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# XO® SIP Service

## Customer Configuration Guide
## Avaya IP Office 4.2.4




XO Communications

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# 1 Overview

***About This Document***

This document describes XO Communications SIP packages 1 and 2 configuration requirements for Avaya IP Office 500 software version 4.2.4, deployed with an XO-provided Cisco 2432 IAD as the router/demarcation device. This document assumes the audience has a general understanding of network provisioning and the connectivity requirements of XO SIP service offering.

***Known Issues***

While XO certifies interoperability between XO SIP service and the PBX as outlined herein, the following known issues were identified during Interoperability testing.  The customer should be aware that certain features and functions may not be fully supportable.  In some cases, special configurations and/or PBX software patches may be available from the vendor:

- In order to make outbound calls with version 4.2.4, the 4.2.42206 patch is required.  Sales should contact XO IP Product Management or Avaya for information on how to obtain the patch.
- Caller ID on Call Forwarded Off-Net – When incoming PSTN calls are delivered to desk phone with Call Forwarding enabled to an off-net PSTN phone, the call will be forwarded but the originating caller ID will not be passed.
- Fax not supported

***Registration Method***

Avaya IP Office uses static registration between IP phones and the IP PBX.

***XO SIP Service Packages Supported***

Avaya IP Office is certified to support XO SIP Service Packages 1 and 2:

| Pkg | Codec | DTMF |
|---|---|---|
| 1 | G.711 | RFC2833 (inband RTP DTMF fallback) |
| 2 | G.729a | RFC2833 |

***Configuration***

The customer premise equipment shall consist of the following components:

- Avaya IP Office Communication System
- XO Managed Router

***Supported Phone Types***

- 2400/5400 series digital
- 4600/5600 series IP
- 6400 series digital
- T3 (IP and digital) –excluding Small Office
- 3701/3711 (IP DECT)
- Analog phones

# 2 Customer Configuration Guide

This section contains the Avaya IP Office Communication System screens that must be configured and updated to support XO SIP.

In order to enable SIP communication you will need a valid SIP trunking license and IP Office with VCM cards.

## How to identify you are running version 4.2.4

Users can identify the version number they are running by looking at the top line of their manager screen and identify the Manager and Core version.
An example is given in Figure 1.

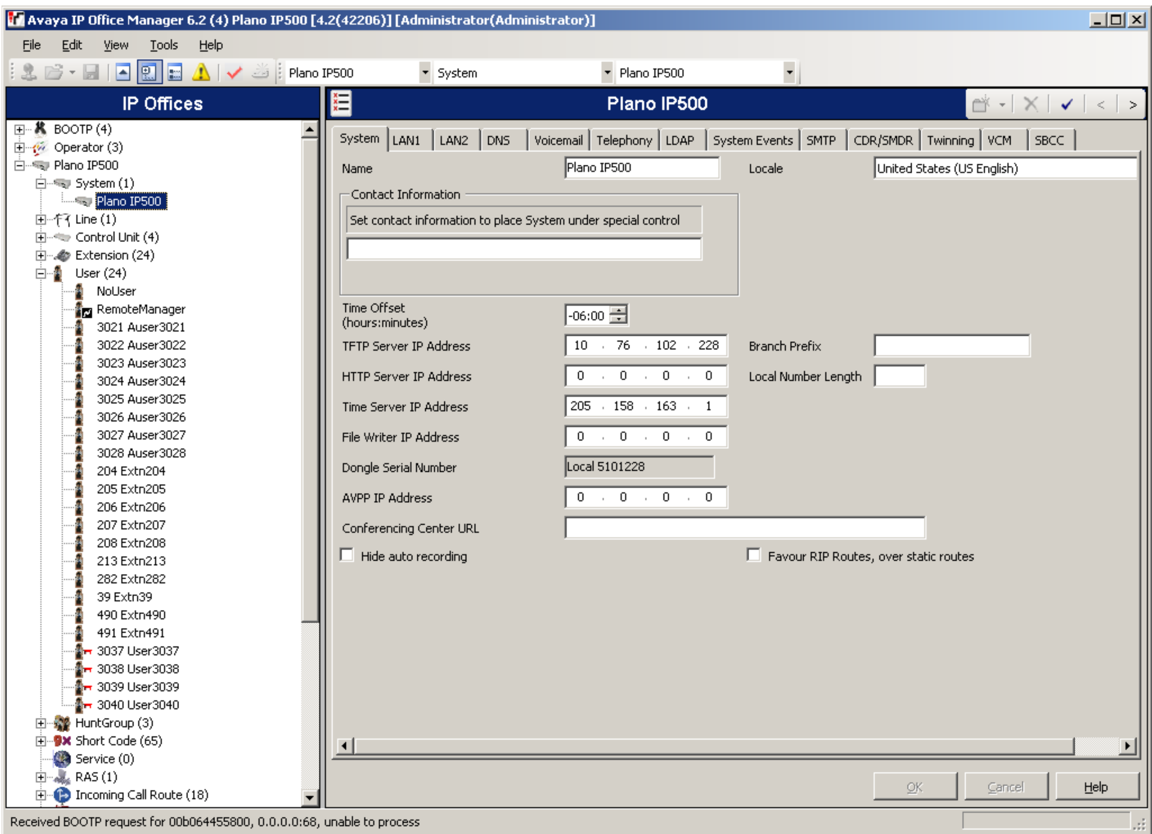


**Figure 1: Example of Manager Screen Shot**

In the example of Figure 1, the Manager Version 6.2(4), and the IP Office Core software version is 4.2.42206.

## How to check for SIP Trunking Licenses

To make calls using SIP you must have a valid license that can be purchased through Avaya business partners, in the number of 1, 4, 5, 8, 20, 28, 255 or a combination of all the above with many instances of the same license. Avaya will provide a license that will need to be inserted into license form. An example is provided in Figure 2. License can be shared among different SIP trunks; the number of instances represents the maximum number of calls that can be dialed or received at the same time by IP Office using any of its SIP trunks.

