Configuring Interoperability between Avaya IP Office and Avaya Communication Manager

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Contents

1.0 Introduction

This document provides a description of the solution where a network of Avaya IP Office systems is connected to a network of Avaya Communication Manager systems through an H323 line.

1.1 Supported Features

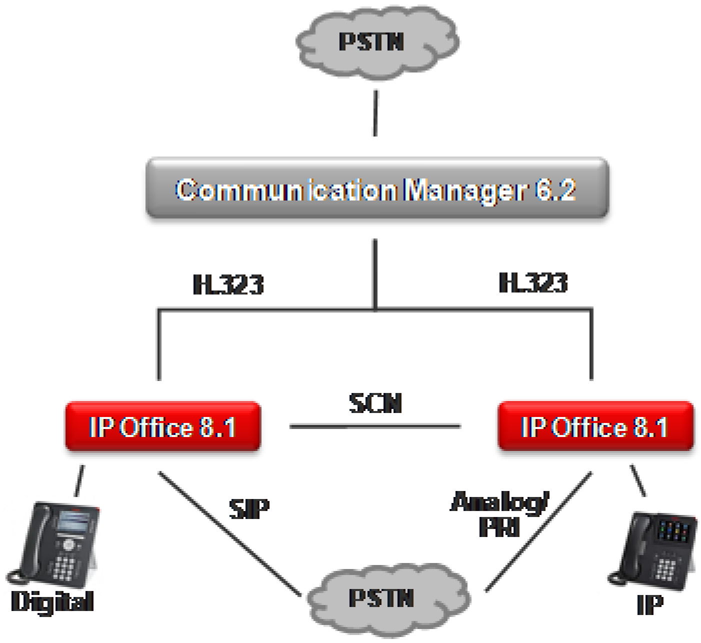
- **Basic Call** – We verified that:
    - Caller receives Ring-back tone
    - Basic calls can be successfully established between IP Office and BCM with two-way talk path.
    - Proper Media Type is used.
- **Basic Call Completion** – We verified that:
    - Terminated calls are properly cleared from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

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

- **Follow Me Here/ Follow Me To** – We verified that:
  - The SIP Trunk calls are successfully redirected based on the Follow Me Here and Follow Me To features.
  - CLID and Contact Name are successfully updated.
- **Twinning** – We verified that:
  - The SIP Trunk calls are successfully redirected based on the Twinning feature (verified only on the IP Office side)

- **Conferencing** – We verified:
  - Both types of conference, Ad hoc and Conference Meet Me.
  - Ad hoc conferences with 3 to 6 parties.
  - That each party has two way speech path.
  - Trunk and resource utilization.
  - Voice quality.
- **PSTN Toll Bypass** – We verified:
  - That PSTN Toll Bypass calls can be successfully established with two-way talk path.
  - That the CLID is properly presented.
  - Involved trunks and resources are properly released when a call is terminated.

1.2 Network Diagram



1.3 Supported Phones

IP Office Phones	
Types	**Series**
DS	1400
	5400
	2400
	9500
	M-series
	T-series
	T3
SIP	1100e
	1200
	Softphone
	A1010/1020/1030/1040
	B179 - Konftel
	3-rd party
H323	1600
	3600
	4600
	5600
	9600
	96x1

	T3
DECT	ASCOM
Analog	B149 - Konftel

Communication Manager **Phones**	
Types	**Series**
H323	9600
	96x1
SIP	9600
	96x1
	A1010/1020/1030/1040

1.4 Software Version

- IP Office 8.1 running Essential, Preferred or Advanced Edition.
- Communication Manager 6.2

1.5 Hardware Platforms

- IP Office 500v2
- IP Office Mid Market

2.0 Configuration Guide

2.1 Avaya IP Office Configuration

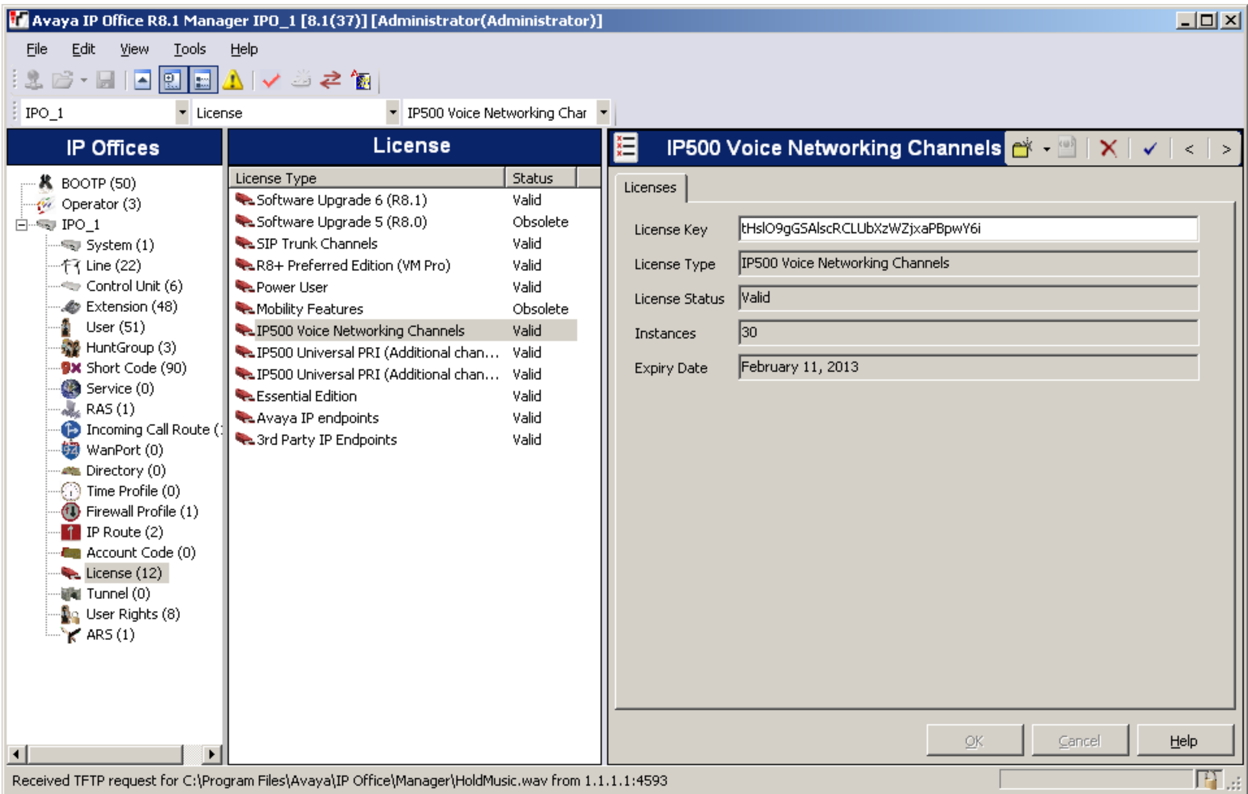
This section provides the procedures for configuring Avaya IP Office using the Avaya IP Office Manager application.

The procedure covers the following areas:

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

2.1.1 IP Office H323 Trunk Licensing

From the configuration tree in the left pane, select **License > IP500 Voice Networking Channels** to display the **IP500 Voice Networking Channels** screen in the right pane. Verify that the **License Status** is ‘Valid’ and that the **Instances** field contains an appropriate number for the system.

