allo.com

NANO 2
User Manual

User Manual

NANO 2

Version 2.7

Table of Contents

1. Product introduction

1.1. Overview

The NanoPBX is a compact system that puts the rich features of a high-end PBX into the reach of small businesses. Its built-in voicemail, multi-level auto attendants, remote extensions and sophisticated call handling features help businesses reduce communications costs, while allowing employees to stay connected worldwide. Setting up and configuring the NanoPBX is a breeze with the user-friendly GUI and this document will show you just how easy it is!

A typical network diagram shows the function of Nano2PBX as below.
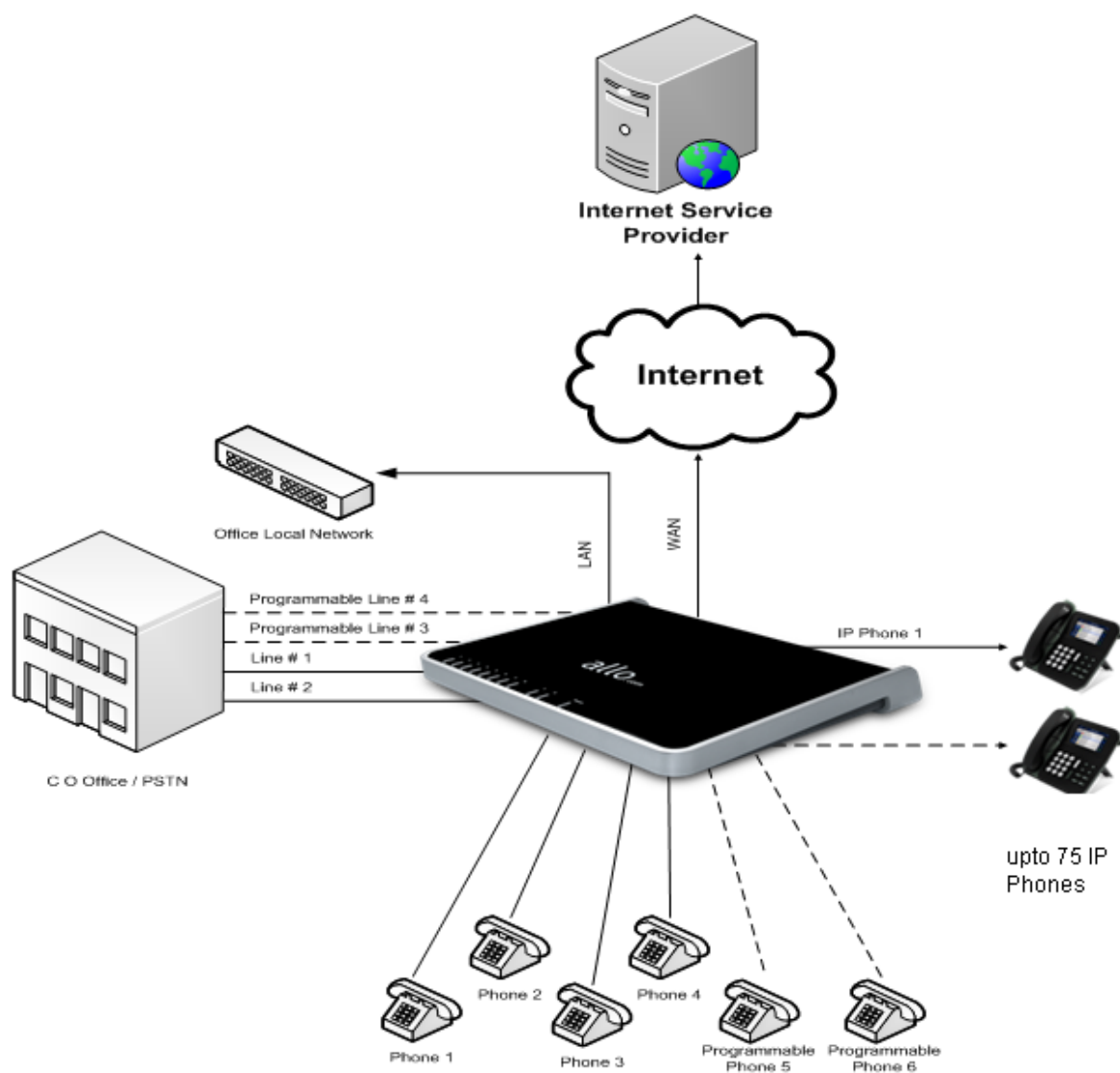
2. Getting Started With the Nano2PBX