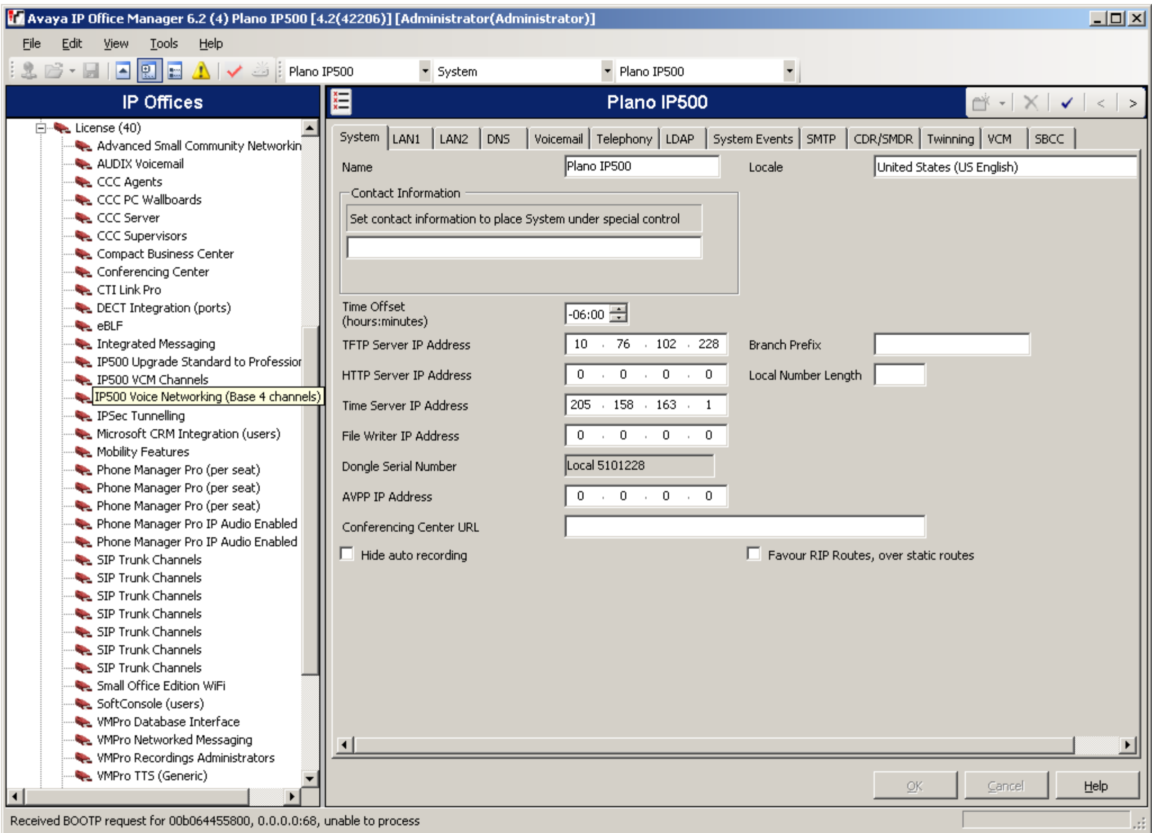


**Figure 2: License Form with valid SIP license**

The fields on the License form are populated as follows:

*License Key* is the license identifier that will be provided by Avaya business partners.

*License Type* must be set to *SIP Trunk Channel*.

*License Status* should be set to *Valid*, if the acquired license is a valid one.

*Instances* will display the number of license instance that have been purchased.

*Expiry Date* will indicate the expiration of the license

## Setting Up IP Route to XO IP Network

Figure 3 illustrates how to set IP Route to Managed Router at Customer Site, where a fictitious IP address has been used.

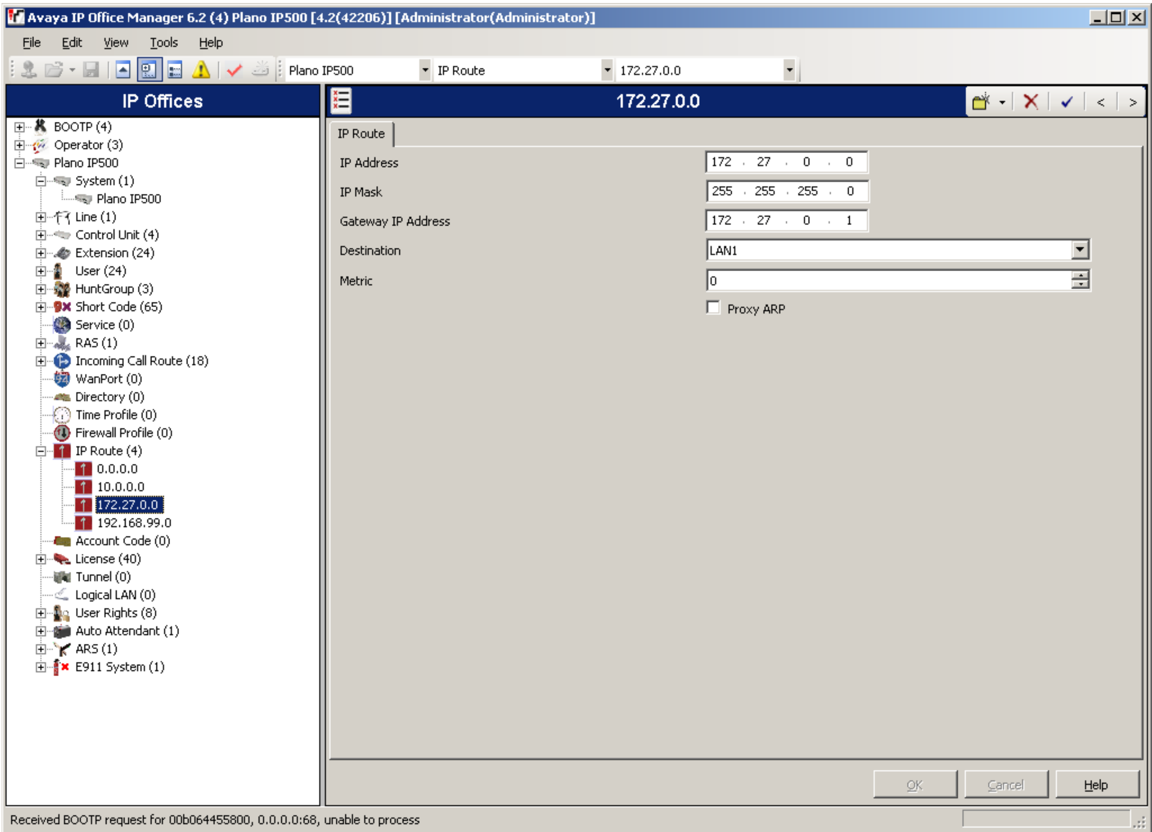


**Figure 3: Setting IP Route to Managed Router at Customer Site**

## Check that TOS settings are the correct default values

TOS settings can be checked for the interface that is going to be connected to XO through the Managed Router at Customer Site. An example configuration is depicted in Figure 4.
**NOTE: Below are the RTP DSCP and SIG DSCP values for use with XO SIP.**

