

# Amiba

## Unified Communications

### Getting Started

### as an Installer

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# 1. Introduction

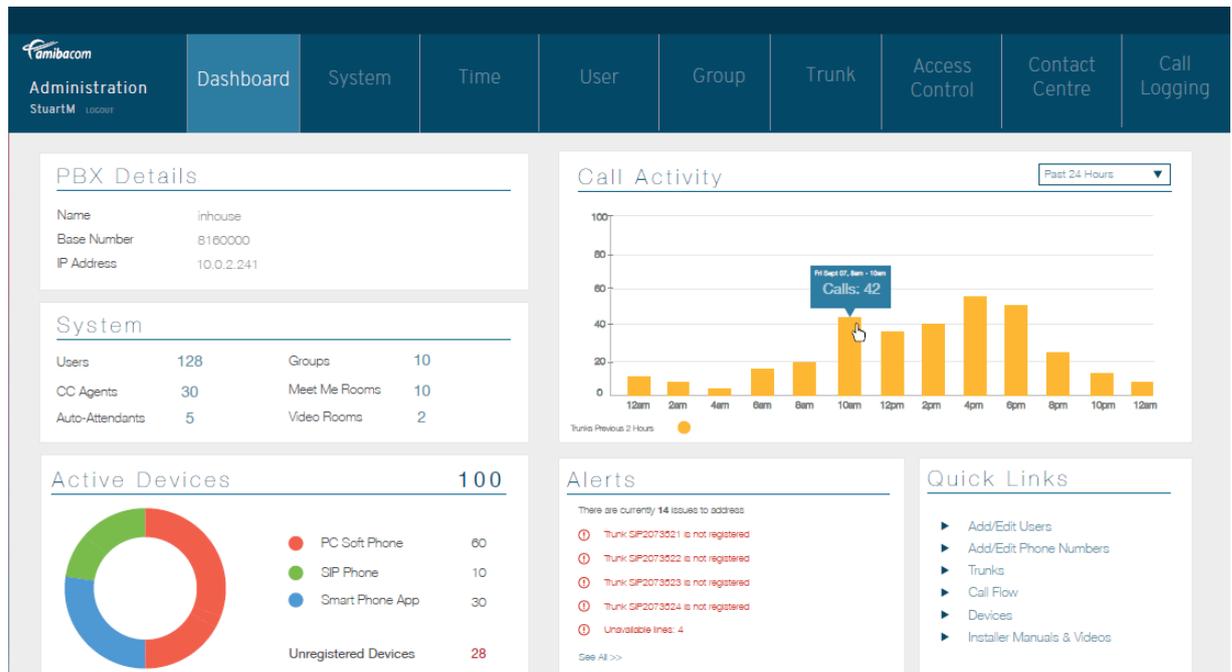
Welcome to the Amiba Unified Communications Cloud-based business system offering:

- Cloud based communications – No clunky hardware
- No Upfront Cost
- Smartphone Apps for iOS and Android
- Redundancy/Resilience
- Your individual online portal
- Chat
- Video Collaboration with document Sharing
- Advanced IVR
- Recording of Voice Calls
- Call Centre Queues
- Fixed Mobile Integration
- Remote Management
- Voicemail to Email
- Presence of Colleagues
- Tele-working
- CTI with leading CRMs
- Inbound/Outbound Call Centre Reports

Amiba Unified Communications supports smartphone apps, tablets, PC softphones and Yealink and Polycom SIP desk phones.

## 2. Getting Started as an Installer

The Installer portal of the Amiba Unified Communications platform gives you full configuration and control of the system. You can add new users and define call flows and advanced features such as Call Centre queues and Automated Attendant with Interactive Voice Response (IVR).



When you login to the Amiba installer portal, the dashboard, at the home icon, gives an overview of the system, the Call Activity, Active Devices and Alerts. Clicking on the tabs at the top of the page allows you to configure individual users and system-wide call settings such as time dependent call flows and auto attendant. Each programming page has a Help button (question mark icon, near the top right, ?) which explains how the feature works and how to set it up.

The following eight steps are necessary to make the system operational.

## 2.1. Enter the Phone Numbers

The SIP trunk interface of the system connects to a SIP trunk to send and receive voice calls. The operator provides a SIP account with one or more public phone numbers.

