

Introduction

This guide assists users 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.

Ixica SIP Trunking

- 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 Trunking with the following Network Layout (Figure 1) and Network Configuration (Figure 2).

Figure 1

Ixica SIP Trunking

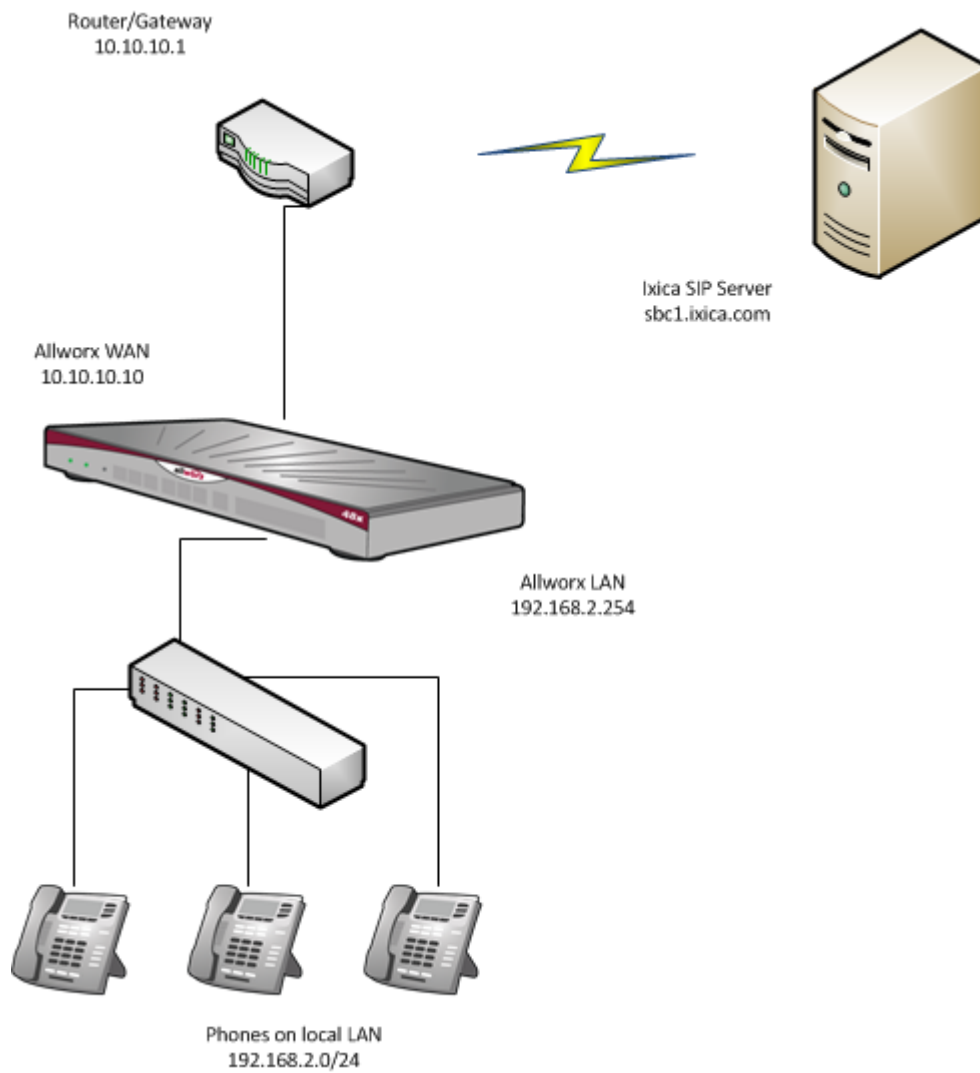


Figure 2

Ixica SIP Trunking

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

<input type="checkbox"/> 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.
<input checked="" type="checkbox"/> 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.
<input checked="" type="checkbox"/> 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.
<input type="checkbox"/> 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
<input checked="" type="checkbox"/>	ETH0 ▼	<input type="checkbox"/>		Local Phones <input type="radio"/> DHCP <input checked="" type="radio"/> Static 192.168.2.254 255.255.255.0 /24 ▼	<input checked="" type="checkbox"/> BLF	
<input checked="" type="checkbox"/>	ETH1 ▼	<input type="checkbox"/>		Description Public <input type="radio"/> DHCP <input checked="" type="radio"/> Static 10.10.10.10 255.255.255.0 /24 ▼	<input type="checkbox"/> BLF	delete

Public Interface

☒ VLAN ETH1/untagged | Public ▼
☐ T1 Port

Default Route

Gateway 10.10.10.1
External IP Address

Ixica SIP Trunking

Allworx Interface Blocking Rules

Block traffic between and [add rule](#)

Firewall

Allworx Services (ports) exposed through firewall:

- ☒ Allworx Reach and Remote Allworx Handsets (UDP 2088, TCP 8081)
- ☐ Allworx View (TCP 54441)
- ☐ DNS Client (UDP 4069)
- ☐ DNS Server (UDP 53)
- ☐ HTTP (TCP 80)
- ☐ HTTPS: Secure Allworx Administration (TCP 8443)
- ☐ HTTPS: Secure My Allworx Manager (TCP 443)
- ☐ IMAP4 (TCP 143)
- ☐ Multisite Voicemail (TCP 25)
- ☐ POP3 (TCP 110)
- ☒ PPTP (TCP 1723)
- ☒ SIP (UDP 5060, TCP 5060)
- ☐ SNMP (UDP 161)
- ☐ SMTP Client (UDP 4068)

Network Address Translation Rules

Public IF Port #	Protocol	IP Address	Local Port #
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		
	TCP ▼		

Host Information

Host Name

Fully Qualified Domain Name (FQDN)

NOTE
It is necessary to restart the Allworx server for new Network Address settings to take effect.

