Amiba
Unified Communications
Getting Started
as an Installer

Contents

1. Introduction

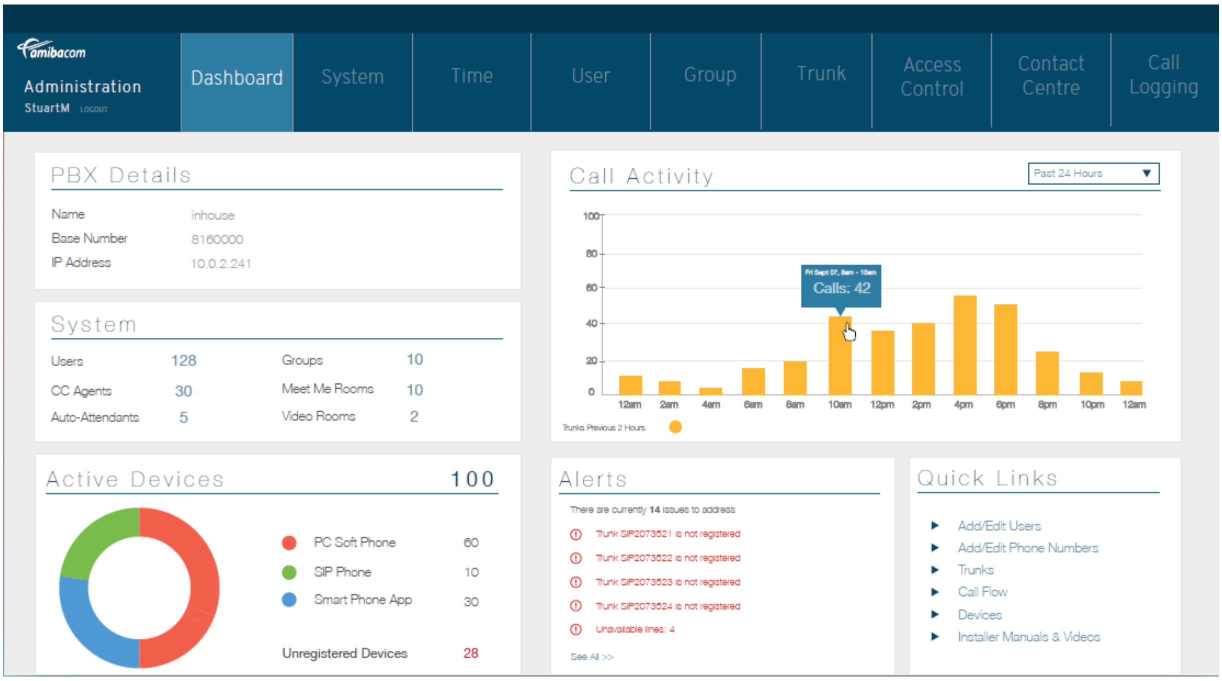
Welcome to the Amiba Unified Communications Cloud-based business system offering:

- 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).



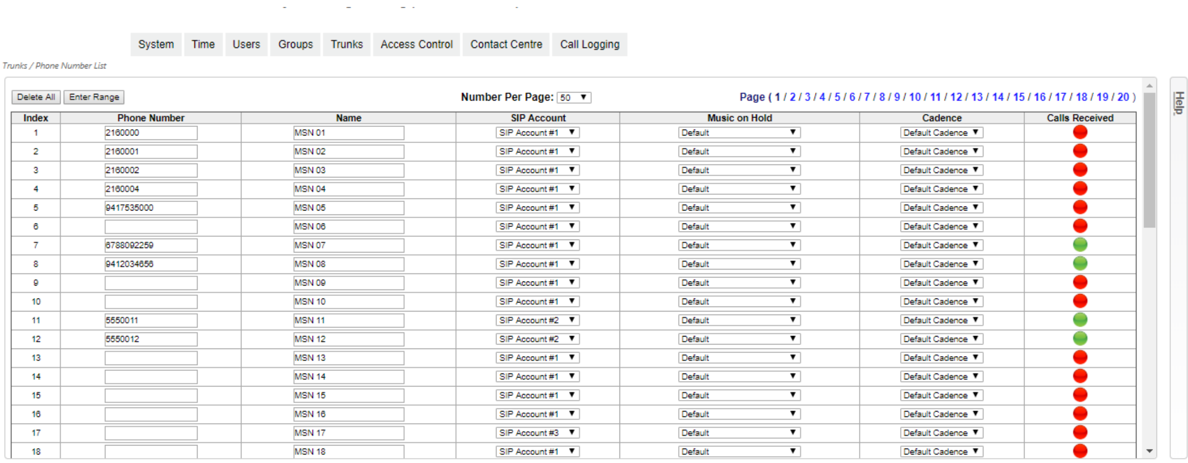
When you login to the Amiba installer portal, the dashboard, at the home icon, gives an overview of the system, the Call Activity, Active Devices and Alerts. Clicking on the tabs at the top of the page allows you to configure individual users and system-wide call settings such as time dependent call 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.

