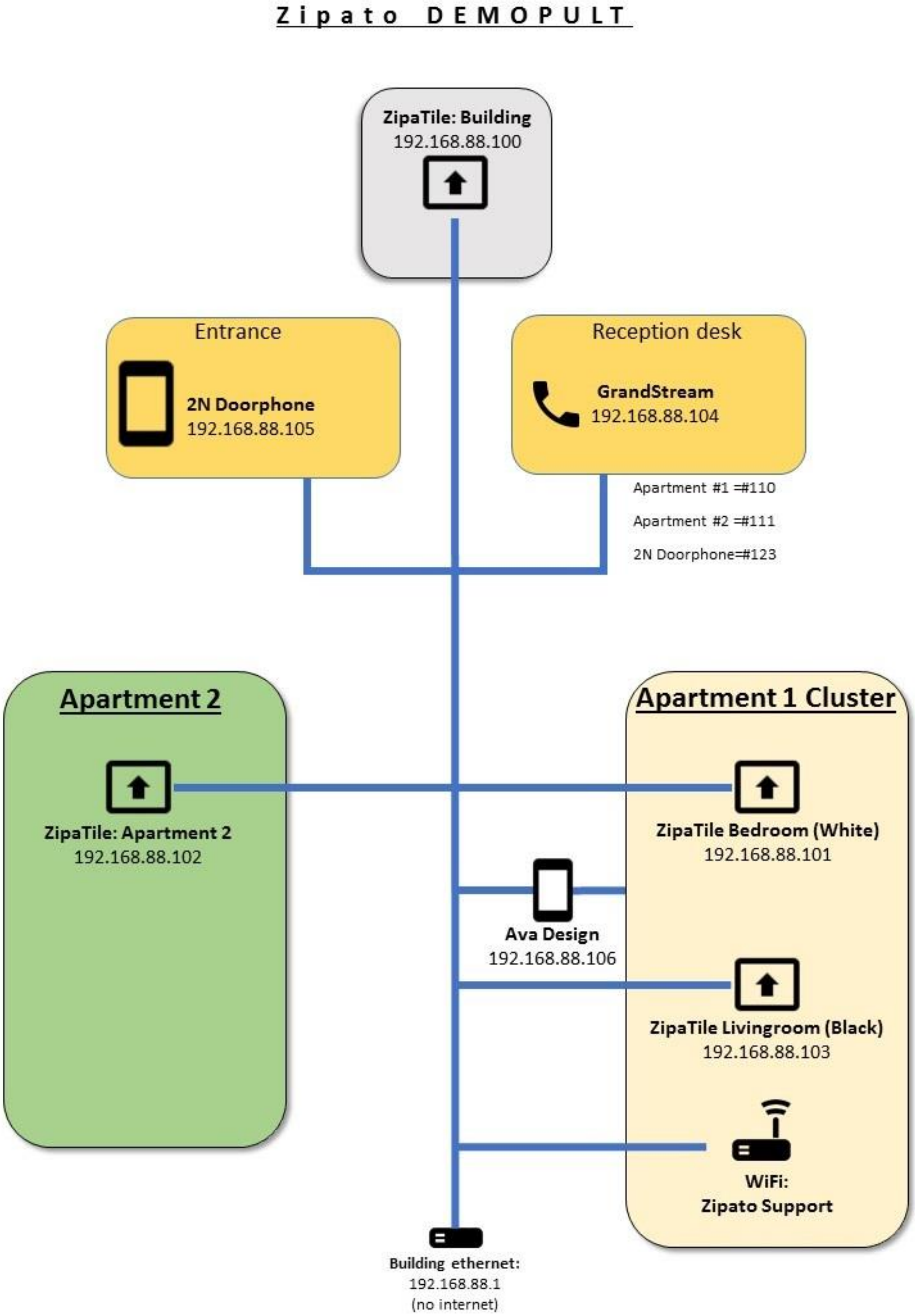
ZIPATO INTERCOM

Content:

1. Zipato's Network:

Scheme of our office SIP Server configuration:

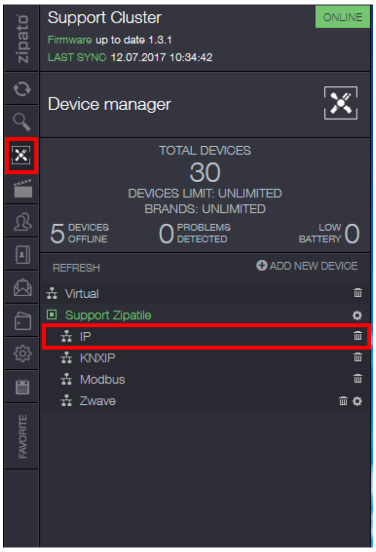


2. How to create a SIP Server:

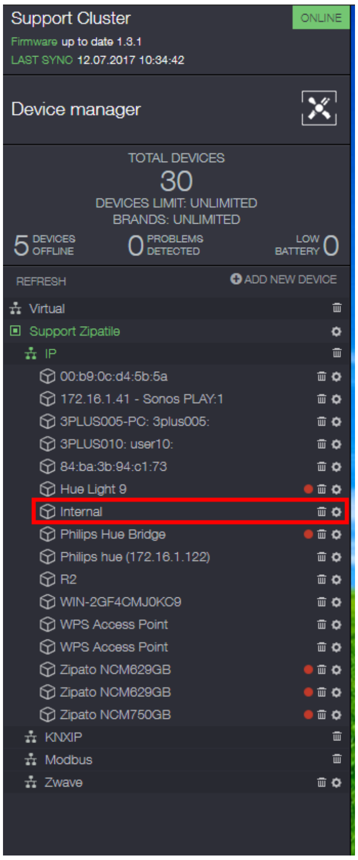
2.1 Creating SIP Server

First of all you have to have PRO License in order to use SIP Server and InterCom.

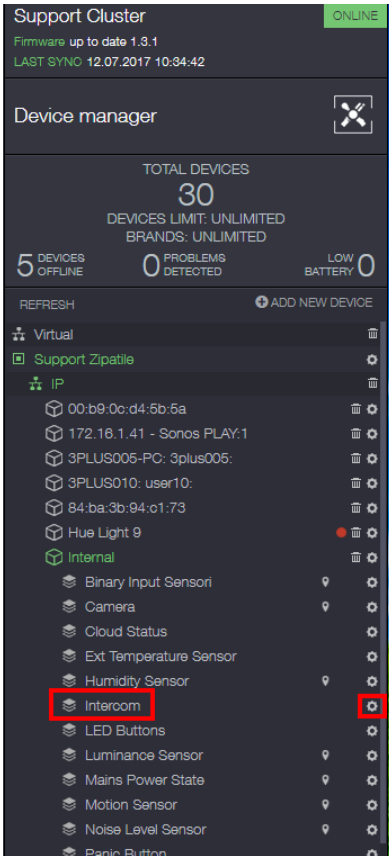
On ZipaTile go to Device Manager and follow the pictures guide.



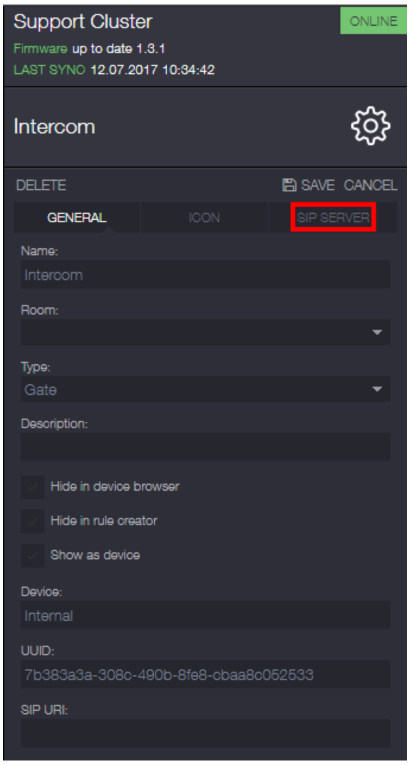
- Go to Device Manager and Select IP



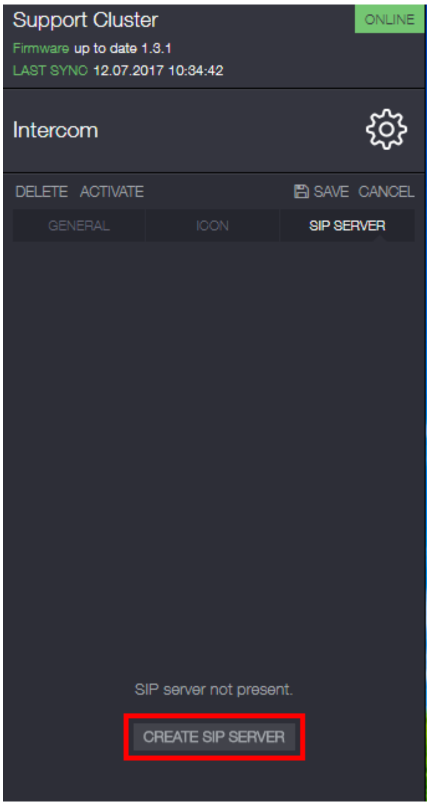
- Select Internal



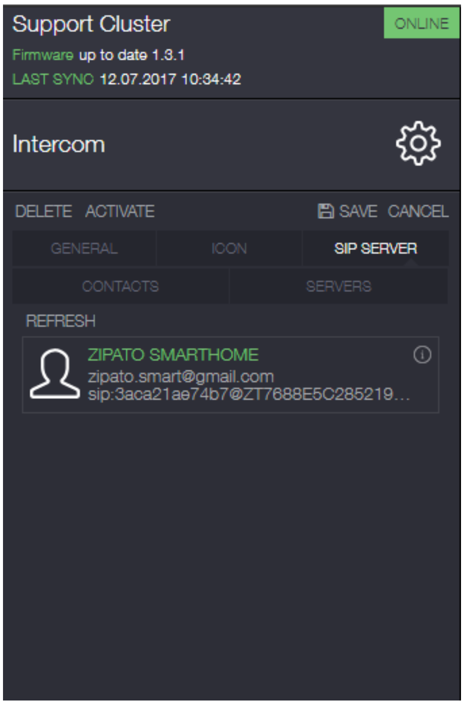
- Go to Intercom settings



- Press on SIP SERVER

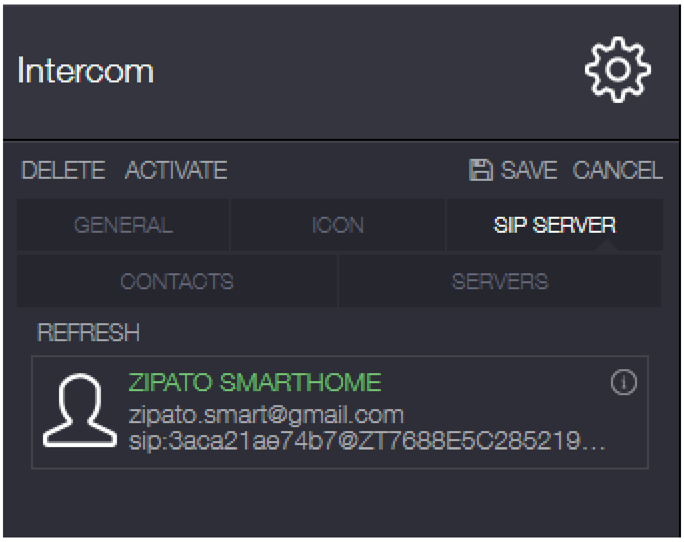


- To create SIP SERVER press CREATE SIP SERVER



- Now you have created SIP Server
- <u>If you are in cluster then you must activate one controller to be the main SIP controller</u>

2.2 Button functionality:



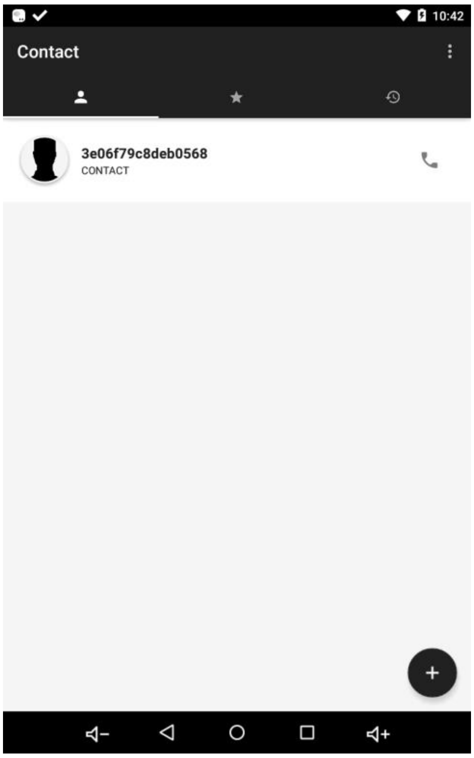
- General: this is general information about Intercom
- Icon: option for changing the default icon
- SIP SERVER: shows 3 new options
- Contacts: it’s option for all the devices that are added to this SIP Server
- Servers: this section is for creating new SIP Server …

2.3 Connecting to InterCom app

Next thing is to connect to your account on ZipaTile and check the status of SIP SERVER.

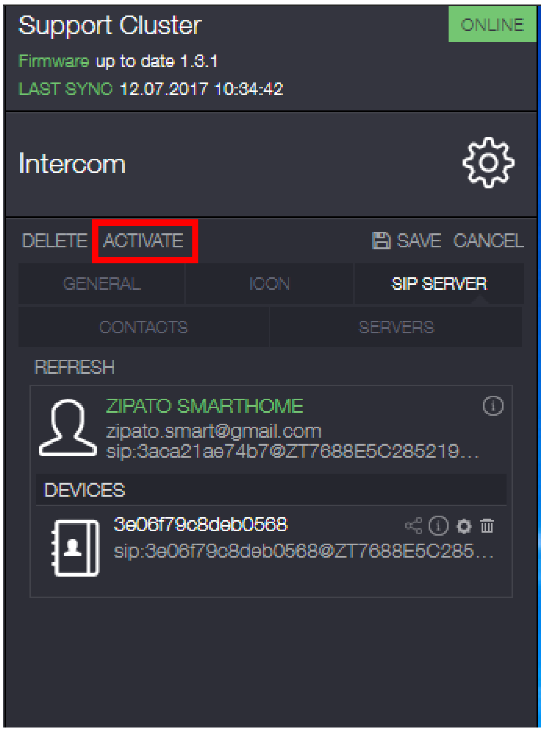
To check status of SIP SERVER on ZipaTile, go to Main menu → Settings → Sip Server

After you checked the status and Sip Server is ready to use, start InterCom app. InterCom app will automatically create new user.



