ALLWORX

Software Version: 8.0.11.5

FUSION SIP TRUNKING

NOVEMBER 2015

INTRODUCTION

This guide assists users to configure the Allworx VoIP Phone System and Fusion SIP Trunking.

PREREQUISITES

- Completed the Allworx Technical training, and the main technician is either an Allworx Certified Administrator (ACA) or Allworx Certified Professional (ACP).
- Setup all other functions within the Allworx system prior to connecting Fusion SIP Trunking (e.g., DHCP settings and installed the latest software version).
- Ordered Fusion SIP Trunking and received the associated configuration information.

IMPORTANT NOTES

This configuration was tested with Allworx server software 8.0.11.5.

The latest software is available at: https://allworxportal.com/support_training/software.aspx.

Currently the Allworx platform does not support T.38 for fax services nor TLS and SRTP.

CONTENTS:

SETTING UP THE ALLWORX SYSTEM

1. Complete and test the following configurations before connecting to the SIP proxy.
    a. Local Area Network has connectivity. Access to the Admin Web GUI. Register at least two local Allworx IP phones on the LAN with the Allworx server and can place station to station calls with each and the server (access voicemail, auto attendants, etc.)
    b. Wide Area Network has connectivity. Log in to the Allworx server admin page, and navigate to Maintenance > Tools. Locate the Network Diagnostics section and enter an IP Address or Domain Name in the field on line 1. Click Ping. Verify the Allworx server successfully pings the gateway IP and an external IP address such as a public DNS server. If either of these fails, contact the Network Administrator to correct any configuration issues before continuing with the SIP Proxy configuration.

    The Allworx server was tested with Fusion SIP Trunking with the following Network Layout (Figure 1) and Network Configuration (Figure 2).