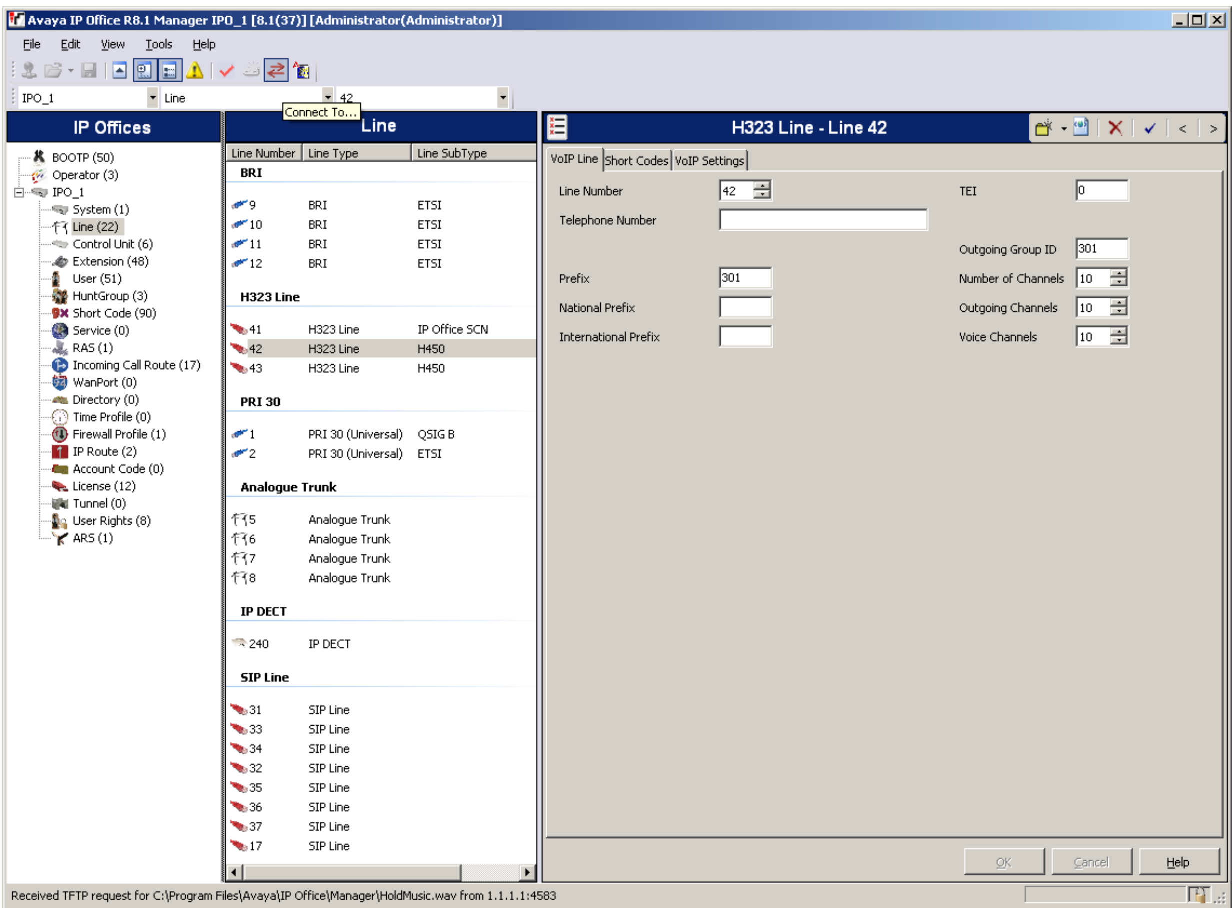


2.1.2 IP Office H323 Line to COMMUNICATION MANAGER

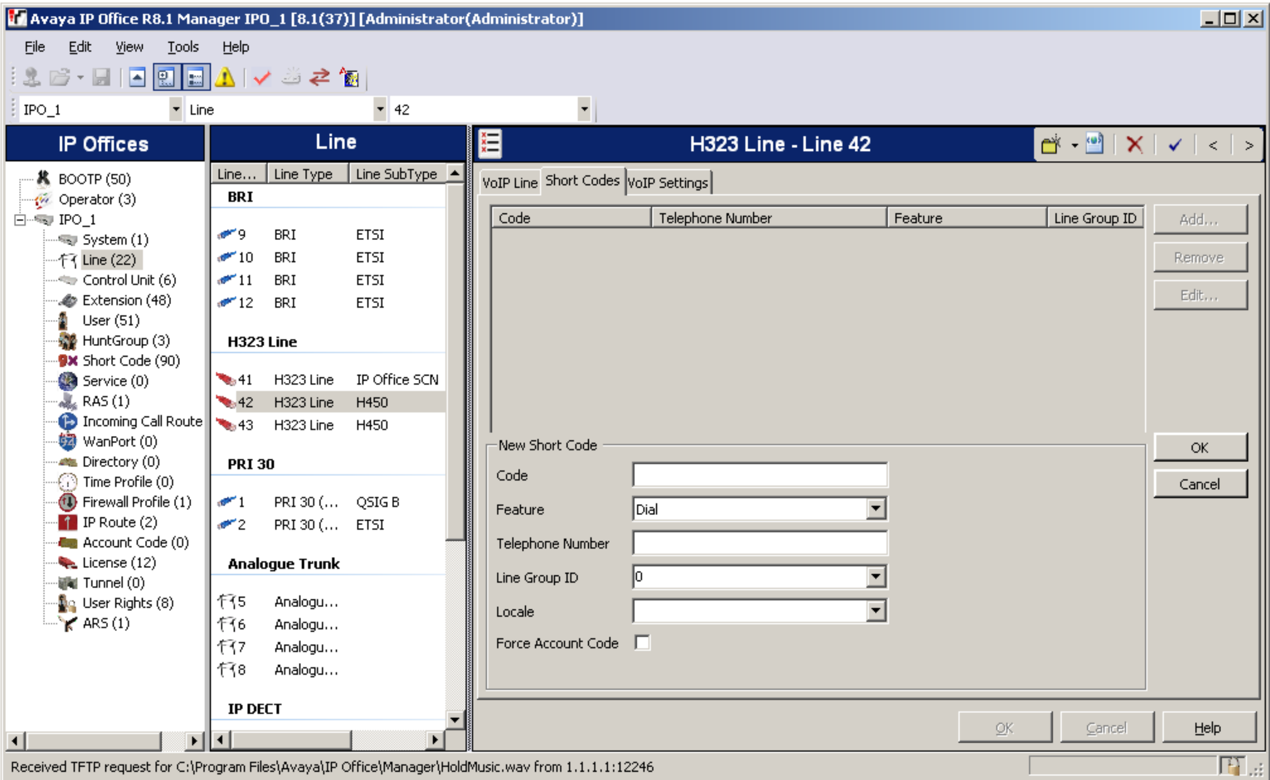
From the configuration tree in the left pane, right-click on **Line** and select **New > H323 Line** to add a new H323 Trunk.

In the **VoIP Line** tab:
- Adjust the **Line Number** as you desire, this must be unique.
- Change default value for **Outgoing Group ID** into a different number. This will be required when defining the short code used to place outgoing calls to this trunk.
- The default values should be retained for all other fields.
- The **Terminal Equipment Identifier** parameter **-** Used to identify each Control Unit connected to a particular ISDN line. For Point to Point lines this is typically (always) 0.

- The **Number of Channels** parameter - Defines the number of operational channels that are available on this line.
- The **Outgoing Channels** parameter - This defines the number of channels available, on this line, for outgoing calls. This should normally be the same as Number of Channels field, but can be reduced to ensure incoming calls cannot be blocked by outgoing calls.
- The **Voice Channels** parameter -The number of channels available for voice use.
- The **Prefix** field – The ISDN messaging tags indicates the call type (National, International or Unknown). If the call type is unknown, then the number in the Prefix field is added to the ICLID.
- The **National Prefix** field – This indicates the digits to be prefixed to an incoming national call.
- The **International Prefix** field - This indicates the digits to be prefixed to an incoming international call.



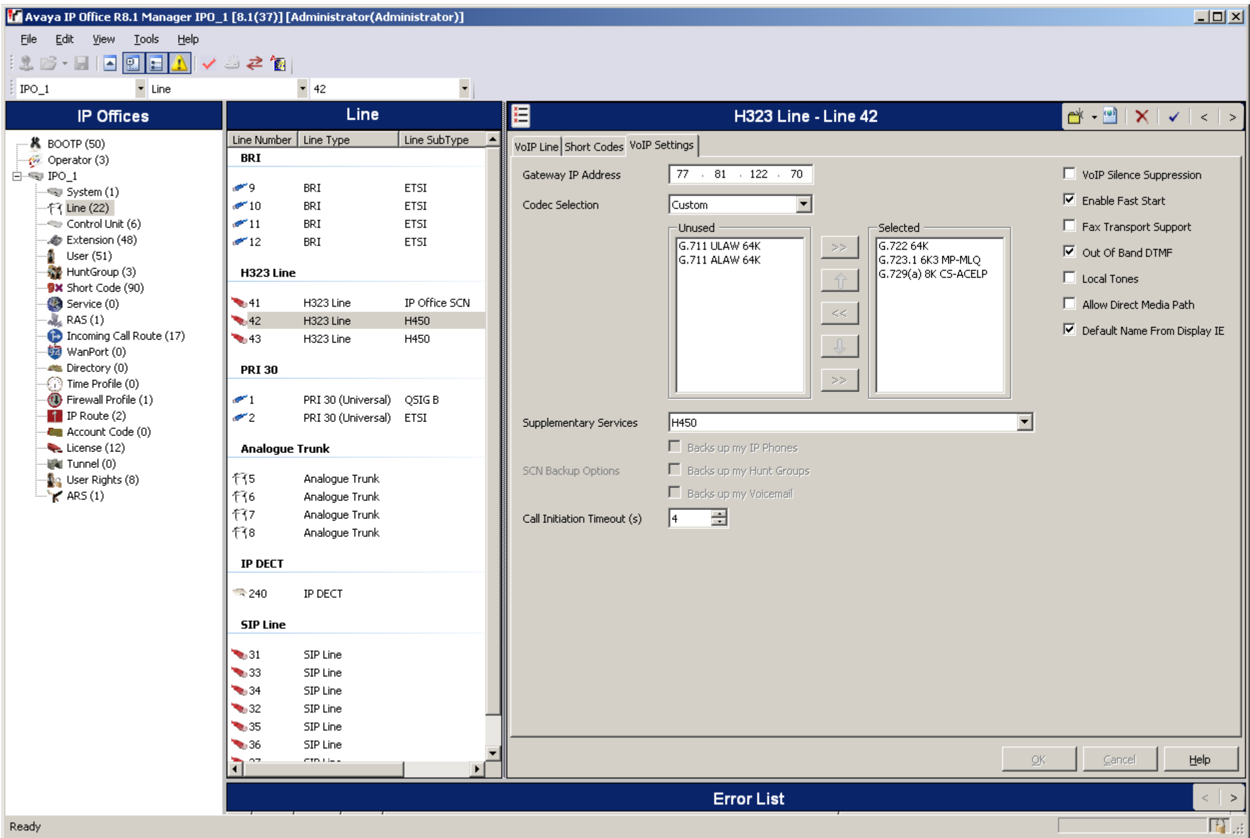
In the **Short codes** tab:
- Line short codes can be applied to the digits received with incoming call.
- Short codes can be added and edited using the **Add**, **Remove** and **Edit** buttons.



In the **VoIP Settings** tab:

- **Codec Selection** - this field defines the codecs 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. Ensure that G.711 Alaw and G.711 Ulaw are excluded as long as Communication Manager doesn't support them anymore.
- **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 Communication Manager.
- **Enable Fast Start –** It should be enabled. A fast connection procedure reduces the number of messages that need to be exchanged before an audio channel is created.
- **Fax Transport Support** – The default setting should be retained, disabled.
- **Out Of Band DTMF –** It should be enabled. DTMF tones can be sent to the remote end either as DTMF tones within the calls audio path (**In Band**) or as separate signals (**Out of Band**). Out of Band is recommended for compression modes such as G.729 and G.723 compression modes where DTMF in the voice stream could become distorted.
- **Local Tones –** It should be disabled. When selected, the tones are generated by the local IP Office system to which the phone is registered.
- **Default Name From Display IE –** It should be **Enabled.**
- **Allow Direct Media Path –** It should be disabled. This setting controls whether H323 calls must be routed via the H323 gatekeeper (the IP Office) or can be routed alternately if possible within the network structure.

- **Supplementary Services –** It should be set to **QSIG**. This is the supplementary service signaling method supported for an H323 trunk with Communication Manager. The remote end of the trunk must support the same option.
- **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.



