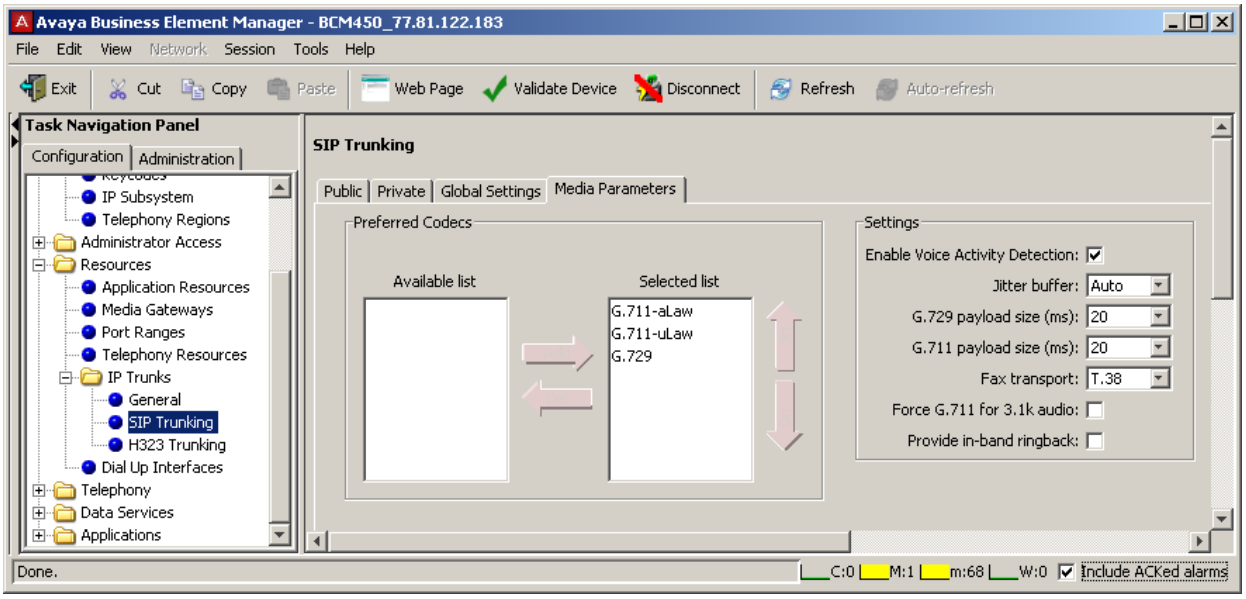


2.2.4 BCM SIP Trunking

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.



In the **Media Parameters** tab.
The SIP media parameters allow 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 and CS1000.
- **Enable Voice Activity Detection** (Silence Suppression) - Should be coordinated with IP Office and CS1000.
- **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.

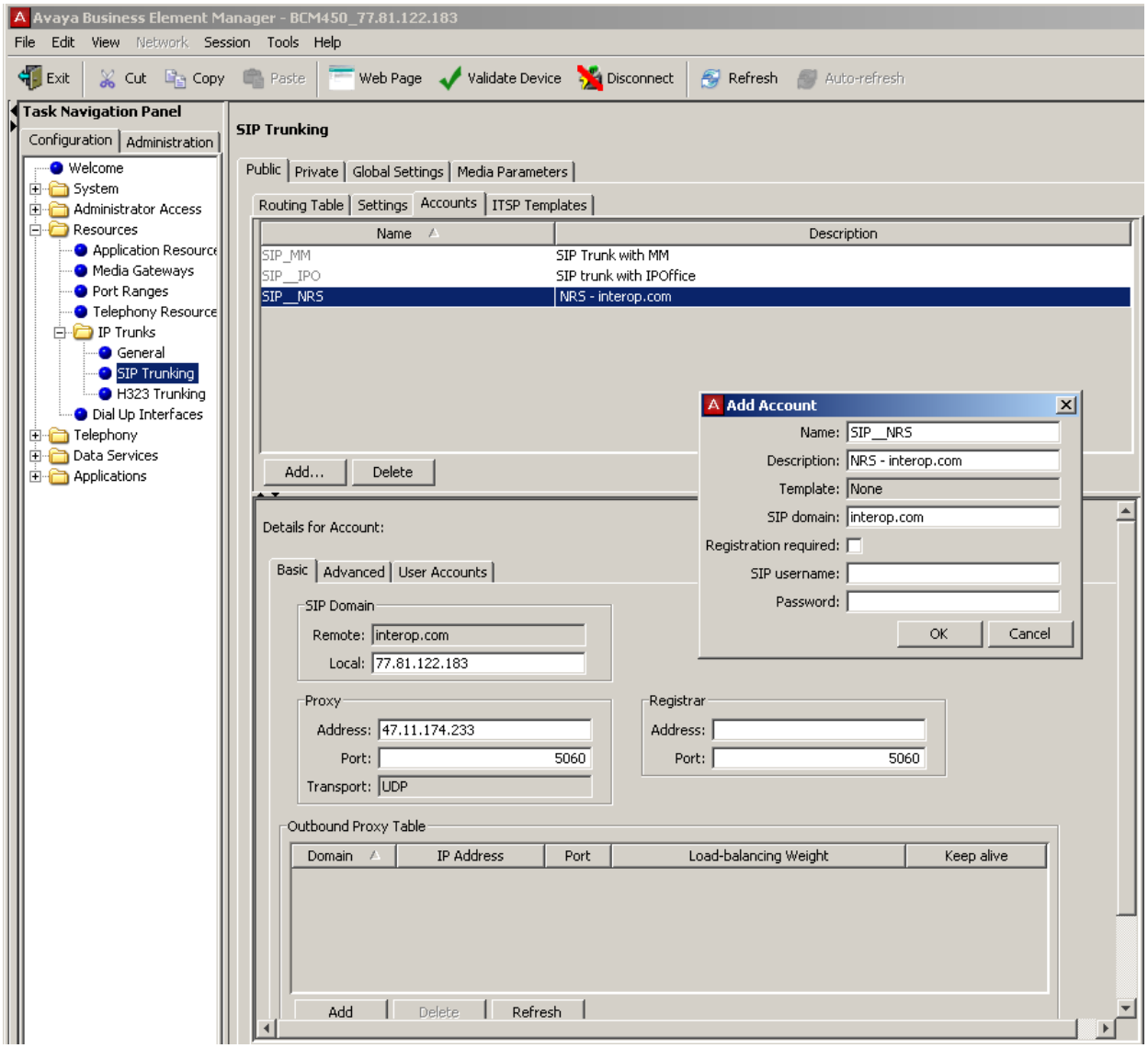


2.2.5 BCM SIP Trunking - Public trunk

Public Trunk should be used to connect the BCM to NRS.
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 NRS before you can add a route for NRS.
To configure an account, under **Public > Accounts** tab click **Add** button:
- **Name –** Introduce the desired name. This will appear in accounts list as 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 domain name as defined in NRS (**interop.com** in our case).
- **Registration Required** – Make sure it is unchecked (not applicable in our case)
- **SIP username** and **Password** - Leave these blank.

Press **OK** to complete and close the **Add Account** window.
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 –** Introduce the IP Address of the NRS.
- 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.



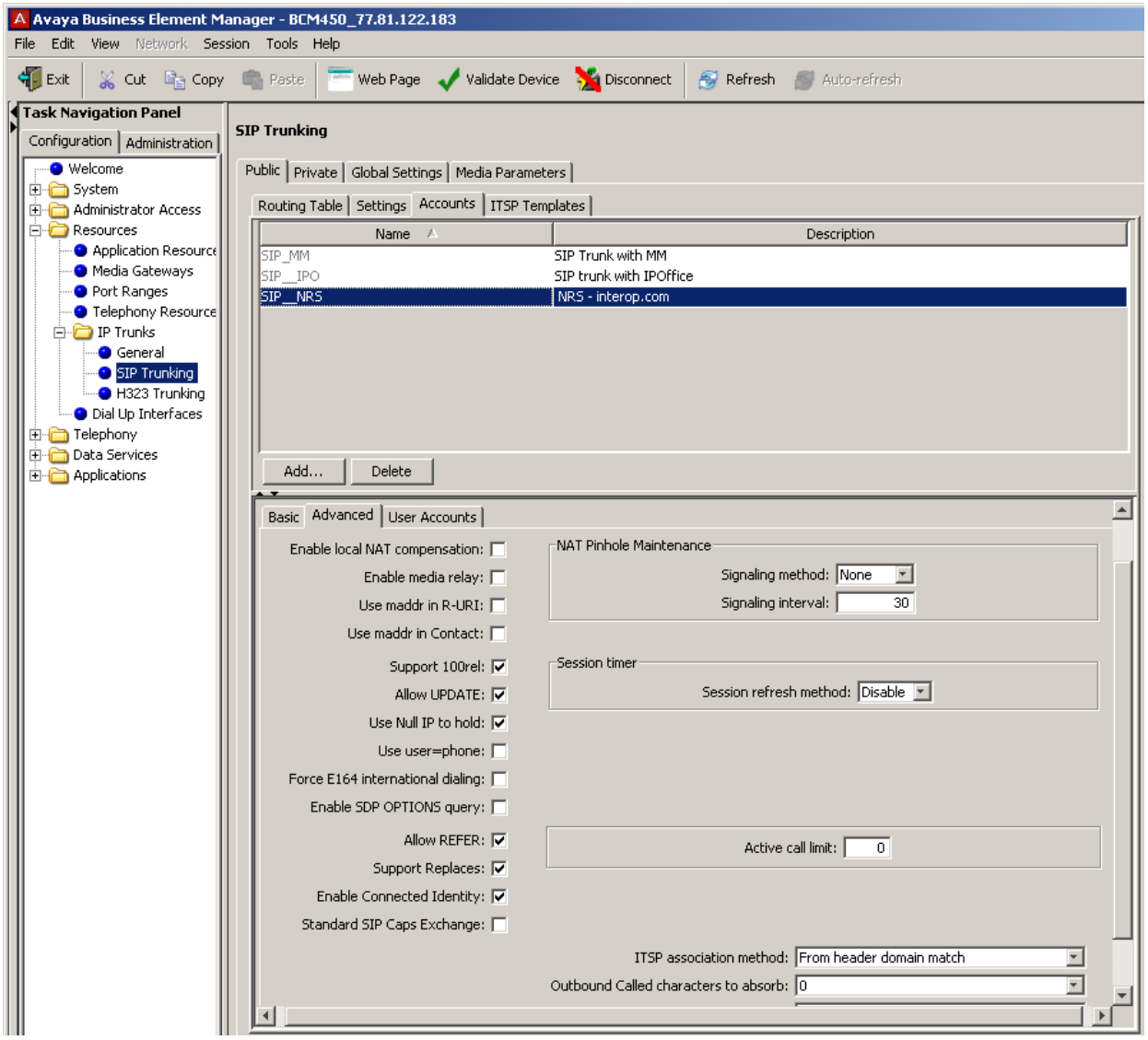
In the **Accounts – Advanced** tab:

- **Enable Connected Identity** – Should be checked. It enables delivery of connected identity across the trunk.
- Retain default values for all other fields.

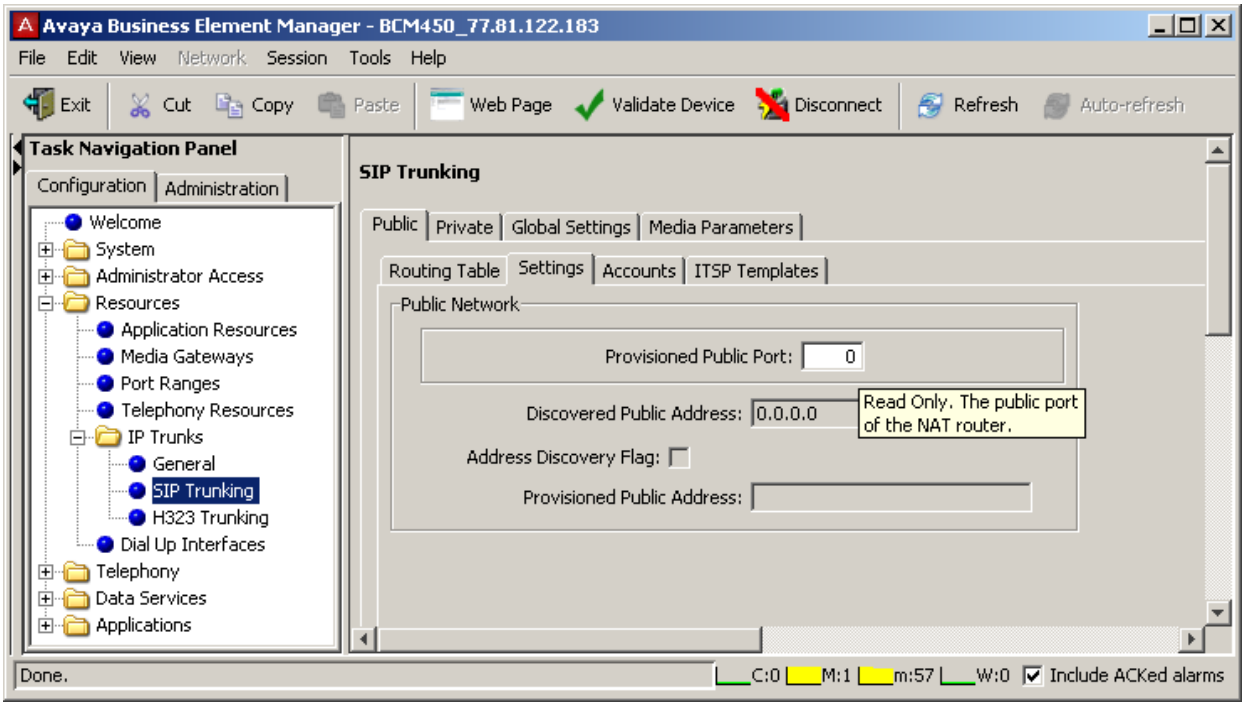
Make sure that the following parameters are enabled:

- **Support 100rel** - Flag indicating if BCM advertises support for 100Rel (PRACK) in the Supported header.
- **Allow Update** - Indicates if BCM advertises support for UPDATE in the Allow Header.
- **Use Null IP to hold** - Determines if 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.

Make sure that ‘Enable SDP Options query’ is disabled .If enabled, speech path issues can be encountered when hold call (issue tracked in IPOFFICE-28404).



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



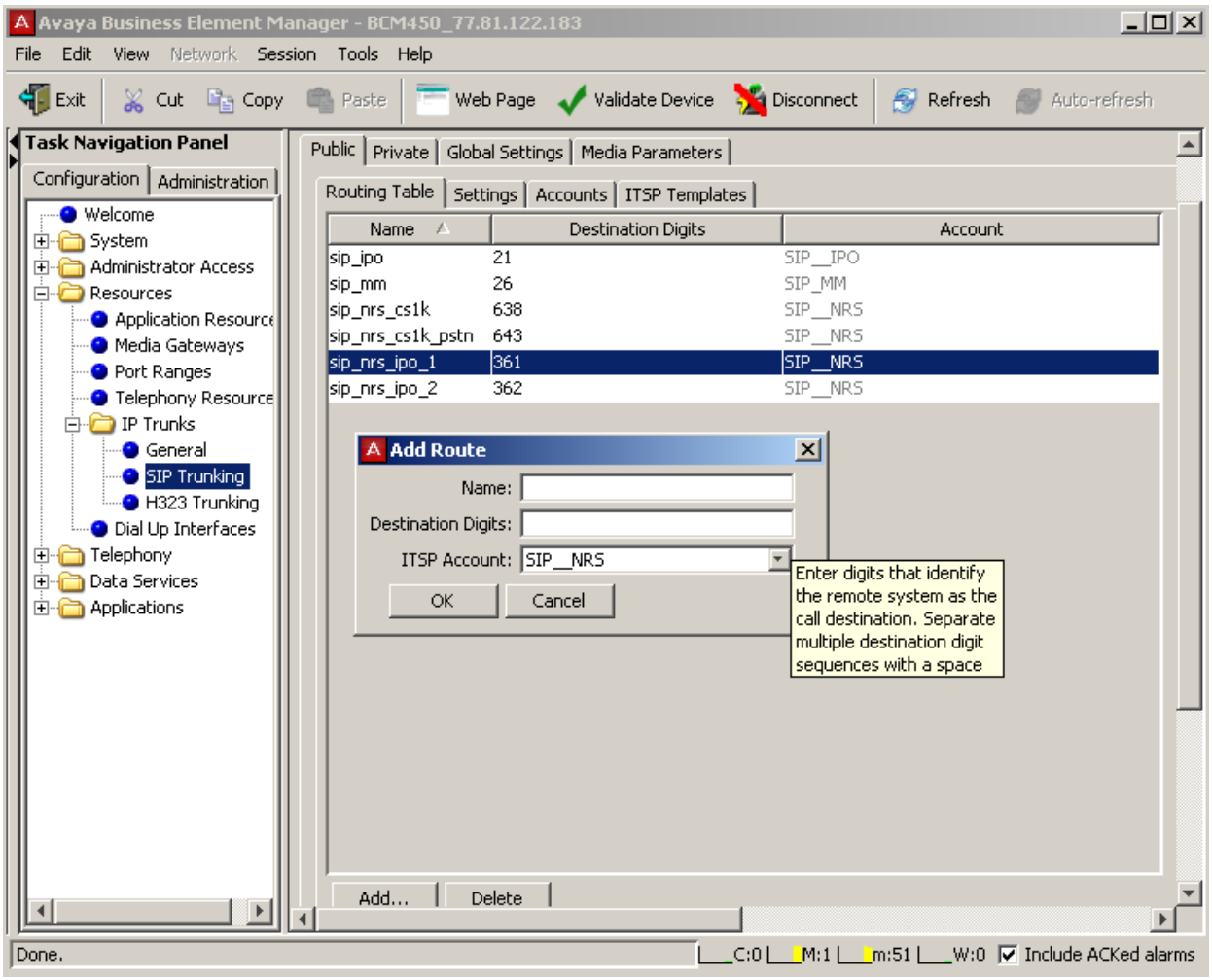
In the **SIP Trunking > Public > Routing Table** tab, click the Add button to add a public route. Enter the required information in the Add Route dialog box:

- 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 under **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.

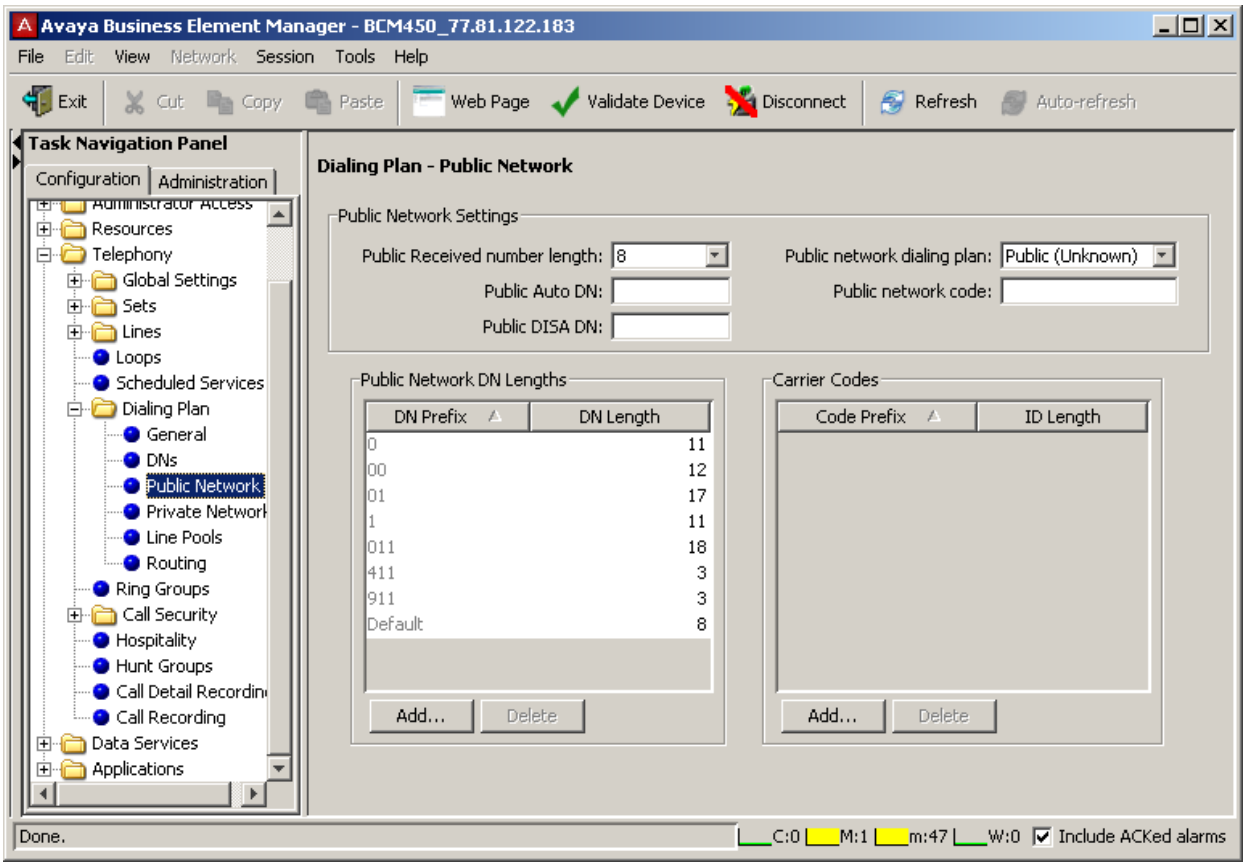
- Click OK to complete and close the Add Route dialog box.



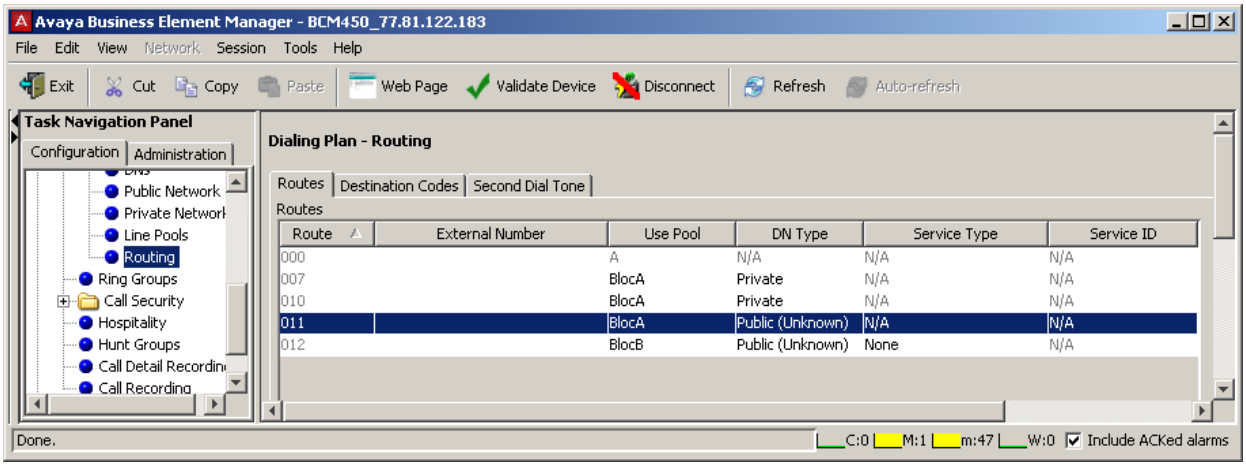
2.2.6 BCM Dialing Plan

Go to the **Telephony > Dialing Plan > Public Network** section:
This section provide the settings that allows the system to determine if an incoming call is meant for 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 numbering plan.
- **Public Network DN Lengths** – Should be set to Public (Unknown) for a variable length.
- **Public network code** – The number entered here concatenates with the Public OLI. It can be set according to 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.

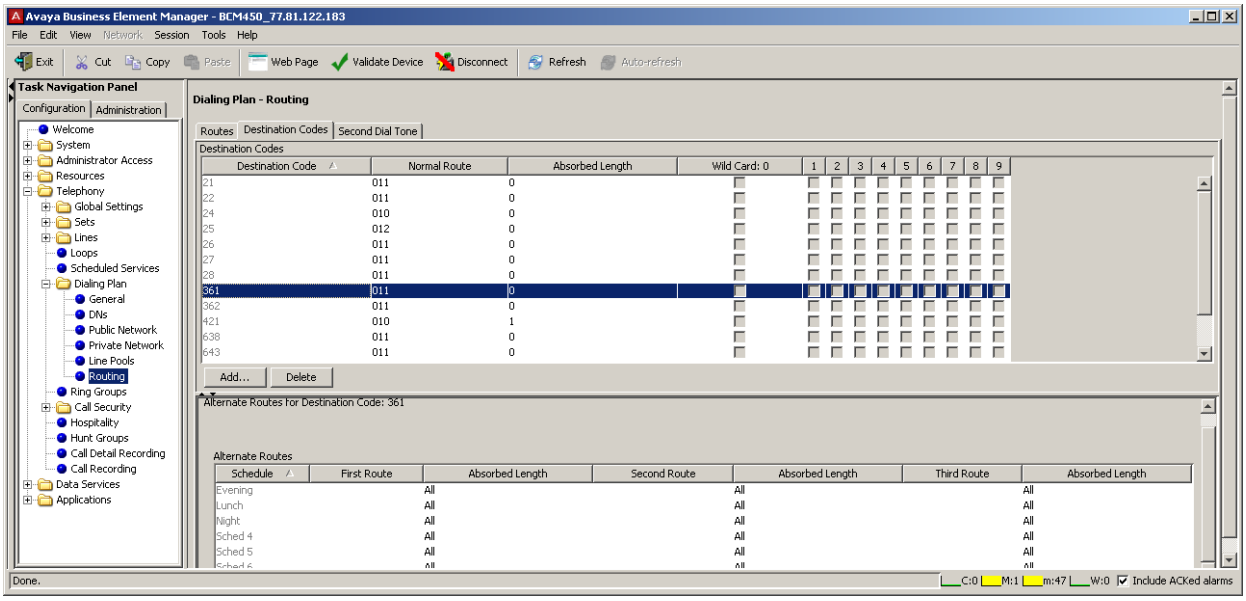
DN Prefix – This is the number (destination code) that must precede a dial string exiting the system to the public network.
DN Length - This number indicates how many numbers, starting from the front of the dial string, the system waits before sending to the public network.



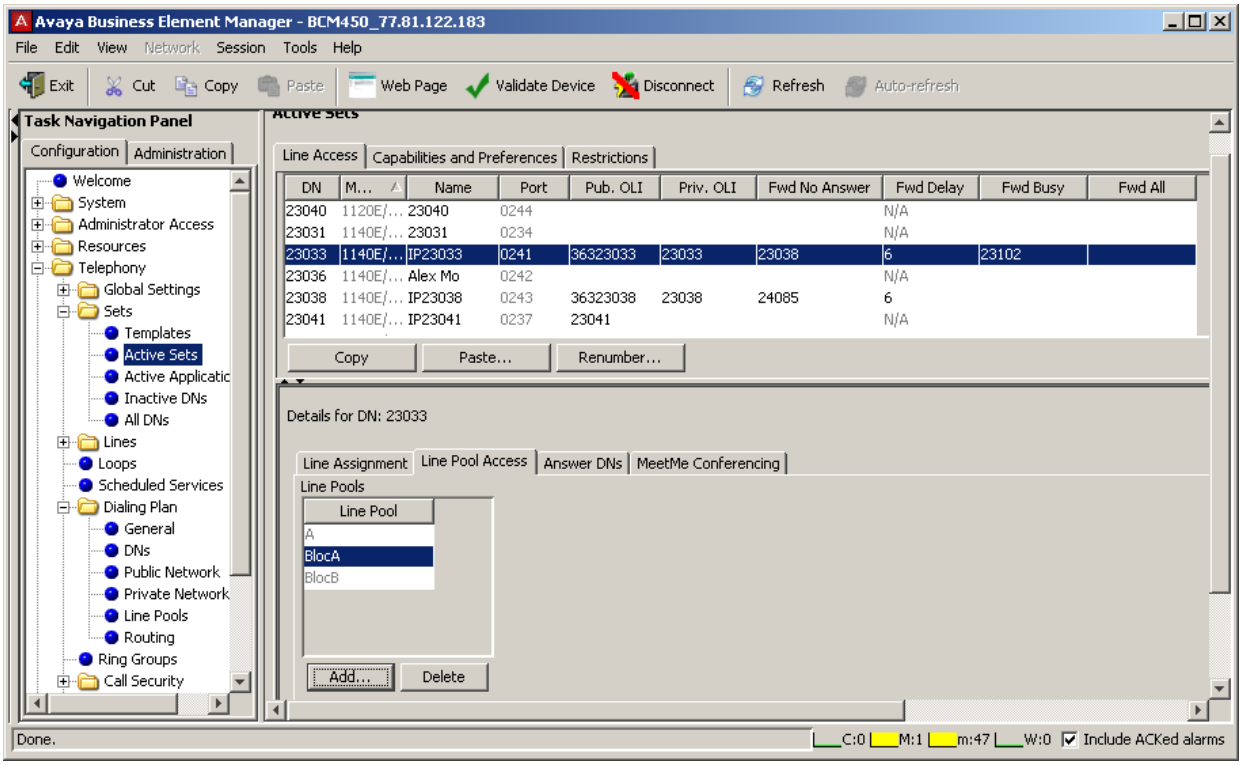
In **Telephony > Dialing Plan > Routing > Routes** tab.
This section gives the option configure the lines and loops to allow the users to dial out of the system.
A route can be used with more than one destination code, but a line pool should only be used with one route.
Click the Add button to add a new Route and fill the required fields:
- Route – This is the route number and it is unique. It can be chose 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 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 the 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 Route 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 which needs to be modified.
- Normal Route – Enter the route number created at previous step. This is the route that system uses when 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 network. Available values are **All, None, 1 – (X-1).** This should be set according to NRS configuration, meaning that BCM should send the digits that NRS 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.

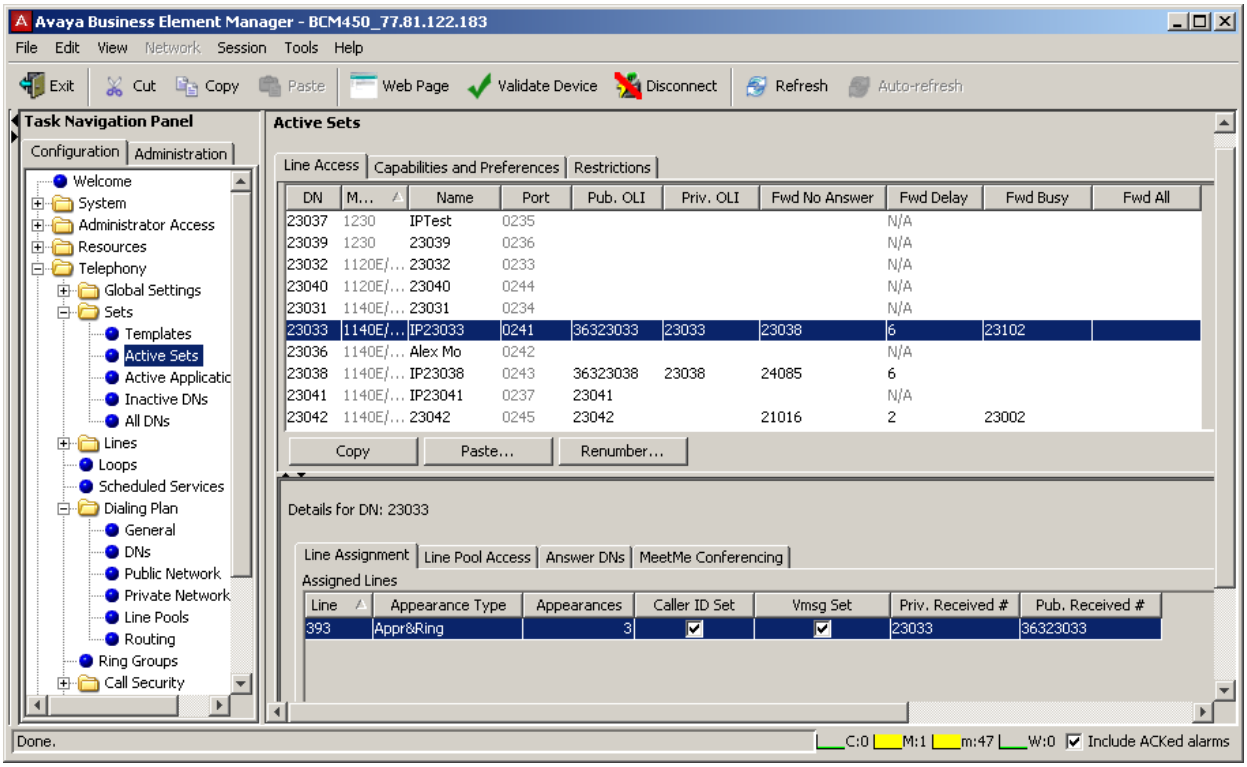


Phone configurations required to originate calls via SIP Trunk.
In the **Telephony > Sets > Active Sets** section**,** highlight a phone from the list and go to the **Line Access** tab:
- Line Pools Access – In order for a phone to be able to originate calls using VoIP lines, it must have access to the Line Pool on which VoIP lines belongs to. By Default, VoIP lines are mapped to **BlocA.** Using **Add** button, add, in 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 **Line Access** tab:
- Line Assignment: Using Add button, assign a **Target** line to selected phone in order for BCM system to be able to route incoming calls to this phone. The entire list of Target lines can be found under 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-mail system 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.