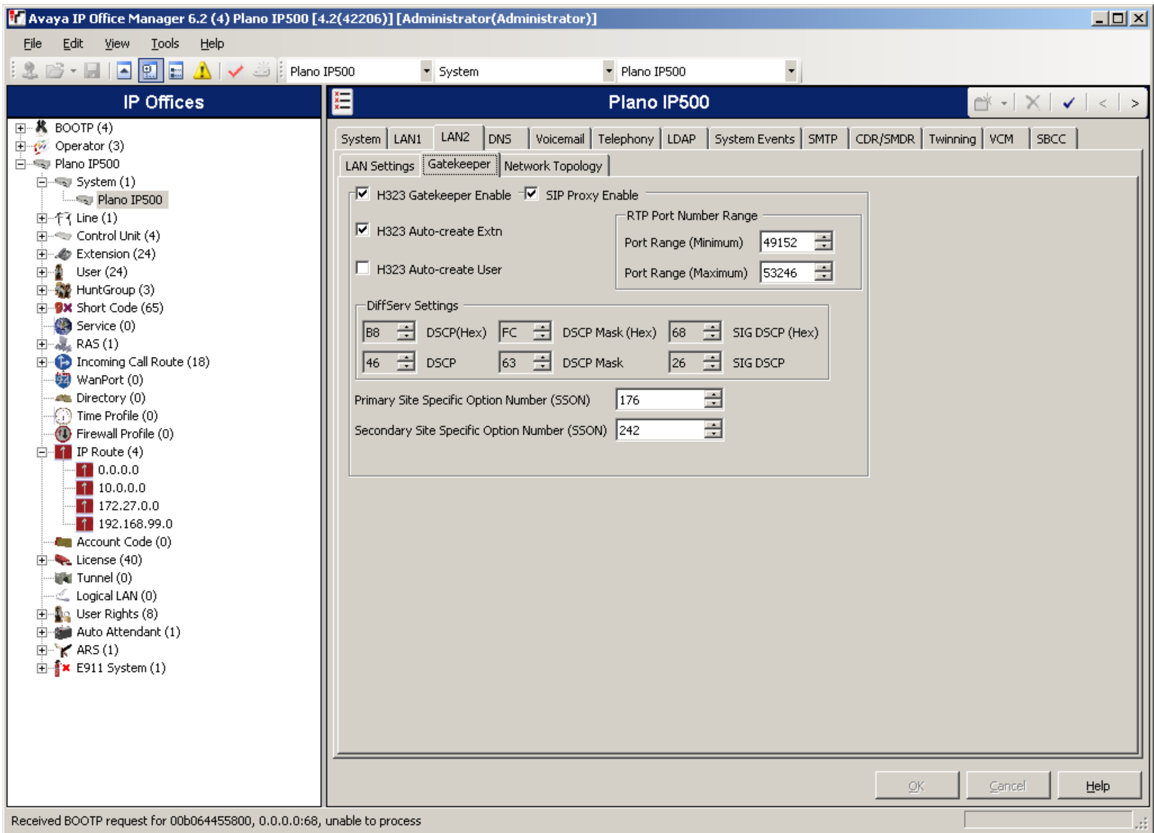


**Figure 4: TOS settings on LAN interface connected to XO network**

It is important to note that for each field in Figure 5, DSCP, DSCP Mask and SIG DSCP, the decimal and hexadecimal fields need to be set individually.

## 2.1 How to Configure the Main SIP Line

Figure 5 shows where to begin with SIP Line configuration – with its main tab.