2.1. Installation
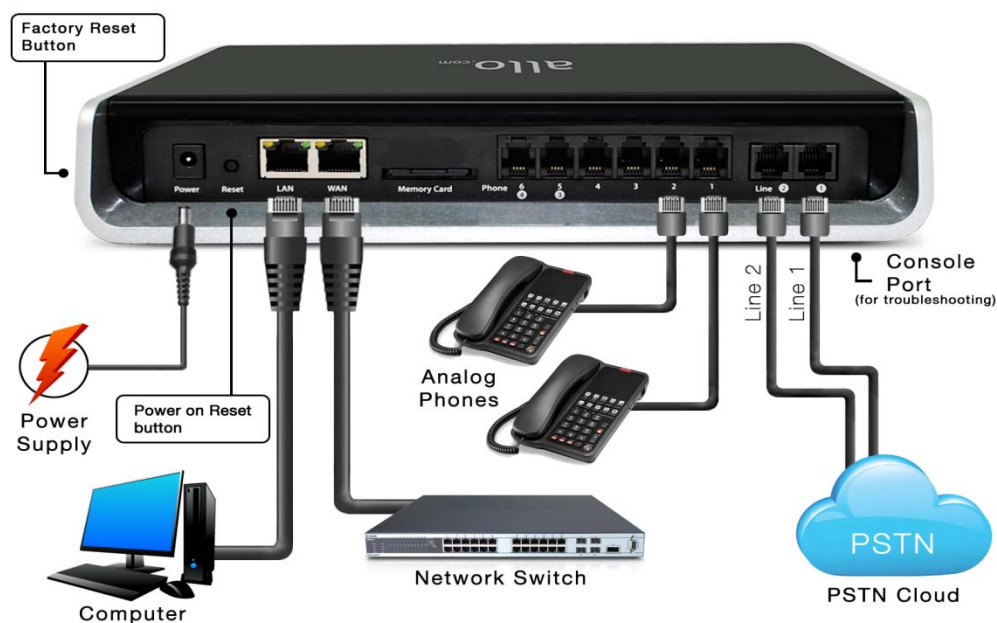
1. Unpack your NANO2 and make sure you received everything.
    Each box contains: 1 x NANO2
    1 x AC adapter (120V-220V) + 2 wall plugs (EU & US)
    1 x Ethernet cable

2. Plug in your NANO2 LAN port to your PC using the network cable provided

3. Plug the power cable to the Nano2PBX. After 10 secs, All the 6 LEDs start blinking and LEDs turn to orange after the completion of boot up.

4. Check your PC network settings and make sure that you have enabled the DHCP

5. Use following IP address in web browser to start the configuration process:
    192.168.113.1:8088

6. Factory default login credentials are:
    Username: admin
    Password: admin

7. Follow the quick setup wizard steps

2.2. Notification LEDs (On the Front Panel):

LED Indicator	LED Status
Power	Power is ON
LAN	Link is connected
WAN	Link is connected
FXS ports (Phone/FAX) (Port No. 1-6)	Off Hook
	On Hook
FXO (Telco Lines) (Port No. ❶ – ❹)	Channels Idle
	Channels Busy

Warning & Error Notifications	LED Status
USB is Not detected	LED ❹ Blinks
Flash Memory is Full	LED ❸ Blinks
Reboot Process	After 10 secs All the Front panel LEDS start blinking and LEDs will turn orange after the completion of Bootup process.
Firmware upgrade	All the Front panel LEDS start blinking and LEDs will turn orange after the completion of Firmware upgrade.
Factory Reset	Hold the reset button till all the LEDs turn Blue (About 5-6 Secs)

Important Note:

a) We strongly recommend you to change the Nano2PBX Admin Password from Factory default to Alpha numeric password to reduce the possibility of a security breach.

b) Use Mozilla Firefox Only!

c) Bootup process of the Nano2PBX takes about 3-4 mins

2.3. Quick Installation Wizard

By following the Wizard, you will create a very basic configuration file allowing you to make and receive calls thus confirming your NANO2 is fully functional.   Most of the features won't be visible until you complete the wizard and access the full fledge user interface.

