OpenScape Office V3

Tutorial

AT&T SIP trunk configuration

Version 1.0

Definitions

HowTo

An OpenScape Business HowTo describes the configuration of an OpenScape Business feature within the OpenScape Office administration. It addresses primarily trained administrators of OpenScape Business.

Tutorial

Within the OpenScape Business tutorials procedures for installation, administration and operation of specific devices, applications or systems, which are connected to OpenScape Business, are described. The tutorial addresses primarily trained administrators of OpenScape Business.

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Table of History

Date	Version	Changes
2013-07-09	1.0	Initial Creation

1 Introduction

The purpose of this document is to provide the user with the suggested configuration steps for connecting the Siemens OpenScape Office Version 3.0 system to the AT&T IP Flexible Reach-Enhanced Features Service using MIS, PNT or AT&T Virtual Private Network.

2 Special Notes

Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

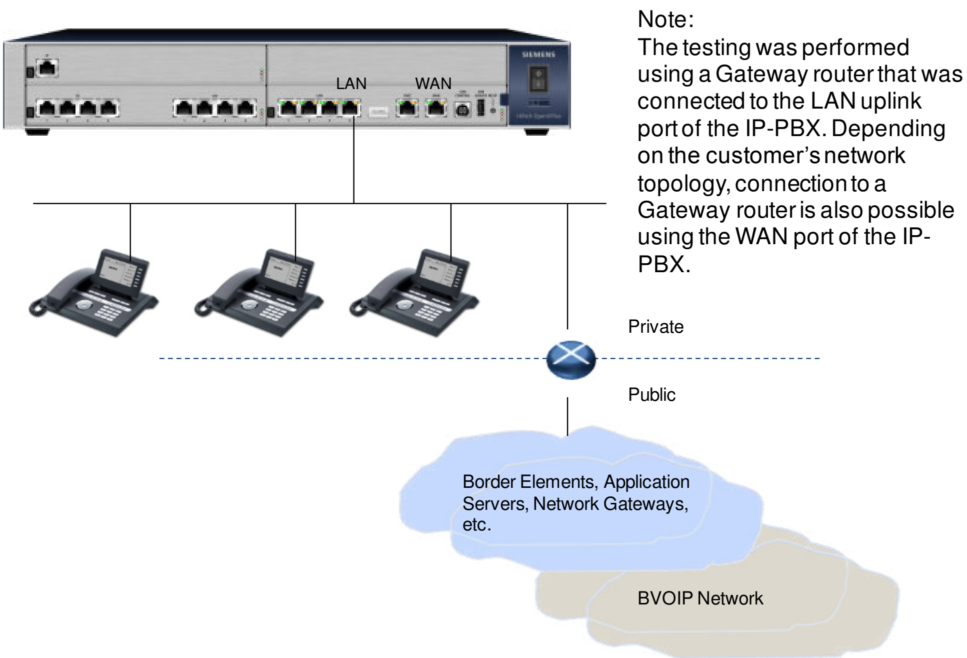
2.1 Items not Supported

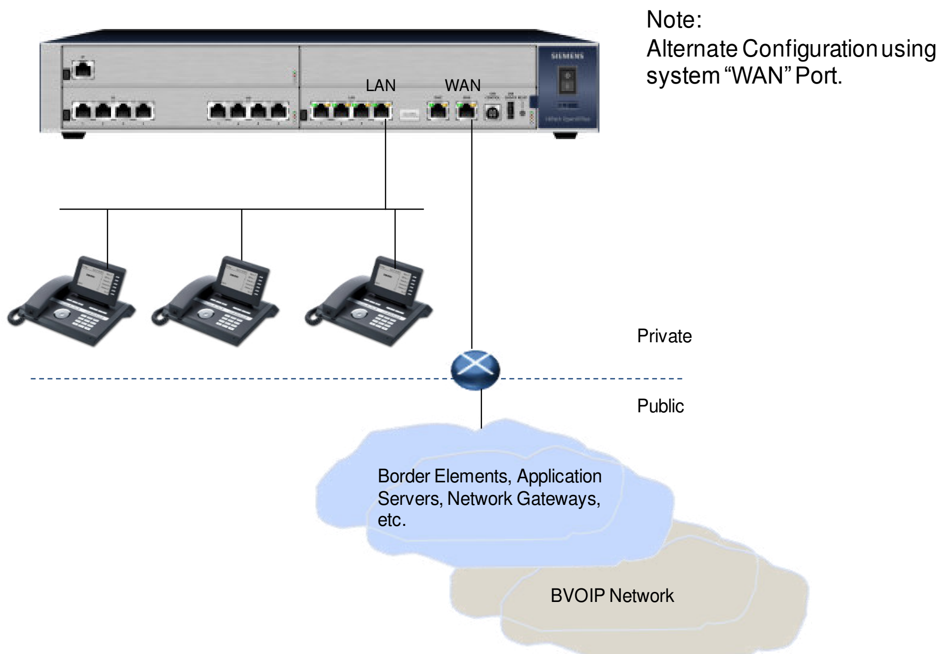
The following Table provides a list of items that are not supported between the Siemens OpenScape Office platform and the AT&T IP Flexible Reach-Enhanced Features Service using MIS, PNT or AT&T Virtual Private Network

