

Allworx / Clearly Setup Application Notes

Version 1.1

March 2013

- 1** Phone system
- 2** Network server
- 3** Message center

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Introduction

This document provides setup instructions to authorized Allworx Resellers for configuring Clearly services on Allworx servers.

Prerequisites

It is assumed that the Reseller has:

- Completed Allworx Technical training and their main technician is certified as either *Allworx Certified Administrator (ACA)* or *Allworx Certified Professional (ACP)*
- Set up all other functions within the Allworx system prior to connecting Clearly services (e.g. DHCP settings, installed latest software version)
- Ordered Clearly services and received the associated configuration information for SIP Trunking

Key Notes

- 1 A. The Allworx server must be running software release 7.4.12.5 or higher. To obtain the latest software, visit the Allworx Authorized Reseller Portal (www.allworxportal.com).
- 2 B. Allworx Customer Support has verified the interoperability of Allworx and Clearly under controlled conditions. The following items were tested:
 - RTP frame rate of 20 for codecs G.711 and G.729a
 - Support of Early Media -183 Session Progress messages
 - Call Hold & Retrieve functionality
 - Call Transfer methods with and without REFER support
 - SIP Diversion and Redirect
 - Direct Inward Dial (DID)
 - Handling of E.164 format for phone numbers
 - Performance of the service with remote Allworx phones
- 3

Allworx testing was successful during the time period in which the testing took place. Subsequent changes to the provider's network or differences in local internet connectivity could alter the performance of the product in the field. It is the Reseller's responsibility to properly place and route to the Allworx server on the local network.

C. Limitations

The following limitations were observed during Allworx testing:

- The Caller ID Name (if populated in the SIP Proxy settings page) will be overridden by the value set in the trunk by Clearly
- Calls not established when Caller ID number (other than the registered DID number) is populated in the SIP Proxy settings page

Setting up the Allworx System:

The following steps must be performed on the Allworx server:

1. Perform steps 1 through 16 of the Allworx Install Checklist including updating the Allworx server software to the latest release (7.4.12.5 or higher).
2. Configure the SIP proxy connection.
3. Configure the VoIP server settings.
4. Create a Dial Plan Service Group for Clearly.
5. Configure Dialing Rules for routing calls through Clearly.
6. If using DID numbers, configure a Routing Plan for the DID numbers.
7. Reboot all phones to download new settings.
8. Verify the connection and its usability.

- #1.** Prior to setting up the connection to Clearly, other basic PBX settings must be configured. Complete steps 1 through 16 of the Allworx Install Checklist including loading software release 7.4.12.5 or higher, if the server is running an earlier version. The checklist is located within the Allworx System Administration Tool.

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The screenshot shows the Allworx System Administration Tool interface. On the left sidebar, the 'Install Checklist' link is highlighted with a red circle and a line pointing to a pop-up window. The pop-up window, titled 'Allworx - Install Checklist - Windows Inte...', displays a list of 8 steps for configuration. The steps are as follows:

Step	Description	Link
1	Set time on server.	Maintenance / Time
2	Program Digital Lines: If connected to T1 interface(s), configure the system to match the settings obtained from the service provider.	Network / Digital Lines
3	Program Network configuration: set the Network Mode, LAN and WAN interface settings, Gateway, server Host Name and Domain Name, and Firewall settings	Network / Configuration
4	Enable/Disable DHCP server.	Servers / DHCP
5	Set DNS server addresses.	Servers / DNS
6	Optional: Configure Port Expanders	Network / Port Expanders
7	Enable VPN, if required.	Network / VPN
8	Restart server for...	

The background interface shows a 'WARNING' message: 'The emergency number dialing rules have not been set. Please [set](#) them now.' The main content area is divided into four sections: PHONE SYSTEM, NETWORK, BUSINESS, and SERVERS, each with a list of configuration options.

#2. To set up the SIP proxy, log into the Allworx System Administration Tool and go to Phone System > Outside Lines > SIP Proxies. Select “Add New SIP Proxy”.

allworx

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WARNING
The emergency number dialing rules have not been set. Please [set](#) them now.