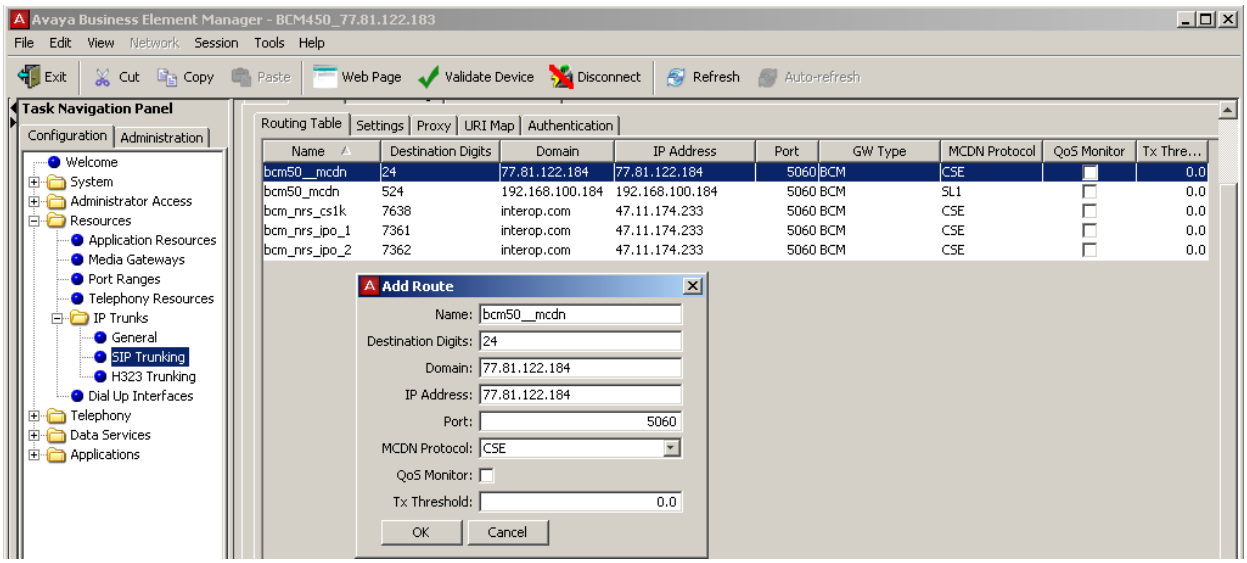
- Pub. Received # - Enter the public received number (DID) that the system will recognize as the target telephone. If the received digits match this number then BCM system will route the call to the corresponding target line and automatically the call will be presented to the mapped phone.



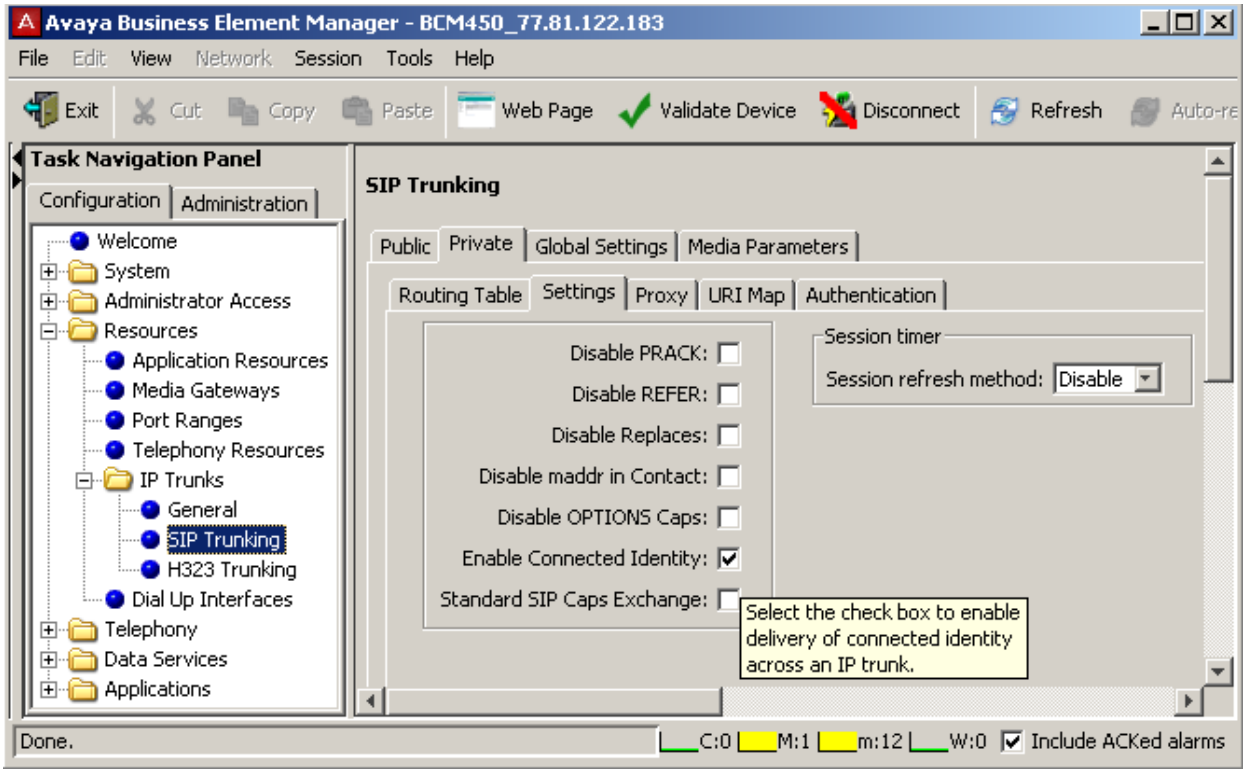
A brief description on how external calls are processed by the BCM system.
Originating calls:
- Based on the Destination Code entered, BCM gets the Route number (Telephony > Dialing Plan > Routing).
- Using the Route number, BCM decides which Pool is involved and if the calling phone has access to that 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 current setup, the BCM is looking 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 this match occurs, BCM will fallow the Route parameters to process the call.
Receiving calls:
- When receiving an external call, BCM is looking in Target Line list for a match between received digits and **Publ. Received #** or **Priv. Received #** fields.
- If this 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.
A MCDN keycode is required to activate this capability.
MCDN is supported for PRI and VoIP (H323 and SIP) trunks.
When PRI trunk is used then **SL-1** Protocol should be selected.
If connected with other BCM through a VoIP trunk then **CSE** protocol should be selected.
In our setup, 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 **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 chose.
- **Destination Digits** – Enter the destination digits for the remote system. It should be the same as Destination Code defined under 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 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 have 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 not part of the numbering plan. It is used in MCDN UDP private networks.

- Private network type – You can specify if 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 have between 3 to 14 digits.

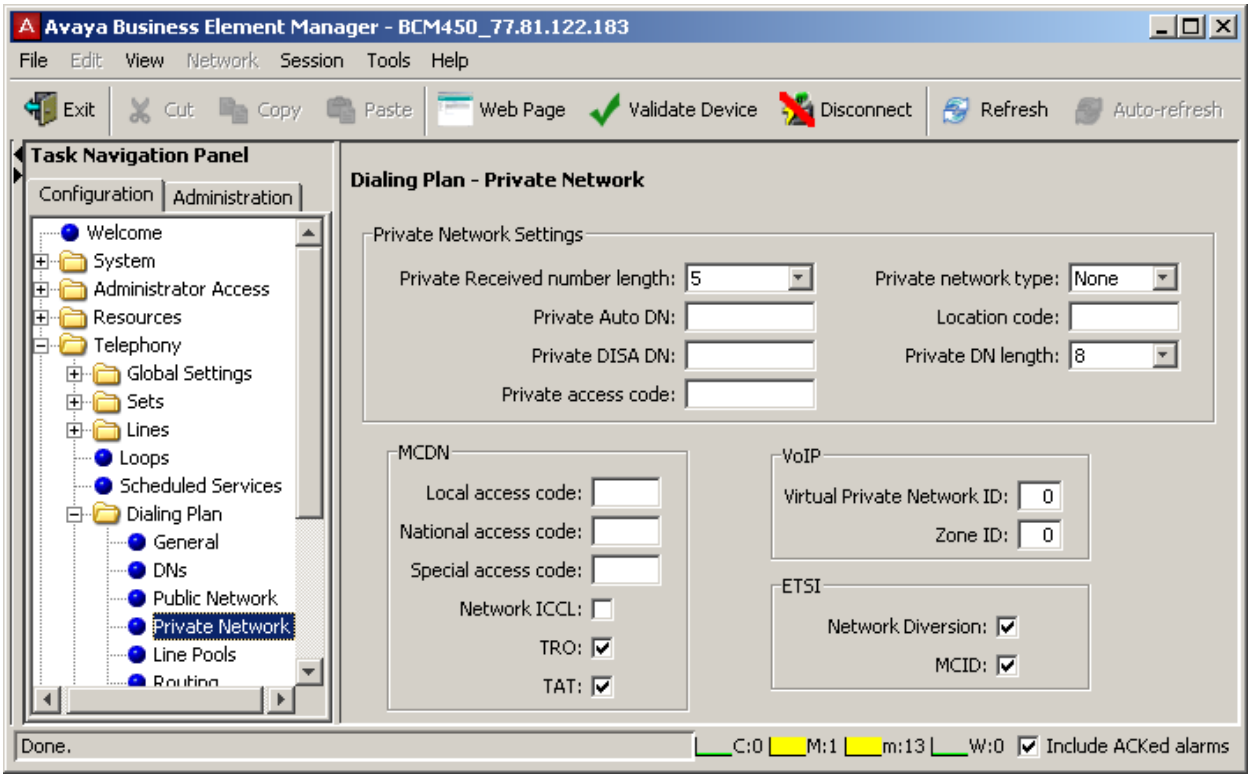
MCDN network values:

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

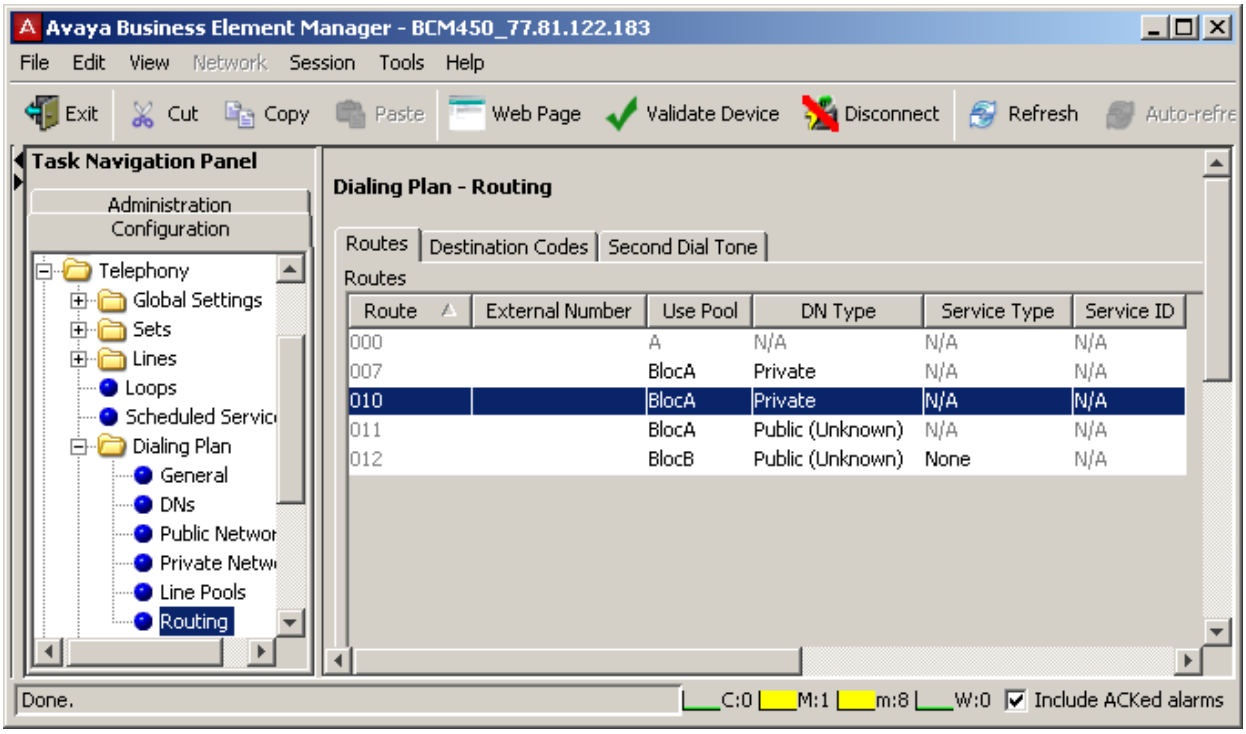
- Local access code – This number is prepended to an incoming E.164/ Local call.

- 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 Business Element Manager, Administration > Utilities.

2.3 Configuration for Avaya Communication Server 1000

This section provides the procedures for configuring CS1000 using Unified Communication Manager (UCM). UCM is a web based application that provides a central launch point for management facilities that oversee multiple network elements to manage the entire network.

The procedure covers the following areas:

- System Limits
- Feature Package
- Customer
- IP Telephony Node
- Bandwidth Zones
- SIP Trunk
- Dial Plan

2.3.1 System Limits

The keycode installed on the Call Server controls the maximum values for system attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales

representative to add additional feature/capacity. Use the Communication Server 1000E system terminal and manually load **Overlay 22** to print the **System Limits** (the required command is SLT), and verify there are sufficient Traditional Telephones, Traditional Trunks, IP Users, Basic IP Users and SIP Access Ports to meet requirements.

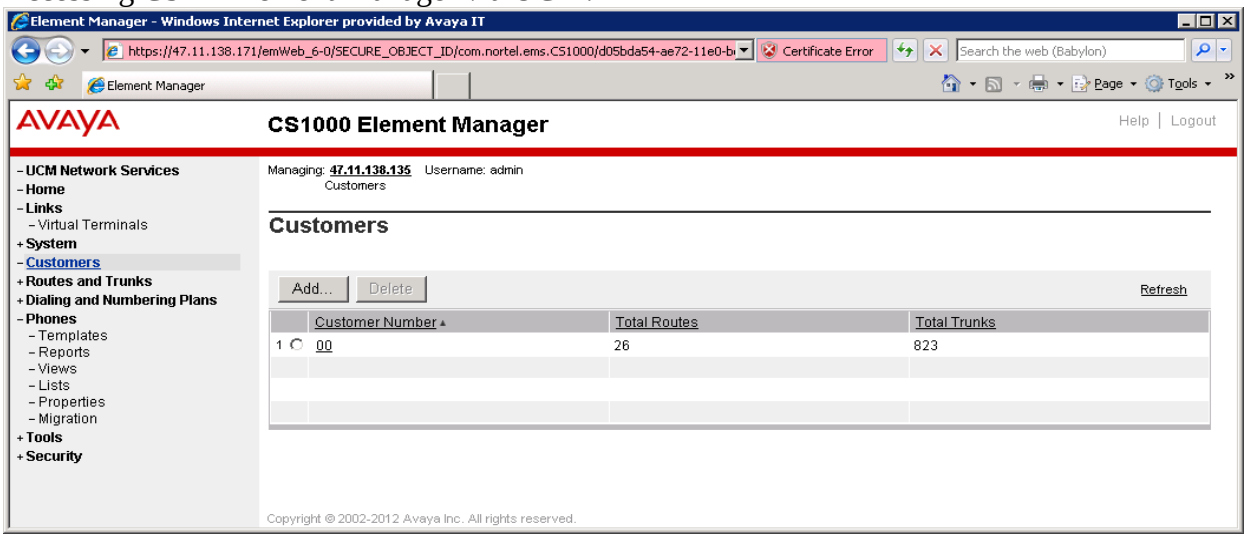
```
                    Overlay 22 - system Limits (SLT) Printout

System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz
IPMGs Registered:              2IPMGs Unregistered:
Configured/unregistered:  0

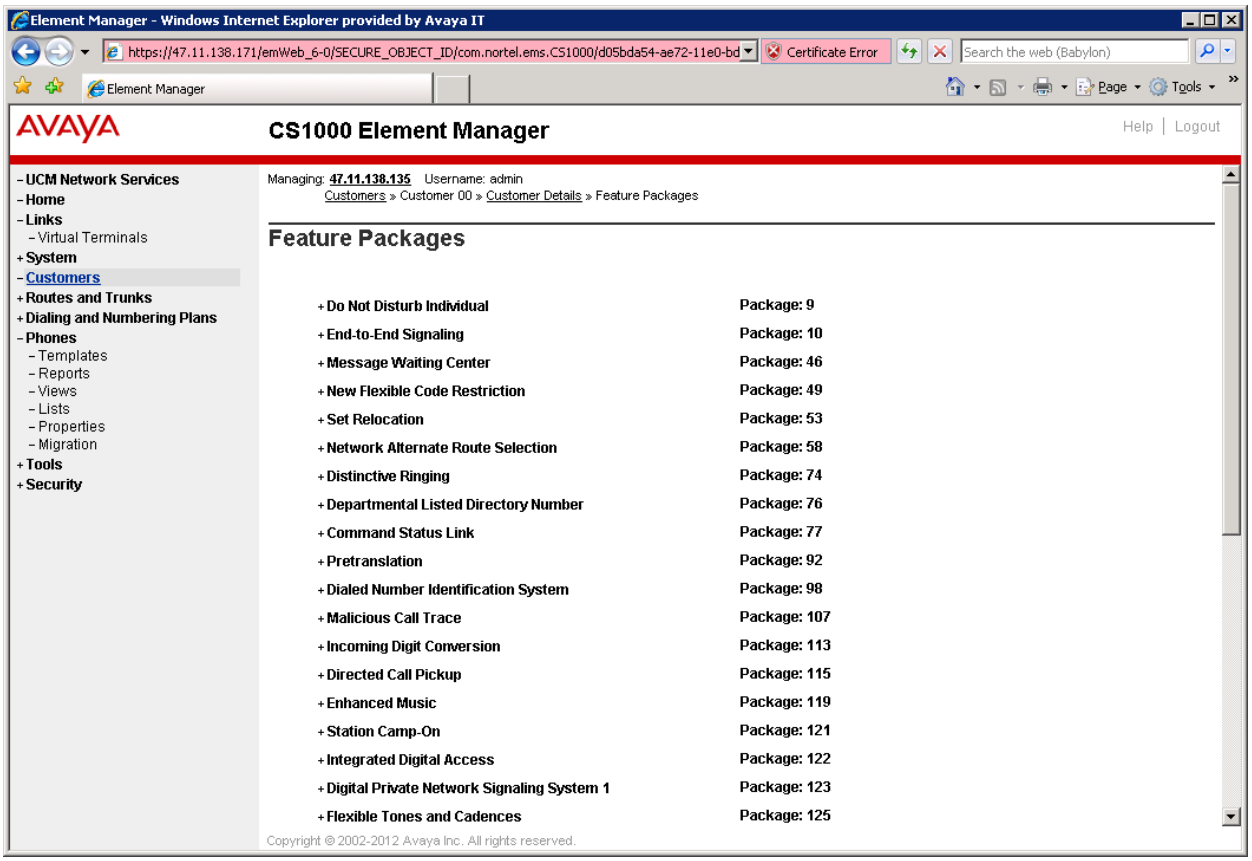
TRADITIONAL TELEPHONES 32767    LEFT 32764    USED     9
TRADITIONAL TRUNKS     32767    LEFT 32677    USED    90
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32744    USED    12
BASIC IP USERS         32767    LEFT 32761    USED     7
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32752    USED     2
MOBILE EXTENSIONS      32767    LEFT 32767    USED     0
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
NORTEL SIP LINES       32767    LEFT 32765    USED     2
THIRD PARTY SIP LINES  32767    LEFT 32761    USED     6
SIP CONVERGED DESKTOPS 32767    LEFT 32767    USED     0
SIP CTI TR87           32767    LEFT 32767    USED     0
SIP ACCESS PORTS       32767    LEFT 32752    USED    30
```

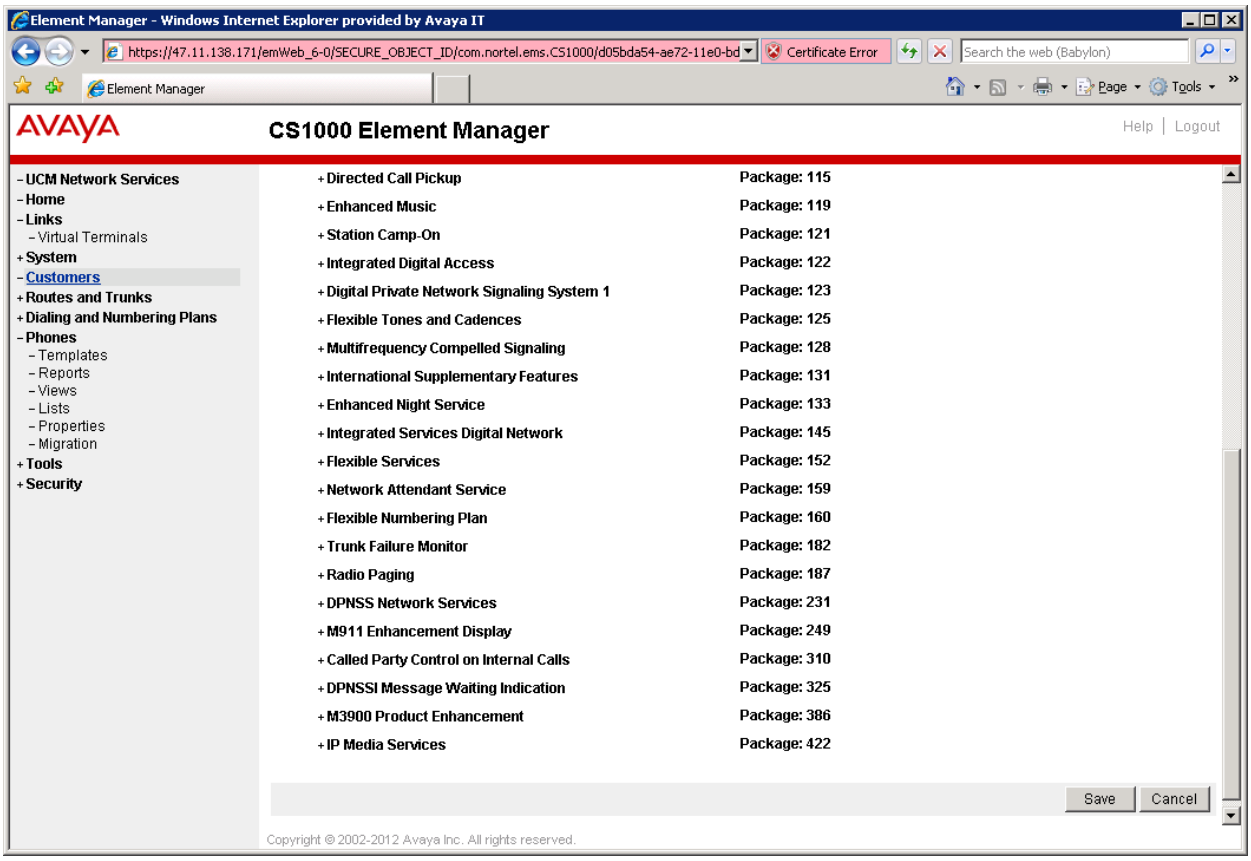
2.3.2 Feature Packages

Accessing CS1K Element Manager via UCM:



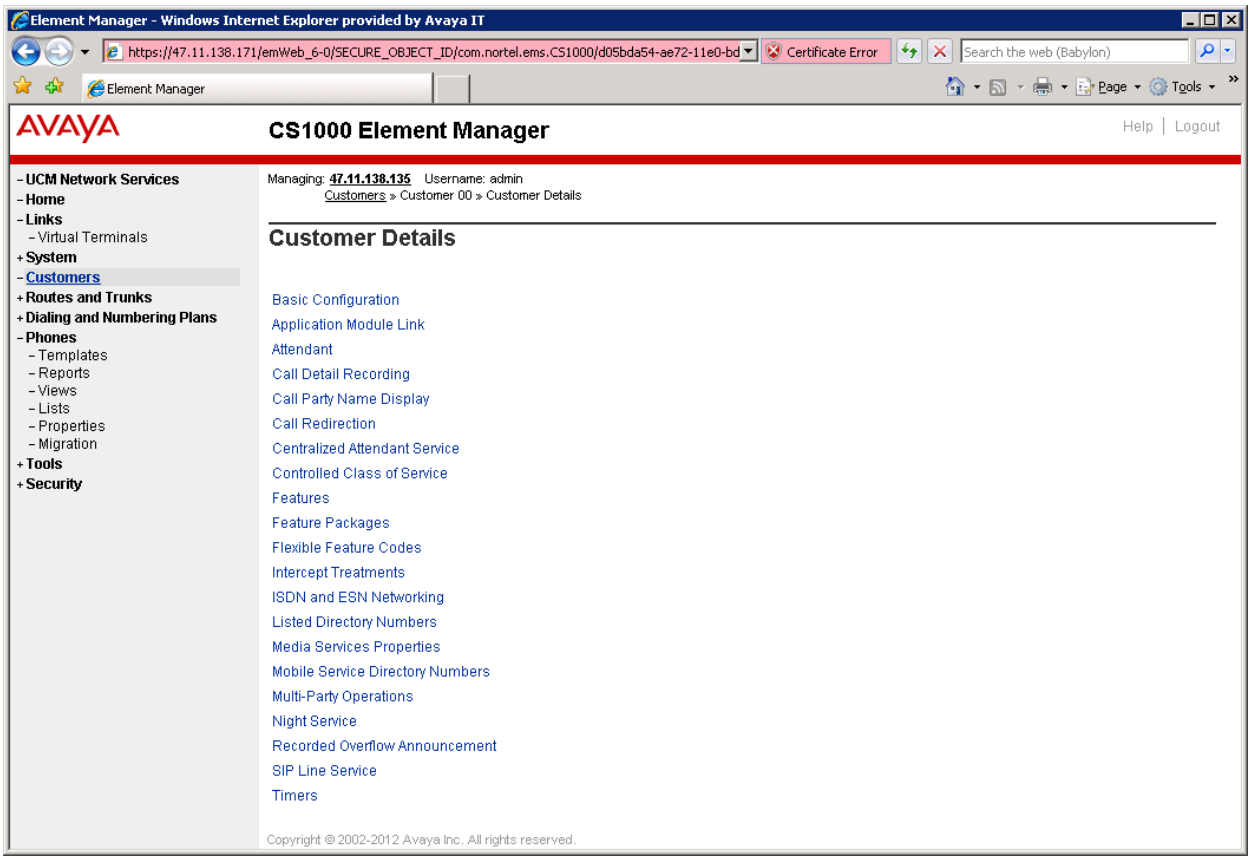
From Element Manager, go to **Customer > Feature Packages** to consult the licensed feature list.


Element Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.138.171/emWeb_6-0/SECURE_OBJECT_ID/com.nortel.ems.CS1000/d05bda54-ae72-11e0-bd
Certificate Error
Search the web (Babylon)
Element Manager
Page
Tools
AVAYA
CS1000 Element Manager
Help | Logout
- UCM Network Services
- Home
- Links
- Virtual Terminals
+ System
- Customers
+ Routes and Trunks
+ Dialing and Numbering Plans
- Phones
- Templates
- Reports
- Views
- Lists
- Properties
- Migration
+ Tools
+ Security
Managing: 47.11.138.135 Username: admin
Customers » Customer 00 » Customer Details » Feature Packages
Feature Packages
+ Do Not Disturb Individual Package: 9
+ End-to-End Signaling Package: 10
+ Message Waiting Center Package: 46
+ New Flexible Code Restriction Package: 49
+ Set Relocation Package: 53
+ Network Alternate Route Selection Package: 58
+ Distinctive Ringing Package: 74
+ Departmental Listed Directory Number Package: 76
+ Command Status Link Package: 77
+ Pretranslation Package: 92
+ Dialed Number Identification System Package: 98
+ Malicious Call Trace Package: 107
+ Incoming Digit Conversion Package: 113
+ Directed Call Pickup Package: 115
+ Enhanced Music Package: 119
+ Station Camp-On Package: 121
+ Integrated Digital Access Package: 122
+ Digital Private Network Signaling System 1 Package: 123
+ Flexible Tones and Cadences Package: 125
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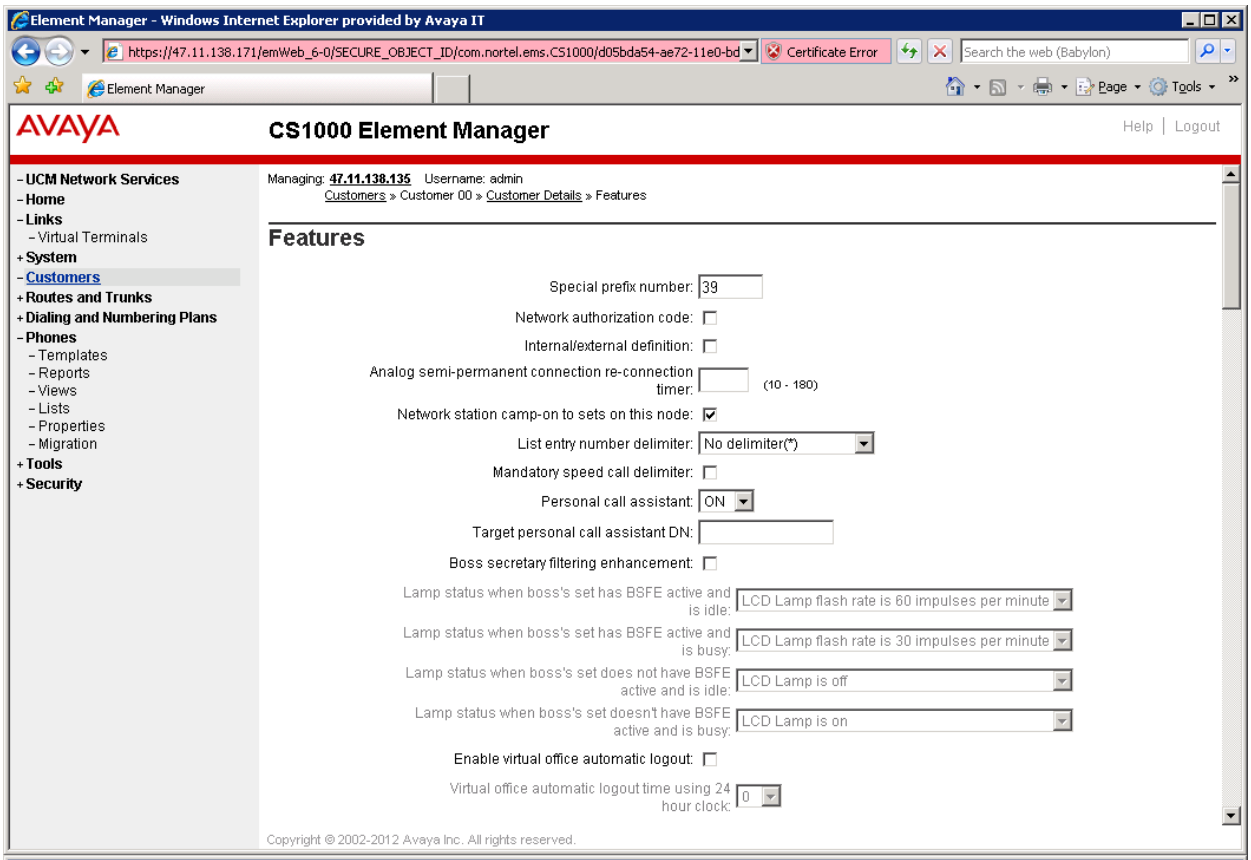