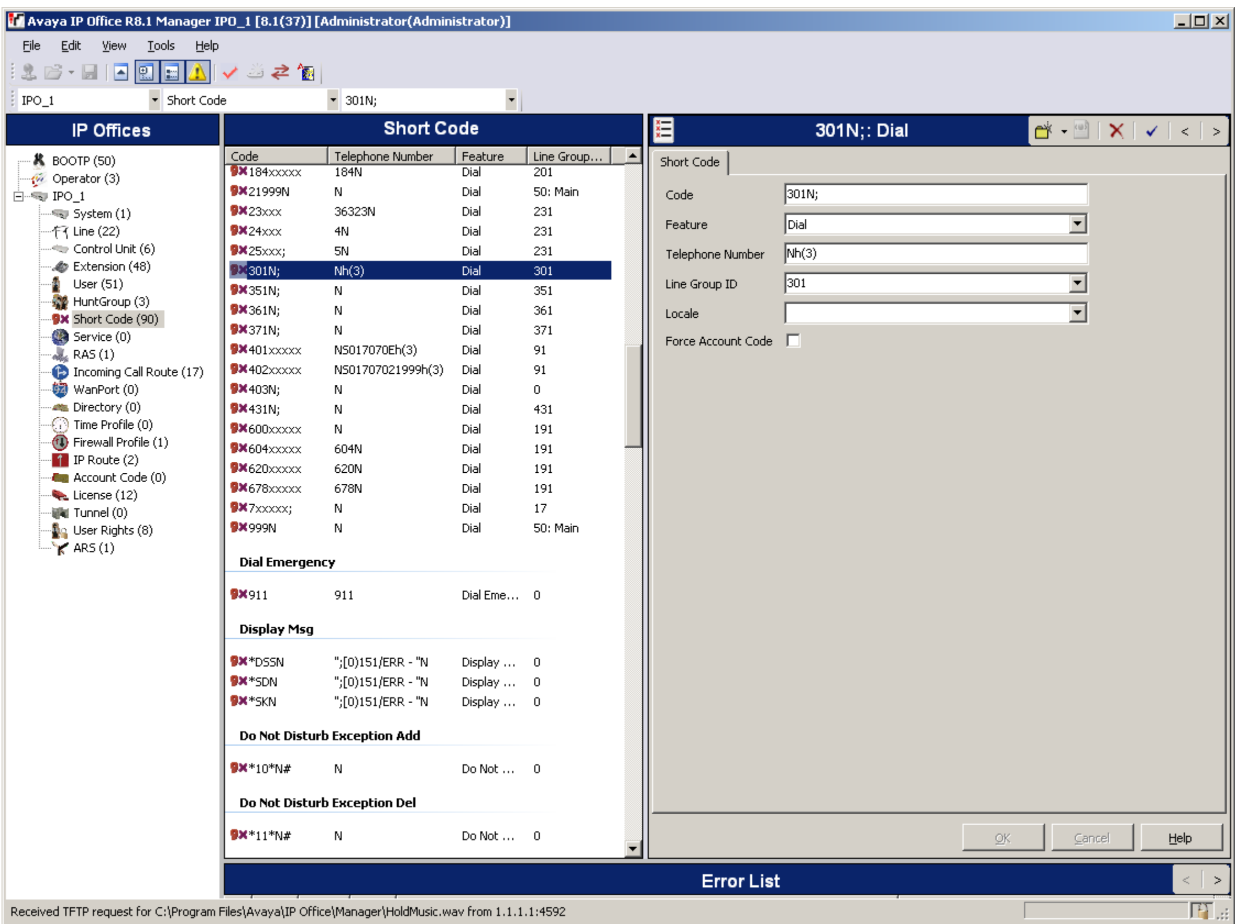
2.1.3 IP Office Incoming Call Route

ICR is no longer required for IP Office to route the H323 trunk incoming calls to internal destinations.

2.1.4 IP Office Short Code

The system uses short codes to access the H323 trunk for outgoing calls.

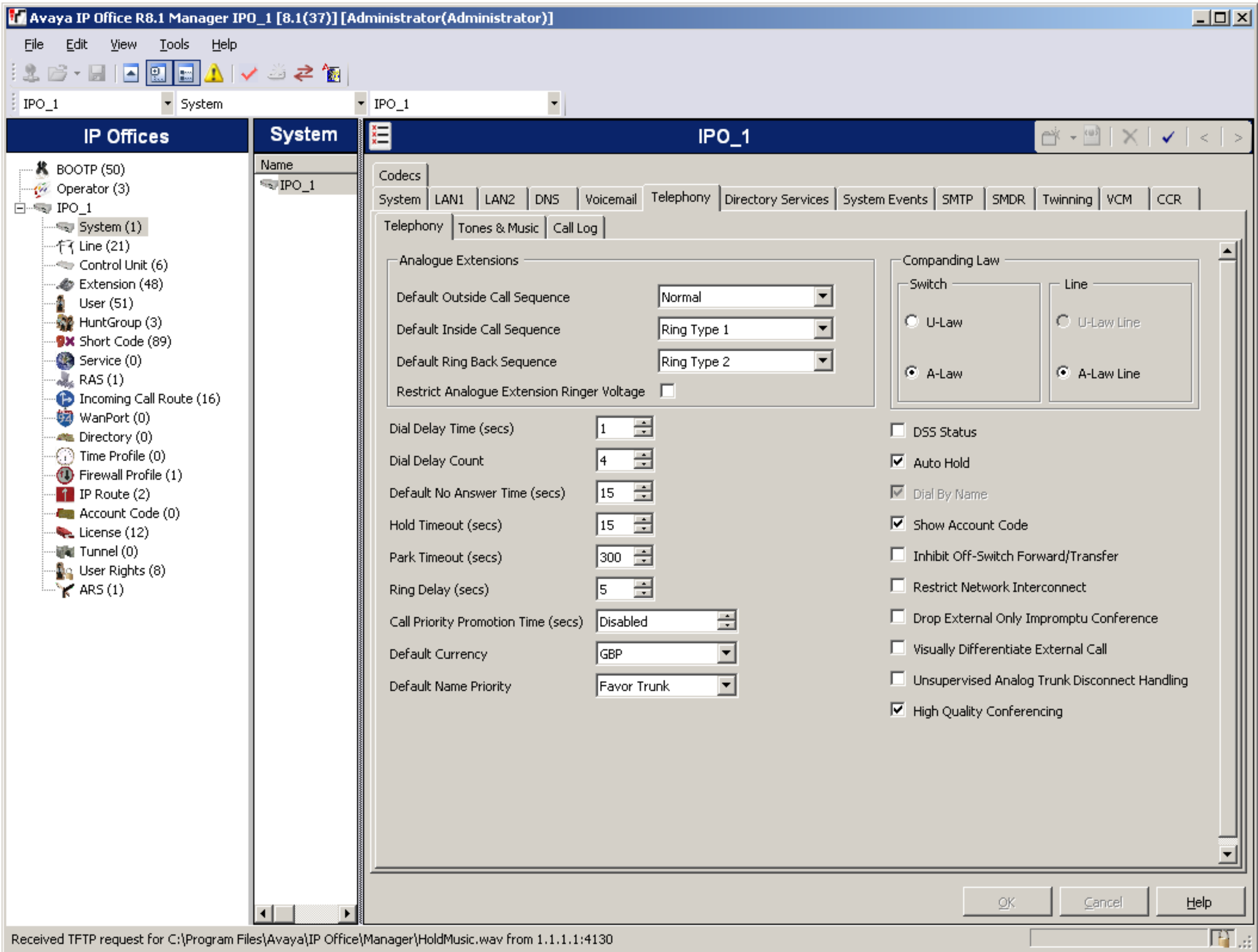
From the configuration tree in the left pane, select **Short Code,** right-click and select **New** to add a new short code.
Introduce the following:
- Code - the dialing digits used to trigger the short code.
- Feature – Select **Dial** action to be performed by the short code.
- Telephone Number – 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 prompted to enter a valid account code before calls are allowed to continue.