PHONE SYSTEM

- Audit PIN Codes
- Auto Attendants
- Call Monitors
- Call Park
- Call Queues
- Conference Center
- Dial Plan
- Emergency CID
- Extensions
- Handsets
- Languages
- Music On Hold
- Outside Lines**
- Paging
- Speed Dial

NETWORK

- Configuration
- Digital Lines
- Multi-Site
- Port Expanders
- Static Routes
- VPN

REPORTS

- Auto Notification
- Call Details
- Configuration
- Digital Lines
- Live Calls
- Network Statistics
- System Events
- Users

BUSINESS

- Contact Information
- Message Aliases
- Schedules
- Users

SERVERS

- DHCP
- DNS
- Email
- VoIP
- Web

MAINTENANCE

- Backup
- Feature Keys
- Import / Export
- Restart
- Time
- Tools
- Update

The New SIP Proxy page is displayed:

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)

☒ **SIP Registration required**

Login ID

Password (maximum 40 characters)

Registrar Port
(if different from Outbound Proxy) (enter IP Address or Domain Name)

Maximum Active Calls (1 to 99, should not exceed proxy capabilities or available bandwidth)

Number of Line Appearances (0 to Maximum Active Calls)

☐ **Send digits as dialed** (without prepending 1 and/or area code)

Digits Sent ☒ (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 1 (x431) ▼

Call Route ⓘ

☐ **Proxy is an "Enterprise Server"** (calls received from this proxy follow the server's internal dial plan)

Calls received from this SIP Proxy go to:

☐ Extension ▼

☒ **Auto Attendant**

☐ Voicemail for user ▼

☐ Routed using DID Block(s): **No DID Blocks have been defined!**

SIP Registration

Configure the settings as listed in the table below then select "Add".

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)

☒ **SIP Registration required**

Login ID

Password (maximum 40 characters)

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

Maximum Active Calls (1 to 99, should not exceed proxy capabilities or available bandwidth)

Number of Line Appearances (0 to Maximum Active Calls)

☐ **Send digits as dialed** (without prepending 1 and/or area code)

Digits Sent (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.

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Description	Recommended Setting
Description	User-assigned label. e.g. "Clearly" (without the quotes)
User ID	Provided by Clearly.
SIP Server	Provided by Clearly.
SIP Server Port	Provided by Clearly. Default value is 5060.
Outbound Proxy	Provided by Clearly
Outbound Proxy Port	Leave Blank
SIP Registration Required	Checked
Login ID	Obtained from Clearly
Password	Obtained from Clearly
Registrar	Leave Blank
Registrar Port	Leave Blank
Maximum Active Calls	Enter the number of SIP trunks purchased from Clearly
Number of Line Appearances	Should be set to a value no greater than the number of sip trunks purchased
Send Digits as dialed	<u>NOT</u> checked
Digits Sent	Select all digits
Default Auto Attendant	This is a customer-specific setting and defines which automated attendant is to be played for each incoming call that ends up at the AA.
Proxy is an Enterprise Server	<u>NOT</u> checked
Calls from this SIP Proxy go to:	Default value is Auto Attendant. Change to suit customer's environment.

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After creating the SIP proxy, the Outside Lines page will be displayed. There are additional settings to configure on the Modify screen. Select “Modify SIP Proxy”. The Modify screen which now includes Advanced Settings is displayed.

Advanced Settings ?

- ☐ Pad DTMF RTP Packets
- ☒ **Enable Early Media** (allow audio from 183 Session Progress responses)
- ☐ **Supports Symmetric Response Routing** (RFC 3581 - include "rport" in requests)
- ☐ **Use SIP Diversion for deflected calls** (draft-levy-sip-diverison-08.txt)
- ☐ **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)