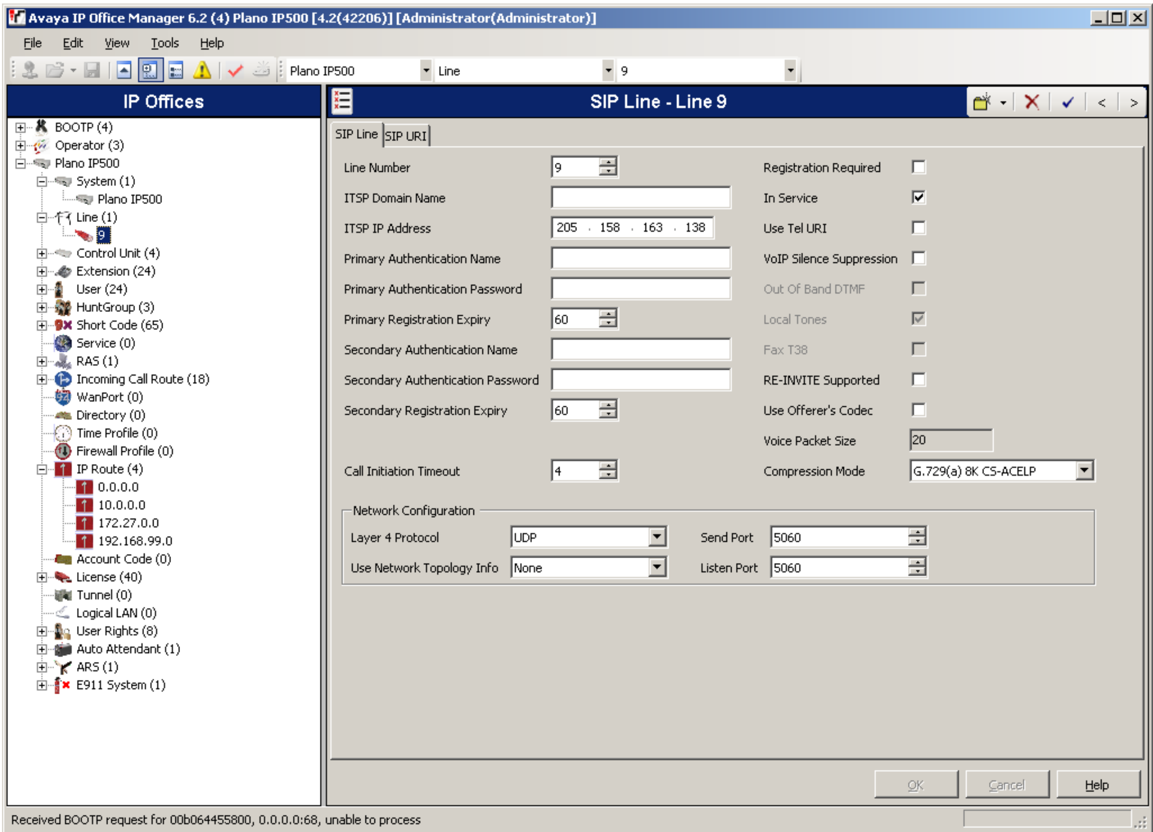


**Figure 5: SIP Main Line**

| Field | What to Enter | Field | What to Enter |
|---|---|---|---|
| Line Number | Automatically assigned in an incremental manner | Secondary Authentication Password | Leave blank |
| ITSP Domain Name | Leave Blank | Secondary Registration Expiry | Left as default |
| Registration Required | Leave un-ticked | In Service | Ticked |
| ITSP IP Address | Set to XO Border Element***** | Use TEL URI | Leave un-ticked |
| Primary Authentication Name | Leave blank | Re-INVITE Supported | Leave un-ticked |
| Primary Authentication Password | Leave blank | Compression Mode | Set to G.729(a) 8K CS-ACELP |
| Primary Registration Expiry | Left as default | Send Port | Set to 5060 (default) |
| Secondary Authentication Name | Leave blank | Listen Port | Set to 5060 (default) |

*Please contact your XO Customer Care Rep. for the XO IP Border Element (IPBE) IP addresses for your specific PBX.

## How to Create the SIP URI tab

For each DID that is assigned to a specific user, a correspondent *SIP URI* needs to be created. Figure 6 depicts a scenario where two fictitious DIDs have been assigned. Each *SIP URI* has a number of fields to set.

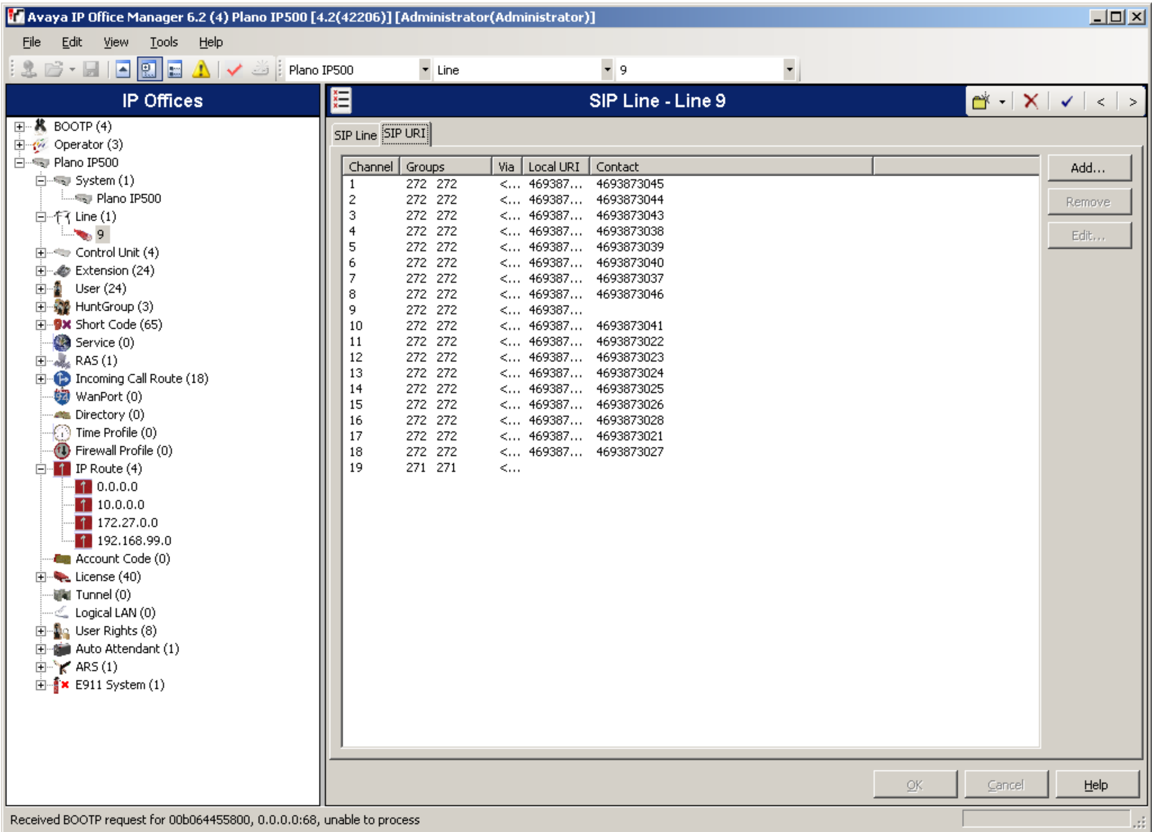


**Figure 6: SIP URI tab**

Each SIP URI has its own Incoming and Outgoing Group ID. Although it is not necessary it is strongly recommended to use a different Group ID for each SIP URI (and each type of Trunk also).

- *Outgoing Group ID* allows selecting using short code, which SIP URI to use to make outbound calls.
- *Incoming Group ID* allows routing of incoming calls to a specific route, if a match between the target of the call and Local URI is produced.

The fields which characterize each URI are Local URI, Contact, and Display Name. Such fields are used to set SIP message headers for both outgoing calls and Incoming calls.

- *Local URI* acts as user part of FROM header for outbound calls, and it is the field used to match user part of TO header from incoming calls.
- *Contact* sets user part of CONTACT field in SIP messages.
- *Display Name* sets the homonymous field in FROM header for outbound calls.

Local URI can be set in two different modes for this configuration type:

- By editing each field individually (Figure 6). This mode allows setting each SIP URI to coincide with a given DID assigned by ITSP. With this mode, SIP URI settings are common for all users in the system.
- By setting in to Use User Data (Figure 7). This setting allows differentiating each SIP URI to a given user in the system. The fields will be filled in using SIP tab in User form.