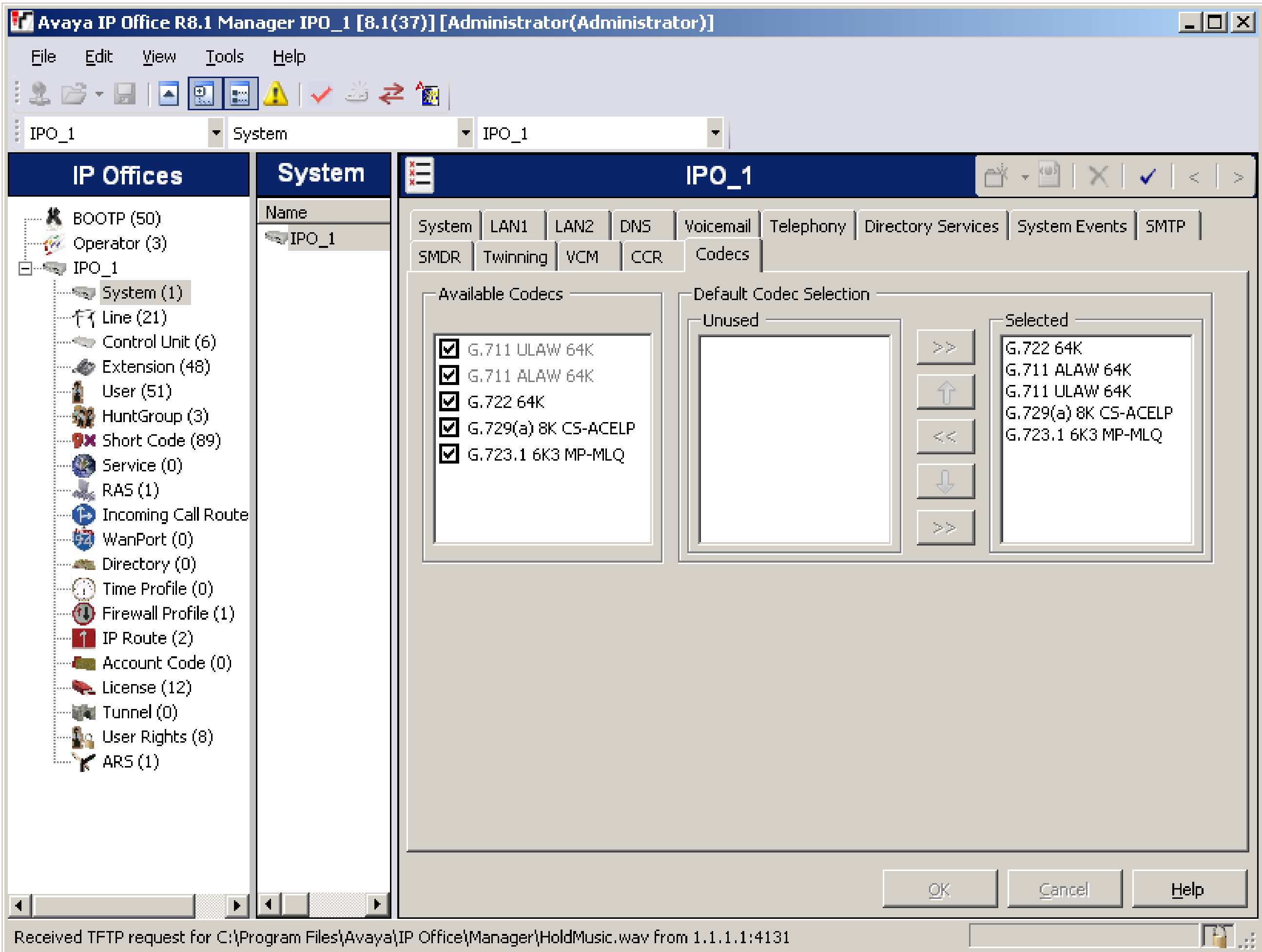
2.1.5 IP Office System Settings

In the **Telephony** tab:
- Companding Law - Choose the appropriate Companding Law depending on the region (U-Law for North America and Japan, A-Law for elsewhere).
- For all other fields, the default values should be retained.

- When **Inhibit Off-Switch Forward/Transfer** is enabled, it stops any user from transferring or forwarding calls externally. Default is Off.



In the **Codecs** tab:
- This tab is used to set the codecs available for use with all IP lines and extensions and to specify 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 - allows to set the codecs preference.



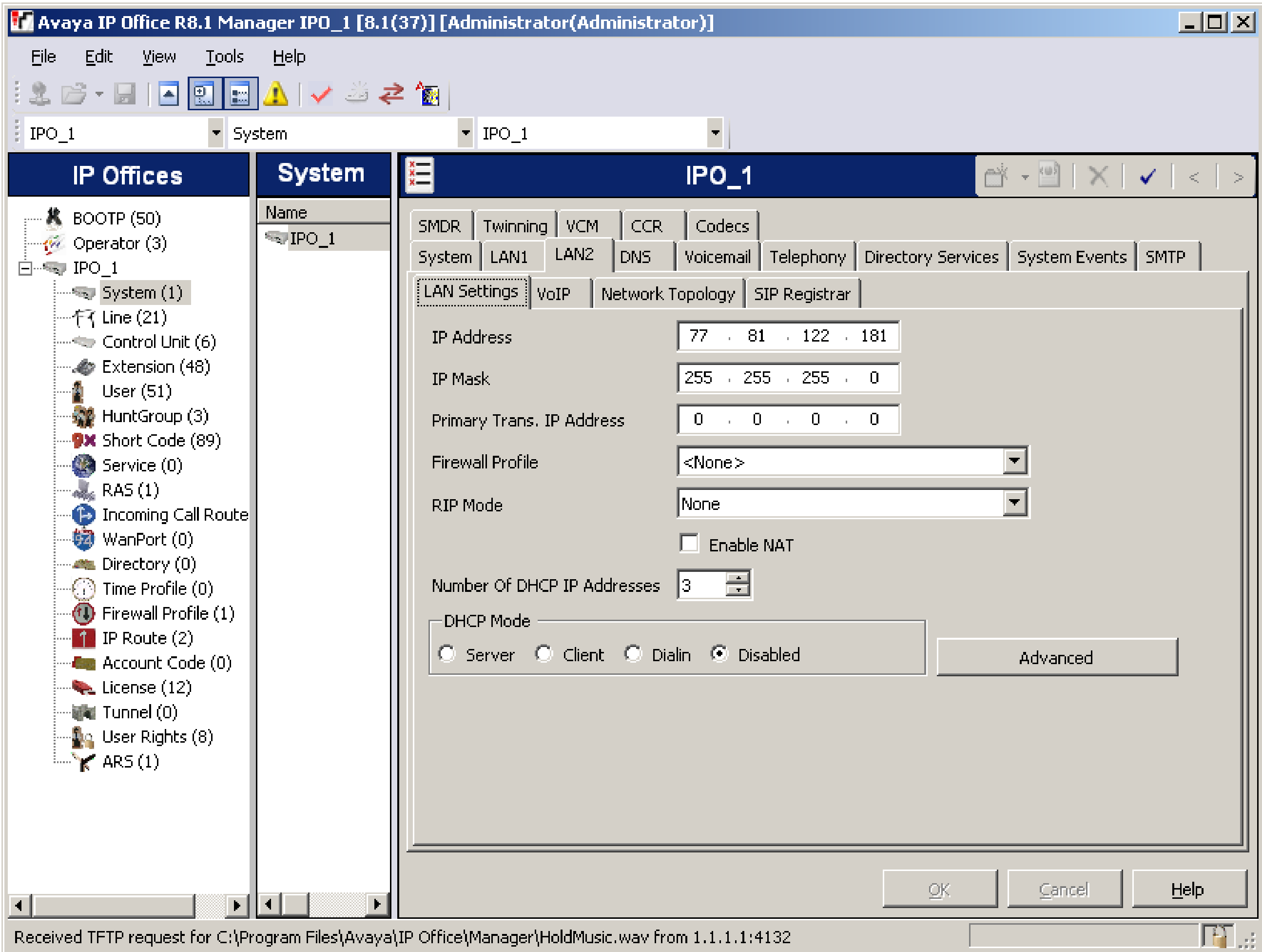
In the **LAN** tab:
Usually LAN1 is used for the local network and LAN2 is used to connect to the Communication Manager, via WAN. LAN2 settings are presented below.

- LAN Settings tab:

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

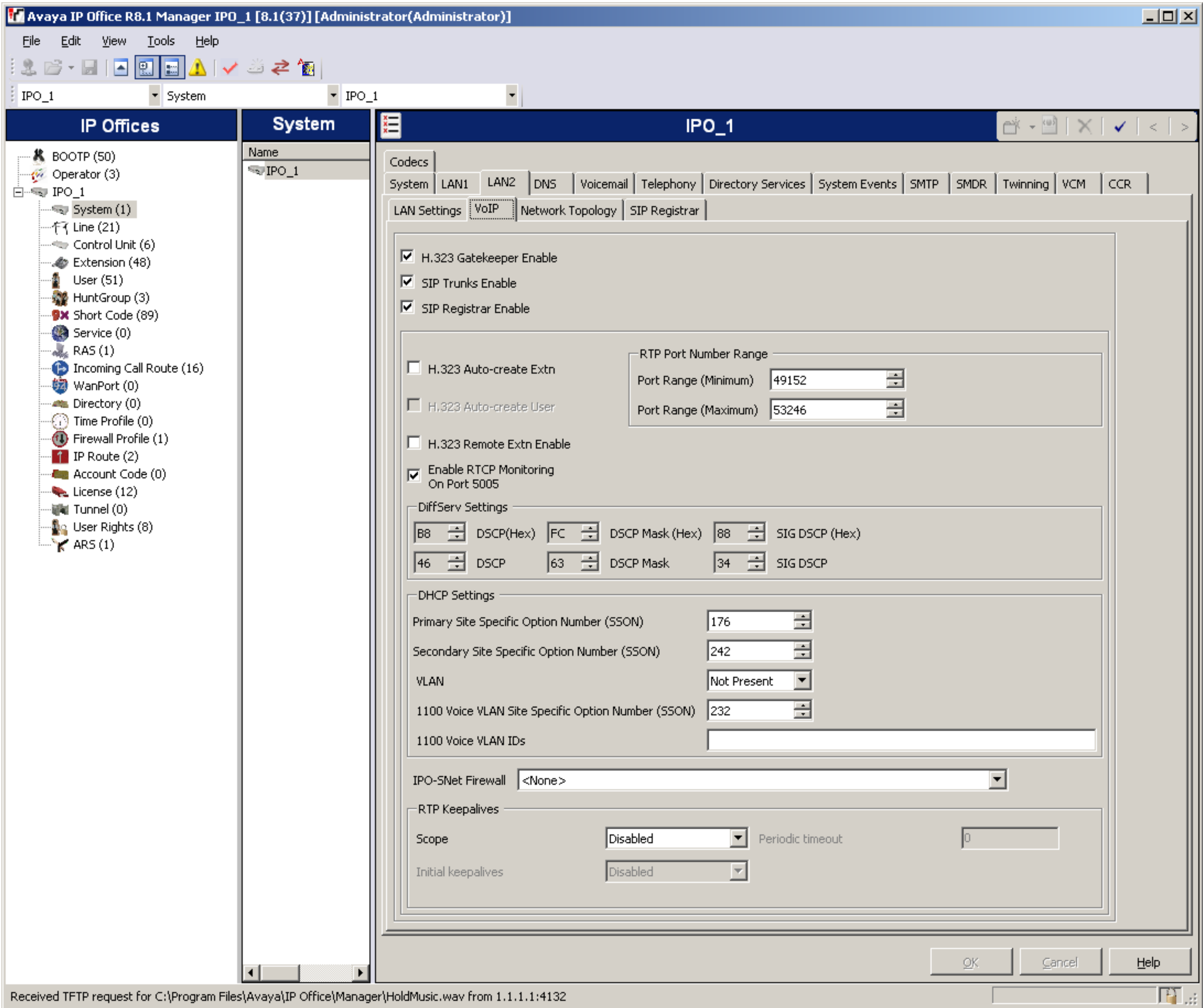
Retain Default values for all other fields.

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



VoIP tab:

- **Default** values should be retained for all fields.
- Ensure that **H.323 Gatekeeper Enable** is checked.



