

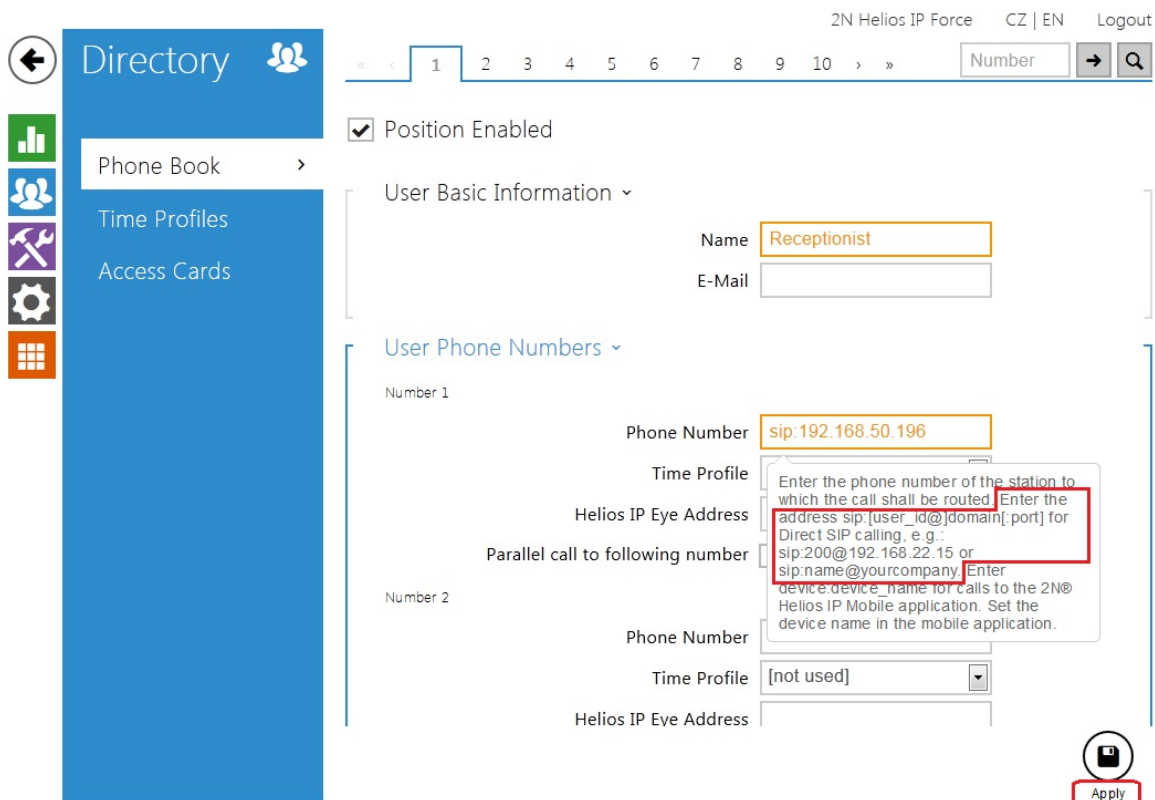
# Grandstream GXV3175v2 - How to configure it with 2N® Helios IP?

This guide describes basic steps for configuration of peer to peer communication between 2N® Helios IP intercom and IP phone Grandstream GXV3175v2. There is also a possibility to register both these devices to the IP PBX (SIP proxy server) and use internal dialling plan for calling between each other but it is not described in this FAQ. Grandstream GXV3175v2 is IP video phone with 7 touch screen colour LCD, integrated Wi-Fi and supporting many different features, audio codecs and video codecs (H.264, H.263 and H.263+). Very useful feature in GXV3175v2 is video preview - it means that you can see the video before you pick up the call. It is the easiest way how you can check who is standing in front of your door before you start to talk with him.

**Note:** Grandstream GXV3175v2 IP video phone is placed in the same LAN (local network) as intercom. IP address of GXV3175v2 is 192.168.50.196 and IP address of 2N® Helios IP intercom is 192.168.50.199. These IP addresses and names are used only such as example - please change it according to your names and network plan.

## How to set 2N® Helios IP intercom?

Settings of 2N® Helios IP intercom is very easy. First of all create a new user and assign him a phone number - we will make SIP direct call (peer-to-peer call) to GXV3175v2 in our scenario. Therefore the number is set in the format: sip:IP\_address (or sip:x@IP\_address). See the picture below for more information.



