Introduction

This guide assistsusers to configure the Allworx VoIP Phone System and Ixica 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 Ixica SIP services (e.g., DHCP settings and installed the latest software version).
- Ordered Ixica services and received the associated configuration information for Ixica SIP Trunking.

Important Notes

Emergency 911/E911 Services:

This configuration was tested with Allworx server software 8.4 and 8.5, on Server platforms for Connect Series and X-series.

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

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 Ixica SIP Trunkingwith the following Network Layout (Figure 1) and Network Configuration (Figure 2).

Figure 1

Ixica SIP Trunking

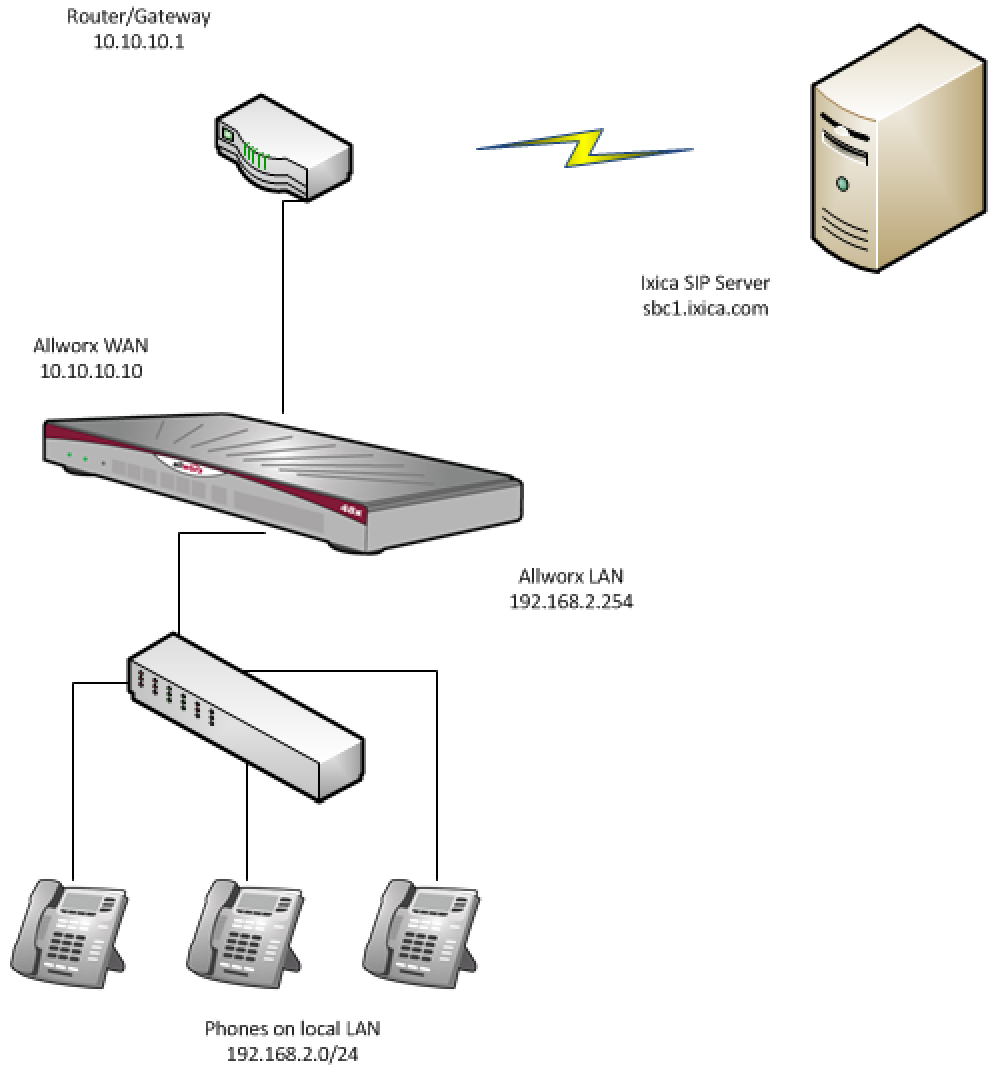
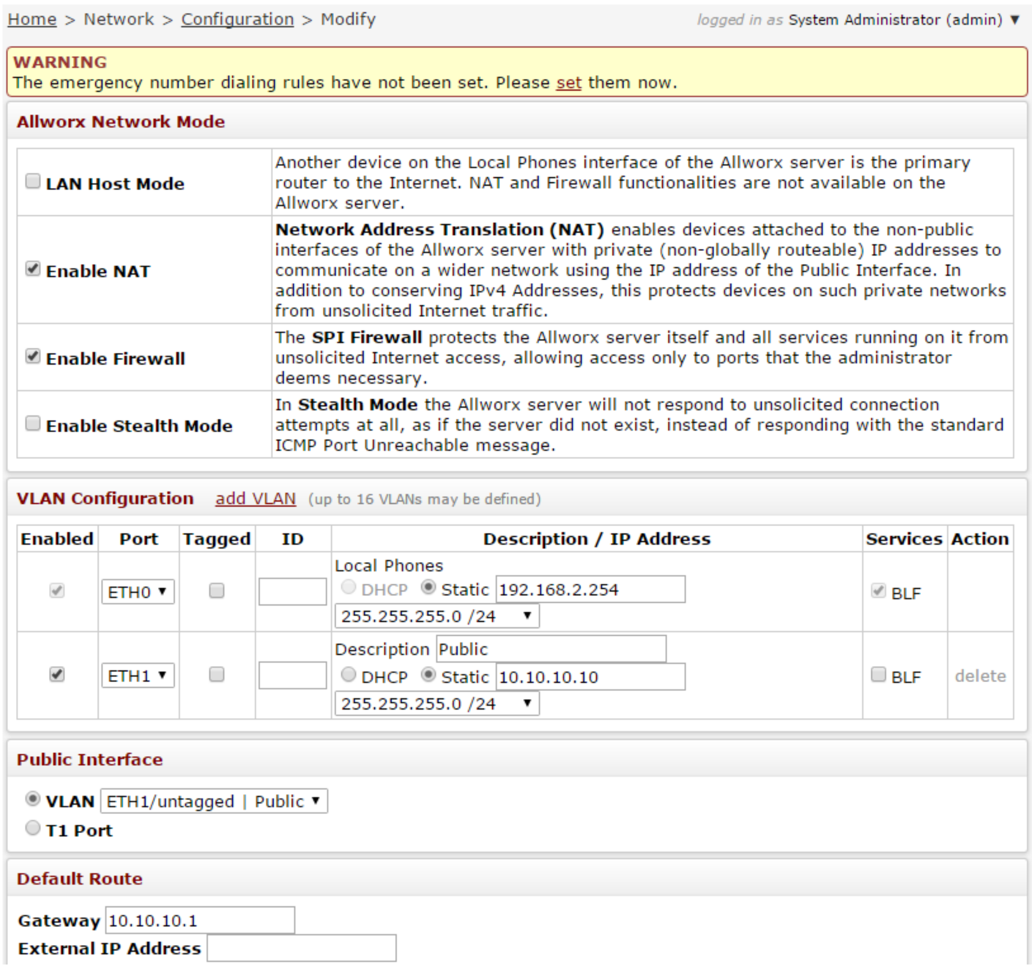
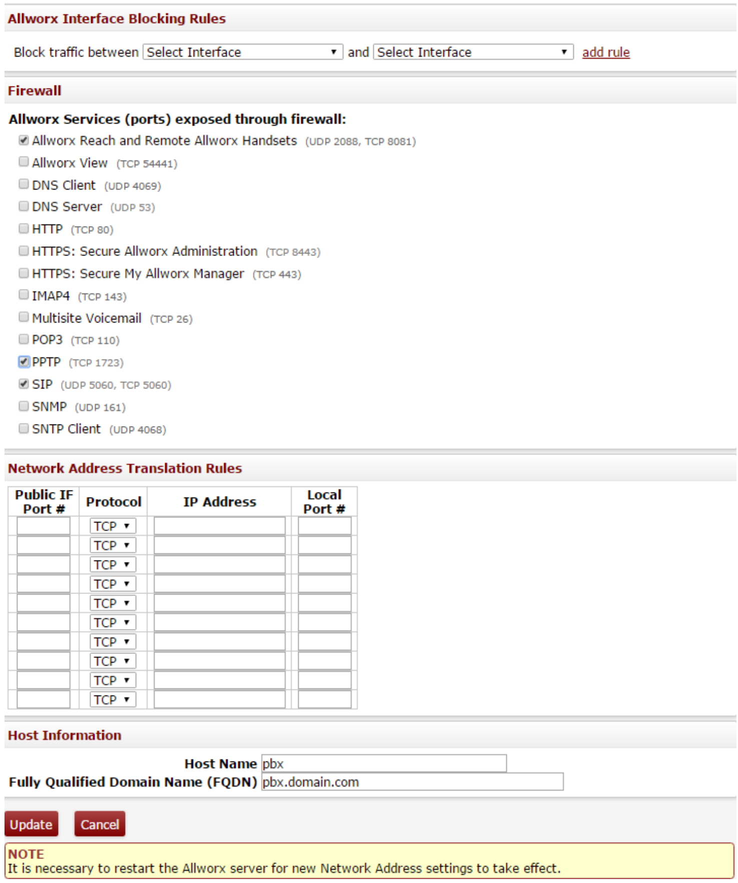