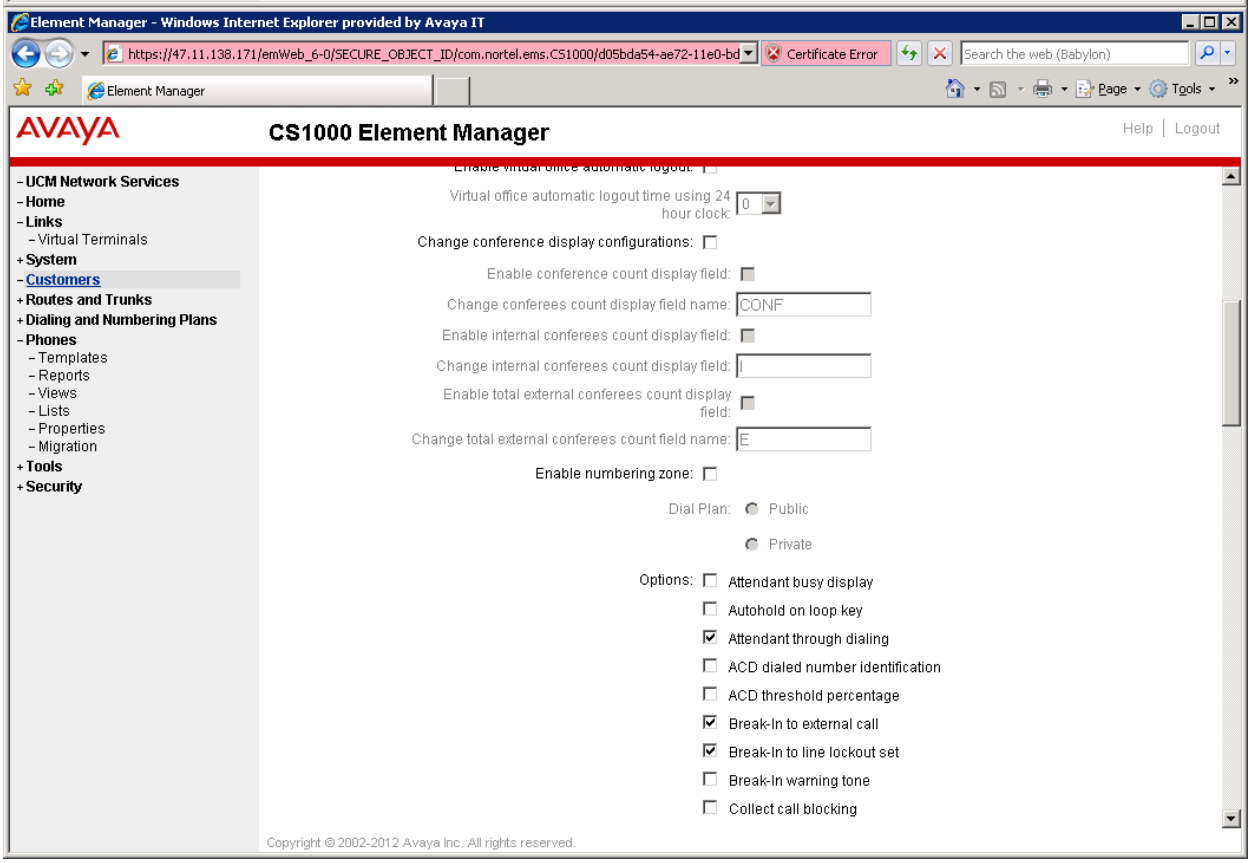
2.3.3 Customers

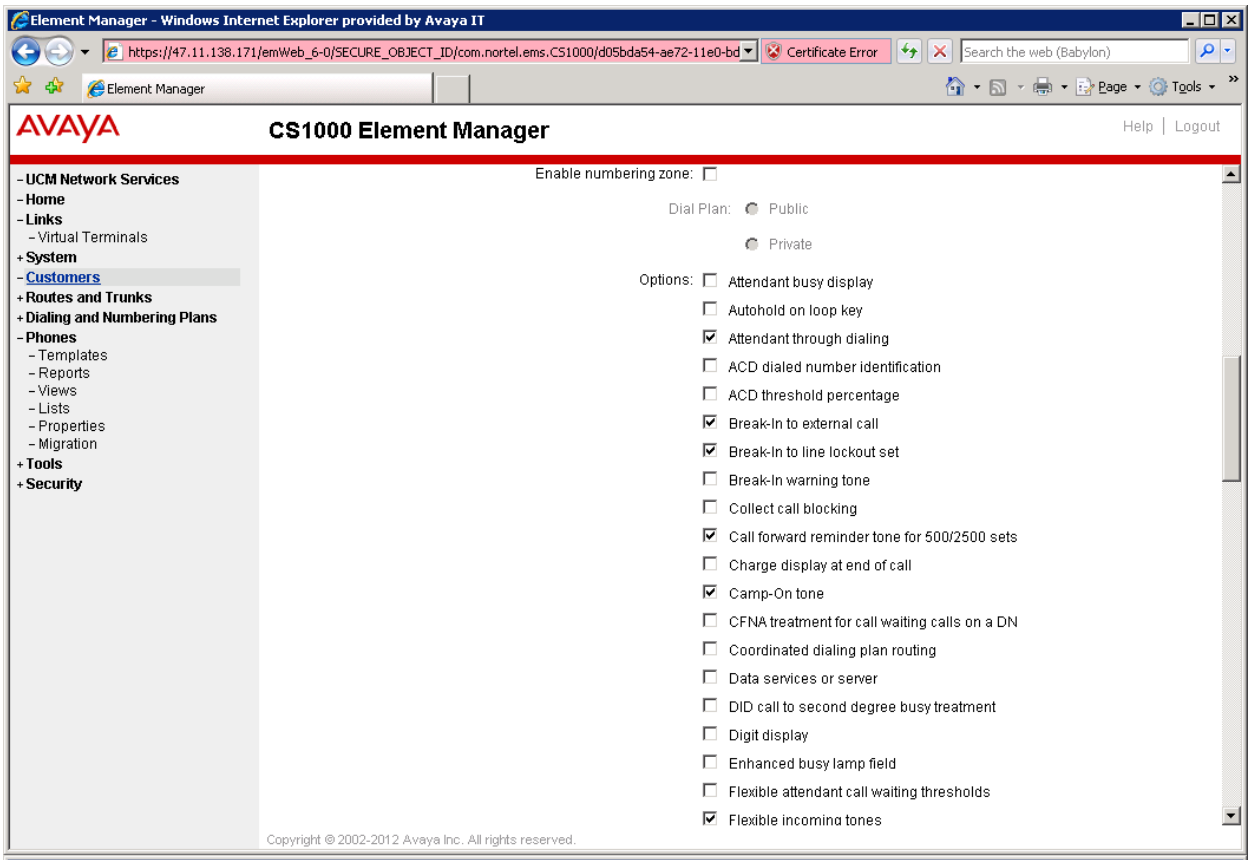
Go to **Customers** and select the appropriate Customer Number to configure and edit the data related to customer. In our case only one customer is defined. The customer field is used to separate the configurations related to different customers.

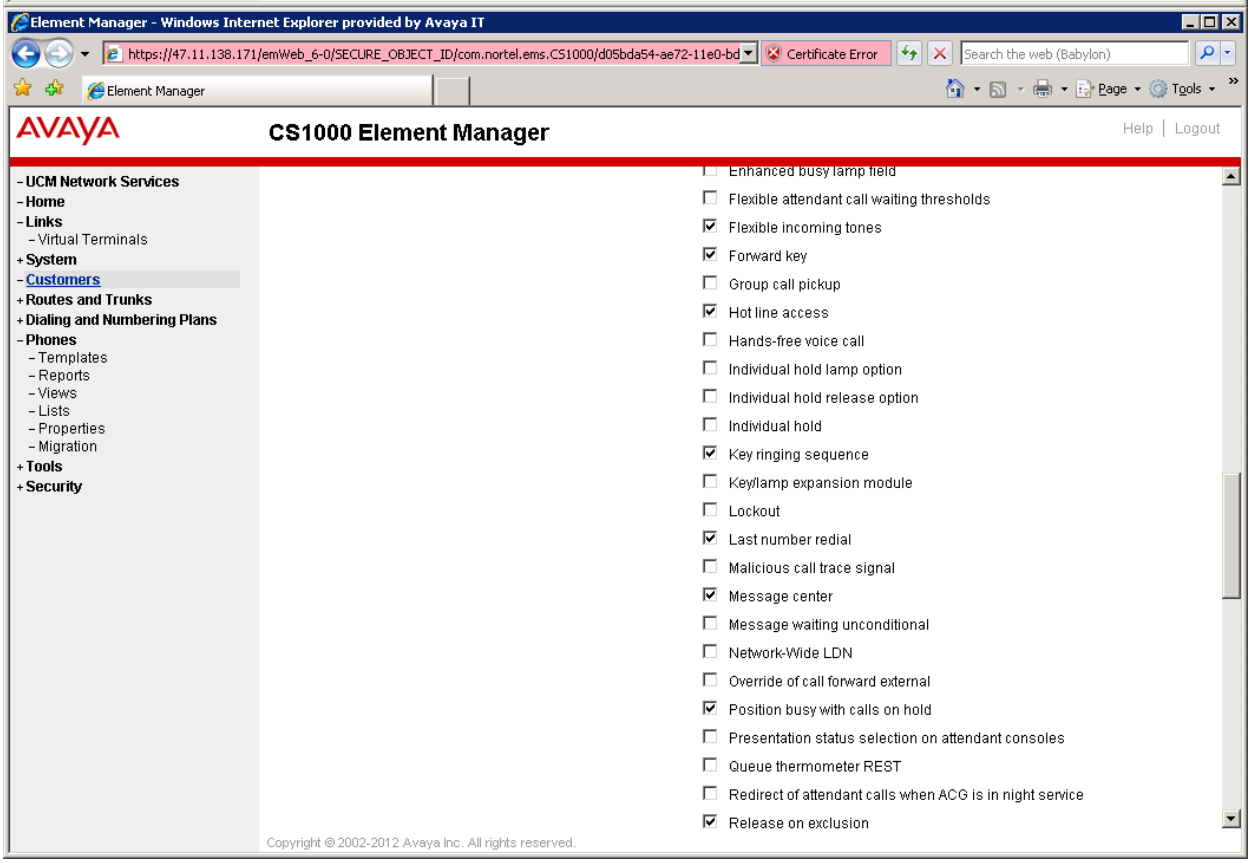


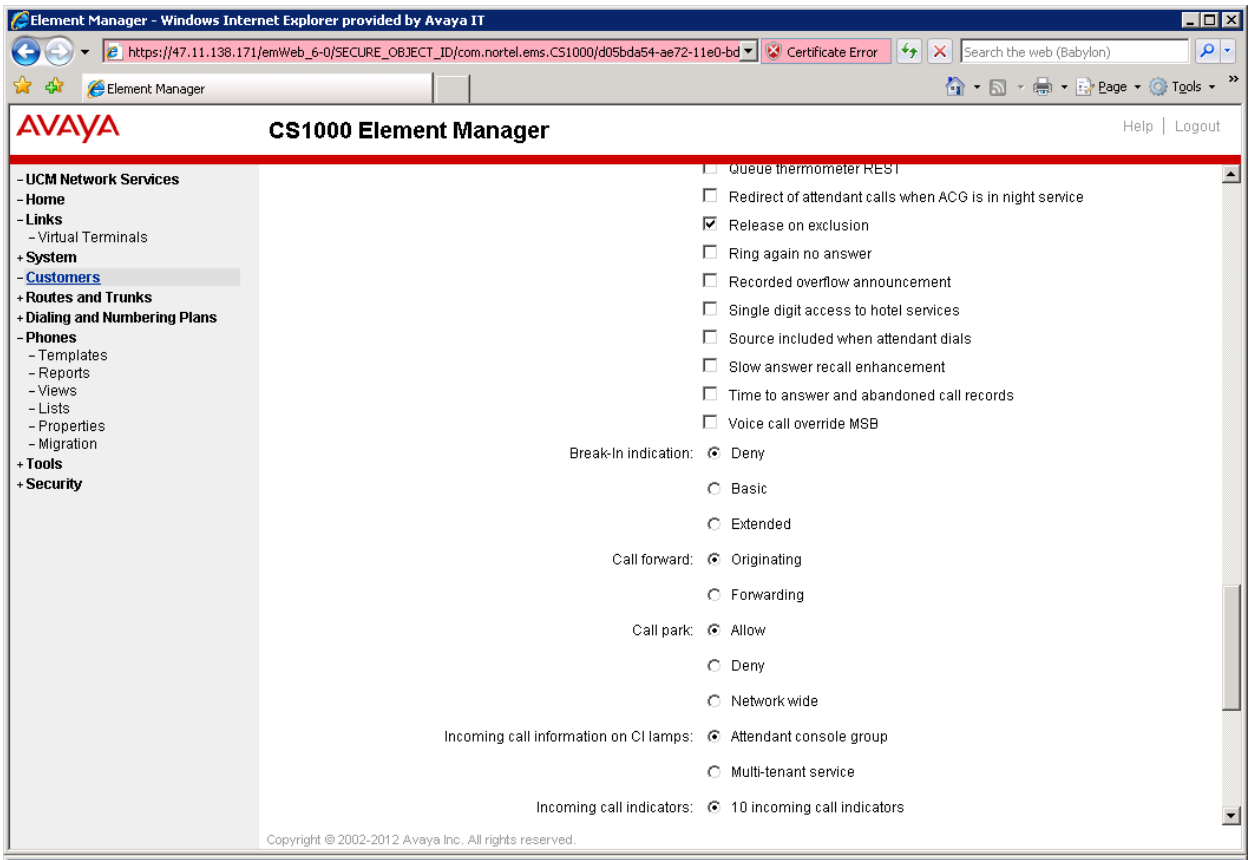
Select **Feature** from the list to access the Customer Features :
- **Call forwarding** – Make sure it is set to Originating. This parameter is used to decide if either the Originating or Forwarding party's Class of Service is used to determine access to services or features on Call Forward.
- Retain default values for all other parameters.

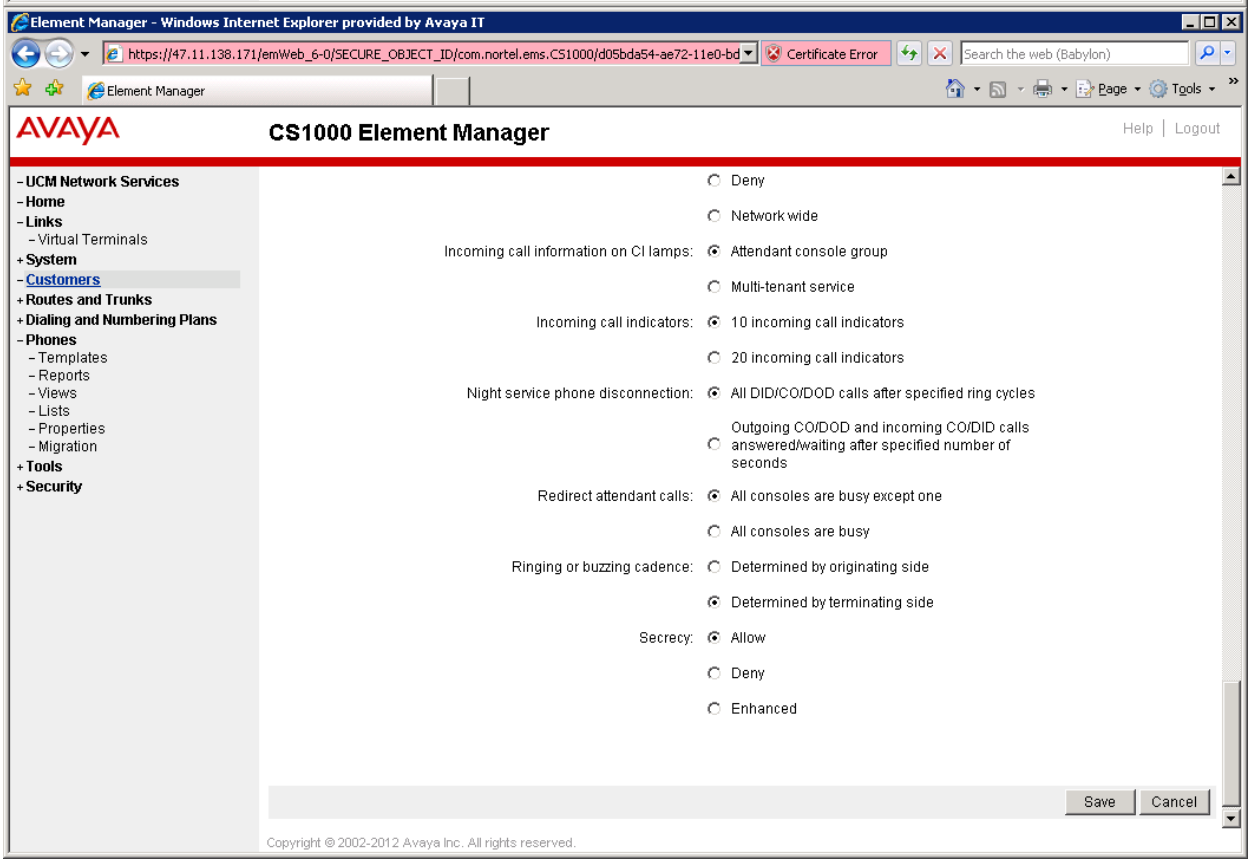

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Managing: 47.11.138.135 Username: admin
Customers » Customer 00 » Customer Details » Features
Features
Special prefix number: 39
Network authorization code:
Internal/external definition:
Analog semi-permanent connection re-connection timer: (10 - 180)
Network station camp-on to sets on this node:
List entry number delimiter: No delimiter(*)
Mandatory speed call delimiter:
Personal call assistant: ON
Target personal call assistant DN:
Boss secretary filtering enhancement:
Lamp status when boss's set has BSFE active and is idle: LCD Lamp flash rate is 60 impulses per minute
Lamp status when boss's set has BSFE active and is busy: LCD Lamp flash rate is 30 impulses per minute
Lamp status when boss's set does not have BSFE active and is idle: LCD Lamp is off
Lamp status when boss's set doesn't have BSFE active and is busy: LCD Lamp is on
Enable virtual office automatic logout:
Virtual office automatic logout time using 24 hour clock: 0
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Virtual office automatic logout time using 24 hour clock: 0
Change conference display configurations:
Enable conference count display field:
Change conferees count display field name: CONF
Enable internal conferees count display field:
Change internal conferees count display field: I
Enable total external conferees count display field:
Change total external conferees count field name: E
Enable numbering zone:
Dial Plan: Public
Private
Options: Attendant busy display
Autohold on loop key
Attendant through dialing
ACD dialed number identification
ACD threshold percentage
Break-In to external call
Break-In to line lockout set
Break-In warning tone
Collect call blocking
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Enable numbering zone:
Dial Plan: Public
Private
Options:
Attendant busy display
Autohold on loop key
Attendant through dialing
ACD dialed number identification
ACD threshold percentage
Break-In to external call
Break-In to line lockout set
Break-In warning tone
Collect call blocking
Call forward reminder tone for 500/2500 sets
Charge display at end of call
Camp-On tone
CFNA treatment for call waiting calls on a DN
Coordinated dialing plan routing
Data services or server
DID call to second degree busy treatment
Digit display
Enhanced busy lamp field
Flexible attendant call waiting thresholds
Flexible incoming tones
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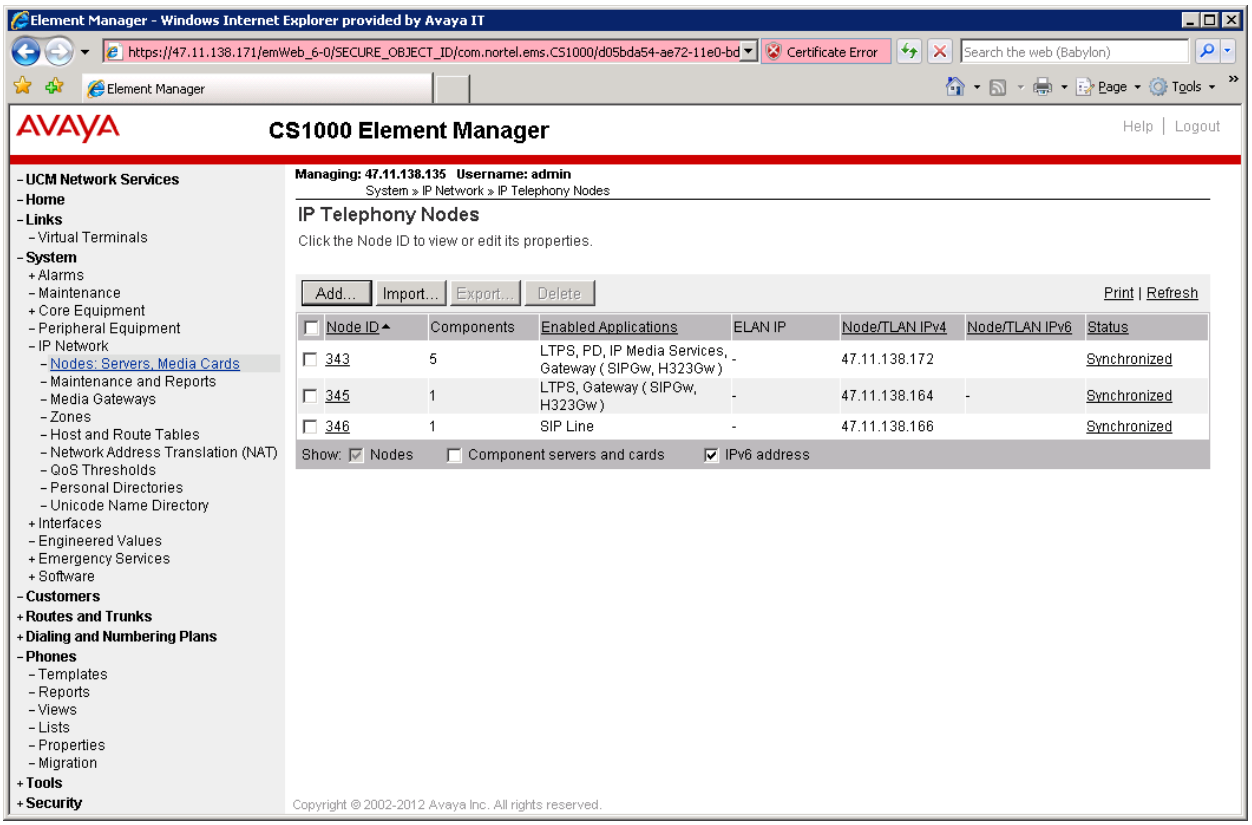

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Enhanced busy lamp field
Flexible attendant call waiting thresholds
Flexible incoming tones
Forward key
Group call pickup
Hot line access
Hands-free voice call
Individual hold lamp option
Individual hold release option
Individual hold
Key ringing sequence
Key/lamp expansion module
Lockout
Last number redial
Malicious call trace signal
Message center
Message waiting unconditional
Network-Wide LDN
Override of call forward external
Position busy with calls on hold
Presentation status selection on attendant consoles
Queue thermometer REST
Redirect of attendant calls when ACG is in night service
Release on exclusion
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Queue thermometer REST
Redirect of attendant calls when ACG is in night service
Release on exclusion
Ring again no answer
Recorded overflow announcement
Single digit access to hotel services
Source included when attendant dials
Slow answer recall enhancement
Time to answer and abandoned call records
Voice call override MSB
Break-In indication: Deny
Basic
Extended
Call forward: Originating
Forwarding
Call park: Allow
Deny
Network wide
Incoming call information on CI lamps: Attendant console group
Multi-tenant service
Incoming call indicators: 10 incoming call indicators
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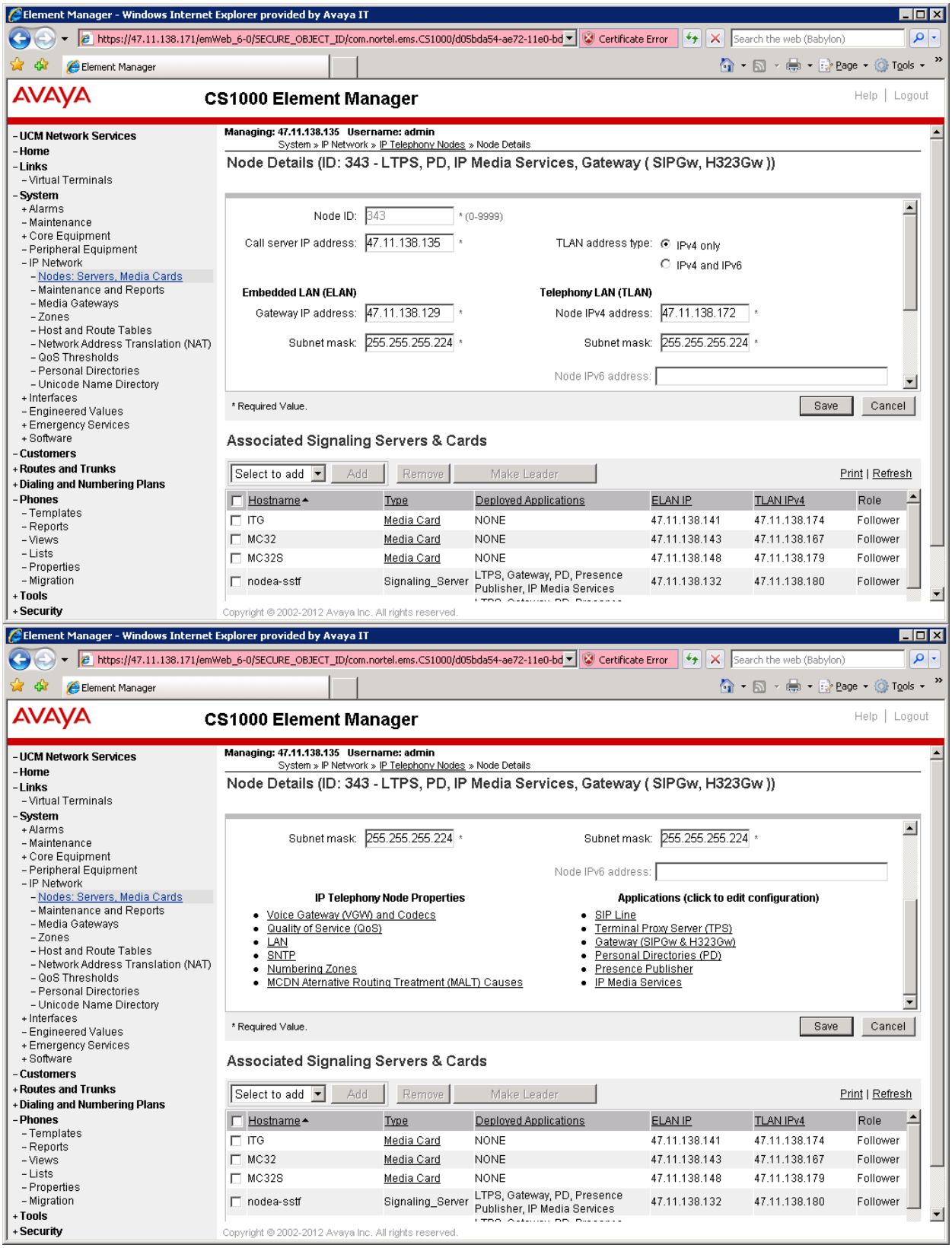

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Deny
Network wide
Incoming call information on CI lamps: Attendant console group
Multi-tenant service
Incoming call indicators: 10 incoming call indicators
20 incoming call indicators
Night service phone disconnection: All DID/CO/DOD calls after specified ring cycles
Outgoing CO/DOD and incoming CO/DID calls answered/waiting after specified number of seconds
Redirect attendant calls: All consoles are busy except one
All consoles are busy
Ringing or buzzing cadence: Determined by originating side
Determined by terminating side
Secrecy: Allow
Deny
Enhanced
Save
Cancel
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2.3.4 IP Telephony Nodes

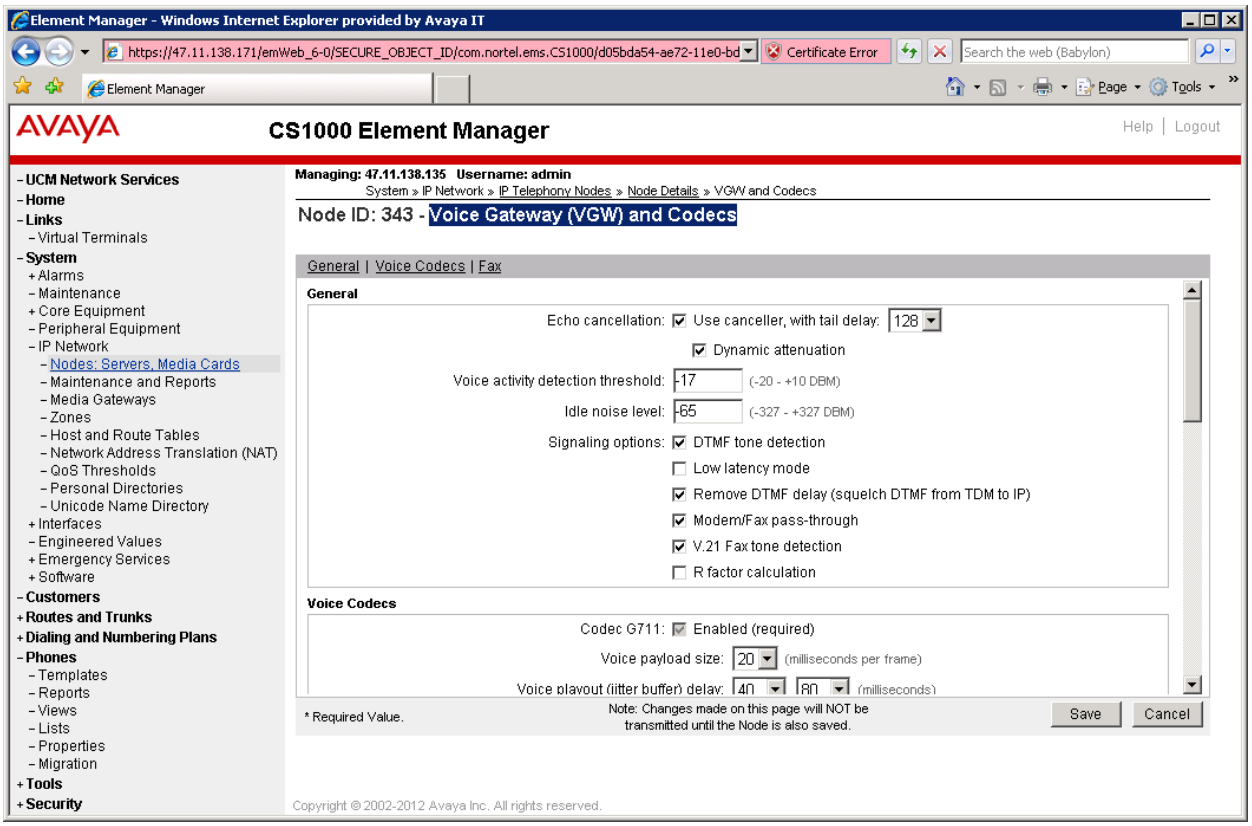
Go to **System -> IP Network -> Nodes: Servers, Media Cards** to view the configured IP Telephony Nodes.

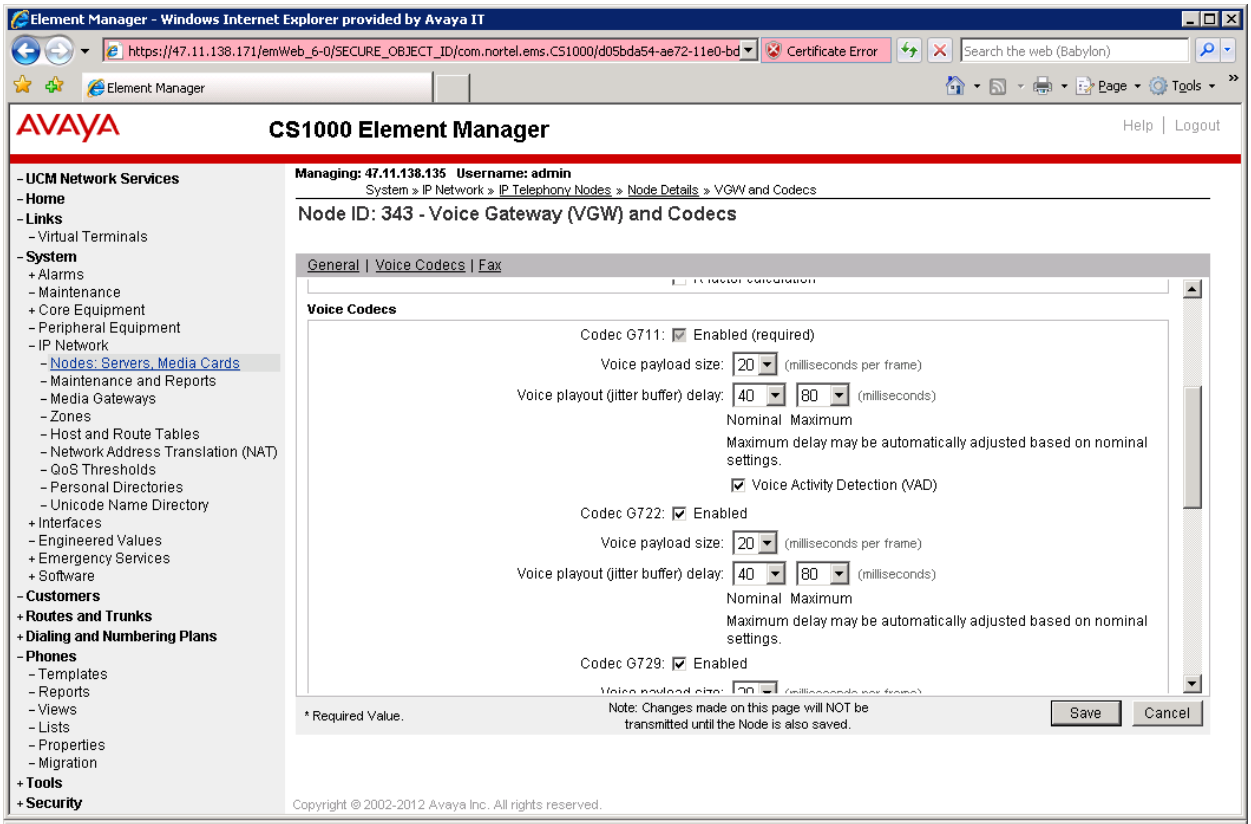


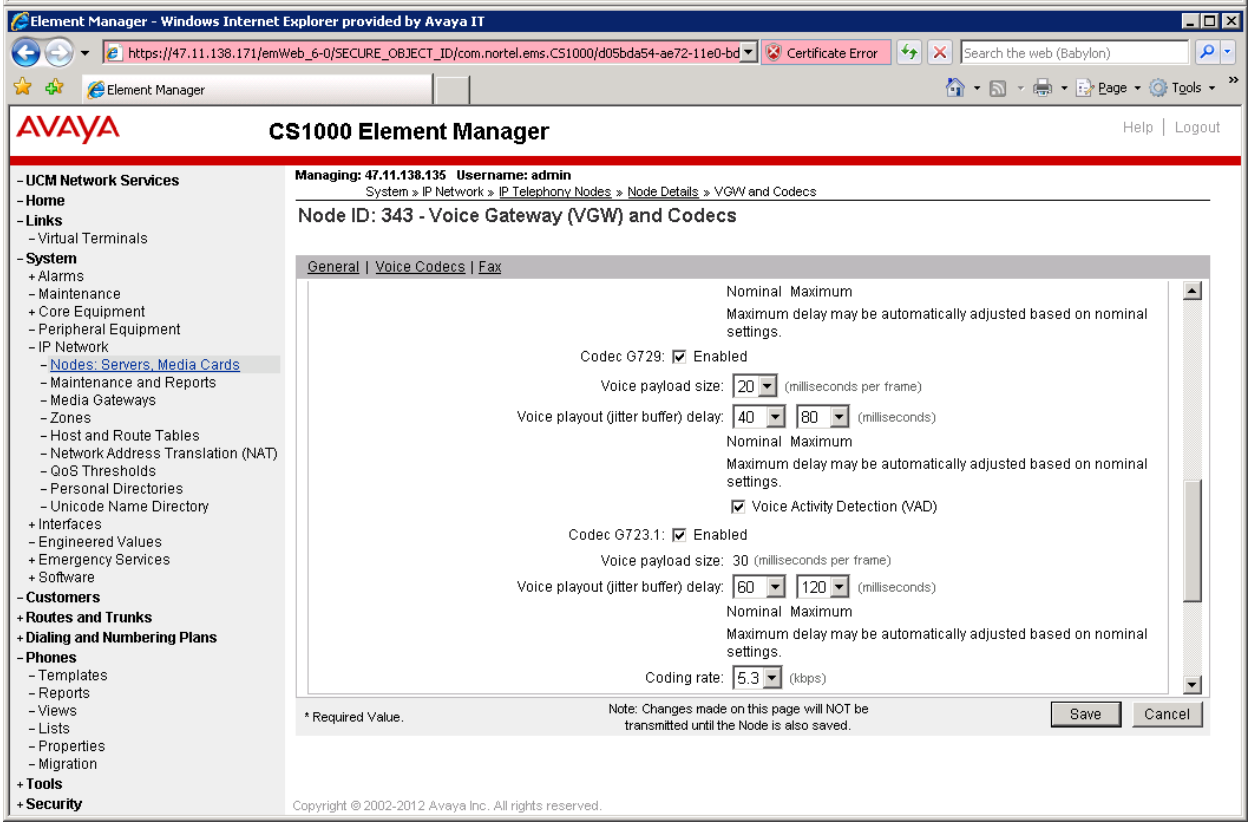
Assuming that an IP Telephony Node is already created, select the apropriate node to access its Configuration.
Take a note of the configured parameters :
- **Call server IP address** – the IP Address of the Call Server .
- **ELAN** – Embeded LAN IP Address – this is used for signaling and management
- **TLAN** – Telephony LAN IP Address – this is used for telephony signaling and data .
- **IP Telephony Node Properties** – Node parameters can be accessed and programmed from here.
- **Application** –Applications handled by this node can be accessd and programmed from here .
- **Associated Signaling Servers & Cards** – It lists the signaling servers and media cards associated to this node.