Description	Comment
Pass display name. Verify display at calling and called parties	Currently Calling Name is not supported. Use of name-number correlation with Siemens system speed lists, external directory import list and Microsoft Contacts list may be used to associate a name with the calling party number at this time. Planned for future release
Mute call from CPE Phone using AT&T Feature set	Not Supported Use of system controlled MUTE at the desktop level does address this feature requirement.
Make a call that fails on both the Primary and the Secondary AT&T IPBE; Call routes to PSTN	Not Supported Unable to test and confirm failover from Primary to Secondary AT&T IPBE. Testing was confirmed from Primary AT&T IPBE to Test ISDN PRI connection to another Siemens System within test lab environment.
G711 fax applications	Not Supported If G3 (G.711) fax machines are to be used then additional non-SIP (Analog or ISDN PRI) trunks must be installed on the Siemens system or the customer must use Siemens approved Super G3 fax machines/applications or Siemens approved T.38 fax applications.
Enhanced IPFR Site with additional contacts and Simultaneous Ring feature	Not supported Please noted that the Siemens system does support station level ring groups for up to 4 parties as well as group level ring groups for up to 8 parties.
Enhanced IPFR Site with additional contacts and Sequential Ring feature	Not Supported Please note that the Siemens system does ring no answer programming to emulate the feature to allow incoming calls to signal additional internal or external destinations on a RNA condition.
Blind Transfer using REFER feature	Not supported

3 System Connectivity Overview

This section must provide an overview of the capabilities that are supported. More specifically, this section will contain a drawing of the equipment components that were tested. Each component and its associated release must be described. A high level call flow must also be provided.





4 Configuration Guide

4.1 Required Information

The following Example Information will be used throughout the configuration note. Please work with Customer's IT team and AT&T for values that will be associated with your project.

Description	Example	Specific Entries
IP Address LAN	192.168.1.8	
SN LAN	255.255.255.0	
DNS Address	192.168.1.1	
IP Address of Public Network Connection	67.164.46.22	
AT&T Domain Name / IP address	207.242.225.210	
AT&T Provider Proxy	207.242.225.210/ Port 5060	
AT&T Internet Telephony Station number	7323204076	
AT&T Internet Telephony Phone Numbers	7323204076 7323204077 7323204078 7323204072 7323204079	

4.2 Assumptions

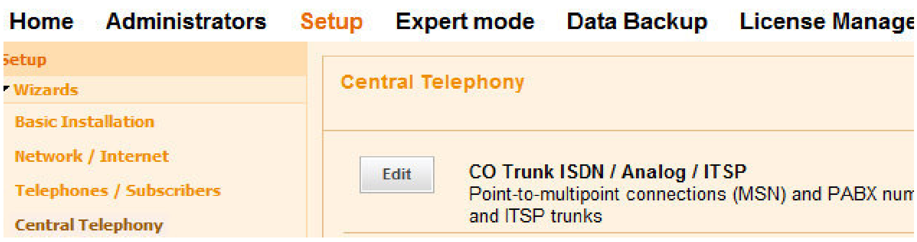
1. The system is running on the latest OpenScape Office Applications Software load
2. The system has been configured to connect to the Internet using the LAN port with an external Router or using the WAN Port with an external router
3. With Assumption 2 completed you are able to use the PING test under the path Expert Mode > Routing > IP Routing > ICMP request > Ping to initiate a successful Ping request to a public IP address and FQDN address of a public sight.
4. The customer engineer installing the SIP trunks has successfully completed the OpenScape Office Installation and Maintenance course curriculum.

5 Configuration Steps

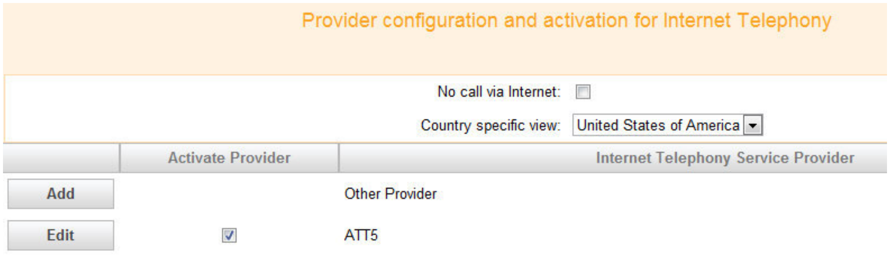
5.1 Step 1. Select and Enable the AT&T Profile and LCR code for SIP Group Access

This step is required to select the AT&T profile and activate the default Least Cost Routing access code "9" for use by the AT&T SIP Route Trunk Group.

