



Grandstream Networks, Inc.

2N Helios IP Door System & Grandstream IP Multimedia Phones Configuration Guide



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INTRODUCTION

The purpose of this document is to provide basic configuration guidance for 2N Helios IP Door System & Grandstream IP Multimedia Phones GXV3140/GXV3175v1/v2.

This guide is applicable to the following Grandstream IP Multimedia Phones (GXV3140 with firmware 1.0.7.76 or higher, GXV3175 with firmware 1.0.3.74 or higher and GXV3175v2 with firmware 1.0.1.46 or higher)

Be aware that different firmware revisions may have different web interface formats and functionality.

This guide is describing 3 basic scenarios:

- 1- Peer 2N Helios IP Door System with a GXV3140 or GXV3175 (Single Peer)
- 2- Peer 2N Helios IP Door System with Multiple Grandstream GXV3140 and/or GXV3175 (Multi-Peers)
- 3- 2N Helios IP Door System with Multiple Grandstream GXV3140 and/or GXV3175v1/v2 using a SIP server

TUTORIAL ENVIRONMENT

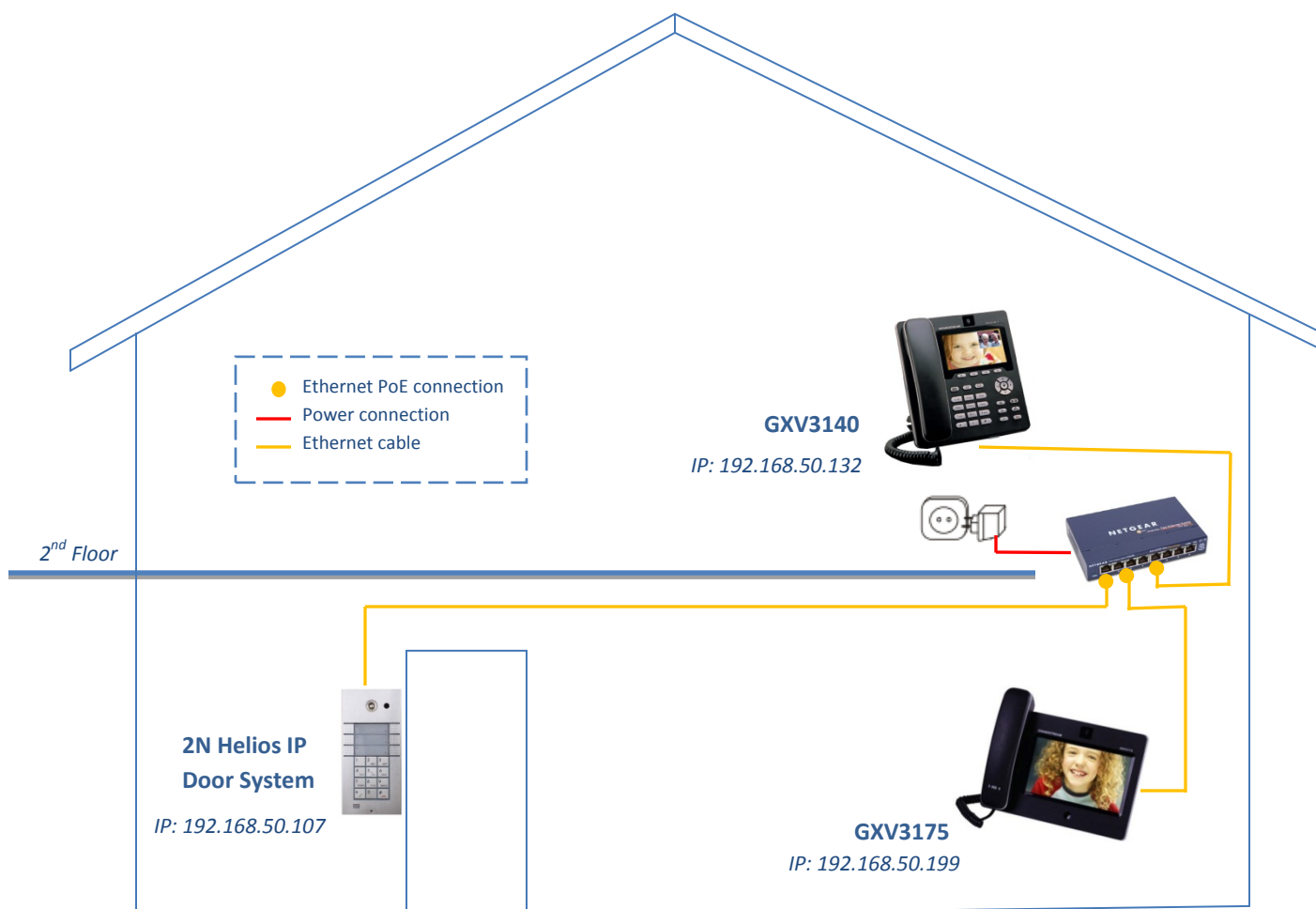
For this tutorial, we will be using Grandstream IP Multimedia Phone GXV3175 with firmware 1.0.3.46 and Grandstream IP Multimedia Phone GXV3140 with firmware 1.0.7.76.