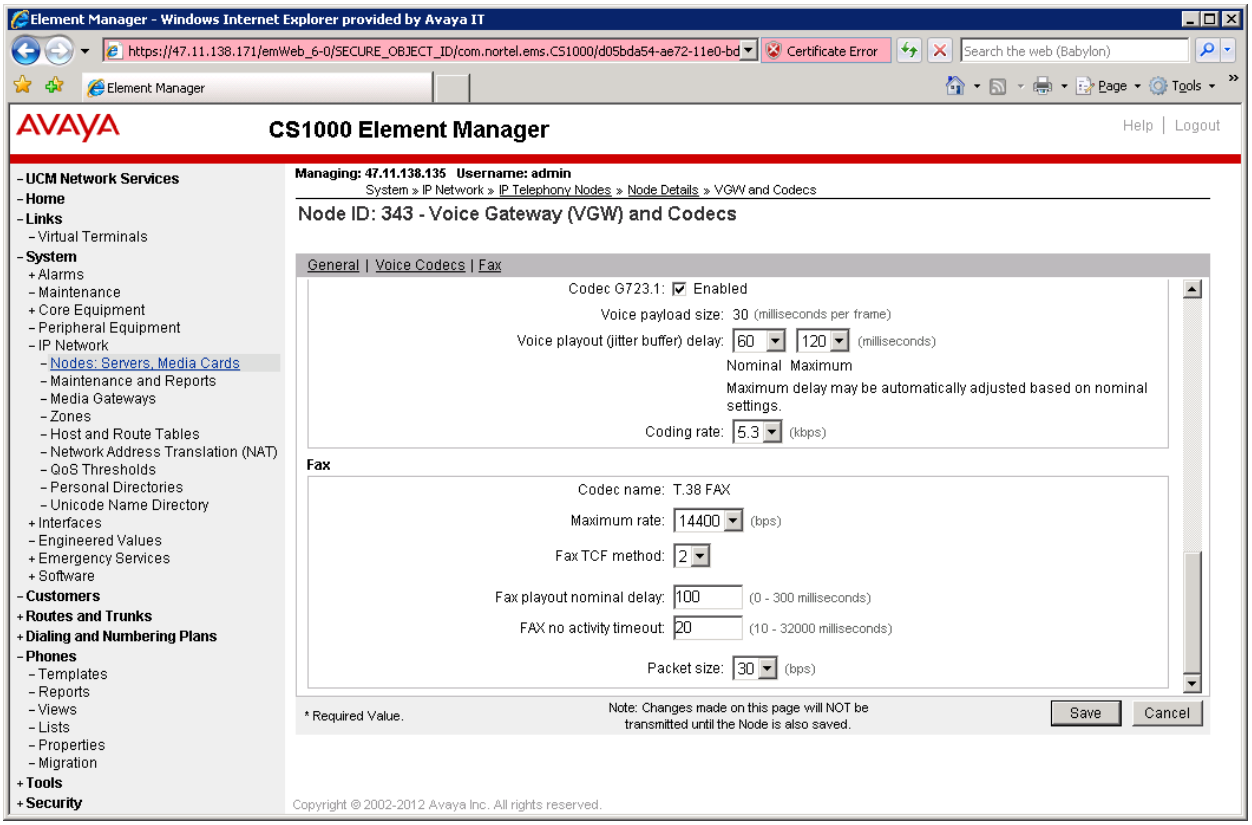

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Managing: 47.11.138.135 Username: admin
System » IP Network » IP Telephony Nodes » Node Details
Node Details (ID: 343 - LTPS, PD, IP Media Services, Gateway (SIPGw, H323Gw))
Node ID: 343 * (0-9999)
Call server IP address: 47.11.138.135 *
TLAN address type: IPv4 only
IPv4 and IPv6
Embedded LAN (ELAN)
Gateway IP address: 47.11.138.129 *
Subnet mask: 255.255.255.224 *
Telephony LAN (TLAN)
Node IPv4 address: 47.11.138.172 *
Subnet mask: 255.255.255.224 *
Node IPv6 address:
* Required Value.
Save
Cancel
Associated Signaling Servers & Cards
Select to add
Add
Remove
Make Leader
Print | Refresh
Hostname
Type
Deployed Applications
ELAN IP
TLAN IPv4
Role
ITG Media Card NONE 47.11.138.141 47.11.138.174 Follower
MC32 Media Card NONE 47.11.138.143 47.11.138.167 Follower
MC32S Media Card NONE 47.11.138.148 47.11.138.179 Follower
nodea-sstf Signaling_Server LTPS, Gateway, PD, Presence Publisher, IP Media Services 47.11.138.132 47.11.138.180 Follower
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Managing: 47.11.138.135 Username: admin
System » IP Network » IP Telephony Nodes » Node Details
Node Details (ID: 343 - LTPS, PD, IP Media Services, Gateway (SIPGw, H323Gw))
Subnet mask: 255.255.255.224 *
Subnet mask: 255.255.255.224 *
Node IPv6 address:
IP Telephony Node Properties
Voice Gateway (VGW) and Codecs
Quality of Service (QoS)
LAN
SNTP
Numbering Zones
MCDN Alternative Routing Treatment (MALT) Causes
Applications (click to edit configuration)
SIP Line
Terminal Proxy Server (TPS)
Gateway (SIPGw & H323Gw)
Personal Directories (PD)
Presence Publisher
IP Media Services
* Required Value.
Save
Cancel
Associated Signaling Servers & Cards
Select to add
Add
Remove
Make Leader
Print | Refresh
Hostname
Type
Deployed Applications
ELAN IP
TLAN IPv4
Role
ITG Media Card NONE 47.11.138.141 47.11.138.174 Follower
MC32 Media Card NONE 47.11.138.143 47.11.138.167 Follower
MC32S Media Card NONE 47.11.138.148 47.11.138.179 Follower
nodea-sstf Signaling_Server LTPS, Gateway, PD, Presence Publisher, IP Media Services 47.11.138.132 47.11.138.180 Follower
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From the **IP Telephony Node Properties** section, click on **Voice Gateway (VGW) and Codecs** to access the codecs configuration.
The voice codec list contains the settings for G.711, G.722, G.723, and G729.
Make sure that for G711, G722 and G729 the **Voice payload size** is set to 20 and the **Voice playout (jitter buffer) delay**, is set to **40** and **80**.
Leave the default values unless Avaya Support directs you to change them.
Scroll down to the bottom of the page and click on the **Save** button.




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Managing: 47.11.138.135 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » VGW and Codecs
Node ID: 343 - Voice Gateway (VGW) and Codecs
General | Voice Codecs | Fax
Voice Codecs
Codec G711: Enabled (required)
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Voice Activity Detection (VAD)
Codec G722: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Codec G729: Enabled
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel
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Managing: 47.11.138.135 Username: admin
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Node ID: 343 - Voice Gateway (VGW) and Codecs
General | Voice Codecs | Fax
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Codec G729: Enabled
Voice payload size: 20 (milliseconds per frame)
Voice playout (jitter buffer) delay: 40 80 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Voice Activity Detection (VAD)
Codec G723.1: Enabled
Voice payload size: 30 (milliseconds per frame)
Voice playout (jitter buffer) delay: 60 120 (milliseconds)
Nominal Maximum
Maximum delay may be automatically adjusted based on nominal settings.
Coding rate: 5.3 (kbps)
* Required Value.
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Save
Cancel
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From the **Applications** section, click on **Gateway** to access the **Virtual Trunk Gateway Configuration**.

Under the **General** tab:

- Vtrk gateway application – It have to be set to **SIPGw and H.323Gw**. It provides the option to select the Gateway application type.

- SIP domain name – It must be set to NRS domain, **interop.com** in our case, or NRS TLAN's IP Address. The SIP Domain Name configured in the Signaling Server properties must match the SIP Service Domain name configured in the NRS. The SIP Domain Name is used in building all SIP messages and appears in the phone context, and must be less than 128 characters in length.

- Local SIP Port – Retain the default value, **5060**. Is the port to which the gateway listens.

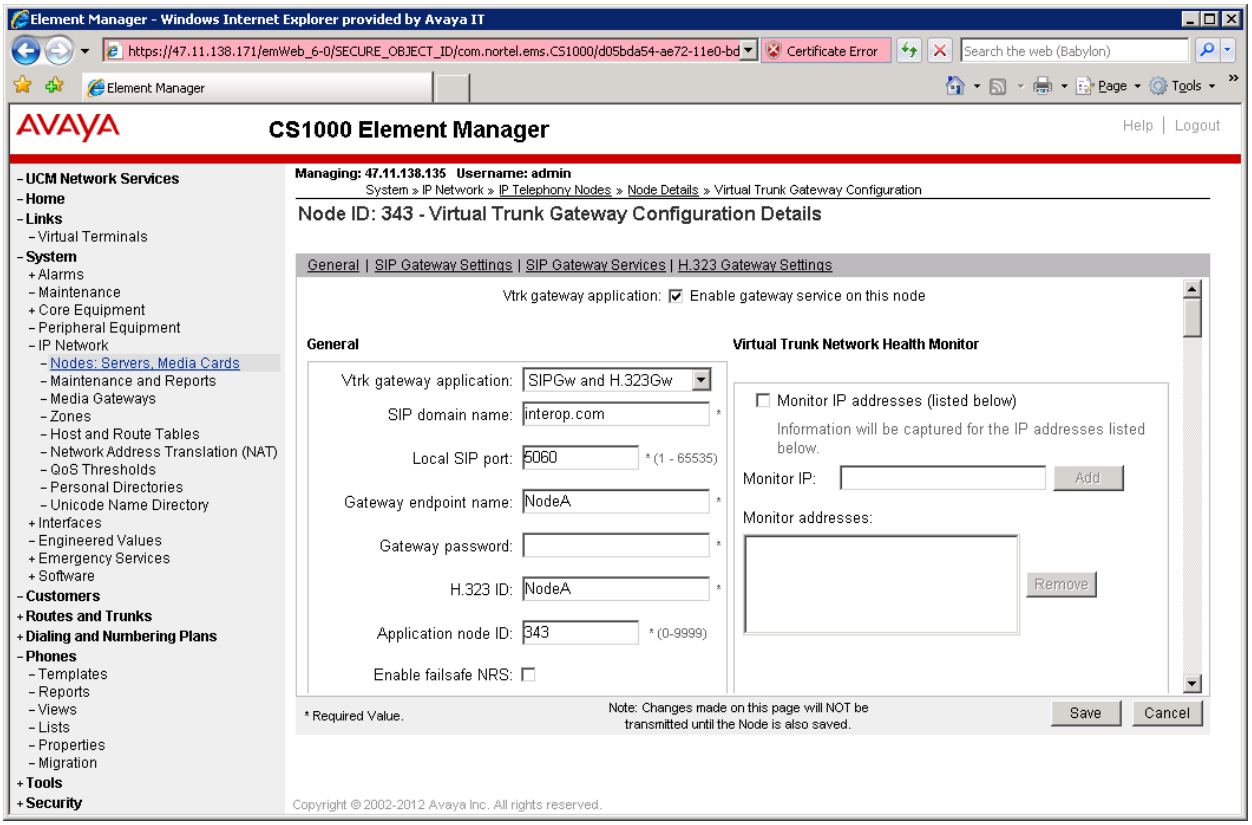
- Gateway endpoint name – It should be the same as Endpoint Name defined in NRS. This is the user name that is used when authenticating this gateway with the NRS.

-Gateway password – Leave this field blank. This is the password that is used when authenticating this gateway with the NRS.

- H.323 ID – It should be the same as defined in NRS. Each H.323 Gatekeeper is configured with an H.323 Gatekeeper alias name.

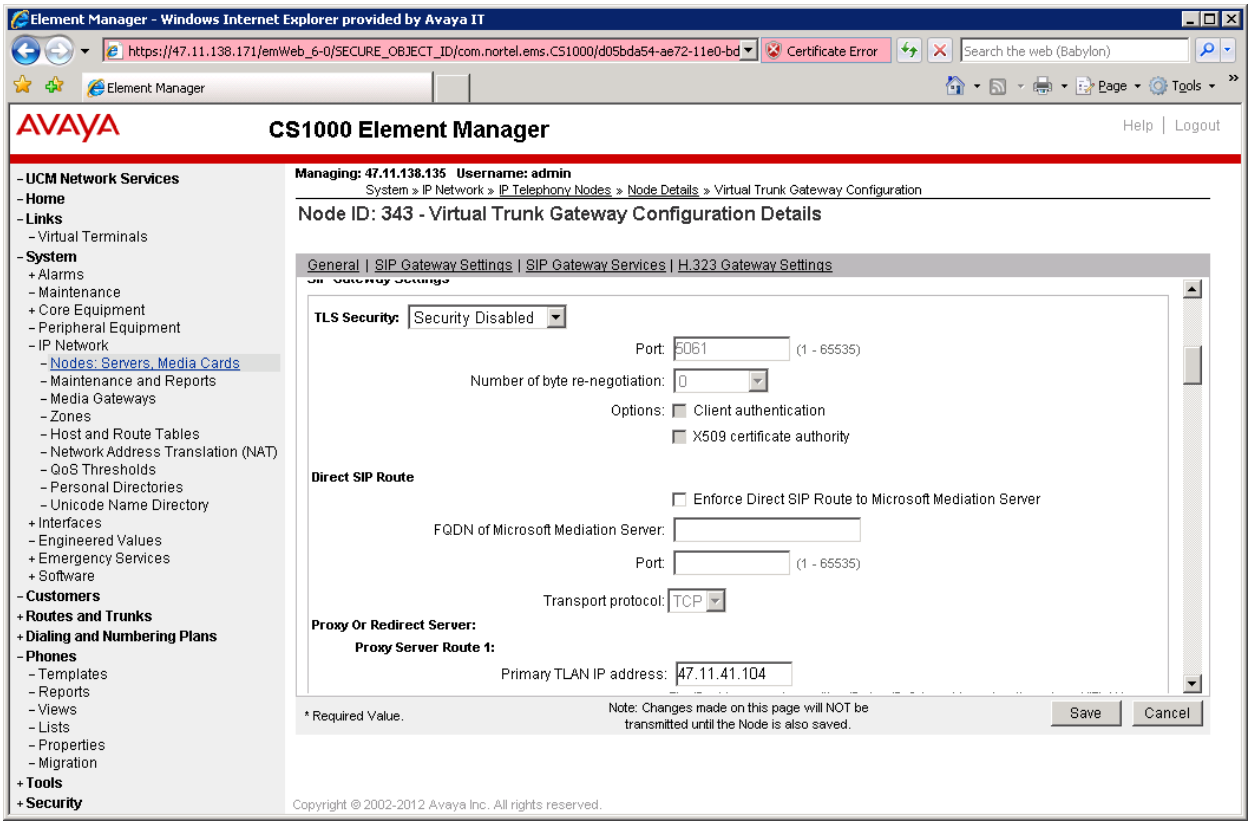
- **Application node ID**, must be the same as the **Node ID**.

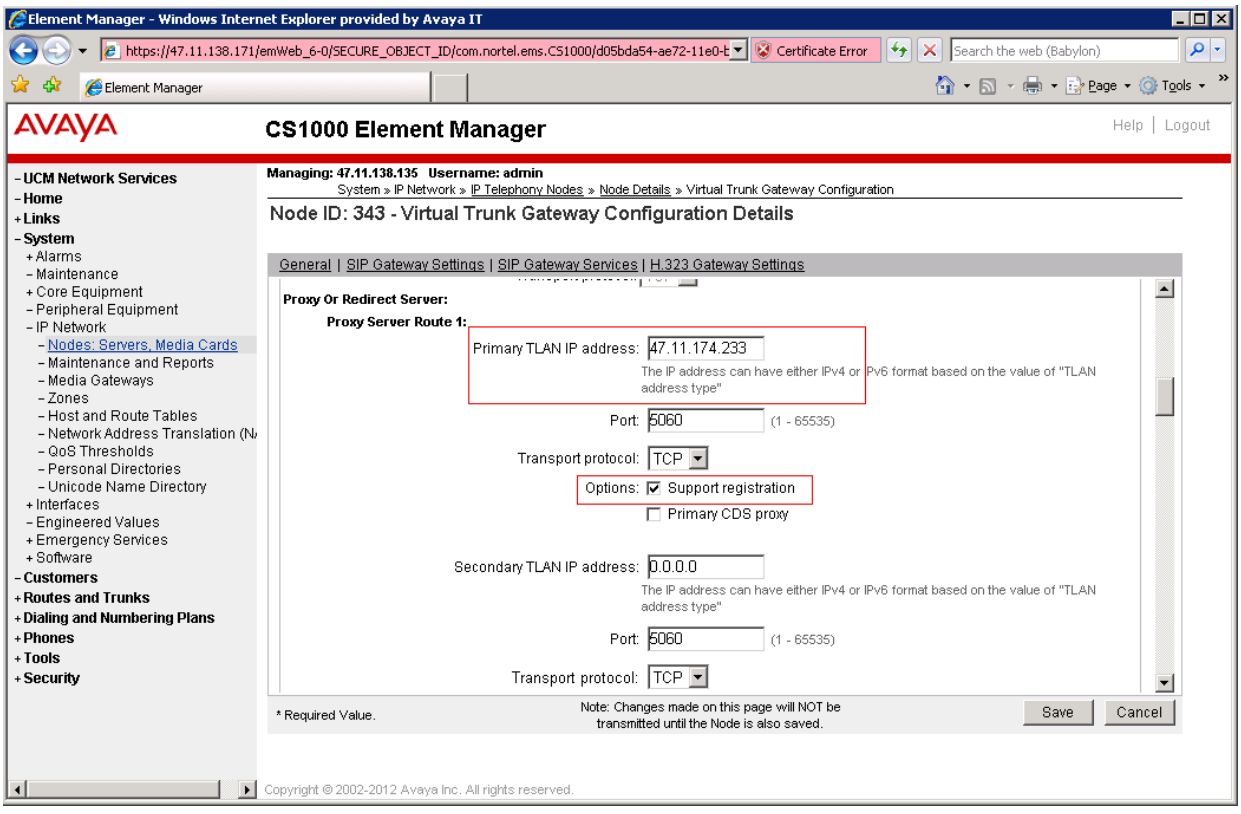
- Retain default values for all other fields.



Under the **SIP Gateway Settings** tab:
- **TLS Security** – Make sure it is set to **Security Disabled**.
- **Primary TLAN IP Address** – Enter the NRS TLAN's IP Address.
- **Port** – Should be set to default, **5060.**
- **Transport Protocol –** Should be set to default, **TCP**.
- **Support Registration –** Must be enabled.
- Retain default values for all other fields.

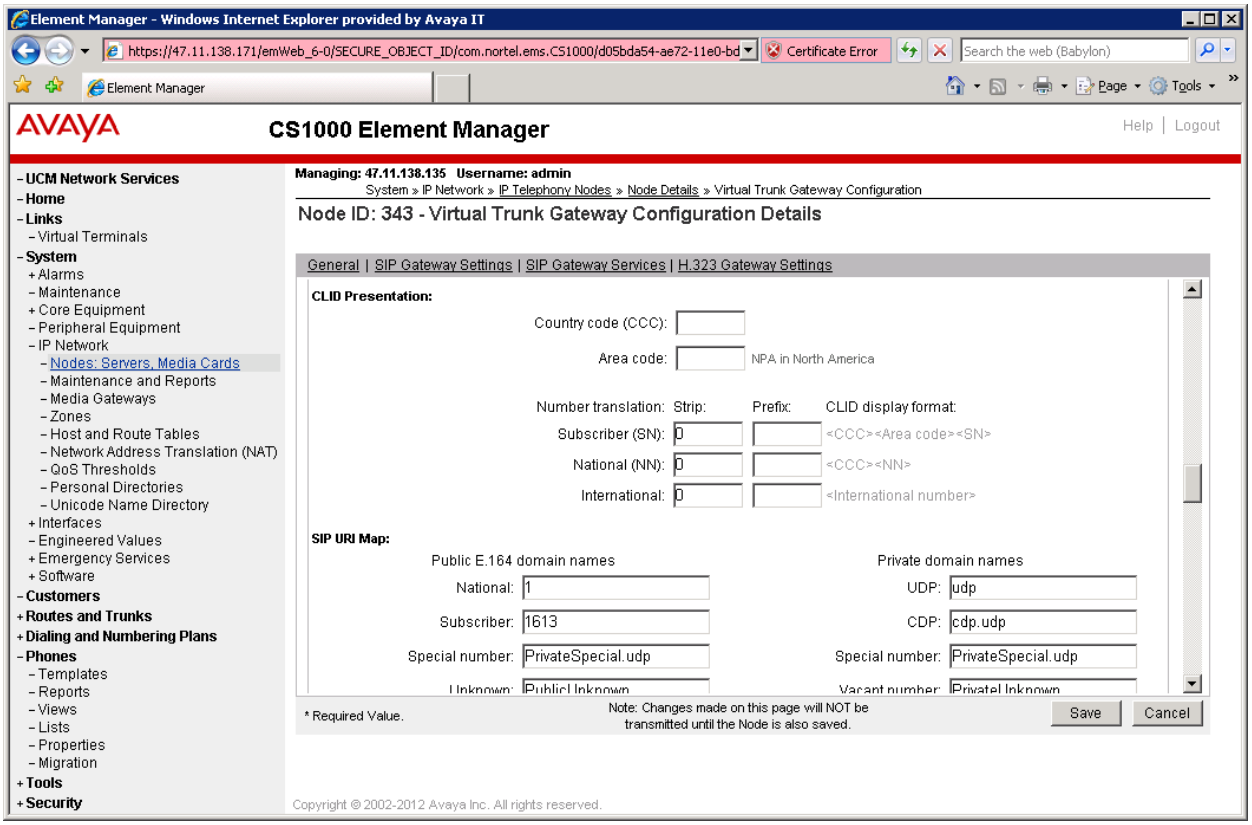
These settings configure the Virtual Trunk Gateway to allow successful registration with the NRS.

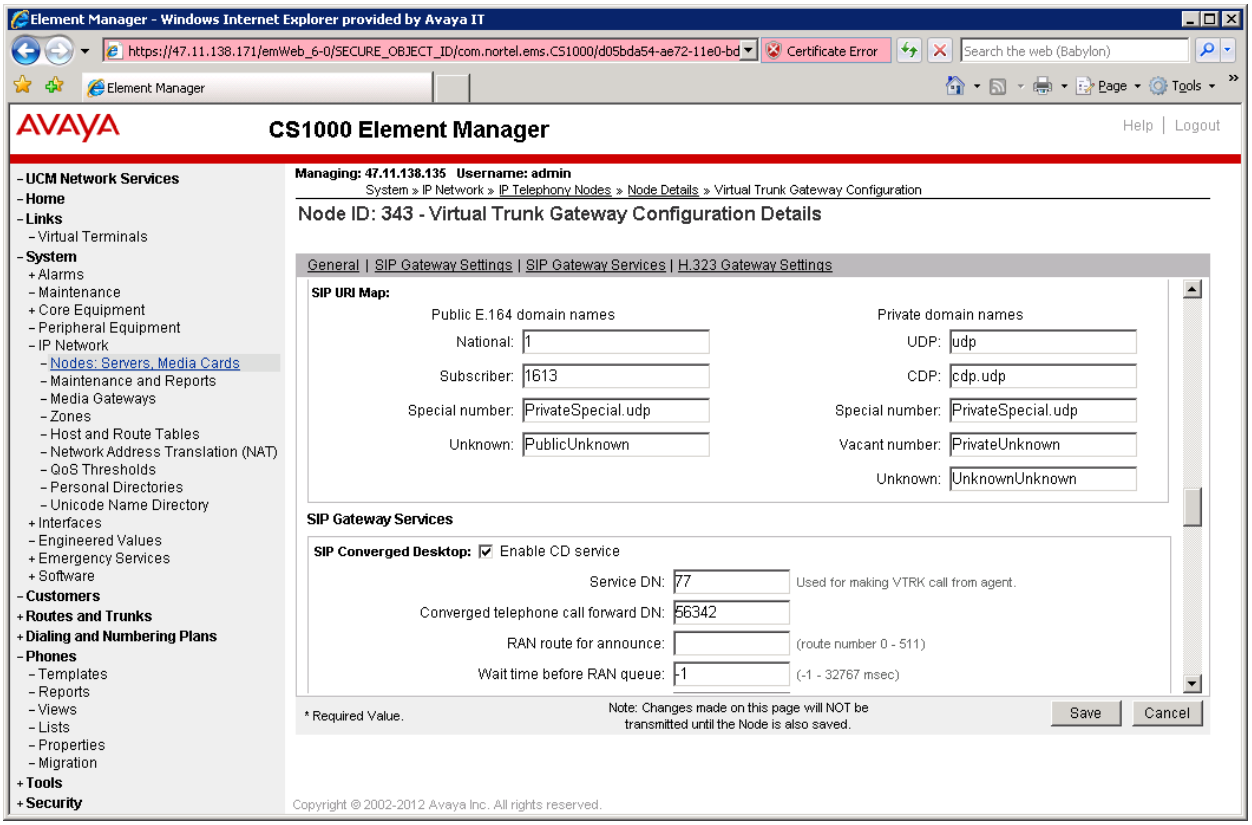

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Managing: 47.11.138.135 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration
Node ID: 343 - Virtual Trunk Gateway Configuration Details
General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings
TLS Security: Security Disabled
Port: 5061 (1 - 65535)
Number of byte re-negotiation: 0
Options: Client authentication
X509 certificate authority
Direct SIP Route
Enforce Direct SIP Route to Microsoft Mediation Server
FQDN of Microsoft Mediation Server:
Port: (1 - 65535)
Transport protocol: TCP
Proxy Or Redirect Server:
Proxy Server Route 1:
Primary TLAN IP address: 47.11.41.104
* Required Value.
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Save
Cancel
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- UCM Network Services
- Home
+ Links
- System
+ Alarms
- Maintenance
+ Core Equipment
- Peripheral Equipment
- IP Network
- Nodes: Servers, Media Cards
- Maintenance and Reports
- Media Gateways
- Zones
- Host and Route Tables
- Network Address Translation (N
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Software
- Customers
+ Routes and Trunks
+ Dialing and Numbering Plans
+ Phones
+ Tools
+ Security
Managing: 47.11.138.135 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration
Node ID: 343 - Virtual Trunk Gateway Configuration Details
General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings
Proxy Or Redirect Server:
Proxy Server Route 1:
Primary TLAN IP address: 47.11.174.233
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)
Transport protocol: TCP
Options: Support registration
Primary CDS proxy
Secondary TLAN IP address: 0.0.0.0
The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type"
Port: 5060 (1 - 65535)
Transport protocol: TCP
* Required Value.
Note: Changes made on this page will NOT be transmitted until the Node is also saved.
Save
Cancel
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.

Configure **CLID** Presentation and **SIP URI Map** according to your dial plan.
Click **Save** from the bottom of the page to save the configurations made.



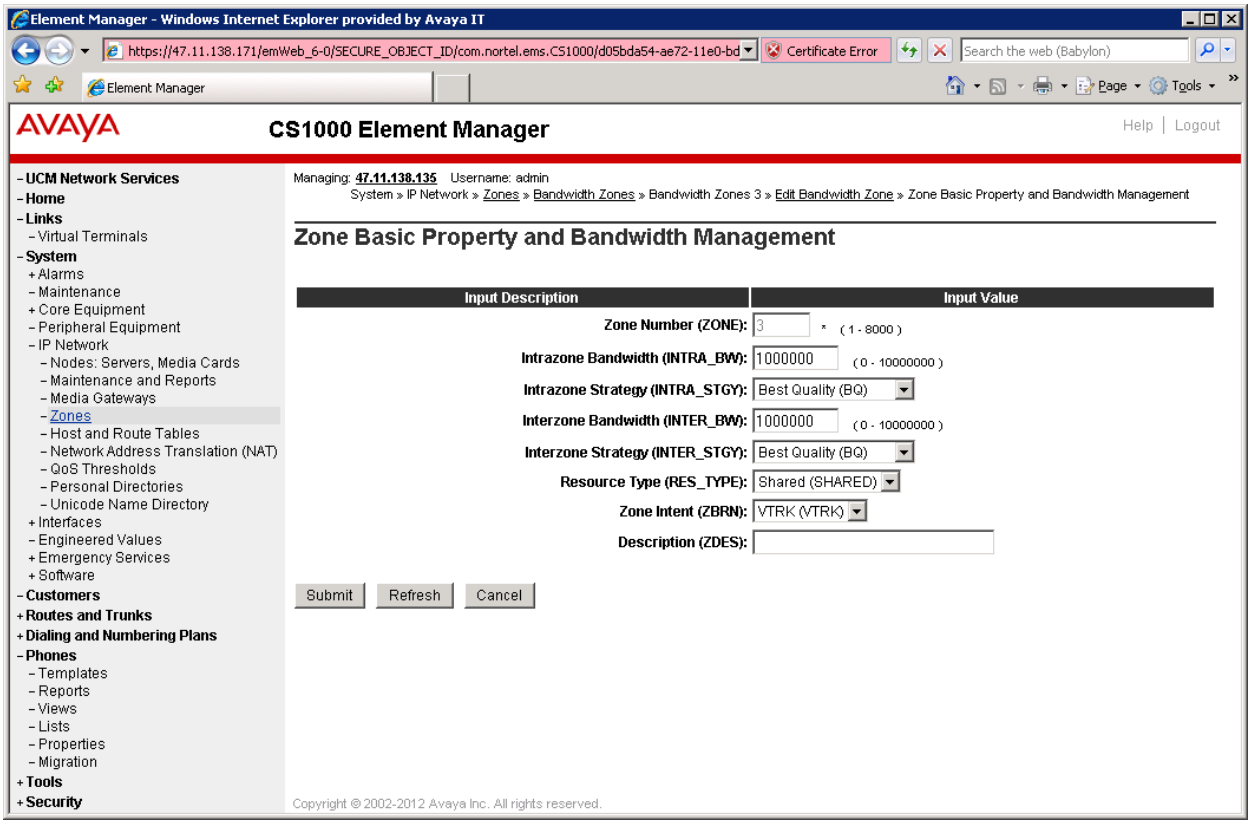


2.3.5 Configure Bandwidth Zones

You must configure a virtual trunk zone for the SIP Line route to work properly.
Go to **System > IP Network > Zones -> Bandwidth Zones** and **Add** new zones as required . Bandwidth Zones are used for alternate call routing between IP telephones and for call Bandwidth Management. SIP trunks require a unique zone, and best practice dictates that IP Trunks, IP telephones and Media Gateways are placed in separate zones.