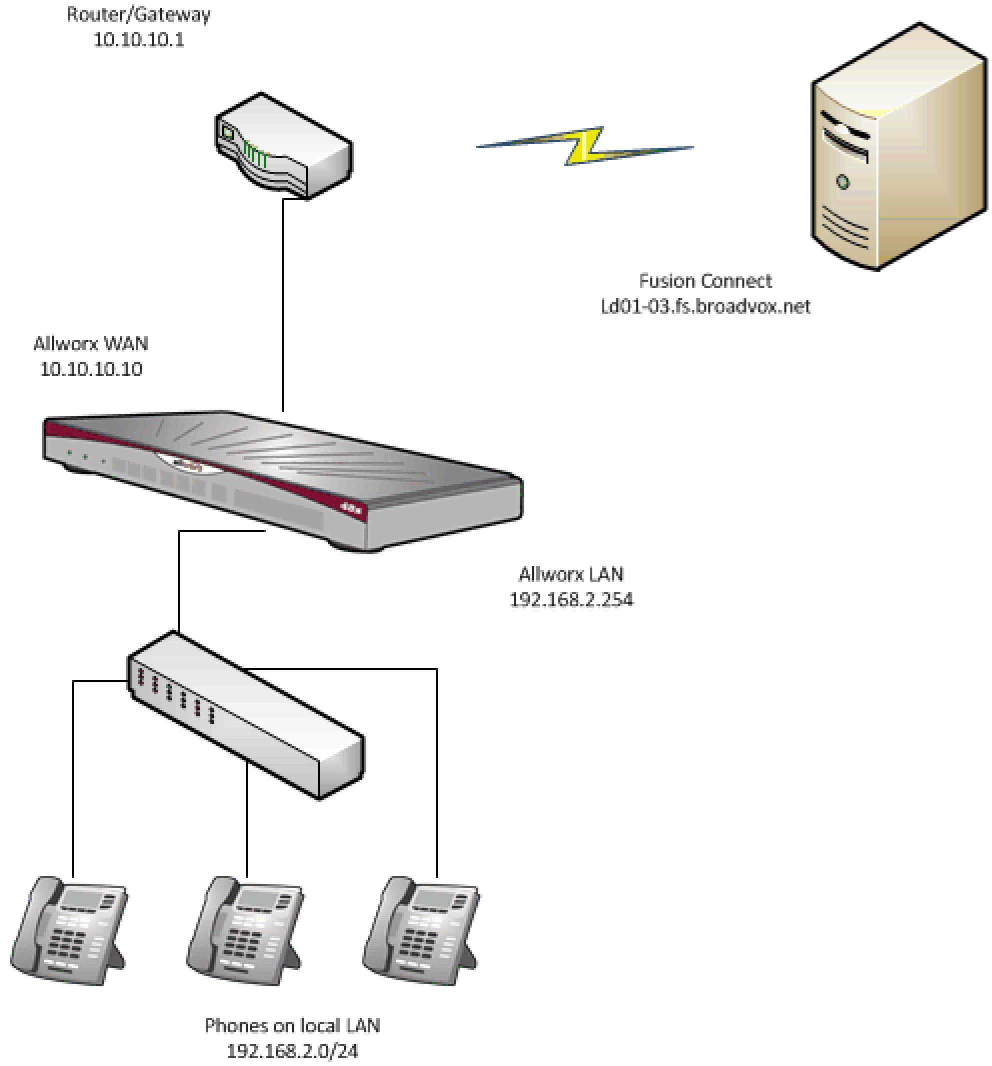

Router/Gateway
10.10.10.1
Fusion Connect
Ld01-03.fs.broadvox.net
Allworx WAN
10.10.10.10
Allworx LAN
192.168.2.254
Phones on local LAN
192.168.2.0/24

Figure 1

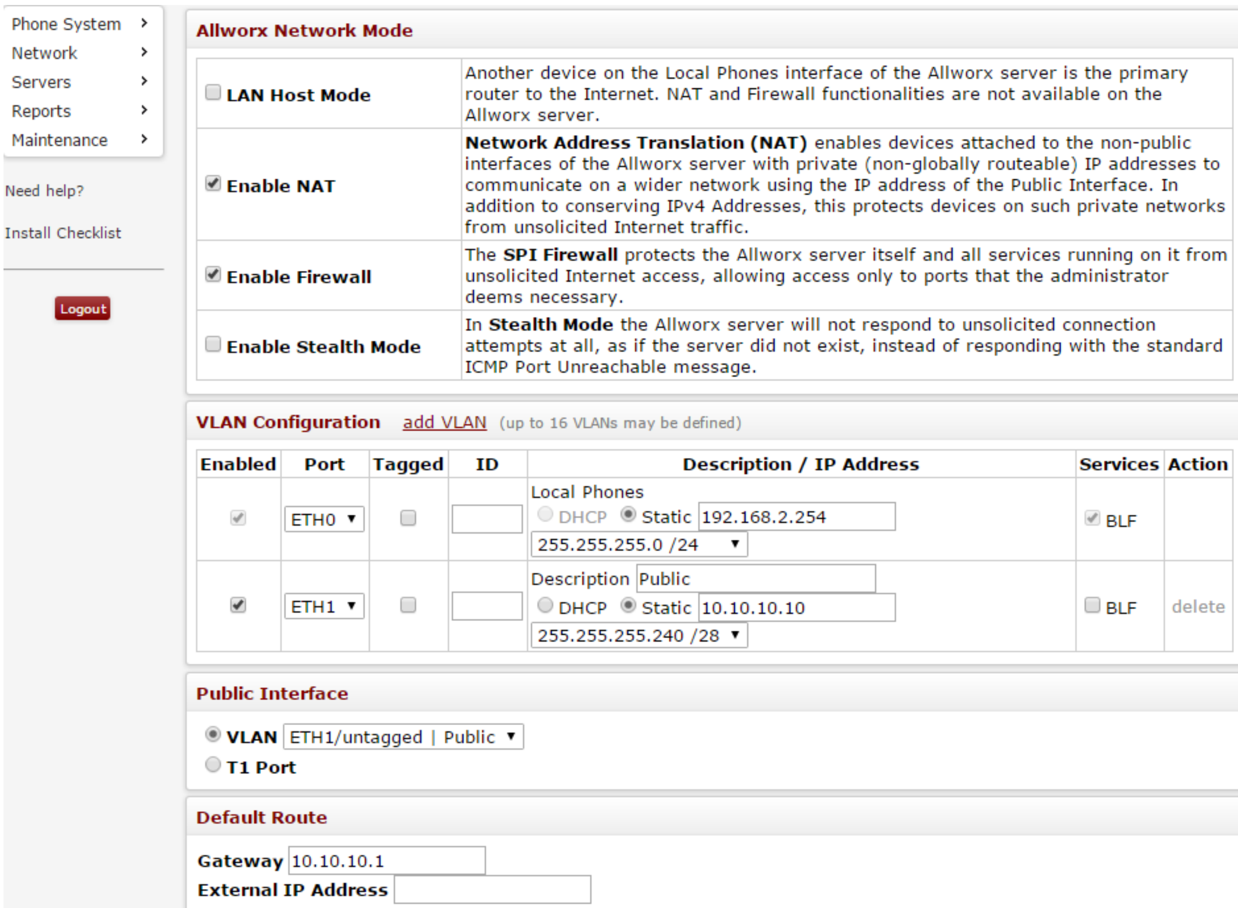
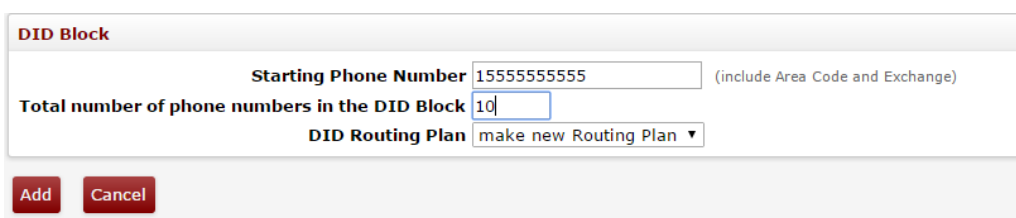