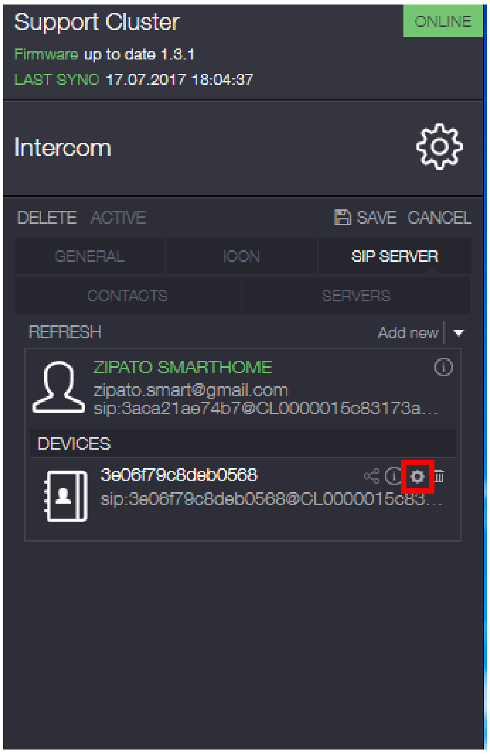
To update the Intercom field on Web UI press REFRESH and the user will be visible.

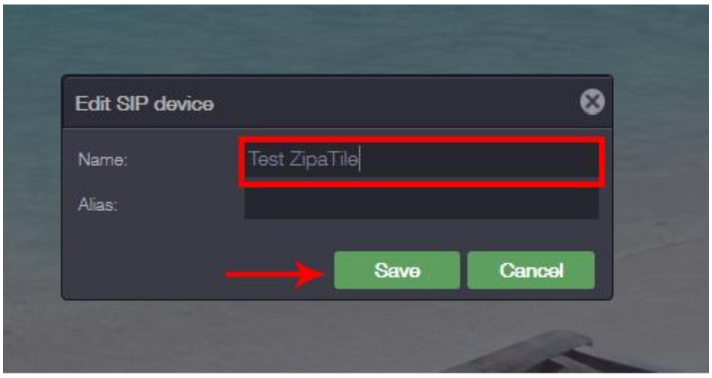
Since we are in cluster first you have to ACTIVATE SIP Server (outside of the cluster you don't have to do this step)



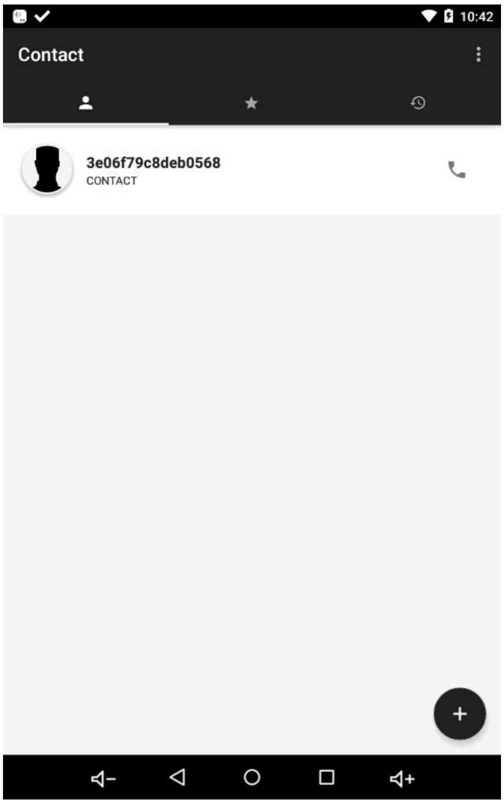
- Press ACTIVATE to activate cluster SIP Server

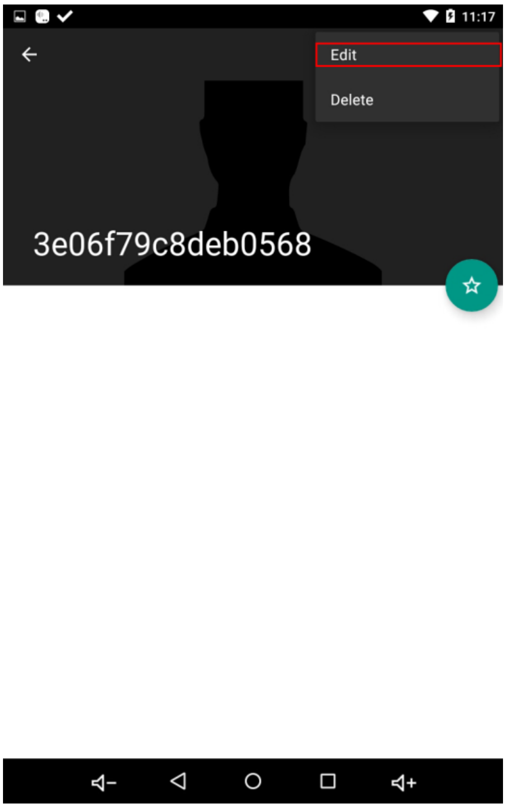
To edit user name press settings option on Web UI, and change the name:

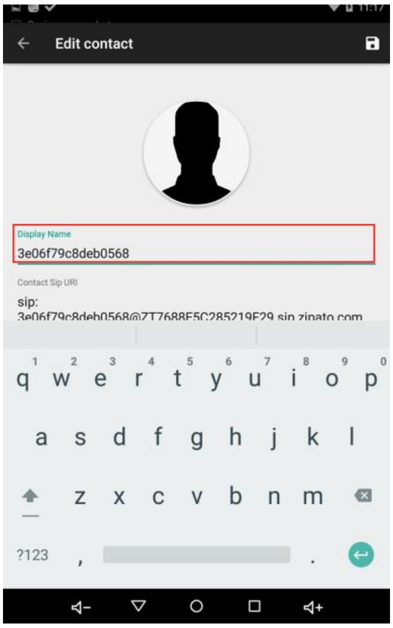




Also, you can do it via InterCom app:







- To edit name, tap on the name you want to change, then press the option on the right top corner and then press Edit, new window will be opened and there you can change the name.
- When you are done with name editing, press SAVE button on the right top corner.

Connecting the 2nd controller on the same account (house installation):

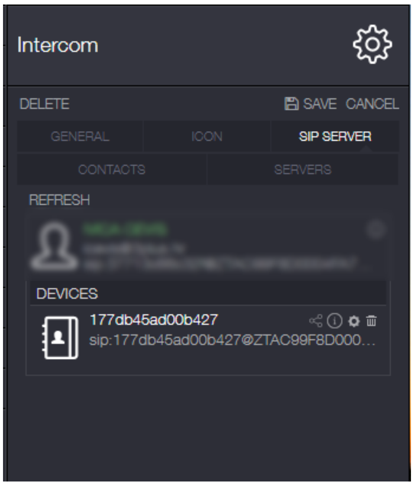
- To connect two ZipaTiles with Intercom you have to have them on the same account and also have PRO License for the other one
- The procedure is the same as it was for the first ZipaTile

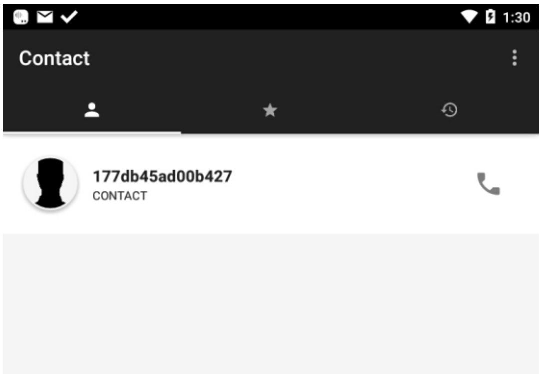
After you are done with configuring you should see all your devices REGistered in Sipserver.

2.4 Making a call/ Connecting 2 ZipaTiles on same account:

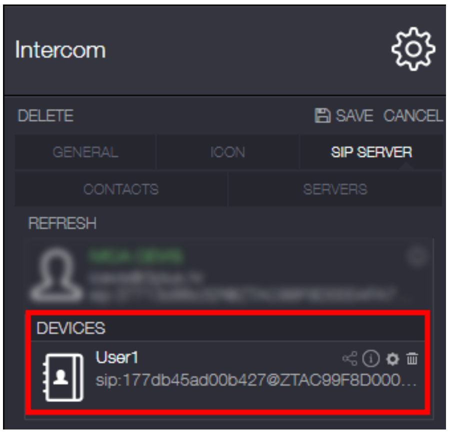
2.4.1 Connect 2 ZipaTiles that are on the same box, follow this procedure:

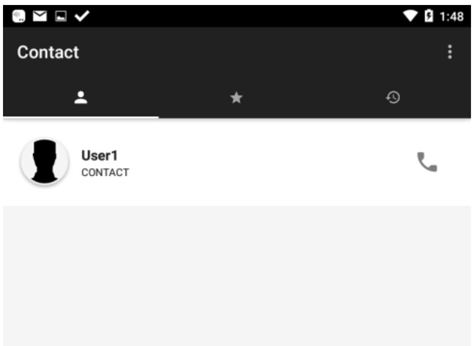
- First switch ZipaTiles to be on the same account (e.g. you have 2 controllers named ZT1 and ZT2, switch the other one to match the first one, so when you do that both controllers would be on the same e.g. ZT1 controller
- Create SIP Server on that account
- After you are done creating SIP Server, open Intercom on first ZipaTile
- New user will be created



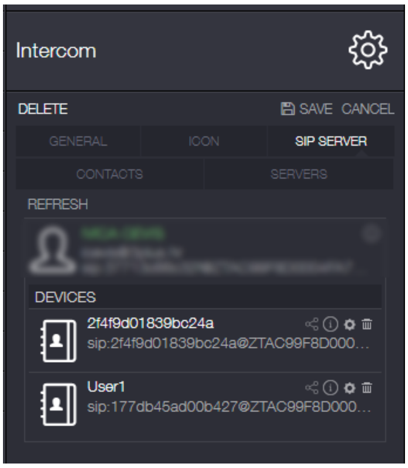


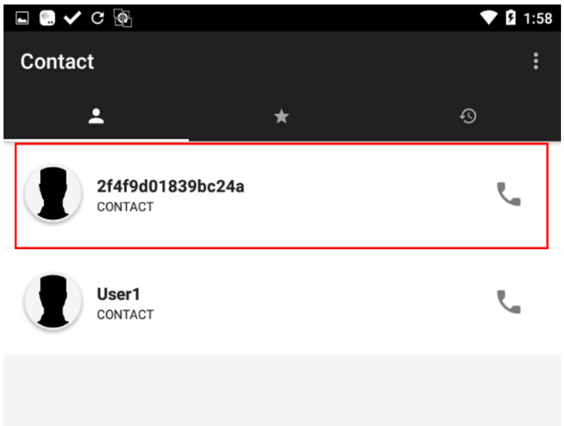
- Change the name of that user, to see the changes on InterCom refresh it



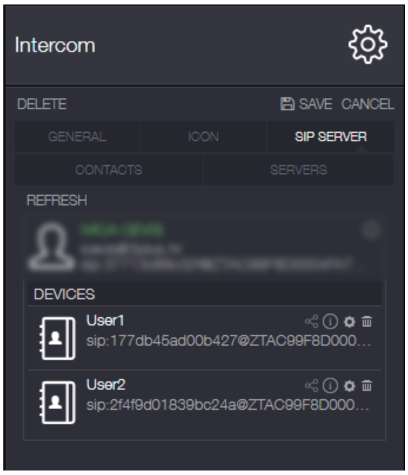


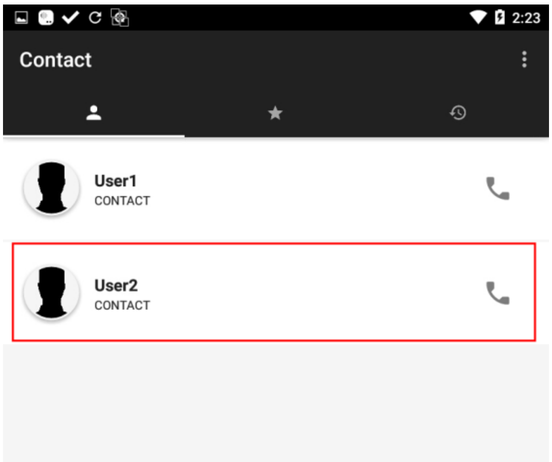
- Now do the same with the other ZipaTile, switch box to the same controller as your first ZipaTile



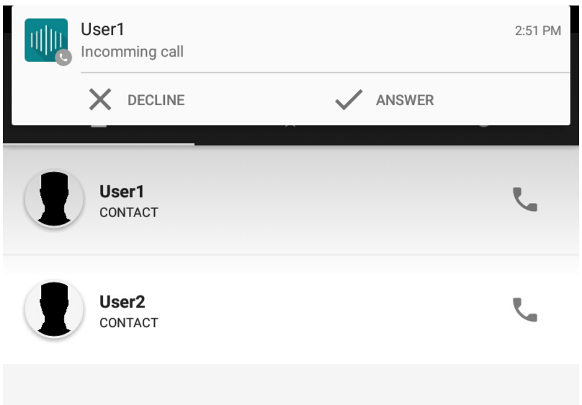


- Change the name of that user



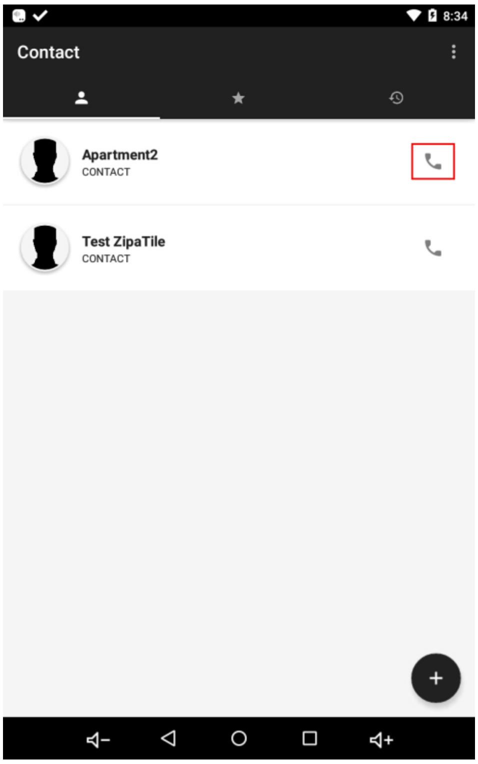


- Now you have connected 2 ZipaTiles to communicate through the Intercom app



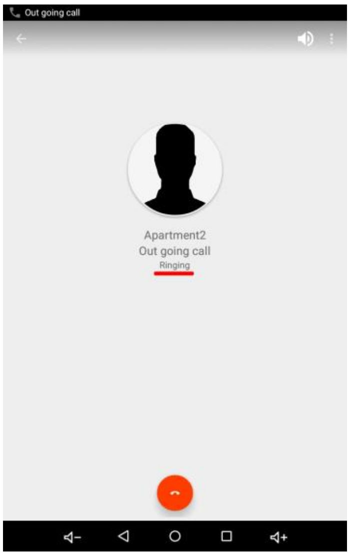
2.4.2 Test call

- If you set everything up as described, you are now ready to make a first call. The important thing is to have a stabile internet connection.
- To make a call press on the client contact in your InterCom app:



- To call, just press on the phone icon

After you placed the call, you will see this:



2.5 SIP CALL STATUS

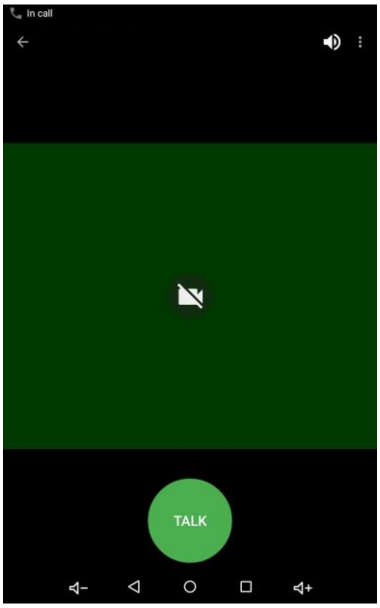
- **Trying** - Extended search being performed may take a significant time so a forking proxy must send a 100 Trying response.
- **Ringing** - Destination user agent received INVITE, and is alerting user of call

- **Decline -** The destination does not wish to participate in the call, or cannot do so, and additionally the destination knows there are no alternative destinations (such as a voicemail server) willing to accept the call.

- **Terminated Dialog** - Can be used by User Agent Server to indicate to upstream SIP entities (including the User Agent Client (UAC)) that an early dialog has been terminated

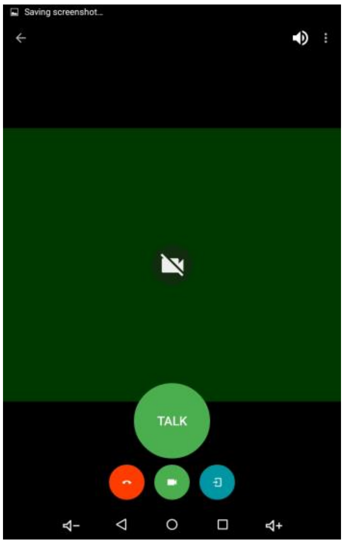
- Those are the most common responses, the whole list of sip responses can be found here: https://en.wikipedia.org/wiki/List_of_SIP_response_codes

Next screen is if the other side has answered looks like this:



- This is normal state of the call, the green screen is protection of unwanted use of your camera
- You are in control of who can see you

To enable the camera, press somewhere on the screen to see the other options in call:



-  button is for the canceling the call
-  button is for enable/disable of ZipaTile camera
-  button is for opening the door
- 

 button works when you press and talk (press to talk option)

3. Building configuration

- All controllers needs PRO License in order to work properly with SIP Server
- Building SIP Server:

- All apartment SIP servers are connected to Building SIP Server and they contain configuration of the building with apartments that are separated

- Ethernet connection with router without internet is used to process calls from doorphone on all apartments (note* controller has to have internet access during configuration)

- It is recommended to set Building SIP server with static IP address (on the main router)

- **1st Apartment:**

- 2 ZipaTiles in a Cluster

- 2 SIP Servers are in cluster and only one is active in cluster

- ZipaTiles are connected with Master SIP Server (Building) in order to receive calls from building doorphone

- **2nd Apartment:**

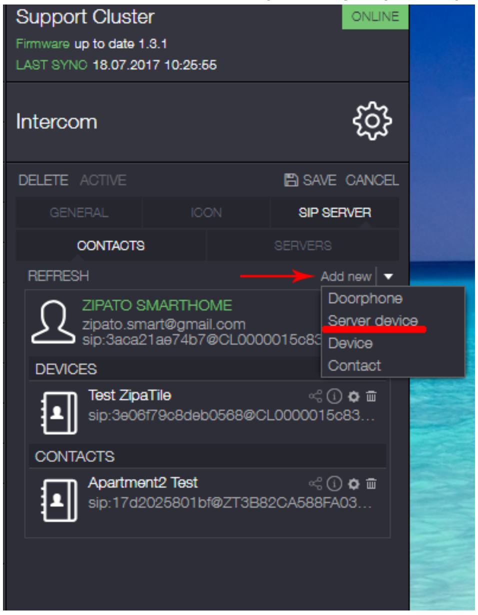
- Single SIP Server communication configuration for Apartment 2

- ZipaTile connected with Master SIP Server (Building) in order to receive calls from building doorphone

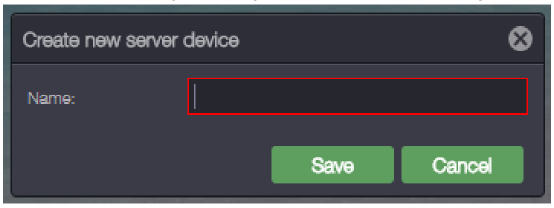
- Create one Building SIP Server and connect all other apartments (accounts) to it

3.1 Procedure for Building ZipaTile, adding controllers to Main Sip Server:

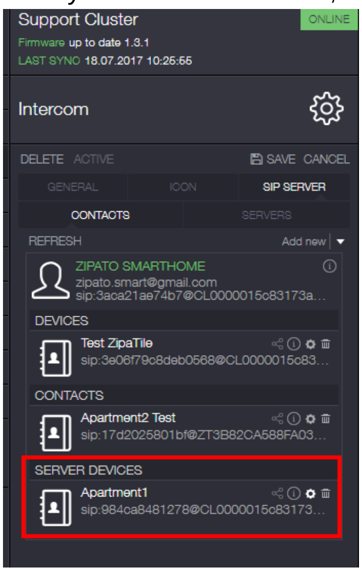
1. Create Server device (Adding ZipaTile)



2. Add name for your ZipaTile contact and press SAVE



3. After you are done with that, the new server device will be created

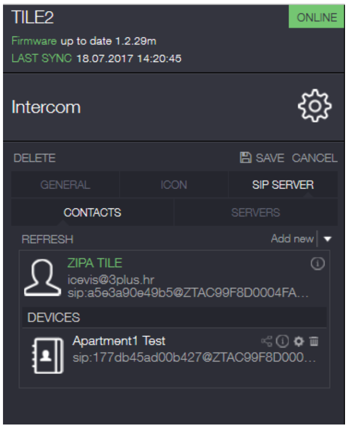


4. When you are done with creating new Server Device (ZipaTile) go to a different ZipaTile account, and create SERVER.

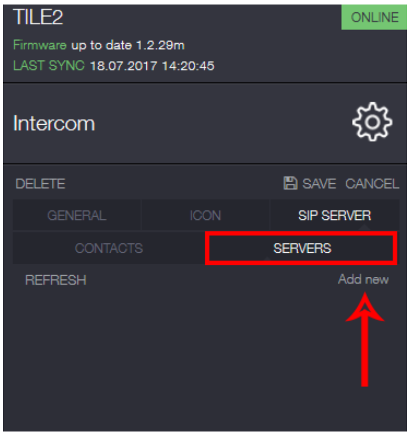
3.2 Procedure for the other accounts:

1. Connect to different account and you will see something like this (note* if you haven't installed SIP Server before, you have to do it.)

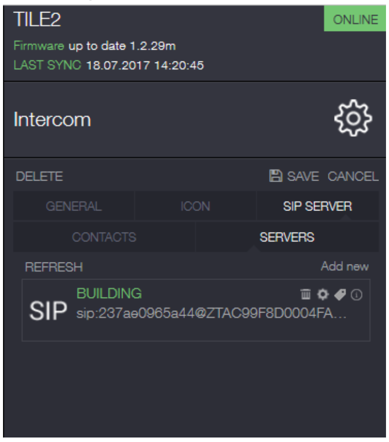
- After you made SIP Server, open InterCom on that ZT to get the client contact



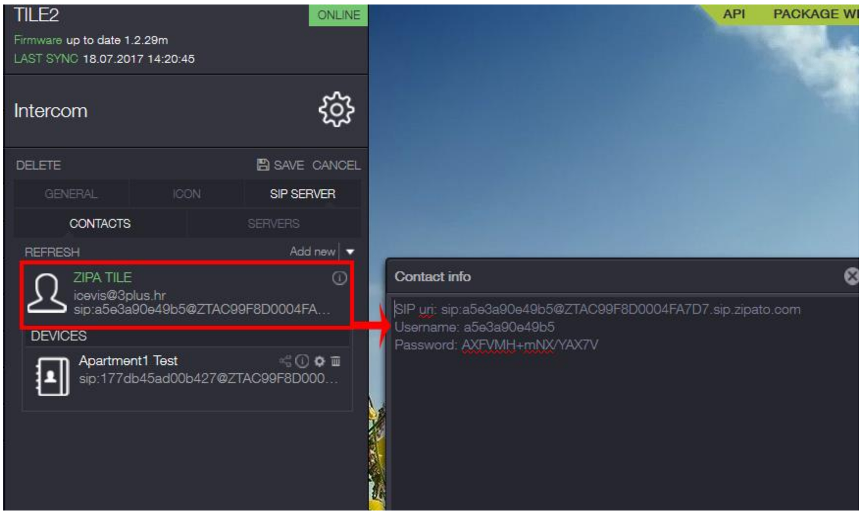
2. Go to SERVERS and create one and name it e.g. Building



3. Building server is created

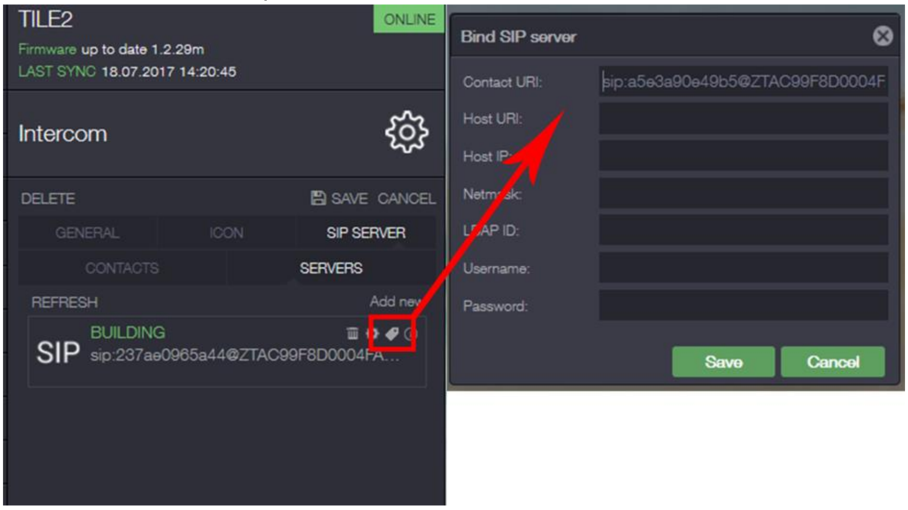


4. To configure it, first copy the current clients uri address

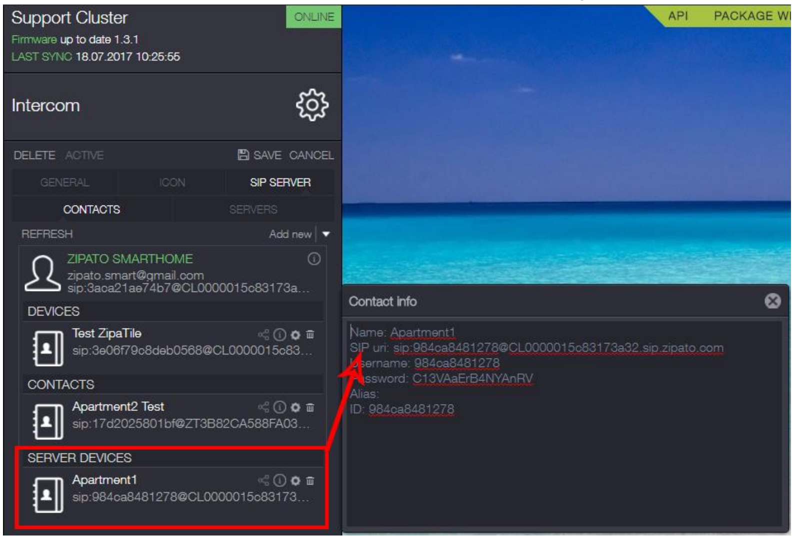


- Copy SIP uri: and paste it in Building configuration under <u>Contact uri</u>

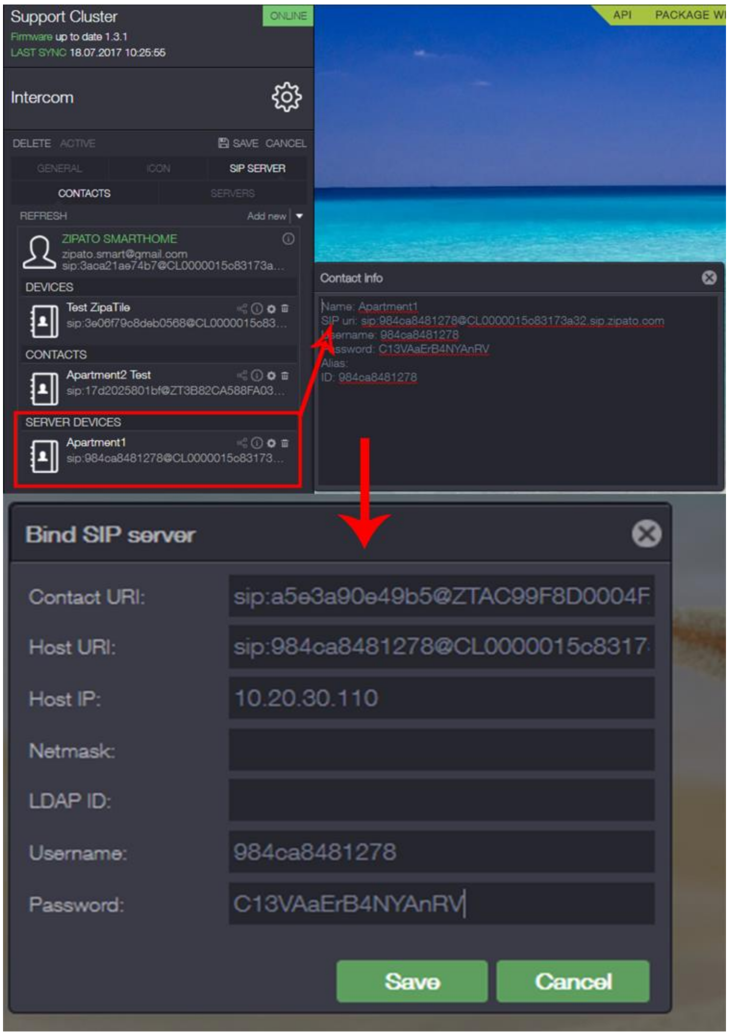
5. Go to SERVERS and paste SIP uri under Contact URI