- **Zone Number** must be a unique non zero value.
- **Intrazone Bandwidth -** Is usually set to the network speed (10, 100 or 1000 Mb/S)
- **Interzone Bandwidth -** Is usually set to the network speed (10, 100 or 1000 Mb/S)
- **Intrazone Strategy –** It sets the preferred codec quality for in zone calls (**BQ** in this example)
- **Interzone Strategy –** It sets the preferred codec quality for zone to zone calls (**BQ** in this example)
- **Resource Type –** It can be set to **Shared**
- **Zone Intent –** It defines the function; in this case it is used for **VTRK** (Virtual Trunks)

Click on the Submit button when completed.



2.3.6 Configure SIP Trunk

Communication Server 1000E virtual trunks will be used for all inbound and outbound calls. Four separate steps are required to configure Communication Server 1000E virtual trunks:

-Configure a D-Channel Handler (DCH)
- Configure a SIP trunk Route Data Block (RDB)
- Configure SIP trunk members

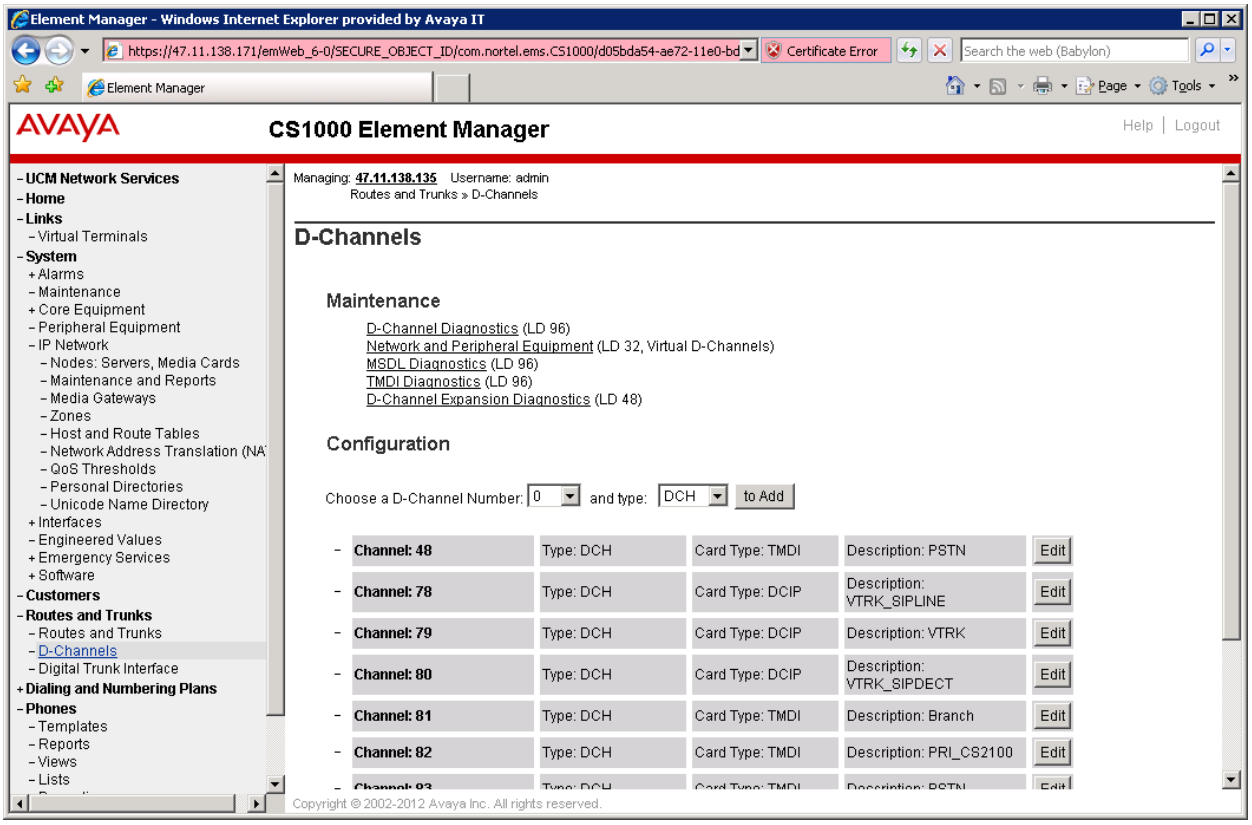
2.3.6.1 Configure D-Channel Handler

The SIP Line Gateway (SLG) application requires a D-channel over IP to communicate with the CS 1000 system. The SIP Line routes are associated with the D-channels and the SLG application running on a Linux server. The SIP Line route is used to communicate with the Call Server.

To configure D-Channel Handler go to **Routes and Trunks > D-Channels**.

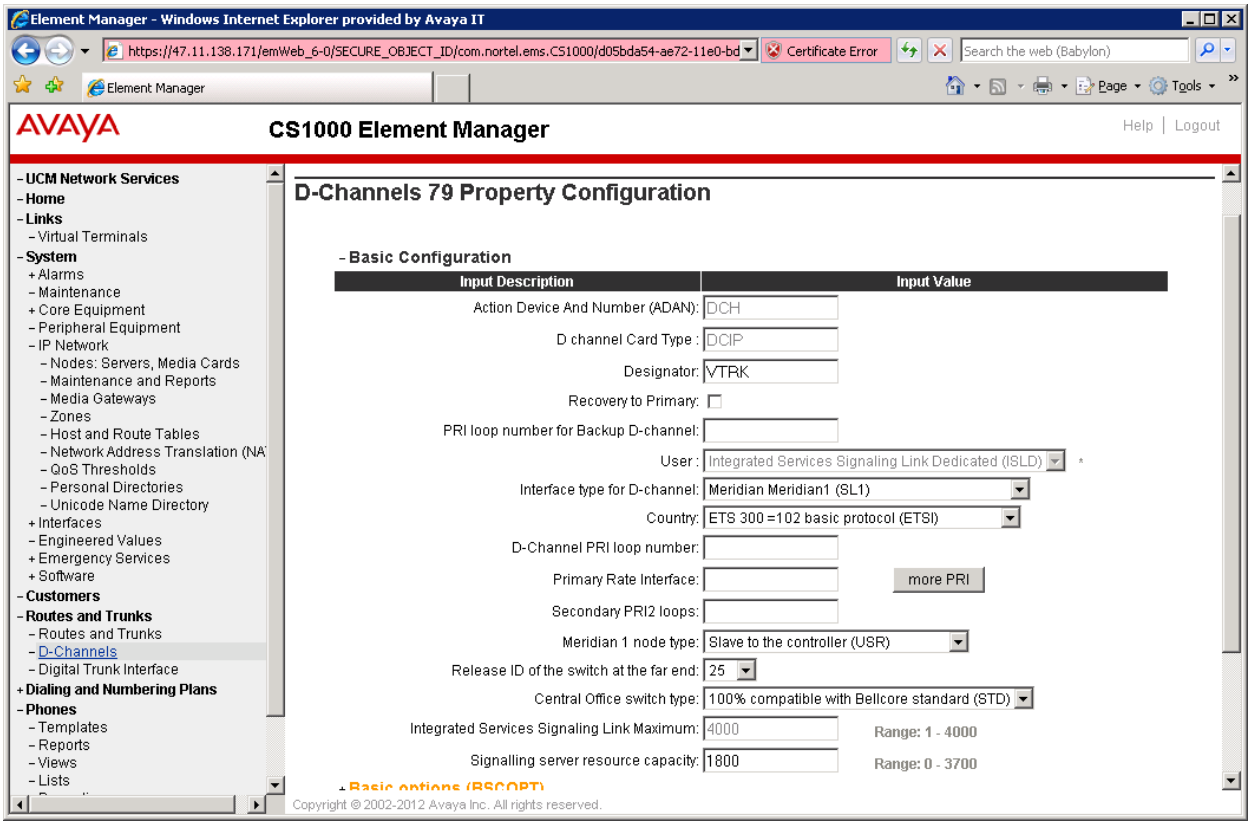
- Under the Configuration section, from the **Choose a D-channel Number** list, select a D-Channel number (79 in our case).

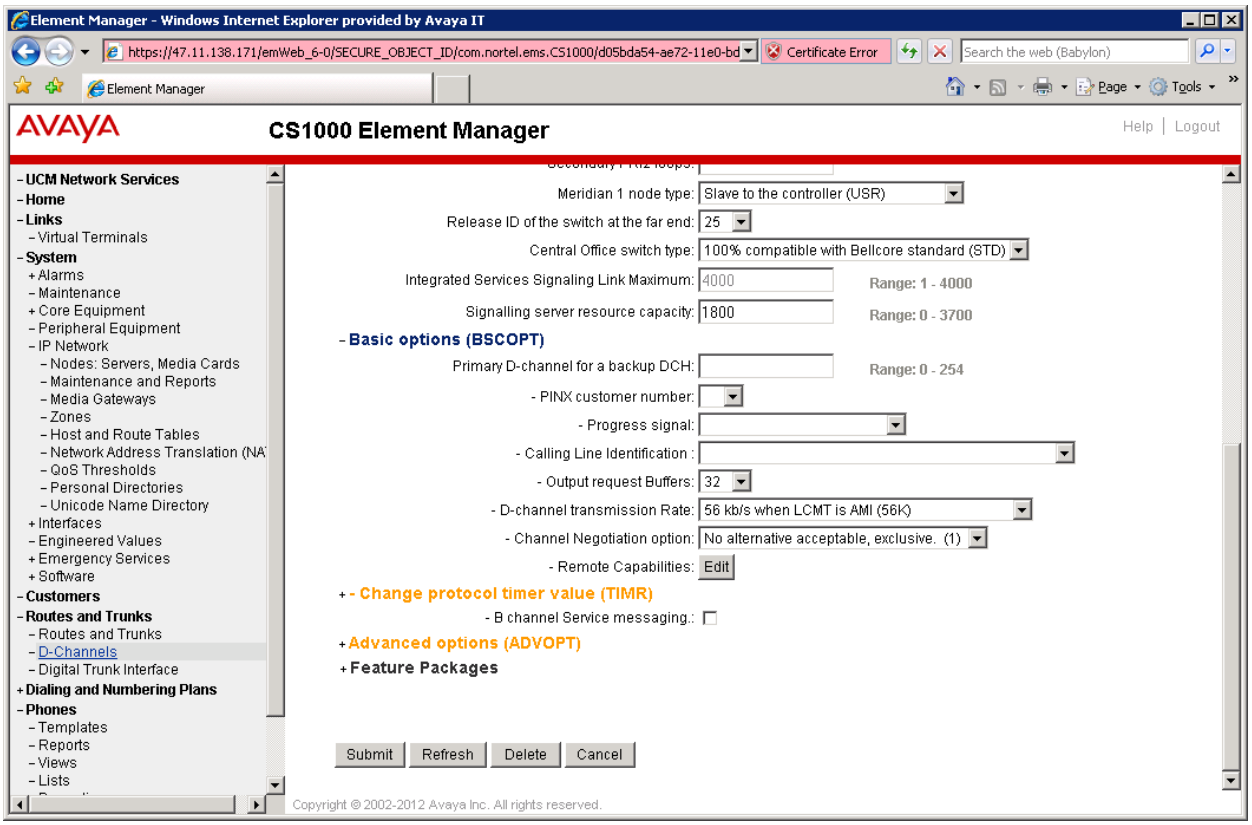
- From the **type** list, select the type of D-Channel, **DCH** and click **to Add** button to add a new DCH.

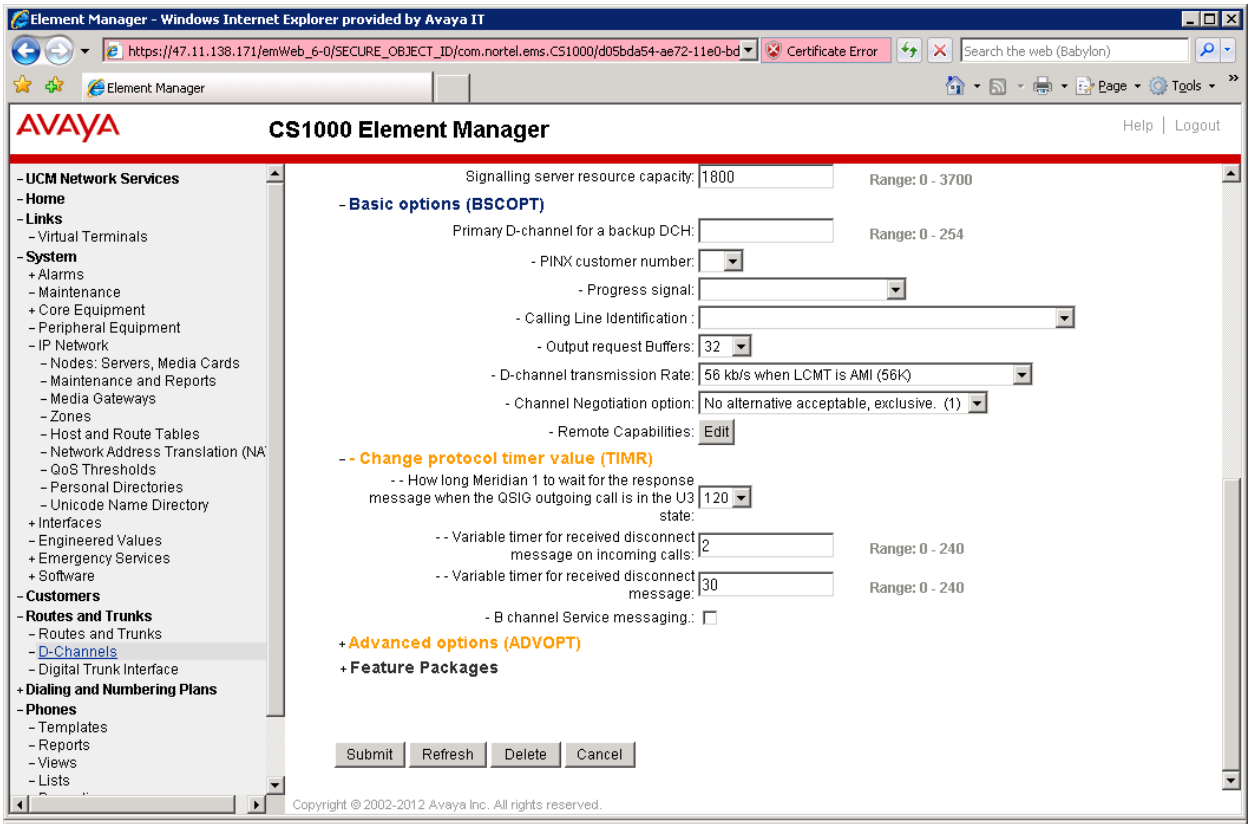


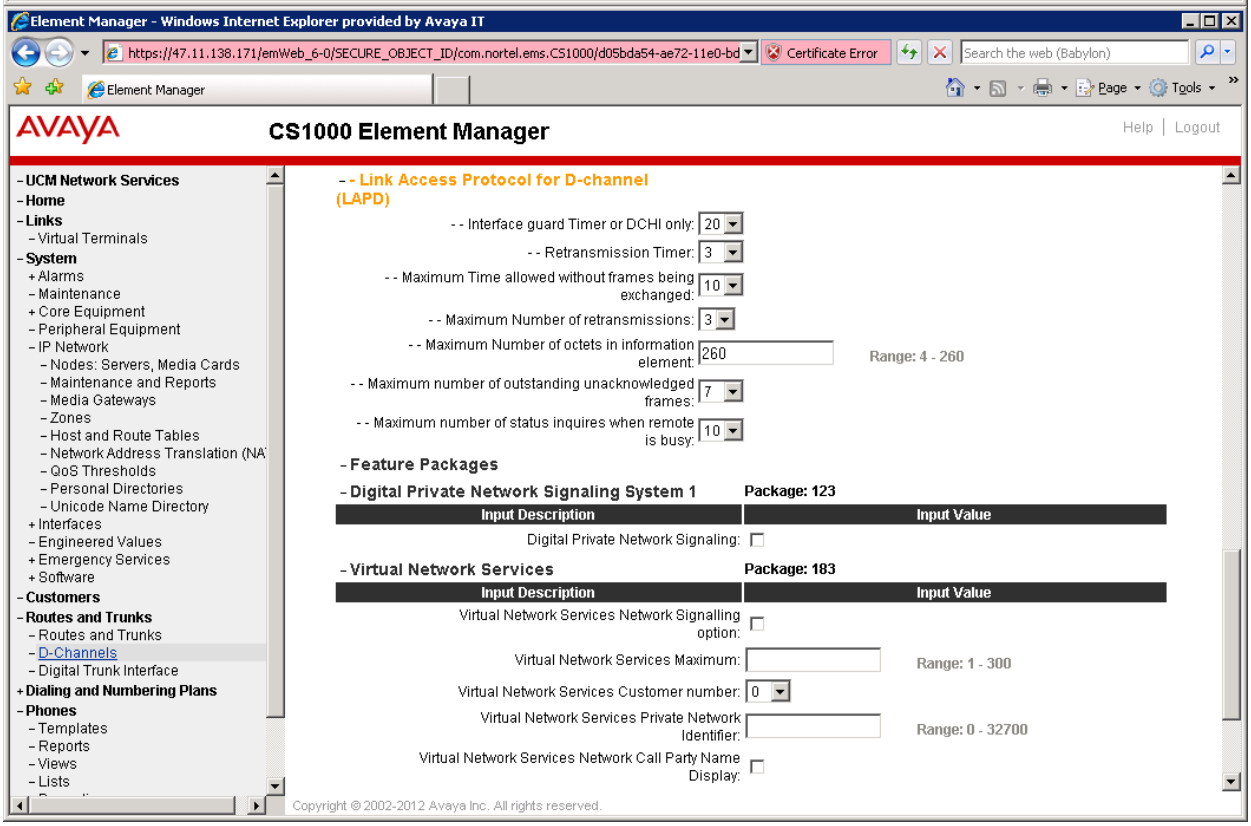
In the **Basic Configuration** section:
- **D channel Card Type** – It should be set to **DCIP** (D channel over IP – SIP or H323)
- **Interface type for D-channel** - It should be set to **Meridian Meridian1 (SL1)**
- Retain default values for all other parameters.

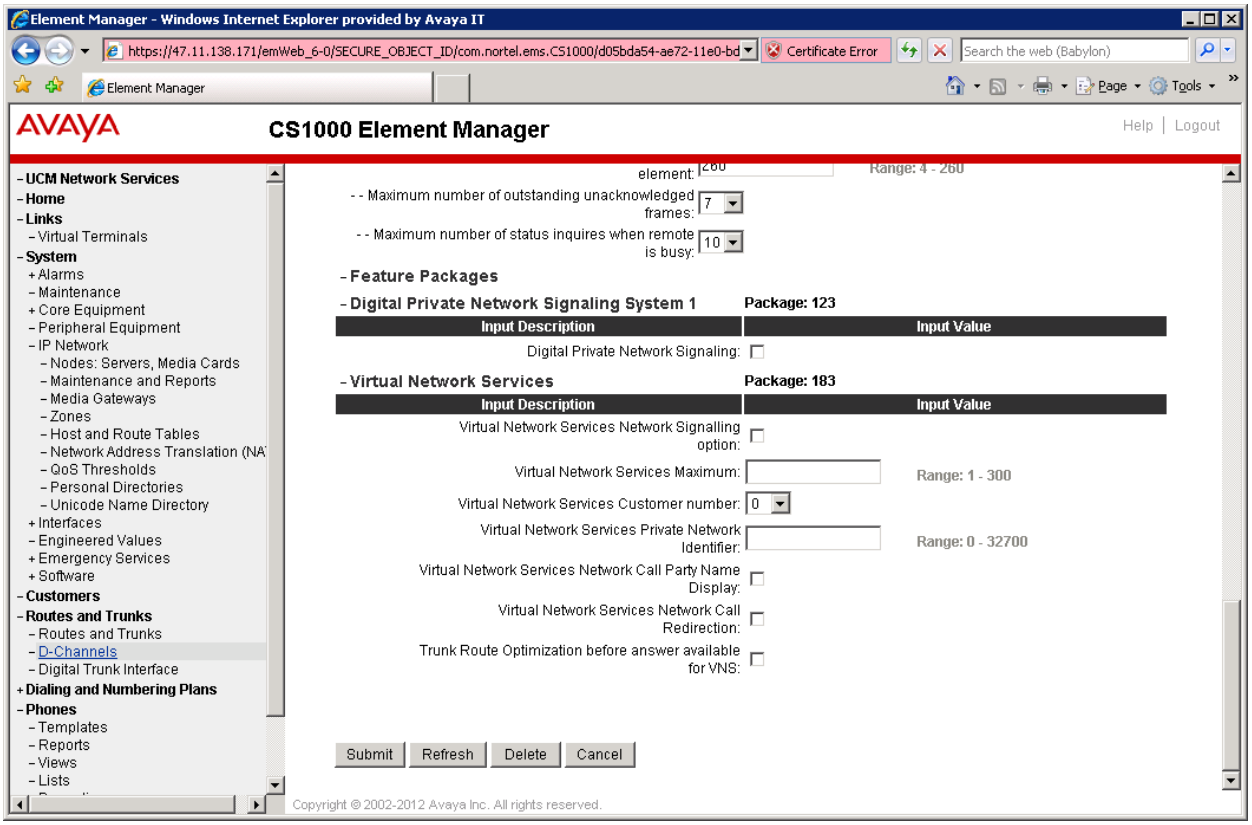
Click **Submit** to save the changes.


Element Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.138.171/emWeb_6-0/SECURE_OBJECT_ID/com.nortel.ems.CS1000/d05bda54-ae72-11e0-bd
Certificate Error
Search the web (Babylon)
Element Manager
Page
Tools
AVAYA
CS1000 Element Manager
Help | Logout
- UCM Network Services
- Home
- Links
- Virtual Terminals
- System
+ Alarms
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- Network Address Translation (NA
- QoS Thresholds
- Personal Directories
- Unicode Name Directory
+ Interfaces
- Engineered Values
+ Emergency Services
+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
+ Dialing and Numbering Plans
- Phones
- Templates
- Reports
- Views
- Lists
D-Channels 79 Property Configuration
- Basic Configuration
Input Description
Input Value
Action Device And Number (ADAN): DCH
D channel Card Type : DCIP
Designator: VTRK
Recovery to Primary:
PRI loop number for Backup D-channel:
User : Integrated Services Signaling Link Dedicated (ISLD)
Interface type for D-channel: Meridian Meridian1 (SL1)
Country: ETS 300 =102 basic protocol (ETSI)
D-Channel PRI loop number:
Primary Rate Interface:
more PRI
Secondary PRI2 loops:
Meridian 1 node type: Slave to the controller (USR)
Release ID of the switch at the far end: 25
Central Office switch type: 100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum: 4000
Range: 1 - 4000
Signalling server resource capacity: 1800
Range: 0 - 3700
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Element Manager - Windows Internet Explorer provided by Avaya IT
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Certificate Error
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- Phones
- Templates
- Reports
- Views
- Lists
Meridian 1 node type: Slave to the controller (USR)
Release ID of the switch at the far end: 25
Central Office switch type: 100% compatible with Bellcore standard (STD)
Integrated Services Signaling Link Maximum: 4000
Range: 1 - 4000
Signalling server resource capacity: 1800
Range: 0 - 3700
- Basic options (BSCOPT)
Primary D-channel for a backup DCH:
Range: 0 - 254
- PINX customer number:
- Progress signal:
- Calling Line Identification :
- Output request Buffers: 32
- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option: No alternative acceptable, exclusive. (1)
- Remote Capabilities: Edit
+ - Change protocol timer value (TIMR)
- B channel Service messaging.:
+ Advanced options (ADVOPT)
+ Feature Packages
Submit
Refresh
Delete
Cancel
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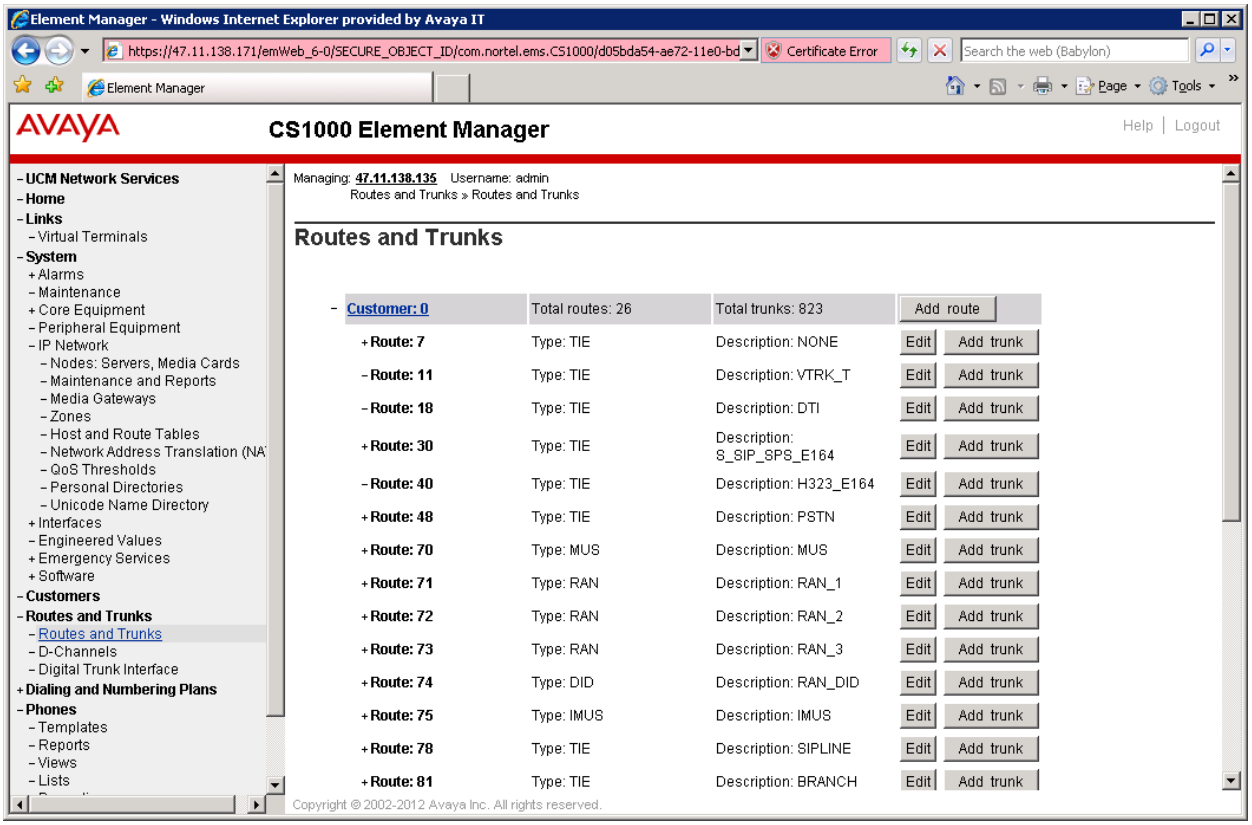

Element Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.138.171/emWeb_6-0/SECURE_OBJECT_ID/com.nortel.ems.CS1000/d05bda54-ae72-11e0-bd
Certificate Error
Search the web (Babylon)
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Tools
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CS1000 Element Manager
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+ Software
- Customers
- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
+ Dialing and Numbering Plans
- Phones
- Templates
- Reports
- Views
- Lists
Signalling server resource capacity: 1800 Range: 0 - 3700
- Basic options (BSCOPT)
Primary D-channel for a backup DCH: Range: 0 - 254
- PINX customer number:
- Progress signal:
- Calling Line Identification :
- Output request Buffers: 32
- D-channel transmission Rate: 56 kb/s when LCMT is AMI (56K)
- Channel Negotiation option: No alternative acceptable, exclusive. (1)
- Remote Capabilities: Edit
- - Change protocol timer value (TIMR)
- - How long Meridian 1 to wait for the response message when the QSIG outgoing call is in the U3 state: 120
- - Variable timer for received disconnect message on incoming calls: 2 Range: 0 - 240
- - Variable timer for received disconnect message: 30 Range: 0 - 240
- B channel Service messaging.:
+ Advanced options (ADVOPT)
+ Feature Packages
Submit Refresh Delete Cancel
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Element Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.138.171/emWeb_6-0/SECURE_OBJECT_ID/com.nortel.ems.CS1000/d05bda54-ae72-11e0-bd
Certificate Error
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- Routes and Trunks
- Routes and Trunks
- D-Channels
- Digital Trunk Interface
+ Dialing and Numbering Plans
- Phones
- Templates
- Reports
- Views
- Lists
- - Link Access Protocol for D-channel (LAPD)
- - Interface guard Timer or DCHI only: 20
- - Retransmission Timer: 3
- - Maximum Time allowed without frames being exchanged: 10
- - Maximum Number of retransmissions: 3
- - Maximum Number of octets in information element: 260 Range: 4 - 260
- - Maximum number of outstanding unacknowledged frames: 7
- - Maximum number of status inquires when remote is busy: 10
- Feature Packages
- Digital Private Network Signaling System 1 Package: 123
Input Description | Input Value
Digital Private Network Signaling:
- Virtual Network Services Package: 183
Input Description | Input Value
Virtual Network Services Network Signalling option:
Virtual Network Services Maximum: Range: 1 - 300
Virtual Network Services Customer number: 0
Virtual Network Services Private Network Identifier: Range: 0 - 32700
Virtual Network Services Network Call Party Name Display:
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2.3.6.2 Configure a SIP trunk Route Data Block (RDB)

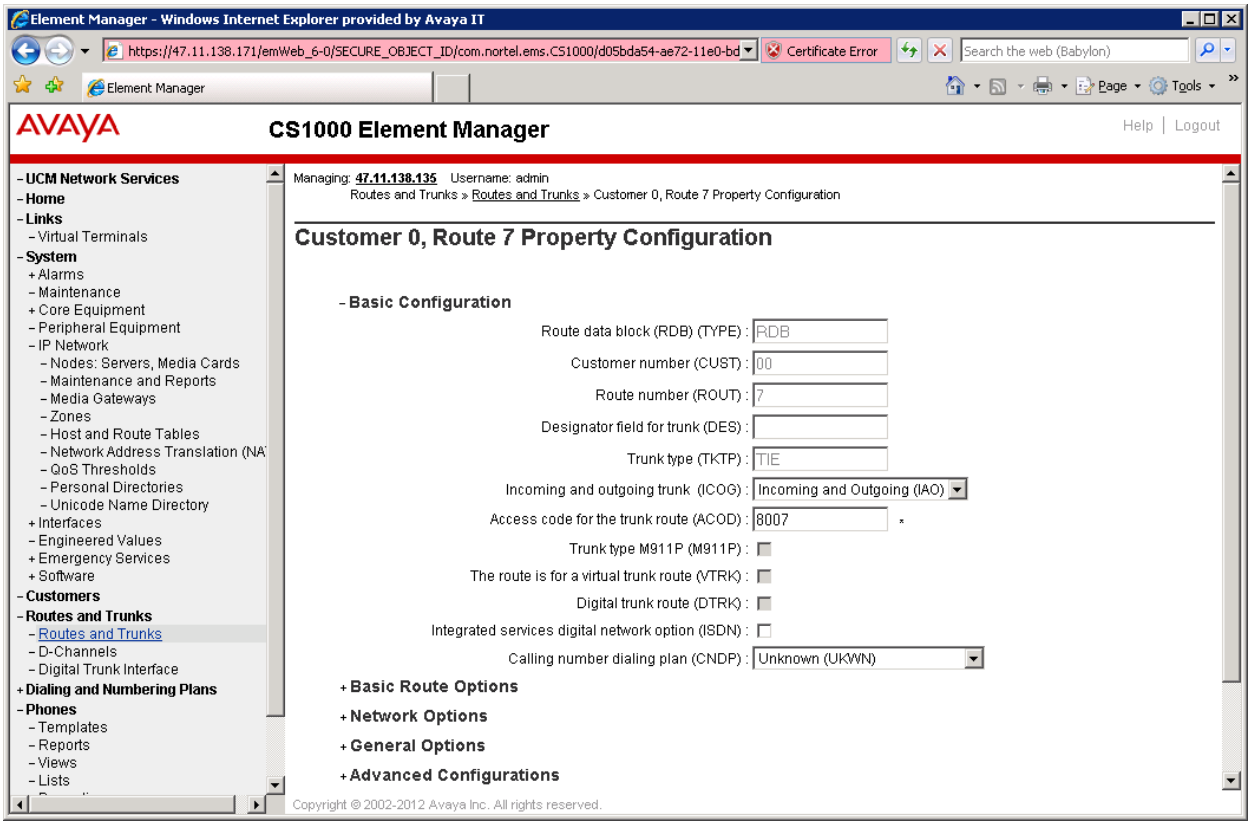
Go to **Routes and Trunks** and expand the appropriate customer number to list all the routes defined for that customer.
Click **Add Route** to add a new route.