After the completion of Quick setup wizard, NANO2 will reboot and launch the basic configuration file you've created. You are ready to test! You will also have access to the full fledge user interface and all the features offered.

Once the reboot is done, Connect the WAN port of the Nano2 to your network switch ,also you can remove the LAN connection and use the WAN IP to access the Full Fledge User Interface of the NANO2. The WAN IP address can be obtained by dialing **# from the Analog phone which is connected to the Nano2PBX

Important Note:
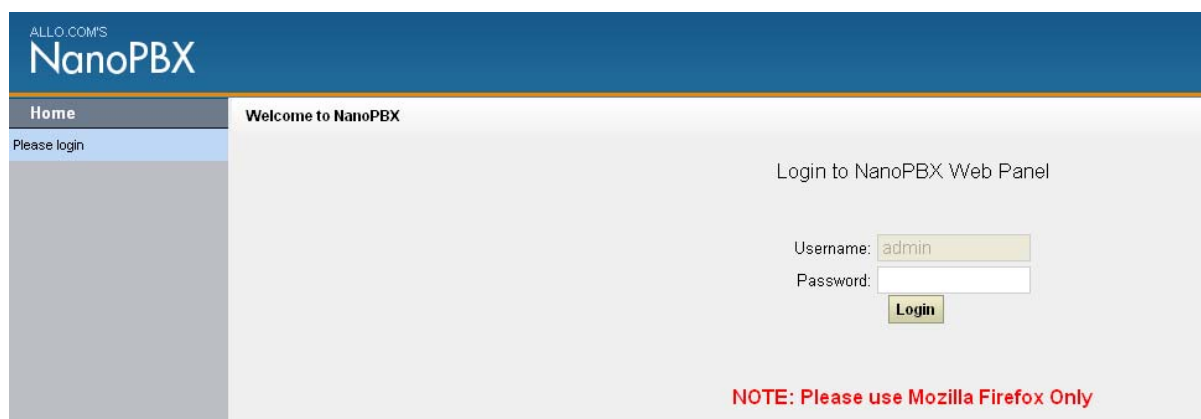
We strongly recommend you to access the Nano2PBX webGUI after completion of Quick setup wizard through WAN IP address. LAN interface only for initial setup.

Make sure to add the port number 8088 at the end of your IP address when logging in. For Eg: 192.168.0.42:8088

The WAN IP address can be obtained by dialing **# from the Analog phone which is connected to the Nano2PBX

2.4. Accessing   the   Main   GUI   (Graphical   User   Interface)

Connect an Analog phone to the any of the FXS ports of the Nano2PBX and dial "* *" to get the WAN IP address of the unit. Using the IP address obtained, open the Web browser and type: http://<WAN IP Address:8088>. Here is an example of how it should look: http://192.168.1.100:8088

Or also you can access the GUI of the NanoPBX by connecting a PC to the LAN port of the NanoPBX. Enable the DHCP option in the Network Settings of the PC, and then enter http://192.168.113.1:8088 in the Web Browser Address field. (Where 192.168.113.1:8088 is the default local LAN IP address of the NanoPBX).

On the login screen, the default username and password is "**admin"** and "**admin**" respectively. Press the Login button to enter the NanoPBX web panel. To change the password, please refer to the **General Settings > Admin Settings → PBX Settings** section in the navigation.

> **Important Note:**
> The nano2 comes with default credentials admin/admin you have to change these as soon as possible to avoid getting hacked. Allo.com cannot be held responsible for unauthorized access to the nano2 PBX. Please make sure you secure your PBX by placing it behind a firewall and changing passwords with frequency.

3. System Status

Navigation to System status, Status of NanoPBX including Memory Status, VoIP Status, Networking Status and Client Status (to see which clients are connected to the system).

After you login, you are brought to a System Status screen, which offers information about the NanoPBX, and help files to assist you in learning about all the different features of the system.