The first step is to program the phone numbers into the system. Click on the tab Trunks on the dashboard and select Phone Number List.

System Time Users Groups Trunks Access Control Contact Centre Call Logging

Trunks / Phone Number List

Delete All Enter Range Number Per Page: 50 Page (1 / 2 / 3 / 4 / 5 / 6 / 7 / 8 / 9 / 10 / 11 / 12 / 13 / 14 / 15 / 16 / 17 / 18 / 19 / 20)

Index	Phone Number	Name	SIP Account	Music on Hold	Cadence	Calls Received
1	2160000	MSN 01	SIP Account #1	Default	Default Cadence	●
2	2160001	MSN 02	SIP Account #1	Default	Default Cadence	●
3	2160002	MSN 03	SIP Account #1	Default	Default Cadence	●
4	2160004	MSN 04	SIP Account #1	Default	Default Cadence	●
5	9417335000	MSN 05	SIP Account #1	Default	Default Cadence	●
6		MSN 06	SIP Account #1	Default	Default Cadence	●
7	8788092259	MSN 07	SIP Account #1	Default	Default Cadence	●
8	9412034858	MSN 08	SIP Account #1	Default	Default Cadence	●
9		MSN 09	SIP Account #1	Default	Default Cadence	●
10		MSN 10	SIP Account #1	Default	Default Cadence	●
11	5550011	MSN 11	SIP Account #2	Default	Default Cadence	●
12	5550012	MSN 12	SIP Account #2	Default	Default Cadence	●
13		MSN 13	SIP Account #1	Default	Default Cadence	●
14		MSN 14	SIP Account #1	Default	Default Cadence	●
15		MSN 15	SIP Account #1	Default	Default Cadence	●
16		MSN 16	SIP Account #1	Default	Default Cadence	●
17		MSN 17	SIP Account #3	Default	Default Cadence	●
18		MSN 18	SIP Account #1	Default	Default Cadence	●

Enter the Phone Numbers on this page and select which SIP account that they belong to and press Save. There are options here to assign a name, select a music-on-hold and a ringing cadence for each number.

It is important to enter numbers in the correct format including prefixes. A departure from the required number format may result in calls being rejected by the trunk.

## 2.2. Program the SIP Account details

Click on Trunks/SIP Accounts and click the Edit button of the account.

System Time Users Groups **Trunks** Access Control Contact Centre Call Logging

Trunks / SIP Accounts

Index	Name	Username	External Number List	Trunk Access Code	Status	Edit
1				*901	Disabled	Edit
2				*902	Disabled	Edit
3				*903	Disabled	Edit
4				*904	Disabled	Edit
5				*905	Disabled	Edit
6				*906	Disabled	Edit
7				*907	Disabled	Edit
8				*908	Disabled	Edit
9				*909	Disabled	Edit
10				*910	Disabled	Edit
11				*911	Disabled	Edit
12				*912	Disabled	Edit
13				*913	Disabled	Edit
14				*914	Disabled	Edit
15				*915	Disabled	Edit
16				*916	Disabled	Edit
17				*917	Disabled	Edit
18				*918	Disabled	Edit
19				*919	Disabled	Edit
20				*920	Disabled	Edit
21				*921	Disabled	Edit
22				*922	Disabled	Edit
23				*923	Disabled	Edit
24				*924	Disabled	Edit

Help

This brings you to the Edit page for the SIP account.

System Time Users Groups Trunks **Access Control** Contact Centre Call Logging

Trunks / SIP Accounts

Account Settings - SIP Account - 1

Name	<input type="text"/>	Enabled	No
Trunk Access Code	*901	Status	Disabled
<b>Provider Settings</b>			
SIP Server	<input type="text"/>	Forced Proxy IP	<input type="text"/>
SIP Server Port	5060	RegInterval	200
Registration Required	Yes		
STUN Server	<input type="text"/>	Use WAN IP	Enabled
SIP Source IP List	<input type="text"/>		
<b>Subscriber Settings</b>			
Username	<input type="text"/>	AuthID	<input type="text"/>
Password	*****	Base Number	None
<b>Audio Settings</b>			
Codec Priority 1	G729	Codec Priority 2	G711-A
Codec Priority 3	G711-LJ	DTMF Method	RTP Event
Use Progress (103) SIP alerting	Disabled		
<b>Service Settings</b>			
Unregister before Register	No	Use REFER	No
Use 302 Moved temporarily	No	NAT Keep-Alive	No
<b>Outgoing Invite Settings</b>			
From Identity	MSN	CLI Restriction	Anonymous From
CLI Header	None		