1. From the Main Menu select Setup > Central Telephony > CO Trunk ISDN / Analog / ITSP click the associated Edit button.

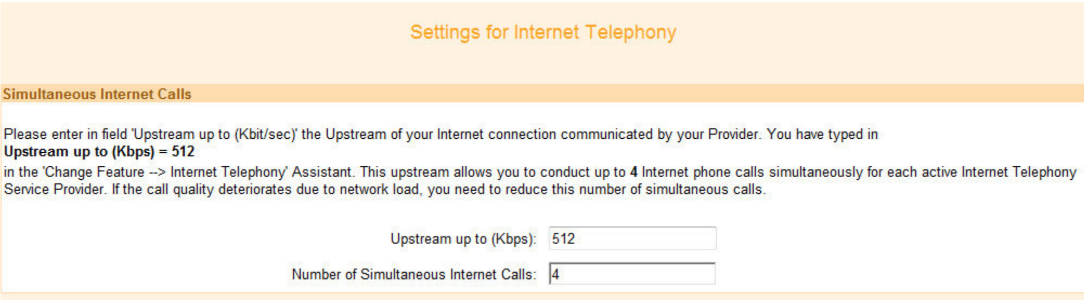


2. Click the "Ok and Next" button on the "Overview: Page to proceed to the next form.
3. On the Provider Configuration and activation for Internet Telephony page



   a. Insure that the "No Call via Internet" check box is disabled
   b. Insure that the "Country specific view" selection is "United States of America"
   c. Enable the Check box associated with the AT&T Profile to activate the Provider
   d. Press the OK & Next Button

4. On the Settings for Internet Telephony form
    a. Enter in the number of simultaneous internet calls that will be managed by the system. The maximum amount of sessions for the OpenScape Office MX is 32. The Maximum amount of sessions for the OpenScape Office LX system is 120.

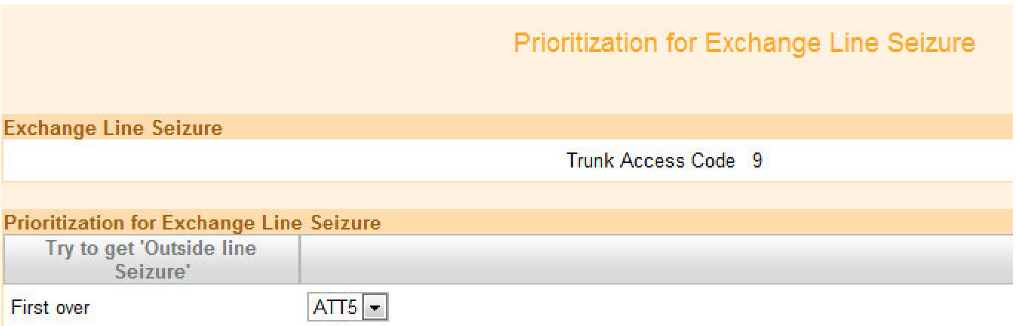


    b. Press the OK & Next Button

5. On the Special Phone Numbers form
    a. Press the OK & Next Button

6. On the Status for the Internet Telephony Service Provider (ITSP) form press the Next button

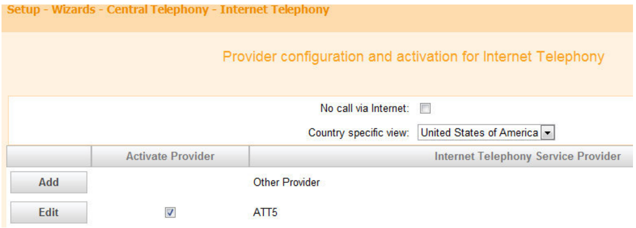
7. On the Prioritization for Exchange Line Seizure form
    a. Select the AT&T profile as the first choice when the user dials the access code "9".
    b. Press the OK & Next Button



8. On the Seizure Code for the 'Outside line Seizure' form
    a. Press the OK & Next Button
    b. Press the Finish button on the final form to complete the Wizard.

5.2 Step 2 Configure SIP Parameters for AT&T SIP trunks

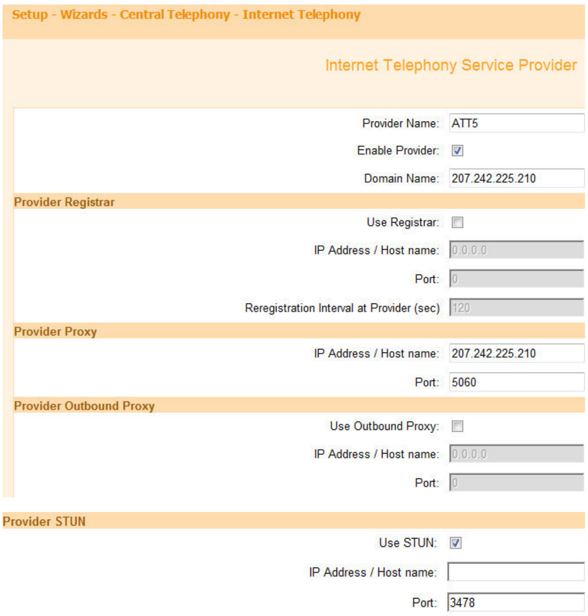
1. From the Main Menu select Setup > Central Telephony > Internet Telephony and then press the Edit button.
2. From the Provider configuration and activation for Internet Telephony Form, press the Edit button associated with the AT&T profile.