Note: The number of simultaneous calls may also dependent on the number of calls supported on the VCM cards.

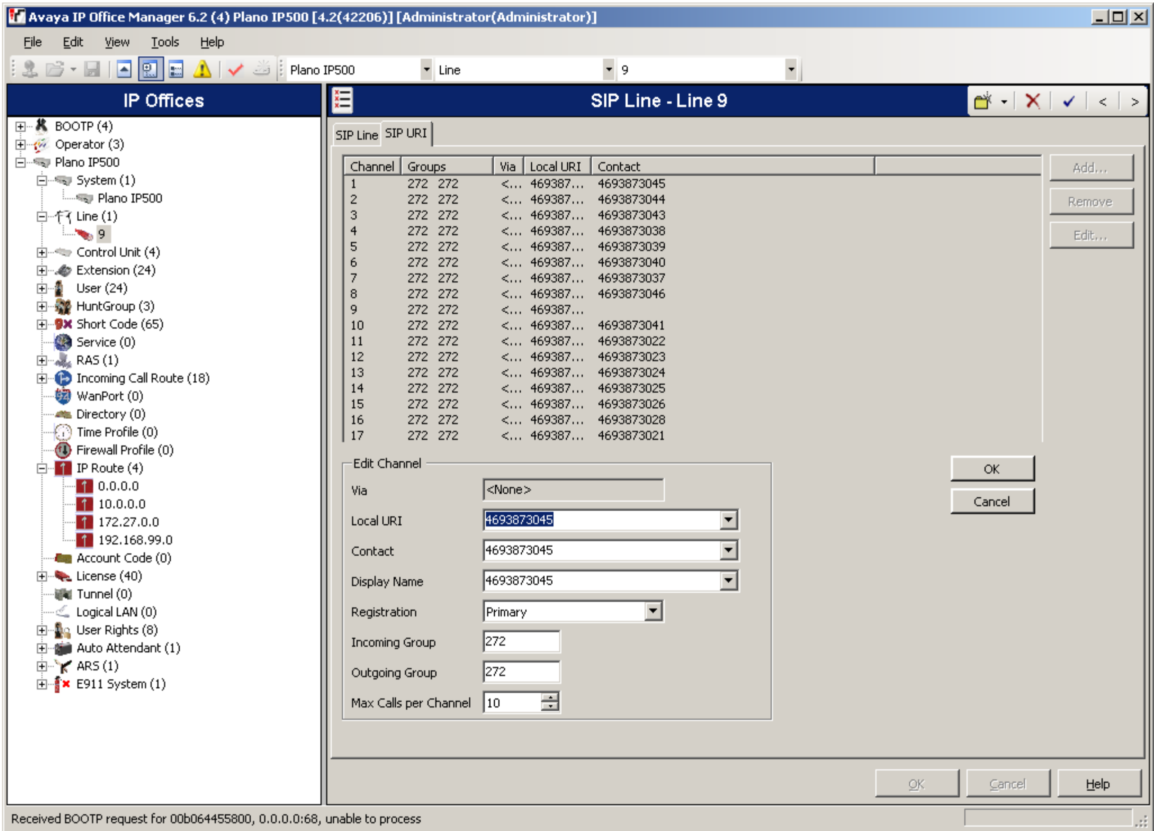


**Figure 7: SIP URI set to USE USER DATA**

## Setting SIP tab in User Field

Figure 8 shows how to set the SIP tab in the User field.