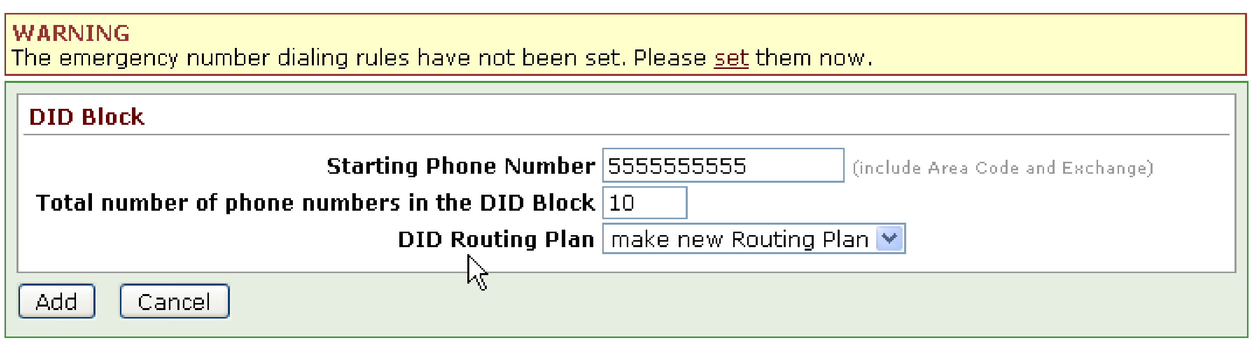
Figure 2


Home > Network > Configuration > Modify
logged in as System Administrator (admin) ▼
WARNING
The emergency number dialing rules have not been set. Please set them now.
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
255.255.255.0 /24
BLF
delete
Public Interface
VLAN
ETH1/untagged | Public
T1 Port
Default Route
Gateway
10.10.10.1
External IP Address