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

WARNING
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DID Block

Starting Phone Number (include Area Code and Exchange)

Total number of phone numbers in the DID Block

DID Routing Plan ▼

Ixica SIP Trunking

- 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.

allworx Home > Phone System > Outside Lines > DID Routing Plan logged in as System Administrator (admin) ▼

About

Phone System

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WARNING
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Routing Plan Information [modify](#)

Description	Routing Plan 1
Default Extension	0 - Operator
Default DNIS Name	{none}
Default Language	Use Source of call
DID Blocks using this plan	(555) 555-5555 / 10 numbers

Phone Number to Extension Mapping

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

[Bulk Edit](#)

▲ Phone Number	Extension	DNIS Name	Default Language	Action
(555) 555-5555	100 - Bruce Batman Wayne	{plan default}	{plan default}	Modify
(555) 555-5556	103 - Clark Superman Kent	{plan default}	{plan default}	Modify
(555) 555-5557	101 - Peter Spiderman Parker	{plan default}	{plan default}	Modify
(555) 555-5558	408 - Conference Center	{plan default}	{plan default}	Modify
(555) 555-5559	431 - Auto Attendant 1	{plan default}	{plan default}	Modify
(555) 555-5560	200 - Queue 0	{plan default}	{plan default}	Modify
(555) 555-5561	102 - Steve Captain America Rogers	{plan default}	{plan default}	Modify
(555) 555-5562	{plan default}	{plan default}	{plan default}	Modify
(555) 555-5563	{plan default}	{plan default}	{plan default}	Modify
(555) 555-5564	{plan default}	{plan default}	{plan default}	Modify


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.

Ixica SIP Trunking

SIP Proxy 

Description

User ID

SIP Server **Port**
(customer domain/realm) (enter IP Address or Domain Name)

Outbound Proxy **Port**
(if different from SIP Server) (enter IP Address or Domain Name)

Login ID (Required for Registration)

Password (6 to 40 characters. Required for Registration.)

☒ **SIP Registration required**

Registrar **Port**
(if different from Outbound Proxy) (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 (1 to 99, should not exceed proxy capabilities or available bandwidth)

Number of Line Appearances (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 (digits from the processed dialed number to send to the proxy)

Default Prompt Language

Default Auto Attendant


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

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

Ixica SIP Trunking

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'





Advanced Settings  [show Codec Filter](#)


☐ **Pad DTMF RTP Packets**
☒ **Enable Early Media** (allow audio from 183 Session Progress responses)
☐ **Supports SIP REFER** (when calls from this proxy are transferred back to this proxy)
☐ **Supports SIP Redirect** (when call requests from this proxy are routed back to the proxy)
☐ **Use E.164 format for phone numbers**
☒ **Offer '100rel' support** (RFC 3262 - PRACK)
☐ **Supports Symmetric Response Routing** (RFC 3581 - include "rport" in requests)
☐ **Supports user=phone parameter** (in Diversion, From, P-Asserted-Identity, and Referred-By Headers)
☐ **Allow SIP P-Asserted-Identity** (RFC 3325 - Adds device to the Trust Domain)

☒ **Use Proxy Caller ID Name**
 Caller ID Name
 User ID
 Domain


☐ **Supports Trunk Groups** (RFC 4904 - Representing Trunk Groups in tel/sip URIs)
 Trunk Group ID
 Trunk Context

☐ **Requires fully qualified domain name** (in Diversion, From, and Referred-By Headers)
 From Domain



Send SIP Diversion header never  (RFC 5806 - Diversion Indication in SIP)
 Obtain DID/DNIS number from SIP To: header field 
 Use dialed number  in Request URI of outbound calls
 Local SIP Port 5060  (for receiving SIP messages from device)

Features 

Prefix String (digits/characters sent by the Allworx to proxy before sending number dialed)

Call Route 

☐ **Proxy is an "Enterprise Server"** (calls received from this proxy follow the Internal Dial Plan)

Calls received from this SIP Proxy go to:
☐ **Extension** choose an extension 
☐ **Auto Attendant**
☐ **Voicemail for user** Bruce Banner (bbanner) 
☒ **Routed using DID Block:**
☒ (555) 555-5555 / 10 Numbers / Routing Plan 1

- 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.

Ixica SIP Trunking

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 V lans for Interact)

Ixica SIP Trunking

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 (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 (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 (Multicast IP/UDP/RTP address, set to 224.0.0.0 through 239.255.254.245)

Paging Port (recommended set to between 49152 through 65534)

Paging Maximum Hop Count (set to between 1 through 255)

Paging Maximum Duration (set to between 1 through 30 minutes)

RTP Base Port (512 ports used, must be an even number from 15000 to 65024)

RTP DTMF Payload (96-127)

RTP DSCP Tag ▼

SIP DSCP Tag ▼

☒ **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**

Ixica SIP Trunking

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 Ixica SIP trunk and click **Add**.

Service Group

A **Service Group** is an ordered list of services (CO Lines, T1 Lines, SIP Gateways, SIP Proxies) the system will use when attempting to make an outside call. Services in a group are tried in order until the outside call can be placed.

Select a service from the list of Services and move it to the Service Group. You can also move services in a group up or down to change the order the system will use.

Description

Services		Service Group	
	<div>move -></div> <div><- move</div>	<div>Ixica (SIP Proxy)</div>	<div>move up</div> <div>move down</div>

Add

Cancel

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

Ixica SIP Trunking

▼ 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

www.allworx.com

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