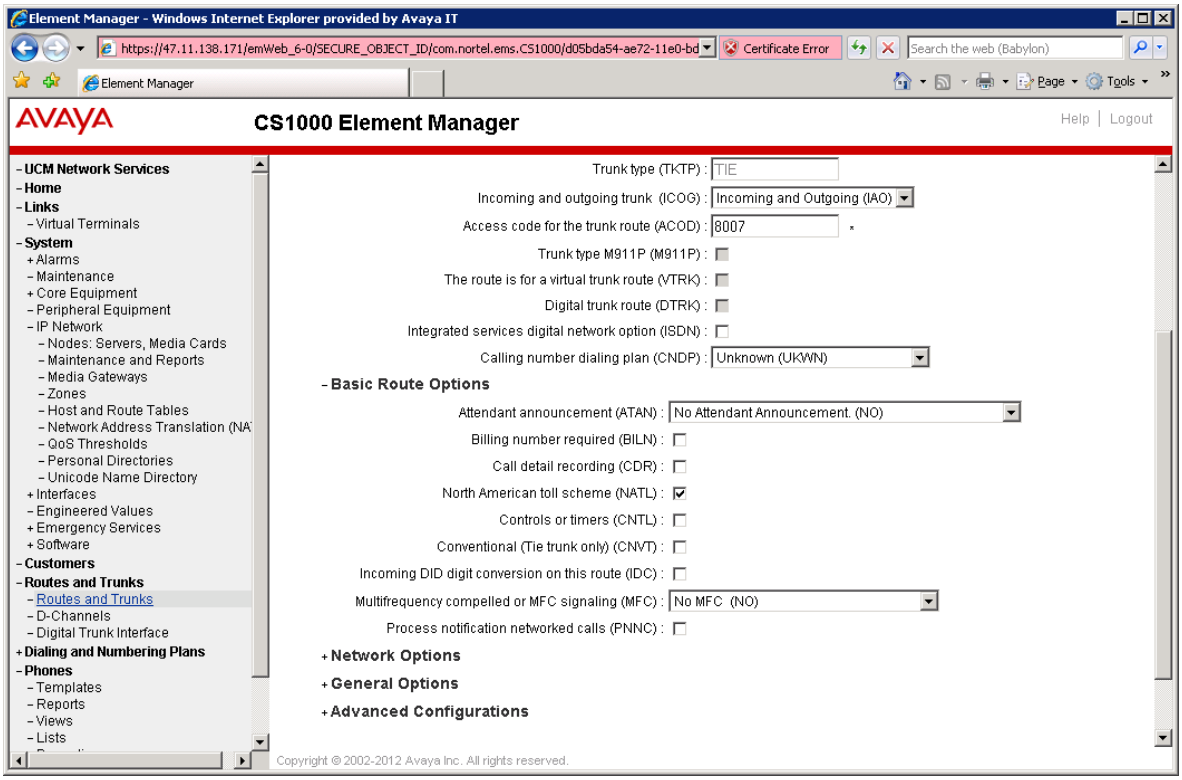
Basic Configuration section

The information entered in the **Basic Configuration** section of this Web page corresponds to Route Data Block information traditionally configured using LD 16 - Route Data Block.

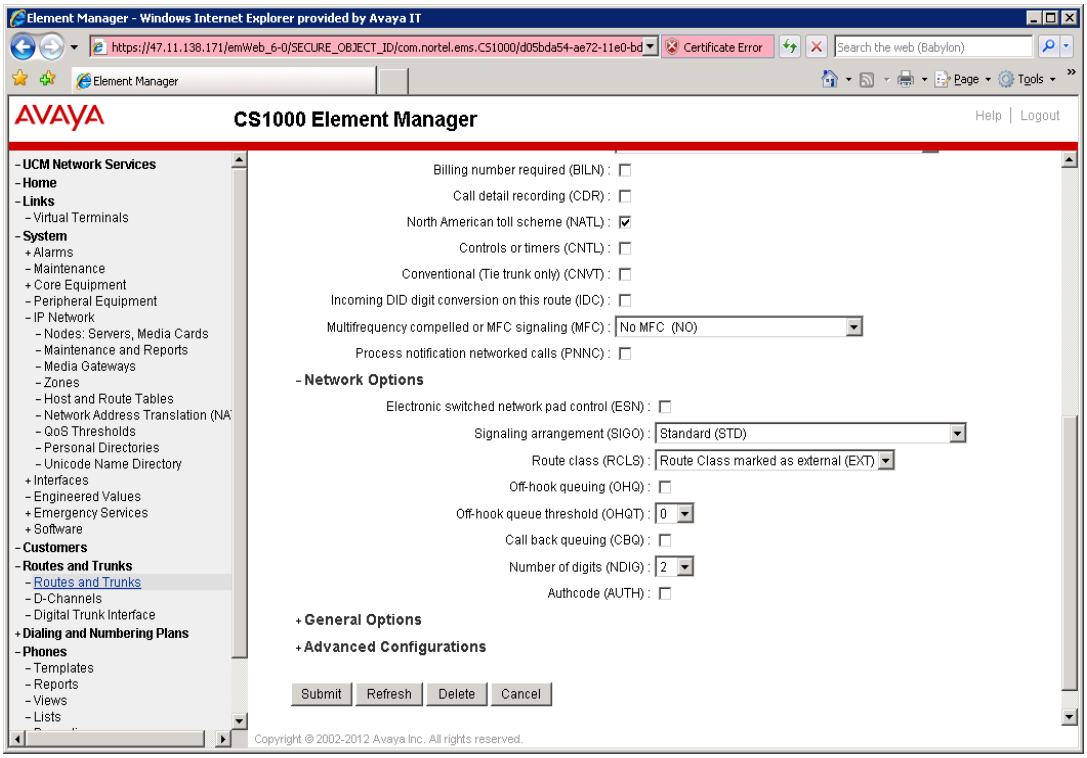
- **Route data block –** RDB - Type of data block.
- **Customer Number** – Enter the appropriate customer number.
- **Route Number** – Enter the number for the route.
- **Designator field for Trunk –** Enter a designation for the trunk (optional).
- **Trunk Type** – It should be set to **TIE**.
- **Incoming and Outgoing trunk –** It should be set to **Incoming and Outgoing (IAO)** to allow calls in both ways, incoming and outgoing.
- **Access Code for the trunk route –** Enter an access code for the route.



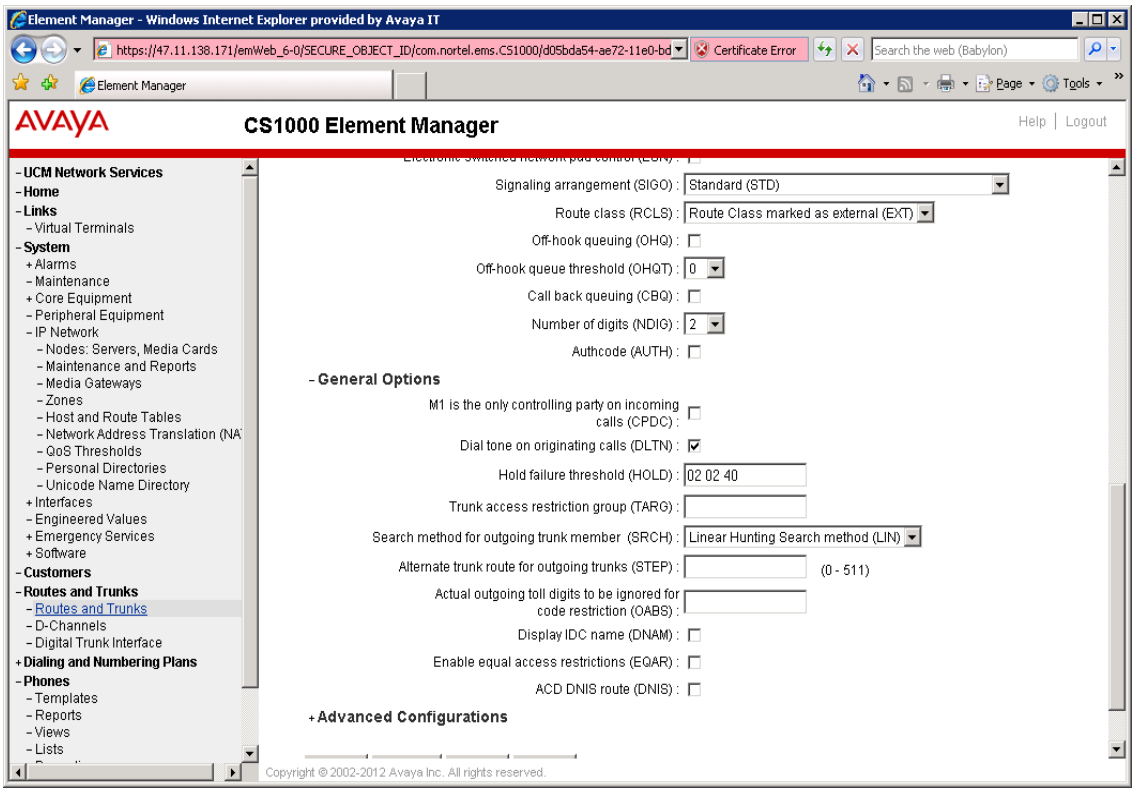
Retain the default values for all **Basic Route Options** parameteres.



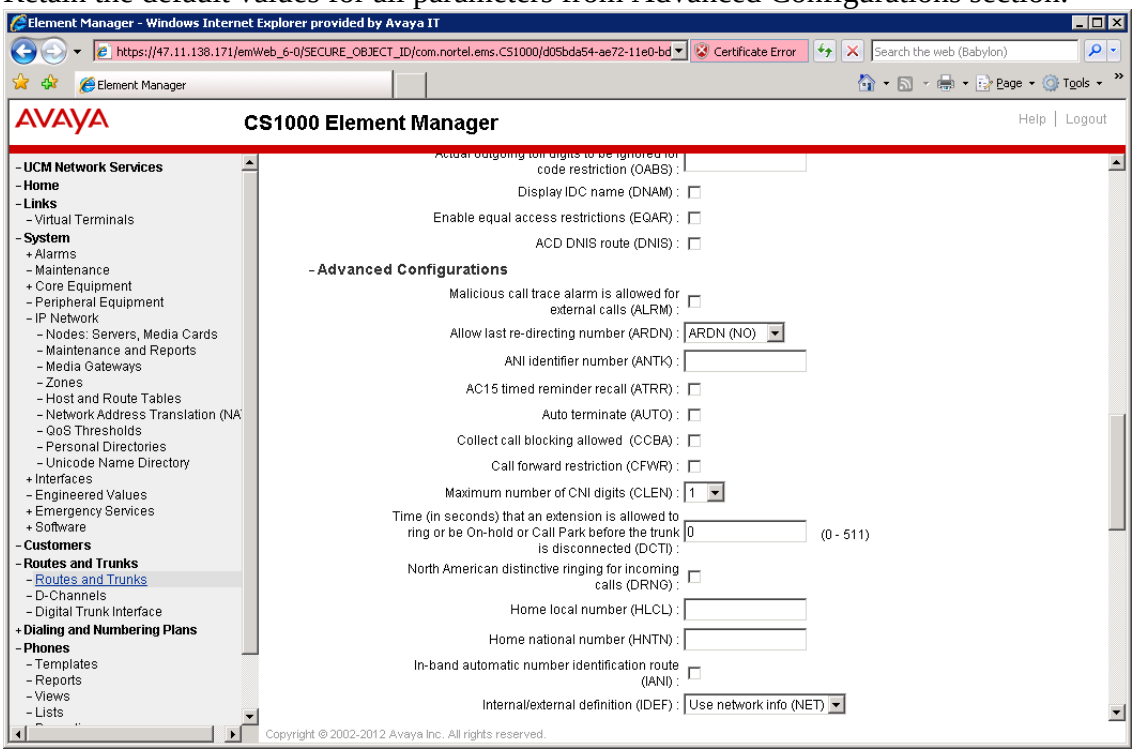
Retain the default values for all **Network Options** parameteres.

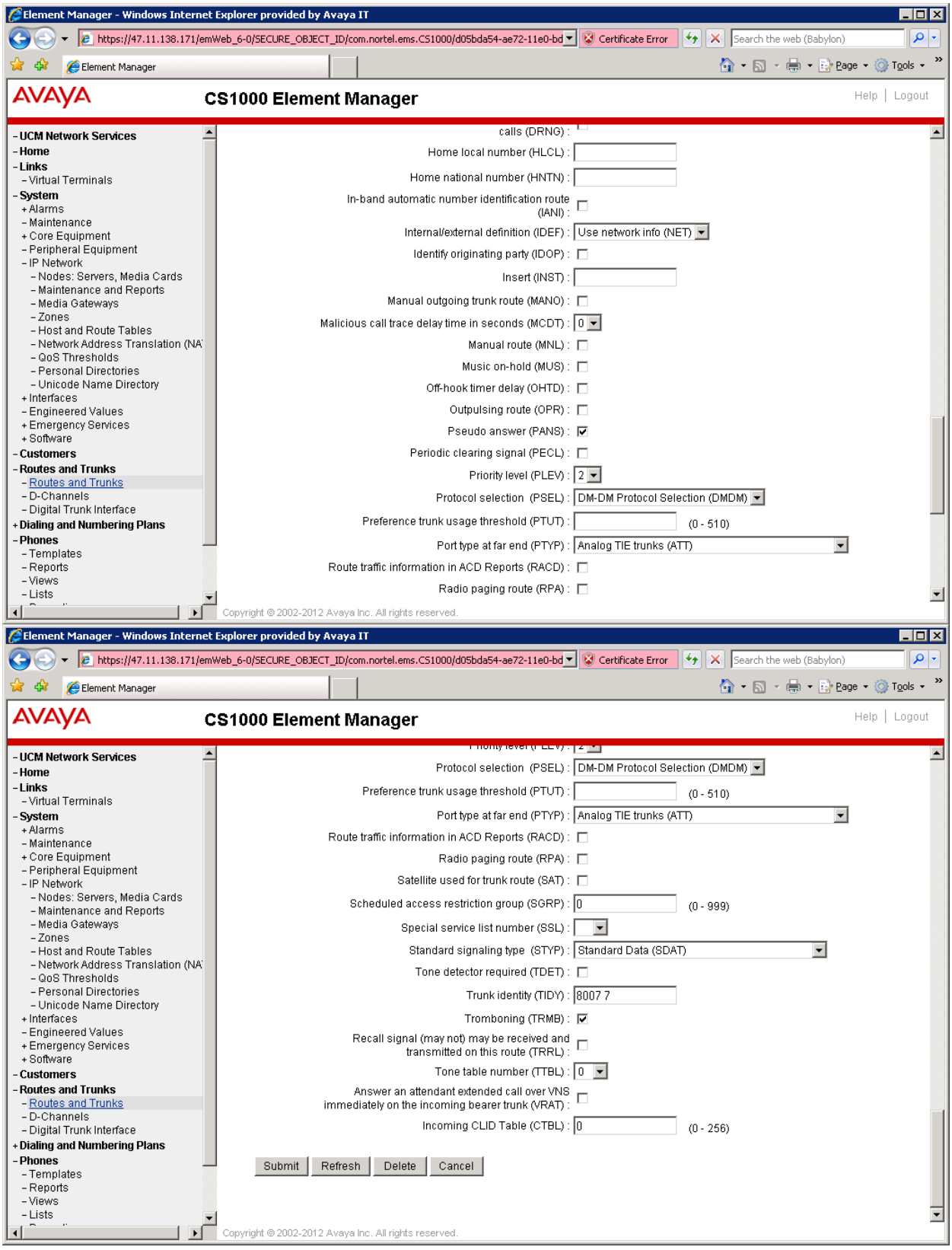


For **General Options** section:
- **Dial Tone on Originating calls** – It should be enabled.
- Retain default values for all parameters apart of this section.



Retain the default values for all parameters from Advanced Configurations section.



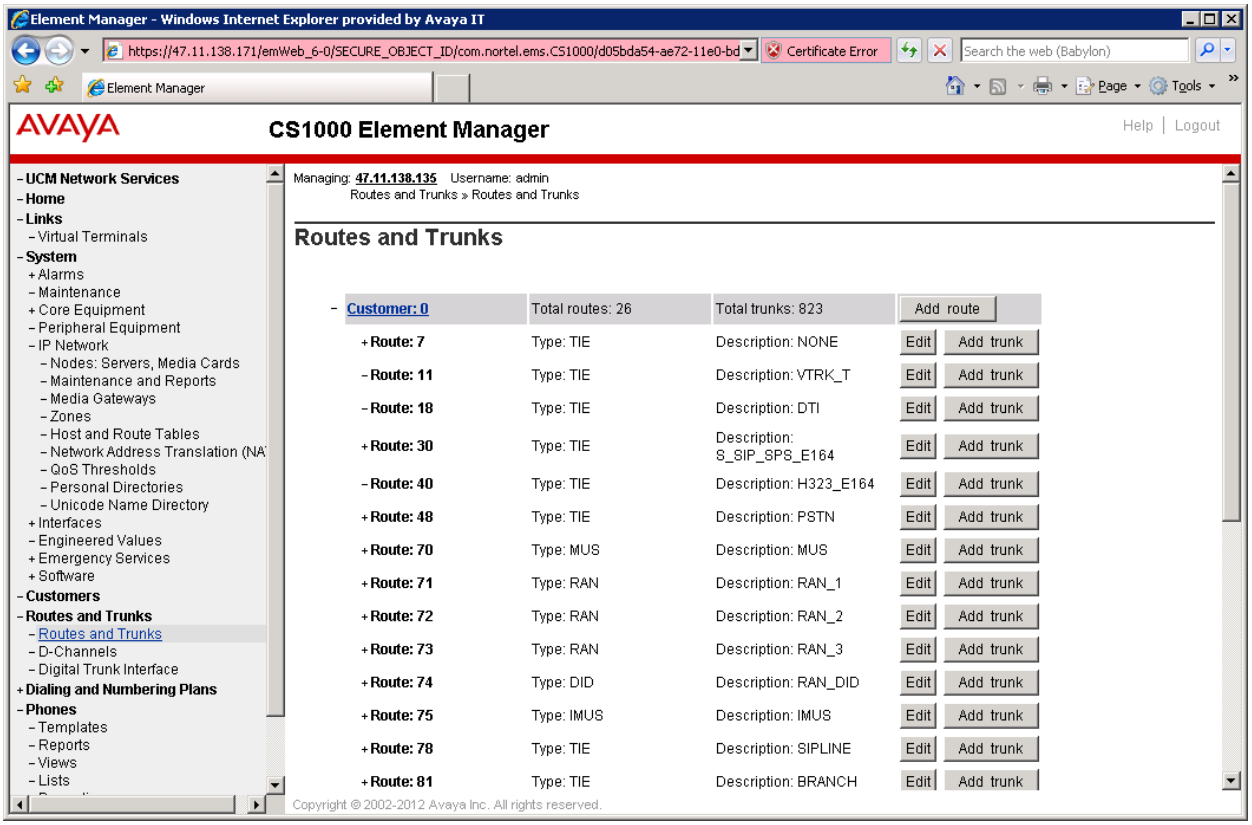


Click Submit to complete the configuration.

2.3.6.3 Configure SIP trunk members

From the **Routes and Trunks > Routes and Trunks**:
(Screenshot needs to be provided)
- Click the appropriate Customer name.
- Click **Add trunk** associated with the route listing, to add new trunk members.
- Select **Basic Configuration**.
- Choose **Multiple trunk input number** if you are using more than one trunk (channel).
- From the **Trunk data block** list, select **IP Trunk (IPTI)**.
- In the **Terminal Number** field, enter a TN.
- In the **Designator field for trunk** field, enter a designator value.
- Select a value for **Extended trunk**.
- Enter a **Member number**.
- Select a value for **Level 3 signaling**.
- Select a value for **Card density**.
- From the **Start arrangement Incoming** list, select a value for the start arrangement for incoming calls.
- From the **Start arrangement Outgoing** list, select a value for the start arrangement for outgoing calls.
- Enter a **Trunk Group Access Restriction** value.
- In the **Channel ID for this trunk** field, enter a channel ID. Select the **Network music** check box to include network music.
- To specify a **Class of Service** for the trunk, click **Edit** and select a **Class of Service**.
- Click Return **Class of Service** to return to the **New Trunk Configuration** Web page.
- Select **Advanced Trunk Configurations**. The Advanced Trunk Configurations list expands.
- Configure Network Class of Service group.
- Click Save to complete.

The Customer Explorer Web page reappears, showing the new trunk members.



2.3.7 Dial Plan

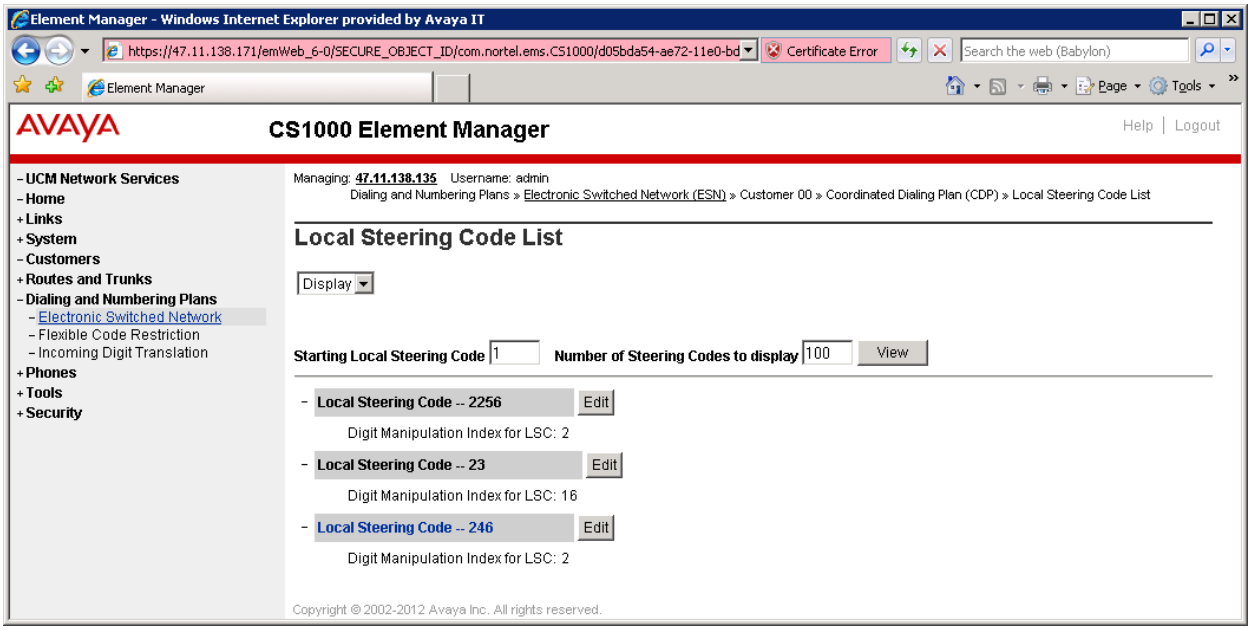
To configure the dial plan for the created SIP Trunk, go to **Dialing and Numbering Plans** > **Electronic Switched Network**

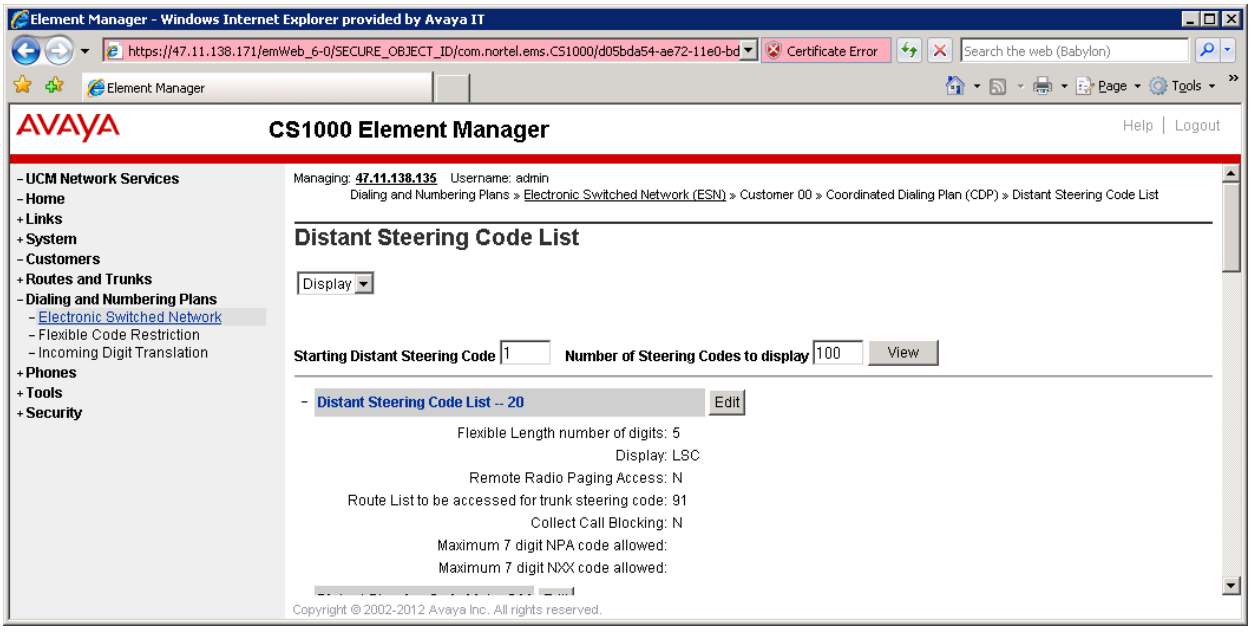
Under **Network Control and Services**, click the links to configure or modify the parameters associated with the following items:

- **Network Control Parameters** (NCTL)
- **ESN Access Codes and Parameters** (ESN)
- **Digit Manipulation Block** (DGT)
- **Home Area Code** (HNPA)
- **Flexible CLID Manipulation Block** (CMDB)
- **Free Calling Area Screening** (FCAS)
- **Free Special Number Screening** (FSNS)
- **Route List Block** (RLB)
- **Incoming Trunk Group Exclusion** (ITGE)
- **Network Attendant Services** (NAS)

Under **Coordinated Dialing Plan (CDP)**, click the links to configure or modify the parameters associated with the following items:

- **Local Steering Code** (LSC)
- **Distant Steering Code** (DSC)
- **Trunk Steering Code** (TSC)

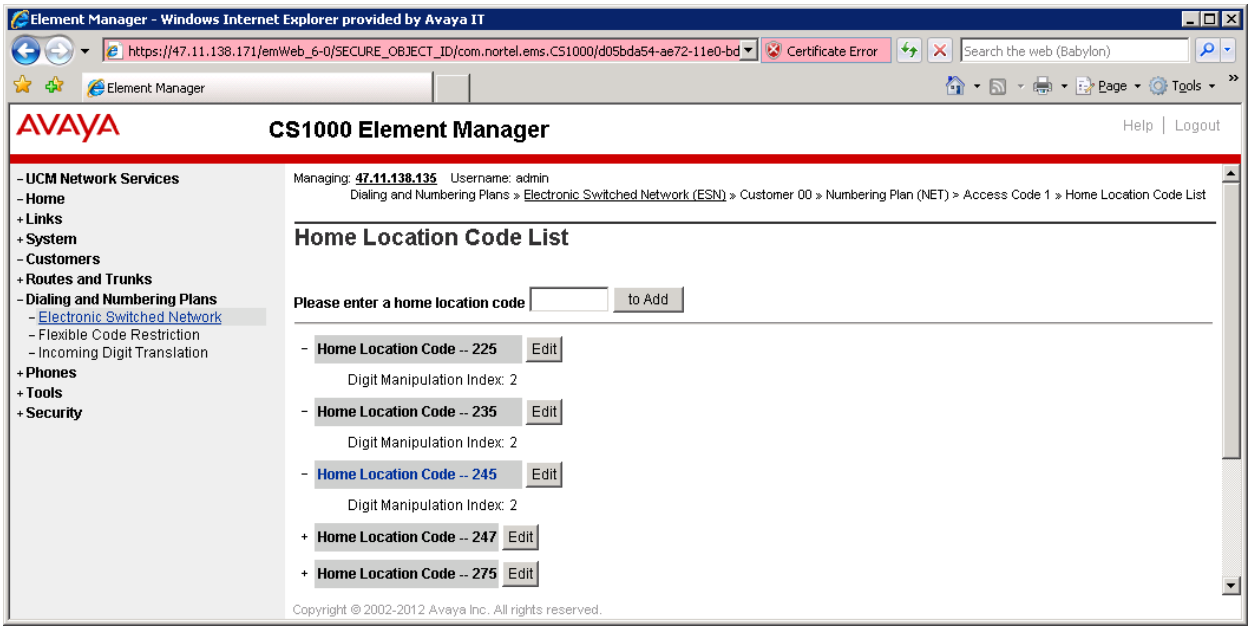




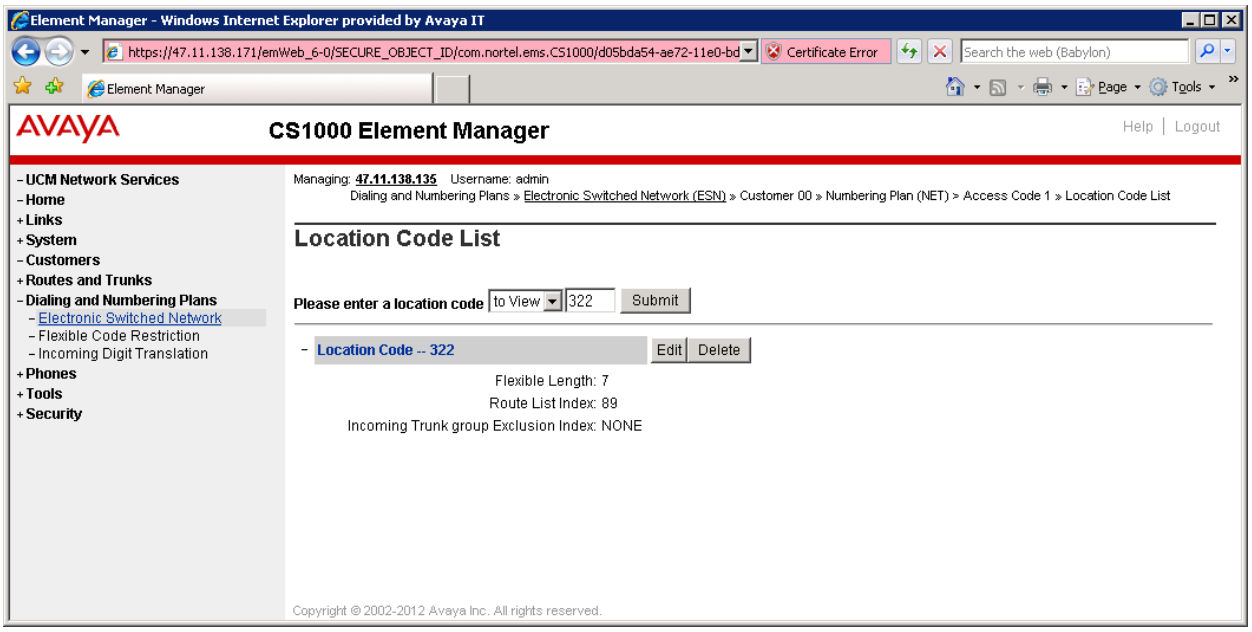
Under **Numbering Plan (NET)**, click the links to configure or modify the parameters associated with **Access Codes**:

- **Home Location Code (HLOC)**

- Location Code (LOC)
- Numbering Plan Area Code (NPA)
- Exchange (Central Office) Code (NXX)
- Special Number (SPN)
- Network Speed Call Access Code (NSCL)



Access Code 1 -> Location Code



2.4 Network Routing Service Configuration

The function of the NRS is to route SIP traffic between the Avaya IP Office, Business Communication Manager and Communication Server 1000.
This section provides a procedure for configuring NRS using NRS Manager.
Network Routing Service Manager can be accessed from Unified Communication Manager.
Before start doing any changes make sure that **Standby Database** is selected.
The NRS numbering plan configuration is stored in XML format in two databases on disk. The active database is used for call processing and the standby database is used for configuration changes. All changes to the numbering plan database are carried out on the standby database. Changes that the administrator makes to the numbering plan database do not affect call processing immediately. The database must first be cut over to the active database. The database is cut over to the active database by executing a database Cut over command.