IP Office Small Community Networking (SCN)

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

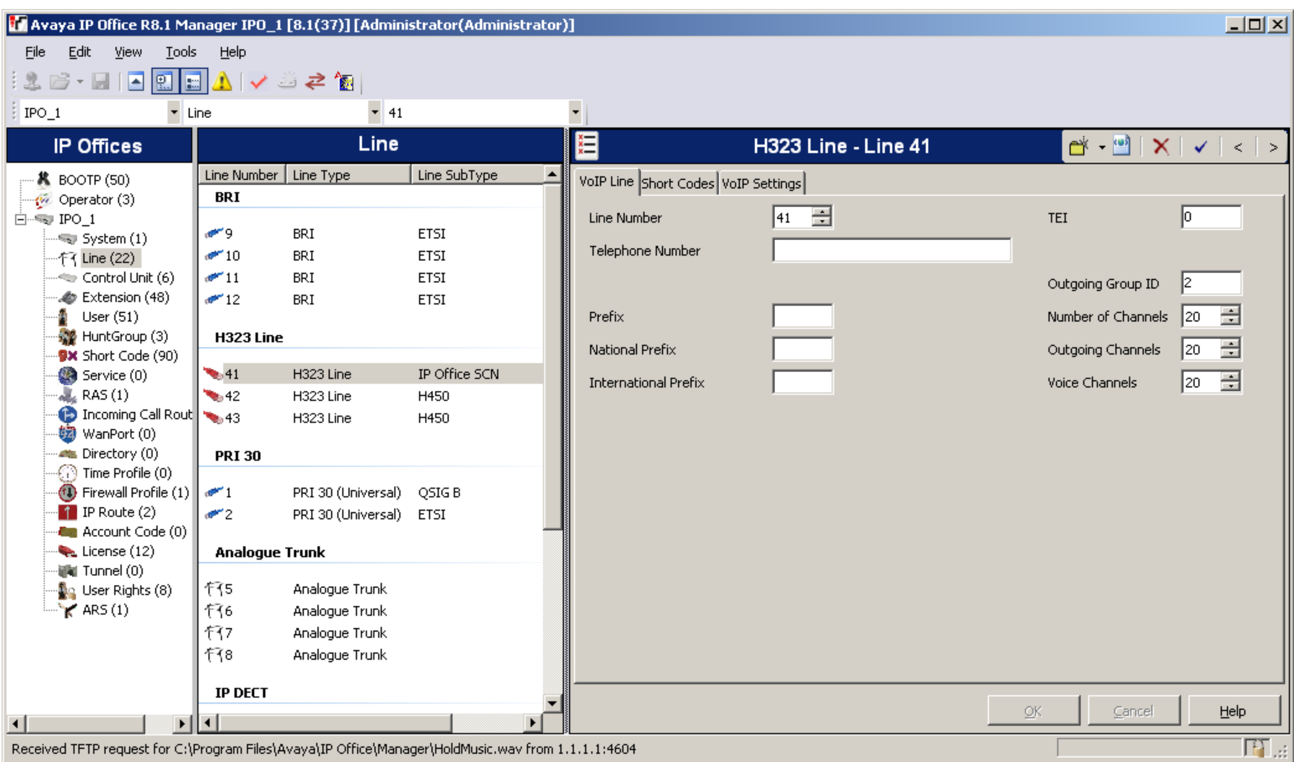
To set up a Small Community Network, the following are required:

- A working H.323 trunk between the systems that has been tested for correct voice and data traffic routing.

- Within a particular network, all SCN trunks should be on the same LAN interface.
- VCM channels are required in all systems.
- The extension, user and group numberings on each system must be unique.
- The user and group names on each system must be unique.
- The Outgoing Group ID on the Small Community Network lines should be changed to a number other than the default , 0 (zero).
- All systems should use the same set of telephony timers, especially the **Default No Answer Time**.
- Only one system should have its **Voicemail Type** set to **Voicemail Pro/Lite**. All other systems must be set to either **Centralized Voicemail** or **Distributed Voicemail**. No other settings are supported.

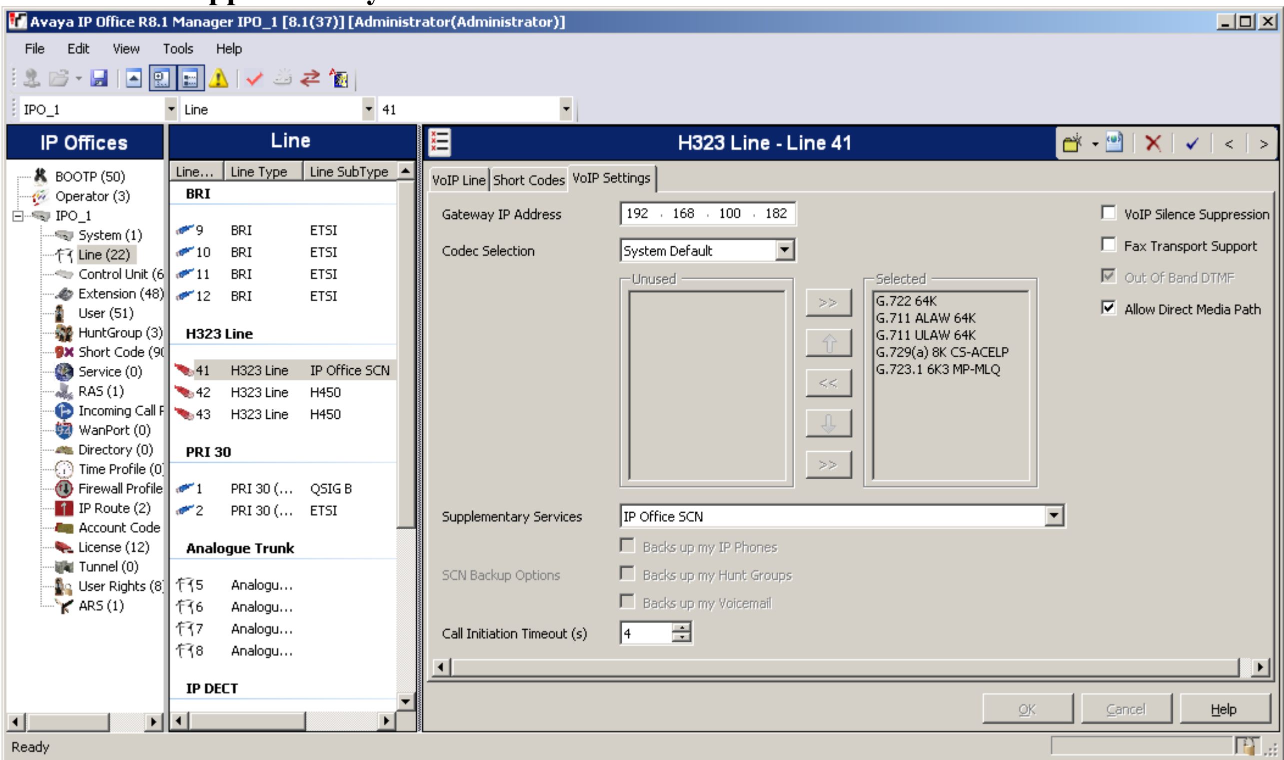
From the configuration tree in the left pane, right-click on **Line** and select **New > H323 Line** to add a new H323 Trunk.

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



In the **VoIP Settings** tab:

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

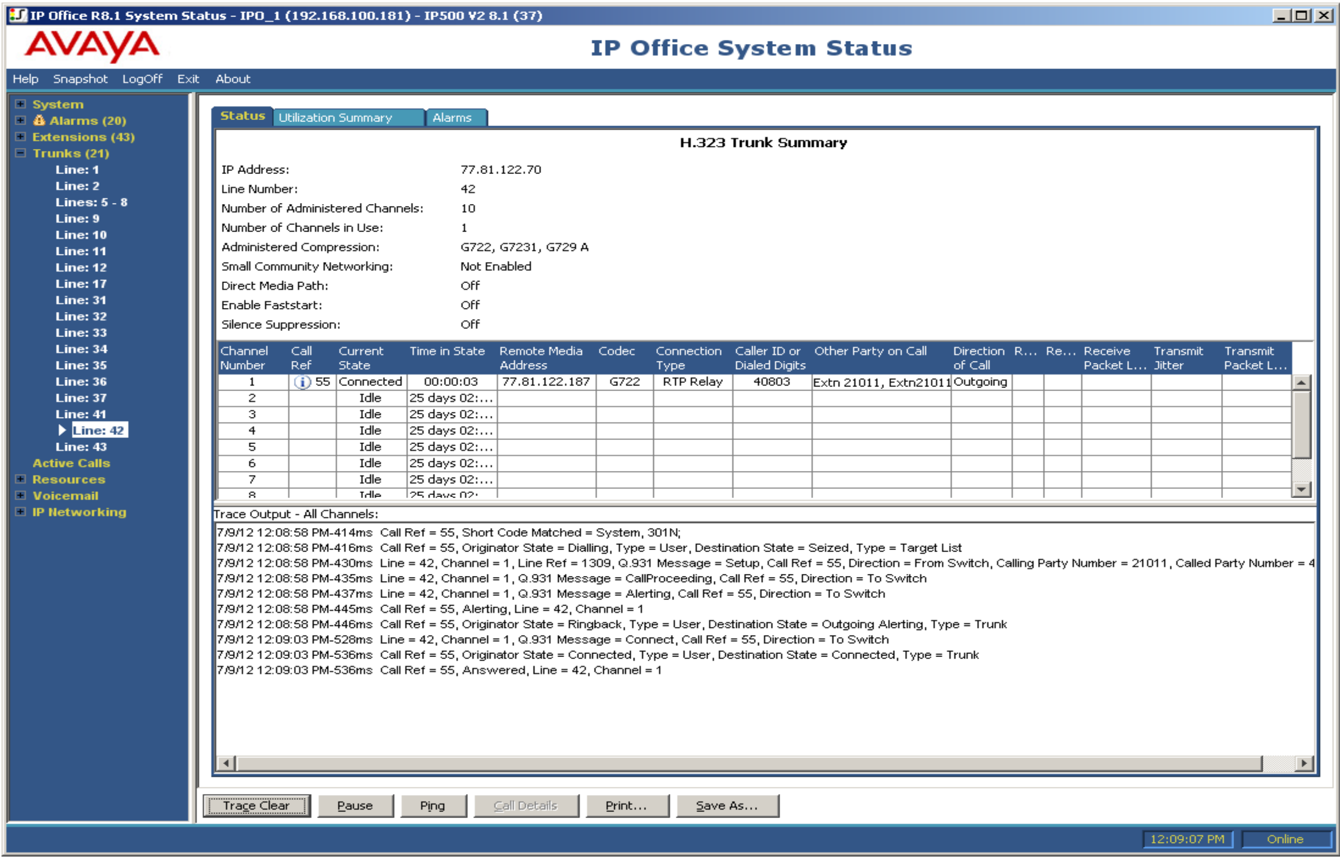


After all IP Office configuration changes are made, save the configurations and send them to the corresponding IP Office unit. Please note that a system reboot will be required for these changes to take effect.