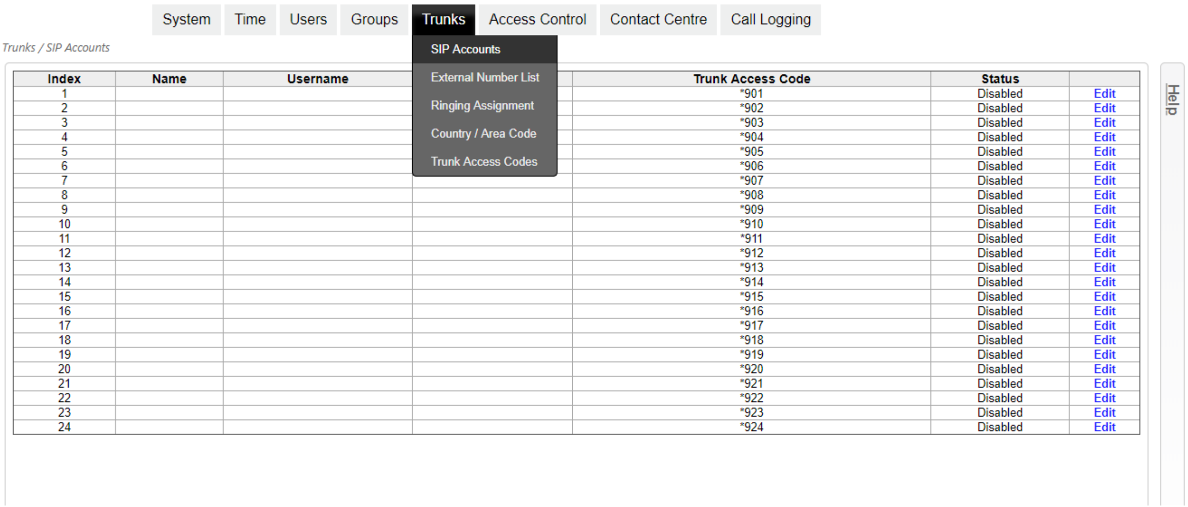


Enter the Phone Numbers on this page and select which SIP account that they belong to and press Save. There are options here to assign a name, select a music-on-hold and a ringing cadence for each number.

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.

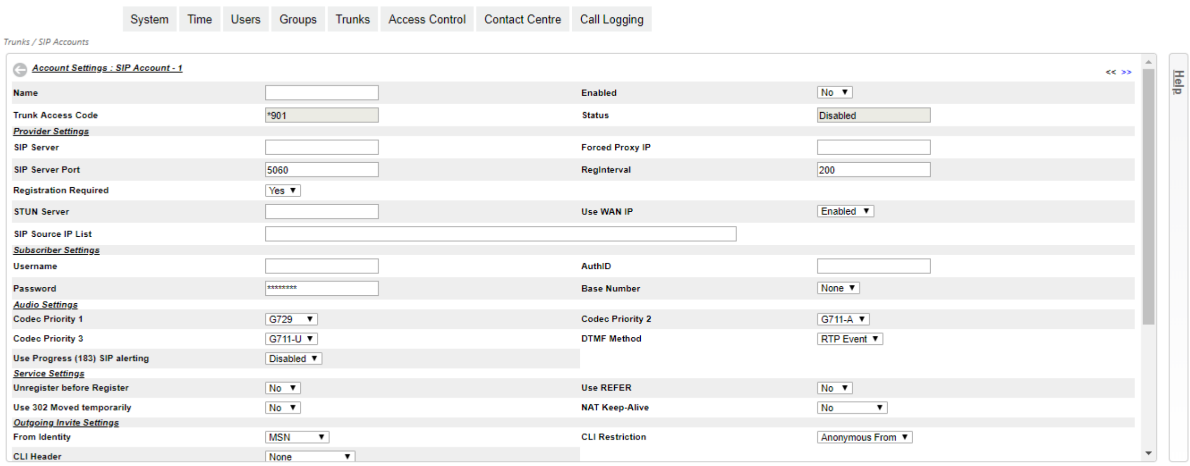
2.2. **Program the SIP Account details**