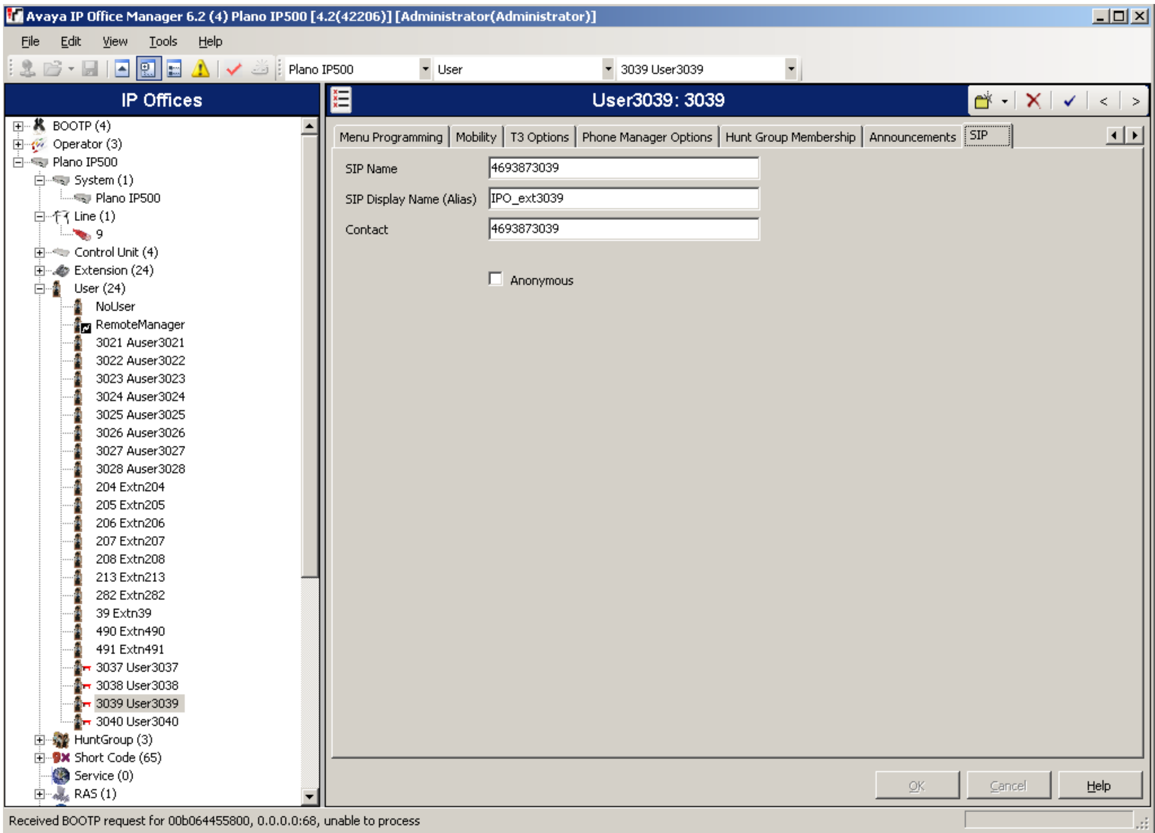


**Figure 8: Setting SIP tab in User Field**

- The User fields are populated as follows: The role of *SIP Name* is identical of *SIP URI* in *SIP URI* form. For inbound calls, it will be compared against the user part of TO header, and if a match is found call will be route according to URI *Incoming Group ID*.
- For outbound calls, *SIP Name* will constitute the user portion of FROM header.
- *Display Name* will fill the homonymous field in FROM header, and *Contact* will fill the user part of CONTACT header.
- *Anonymous* allows enabling Privacy Mechanism for outbound calls.

## Routing Calls to XO

This section describes the IP Office configuration required for sending calls to the XO network.

## Setting the Calling Plan

The usual system of SHORT CODE object can be used to direct calls to the XO network. Figure 9 shows an example with a fictitious IP address.

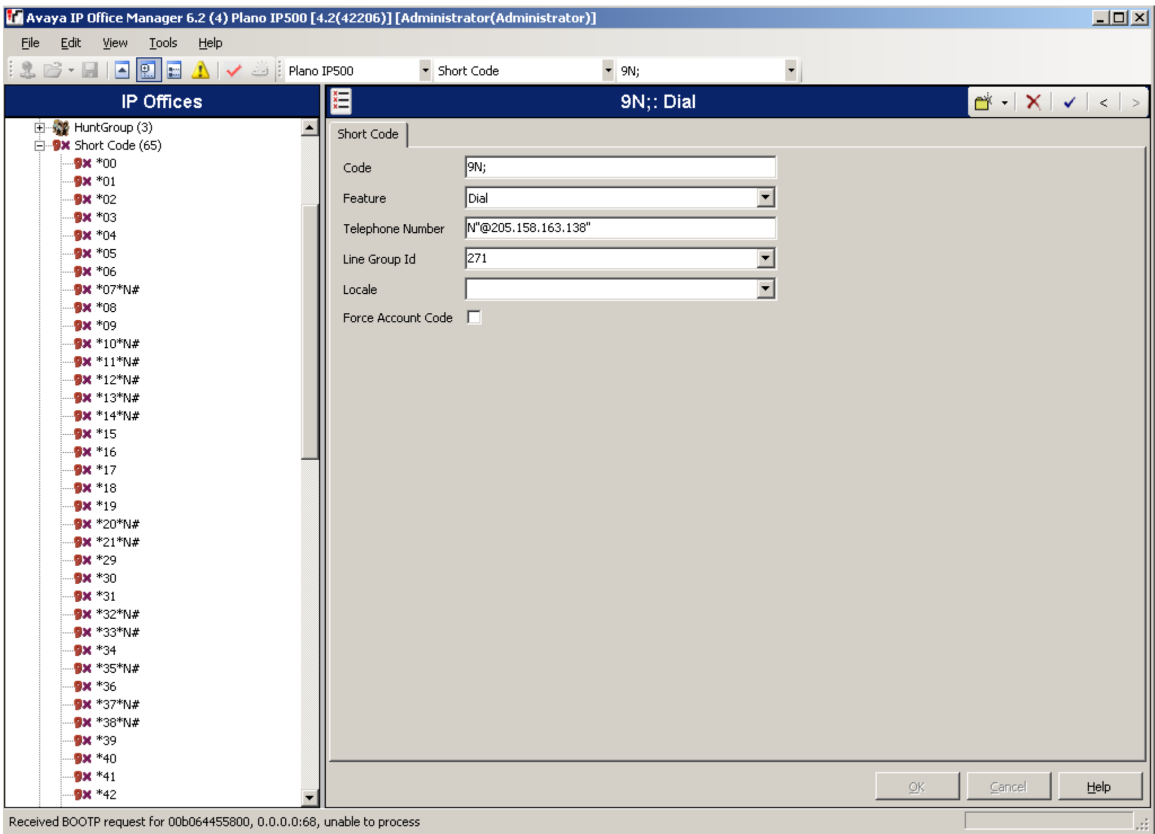


**Figure 9: Setting Calling Plan**

The fields are populated as follows:

- *Short code* – This field is set to "<trunk identifier>N#". <trunk identifier> is a series of number or characters that uniquely identify while dialing out. The character "#" is sued to terminate the string.
- *Line Group ID* – Set this field to the SIP *URI Outgoing Group* ID that it is used to connect to the XO network. A subsequent section will describe how to configure IP Office to route to the 2 XO Border Elements in a primary/secondary arrangement.
- *Feature* – Set this field to "Dial".
- *Telephone number* - is set to "N@<IP address of XO server>". Beware that appending IP address of remote server is necessary to select a SIP URI rather than a TEL URI.

## Routing to a Primary and Secondary XO Border Element

Figure 10 depicts an example of Alternate Route Selection (ARS) that can be used to try different target for the same number.