2N Helios IP Force CZ | EN Logout

« < 1 2 3 4 5 6 7 8 9 10 > » Number → 🔍

Position Enabled

User Basic Information ▾

Name

E-Mail

User Phone Numbers ▾

Number 1

Phone Number

Time Profile

Helios IP Eye Address

Parallel call to following number

Number 2

Phone Number

Time Profile

Helios IP Eye Address

Enter the phone number of the station to which the call shall be routed. Enter the address sip:[user\_id@[domain[:port]]] for Direct SIP calling, e.g.: sip:200@192.168.22.15 or sip:name@yourcompany. Enter device:device\_name for calls to the 2N® Helios IP Mobile application. Set the device name in the mobile application.

In the next step we will move to the section "Services - Phone - SIP" and fill in the IP address of 2N® Helios IP into the "Domain" field.

Services

Phone

Streaming

Onvif

E-Mail

Automation

User Sounds

Web Server

Audio Test

SIP

Intercom Identity

Display Name: 2N Helios IP Force

Phone Number (ID): 111

Domain: 192.168.50.199

Authentication

Use Authentication ID:

Authentication ID: [Greyed out]

Password: [Masked]

SIP Proxy

Proxy Address: 192.168.1.1

Proxy Port: 5060

SIP Registrar

Apply

Finally we will modify the settings of Video codecs (section "Services - Phone - Video"). You can set the highest possible resolution (VGA or HD which is available only for 2N<sup>®</sup> Helios IP Verso) and also modify the bitrate and framerate parameters if needed. Consider the bandwidth consumption during the settings.

Services

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Calls

Audio

Video

Preferred Video Codecs

Codec 1: H.264

Codec 2: H.263+

Codec 3: H.263

Codec 4: None

H.264 Video Parameters

Video Resolution: VGA (640x480)

Video Framerate: 30 fps

Video Bitrate: 1024 kbps

H.263 Video Parameters

Video Resolution: CIF (352x288)

Video Framerate: 15 fps

Video Bitrate: 1024 kbps

Apply

## How to set Grandstream GXV3175v2 IP phone?

The easiest way how to set up this IP phone is via web interface but you can also use touch screen of this phone and set all necessary parameters this way. Complete setting via web interface is described below.

In the first step you have to enable "Account 1" and set all necessary information for this account in the "General" section. We will set peer-to-peer connection between 2N<sup>®</sup> Helios IP and GXV3175v2. Therefore set the IP address of the intercom into the "SIP Server" field as shown in the picture below. Authentication ID and password aren't used in this case.



The screenshot displays the web interface for the GXV3175v2 IP phone. The main heading is "GXV3175v2 -- Multimedia Phone Administration Interface". Below this, there are navigation tabs: "Status", "Account 1", "Account 2", "Account 3", "Advanced Setting", "Maintenance", and "Application Setting". The "Account 1" tab is selected. On the left, there is a sidebar with "General Settings" (highlighted in orange) and other options: "Network Settings", "SIP Settings", "Codec Settings", and "Call Settings". The main content area is titled "General Settings" and contains the following fields:

Account Active :	<input checked="" type="checkbox"/> Yes
Account Name :	Grandstream
SIP Server :	192.168.50.199
SIP User ID :	105
SIP Authentication ID :	105
SIP Authentication Password :	•••
Voice Mail UserID :	*26
Name :	
Tel URI :	Disable

At the bottom of the form, there are "Save" and "Cancel" buttons.

Go to the "Call settings" section and modify the "Dial Plan" field based on the picture below. You can also copy this dial plan from the Account 2 or from the Account 3.

Status Account 1 Account 2 Account 3 **Advanced Setting** Maintenance Application Setting

**General Settings**  
 Network Settings  
 SIP Settings  
 Codec Settings  
**Call Settings**

### Call Settings

Start Video Automatically :  Yes  
 Remote Video Request : Prompt  
 Dial Plan Prefix :  
 DialPlan : {x+|\+x+|\*x+|\*xx\*x+}  
 Early Dial :  Yes  
 Refer-To Use Target Contact :  Yes  
 Auto Answer : No  
 Send Anonymous :  Yes  
 Anonymous Call Rejection :  Yes

In the next step move to the section "Advanced settings - General settings" and check if the "Use Random Port" check box is empty.