6. After that fill the rest from the Server Device that you created before named Apartment1



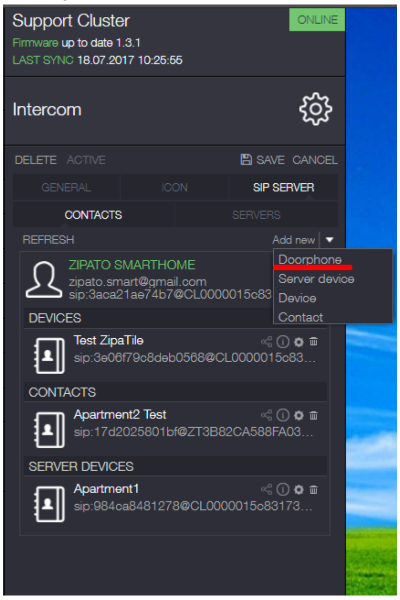
7. This is how it should look when Bind SIP Server form is filled



- Make sure that you enter the right credentials, and to enter your Host IP, the best is to enter the SIP Server on ZT and check the IP Address (Main Menu → Zipato Settings → SIP Server → check the IP address)
- *note: Netmask is not necessary, and LDAP ID will be added automatically after SAVE
- Now every configuration is the same for the other apartments that you want to add on Building SIP Server, the Host IP stays the same, and other information needs to be changed

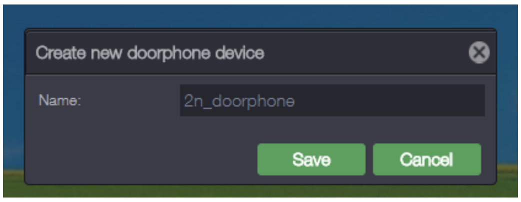
4. Adding 2N Doorphone into SIP Server:

1. First you have to add it on Building SIP Server:

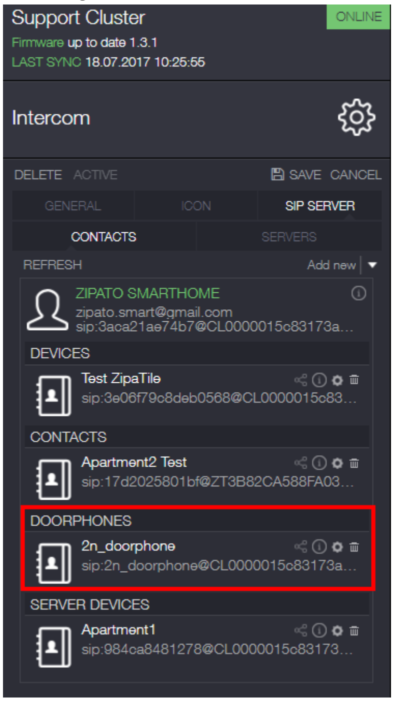


- Go to Add new, and select Doorphone

2. Enter the name and SAVE it:



3. You'll get this:

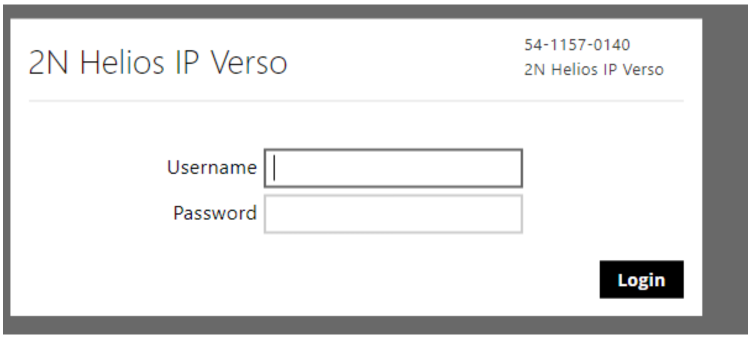


4. After you named the 2n Doorphone, now it's time to enter it's own UI and configure the rest there

4.1 2N DOORPHONE configuration:

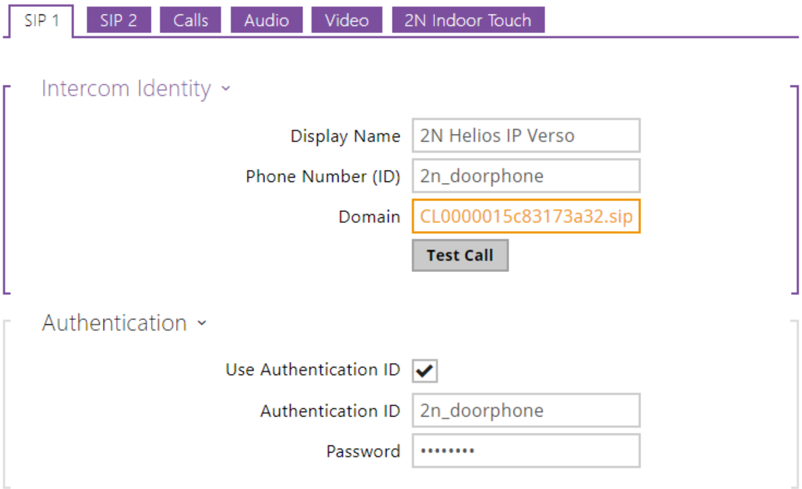
1. Enter the 2N IP address in web browser, to find the IP Address of 2N, plug it to power and press the specific quick dial button five times **within 30 seconds** after the first sound signal (if the address is 0.0.0.0, it means that the intercom has not obtained the IP address from the DHCP server)

2. Enter the credentials:

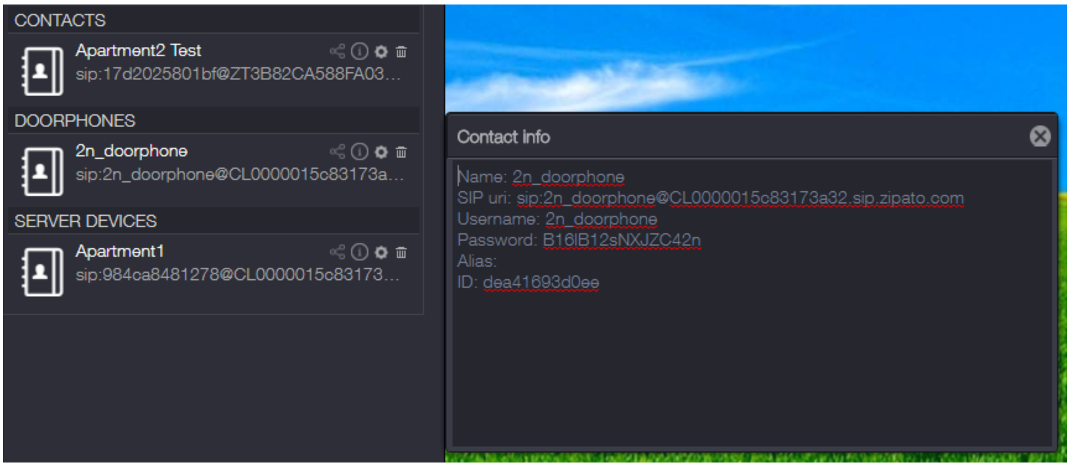


3. The front page:

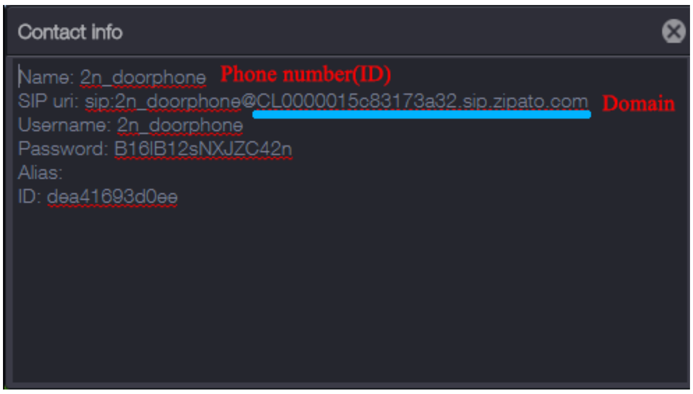
4. Go to services and enter the credentials to register 2N Doorphone



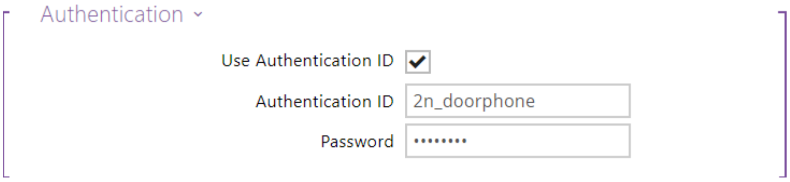
5. Credentials could be found in Web UI on Building SIP account where we created 2n_doorphone device



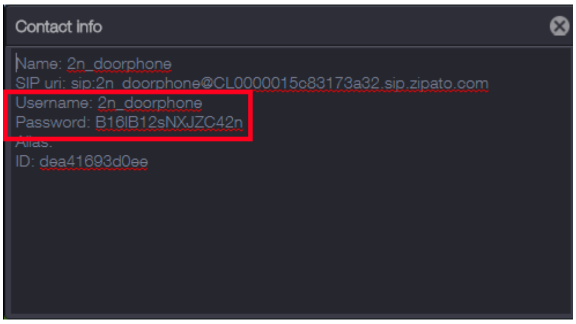
6. Enter the Display name, after that enter the Phone Number which is basically the name you gave to doorphone when you created it.



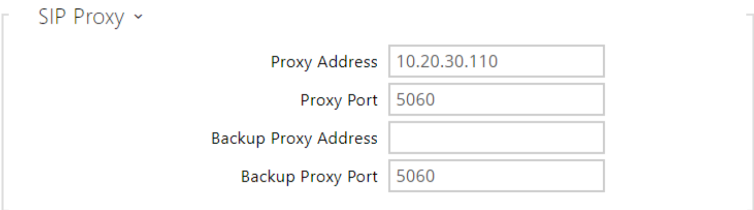
7. Next is Authentication:



- In authentication window enter the username and password for 2N doorphone, you can find it here:

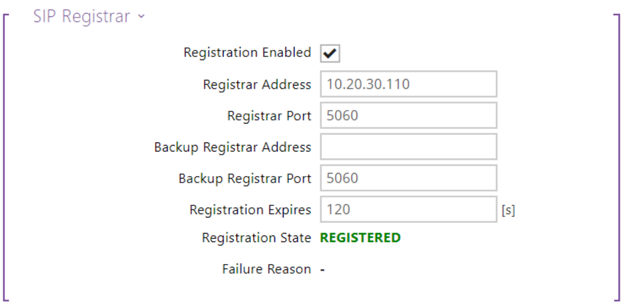


8. SIP Proxy:

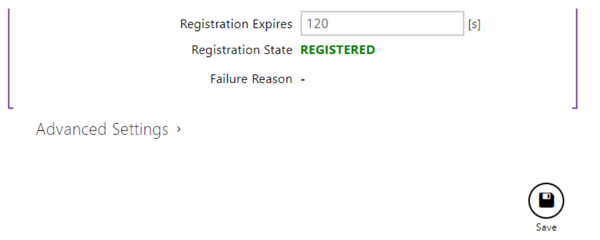


- In SIP Proxy we have to enter IP address of Master SIP Server in this case it's Support Cluster, so its IP address is 10.20.30.110, so we enter that under Proxy Address, Proxy port and Backup Proxy Port by default are 5060

9. Last is SIP Registrar:



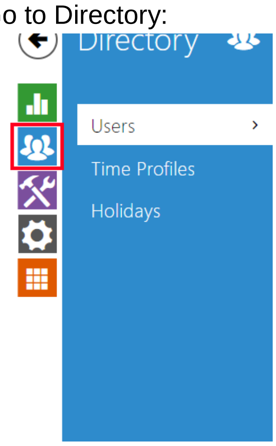
- We enable it, and enter the following data
- In registrar address we have to enter the IP address of Master SIP Server (in this case Support Cluster)
- Registrar Port and Backup Registrar Port are by default 5060
- In Registration Expires enter 120s or 3 minutes which is enough for registration
- Registration State will be not registered, after SAVE



- Press SAVE, and if you enter the correct data, your Registration status should be REGISTERED

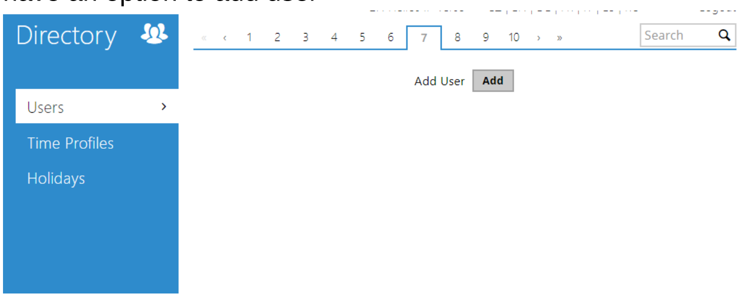
4.2 Adding user on 2n Doorphone:

1. Go to Directory:



2. Adding user

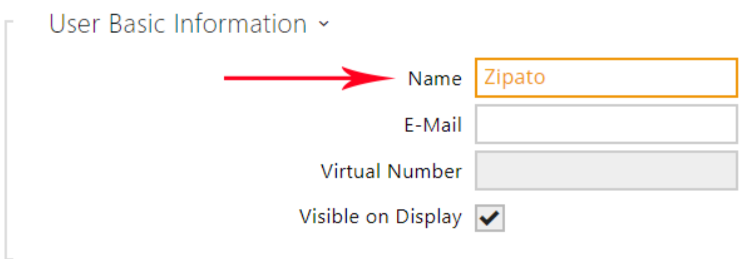
- Under users at top you'll have numbers, and those numbers presents potential users, you have an option to add user



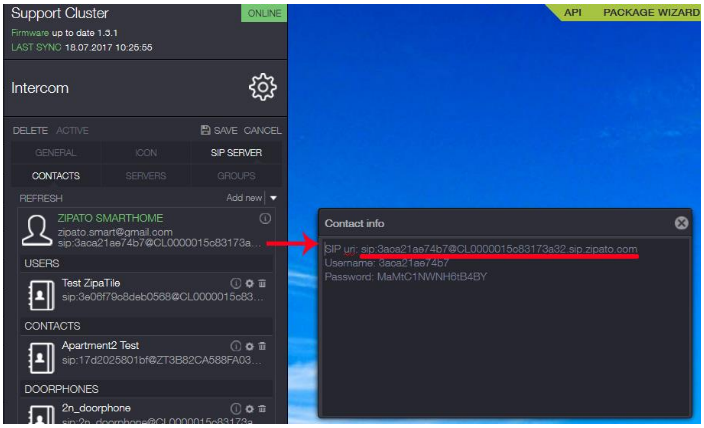
- Add user, and after that you'll have to fill the form in order to register that user

3. Filling the form:

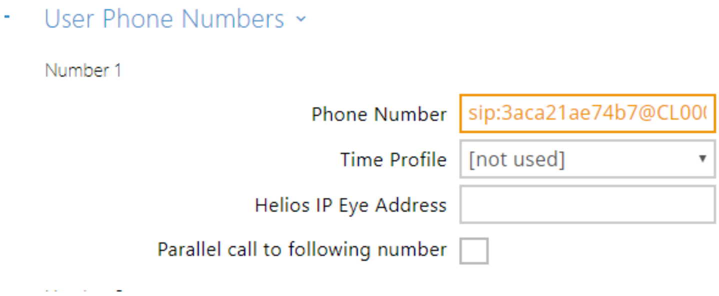
- First enter the name of this user



- After that, add a phone number, the phone number is your SIP uri address



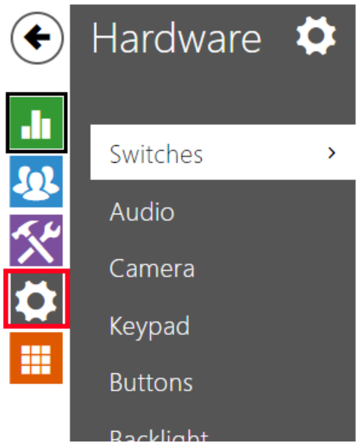
- So it should looks like this:



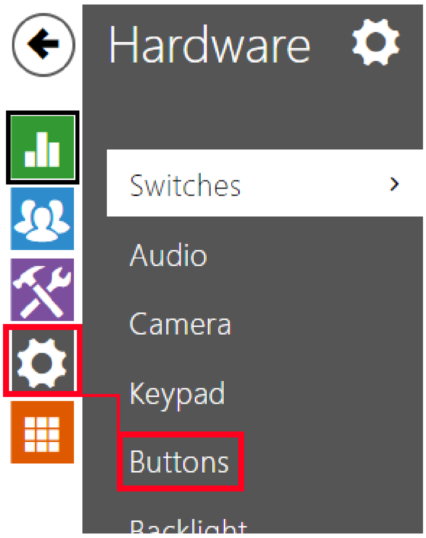
- After you are done, press SAVE (Save)

4.3 Binding user with button:

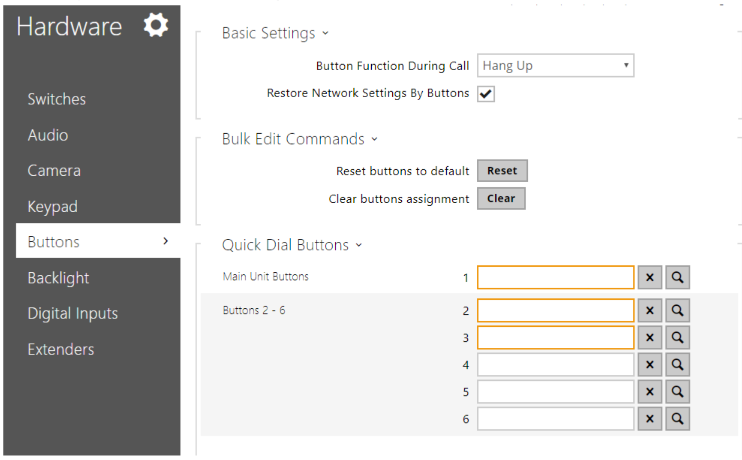
1. Go to Hardware



- And enter Buttons

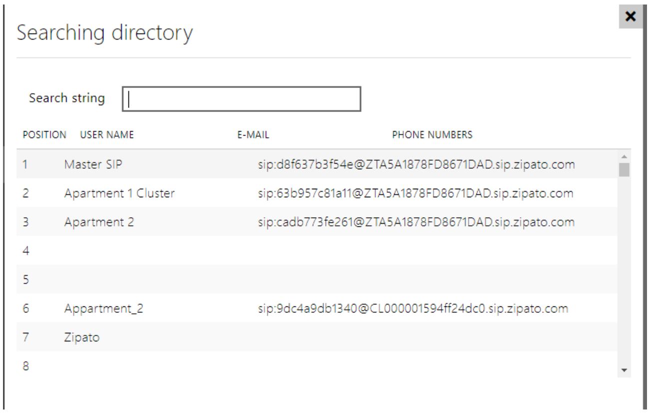


- In buttons option you have to choose which Dial button will be for which user, since we already created user Zipato we will add it

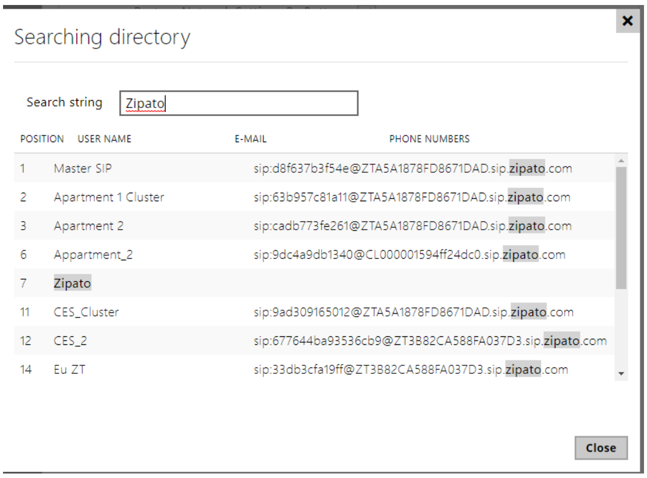


- In Quick Dial Buttons you have to choose which user will be applied on doorphone button, to do so press on search icon

- New window will be opened and you can choose between 2 types of search, first one is straight forward with all names showed, and the second one is search by words



- If you search by words enter the few letters and you will get the search results only with that letters



- When you are done with adding buttons to 2N Doorphone, press SAVE (Save) and you can now use 2N Doorphone for calling ZipaTile

5. How to add AVA Design DP104

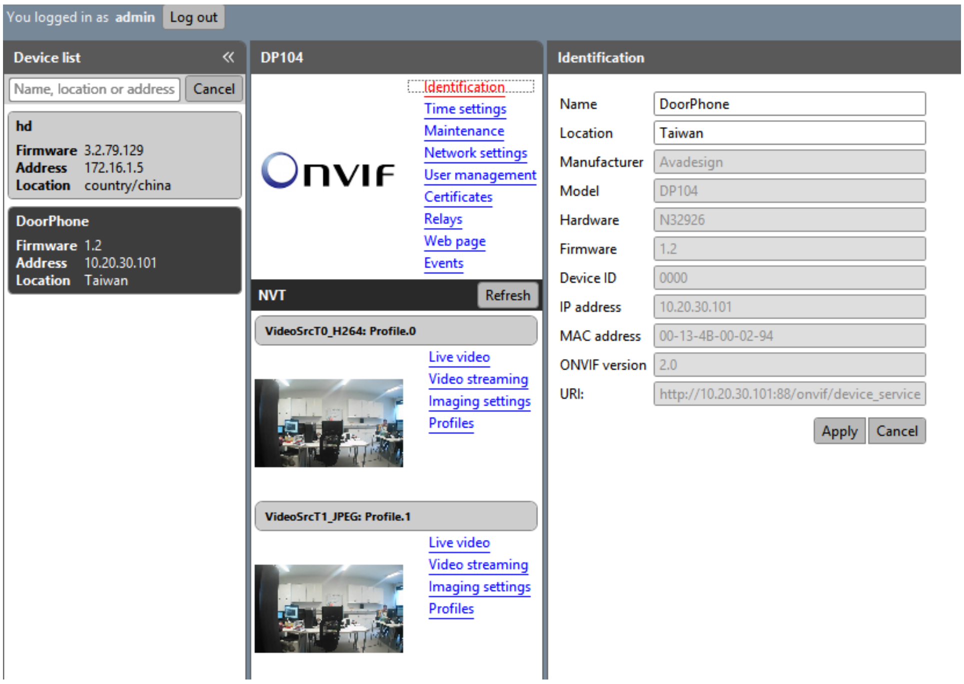
5.1 Installation

After connecting DoorPhone to Ethernet network and powering device, please run ONVIF Device Manager (https://sourceforge.net/projects/onvifdm/)

Run ONVIF Device manager and log in with credentials:

- Username: admin
- Password: admin

Scan network to get doorphone IP (your PC should be connected to the same network as DoorPhone)

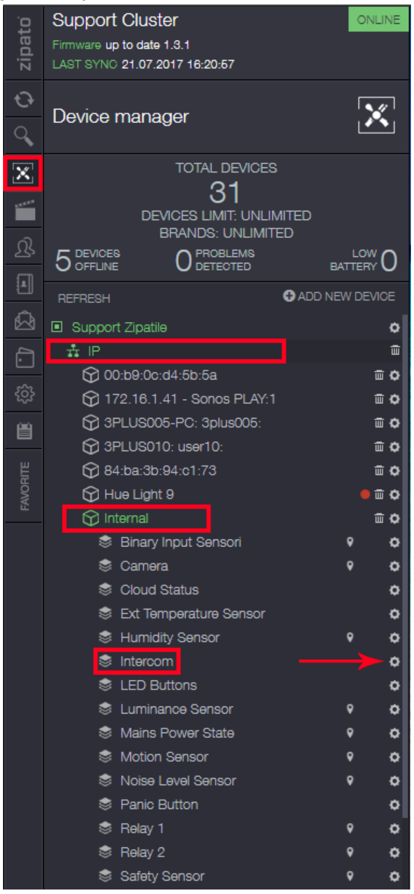


- After you found the IP address, please open that address in your web browser to get to the DoorPhone web interface

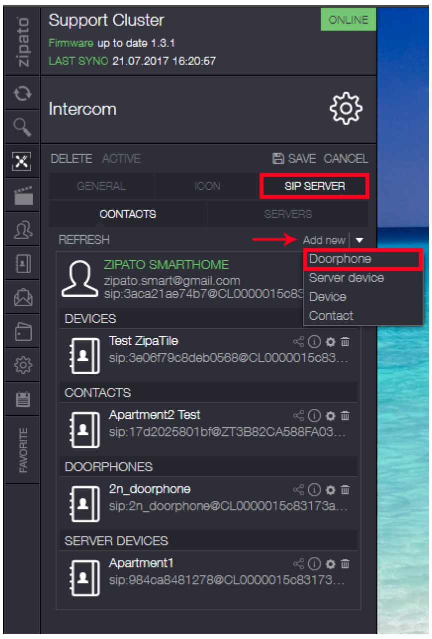
5.2 Creating Ava Doorphone in Zipato Web UI

First, enter the Zipato Web UI (https://my.zipato.com/zipato-web/app2login) and check SIP Server configuration

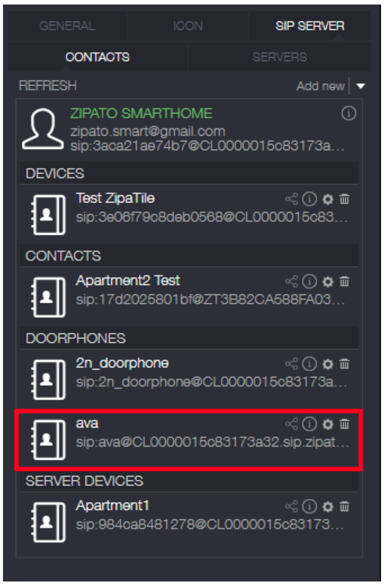
- Go to Device manager, enter your ZipaTile select IP, find Intercom and enter its settings



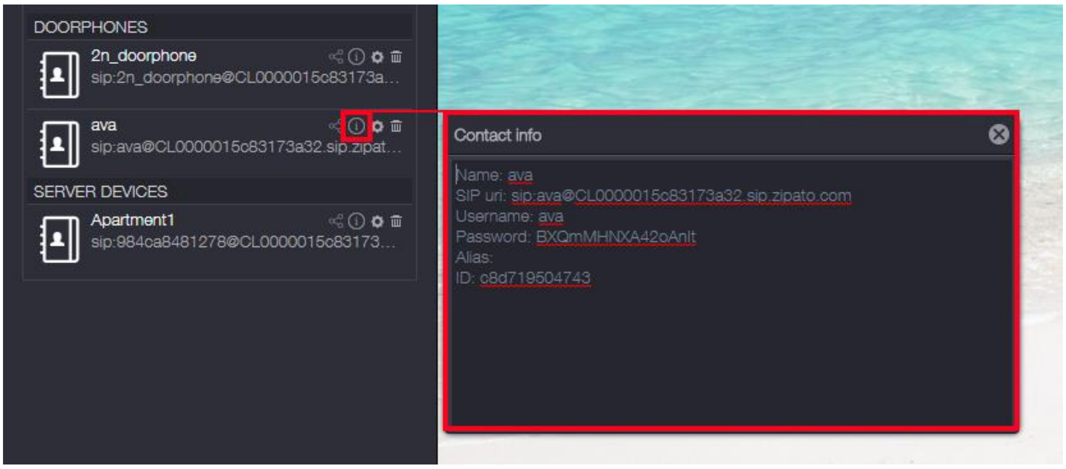
Select SIP Server, and add new Doorphone, press on add new, and enter the name of DoorPhone:



- We created a new doorphone:



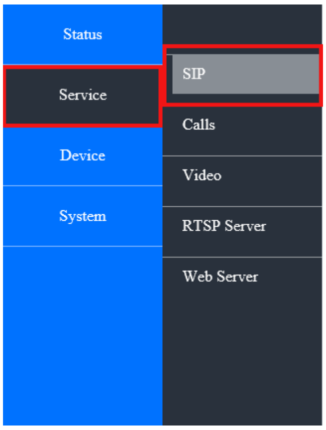
- The important thing is to check the contact info on your DoorPhone, and copy that info into DP-104 configuration



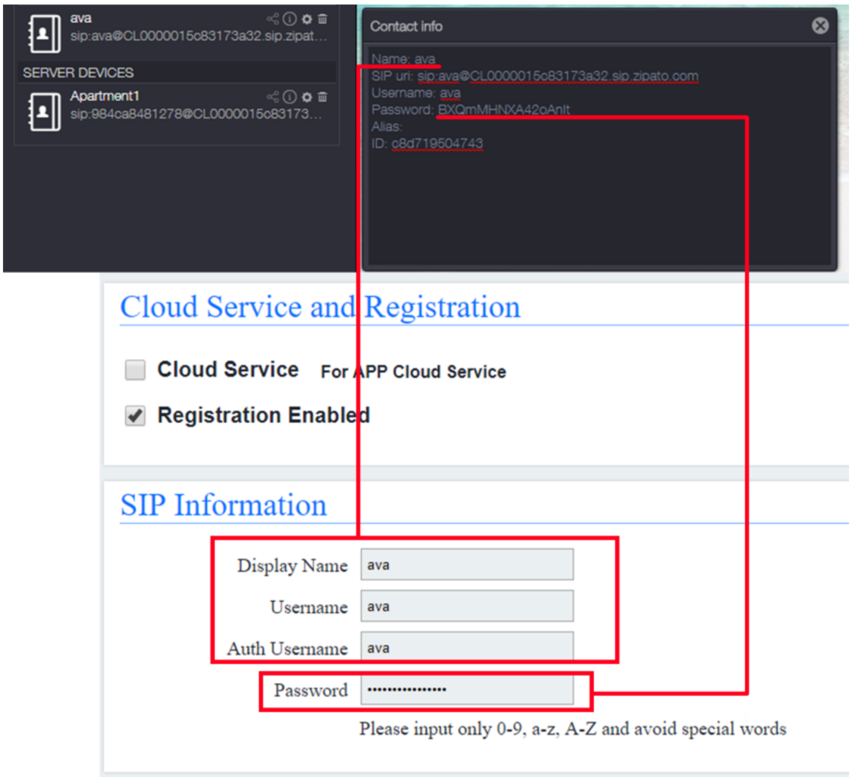
5.3 Ava DP-104 configuration

Enter the DoorPhone configuration in web browser as described in installation.

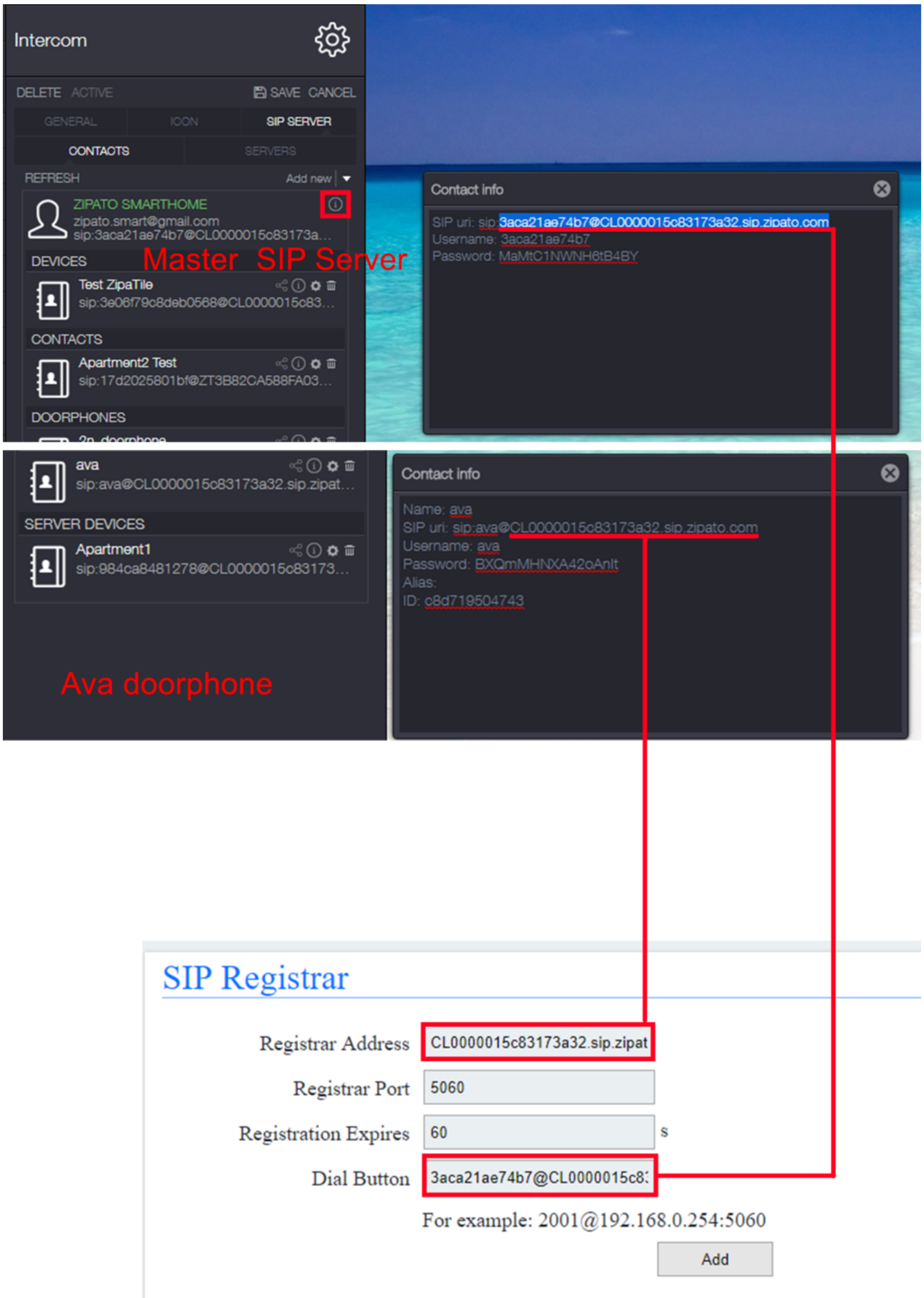
- First go to Service → SIP, and after that enter the credentials and other information as described



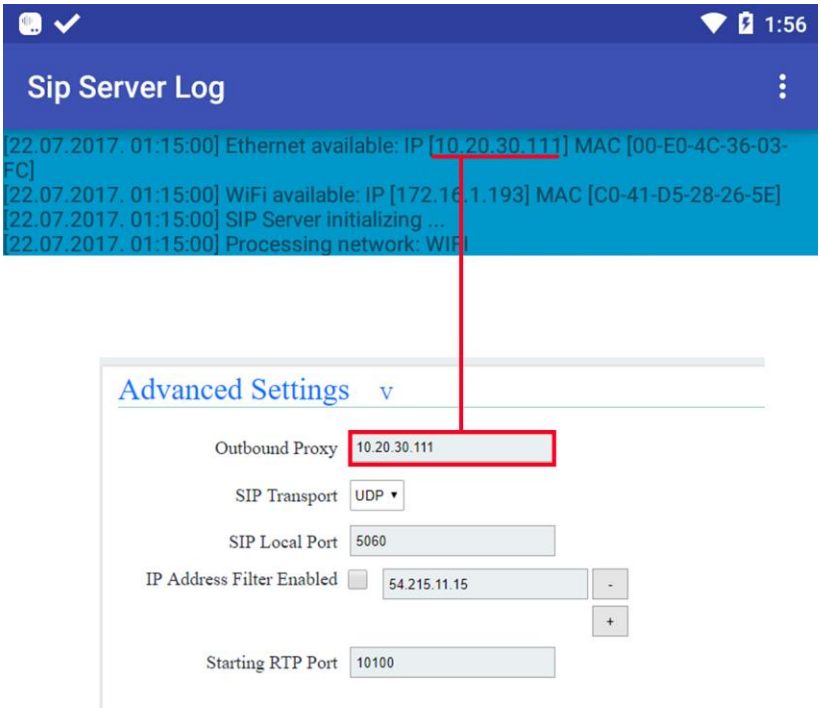
- Cloud Service and Registration, this section have to be checked under **Registration Enabled**
- Enter the credentials in SIP Information as described:
    - Copy name (in this example) ava to Display name, Username and Auth Username, all should be the same
    - Copy password from ava doorphone in Web UI to SIP Information (Password)



- SIP Registrar:
    - Open up the master sip server, there you also created ava doorphone
    - Registrar Address: copy Ava's SIP uri: after the @
    - Registrar Port: default port 5060
    - Registration Expires: default is 60s
    - Dial Button: this is what will Ava call when the call button is pressed, we added the main user (cluster user) as a contact, so if you have 2 ZT in cluster then both will ring, copy SIP uri after the sip: to dial button

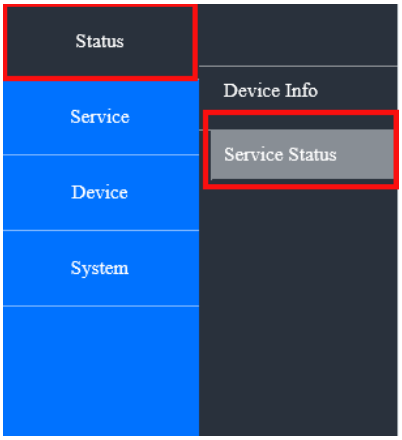


- Advanced settings:
    - In advanced settings we have to enter the Outbound Proxy
    - To find it, just go to the ZipaTile on which you are trying to add Ava
    - Go to Zipato app → open Main Menu →Zipato Settings → Sip Server and there you will find this information, copy the Ethernet IP address

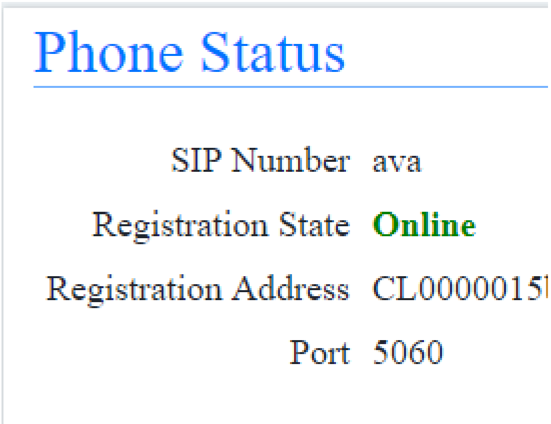


- After you are done, please press SAVE on the bottom right and check if the doorphone is connected

5.4 Checking registration state:



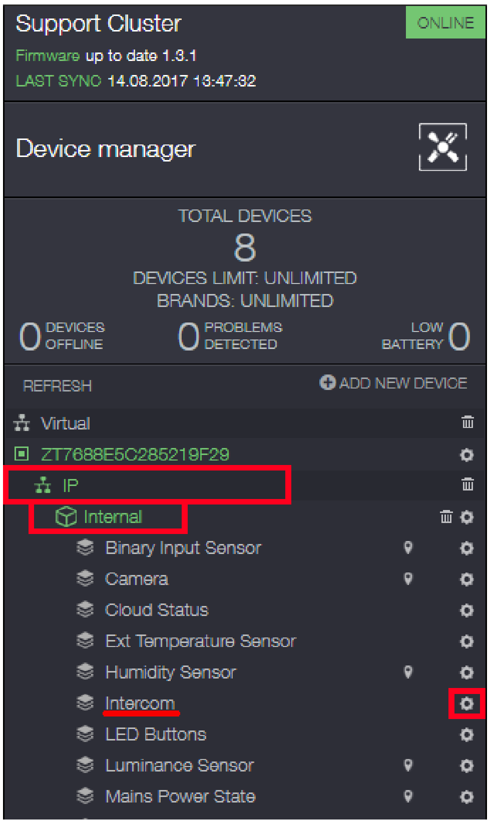
- Under Phone status you will see the Online state, if it's offline, then the configuration is not right, and has to be configured properly



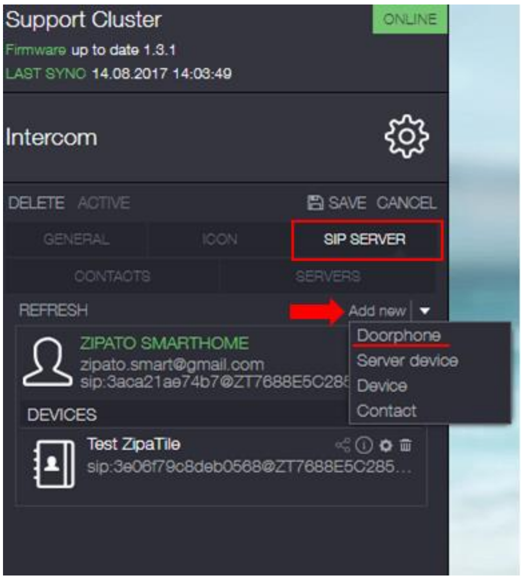
6. How to add Akuvox R27A Doorphone

6.1 Adding Doorphone to Building SIP server

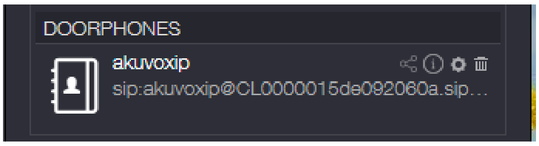
- Go to Device manager, enter your ZipaTile select IP, find Intercom and enter its settings