2.1.8 Verify Basic Connectivity

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

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



2.2 Avaya Communication Manager Configuration

This section provides the procedures for configuring the Avaya Communication Manager using a system terminal.

The procedure covers the following areas:

- Communication Manager Keycodes
- Communication Manager H.323 Trunks

2.2.1 Communication Manager Keycodes

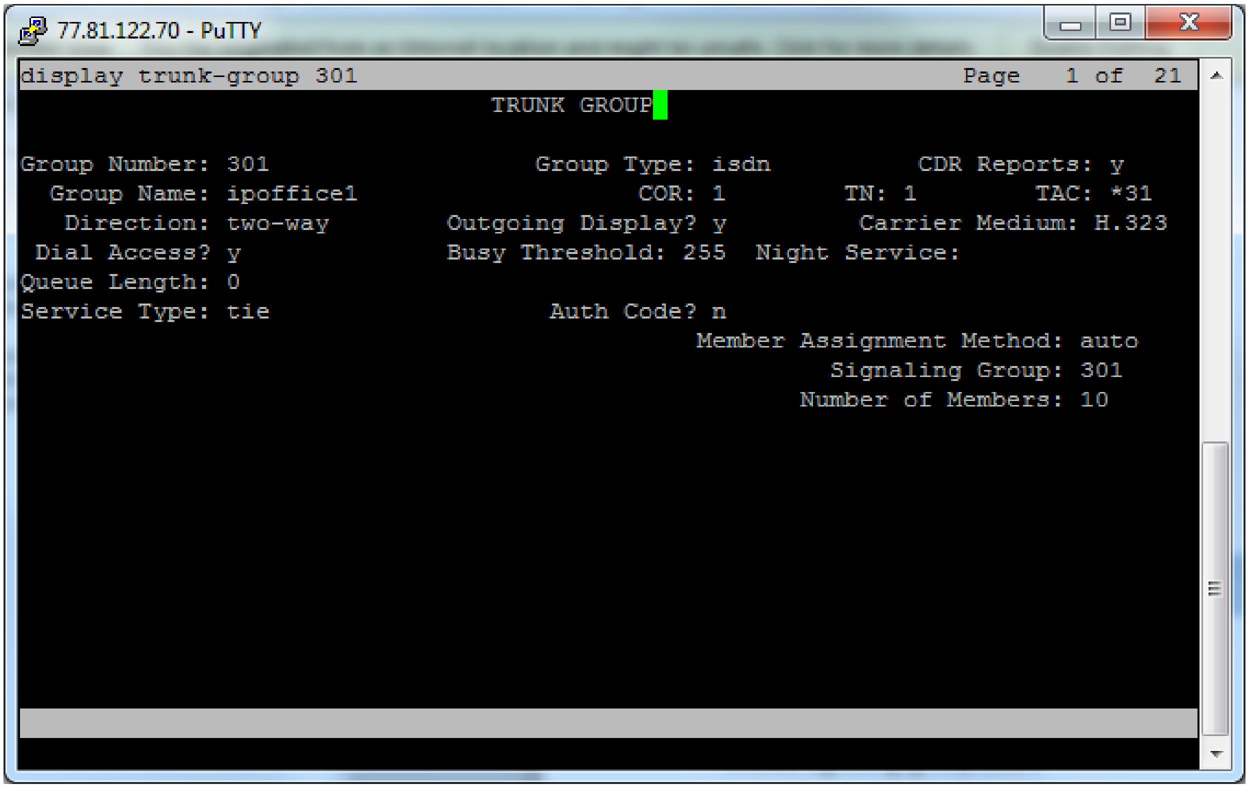
No license is required when creating H.323 trunks on Avaya Communication Manager.

2.2.2 Communication Manager H.323 Trunks

Connect to system terminal over SSH with a user that has administration rights.
To add a new H.323 line enter the command **Add trunk**

The **Display trunk-group** page is presented with the following parameters:
- **Group Number –** Enter the group number for this trunk.
- Group Name – A unique name that provides information about the trunk group.
- Direction – Should be set to **two-way** to have traffic in both directions.
- Dial Access – Should be set to **y** (enabled)**.** Users can gain access to the trunk group by dialing its access code**.** It controls whether users can route outgoing calls through an outgoing or two-way trunk group by dialing its trunk access code. Allowing dial access does not interfere with the operation of AAR/ARS.
- Queue Length – Should be set to default, **0.** In this case, the callers receive a busy signal when no trunks are available. It represents the number of outgoing calls that can wait in queue when all trunks in a trunk group are busy.
- Service Type – Should be set to **tie.** The service for which this trunk group is dedicated. Tie trunks are for general purpose.
- Group Type – Should be set to **ISDN.** Used when digital trunks are needed that can integrate voice, data, and video signals and provide the bandwidth needed for applications such as high-speed data transfer and video conferencing. ISDN trunks can also efficiently combine multiple services on one trunk group**- COR –** It should be set to default, **1.** The Class of Restriction (COR) for the trunk group. Classes of restriction control access to trunk groups, including trunk-to-trunk transfers. Decisions regarding the use of Class of Restriction (COR) and Facility Restriction Levels (FRLs) should be made with an understanding of their implications for allowing or denying calls when AAR/ARS/WCR route patterns are accessed.**- Outgoing Display? –** It should be set to **y** (enabled). With Outgoing Display, telephones can show the name and number of the trunk group used for an outgoing call before the call is connected.
- Busy Threshold – Should be set to default, **255.** The number of trunks that must be busy to alert attendants to control access to outgoing and two-way trunk groups during periods of high use. When the threshold is reached and the warning lamp for that trunk group lights, the attendant can activate trunk group control: internal callers who dial out using a trunk access code are connected to the attendant. Calls handled by AAR and ARS route patterns go out normally.
- CDR Reports – Should be set to **y** (enabled). All outgoing calls on this trunk group generate call detail records.
- TN – Should be set to default, **1.** A tenant partition number assigned to this trunk group. If an unassigned tenant partition number is used, the system accepts the entry but calls cannot go to the trunk group.
- TAC – Enter a trunk access code. The trunk access code (TAC) that must be dialed to access the trunk group. A different TAC must be assigned to each trunk group. CDR reports use the TAC to identify each trunk group.
- Carrier Medium – Should be set to **H.323.** The type of transport medium interface used for the ISDN trunk group.
- Auth Code – It should be set to **n** (disabled).When enabled, Communication Manager performs an auth code check for the incoming trunk call that is being routed over another trunk.
- Member Assignment Method – Should be set to **auto.** The system automatically generates members to a specific signaling group.

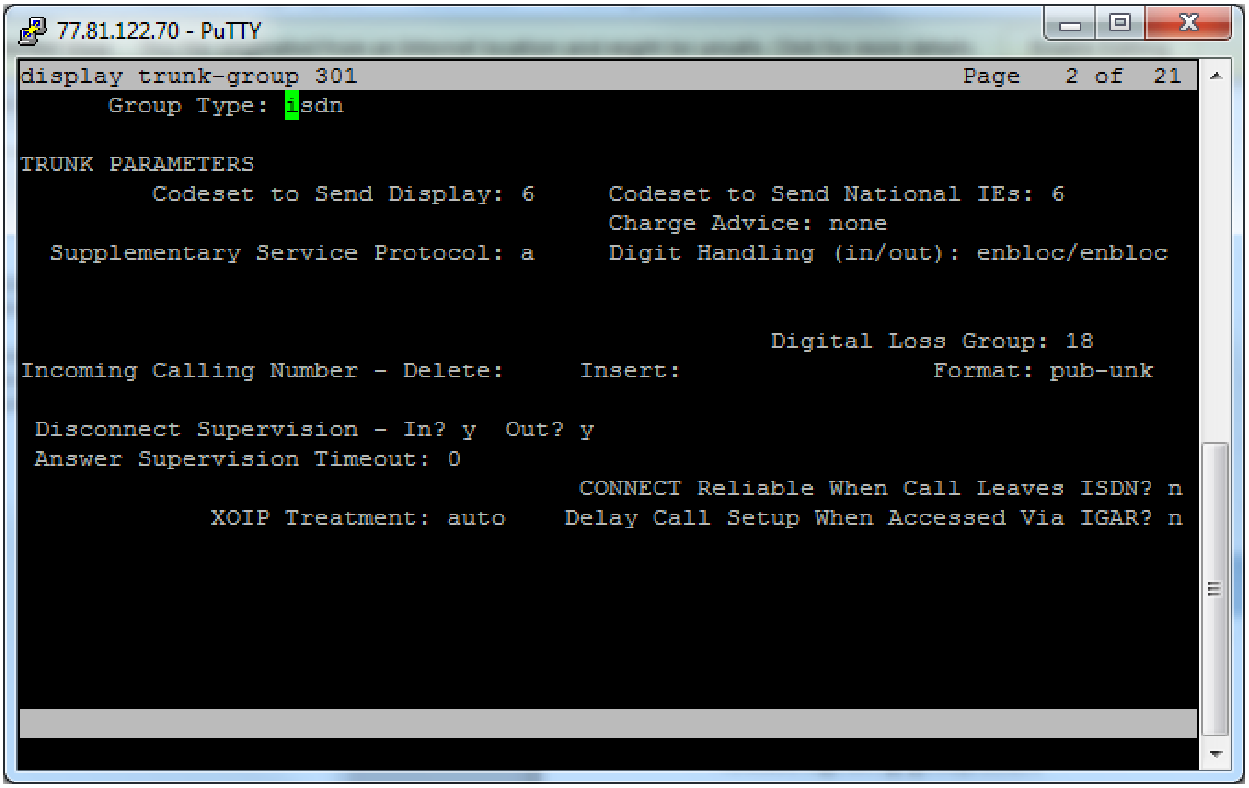
- Signaling Group – Enter the same value as **Group Number, 301** in our case. Available only if the **Carrier Medium** is H.323 and the **Member Assignment Method** is auto.
- Number of Members – Enter a value based on the system resources and setup needs. The number of virtual trunk members automatically assigned to the signaling group.



```
77.81.122.70 - PuTTY
display trunk-group 301                                         Page   1 of  21
                                TRUNK GROUP

Group Number: 301                   Group Type: isdn          CDR Reports: y
  Group Name: ipoffice1                    COR: 1        TN: 1        TAC: *31
   Direction: two-way         Outgoing Display? y         Carrier Medium: H.323
 Dial Access? y               Busy Threshold: 255  Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 301
                                                     Number of Members: 10
```