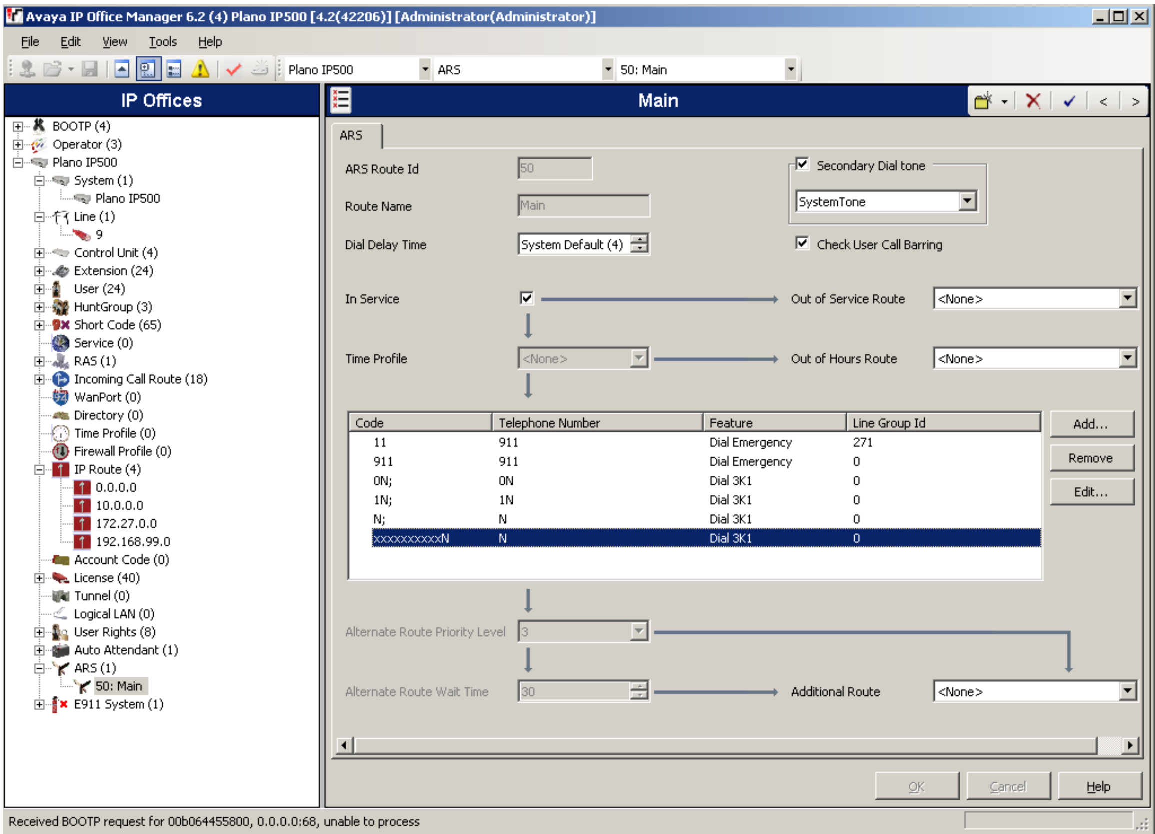


**Figure 10: ARS form**

Each ARS for is entered using a short-code, like that of Figure 11.

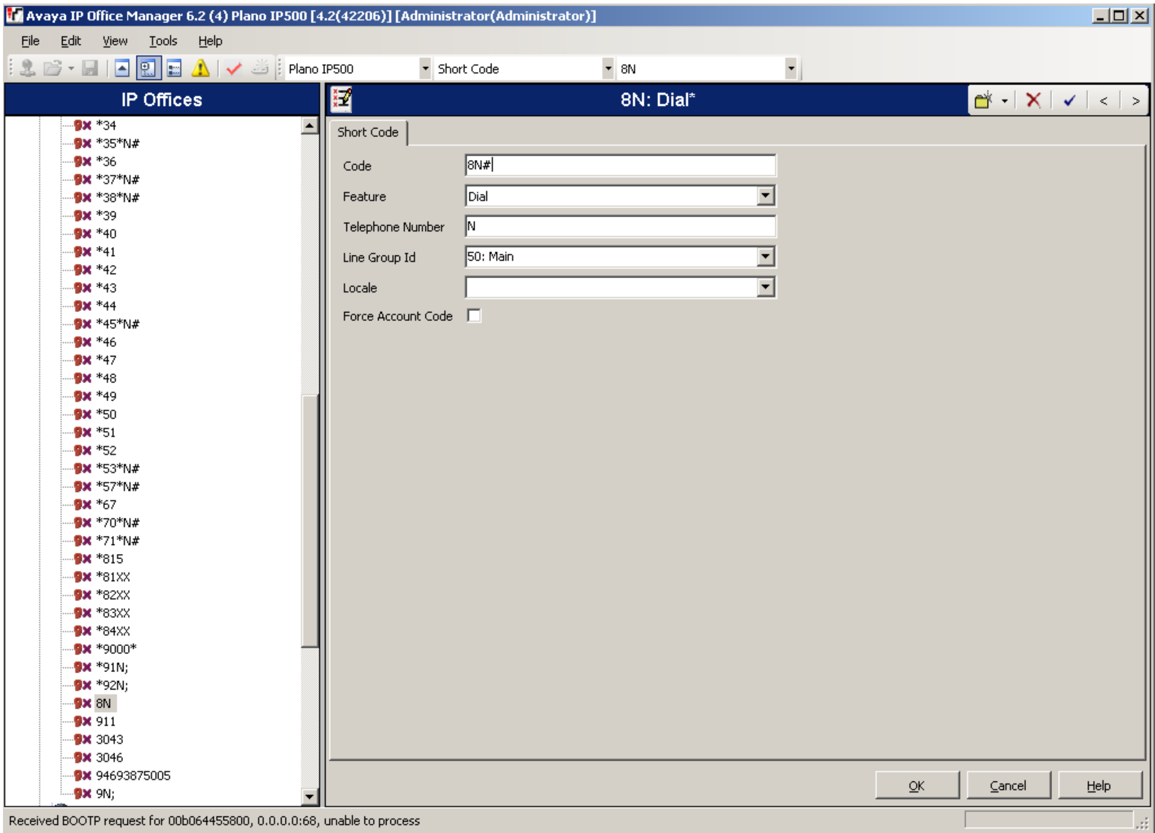


**Figure 11: Short Code Setting to Dial ARS**

Each target listed in Figure 10 is attempted from bottom to top, and Invite Timeout is fixed to approximately 32 seconds.