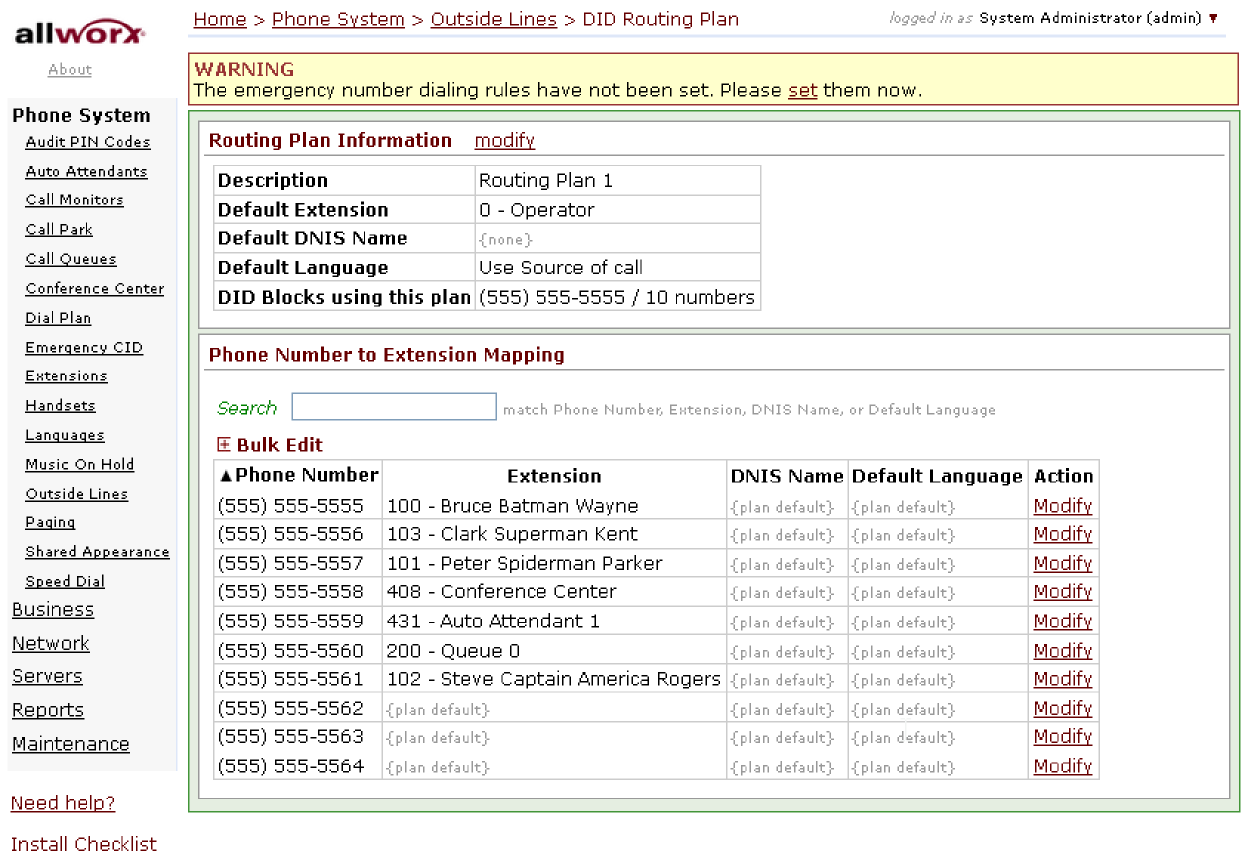


2. (Optional) Setup the DID Block and DID Routing Plan to use with the SIP Proxy. The cut-sheet received from Ixicaprovides the available numbers.

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.