Go to the 2nd page for Trunk Parameters
- Retain the default values for all parameters:
- **Supplementary Service Protocol –** Ensure it is set to **a**. This is used for **National** numbering plan.
The supplementary service protocol to use for services over this trunk group. Supplementary service protocols are mutually exclusive.
- Codeset to Send Display – Ensure it is set to default, **6.** Defines the codeset for sending the information element for display. The value depends on the type of server or switch used for the connection.
- Codeset to Send National IEs - Ensure it is set to default, **6**. The codeset for sending the information element (IE) for national IEs. National IEs include all IEs previously sent only in code set 6 (such as DCS IE). Now these national IEs, including Traveling Class Marks (TCMs) and Look Ahead Interflow (LAI), can be sent in code set 6 or 7.The value depends on the type of server/switch to which the user is connected.
- **Digit Handling –** Ensure it is set to default, **enbloc/enbloc.** Defines whether overlap receiving and overlap sending features are enabled. enbloc disables overlap receiving and overlap sending. The first field value indicates digit receiving and the second value indicates digit sending.

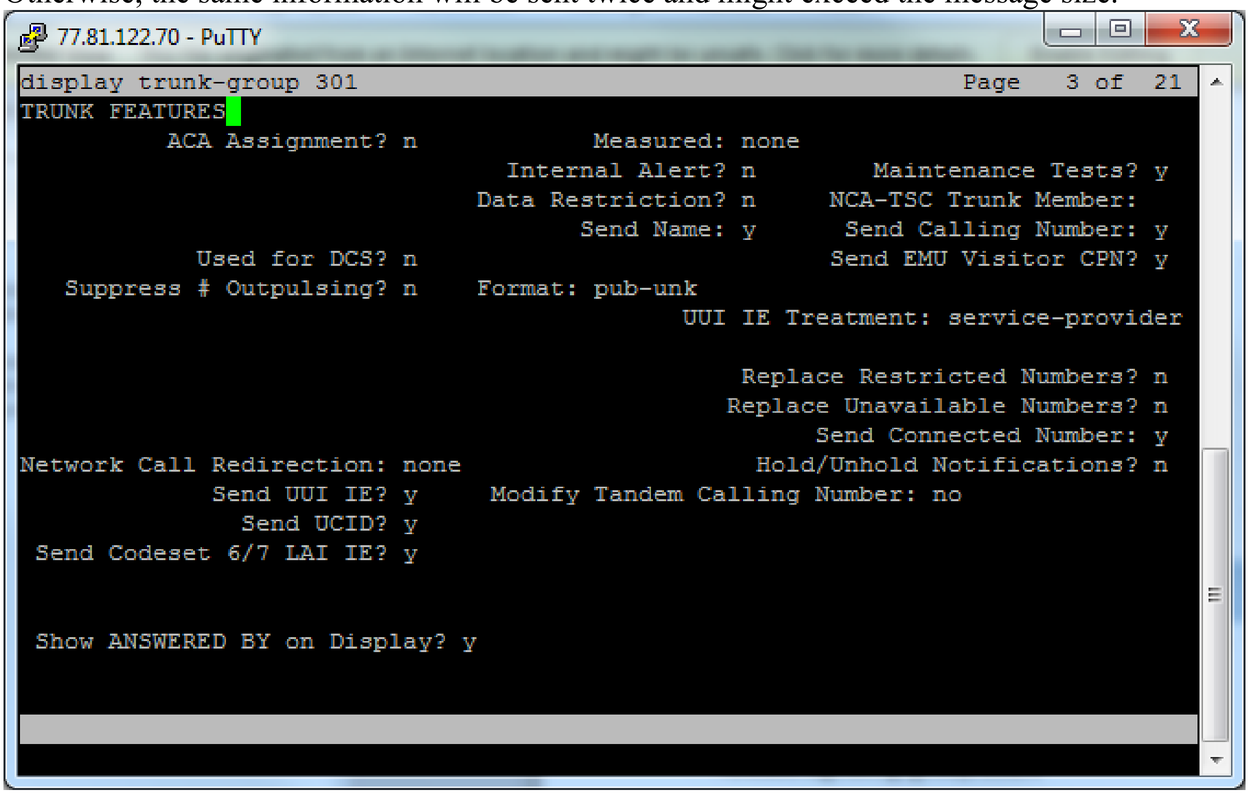
- **Disconnect Supervision In -Out** – Ensure it is set to default, **In? y, Out? y.** Allows trunk-to-trunk transfers involving trunks in this group. The far-end sends a release signal when the called party releases an outgoing call, and the far-end is responsible for releasing the trunk.
- **Connect Reliable When Call Leaves ISDN** – Ensure it is set to default, **n.** If a call is not end-to-end ISDN, the CONNECT message is considered unreliable.




```
display trunk-group 301                                         Page   2 of  21
      Group Type: isdn

TRUNK PARAMETERS
         Codeset to Send Display: 6     Codeset to Send National IEs: 6
                                        Charge Advice: none
  Supplementary Service Protocol: a     Digit Handling (in/out): enbloc/enbloc


                                                   Digital Loss Group: 18
Incoming Calling Number - Delete:     Insert:                 Format: pub-unk

 Disconnect Supervision - In? y  Out? y
 Answer Supervision Timeout: 0
                                      CONNECT Reliable When Call Leaves ISDN? n
             XOIP Treatment: auto    Delay Call Setup When Accessed Via IGAR? n
```

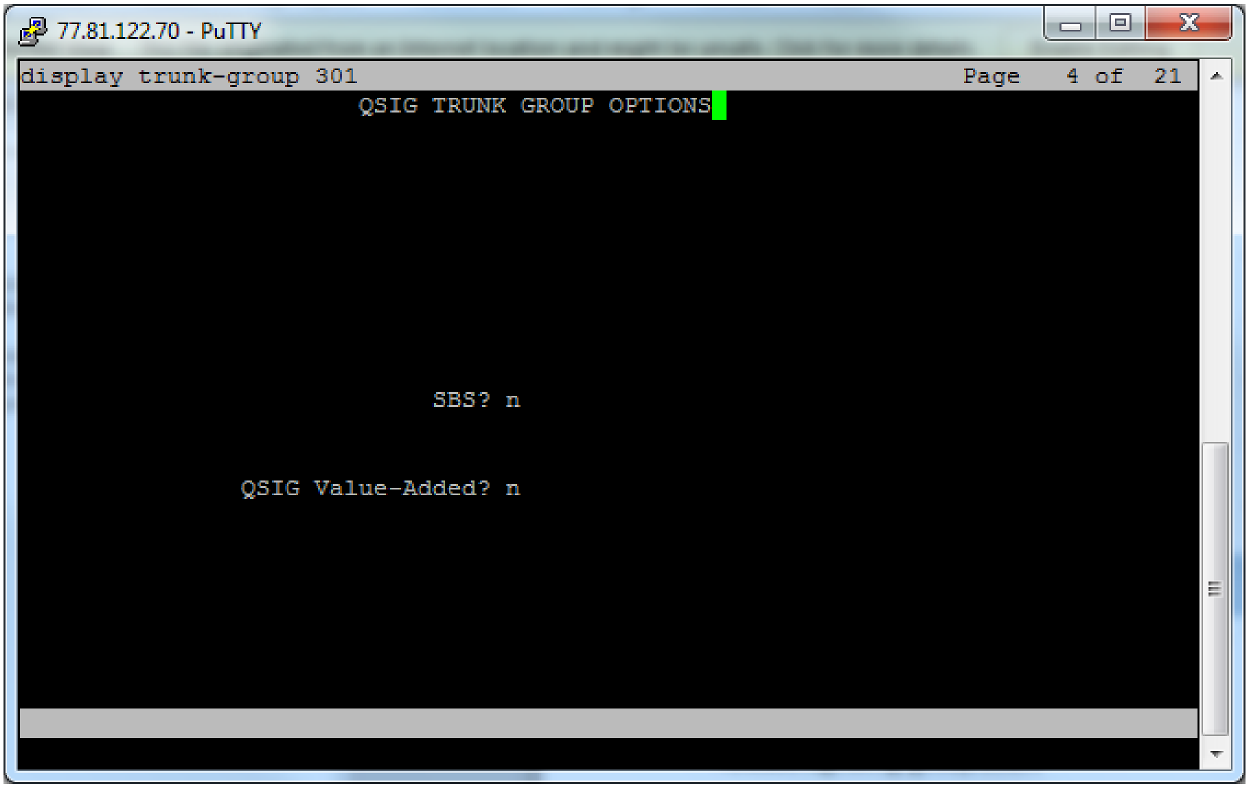
Go to the 3rd page to access **Trunk Features:**
- Send Name – Should be enabled, **y**. Specifies whether the calling/connected/called/busy party's administered name, or the name on a redirected call, is sent to the network on outgoing/incoming calls. If name information is not administered for the calling station or the connected/called/busy station, the system sends the extension number in place of the name.
- **Send Calling Number** – Should be enabled, **y**. Specifies whether the calling party's number is sent on outgoing or tandemed ISDN calls.
- **Send EMU Visitor CPN?** – Should be enabled, **y.** Controls which calling party identification (extension of the primary telephone or extension of the visited telephone) is used when a call is made from a visited telephone. When enabled it sends calling party identification information on the extension of the EMU user's telephone.
- **Send Connected Number** –Should be enabled, **y**. Specifies if the connected party's number is sent on incoming or tandemed ISDN calls. When enabled, the Numbering - Public/Unknown Format is accessed to construct the actual number sent, or the Numbering - Private Format is used.
- **Send UCID?** – Should be enabled, **y**. When enabled, the trunk transmits Universal Call IDs.