Obtain DID/DNIS number from **SIP To: header field**

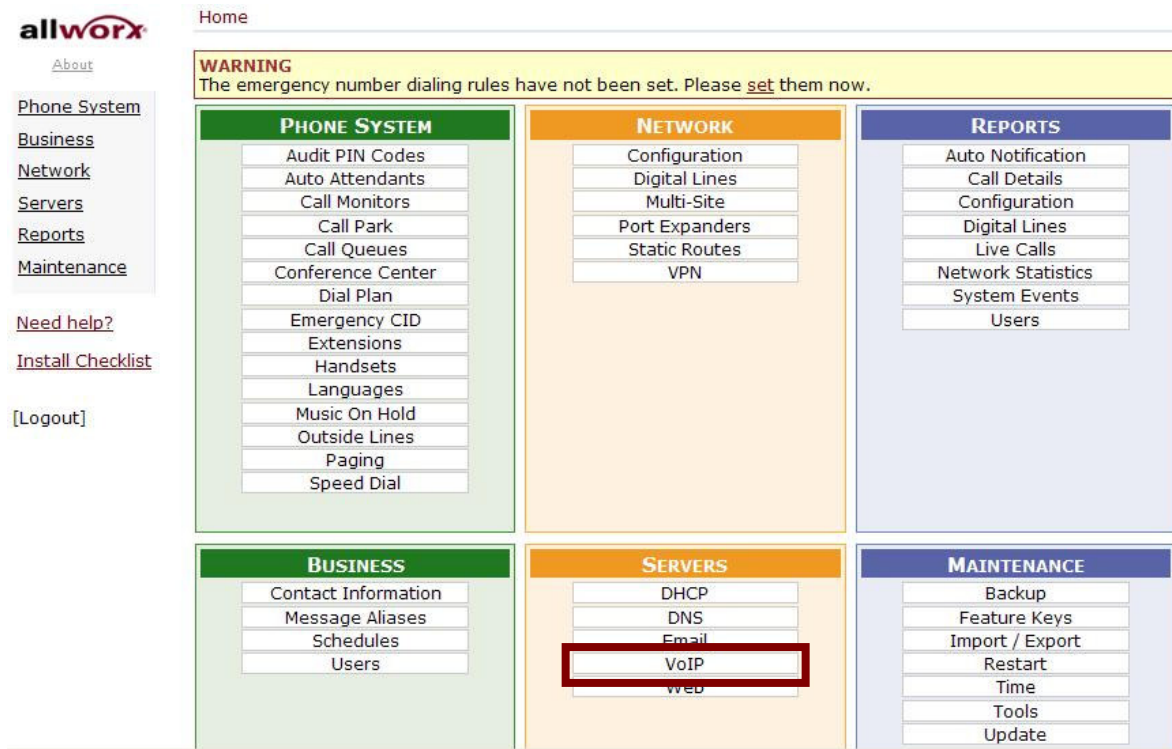
Use **dialed number** in Request URI of outbound calls

Configure the Advanced Settings as listed in the table below then select “Update”

Description	Recommended Setting
Pad DTMF RTP Packets	<u>NOT</u> checked
Enable Early Media	Checked
Supports Symmetric Response Routing	<u>NOT</u> checked
Use SIP Diversion for deflected calls	<u>NOT</u> checked
Supports SIP REFER	<u>NOT</u> checked
Supports SIP Redirect	<u>NOT</u> checked
Use E.164 format for phone numbers	<u>NOT</u> checked
Offer '100rel' support	Checked Note: This setting is available in Release 6.8 or higher
Obtain DID/DNIS number from:	SIP Request URI
Use in Request URI of outbound calls	Dialed number

Note: The appropriate advanced features can be checked according to features required and supported.

#3. To set up the Allworx VoIP Server, go to Servers > VoIP and select “Modify”.



The Modify screen is displayed.

The screenshot shows the 'VoIP Server' configuration screen. The title is 'VoIP Server' with a help icon. The configuration options are as follows:

- 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:** 8 (set to at least 1, for SIP Trunks, remote phones, remote sites as bandwidth allows)
- 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) (dropdown menu)
- SIP DSCP Tag:** Assured Forwarding 41 (AF41) (dropdown menu)
- ☐ **Disable Phone Creates via LAN Plug and Play**
- ☐ **Disable Phone Creates via WAN (Remote Phone) Plug and Play**
- ☐ **Disable Assign User at Phone**

At the bottom are three buttons: Update, Start Over, and Cancel.

Configure the settings as listed in the table below then select "Update".

Description	Recommended Setting
BLF Port	Typically set to 2088, change if needed for firewall
Secure BLF	<u>NOT</u> checked
Force Remote Phone audio through server	Checked
Plug and Play Secret Key	6-64 characters use 0-9, and #. Keep this key closely guarded
Phone Administration Password	4-32 character password to access handset administration
Maximum Active Remote Calls	Should be set to 8. Check with Clearly if more than 8 calls will be permitted.
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	Use the default setting of "1"
Paging Maximum Duration	Use the default setting of "1"
RTP Base Port	Use the default setting of "15000"
RTP DTMF Payload	Use the setting of "101"

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#4. To create a Dial Plan Service Group, go to Phone System > Dial Plan

allworx Home

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WARNING
The emergency number dialing rules have not been set. Please [set](#) them now.