- ✓ GXV3175 IP is 192.168.50.199
- ✓ GXV3140 IP is 192.168.50.132
- ✓ 2N Helios IP is 192.168.50.107

Note: *Make sure that devices are set to use Static IPs; otherwise, the communication cannot be established if one of them changes its IP from original one. (This apply to First and Second scenario only)*

1st Scenario:

PEER THE 2N HELIOS IP DOOR SYSTEM WITH A GXV3140 OR GXV3175V1/V2 (SINGLE PEER)



STEP 1: 2N HELIOS IP DOOR SYSTEM CONFIGURATION

- 1- Access the web interface of your Helios
- 2- Go to **Basic Settings > Phonebook**
- 3- Select a contact which is mapped for one touch call (for example for button #1 on your HeliosIP). You need to modify Phone number for this contact according to your peer telephone. In our example, you can use either GXV3140 with IP 192.168.50.132 or GXV3175 with IP 192.168.50.199
Number value needs to be like **sip: x@192.168.50.132**
(x can be replaced by any letter(s) or digit(s))



The screenshot shows the HeliosIP web interface for configuring a contact in the phonebook. The contact is labeled '1'. The configuration is divided into several sections:

- General settings:** Position enabled (Yes), Position name (Mr. Test), E-Mail.
- User activation & deactivation:** Activation code, Deactivation code, User current state (Active).
- Phone numbers:** Number 1 (sip:1@192.168.50.132), Time profile ([not used]), Station name. Similar fields are present for Number 2 and Number 3.
- User switch codes:** Switch 1 code, Switch 2 code.
- Card reader:** User card ID.

- 4- Press **Save** to apply settings
- 5- Go to **Advanced Settings > SIP Settings** and set the following settings as shown in next figure.

IP addresses should match your network plan (in our example: 192.168.50.x)

SIP registration > Enable registration should be set to **No**

User ID/Auth ID/Password can be set to any extension

Keep **Other settings** to default (refer to above figure)

6- Press **Save** to apply settings



The screenshot shows the HeliosSIP SIP Settings web interface. The page title is "SIP Settings" and the user is logged in as "2N". The interface is divided into several sections:

- User settings:**
 - Display name: 2N
 - User ID: 200
 - Domain: 192.168.50.192
 - Use auth ID: No
 - Auth ID: 200
 - Password:
- SIP proxy settings:**
 - Proxy address: 192.168.50.192
 - Proxy port: 5060
- SIP registration:**
 - Enable registration: No
 - Registration expires: 120 s
 - Registrar address: 192.168.50.192
 - Registrar port: 5060
- Other settings:**
 - Local SIP port: 5060
 - Send keepalive packets: No
 - Starting RTP port: 5000
 - RTP Timeout: 2 s

The left sidebar contains a navigation menu with the following items: Information, Basic Settings, Advanced Settings (selected), Network, Date and Time, SIP Settings, Web Server, Mic & Speaker, Camera, Audio Codecs, Video Codecs, Streaming, Auto Updates, Display, System log, E-mail, Multicast, Miscellaneous, Card reader, Tools, and Logout.