Phone System
Network
Servers
Reports
Maintenance
Need help?
Install Checklist
Logout
Allworx Network Mode
LAN Host Mode
Another device on the Local Phones interface of the Allworx server is the primary router to the Internet. NAT and Firewall functionalities are not available on the Allworx server.
Enable NAT
Network Address Translation (NAT) enables devices attached to the non-public interfaces of the Allworx server with private (non-globally routeable) IP addresses to communicate on a wider network using the IP address of the Public Interface. In addition to conserving IPv4 Addresses, this protects devices on such private networks from unsolicited Internet traffic.
Enable Firewall
The SPI Firewall protects the Allworx server itself and all services running on it from unsolicited Internet access, allowing access only to ports that the administrator deems necessary.
Enable Stealth Mode
In Stealth Mode the Allworx server will not respond to unsolicited connection attempts at all, as if the server did not exist, instead of responding with the standard ICMP Port Unreachable message.
VLAN Configuration
add VLAN
(up to 16 VLANs may be defined)
Enabled
Port
Tagged
ID
Description / IP Address
Services
Action
ETH0
Local Phones
DHCP
Static
192.168.2.254
255.255.255.0 /24
BLF
ETH1
Description
Public
DHCP
Static
10.10.10.10
255.255.255.240 /28
BLF
delete
Public Interface
VLAN
ETH1/untagged | Public
T1 Port
Default Route
Gateway
10.10.10.1
External IP Address