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



Index	Name	Username		Trunk Access Code	Status	
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

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

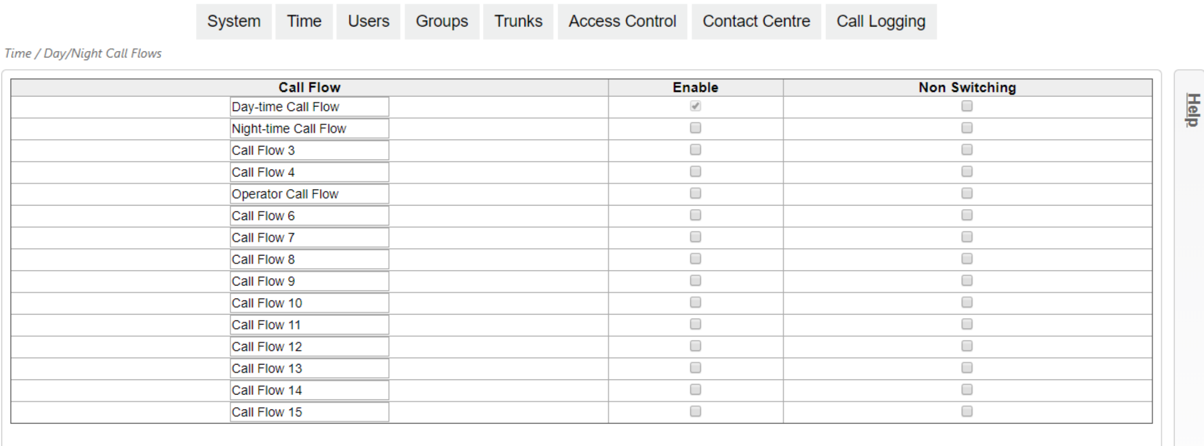


If the SIP account requires registration, you must enter the registration details on this page. These are the SIP server URL or IP address, the Username and the Password. Many other important parameters of the SIP trunk are programmed here. You should check with the SIP operator how to configure the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

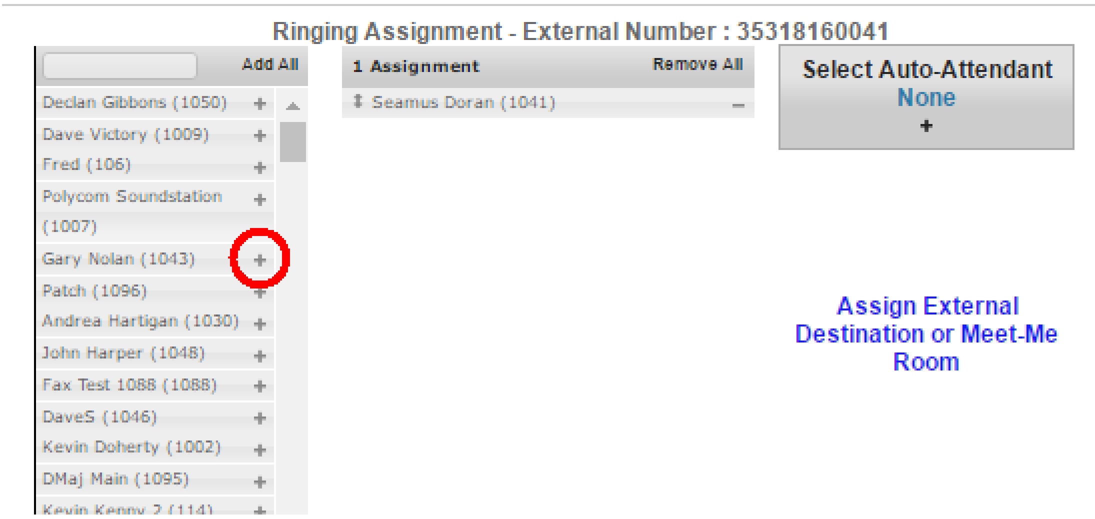
Click on Time followed by Day/Night Call Flows to define a call flow and give it a name such as Day-time Call Flow, Night-time Call Flow. You may define up to 15 different time-based call flows.