STEP 2: GRANDSTREAM GXV3140 OR GXV3175V1/V2 CONFIGURATION

1. Access to the web interface of your IP Multimedia Phone by entering the IP of the IP Multimedia Phone on your browser.
http://<IP_Multimedia_Phone_IP> (Default; username: admin, password: admin)
(i.e. <http://192.168.50.132>)

2. Go to **Advanced Settings > General Settings** and uncheck **Use Random Port** (Default is Yes).

Use Random Port : <input type="checkbox"/> Yes
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3. Click **Save** on the bottom of the page and then apply your settings by clicking **Apply** on the top of the page.
4. Go to **Account 1** and enter the following :
 - a. **Account Active** : Yes (by enabling the check box)
 - b. **Account Name** : Any name (for example : Door Phone1)
 - c. **SIP Server** : Enter the IP of 2N Helios System (in our example : 192.168.50.107)
 - d. **SIP User ID** : Any extension number (for example : 100)
 - e. **Authenticate ID** : Same as SIP User ID (for example : 100)
 - f. **Authenticate Password** : Any password, can be same as SIP User ID (for example : 100)
 - g. **Name** : Any name (for example : Door Phone1)

General Settings

* Account Active : Yes

* Account Name :

* SIP Server :

* SIP User ID :

* Authenticate ID :

* Authenticate Password :

Voice Mail UserID :

* Name :

* User ID is phone number : Yes

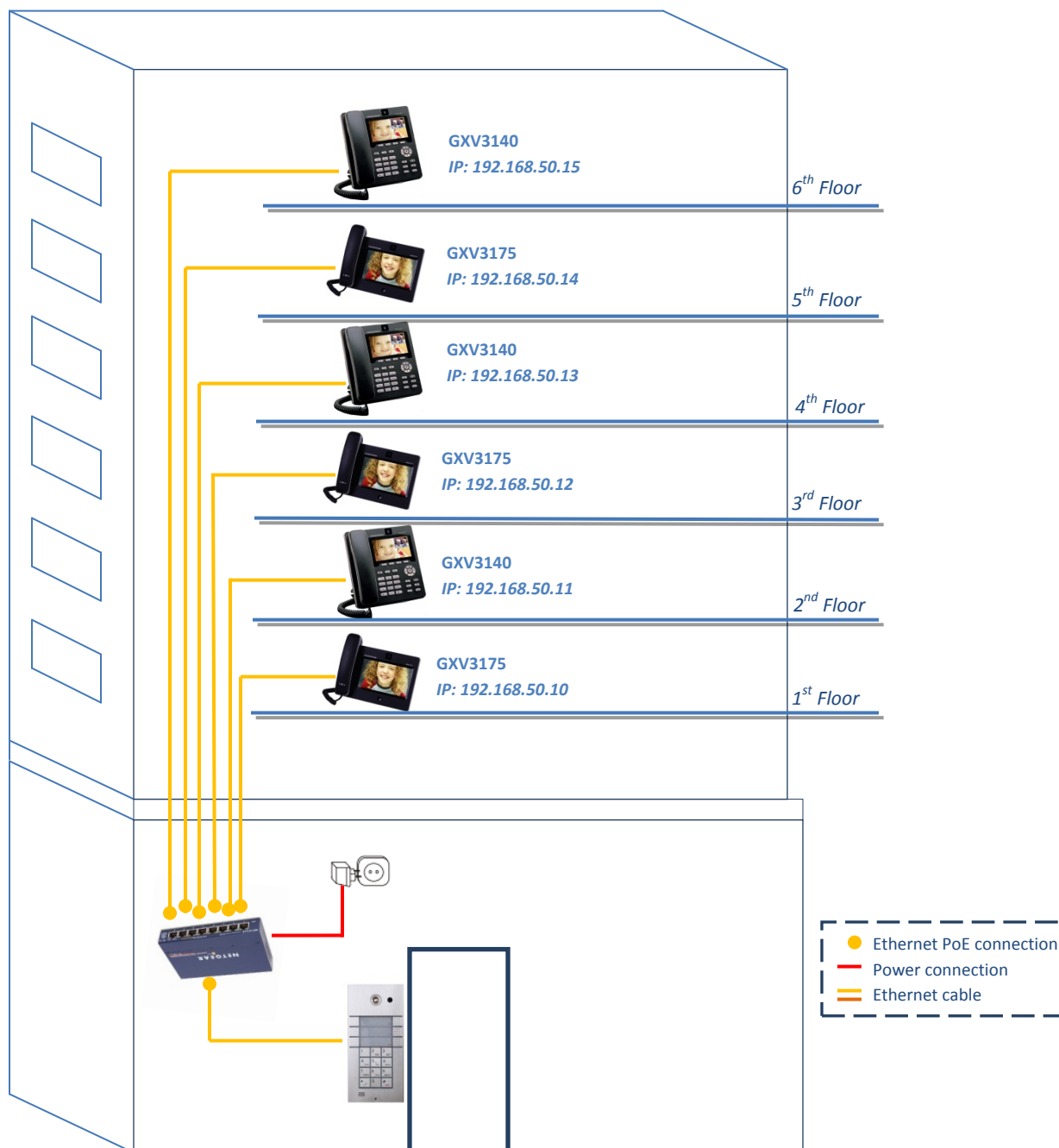
5. Repeat Step 3 to save and apply your settings.
6. Go to **Account 1 > SIP Settings** and set the following :
 - a. **SIP Registration** : No (Disable the check box)
 - b. **Unregister On Reboot** : Yes (Default is Yes)
 - c. **Local SIP port** : 5060 (Default is 5060)
7. Go to **Account 1 > Network Settings**, set **NAT Traversal** to **NAT NO** and click **Save**.