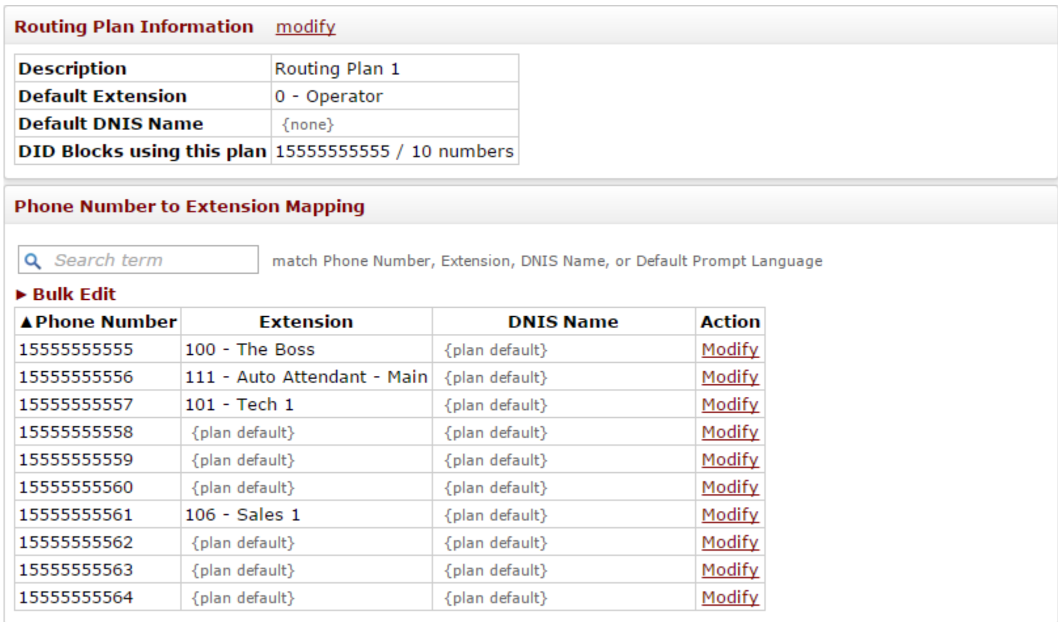
Figure 2

Figure 2 (con't)

2. (Optional) Setup the DID Block and DID Routing Plan to use with the SIP Proxy. The cut-sheet received from Fusion provides the available numbers.
    a. DID block: Log in to the Allworx server admin page, and navigate to **Phone System** > **Outside Lines**. Locate the Direct Inward Dial Blocks section and click **add new DID block**.



b. Build the routing plan and map each DID to the appropriate extensions or destinations such as Call Queues, Auto Attendants, Conference Center, etc. Navigate to Phone System > Outside Lines > DID Routing Plan. Locate the Phone Number to Extension Mapping section, and click the appropriate Modify link. Using the Extension drop-down arrow, select the extension.



Routing Plan Information modify

Description	Routing Plan 1
Default Extension	0 - Operator
Default DNIS Name	{none}
DID Blocks using this plan	15555555555 / 10 numbers

Phone Number to Extension Mapping

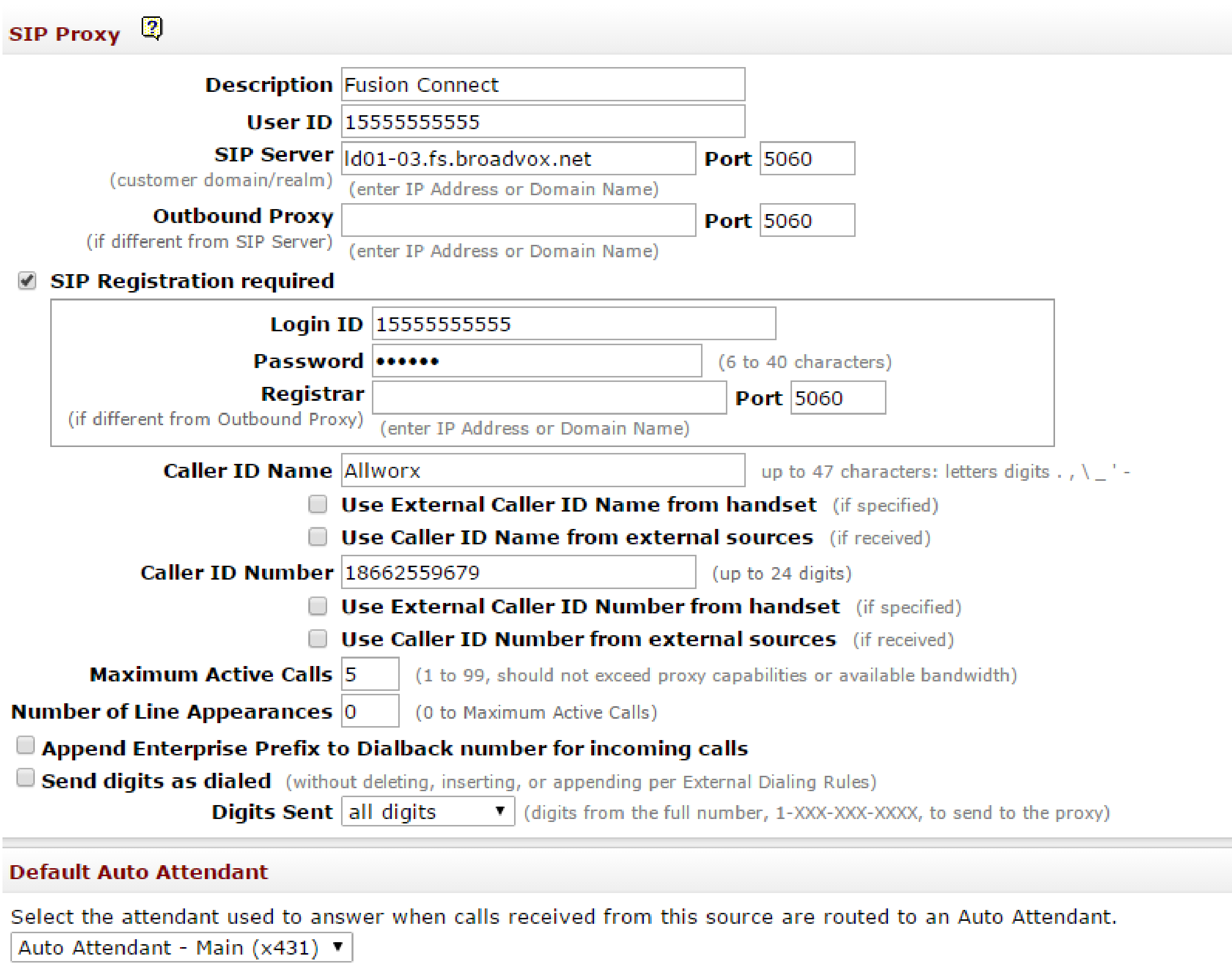
Search term match Phone Number, Extension, DNIS Name, or Default Prompt Language