PHONE SYSTEM	NETWORK	REPORTS
Audit PIN Codes Auto Attendants Call Monitors Call Park Call Queues Conference Center Dial Plan Emergency CID Extensions Handsets Languages Music On Hold Outside Lines Paging Speed Dial	Configuration Digital Lines Multi-Site Port Expanders Static Routes VPN	Auto Notification Call Details Configuration Digital Lines Live Calls Network Statistics System Events Users

BUSINESS	SERVICES	MAINTENANCE
Contact Information Message Aliases Schedules Users	DHCP DNS Email VoIP Web	Backup Feature Keys Import / Export Restart Time Tools Update

Create a new service group for Clearly or add it to a previously-defined service group. To create a new one, select “Add new Service Group” near the bottom of the page. The New Service Group screen is displayed.

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Configure the settings as listed in the table below then select “Add”.

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Description	Recommended Setting
Description	Enter any descriptive name. It is common to use the name of the ITSP, i.e. Clearly.
Service Group	Put Clearly in the Service Group by choosing it and selecting “move ->”.

#5. Once the Service Group is created, configure the dialing rules.

On the Dial Plan page in the External Dialing Rules section, the “North American numbering Plan Administration (NANPA)” should be set to “Enabled”. If it is “Disabled”, select “Modify”, check the box labeled “Enable North American Number Plan Administration (NANPA)”, then select “Update”.

The next section of the Dial Plan screen lists Area Codes and Exchanges. Select “Modify” in this section to set up rules to route outgoing calls through Clearlyfly.

Area Code	Exchange	Number Dialed	Service Group	Action
any		9+1+aaa-xxx-nnnn	All CO Lines, SIP Gateways & SIP Proxies	Modify

aaa - area code xxx - exchange nnnn - number

The Modify screen is displayed.

Dialing Rules

The Allworx uses the table below to determine how numbers in your region are dialed and which Service Group is used to complete the call. Enter your **Home** Area Code and any area codes that do not require dialing 1 before the area code. If some exchanges inside an area code require dialing 1 while others do not, you need only to enter the area code/exchanges that require dialing 1. You may also enter any area codes or area code/exchanges for which you require a specific Service Group to be used to complete the call.

Area Code	Exchange	Dial Method	Service Group
add new row			
Home <input type="text"/>		Area Code NOT dialed	All Digital Lines, CO Lines & SIP Gateways
all others		1 + Area Code dialed	Clearlyfly

NOTE

If the **Home** Area Code has been set, seven digit ph...
Group selected for the **Home** Area Code. If the **Home**...
routed using the "All Trunk Devices" Service Group.

the Service
numbers will be

Update Cancel

Enter the home area code and exchanges to be serviced by Clearlyfly into the appropriate boxes. Under Service Group, select the Clearlyfly Service Group from the drop-down menus for all appropriate area codes and exchanges then select “Update”.

The next section of the Dial Plan screen lists special dialing cases including Emergency and Operator. Select "Modify" in this section to direct these calls to Clearly as required.

Type	Number Dialed	Service Group	Action
Emergency	9+911	see Dialing Privileges Group for source of call	Modify
Phone Services (211,311,411,511,611,711,811)	9+n11	All Trunk Devices	
Operator	9+0	All Trunk Devices	
Long Distance Services	9+1010...	All Trunk Devices	
International Calls	9+011...	All Trunk Devices	
Public SIP Directory	8+nnnn (4 digits)	All SIP Proxies	
PIN Code	78+nnnnn (5 digits)	All Digital Lines & CO Lines	
Outside Line Seizure	9#	All Trunk Devices	

The Modify screen is displayed.

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External Dialing Rules

Description	Number Dialed	Service Group
Phone Services (211,311,411,511,611,711,811)	9+n11	All Trunk Devices
Operator	9+0	All Trunk Devices
Long Distance Services	9+1010...	All Trunk Devices
International Calls	9+011...	All Trunk Devices
Public SIP Directory	8+ 11 digits	Clearly
PIN Code	78+ 5 digits	All CO Lines All Digital Lines All Digital Lines & CO Lines All Digital Lines, CO Lines & SIP Gateways All SIP Gateways All SIP Proxies All Trunk Devices Clearly
Outside Line Seizure	9#	

NOTE
Allworx phones must be rebooted when change **Public SIP Directory** or **PIN Code** values.

Update Cancel

Under Service Group, select the Clearly Service Group from the drop-down menus for all required special phone numbers then select "Update".