NAT Traversal : ▼

8. Go to **Account 1 > Call Settings** and set the following:
 - a. **Start Video Automatically**: Yes (Enable check box)
 - b. **Remote Video Request**: Select "Accept" from the dropdown list

2nd Scenario:

PEER THE 2N HELIOS IP DOOR SYSTEM WITH MULTIPLE GRANDSTREAM GXV3140 AND/OR GXV3175V1/V2 (MULTI-PEERS)



STEP 1: IP HELIOS DOOR SYTEM CONFIGURATION

Please refer to “*Step 1: IP Helios Door System Configuration*” described in **First scenario**, the steps are the same, except for 3rd step.

The **Phonebook entry number / Position enabled / Position Name / Phone Numbers (Number 1)** need to be adjusted to match the actual scenario. You can refer to the next table for an example of configuration.

Phone	Phonebook entry number	Position enabled	Position Name	Phone Numbers (Number 1)
Floor1	1	Yes	Floor1	sip:100@192.168.50.10
Floor2	2	Yes	Floor2	sip:101@192.168.50.11
Floor3	3	Yes	Floor3	sip:102@192.168.50.12
Floor4	4	Yes	Floor4	sip:103@192.168.50.13
Floor5	5	Yes	Floor5	sip:104@192.168.50.14
Floor6	6	Yes	Floor6	sip:105@192.168.50.15