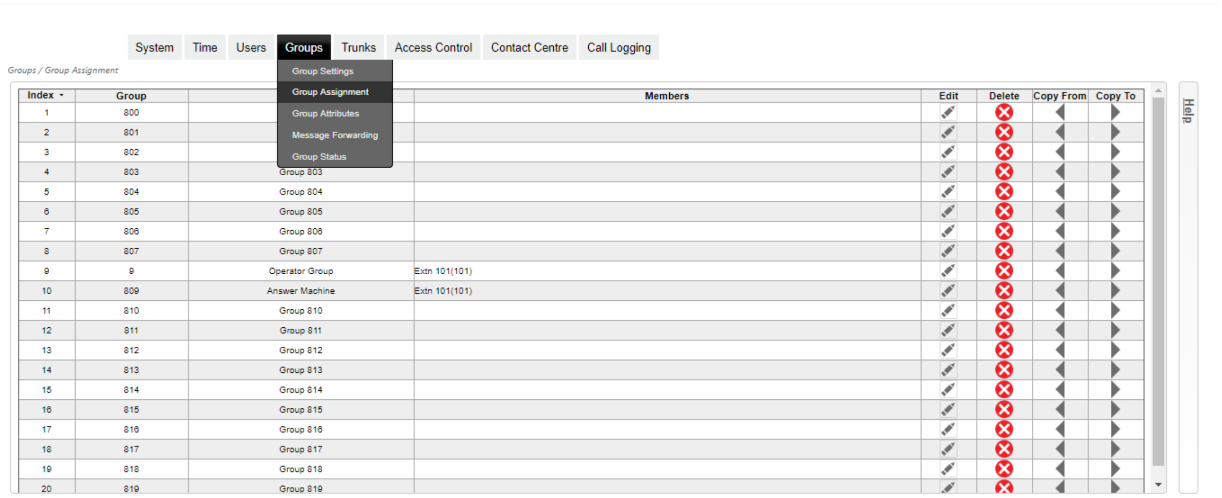
Click on Time followed by Day/Night Switch Times and program a start time for each call flow for every day of the week.

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.

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.



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.

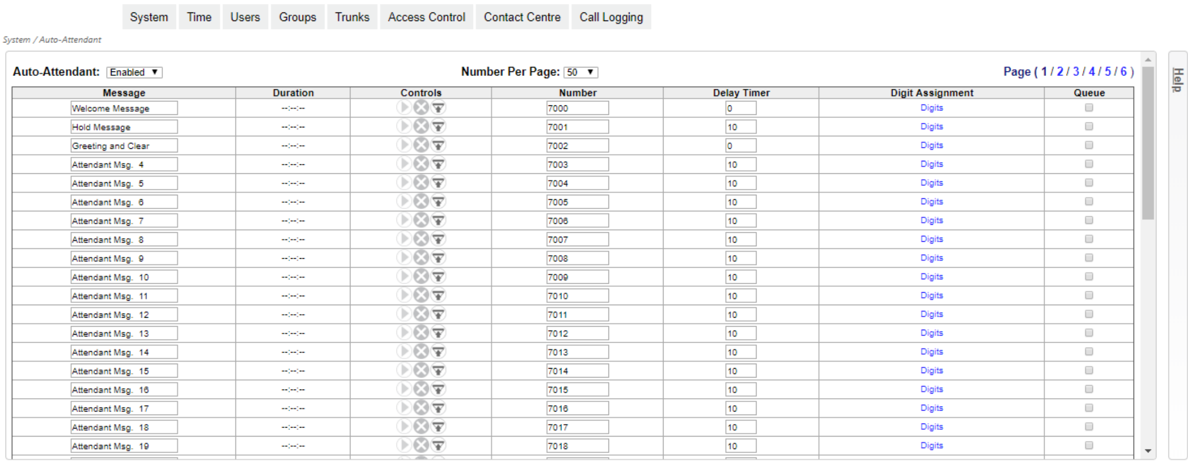


If you want to play an Auto Attendant message while users are being rung, a group must first be created, to include the Auto Attendant message and the users.

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.



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.

2.7. **Firewall**

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.

2.8. **Provisioning Yealink and Polycom phones**

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.