Help

If the SIP account requires registration, you must enter the registration details on this page. These are the SIP server URL or IP address, the Username and the Password. Many other important parameters of the SIP trunk are programmed here. You should check with the SIP operator how to configure the trunk parameters. All SIP trunk parameters are explained in the Help text.

## 2.3. Call Flows

Call flows determine how incoming calls are routed to internal users or groups at different times of the day and days of the week.

Click on Time followed by Day/Night Call Flows to define a call flow and give it a name such as Day-time Call Flow, Night-time Call Flow. You may define up to 15 different time-based call flows.

System Time Users Groups Trunks Access Control Contact Centre Call Logging

Time / Day/Night Call Flows

Call Flow	Enable	Non Switching
Day-time Call Flow	<input checked="" type="checkbox"/>	<input type="checkbox"/>
Night-time Call Flow	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 3	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 4	<input type="checkbox"/>	<input type="checkbox"/>
Operator Call Flow	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 6	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 7	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 8	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 9	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 10	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 11	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 12	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 13	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 14	<input type="checkbox"/>	<input type="checkbox"/>
Call Flow 15	<input type="checkbox"/>	<input type="checkbox"/>

Help

Click on Time followed by Day/Night Switch Times and program a start time for each call flow for every day of the week.

Click on Trunks followed by Call Flow to program the call flow destinations. Select the call flow to be programmed from the drop-down list at the top left of the page. To program a call flow for a particular phone number, press the edit button for the number.

## Amiba Unified Communications Getting Started as an Installer

On the edit page for the number, you can route to an internal destination by pressing the + icon beside the destination in the left column or by dragging and dropping the destination into the central assignment column. The destination may be either an individual user, a group, an Auto-Attendant message, an external number or a Meet-Me audio conference room.

**Ringing Assignment - External Number : 35318160041**

<div style="text-align: right; padding-right: 5px;">Add All</div> <ul style="list-style-type: none"> <li>Declan Gibbons (1050) + ▲</li> <li>Dave Victory (1009) +</li> <li>Fred (106) +</li> <li>Polycom Soundstation (1007) +</li> <li>Gary Nolan (1043) + <span style="border: 2px solid red; border-radius: 50%; padding: 2px;">+</span></li> <li>Patch (1096) +</li> <li>Andrea Hartigan (1030) +</li> <li>John Harper (1048) +</li> <li>Fax Test 1088 (1088) +</li> <li>DaveS (1046) +</li> <li>Kevin Doherty (1002) +</li> <li>DMaj Main (1095) +</li> <li>Kevin Kenny 2 (114) +</li> </ul>	<div style="text-align: center; border-bottom: 1px solid #ccc; margin-bottom: 5px;"> <span style="font-weight: bold;">1 Assignment</span> <span style="float: right;">Remove All</span> </div> <ul style="list-style-type: none"> <li>Seamus Doran (1041) -</li> </ul>	<div style="text-align: center; border: 1px solid #ccc; padding: 10px; margin: 10px auto; width: 80%;"> <span style="font-weight: bold; color: blue;">Select Auto-Attendant</span>  <span style="font-weight: bold; color: blue;">None</span>  <span style="color: blue;">+</span> </div> <div style="text-align: center; color: blue; margin-top: 20px;"> <b>Assign External Destination or Meet-Me Room</b> </div>
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If you wish to route calls on a particular number to several different users, you must first create a group with these users. Then select the group as the call flow destination. Groups are set up on the page Groups/ Group Assignment.