► **Bulk Edit**

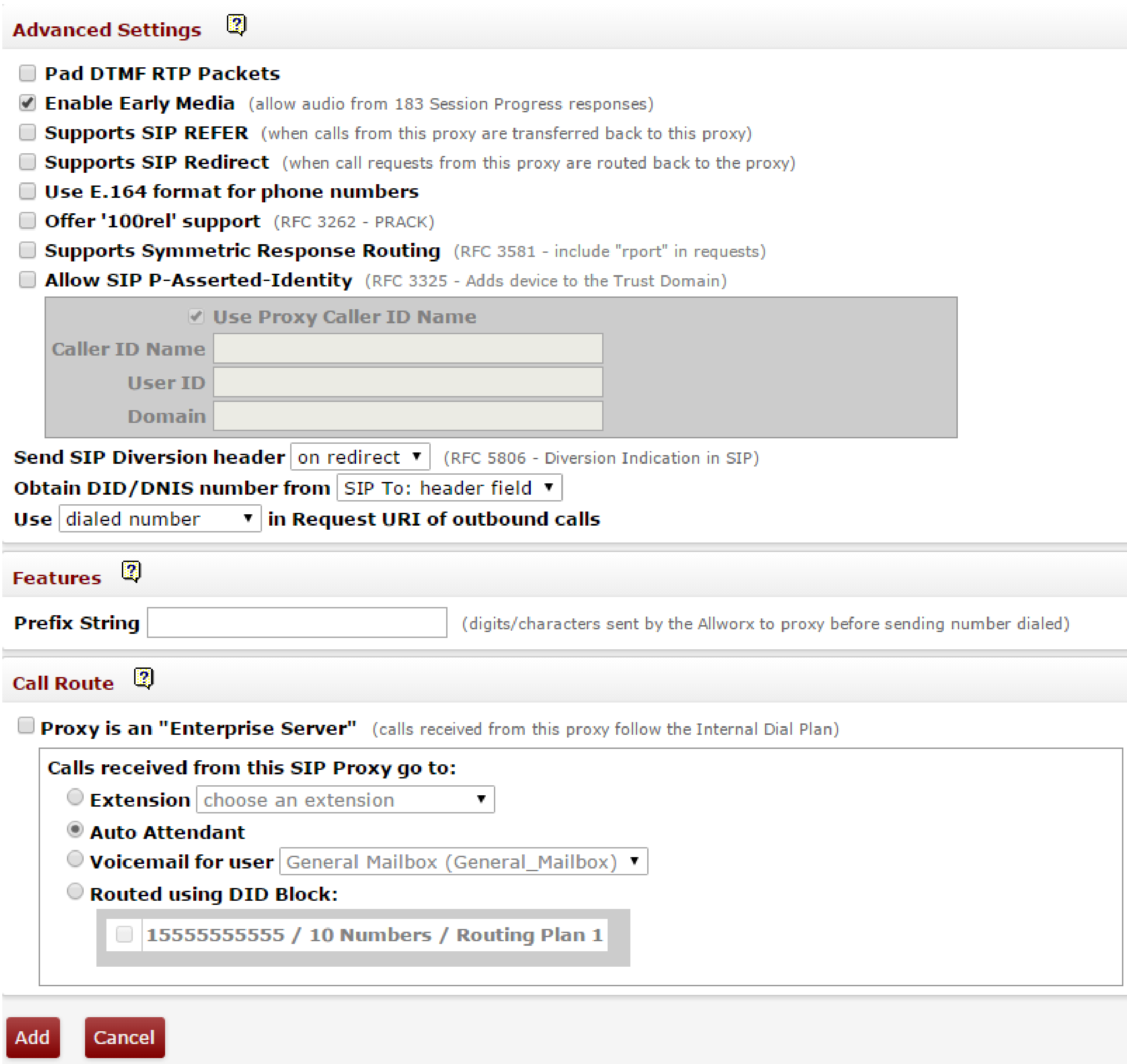
▲Phone Number	Extension	DNIS Name	Action
15555555555	100 - The Boss	{plan default}	Modify
15555555556	111 - Auto Attendant - Main	{plan default}	Modify
15555555557	101 - Tech 1	{plan default}	Modify
15555555558	{plan default}	{plan default}	Modify
15555555559	{plan default}	{plan default}	Modify
15555555560	{plan default}	{plan default}	Modify
15555555561	106 - Sales 1	{plan default}	Modify
15555555562	{plan default}	{plan default}	Modify
15555555563	{plan default}	{plan default}	Modify
15555555564	{plan default}	{plan default}	Modify