STEP 2: GRANDSTREAM GXV3140/GXV3175V1/V2 CONFIGURATION

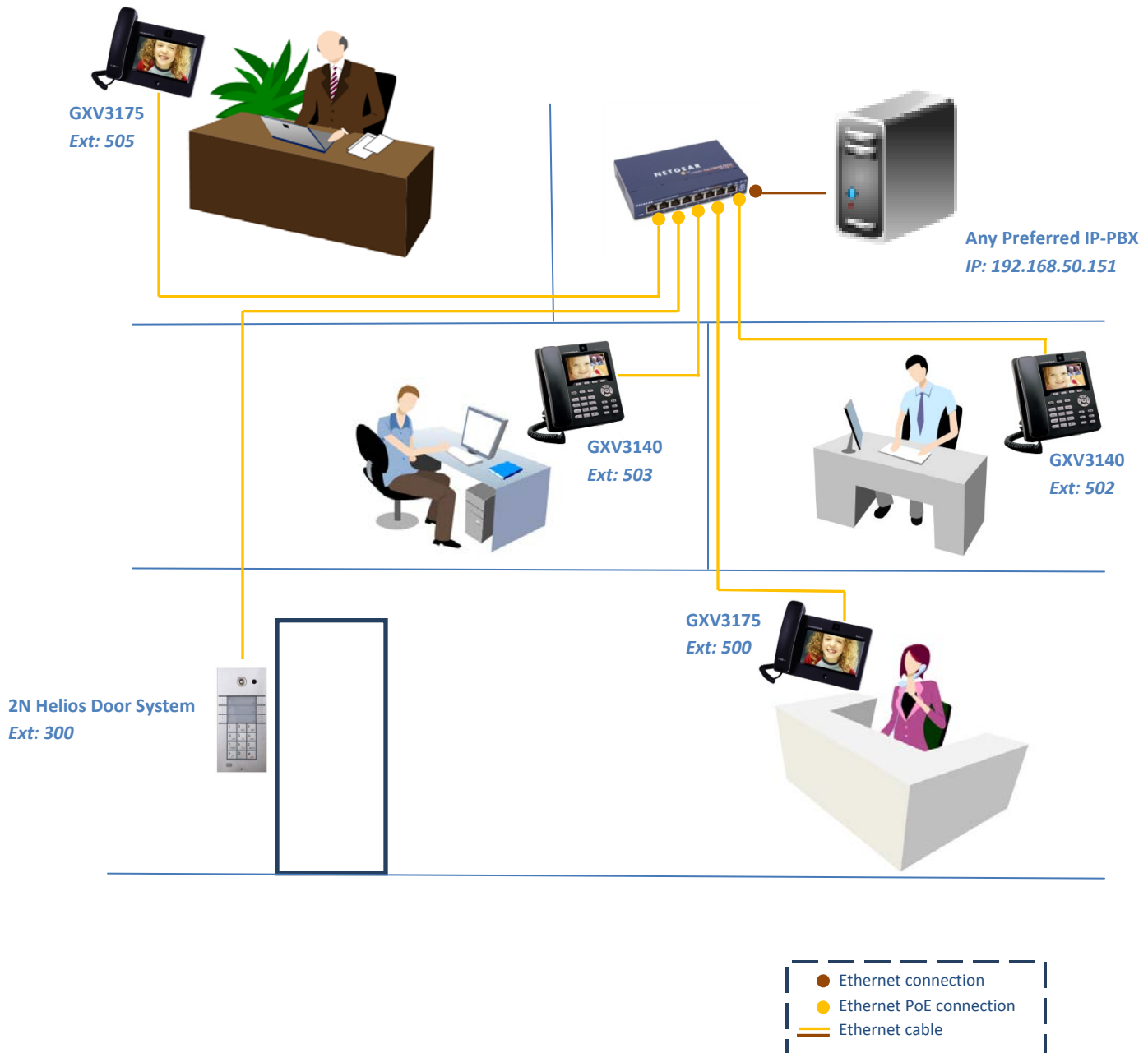
Please refer to “*Step 2: GXV3140/GXV3175 Configuration*” described in **First scenario**, the steps are the same and should be done on each phone.

The **Account Name / SIP User ID / Authenticate ID / Authenticate Password / Name** need to be different on each phone. You can refer to the next table for an example of configuration.

Phone/IP	Account Active	Account Name	SIP Server	SIP User ID	Auth. ID	Authenticate Password	Name
Phone1 192.168.50.10	Yes	Floor1	192.168.50.107	100	100	100	Floor1
Phone2 192.168.50.11	Yes	Floor2	192.168.50.107	101	101	101	Floor2
Phone3 192.168.50.12	Yes	Floor3	192.168.50.107	102	102	102	Floor3
Phone4 192.168.50.13	Yes	Floor4	192.168.50.107	103	103	103	Floor4
Phone5 192.168.50.14	Yes	Floor5	192.168.50.107	104	104	104	Floor5
Phone6 192.168.50.15	Yes	Floor6	192.168.50.107	105	105	105	Floor6

3rd Scenario:

2N HELIOS IP DOOR SYSTEM WITH MULTIPLE GRANDSTREAM GXV3140 AND/OR GXV3175 USING A SIP SERVER



STEP 1: IP HELIOS DOOR SYSTEM CONFIGURATION

- 1- Access the web interface of your Helios
- 2- Go to **Basic Settings > Phonebook**
- 3- Select a contact which is mapped for one touch call (for example for button #1 on your HeliosIP). You need to modify Phone number for this contact according to an extension registered on a Grandstream Multimedia IP Phone.
In our example, you can use either a GXV3140 or a GXV3175 with an extension registered to your SIP server (example: 501)
Number value needs to be like **501**



The screenshot shows the HeliosIP web interface for configuring a phone book entry. The interface includes a sidebar with navigation options and a main content area with several configuration sections.