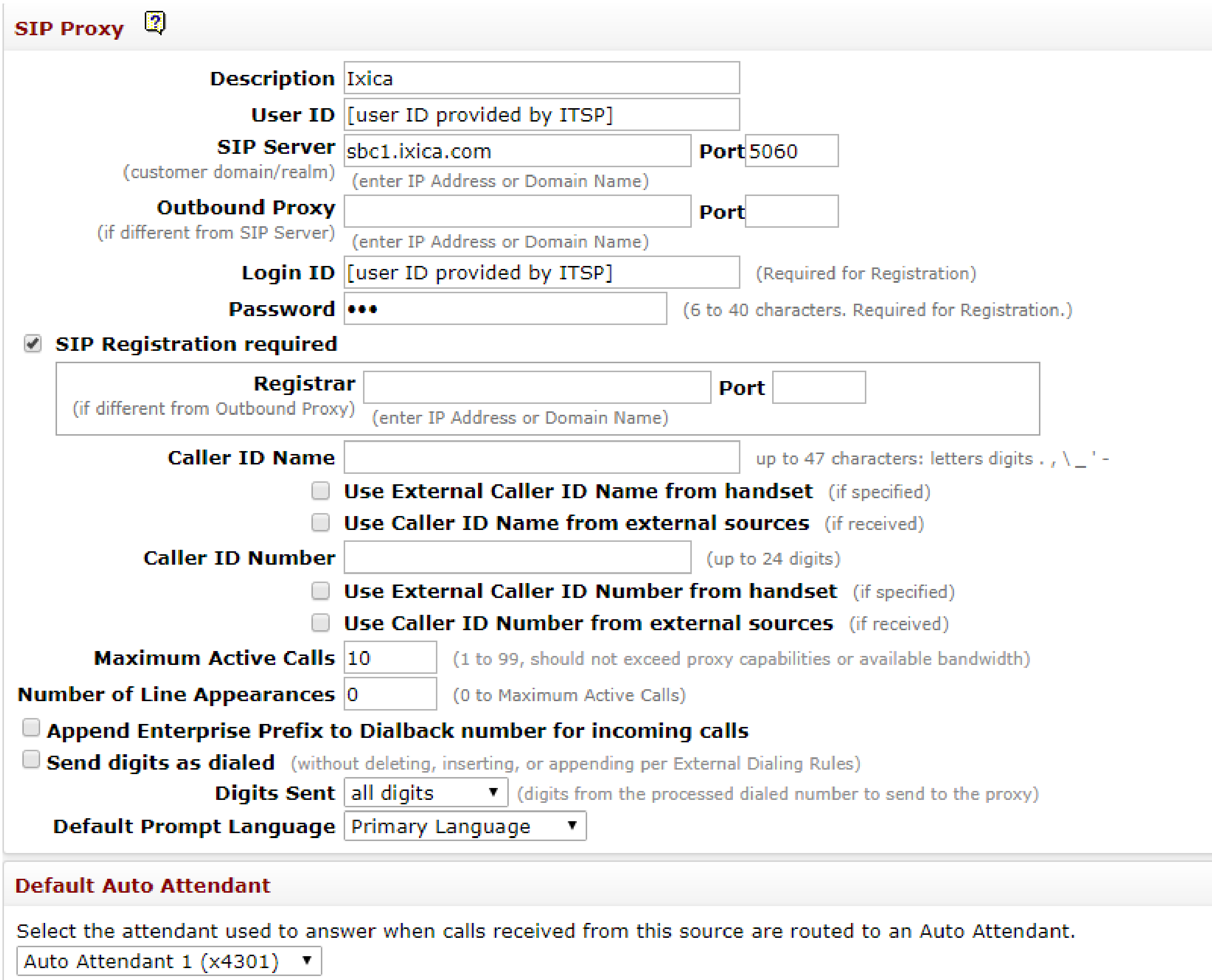
3. Configure the SIP Proxy.

a. Navigate to **Phone System**>**Outside Lines**>**SIP Proxies**>**add new SIP Proxy**.

Field	Recommend Setting
SIP PROXY	

Ixica SIP Trunking

Description	User assigned label such as, Ixica SIP.
User ID	Provided by Ixica.
SIP Server	Provided by Ixica
SIP Server Port	Default value is 5060.
Outbound Proxy	Leave Blank.
Outbound Proxy Port	Leave Blank.
Login ID	Provided by Ixica.
Password	Provided by Ixica.
SIP Registration	Checked
Registrar	Leave Blank.
Registrar Port	Leave Blank.
Caller ID Name	Leave Blank.
Caller ID Number	Leave Blank.
Maximum Active Calls	Provided by Ixica.
Number of Line Appearances	Default value of 0.
Append Enterprise prefix…	Leave Blank
Send Digits as dialed	Unchecked (dialing rules will be applied as configured).
Digits Sent	Default setting of ‘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	Ixica	
User ID	[user ID provided by ITSP]	
SIP Server (customer domain/realm)	sbc1.ixica.com — Port 5060	(enter IP Address or Domain Name)
Outbound Proxy (if different from SIP Server)	Port	(enter IP Address or Domain Name)
Login ID	[user ID provided by ITSP]	(Required for Registration)
Password	•••	(6 to 40 characters. Required for Registration.)