3. Configure the SIP Proxy.
    a. Navigate to **Phone System** > **Outside Lines** > **SIP Proxies** > **add new SIP Proxy**. Have the main telephone number available, which is usually referred to as the BTN, Billing Telephone Number and the IP address of the SIP server. In the example the BTN is 1-555-555-5555 and the SIP server is ld01-03.fs.broadvox.net.

Field	Recommended Setting
SIP Proxy	
Description	User assigned label such as, Fusion SIP Trunking.
User ID	Provided by Fusion should be the main telephone number and/or BTN.
SIP Server	Provided by Fusion, tested with ld01-03.fs.braodvox.net.
SIP Server Portal	Default value is 5060.
Outbound Proxy	Leave Blank.
Outbound Proxy Port	Leave Blank.
SIP Registration Required	Checked.
Login ID	Provided by Fusion.
Password	Provided by Fusion.
Registrar	Leave Blank.
Registrar Port	Leave Blank.
Caller ID Name	User/Fusion Supplied.
Caller ID Number	User/Fusion Supplied.
Maximum Active Calls	Provided by Fusion.
Number of Line Appearances	Default value of 0.
Append Enterprise prefix…	Leave Blank.
Send Digits as dialed	Unchecked if using ARS, checked if always dialing 11 digits.
Digits Sent	Select all digits.
Default Language	User specified.
Default Auto Attendant	This is a customer specific setting and defines which automated attendant plays for each incoming call that ends up at the AA.