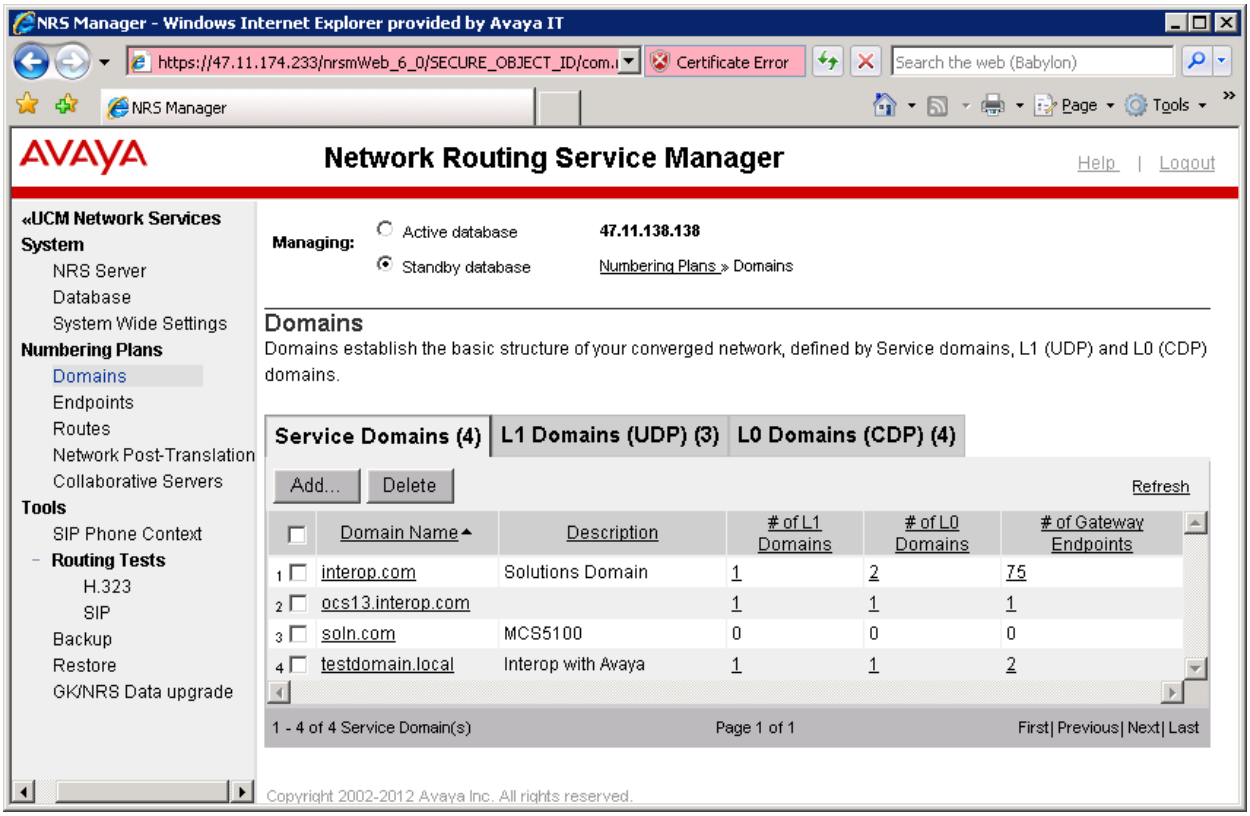
The procedure covers the following areas:
- SIP Service Domain, L1 and L0 domains
- Endpoints
- Routing
- Updating Database

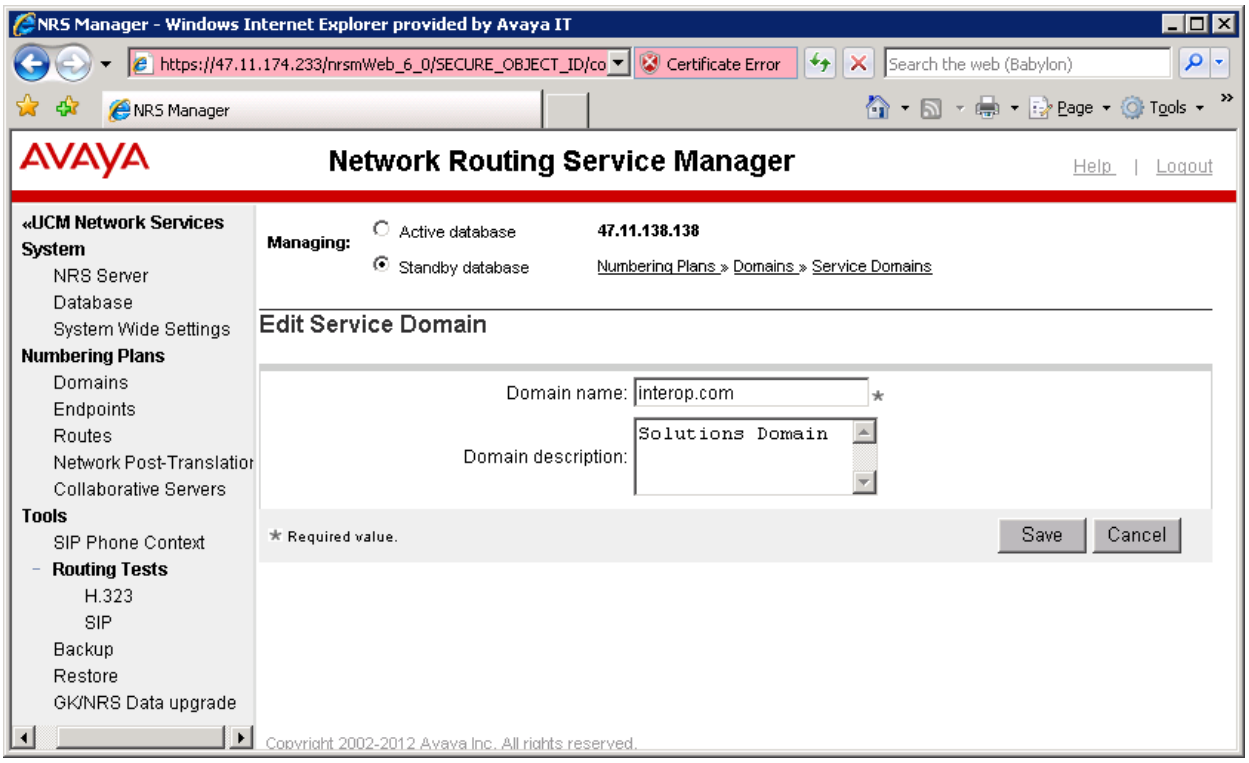
2.4.1 Configuring SIP Service Domain, L1 and L0 domains

To add a domain, go to **Numbering Plans > Domain** and under **Service Domain** tab click on the **Add** button.
- Enter the domain name, **interop.com** in our case.
- Click save button when finished.

Under **L1 Domains** tab, select the service domain (interop.com) previously configured from the **Filter by Domain** drop-down box to start editing.
In the **Edit L1 Domain** page:
- **Domain name -** Enter the name for L1 domain, UDP.
- **Endpoint authentication enabled –** Make sure it is set to **authentication off**.
- Retain the default values for all other parameters.
- Click on the **Save** button when completed.

Under **L0 Domains** tab, select the service domain (interop.com) previously configured from the **Filter by Domain** drop-down box and then the previously configured L1 domain from the **Filter by L1 domain** drop-down box.
In the **Edit L0 Domain** page:
- **Domain Name –** Enter the name for L0 domain, CDP.
- Ensure that **Endpoint authentication enabled** is set to **Not configured**.
- Retain default values for all other fields.
- Click on the **Save** button when completed.

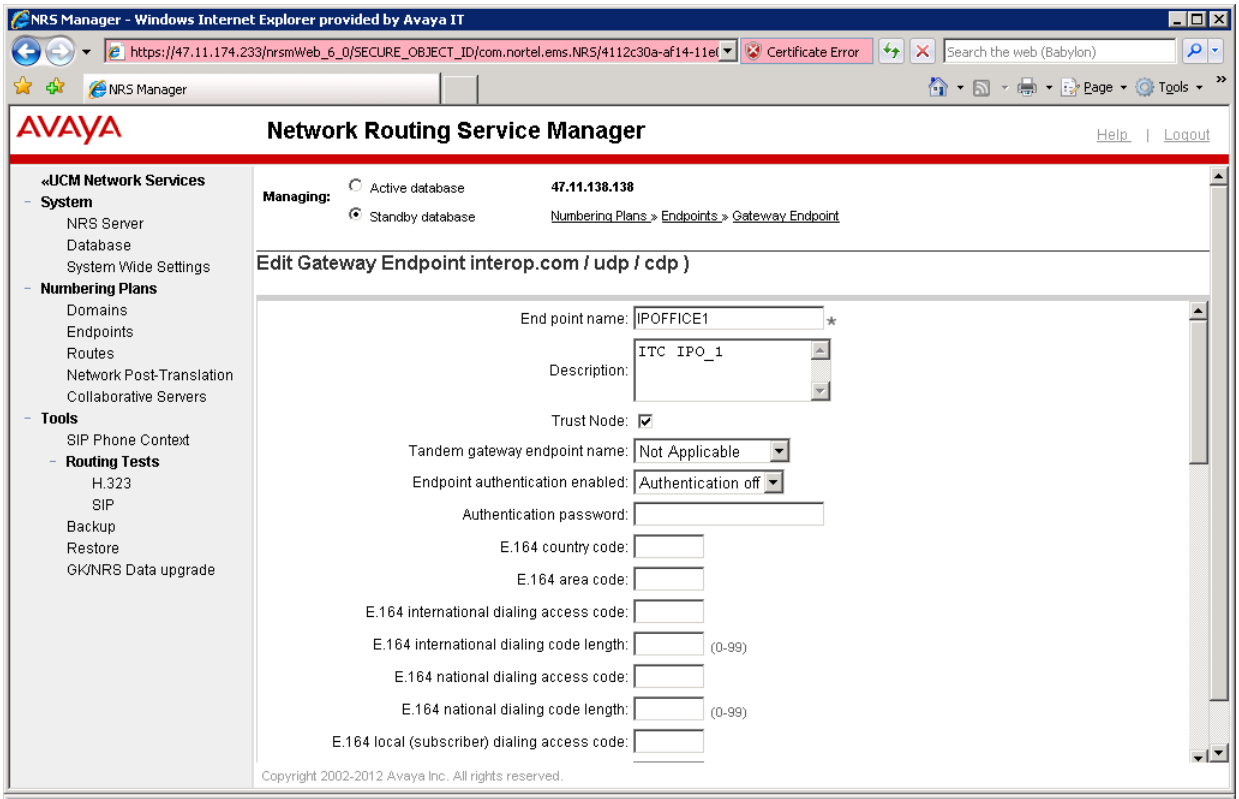

NRS Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.174.233/nrsmWeb_6_0/SECURE_OBJECT_ID/com.
Certificate Error
Search the web (Babylon)
NRS Manager
Page
Tools
AVAYA
Network Routing Service Manager
Help | Logout
«UCM Network Services
System
NRS Server
Database
System Wide Settings
Numbering Plans
Domains
Endpoints
Routes
Network Post-Translation
Collaborative Servers
Tools
SIP Phone Context
Routing Tests
H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
47.11.138.138
Numbering Plans » Domains
Domains
Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.
Service Domains (4)
L1 Domains (UDP) (3)
L0 Domains (CDP) (4)
Add...
Delete
Refresh
Domain Name
Description
# of L1 Domains
# of L0 Domains
# of Gateway Endpoints
1 interop.com Solutions Domain 1 2 75
2 ocs13.interop.com 1 1 1
3 soln.com MCS5100 0 0 0
4 testdomain.local Interop with Avaya 1 1 2
1 - 4 of 4 Service Domain(s)
Page 1 of 1
First| Previous| Next| Last
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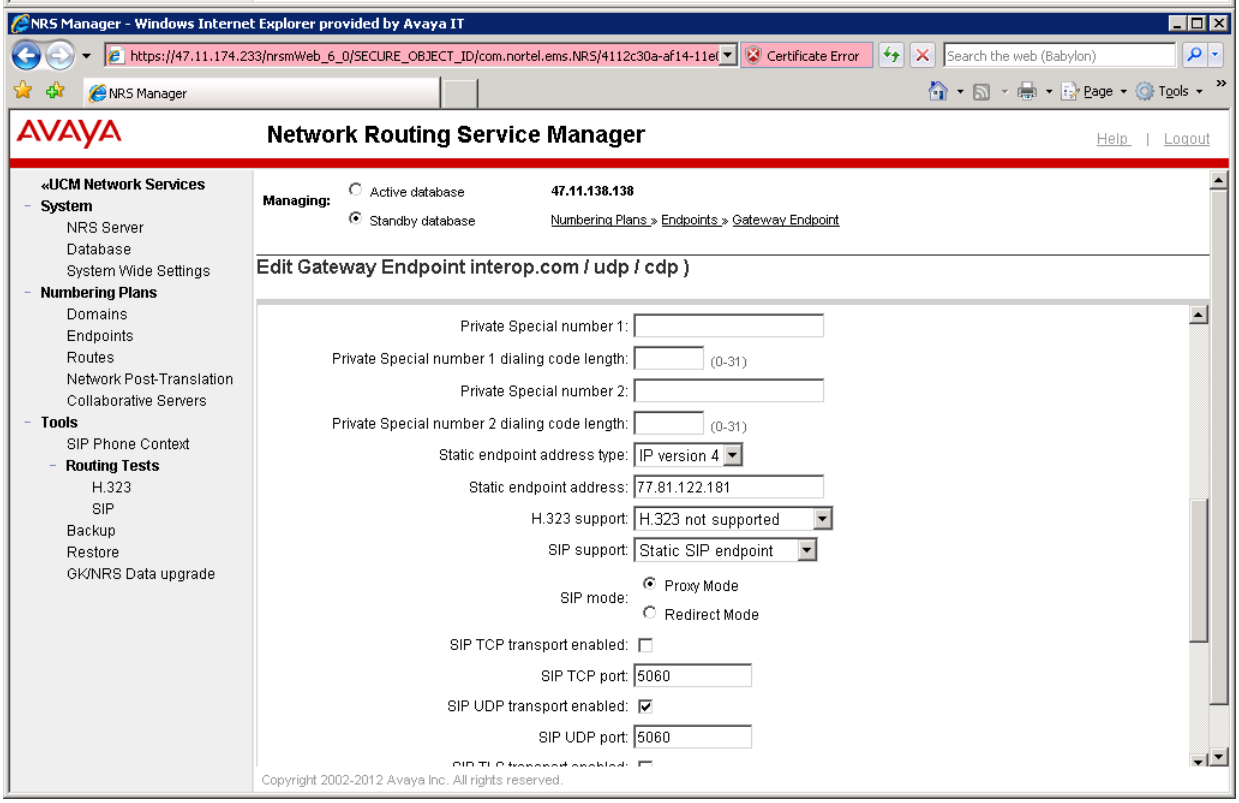

NRS Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.174.233/nrsmWeb_6_0/SECURE_OBJECT_ID/co
Certificate Error
Search the web (Babylon)
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Tools
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«UCM Network Services
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SIP Phone Context
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H.323
SIP
Backup
Restore
GK/NRS Data upgrade
Managing:
Active database
Standby database
47.11.138.138
Numbering Plans » Domains » Service Domains
Edit Service Domain
Domain name: interop.com *
Domain description: Solutions Domain
* Required value.
Save
Cancel
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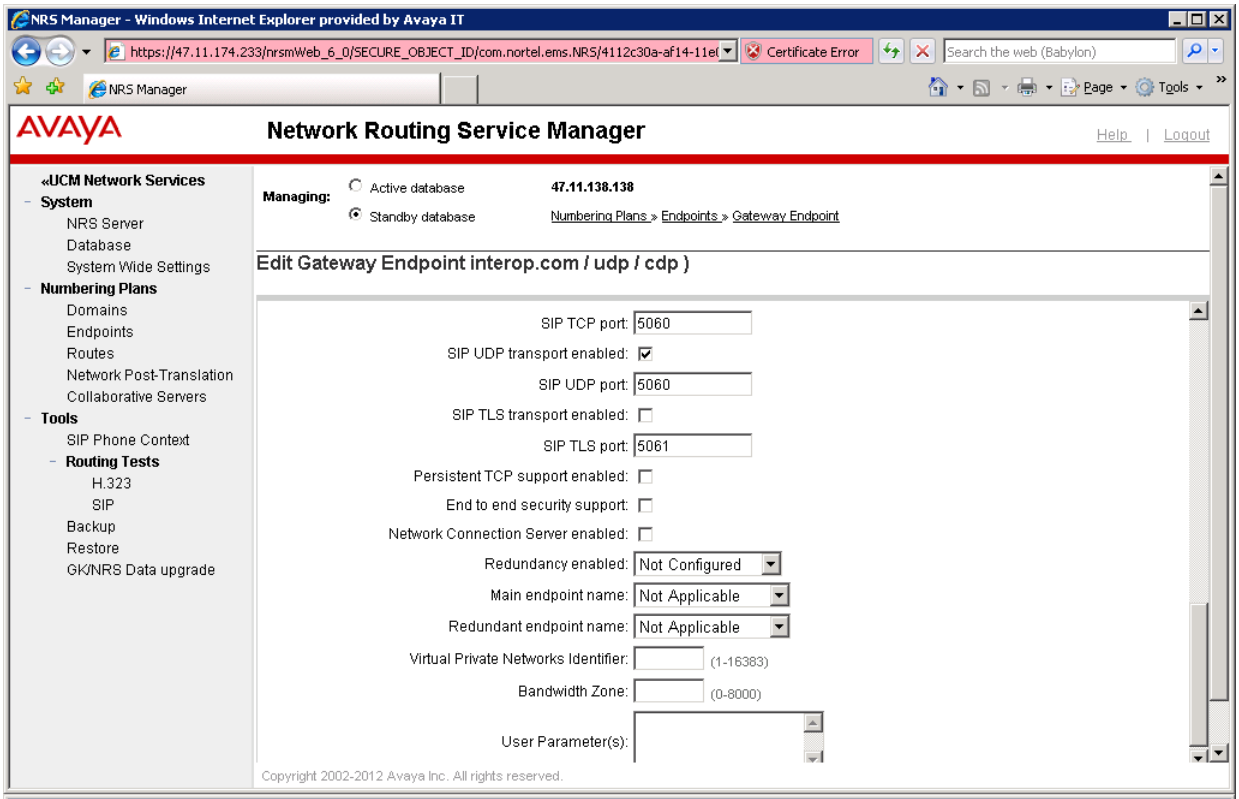
2.4.2 Configure Endpoints

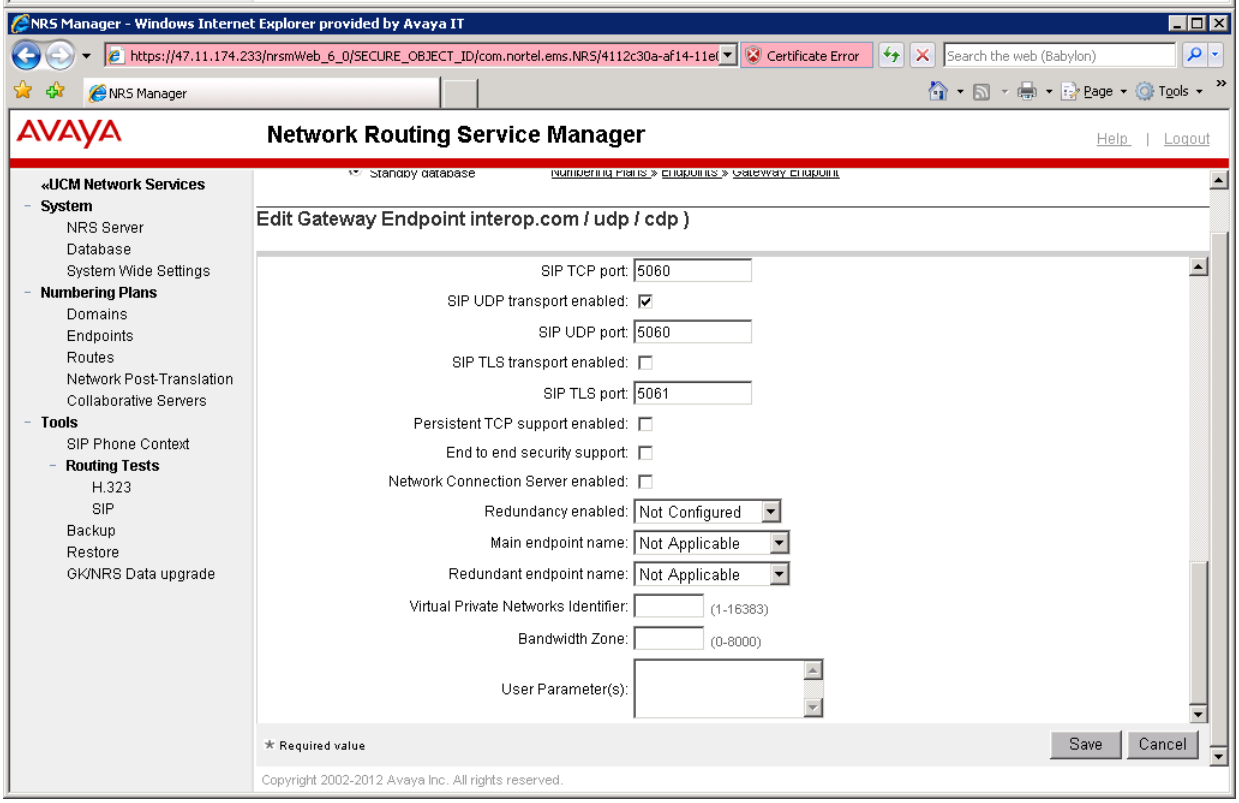
Go to **Numbering Plans > Endpoints** and click on **Add** button to define an endpoint.

IP Office endpoint:
- **End point name** - Enter a distinct name for this endpoint
- **Description –** A brief description can be entered, optional
- **Trust Node –** Ensure it is enabled
- **Tandem gateway endpoint name** - Ensure it is set to **Not applicable**
- **Endpoint authentication enabled –** Ensure it is set to **Authentication off**
- **Static endpoint address –** Enter the IP Address of the IP Office
- **H.323 support –** Ensure it is set to **H.323 not supported**
- **SIP Support -** Ensure it is set to **Static SIP endpoint**
- **SIP Mode –** Ensure it is set to **Proxy Mode**
- **SIP TCP transport enabled** – make sure the box is **disabled**.
- **SIP UDP transport enabled** – This box must be **enabled** as long as UDP is the supported protocol for SIP trunks with IP Office.
- **SIP UDP port –** Retain default value, **5060.**
- **SIP TLS transport enabled –** make sure the box is disabled.
- Retain default values for all other fields.
- Click on **Save** button when finish.


NRS Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.174.233/nrsmWeb_6_0/SECURE_OBJECT_ID/com.nortel.ems.NRS/4112c30a-af14-11e(
Certificate Error
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Managing:
Active database
Standby database
47.11.138.138
Numbering Plans » Endpoints » Gateway Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
End point name: IPOFFICE1
Description: ITC IPO_1
Trust Node:
Tandem gateway endpoint name: Not Applicable
Endpoint authentication enabled: Authentication off
Authentication password:
E.164 country code:
E.164 area code:
E.164 international dialing access code:
E.164 international dialing code length: (0-99)
E.164 national dialing access code:
E.164 national dialing code length: (0-99)
E.164 local (subscriber) dialing access code:
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NRS Manager - Windows Internet Explorer provided by Avaya IT
https://47.11.174.233/nrsmWeb_6_0/SECURE_OBJECT_ID/com.nortel.ems.NRS/4112c30a-af14-11e(
Certificate Error
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47.11.138.138
Numbering Plans » Endpoints » Gateway Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
Private Special number 1:
Private Special number 1 dialing code length: (0-31)
Private Special number 2:
Private Special number 2 dialing code length: (0-31)
Static endpoint address type: IP version 4
Static endpoint address: 77.81.122.181
H.323 support: H.323 not supported
SIP support: Static SIP endpoint
SIP mode: Proxy Mode
Redirect Mode
SIP TCP transport enabled:
SIP TCP port: 5060
SIP UDP transport enabled:
SIP UDP port: 5060
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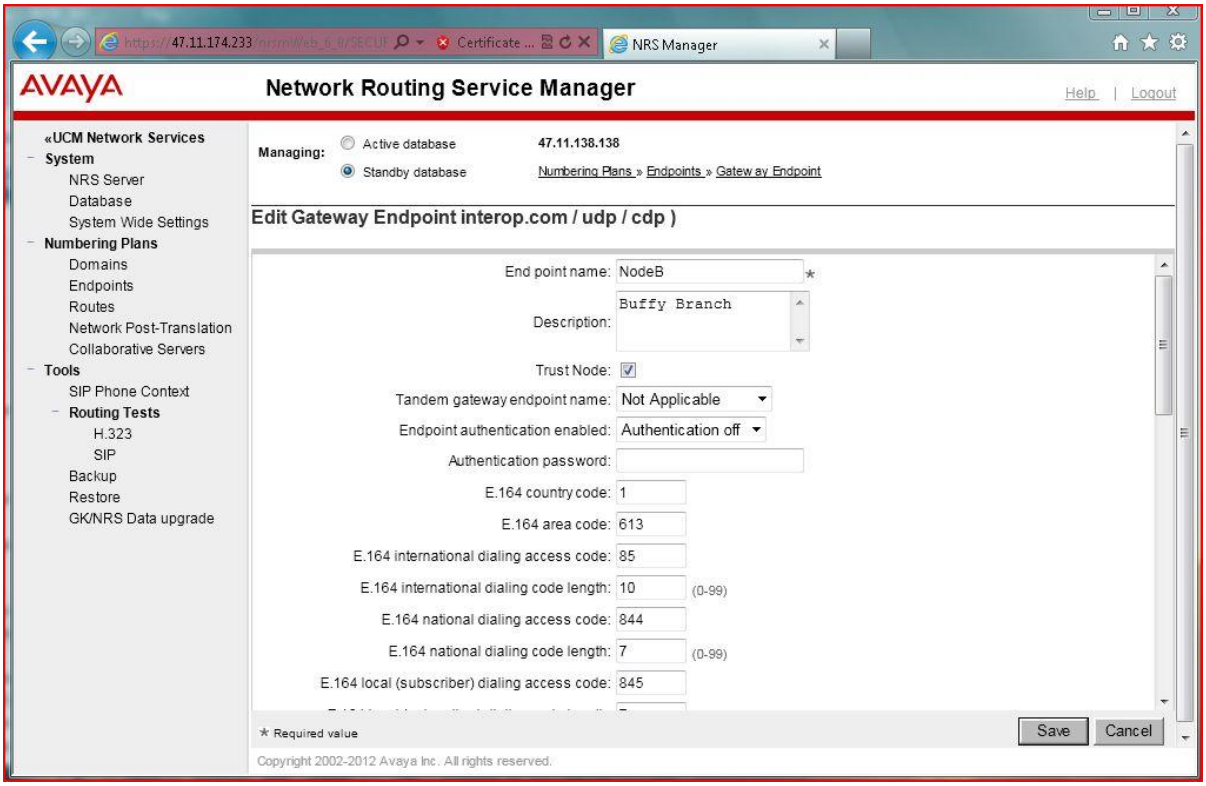


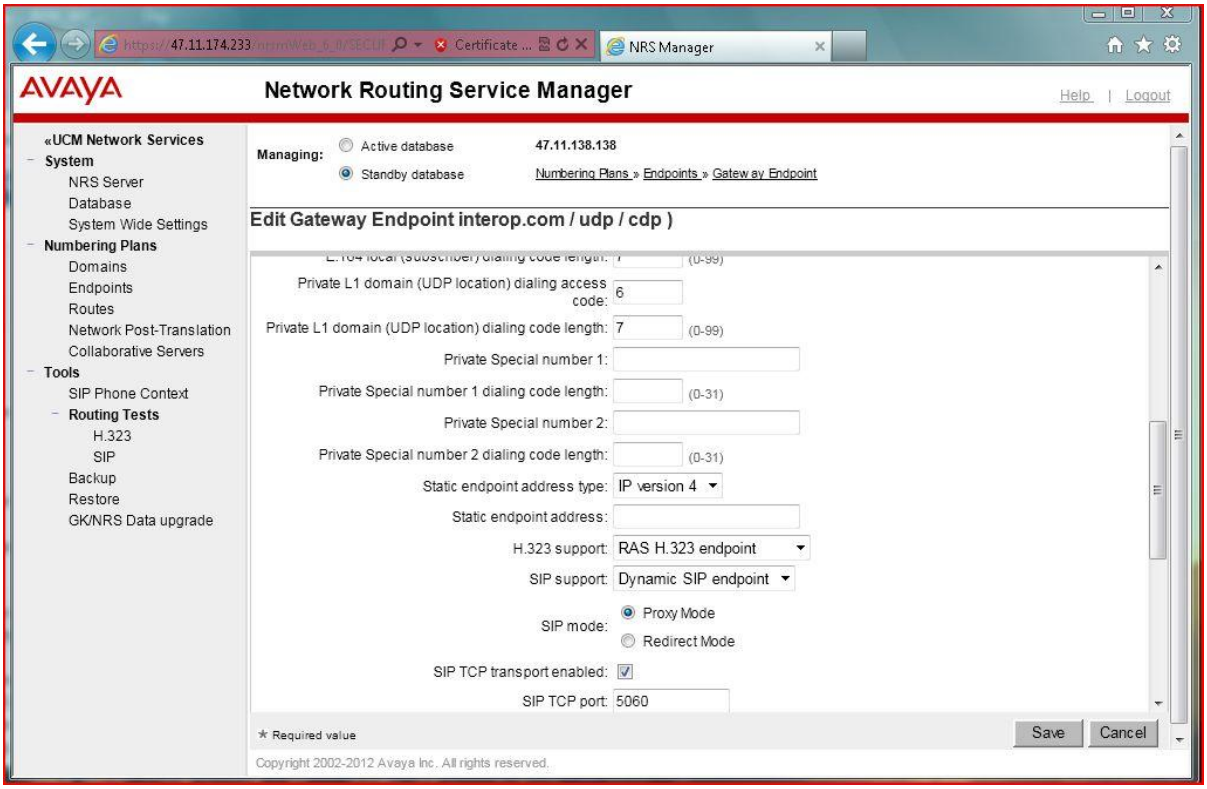


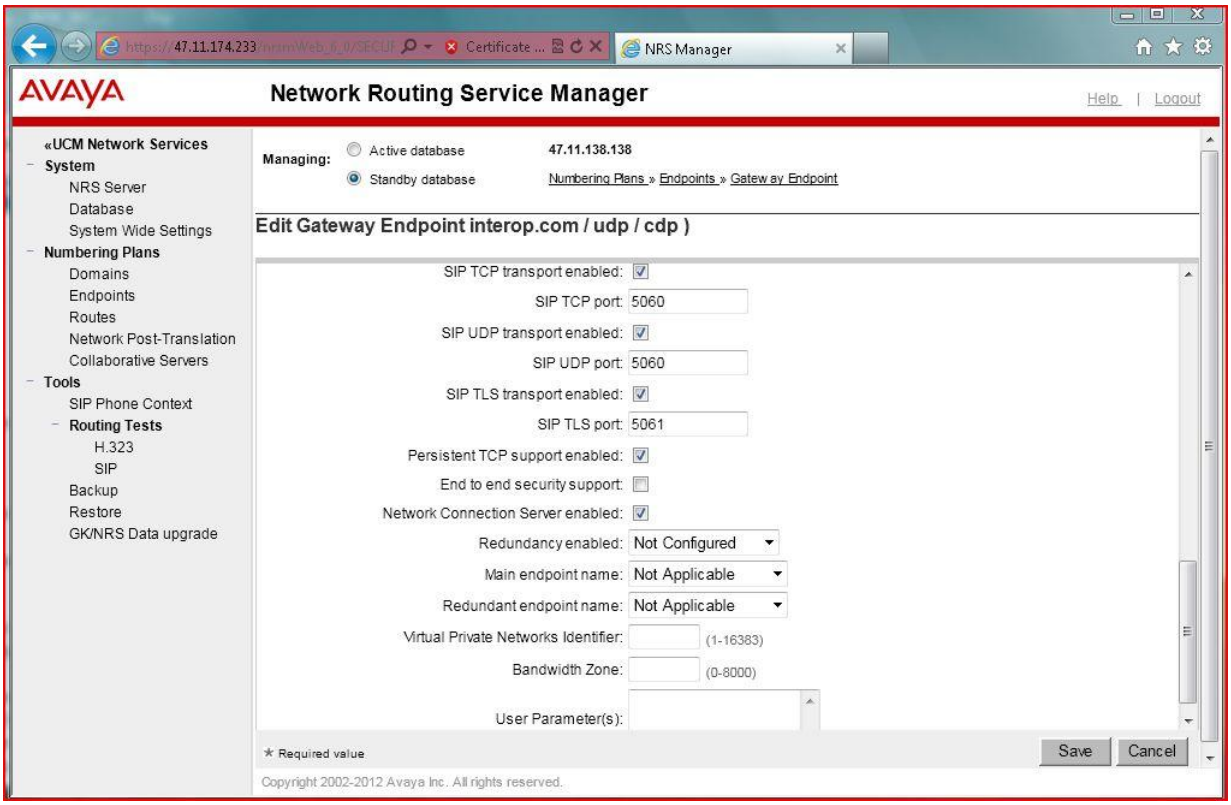
CS1000 endpoint:

- End point name - Enter a distinct name for this endpoint

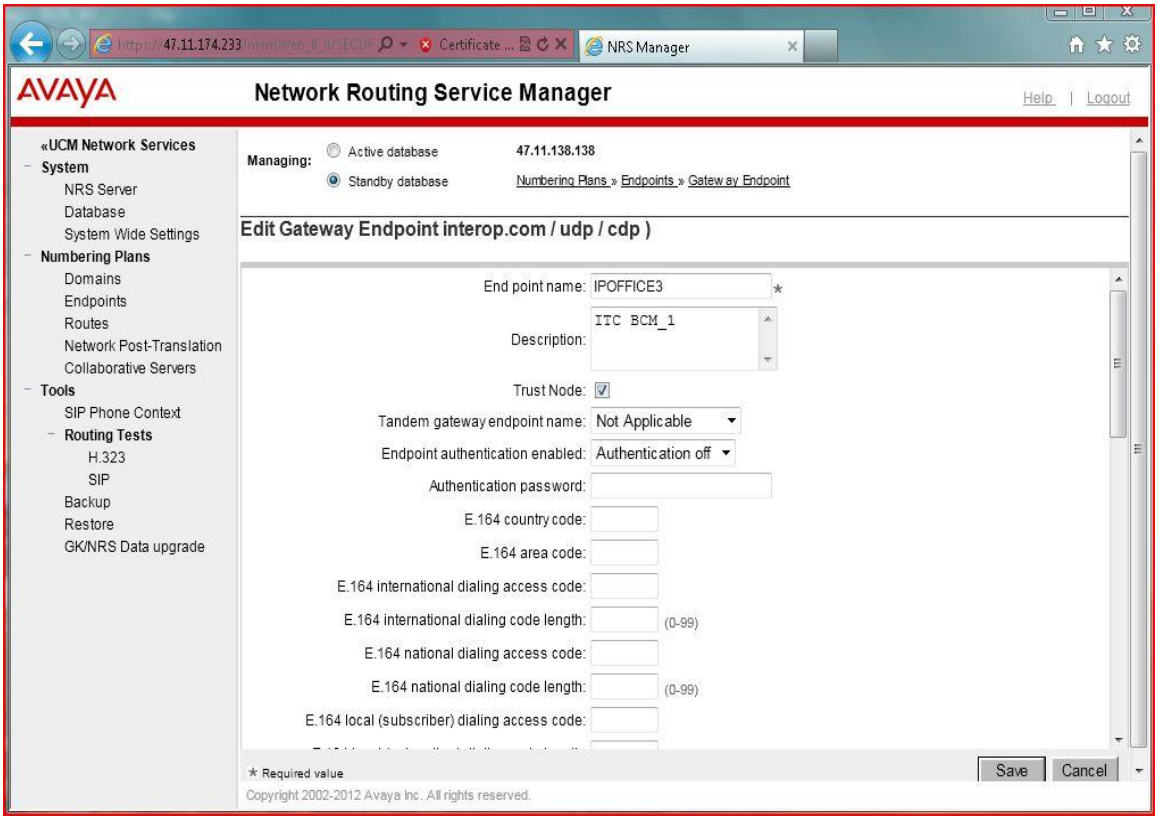
- **Description –** A brief description can be entered, optional
- **Trust Node –** Ensure it is enabled
- **Tandem gateway endpoint name** - Ensure it is set to **Not applicable**
- **Endpoint authentication enabled –** Ensure it is set to **Authentication off**
- **Static endpoint address –** Leave it blank.
- **H.323 support –** Ensure it is set to **H.323 not supported**.
- **SIP Support** - Ensure it is set to **Static SIP endpoint**.
- **SIP Mode –** Ensure it is set to **Proxy Mode**.
- Enter the required parameters according to your dial plan.
- Retain default values for all other fields.
- Click on **Save** button when finish.