☑ SIP Registration required

Registrar (if different from Outbound Proxy)	Port	(enter IP Address or Domain Name)
Caller ID Name		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		(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	10	(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 processed dialed number to send to the proxy)
Default Prompt Language	Primary Language	

Default Auto Attendant

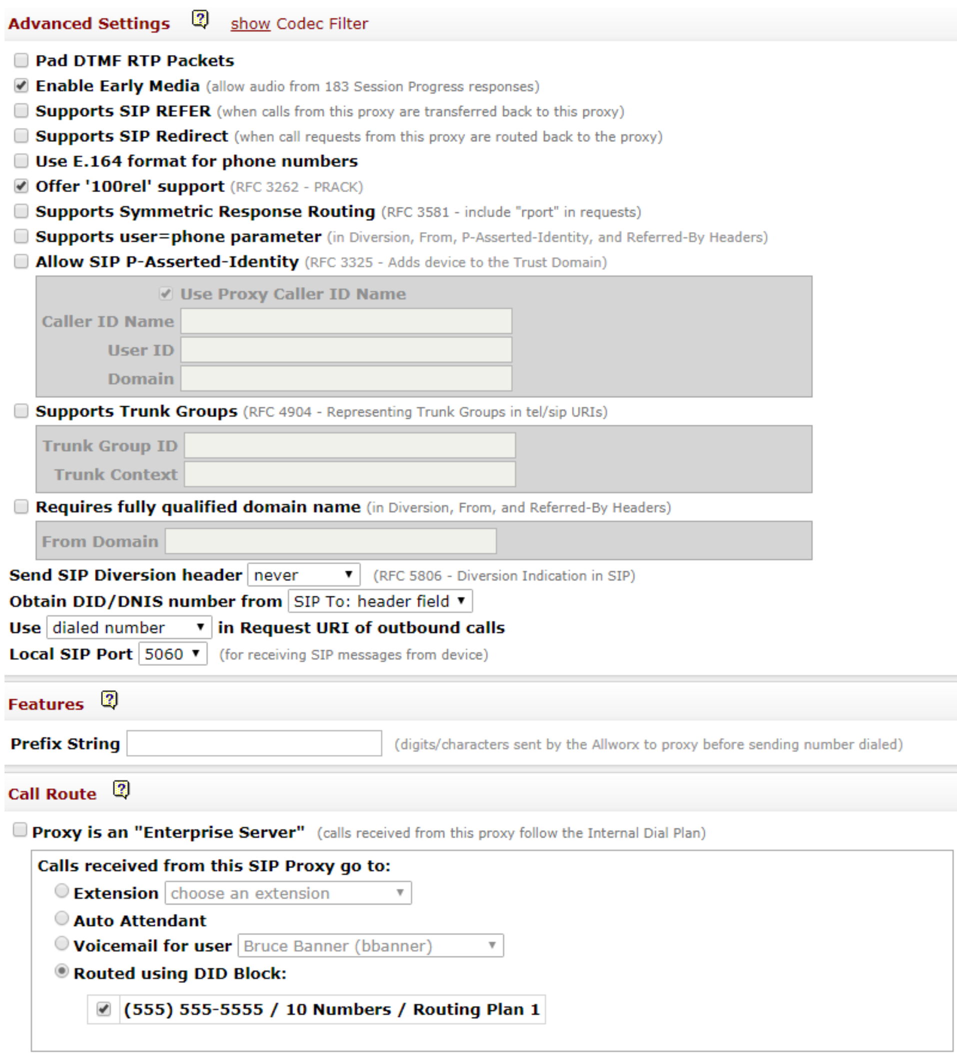
Select the attendant used to answer when calls received from this source are routed to an Auto Attendant.