SIP Proxy
Description
Fusion Connect
User ID
15555555555
SIP Server
(customer domain/realm)
ld01-03.fs.broadvox.net
Port
5060
(enter IP Address or Domain Name)
Outbound Proxy
(if different from SIP Server)
Port
5060
(enter IP Address or Domain Name)
SIP Registration required
Login ID
15555555555
Password
••••••
(6 to 40 characters)
Registrar
(if different from Outbound Proxy)
Port
5060
(enter IP Address or Domain Name)
Caller ID Name
Allworx
up to 47 characters: letters digits . , \ _ ' -
Use External Caller ID Name from handset (if specified)
Use Caller ID Name from external sources (if received)
Caller ID Number
18662559679
(up to 24 digits)
Use External Caller ID Number from handset (if specified)
Use Caller ID Number from external sources (if received)
Maximum Active Calls
5
(1 to 99, should not exceed proxy capabilities or available bandwidth)
Number of Line Appearances
0
(0 to Maximum Active Calls)
Append Enterprise Prefix to Dialback number for incoming calls
Send digits as dialed (without deleting, inserting, or appending per External Dialing Rules)
Digits Sent
all digits
(digits from the full number, 1-XXX-XXX-XXXX, to send to the proxy)
Default Auto Attendant
Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.
Auto Attendant - Main (x431)

Advanced Settings	
Pad DTMF RTP Packets	Unchecked
Enable Early Media	Checked
Supports SIP REFER	Unchecked
Supports SIP Redirect	Unchecked
Use E.164 format...	Unchecked
Offer '100rel' support	Unchecked
Supports Symmetric...	Unchecked
Allow SIP P-Asserted...	Unchecked
Send Diversion Header	Select 'on redirect'
Obtain DID/DNIS number...	Select 'SIP TO: header field'
Use < > in Request URI of outbound calls	Select 'dialed number'



b. (Optional) Route DID to specific locations. Navigate to **Phone System** > **Outside Lines** > **New SIP Proxy**. Locate the Call Route section. Select the **Routed using DID Block**: option, and then select the DID block created earlier.

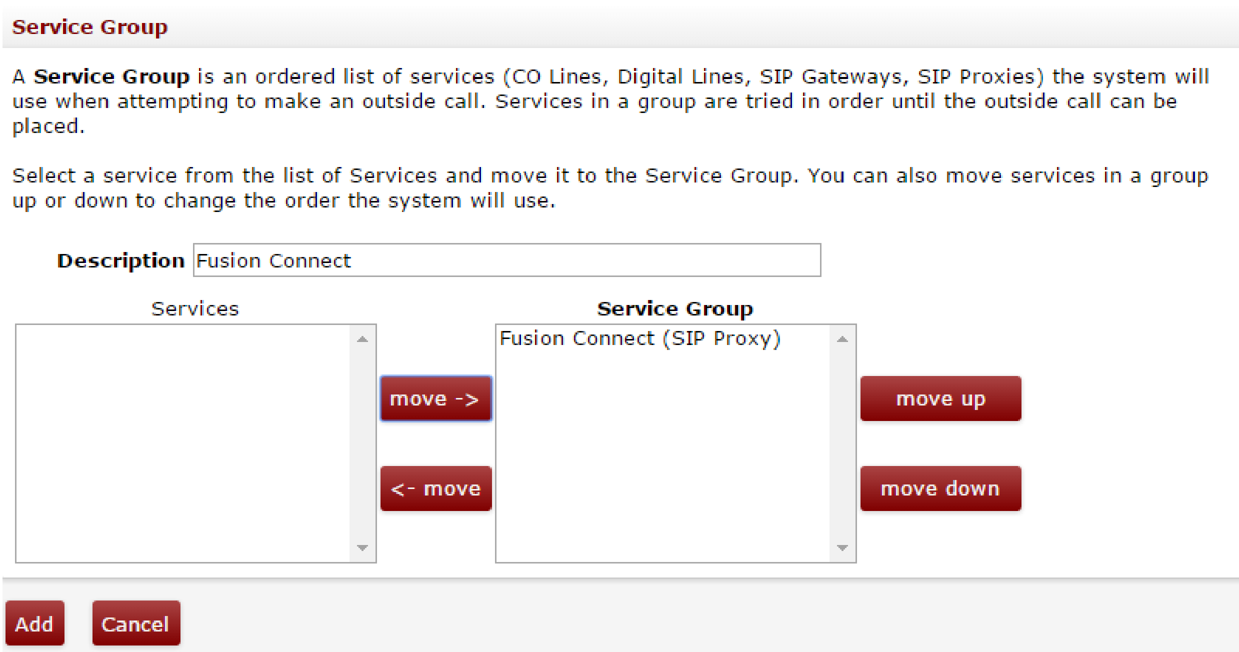
4. Setup the Allworx VoIP Server parameters. Navigate to Servers > VoIP. Click Modify to change any of the settings.