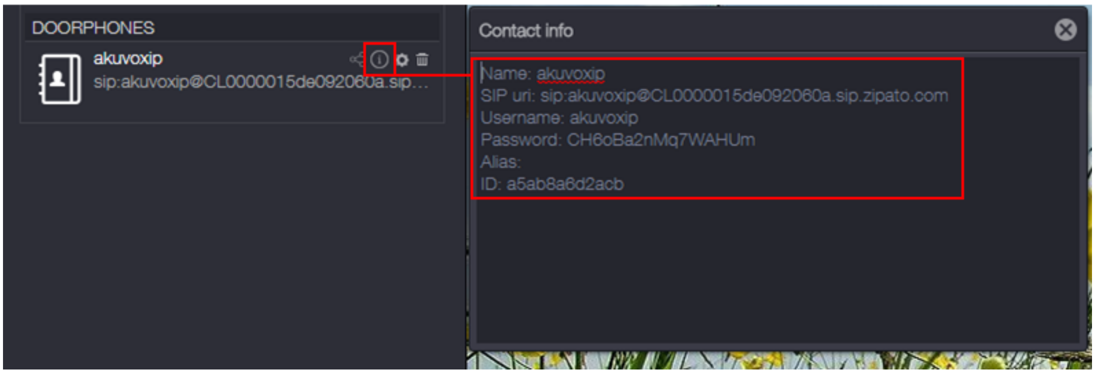
Select SIP Server, and add new Doorphone, press on Add new, and enter the name of DoorPhone:



- We created a new doorphone:



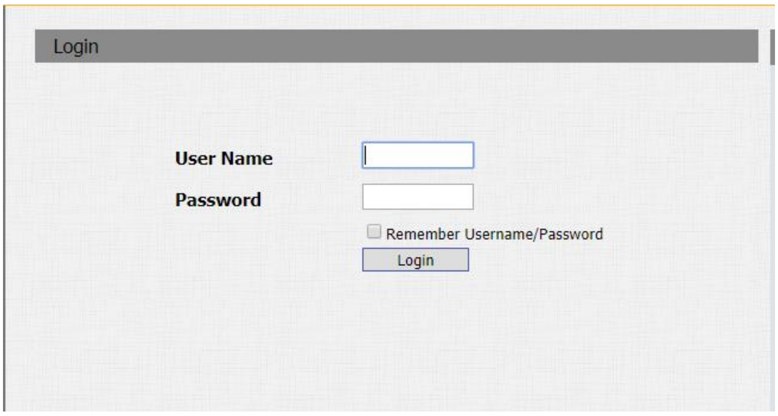
- The important thing is to check the contact info of your DoorPhone, and copy that info into Akuvox R27A configuration



6.2 Configuration of Akuvox R27A

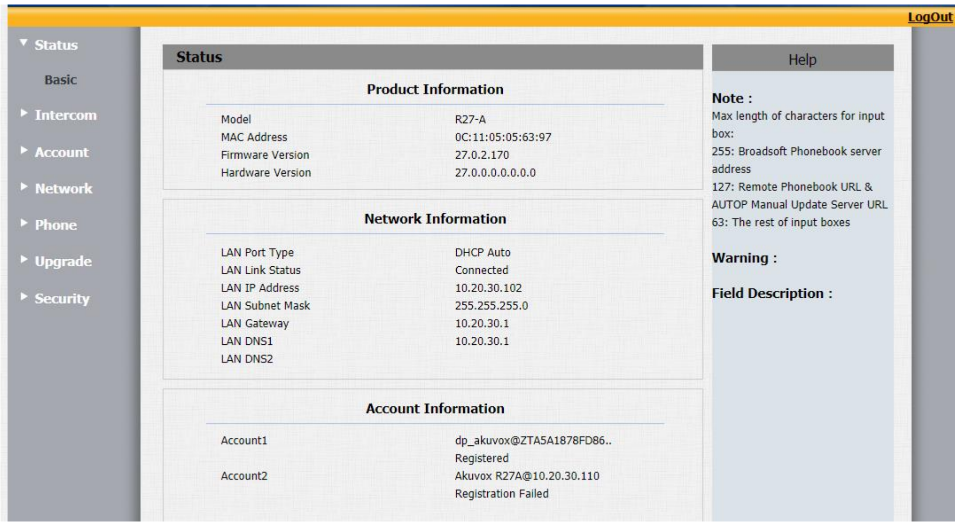
- To find the IP address of Doorphone, press *2396# on dial pad and click (1) System information

Enter that IP address into your web browser, in our example is 10.20.30.102, and log in

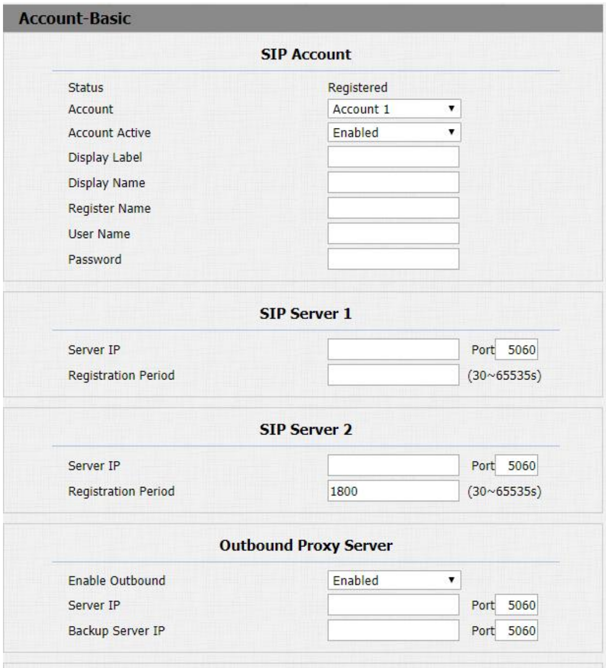


  - Enter User Name and Password, which is admin for both

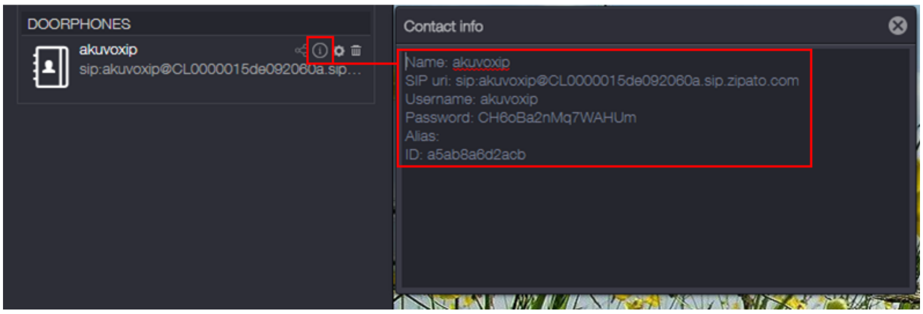
- You get the new window:



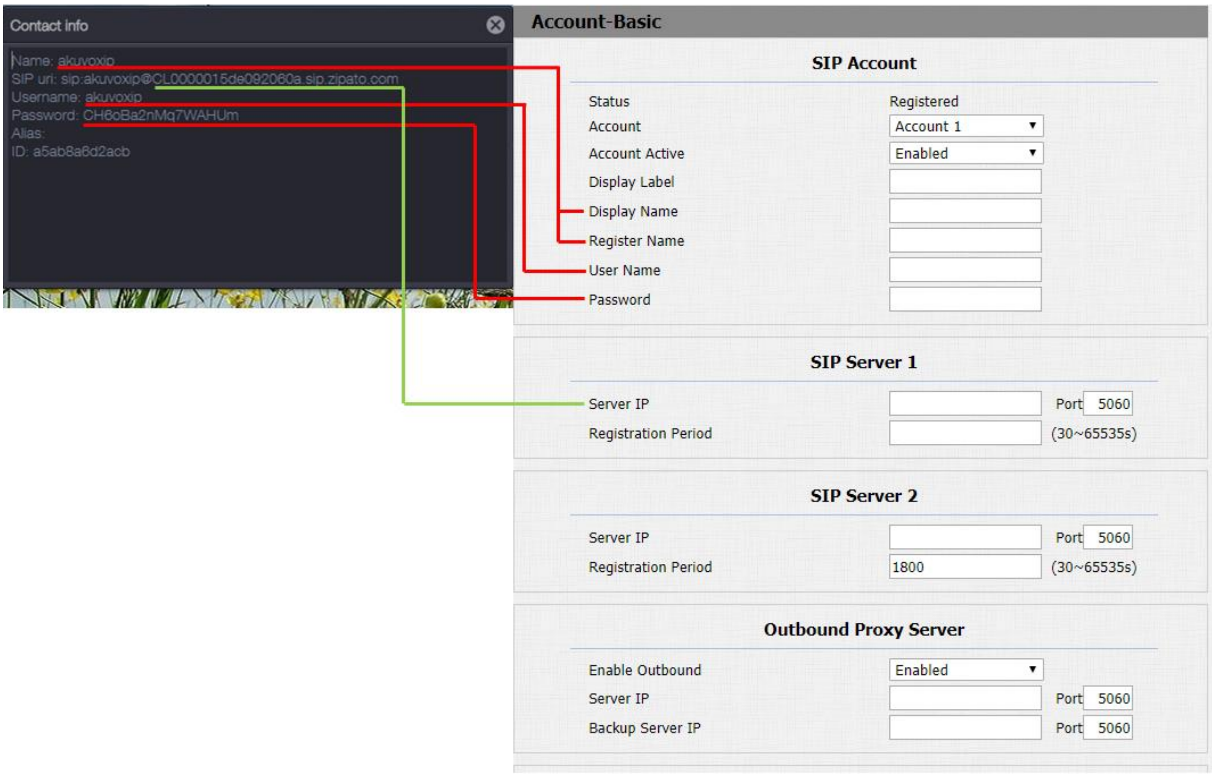
- Enter Account- Basic



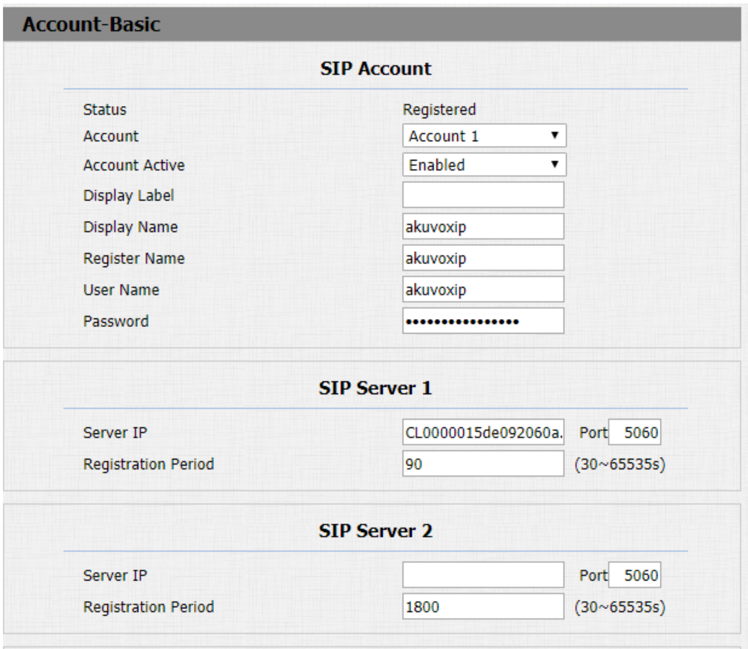
Now we have to configure everything, and to do that we will need contact information of created Doorphone on SIP Server:



This is how you should configure it:

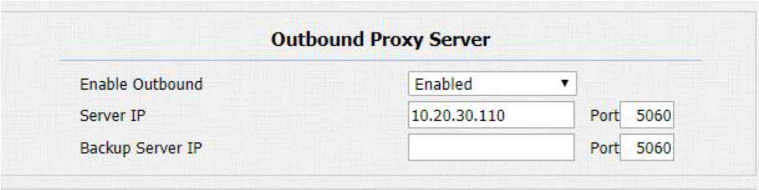


And after you enter everything that you need, you'll have this:



- Under SIP Server 1, the Server IP could start with ZT or CL, but in most cases you'll have SIP Server that is not in cluster

- **Outbound Proxy Server**- this is important, enter the Server IP which means you have to enter SIP Server IP address

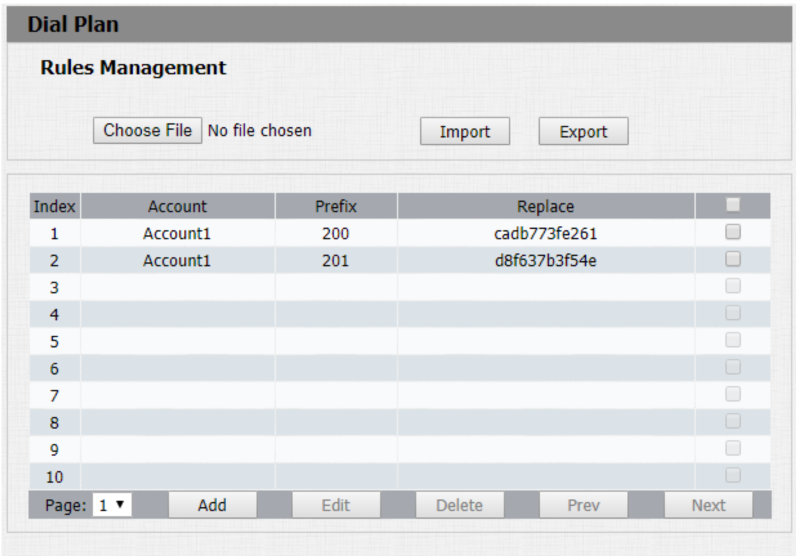


- After you are done, press Submit button, shortly after that your account should be REGISTERED