3. On the Internet Telephony Service Provider form the information associated with the selected profile will be displayed. Please insure that the following information is in place and confirm the entries with your AT&T representative. The default profile for AT&T is based on the results of the testing in the Siemens Lab. Any changes to the default information may require the addition of a new profile to the OpenScape Office system database.

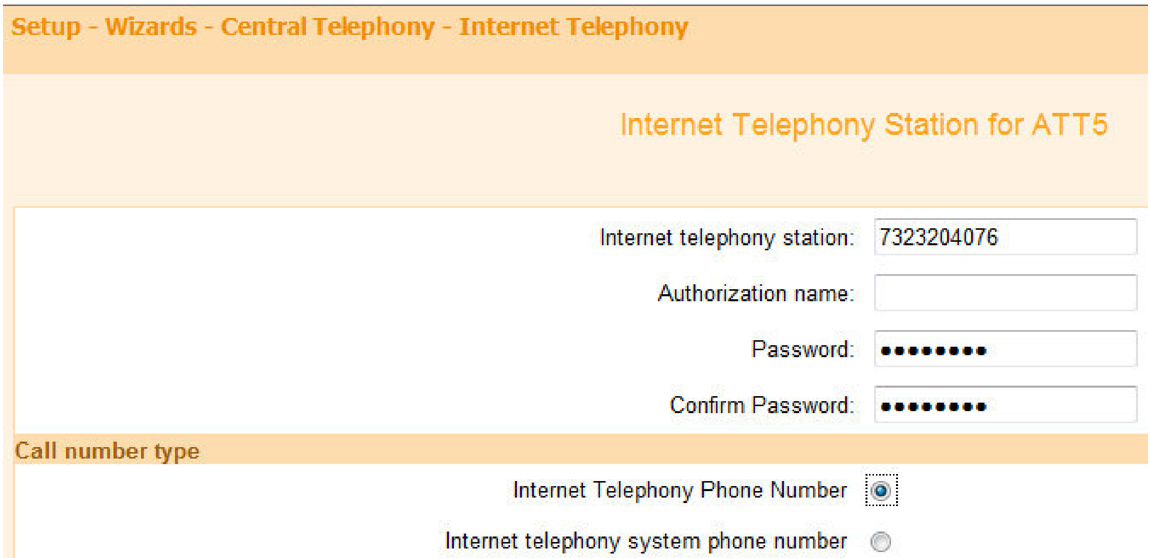
    -

    a. Provider Name is listed
    b. The Enable Provider box is enabled
    c. The Domain Name or IP address is displayed.
    - It is recommended that the Domain naming information be confirmed with AT&T
    d. The Provider Registrar > Use Registrar box is disabled
    e. The Provider Proxy IP Address/Host name is filled in and the displayed port is 5060.
    - It is recommended that the Domain naming information be confirmed with AT&T
    f. The Provider Outbound Proxy box is disabled
    g. The Provider STUN box is enabled and the Port ID is 3478.
    h. After confirming the information press the Ok & Next Button

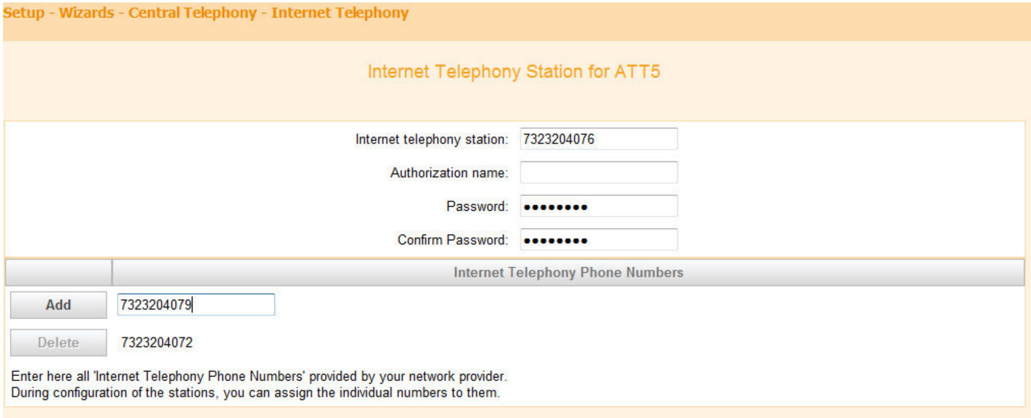


4. From the Internet Telephony Stations for ATT5 From, press the ADD Button to configure your first Internet Telephony station number. Typically only 1 entry per system will be required.

5. On the Internet Telephony Station for the AT&T profile enter the main Internet telephony station number (Telephone number) received from AT&T. Highlight and delete any information in the Authorization name, Password and Confirm Password fields.