Field	Recommended Setting
BLF Port	Leave as default 2088
Secure BLF	Unchecked
Force Remote Phone audio through server	Checked.
Plug and Play Secret Key	6 to 20 characters use 0-9, and #
Phone Administration Password	0 to 6 characters, use alphanumeric and #
Global SIP Connection Limit	Set to maximum number of concurrent calls allowed plus the number of remote handsets
Paging Base IP address	Use the default setting of 239.255.10.0.
Paging Port	Use the default setting of 56586.
Paging Maximum Hop Count	Typically us the default setting of 1.
Paging Maximum Duration	Set between 1 and 30 minutes
RTP Base Port	User/Fusion specified. By default 15000, some providers require a specific starting port such as 16384.
RTP DTMF Payload	Set to 101
RTP DSCP Tag	Select 'Expedited Forwarding (EF)'
SIP DSCP Tag	Select 'Assured Forwarding 41 (AF41)'
Disable Phone Creates via LAN Plug and Play	Typically Unchecked but once all phones have been added to the system for security purposes can be Checked.
Disable Phone Creates via WAN Plug and Play	Typically Unchecked but once all remote phones have been added to the system for security purposes can be Checked.
Disable Assign User at Phone	Typically Unchecked but once all remote phones have been added to the system for security purposes can be Checked.
Enable PCP Proxy	Typically enabled, allows PCP between PC and Phones on different VLANs. Refer to Admin Guide.

VoIP Server modify

	Current Value
BLF Port	2088
BLF Secure	disabled
Force Remote Phone audio through server	enabled
Plug and Play Secret Key	****** show
Phone Administration Password	****** show
Global SIP Connection Limit	2
Paging Base IP Addr	239.255.10.0
Paging Port	56586
Paging Max Hop Count	1
Paging Maximum Duration (minutes)	1
RTP Base Port	15000
RTP DTMF Payload	101
RTP DSCP Tag	Expedited Forwarding (EF)
SIP DSCP Tag	Assured Forwarding 41 (AF41)
Phone Creates via LAN Plug and Play	disabled
Phone Creates via WAN (Remote Phone) Plug and Play	disabled
Assign User at Phone	disabled
PCP Proxy	enabled

5. Configure the Dial Plan. Navigate to **Phone System** > **Dial Plan**.
    a. Create a service group for the SIP trunk. Locate the Service Groups section and click **Add New Service Group**. Select the Fusion SIP trunk and click **Add**.



    b. Modify the existing rules and set the Service Group to the newly created custom service group.

▾ External Dialing Rules

North American Numbering Plan Administration (NANPA)	enabled	Modify
Home Area Code		Modify

Automatic Route Selection add new rule

Number Dialed	Output Dial String	Service Group	Action
9+1nnnnnnnnnn	1nnnnnnnnnn	Fusion Connect	Modify

n - number (0-9)

Emergency

Type	Number Dialed	Service Group	Action
Emergency	9+911 911	see Dialing Privileges Group for source of call	Modify

Emergency Call Email Notifications are not enabled. Modify

Services

Type	Number Dialed	Service Group	Action
Phone Services (211,311,411,511,611,711,811)	9+n11	Fusion Connect	Modify
Operator	9+0	Fusion Connect	
Long Distance Services	9+1010...	Fusion Connect	
International Calls	9+011...	Fusion Connect	
Public SIP Directory	8+nnnnnnnnnnn (11 digits)	No Devices	
PIN Code	78+nnnnn (5 digits)	No Devices	
Outside Line Seizure	9#	No Devices	

SUPPORT

Allworx
Allworx Technical Support:
1-866-Allworx (255-9679)
Monday - Friday 8:00 am to 8:00 pm EST
support@allworx.com

Fusion
Customer Support Email: customersupport@fusionconnect.com
SIP Trunking Customer Support: 888.849.9608
Technical Support Email: sipsupport@fusionconnect.com
SIP Trunking Technical Support: 888.849.9608 (Press 3 for trunk turn up, Press 4 for Support)