#6. To create new DID blocks and routing plans, go to Phone Systems > Outside Lines

The screenshot shows the Allworx Home page. On the left is a navigation menu with links: About, Phone System, Business, Network, Servers, Reports, Maintenance, Need help?, Install Checklist, and [Logout]. The main content area is titled 'Home' and contains a yellow warning banner: 'WARNING The emergency number dialing rules have not been set. Please set them now.' Below the banner are six categorized menu sections: PHONE SYSTEM (green header), NETWORK (orange header), REPORTS (blue header), BUSINESS (green header), SERVERS (orange header), and MAINTENANCE (blue header). The 'PHONE SYSTEM' section lists: Audit PIN Codes, Auto Attendants, Call Monitors, Call Park, Call Queues, Conference Center, Dial Plan, Emergency CID, Extensions, Handsets, Languages, Music On Hold, **Outside Lines** (highlighted with a red box), Paging, and Speed Dial. The 'NETWORK' section lists: Configuration, Digital Lines, Multi-Site, Port Expanders, Static Routes, and VPN. The 'REPORTS' section lists: Auto Notification, Call Details, Configuration, Digital Lines, Live Calls, Network Statistics, System Events, and Users. The 'BUSINESS' section lists: Contact Information, Message Aliases, Schedules, and Users. The 'SERVERS' section lists: DHCP, DNS, Email, VoIP, and Web. The 'MAINTENANCE' section lists: Backup, Feature Keys, Import / Export, Restart, Time, Tools, and Update.

Select “add new DID block”.

Direct Inward Dial Blocks add new DID Block
No DID Blocks have been defined.
Direct Inward Dial Routing Plans
No DID Routing Plans have been defined. (new plans can be created when a DID Block is added or modified)

The DID block page is displayed.

DID Block	
Starting Phone Number	<input type="text"/> (include Area Code and Exchange)
Total number of phone numbers in the DID Block	<input type="text"/>
DID Routing Plan	<input type="button" value="make new Routing Plan"/> ▼
<input type="button" value="Add"/> <input type="button" value="Cancel"/>	

In this section, enter the DID information provided by Clearly.

To map DID numbers to extensions, Select “Details” under the DID Routing Plans.

Select “add number to table” in this section.

Routing Plan Information [modify](#)

Description	Routing Plan 1
Default Extension	0 - Operator
Default DNIS Name	{none}
DID Blocks using this plan	(406) 794-0983 / 2 numbers

Phone Number to Extension Mapping

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

Bulk Edit

Phone Number	Extension	DNIS Name	Action
(406) 794-0983	1000 - Clearly User One	{plan default}	Modify

Extension 1000 - Clearly User One

DNIS Name Use Default Extension
0 - Operator
1000 - Clearly User One
1001 - Clearly User Two
1002 - Clearly Remote Phone
1199 - System Administrator
404 - Voicemail
431 - Auto Attendant 1
432 - Auto Attendant 2
433 - Auto Attendant 3
434 - Auto Attendant 4
435 - Auto Attendant 5
436 - Auto Attendant 6
437 - Auto Attendant 7
438 - Auto Attendant 8
439 - Auto Attendant 9
408 - Conference Center

Update

(406) 794-0983

The desired DID to extension mapping can be done as shown above.

The SIP proxy setup page also has to be modified to route the inbound calls to the DID block.

Under Outside Lines, select “Modify” to modify the SIP proxy settings. Check “Routed using DID Block(s)” and the corresponding DID blocks.

Call Route [?](#)

☐ Proxy is an "Enterprise Server" (calls received from this proxy follow the server's internal dial plan)

Calls received from this SIP Proxy go to:

☐ Extension

☐ Auto Attendant

☐ Voicemail for user

☒ Routed using DID Block:

☒ (406) 794-0983 / 2 Numbers / Routing Plan 1

Update Cancel

- #7.** All phones must be rebooted so that they acquire the new settings from the Allworx server. Reboot them either using each handset's on-phone menu (Config > Reboot phone) or the Allworx System Administration Tool (Phone System > Handsets).
- #8.** The Allworx system is now properly configured to receive and send calls via the Clearly service. Verify that both Allworx and Clearly are working properly by making test calls using a variety of dialing plans. If you have any questions or need additional technical support, please call:

Allworx: 866-Allworx (866-255-9679)
Monday – Friday, 8:00am – 8:00pm Eastern Time
support@allworx.com

For detailed configuration instructions, access the latest Allworx System Administration Guide by visiting the Allworx Authorized Reseller Portal (www.allworxportal.com).