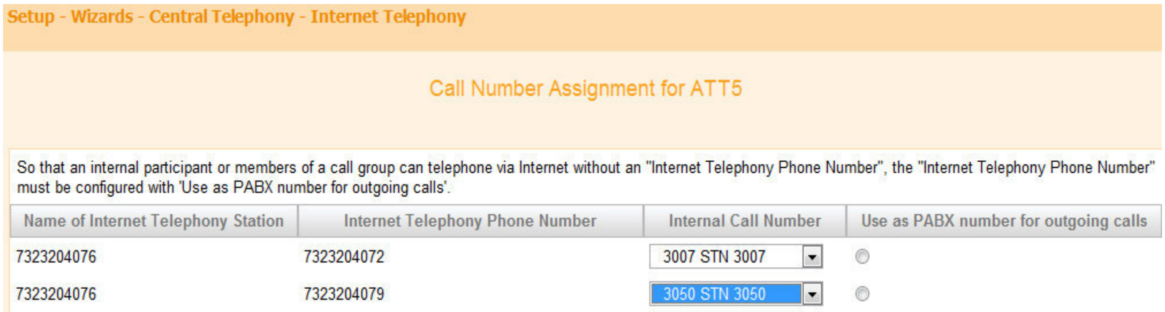
6. On the lower portion Internet Telephony Station for the AT&T From, enter the all of the Internet telephone numbers received from AT&T. After entering each telephone number press the Add button to store the information. Continue the process for the balance of the numbers. Press the OK & Next button after all the Internet Telephony Phone Numbers have been entered.



7. Press the OK & Next button when the Internet Telephony Station for the AT&T profile > Name of Internet Telephony Station form appears

8. On the Internet Telephony Station for the AT&T profile > Call Number Assignment for AT&T Profile form,
    a. Select the target station number or group from the list box associated with each Telephone number. This will provide the direct inward dial assignment for each telephone number.
    It will also provide the calling number information that will be displayed when the station places an outbound call.
    Enabling the “Use as PABX number for outgoing calls” button next to one of the Call Number assignments will allow internal stations or members of a call group not assigned a Internet Telephony Phone Number to place outbound calls and display the selected Telephone number information. Only one number may be selected per system.

    b. Press the OK & Next Button after entering all of the assignments



Setup - Wizards - Central Telephony - Internet Telephony

Call Number Assignment for ATT5

So that an internal participant or members of a call group can telephone via Internet without an "Internet Telephony Phone Number", the "Internet Telephony Phone Number" must be configured with 'Use as PABX number for outgoing calls'.

Name of Internet Telephony Station	Internet Telephony Phone Number	Internal Call Number	Use as PABX number for outgoing calls
7323204076	7323204072	3007 STN 3007	
7323204076	7323204079	3050 STN 3050	

9. Press the OK & Next button when the Provider configuration and activation for Internet Telephony form is displayed.
10. Press the OK & Next button when the "Settings for Internet Telephony" form is displayed.
11. Press the OK & Next button when the Special phone numbers form is displayed.
12. When the Status for the Internet Telephony Service Provider (ITSP) form appears confirm that the AT&T profile is registered.
    a. Press the next key

Status for the Internet Telephony Service Provider (ITSP)

	Provider		User	
	ATT5	Enabled	7323204076	registered
	COLT UK & Europe	Disabled		

If the Profile did not register the please refer to the Trouble shooting connection.

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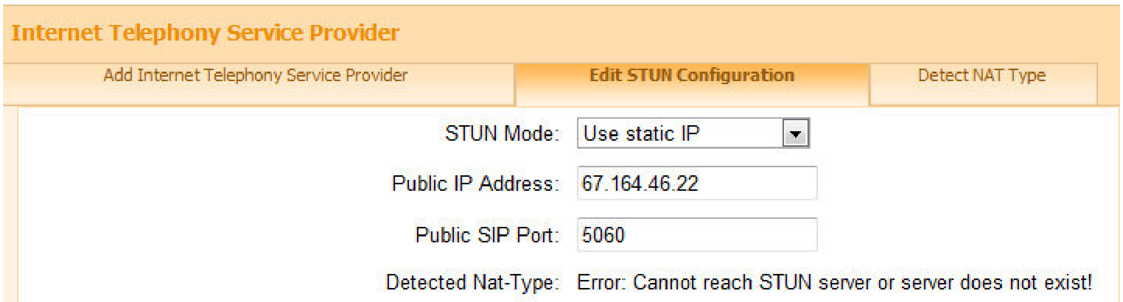
13. When the “The changes for the feature “‘Internet Telephony are completed” form is displayed press the Finish button to complete the wizard.