**G X V 3 1 7 5 v 2**  
 -- Multimedia Phone Administration Interface

Status Account 1 Account 2 Account 3 **Advanced Setting** Maintenance Application Setting

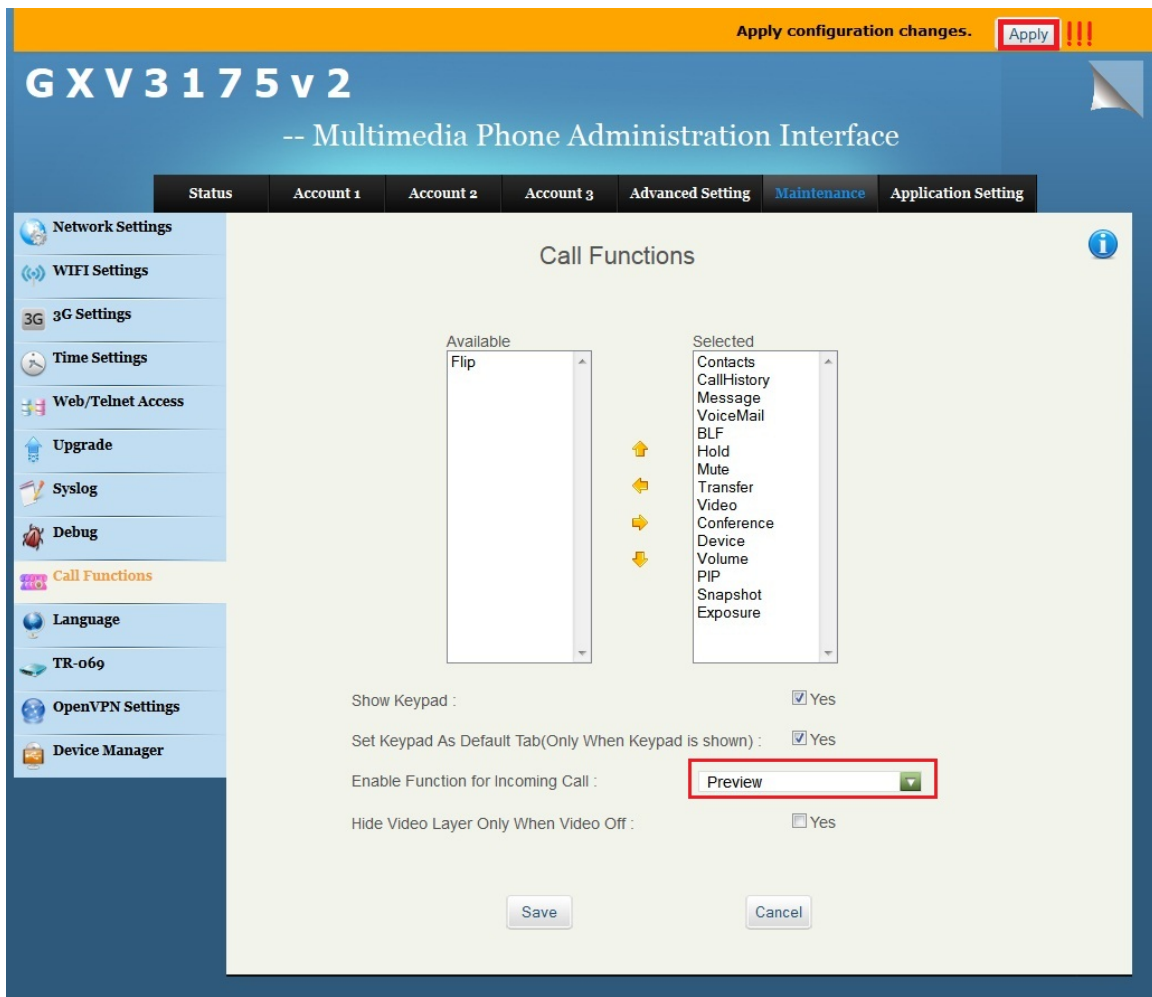
**General Settings**  
 Call Features  
 Video Settings  
 Ring Tone

### General Settings

Local RTP Port : 5004  
 Use Random Port :  Yes  
 \* Disable PC Port :  Yes  
 Disable in-call DTMF display :  Yes  
 Keep-alive Interval (s) : 20  
 STUN Server : stun.ipvideotalk.com  
 Use NAT IP :  
 SIP TLS Certificate :  
 SIP TLS Private Key :  
 SIP TLS Private Key Password :

Now go to section "Maintenance - Call Functions" and set the Preview as a function for incoming calls. Thanks to

this feature you can see the video from intercom before you pick up the call and start to talk with calling person. Don't forget to confirm all previous settings by "Apply" button on the upper right corner!



Finally you can double check that GXV3175v2 was properly registered to the intercom and that you can make a call between these devices.