Auto Attendant 1 (x4301)

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	Checked
Supports Symmetric…	Unchecked
Supports user=…	Unchecked
Allow SIP P-Asserted…	Unchecked

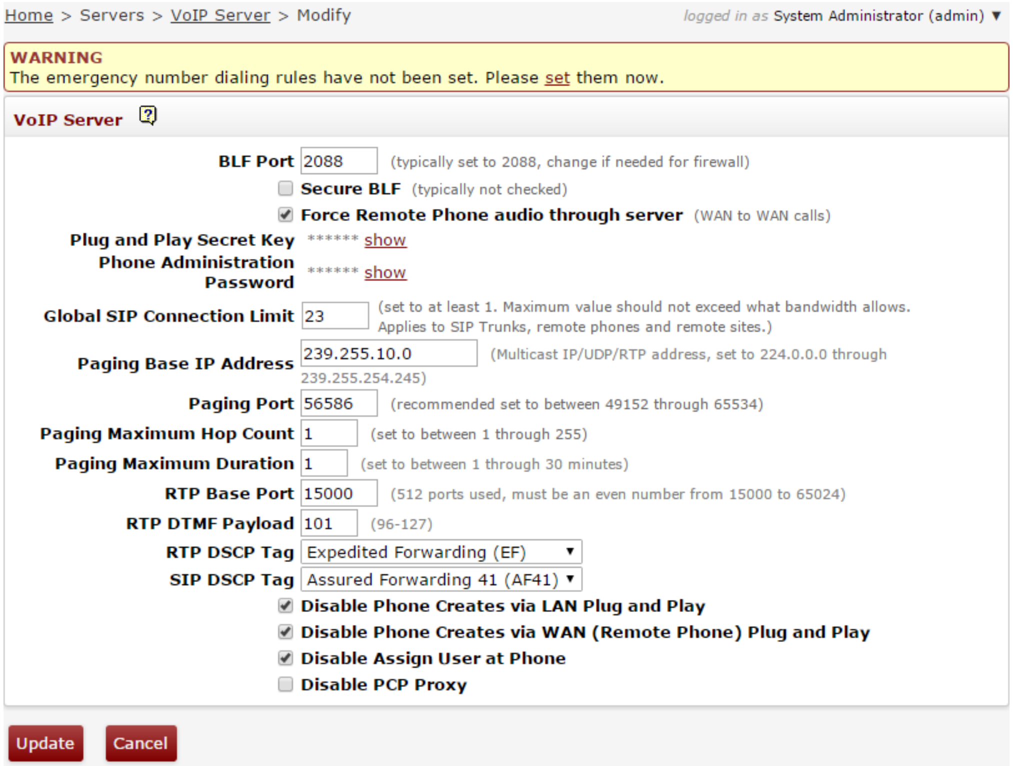
Supports Trunk…	Unchecked
Requires fully…	Unchecked
Send Diversion Header	Select 'Never'
Obtain DID/DNIS number…	Select 'SIP To…'
Use <> in Request URI of outbound calls	Select 'dialed number'



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.