5.3 Step 3. Configure the STUN setting

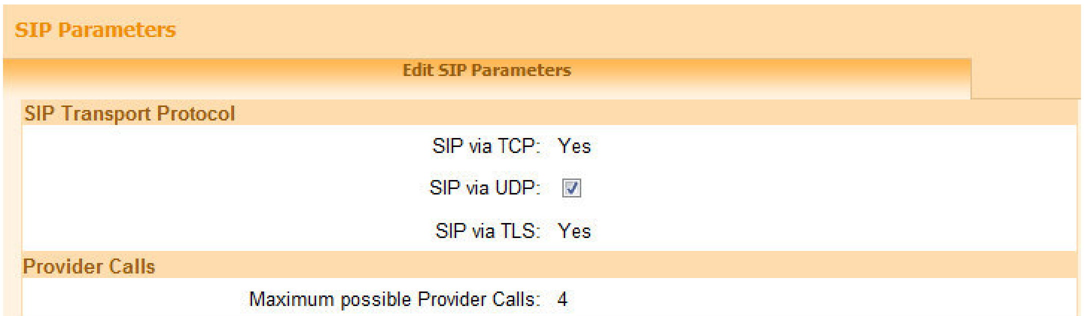
1. From the main menu select Expert Mode > Telephony Server > Voice Gateway > Internet Telephony Service Provider > Edit STUN Configuration.



   a. Select "Use Static" IP from the STUN Mode List box
   b. Enter the public IP address for your Internet connection
   c. Enter the Public SIP Port as "5060"
   d. Press the "Apply" button to accept the changes
   -

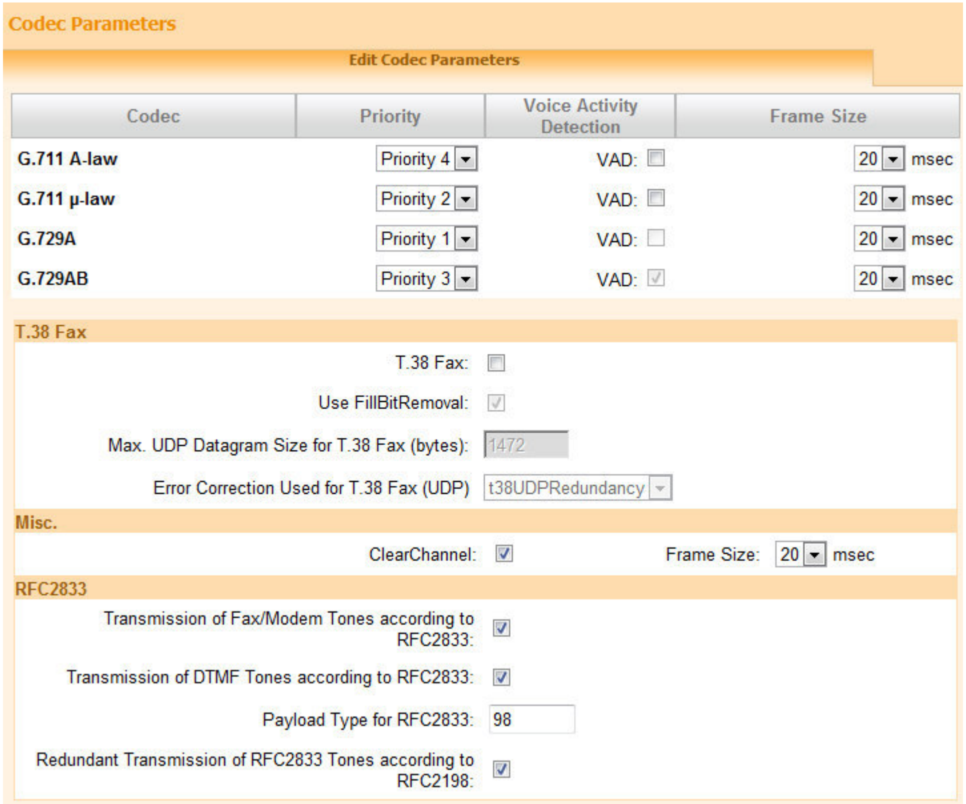
5.4 Step 4. Configure the System SIP Parameters

1. From the main menu select Expert Mode > Telephony Server > Voice Gateway > SIP Parameters > Edit SIP Parameters form
   a. Insure that the SIP via UDP check box is enabled