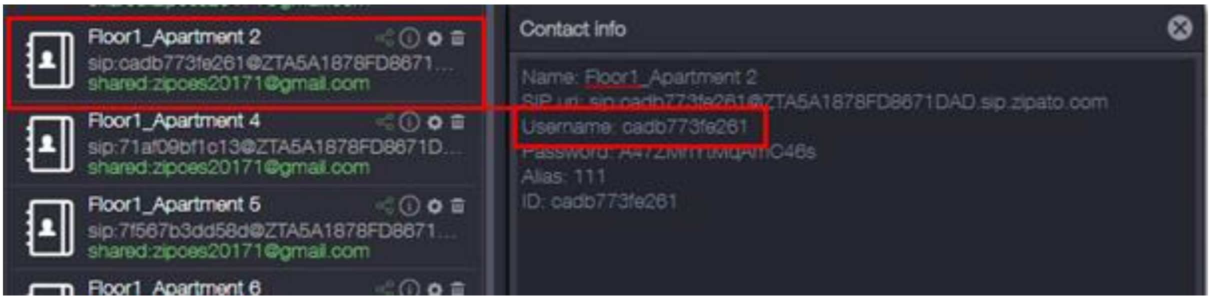
6.3 Associating apartments with doorphone

- To associate apartments to a dial number from doorphone, go to Phone- Dial Plan

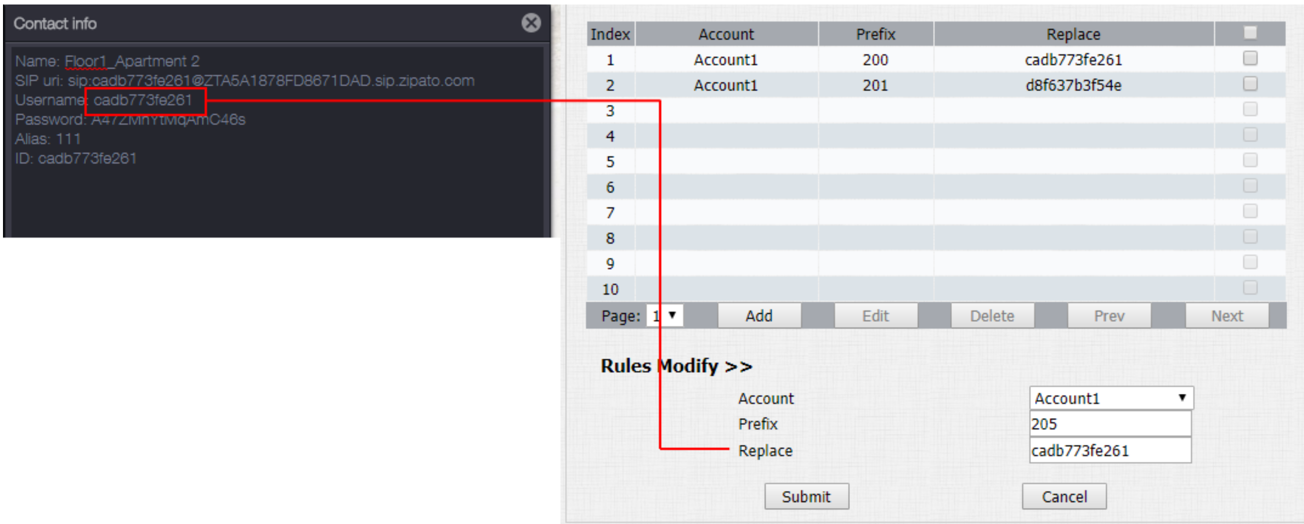
And you will get this window:



- To associate apartment number and dial pad number you will need username from other apartment that you created on Building SIP Server



- Press Add, choose Account- Account1 (which is registered and associated with your Building SIP Server)
- Prefix- is the number on dial pad
- Replace- in this box you have to enter the username of apartment that you created on Building SIP Server

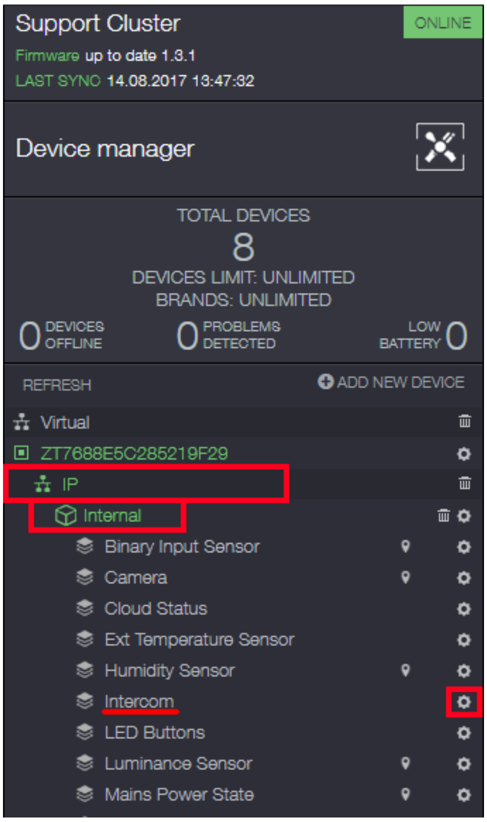


- After you are done, press Submit and now the apartment is configured

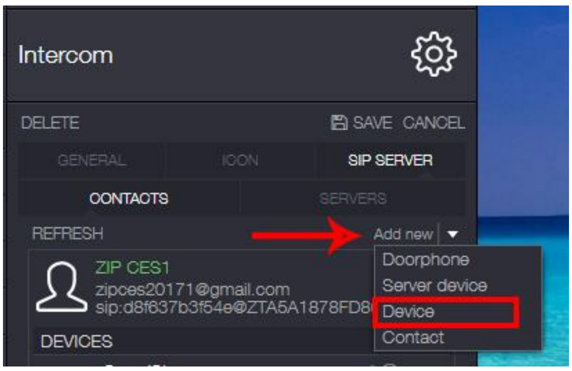
7. GrandStream

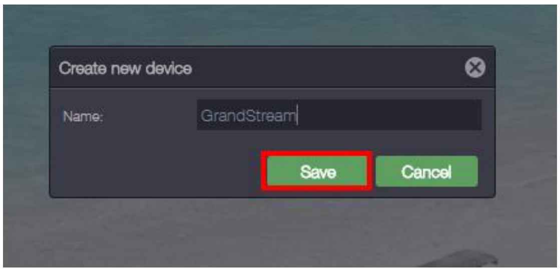
7.1 Adding GrandStream to Building SIP server

- Go to Device manager, enter your ZipaTile select IP, find Intercom and enter its settings

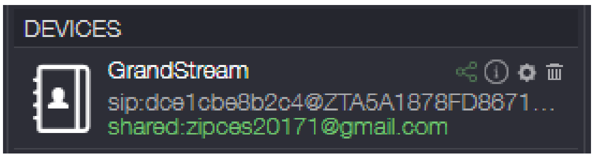


Select SIP Server, and add new Device, press on Add new, and enter the name of device:





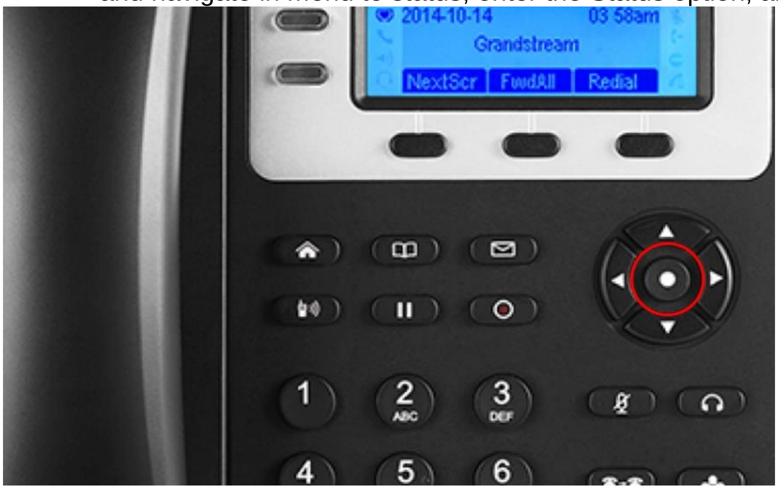
- We created a new device:



- The important thing is to check the contact info of your GrandStream (device), and copy that info into GrandStream configuration

7.2 Configuration of GrandStream

- To find the IP address of Grandstream, press middle button as shown In the picture below, and navigate in menu to status, enter the Status option, and there you will have IP address

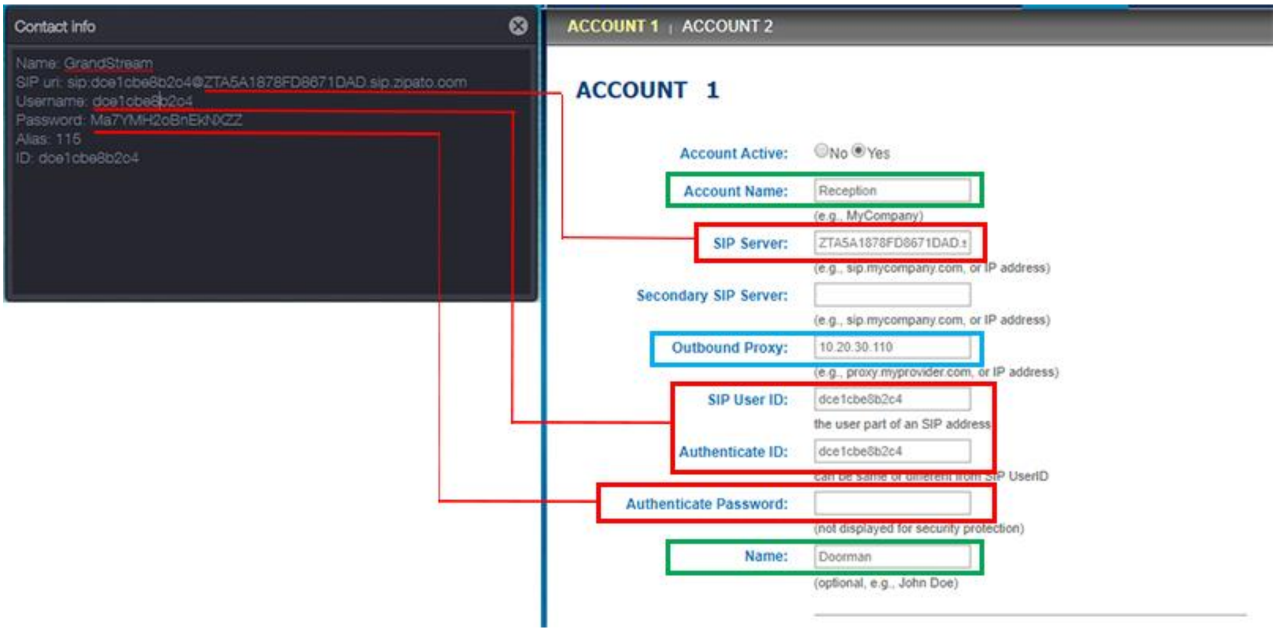


Enter that IP address into your web browser, in our example is 10.20.30.111, and log in

- Enter Password

- You get the new window:

  - Enter Accounts



- Green squares are for names, so the first is Account Name that's what will be displayed on GrandStream, the 2nd can be like in this example.
- Enter sip server address in SIP Server field
- Username will go into SIP User ID and Authenticate Password
- Password will go into Authenticate Password
- Blue square is Outbound Proxy and you will have to enter Building SIP Server address, in this case it's 10.20.30.110

When you are done with the configuration update and reboot GrandStream

8. FAQ:

1. My Intercom is not working, it says disconnect.
- In order to have a working InterCom you have to reconnect it, if that doesn't help, procedure is next: Go to Main menu → Zipato Settings → SIP Server → click on upper right corner and press restart, after that wait for a few seconds until all connections are established
- Go to InterCom and it should work
- Check if you are on the right account, sometimes if you have more than one controller, you may be connected to the wrong one

2. What is ICE protocol?

This **protocol** is called **Interactive Connectivity Establishment** (**ICE**). **ICE** makes use of the Session Traversal Utilities for NAT (STUN) **protocol** and its extension, Traversal Using Relay NAT (TURN). **ICE** can be used by any **protocol** utilizing the offer/answer model, such as the Session Initiation **Protocol** (SIP).

NAT- (**Network Address Translation** or Network Address Translator) is the virtualization of Internet Protocol (IP) addresses. **NAT** helps improve security and decrease the number of IP addresses an organization needs.

3. **What is Intercom?**

An intercom (intercommunication device), talkback or doorphone is a stand-alone voice communications system for use within a building or small collection of buildings, functioning independently of the public telephone network (Azori 2016). Intercoms are generally mounted permanently in buildings and vehicles. Intercoms can incorporate connections to public address loudspeaker systems, walkie talkies, telephones, and to other intercom systems. Some intercom systems incorporate control of devices such as signal lights and door latches.

4. What is SIP in Intercom?

Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify, and terminate multimedia sessions (conferences). A session is considered as an exchange of data between an association of participants, such as Internet telephony calls and video telephony.

5. How does a SIP server work?

SIP (Session Initiation Protocol) is a signaling protocol, widely used for setting up, connecting and disconnecting communication sessions, typically voice or video calls over the Internet. SIP is a standardized protocol with its basis coming from the IP community and in most cases uses UDP or TCP.

There are two types of Internet Protocol (IP) traffic. They are **TCP** or Transmission Control Protocol and **UDP** or User Datagram Protocol. TCP is connection oriented – once a connection is established, data can be sent bidirectional. UDP is a simpler, connectionless Internet protocol.

6. What is SIP Call?

A **SIP** connection is a marketing term for voice over Internet Protocol (VoIP) services offered by many Internet telephony service providers (ITSPs). The service provides routing of telephone **calls** from a client's private branch exchange (PBX) telephone system to the public switched telephone network (PSTN).

7. What is the SIP Server?

A **SIP server** is the main component of an IP PBX, and mainly deals with the management of all **SIP** calls in the network. A **SIP server** is also referred to as a **SIP**Proxy or a Registrar.

8. What is SIP URI?

In other words, a **SIP URI** is a user's **SIP** phone number. The **SIP URI** resembles an e-mail address and is written in the following format: **SIP**-**URI** = **sip**:x@y:Port where x=Username and y=host (domain or IP)

9. Why would you use a SIP?

SIP allows people around the world to communicate using their computers and mobile devices over the Internet. It is an important part of Internet Telephony and allows you to harness the benefits of VoIP (voice over IP) and have a rich communication experience.

10. IP PBX

An IP PBX is a private branch exchange (telephone switching system within an enterprise) that switches calls between VoIP (voice over Internet Protocol or IP) users on local lines while allowing all users to share a certain number of external phone lines. The typical IP PBX can also switch calls between a VoIP user and a traditional telephone user, or between two traditional telephone users in the same way that a conventional PBX does.