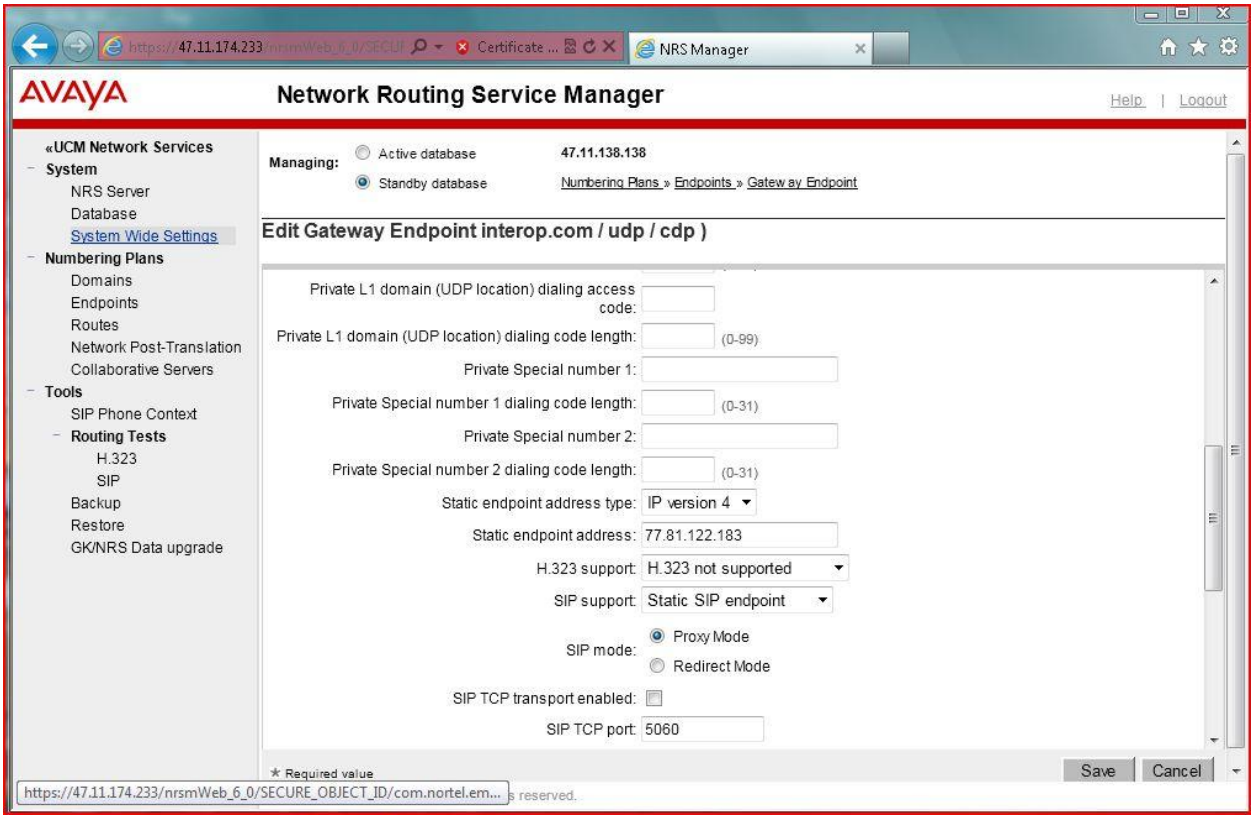


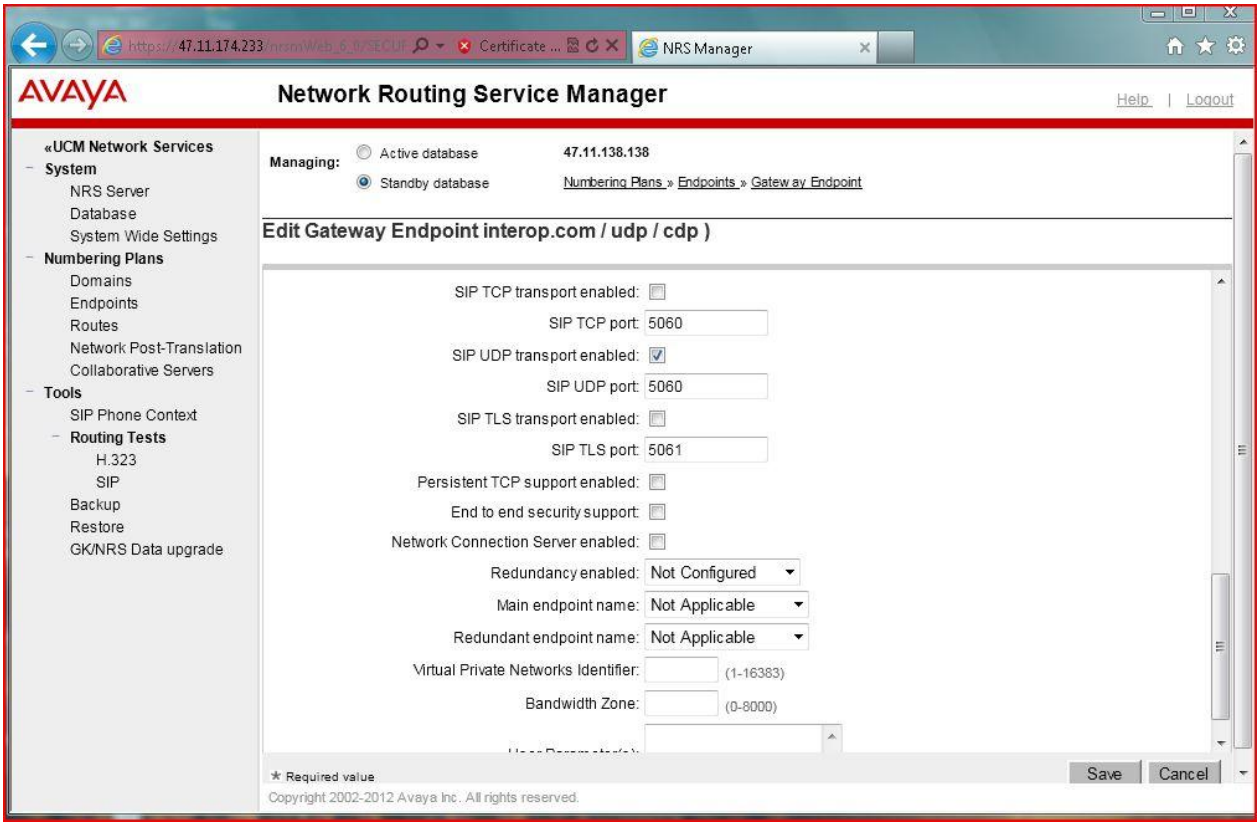

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Standby database
47.11.138.138
Numbering Plans » Endpoints » Gatew ay Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
Private L1 domain (UDP location) dialing access code: 6
Private L1 domain (UDP location) dialing code length: 7 (0-99)
Private Special number 1:
Private Special number 1 dialing code length: (0-31)
Private Special number 2:
Private Special number 2 dialing code length: (0-31)
Static endpoint address type: IP version 4
Static endpoint address:
H.323 support: RAS H.323 endpoint
SIP support: Dynamic SIP endpoint
SIP mode: Proxy Mode
Redirect Mode
SIP TCP transport enabled:
SIP TCP port: 5060
★ Required value
Save
Cancel
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Managing:
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Standby database
47.11.138.138
Numbering Plans » Endpoints » Gatew ay Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
SIP TCP transport enabled:
SIP TCP port: 5060
SIP UDP transport enabled:
SIP UDP port: 5060
SIP TLS transport enabled:
SIP TLS port: 5061
Persistent TCP support enabled:
End to end security support:
Network Connection Server enabled:
Redundancy enabled: Not Configured
Main endpoint name: Not Applicable
Redundant endpoint name: Not Applicable
Virtual Private Networks Identifier: (1-16383)
Bandwidth Zone: (0-8000)
User Parameter(s):
★ Required value
Save
Cancel
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BCM endpoint:
- **Endpoint name** - Enter a distinct name for this endpoint
- **Description –** A brief description can be entered, optional.
- **Trust Node –** Ensure it is enabled.
- **Tandem gateway endpoint name** - Ensure it is set to **Not applicable**
- **Endpoint authentication enabled –** Ensure it is set to **Authentication off**
- **Static endpoint address –** Enter the IP Address of the BCM system.
- **H.323 support –** Ensure it is set to **H.323 not supported**.
- **SIP Support** - Ensure it is set to **Static SIP endpoint**.
- **SIP Mode –** Ensure it is set to **Proxy Mode**.
- **SIP TCP transport enabled** – make sure the box is **disabled**.
- **SIP UDP transport enabled** – This box must be **enabled** as long as UDP is the only supported protocol for BCM SIP trunks.
- **SIP UDP port –** Retain default value, **5060.**
- **SIP TLS transport enabled –** make sure the box is disabled.
- Retain default values for all other fields.
- Click on **Save** button when finish.




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Managing:
Active database
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47.11.138.138
Numbering Plans » Endpoints » Gatew ay Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
Private L1 domain (UDP location) dialing access code:
Private L1 domain (UDP location) dialing code length:
(0-99)
Private Special number 1:
Private Special number 1 dialing code length:
(0-31)
Private Special number 2:
Private Special number 2 dialing code length:
(0-31)
Static endpoint address type:
IP version 4
Static endpoint address:
77.81.122.183
H.323 support:
H.323 not supported
SIP support:
Static SIP endpoint
SIP mode:
Proxy Mode
Redirect Mode
SIP TCP transport enabled:
SIP TCP port:
5060
Required value
Save
Cancel
https://47.11.174.233/nrsmWeb_6_0/SECURE_OBJECT_ID/com.nortel.em...

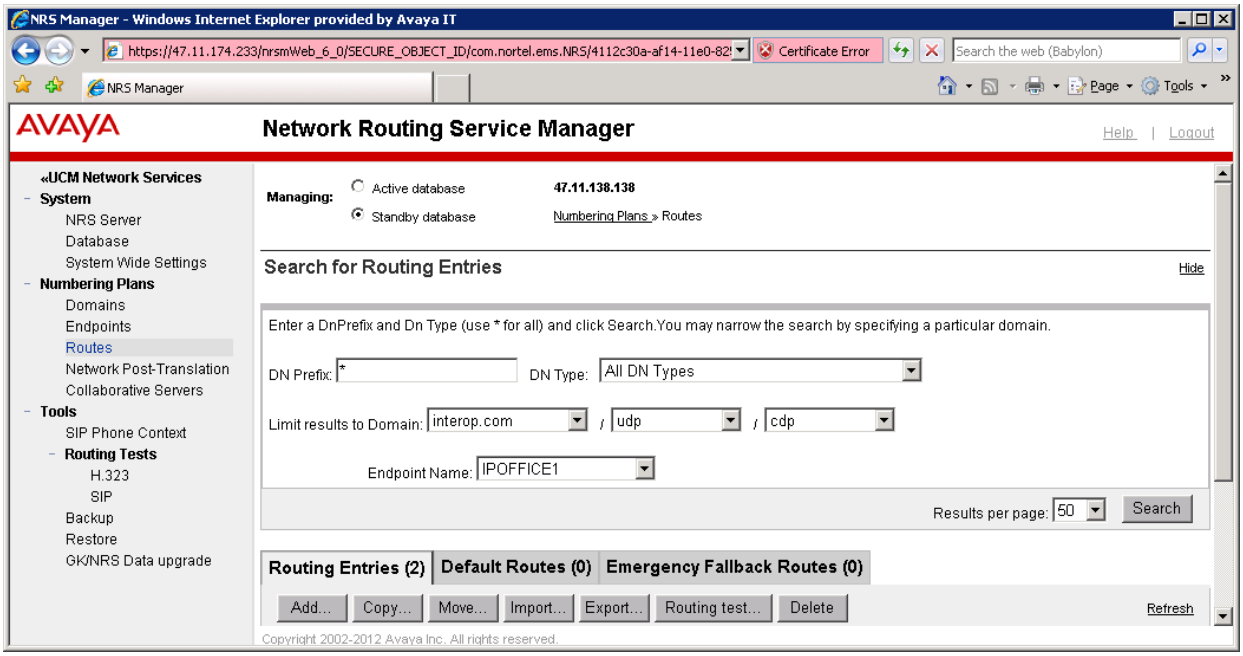

NRS Manager
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Managing:
Active database
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47.11.138.138
Numbering Plans » Endpoints » Gatew ay Endpoint
Edit Gateway Endpoint interop.com / udp / cdp)
SIP TCP transport enabled:
SIP TCP port:
5060
SIP UDP transport enabled:
SIP UDP port:
5060
SIP TLS transport enabled:
SIP TLS port:
5061
Persistent TCP support enabled:
End to end security support:
Network Connection Server enabled:
Redundancy enabled:
Not Configured
Main endpoint name:
Not Applicable
Redundant endpoint name:
Not Applicable
Virtual Private Networks Identifier:
(1-16383)
Bandwidth Zone:
(0-8000)
Required value
Save
Cancel
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2.4.3 Configure routing

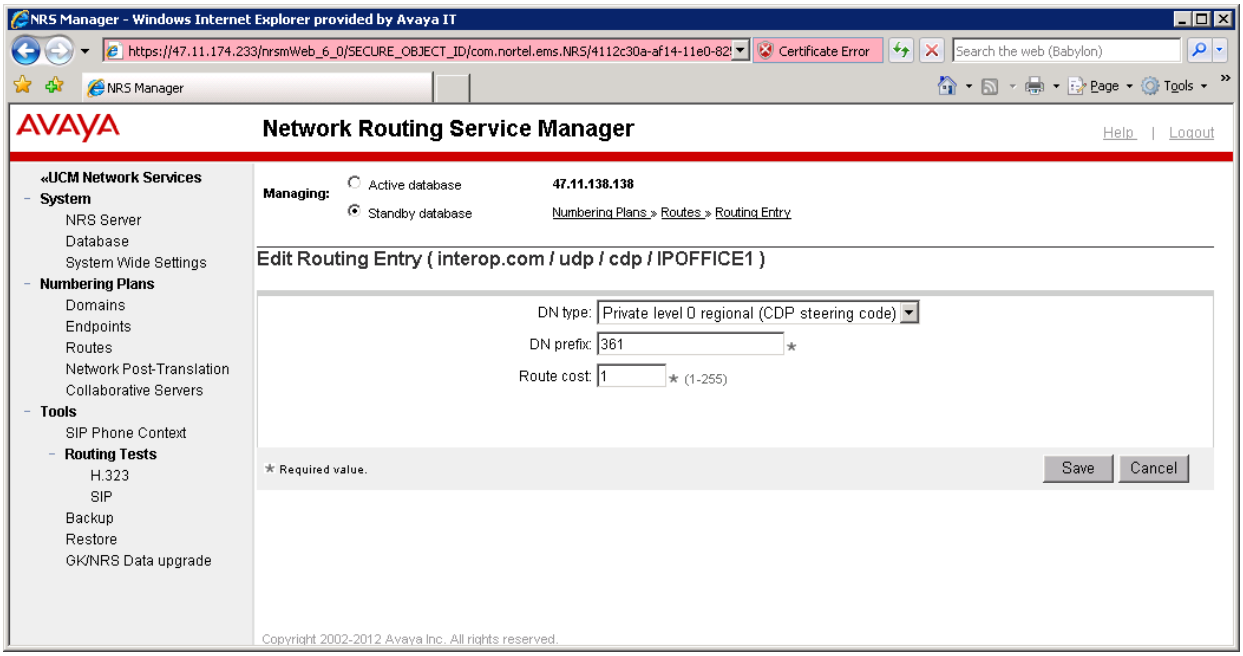
The NRS is the central location where the numbering plan information is configured. The identity of each endpoint is configured in the NRS with the numbers it can reach.
Routing entries are telephone numbers associated with an endpoint. When a telephone number is dialed, the NRS searches the endpoint database to find a match and then directs the call to the endpoint with the first returned match.

Go to **Numbering Plans > Routing**.
- Filter the results by introducing the related domains, interop.com / udp /cdp, in the **Limits results to Domain** field.
- Select the **Endpoint Name** from the drop-down list.
- **Add** button is activated and a new route can be added.

2.4.3.1 IP Office Routing entries:



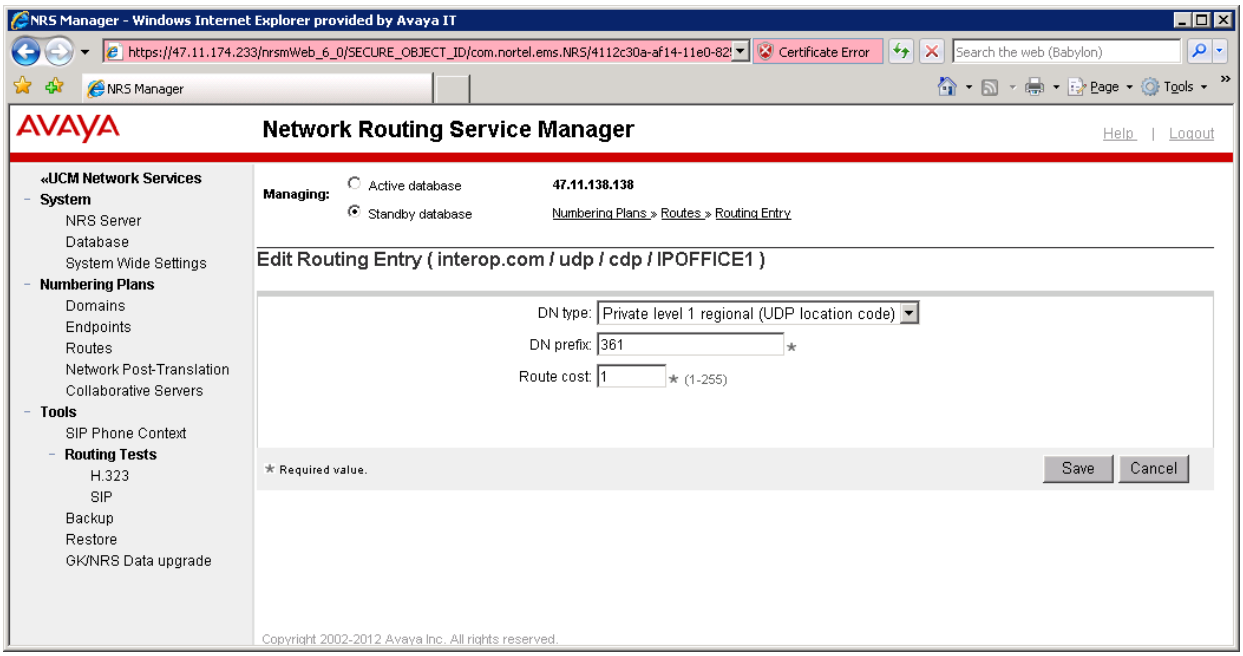
L0 domain:
- Click on the **Add** button and enter the route data:
- **DN Type –** Select **Private level 0 regional (CDP steering code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 361 in our case.
- **Route cost –** It should be left to default, 1.



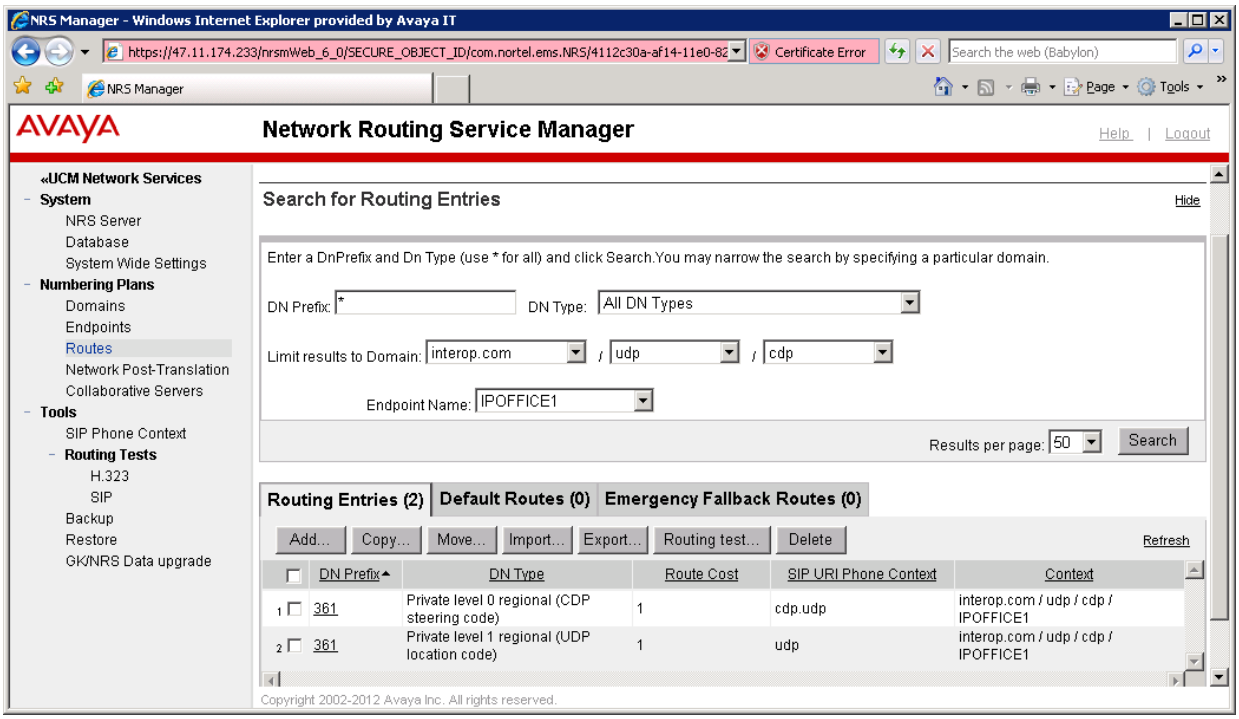
L1 domain:

- Click on the **Add** button and enter the route data:

- **DN Type –** Select **Private level 1 regional (UDP location code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 361 in our case.
- **Route cost –** It should be left to default, 1.



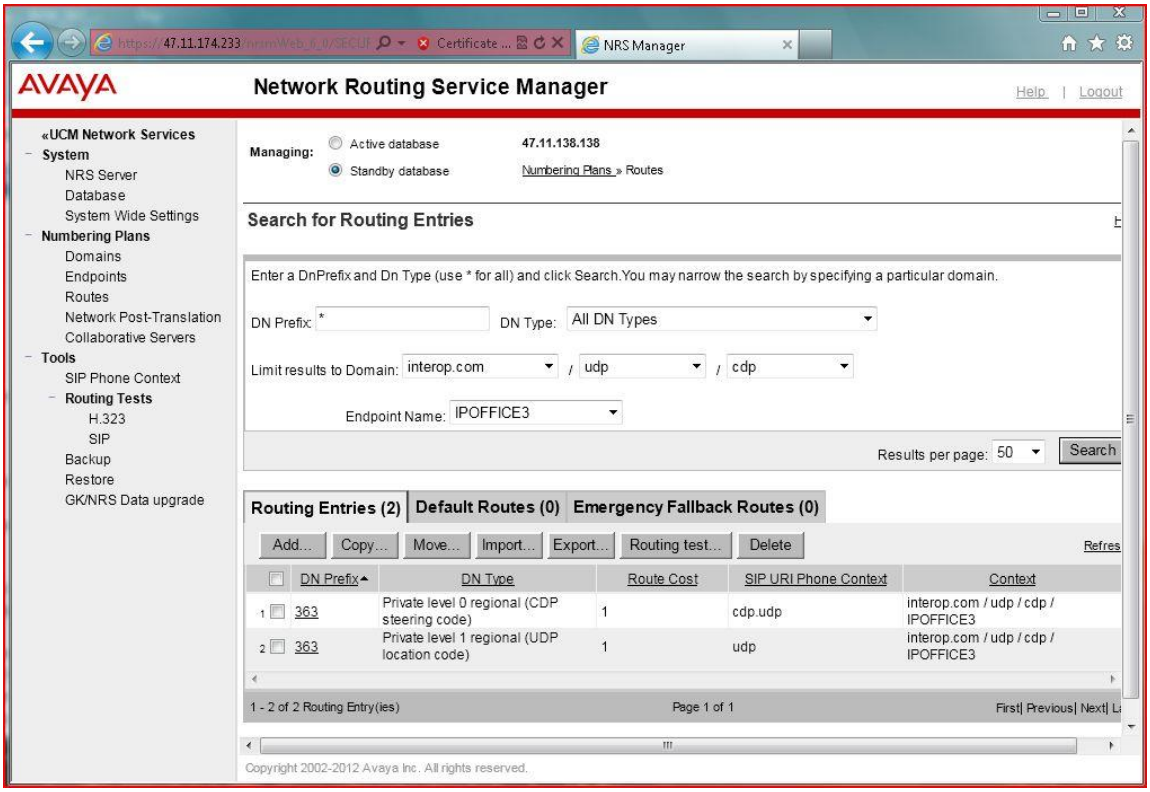
Click the **Save** button when finished.



2.4.3.2 BCM Routing entries

In the same manner we configure the routing entries for BCM.
L0 domain:
- Click on the **Add** button and enter the route data:
- **DN Type –** Select **Private level 0 regional (CDP steering code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 363 in our case.
- **Route cost –** It should be left to default, 1.

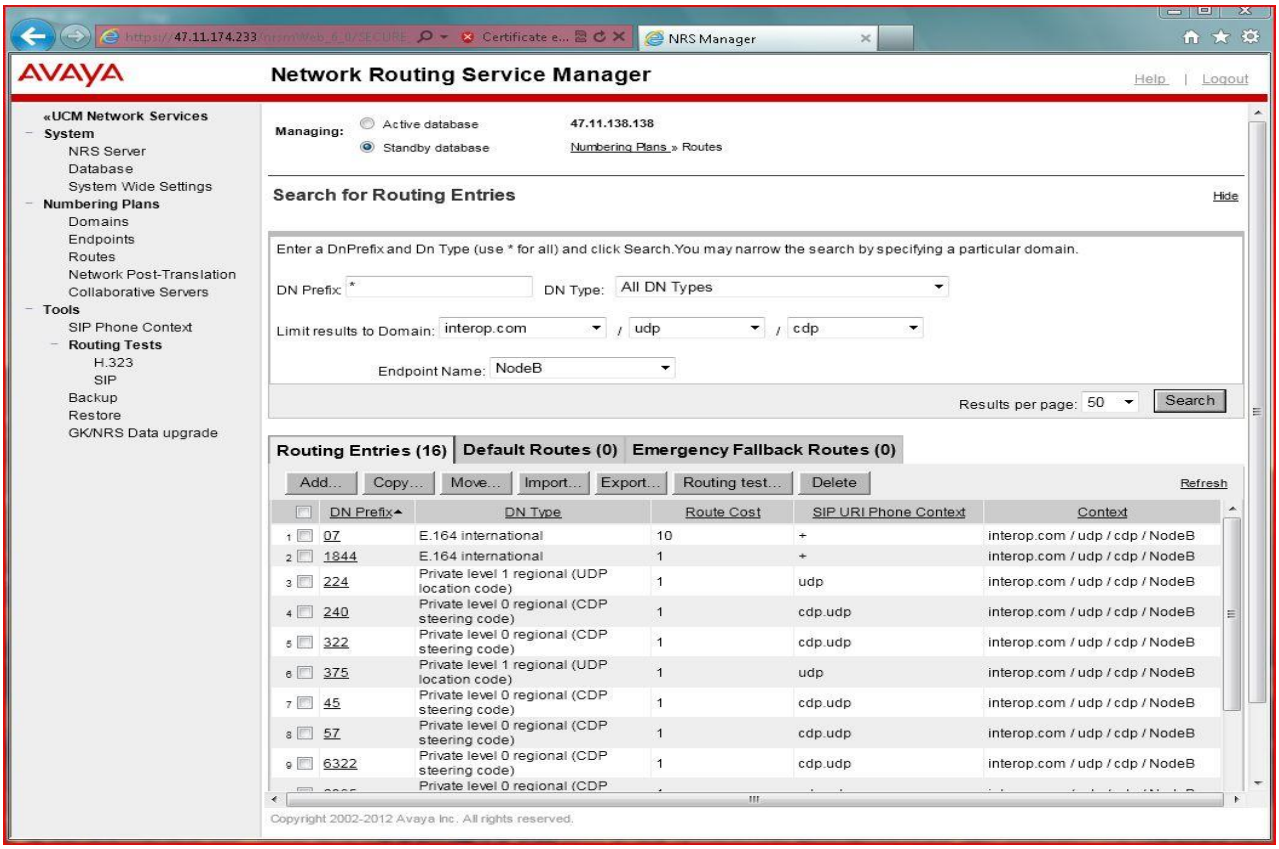
L1 domain :
- Click on the **Add** button and enter the route data:
- **DN Type –** Select **Private level 1 regional (UDP location code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 363 in our case.
- **Route cost –** It should be left to default, 1.
- Click the **Save** button when finished.



2.4.3.2 CS1K Routing entries

L0 domain:
- Click on the **Add** button and enter the route data:
- **DN Type –** Select **Private level 0 regional (CDP steering code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 638 our case.
- **Route cost –** It should be left to default, 1.

L1 domain:
- Click on the **Add** button and enter the route data:
- **DN Type –** Select **Private level 1 regional (UDP location code)** from the drop-down list.
- **DN Prefix –** Enter the digits related to IP Office endpoint – 638 our case.
- **Route cost –** It should be left to default, 1.
- Click the **Save** button when finished.



2.4.4 Updating the database

- From NRS Manager go to **System > Database**
- Click the **Cut over** button – The Cut over command is issued and the database is placed into a Switched over state.
- Click the **Commit** button. The database is placed into the Committed state. After a database Cut over, the Commit command copies data from the active database to the standby database. The previous configuration data is overwritten with the new configuration data. The standby database is synchronized with the active database.

3.0 Found Issues

IPOFFICE-28404 - One-Way speech path when BCM places an IP Office Call on Hold and "Enable SDP OPTIONS query" is active
IPOFFICE-27611 - Wrong CLID/Named displayed when CS1K FwdNoAns an IPOffice call back to IP Office
IPOFFICE-27584 - No talk path when BCM blind transfers an IP Office call back to IP Office
IPOFFICE-27582 - One way talk path when BCM blind transfers an IP Office call to CS 1000 via NRS

IPOFFICE-26887 -Hold Music is not always played when CS1K phone holds an IP Office call

4. Limitations

- Avaya IP Office doesn't support SIP trunk over **TCP** with NRS. When TCP is used, as transport protocol, issues related to call clearing or call signaling can be encountered.
- BCM – When "Enable SDP options query" is active for the Public SIP trunk under account, advanced settings, talk-path issue can be encountered.
- In case of using BCM50e, it is not supported to use the Embedded Router to route the telephony traffic (local LAN) to NRS over WAN port. An additional router should be use to route the traffic in case that NRS is not apart of BCM Telephony LAN.
- IP Office SIP terminals, 11xx,12xx acts different when transferring calls, meaning that when transferring a call, the call is presented with SIP(11xx,12xx) CLID information, instead of the transferred party's CLID, and this information is no updated after the transfer is completed.
- Voice Talk Path issues can happens if Audio Codecs are not coordinated between the systems.

….