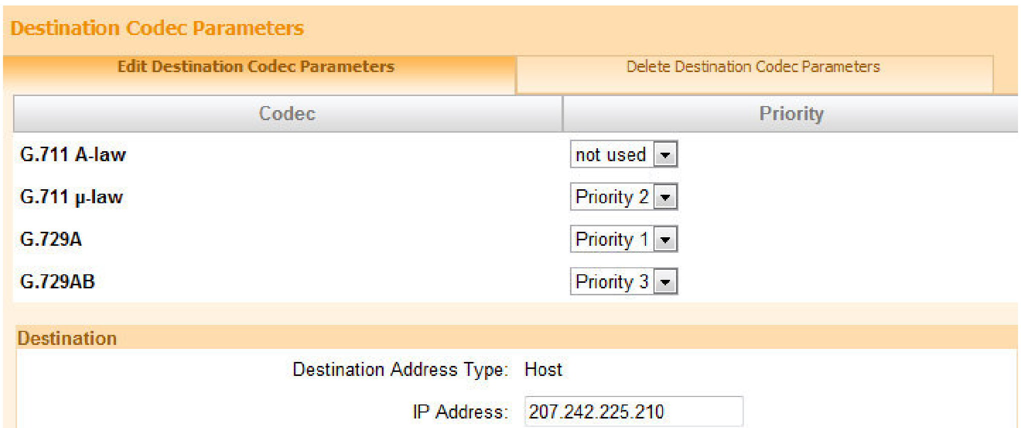


5.5 Step 5 Configure the Codec Parameters

1. From the main menu select Expert Mode > Telephony Server > Voice Gateway > Codec Parameters.
    a. Fill in the form as depicted in the diagram below.
    b. Press the Apply button to write and save the information to the database



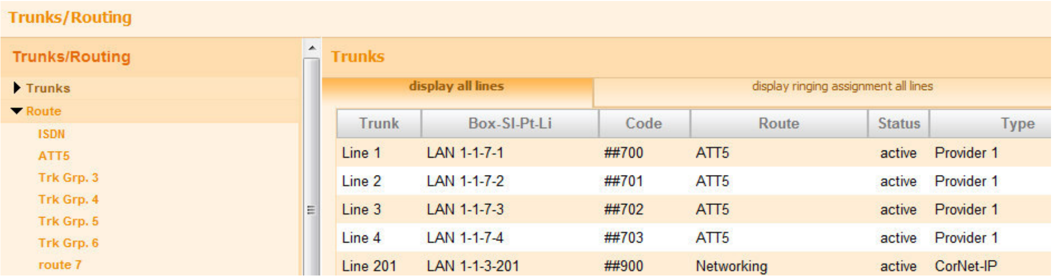
2. From the main menu select Expert Mode > Telephony Server > Voice Gateway > Codec Parameters > Destination Codec Parameters > Edit Destination Codec Parameters
    a. Fill in the form as depicted in the diagram below. The IP address represents the destination address for the AT&T services
    b. Press the Apply button to write and save the information to the database



5.6 Step 6 Configuration Least Cost Routing

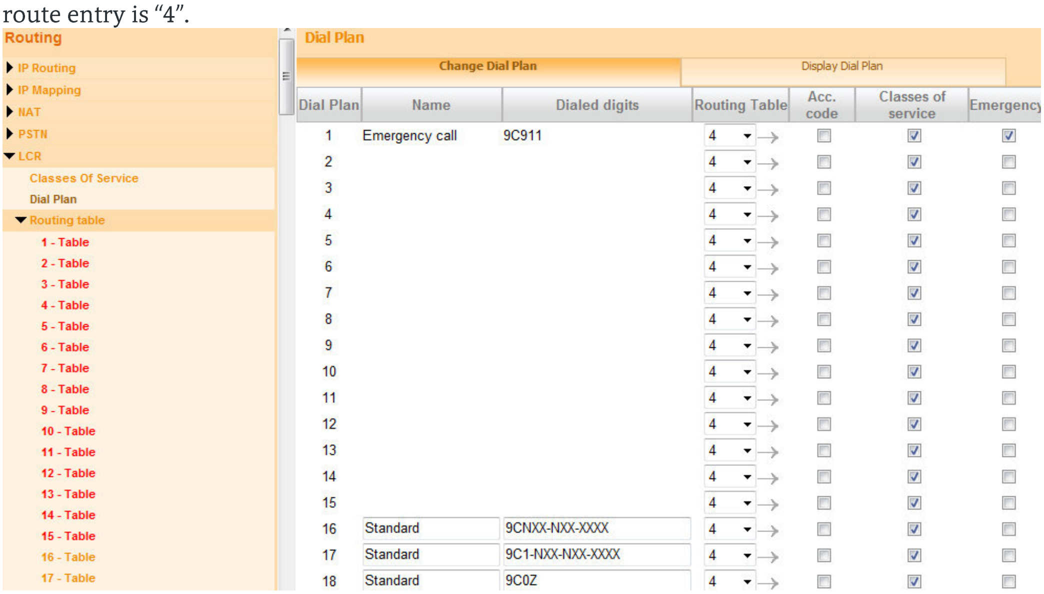
1. Confirm that the AT&T Trunk Route Group and configured channels have been created.
   By selecting the AT&T profile (Step 2 > Item 2) as the only provider to be used by the system, the AT&T group will automatically be assign to Trunk Route Group number 2. You can confirm this by selecting the path Expert > Trunks/Routing > Route

   You can also display the confirm the number of sessions entered as well as the connection status by selecting the path Expert > Trunks/Routing > Trunks