System   Time   Users <b>Groups</b> Trunks   Access Control   Contact Centre   Call Logging									
Groups / Group Assignment									
Index	Group	Members	Edit	Delete	Copy From	Copy To			
1	800			✘					
2	801			✘					
3	802			✘					
4	803			✘					
5	804			✘					
6	805			✘					
7	806			✘					
8	807			✘					
9	8	Operator Group	Extn 101(101)	✘					
10	808	Answer Machine	Extn 101(101)	✘					
11	810			✘					
12	811			✘					
13	812			✘					
14	813			✘					
15	814			✘					
16	815			✘					
17	816			✘					
18	817			✘					
19	818			✘					
20	819			✘					

If you want to play an Auto Attendant message while users are being rung, a group must first be created, to include the Auto Attendant message and the users.

## 2.4. Auto-Attendant

The Auto-Attendant may be programmed to answer incoming calls automatically and to guide the caller through a series of announcements to the desired destination. If you have routed incoming call flows to the Auto-Attendant, you must also program how the Auto-Attendant will handle these calls and program the interactive responses of the Auto-Attendant to the selections made by callers.

To do this, go to the System/Auto-Attendant page and consult the online help on how to configure the Auto-Attendant.

System Time Users Groups Trunks Access Control Contact Centre Call Logging

System / Auto-Attendant

Auto-Attendant: Enabled Number Per Page: 50 Page ( 1 / 2 / 3 / 4 / 5 / 6 )

Message	Duration	Controls	Number	Delay Timer	Digit Assignment	Queue
Welcome Message	----	▶ ⊗ ▼	7000	0	Digits	☐
Hold Message	----	▶ ⊗ ▼	7001	10	Digits	☐
Greeting and Clear	----	▶ ⊗ ▼	7002	0	Digits	☐
Attendant Msg. 4	----	▶ ⊗ ▼	7003	10	Digits	☐
Attendant Msg. 5	----	▶ ⊗ ▼	7004	10	Digits	☐
Attendant Msg. 6	----	▶ ⊗ ▼	7005	10	Digits	☐
Attendant Msg. 7	----	▶ ⊗ ▼	7006	10	Digits	☐
Attendant Msg. 8	----	▶ ⊗ ▼	7007	10	Digits	☐
Attendant Msg. 9	----	▶ ⊗ ▼	7008	10	Digits	☐
Attendant Msg. 10	----	▶ ⊗ ▼	7009	10	Digits	☐
Attendant Msg. 11	----	▶ ⊗ ▼	7010	10	Digits	☐
Attendant Msg. 12	----	▶ ⊗ ▼	7011	10	Digits	☐
Attendant Msg. 13	----	▶ ⊗ ▼	7012	10	Digits	☐
Attendant Msg. 14	----	▶ ⊗ ▼	7013	10	Digits	☐
Attendant Msg. 15	----	▶ ⊗ ▼	7014	10	Digits	☐
Attendant Msg. 16	----	▶ ⊗ ▼	7015	10	Digits	☐
Attendant Msg. 17	----	▶ ⊗ ▼	7016	10	Digits	☐
Attendant Msg. 18	----	▶ ⊗ ▼	7017	10	Digits	☐
Attendant Msg. 19	----	▶ ⊗ ▼	7018	10	Digits	☐

Help

## 2.5. Email

To configure an email account on the system to send voicemail messages to users via email, click on System/External Servers, select SMTP from the drop-down box and program the SMTP server for the system as explained in the help.

## 2.6. Users

Go to the page Users/User Settings to enter the following details for each user: Name, E-mail address, mobile number, PIN code, for user login, and outgoing CLI. On this page you may also activate the user smartphone app on the system, connect to the user portal, import or export the user contacts or send a welcome email to the user.

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## **2.7. Firewall**

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The system allows remote access to the Installer, Admin and User portals, xHTML, CTI, Provisioning of SIP Desk phones and SOAP API's. The system firewall controls access to these services by defining whitelists of authorised remote IP addresses. These are configured on the pages System/Firewall and System/Whitelist (RAS) IP addresses.

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## **2.8. Provisioning Yealink and Polycom phones**

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Yealink T-series and Polycom VVX series SIP desk phones can be provisioned automatically by the system based on the phone's MAC address. This means that the SIP registration details and user preferences can be downloaded automatically from the system allowing Plug and Play installation of the phones.

The provisioning server is configured on the page System/ Provisioning Server and the phones are provisioned on the page Users/SIP Phone Provisioning.