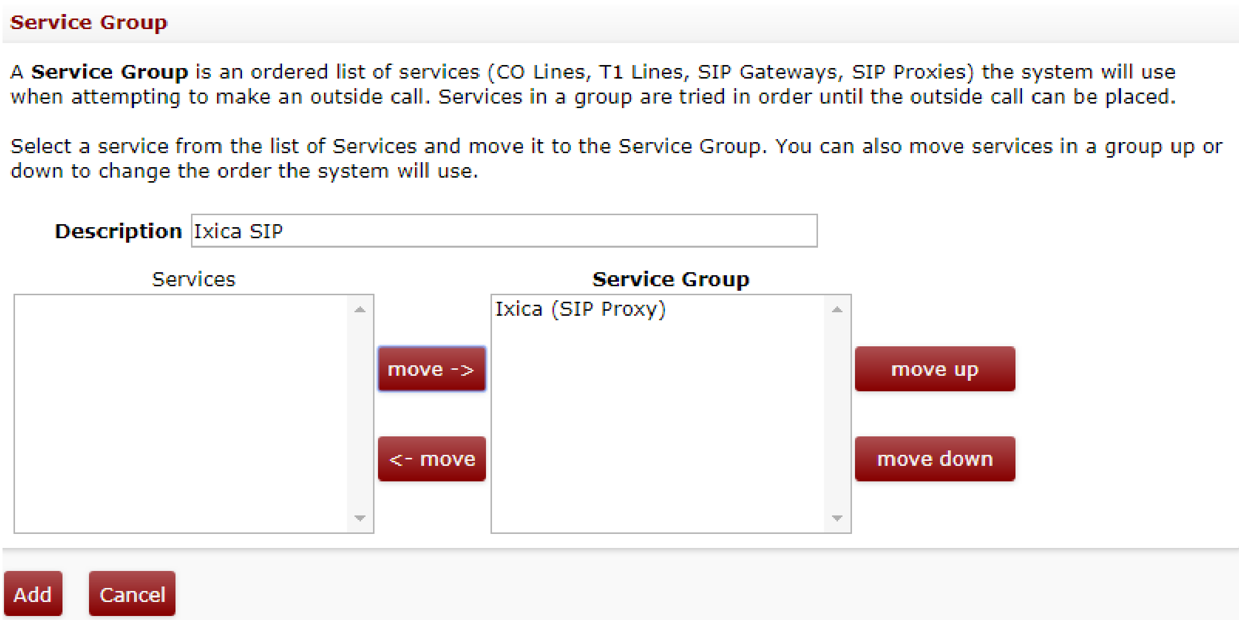
4. Setup the Allworx VoIP Server parameters. Navigate to **Servers**>**VoIP**. Click **modify** to change any of the settings.

Field	Recommend 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 use the default setting of 1.
Paging Maximum Duration	Set between 1 and 30 minutes
RTP Base Port	Default value of 15000
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.
Disable PCP Proxy	Typically Unchecked (used to proxy some data from Data and Voice Vlans for Interact)


Home > Servers > VoIP Server > Modify
logged in as System Administrator (admin)
WARNING
The emergency number dialing rules have not been set. Please set them now.
VoIP Server
BLF Port 2088 (typically set to 2088, change if needed for firewall)
Secure BLF (typically not checked)
Force Remote Phone audio through server (WAN to WAN calls)
Plug and Play Secret Key ****** show
Phone Administration Password ****** show
Global SIP Connection Limit 23 (set to at least 1. Maximum value should not exceed what bandwidth allows. Applies to SIP Trunks, remote phones and remote sites.)
Paging Base IP Address 239.255.10.0 (Multicast IP/UDP/RTP address, set to 224.0.0.0 through 239.255.254.245)
Paging Port 56586 (recommended set to between 49152 through 65534)
Paging Maximum Hop Count 1 (set to between 1 through 255)
Paging Maximum Duration 1 (set to between 1 through 30 minutes)
RTP Base Port 15000 (512 ports used, must be an even number from 15000 to 65024)
RTP DTMF Payload 101 (96-127)
RTP DSCP Tag Expedited Forwarding (EF)
SIP DSCP Tag Assured Forwarding 41 (AF41)
Disable Phone Creates via LAN Plug and Play
Disable Phone Creates via WAN (Remote Phone) Plug and Play
Disable Assign User at Phone
Disable PCP Proxy
Update
Cancel

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 theIxica 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	Ixica SIP	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>	Ixica SIP	Modify
Operator	9<0>	Ixica SIP	
Long Distance Services	9<1010...>	Ixica SIP	
International Calls	9<011...>	Ixica SIP	
Public SIP Directory	8<nnnn> (4 digits)	No Devices	
PIN Code	78<nnnnn> (5 digits)	No Devices	
Outside Line Seizure	9#	No Devices	

NOTE: If you do not intend to make international calls, it is highly recommended setting the Service Group for International Calls to ‘No Devices’. If International Calls are required it is recommended to create a custom Dial Plan that only allows calls to the required country codes. International fraud is a major issue; these configuration steps add another level of security to protect you from this costly exploit.

Support

Allworx

Allworx Technical Support:
1-866-Allworx (255-9679)
Monday - Friday 8:00 am to 8:00 pm EST
support@allworx.com
Toll Free 1-800-ALLWORX * 585-421-3850
wwww.allworx.com
Revised: 5/15/20

Ixica