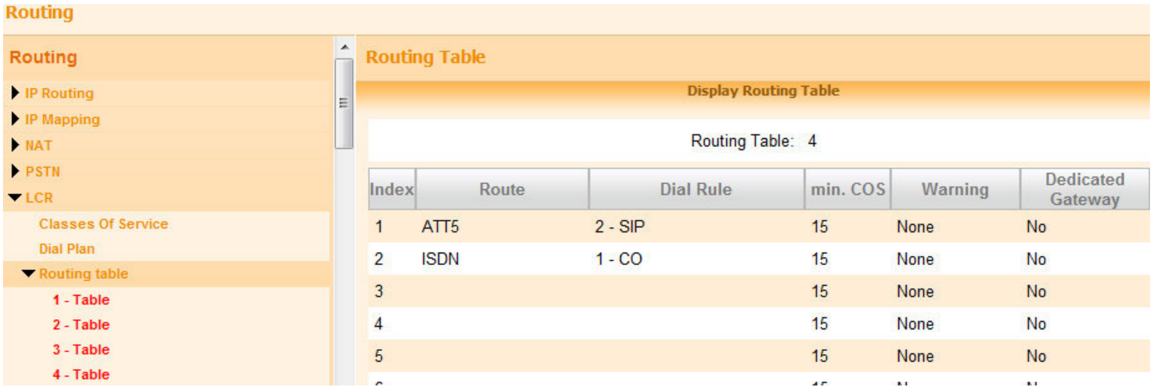
Trunk	Box-SI-Pt-Li	Code	Route	Status	Type
Line 1	LAN 1-1-7-1	##700	ATT5	active	Provider 1
Line 2	LAN 1-1-7-2	##701	ATT5	active	Provider 1
Line 3	LAN 1-1-7-3	##702	ATT5	active	Provider 1
Line 4	LAN 1-1-7-4	##703	ATT5	active	Provider 1
Line 201	LAN 1-1-3-201	##900	Networking	active	CorNet-IP

2. Access the Least Cost Routing Dial Plan table using the following Path; Expert > Routing > LCR > Dial plan. Confirm that all of the standard dial plan entries are assigned a Route Table entry. In our example the default route entry is “4”.



Dial Plan	Name	Dialed digits	Routing Table	Acc. code	Classes of service	Emergency
1	Emergency call	9C911	4	☐	☑	☑
2			4	☐	☑	☐
3			4	☐	☑	☐
4			4	☐	☑	☐
5			4	☐	☑	☐
6			4	☐	☑	☐
7			4	☐	☑	☐
8			4	☐	☑	☐
9			4	☐	☑	☐
10			4	☐	☑	☐
11			4	☐	☑	☐
12			4	☐	☑	☐
13			4	☐	☑	☐
14			4	☐	☑	☐
15			4	☐	☑	☐
16	Standard	9CNXX-NXX-XXXX	4	☐	☑	☐
17	Standard	9C1-NXX-NXX-XXXX	4	☐	☑	☐
18	Standard	9C0Z	4	☐	☑	☐

3. Display the Routing Table entry using the Path; Expert > Routing > LCR > Routing Tables. Please confirm that the AT&T Trunk Route Group is the first choice.



Note: If the customer is planning on using a fax machine or the OpenScape Office personal Fax application you will need to select another Routing Table, not highlighted in bold red, and replicate all of the information found in the default Route Table 4 with the exception of the Min COS setting. When Fax applications are used the Min COS setting must be configured as a class 1.

Once you have defined the new Route Table, return to the dial plan Table and change the default entry to the new table.

6 Trouble Shooting Information

6.1 No indication of any call set up information is seen in a Wire shark Trace.

- In the case of the OpenScape Office system confirm that you have configured a port (LAN Port or WAN Port) that will be used for Internet access.
- Insure that the system is able to reach the Public Internet over the configured LAN or WAN port using the PING command.
- Insure that the AT&T profile has been activated and is designated as SIP provider 1
- Insure that the AT&T Domain name or entered IP address from the profile covered in step 2.3 is correct.
- Insure that the Provider Proxy name or entered IP address from the profile covered in step 2.3 is correct.
- Insure that the static public IP address information you provided to AT&T is correct
- Insure that you are physically connected to the Internet connection on the defined port (LAN Port or WAN Port).

6.2 Problem: Not able to receive incoming calls

- Insure that each IP SIP telephone number is associated with a system station or group.

6.3 Problem: Not able to place an outbound calls

- Insure that the Provider Proxy name or entered IP address from the profile covered in step 2.3 is correct.
- Confirm that the dial 9 access code has been configured. Step 1.1 through step 1.8.
- Confirm that the number pattern dialed is included in the LCR dial plan table and a valid LCR Route has been assigned.
- Insure that the LCR Route selected contains the AT&T trunk route group.
- Confirm that the station dialing has been assigned a telephone number or one of the numbers has been flagged with the "Use as PABX number for Outgoing calls".

About Unify

Unify is one of the world's leading communications software and services firms, providing integrated communications solutions for approximately 75 percent of the Fortune Global 500. Our solutions unify multiple networks, devices and applications into one easy-to-use platform that allows teams to engage in rich and meaningful conversations. The result is a transformation of how the enterprise communicates and collaborates that amplifies collective effort, energizes the business, and enhances business performance. Unify has a strong heritage of product reliability, innovation, open standards and security.

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