HeliosIP (2N TELECOMMUNICATIONS)

Phone book

Information

Basic Settings

- Phone book
- Profiles
- Switch 1
- Switch 2

Advanced Settings

Card reader

Tools

Logout

General settings

Position enabled: Yes

Position name: Phone1

E-Mail:

Phone numbers

Number 1: 500

Time profile: [not used]

Station name:

Number 2:

Time profile: [not used]

Station name:

Number 3:

Time profile: [not used]

Station name:

Substitute if inaccessible: [none]

User activation & deactivation

Activation code:

Deactivation code:

User current state: Active

User switch codes

Switch 1 code:

Switch 2 code:

Card reader

User card ID:

- 4- Press **Save** to apply settings

Note: Repeat steps 4 & 5, in other Phonebook Entries if you need to configure other phones specifying the extension of each in "Number 1" field.

- 5- Go to **Advanced Settings > SIP Settings** and set the following settings as shown in next figure.



The screenshot shows the HelioSIP SIP Settings configuration page. The page has a dark theme with a sidebar on the left containing navigation options: Information, Basic Settings, Advanced Settings (with sub-items like Network, Date and Time, SIP Settings, Web Server, Mic & Speaker, Camera, Audio Codecs, Video Codecs, Streaming, Auto Updates, Display, System log, E-mail, Multicast, Miscellaneous), Card reader, Tools, and Logout. The main content area is titled 'SIP Settings' and is divided into three sections:

- User settings:**
 - Display name: 2N
 - User ID: 300
 - Domain: 192.168.50.151
 - Use auth ID: Yes (dropdown)
 - Auth ID: 300
 - Password:
- SIP proxy settings:**
 - Proxy address: 192.168.50.151
 - Proxy port: 5060
- SIP registration:**
 - Enable registration: Yes (dropdown)
 - Registration expires: 120 s
 - Registrar address: 192.168.50.151
 - Registrar port: 5060
- Other settings:**
 - Local SIP port: 5060
 - Send keepalive packets: No (dropdown)
 - Starting RTP port: 5000
 - RTP Timeout: 2 s

Domain / Proxy address / Registrar address need to be set to the IP or FQDN of your SIP server (in our example: 192.168.50.151)

SIP registration > Enable registration should be set to **Yes**

User ID/Auth ID/Password should be entered as configured on the SIP server (in our example: 300 / 300 / password)

Keep **Other settings** to default (refer to above figure)

- 6- Press **Save** to apply settings.
- 7- You can check if the registration was successful from **Information** page

STEP 2: GRANDSTREAM GXV3140 OR GXV3175V1/V2 CONFIGURATION


1. Access to the web interface of your IP Multimedia Phone by entering the IP of the IP Multimedia Phone on your browser.
http://<IP_Multimedia_Phone_IP> (Default; username: admin, password: admin)
(i.e. <http://192.168.50.132>)
2. Go to **Account 1** and enter the following :
 - a. **Account Active** : Yes (by enabling the check box)
 - b. **Account Name** : Any name (for example : Door Phone1)
 - c. **SIP Server** : Enter the IP of the SIP Server (in our example : 192.168.50.151)
 - d. **SIP User ID** : Enter SIP User ID as configured in SIP Server (for example : 500)
 - e. **Authenticate ID** : Enter Auth. ID as configured in SIP Server (for example : 500)
 - f. **Authenticate Password** : Enter Auth. Password as configured in SIP Server (for example : Grandstream)
 - g. **Name** : Any name (for example : Door Phone1)

General Settings

Account Active :	<input checked="" type="checkbox"/> Yes
Account Name :	<input type="text" value="500"/>
SIP Server :	<input type="text" value="192.168.50.151"/>
SIP User ID :	<input type="text" value="500"/>
Authenticate ID :	<input type="text" value="500"/>
Authenticate Password :	<input type="password" value="..."/>
Voice Mail UserID :	<input type="text" value="*26"/>
Name :	<input type="text" value="500"/>
Tel URI :	<input type="text" value="User=Phone"/>

3. Repeat **Save** and **Apply** to save and apply your settings.

4. Go to **Account 1 > SIP Settings** and set the following :
 - c. **SIP Registration** : Yes (Enable the check box)
 - d. **Unregister On Reboot** : Yes (Default is Yes)
 - e. **Local SIP port** : 5060 (Default is 5060)
5. Go to **Account 1 > Network Settings**, set **NAT Traversal** to **NAT NO** and click **Save**.

NAT Traversal :	NAT NO	
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6. Go to **Account 1 > Call Settings** and set the following:
 - f. **Start Video Automatically**: Yes (Enable check box)
 - g. **Remote Video Request**: Select "Accept" from the dropdown list