## Receiving Calls from XO

### Called Number Translation (Local)

The translation of each DID to an IP Office extension is done using the Incoming Call Rouging (ICR) field, as depicted in Figure 12

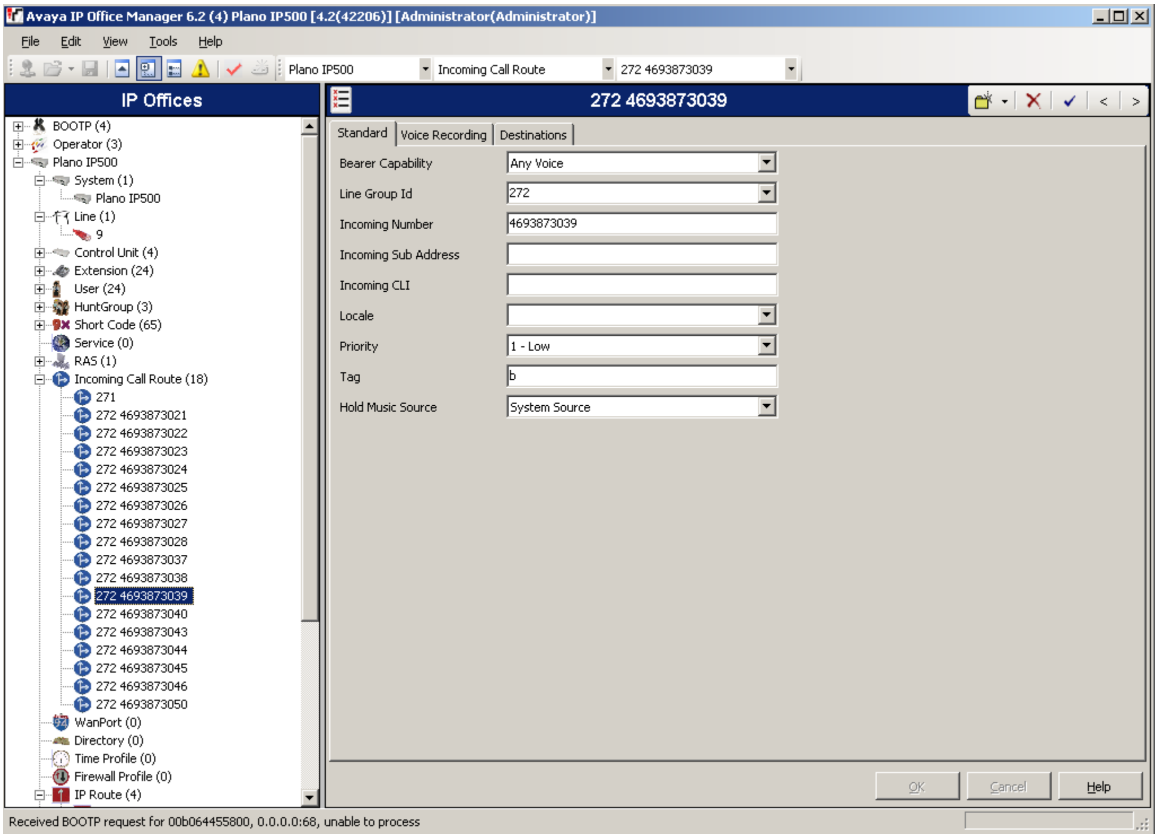


**Figure 12: Incoming Call Route**

The fields are populated as follows:

- *Bearer Capability* shall be set to *Any Voice.*
  *Line Group Id* shall be set to the *Incoming Group ID* of the *SIP URI* that is used to receive external phone calls.
- *Destination* shall be set to the desired target of incoming calls that can be an extension or hunt-group.

## 2.2 IP Phone Configuration

An example of IP phone configuration is given in Figure 13.

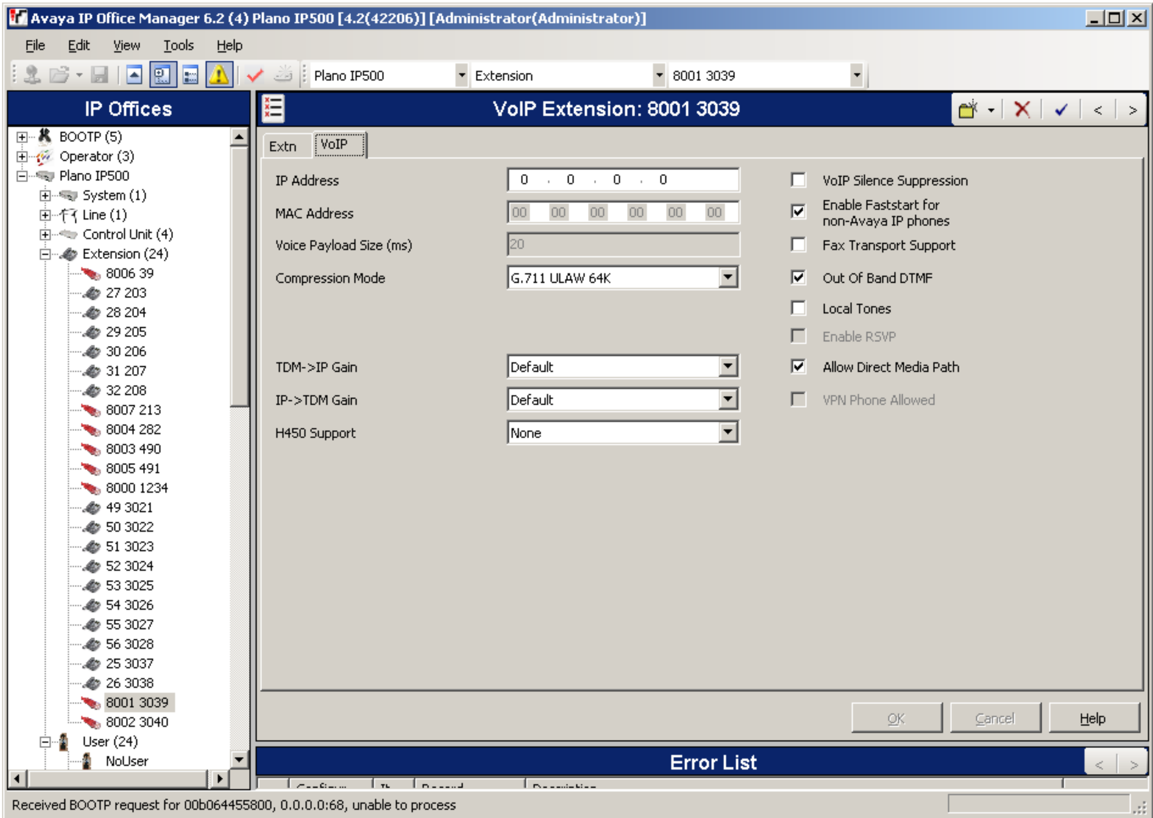


**Figure 13: IP Phone Configuration**

*Enable Fast Start for non Avaya phones* is enabled
*Allow Direct Media Path* can be left ticked or un-ticked depending on user preference. SIP trunks will always use relay.