- **Show ANSWERED BY on Display?** – Ensure that it is enabled, **y**. The words "ANSWERED BY" display in addition to the connected telephone number on calls over this trunk. This is the default value.
- Retain default values for all other parameters.
- **ACA Assignment –** Ensure it is set to **n.** Indicates whether Automatic Circuit Assurance (ACA) measurements are taken for this trunk group.
- Send UUI IE – Ensure it is set to default, **y.** When enabled, sends UUI information on a per trunk group basis.
- **UUI IE Treatment –** Ensure it is set to default, **service-provider.** Specifies whether the user Information Element (IE) is shared.
- **Send Codeset 6/7 LAI IE?** – Ensure it is set to default, **y.** If enabled, the ISDN trunk transmits information in Codeset 6/7.If the UUI IE Treatment is shared, then this field should be disabled. Otherwise, the same information will be sent twice and might exceed the message size.

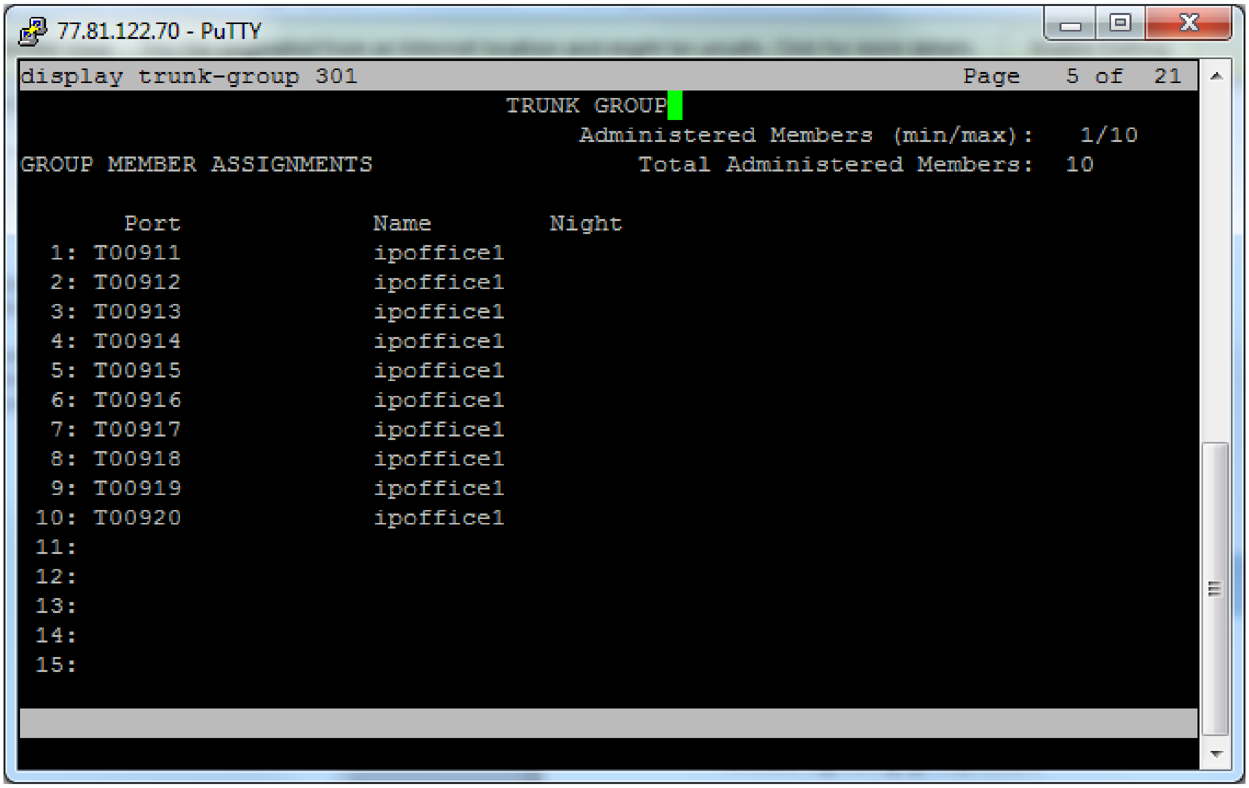


```
77.81.122.70 - PuTTY
display trunk-group 301                                         Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 Internal Alert? n        Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? y
   Suppress # Outpulsing? n    Format: pub-unk
                                             UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: y
Network Call Redirection: none                     Hold/Unhold Notifications? n
             Send UUI IE? y     Modify Tandem Calling Number: no
               Send UCID? y
 Send Codeset 6/7 LAI IE? y


 Show ANSWERED BY on Display? y
```

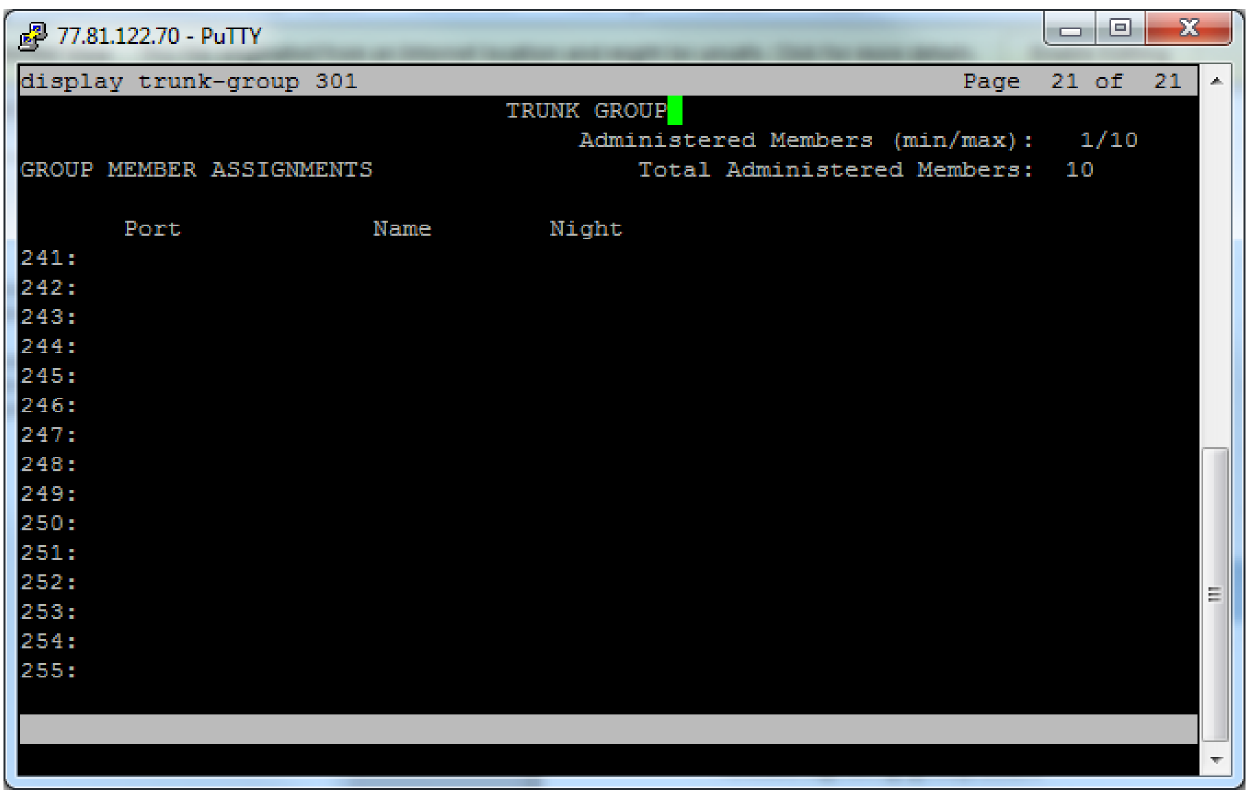
Go to the 4th page to access Qsig Trunk Group Options :
- Retain default values for all the parameters.
- **SBS?** – Ensure it is set to **n** (disabled). Enables or disables Separation of Bearer and Signaling.
- QSIG Value-Added? – Ensure it is set to **n** (disabled). Enables or disables QSIG-VALU services.



Go to the 5th page to see the Members Assigned. The number of the members was defined on the 1st page.



```
77.81.122.70 - PuTTY
display trunk-group 301                                         Page   5 of  21
                              TRUNK GROUP
                                   Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  10

      Port              Name        Night
  1: T00911            ipoffice1
  2: T00912            ipoffice1
  3: T00913            ipoffice1
  4: T00914            ipoffice1
  5: T00915            ipoffice1
  6: T00916            ipoffice1
  7: T00917            ipoffice1
  8: T00918            ipoffice1
  9: T00919            ipoffice1
 10: T00920            ipoffice1
 11:
 12:
 13:
 14:
 15:
```



```
77.81.122.70 - PuTTY
display trunk-group 301                                         Page  21 of  21
                              TRUNK GROUP
                                   Administered Members (min/max):   1/10
GROUP MEMBER ASSIGNMENTS                  Total Administered Members:  10

      Port              Name        Night
241:
242:
243:
244:
245:
246:
247:
248:
249:
250:
251:
252:
253:
254:
255:
```

3.0 Found Issues

…

4.0 Limitations

- Avaya Communication Manager doesn’t support G711 codec.
- Avaya Communication Manager doesn’t update the phone display in case of transfer or forward.
- Avaya IP Office SIP Terminals (11xx,12xx) acts different than H323 phones in case of a Transfer, meaning that when transferring a call, the SIP phone CLID is presented to the remote party instead of the transferred party CLID and after the transfer is completed the displays are not updated.