1. Under **Trunks** you will see the Registration Status of the VoIP Account(s) configured. If it displays "Registered" then it is successfully configured and connected.
2. Under **Conference Rooms** you will see all Conference Extensions available or unavailable.
3. Under **Parking Lot** you will see all Parking Extensions available or unavailable.
4. Under **Extensions** you will see all the users and the extensions connected to the NanoPBX. Also you can sort out the extension by clicking in the extension
5. Under **Queues** you will see all Queue Extensions available or unavailable.
6. Under **System status > System Info > General** you will have a summary of your General information, such as Hostname, Server Date & time zone, Uptime.
7. Under **System status > System Info > Memory Status** you will see total memory resources, including RAM usage, Compact Flash usage (to store voice files and voicemail) and Inbox status (for voicemail inbox).
8. Under **System status > System Info > Network Status** you will have a summary of your Network information, such as Hostname, WAN IP Address, Subnet Mask, WAN MAC Address and Default Gateway (you may refer to Settings > Network Settings for more info).
9. Under **System status > System Info > Disk** you will have a summary of Disk usage and Disk free space available on the file system.

4. Setting up the Features

4.1. Configure hardware

Analog Hardware Setup & Configuration

Port Configuration: Port Configuration allows you to manually configure Ports 5 and 6 either as FXS ports or FXO ports, which are configurable ports on the NanoPBX. That means you can configure these ports to be either as FXS ports (to connect Analog Phones) or FXO ports (to connect PSTN line).

> **Important Note:**
>
> 1. After choosing the port configuration click on Update Settings Button, NanoPBX will reboot for the changes to take effect. Once the system comes up you can use the ports as per your configuration.
>
> 2. Please make sure to disconnect incoming PSTN Line connected to the Port Number 5 and 6 before you switch from 4FXS and 4FXO Mode to 6FXS and 2FXO Mode. It may damage your NanoPBX Unit.

4.2. Extensions (Analog & SIP)

Extensions are the core of the NanoPBX. An extension is a number mapped to a person. So basically, every employee that is connected to the NanoPBX should have their own unique extension number, so that he/she can be reached, and be able to place calls.

The NanoPBX supports 2 types of Extensions: IP Extensions and Analog Extensions.

IP Extensions:

IP extension are devices that have only data networking connection such as Ethernet and they communicate with the NanoPBX using IP based protocol for signaling and Voice, examples are IP Phone, Soft Phone application. The NanoPBX can support up to 75 IP Extensions registration.

Analog Extensions:

An Analog Extension is used with a regular telephone system which can be connected to an available FXS port on the back of the NanoPBX. The NanoPBX can have up to 6 Analog Extensions.All the features are supported by both the Analog and IP extensions. Extensions can be part of other features, such as: Queues and Voicemail Groups. Also, the extension can have a voicemail of its own.

4.2.1. Create New User:

You have to create atleast one Dialplan using Dialplan option before trying to create Analog/IP Extensions

Navigation**: Users > Create New user**: This is where you setup your Analog/IP extensions

Technology	
SIP	Select "SIP" if the user is using SIP or a SIP device
Analog Station	select the FXS port need to be configured from the drop down menu.
Codec Preference	Select audio and video codec for the extension. The available codecs are: G711U-law, G711A-law, GSM, G.726, G.722, G.729, H.263 and H.263p.
General	
Extension	The extension number associated with the user.
CallerID Name	Configure the CallerID Name associated with the user
Dial Plan	Select one from the dropdown box (appears only if dialplans are created using Dialplans).By Default: Dialplan1
CallerID Number	Configure the CallerID Number that would be applied for outbound calls from this user. **Note:** The ability to manipulate your outbound Caller ID may be limited by your VoIP provider.

Enable Voicemail for the User: Enable this extension so that it can be visible when adding members to the Voicemail Group and Voicemail. The Voicemail PIN password will be same as the password provided to the Extension while creating an Extension. This will applies to both Analog and IP Extension. If you are trying to access or retrieve Voice Message, use that particular Extension password.

Email Address: Mail address which the user wishes to receive mails (for Voicemail)

VoIP Settings

SIP Password	Configure the password for the user.
NAT	Use NAT when the Nano2PBX is on a public IP communicating with devices hidden behind NAT (e.g., broadband router). If there is one-way audio issue, usually it's related to NAT configuration or Firewall's support of SIP and RTP ports.
Can Reinvite	By default, the Nano2PBX will route the media steams from SIP endpoints through itself. If enabled, the PBX will attempt to negotiate with the endpoints to route the media stream directly. It is not always possible for the Nano2 to negotiate endpoint-to-endpoint media routing. The default setting is "No".
Insecure	Port: Allow peers matching by IP address without matching port number. • Very: Allow peers matching by IP address without matching port number. Also, authentication of incoming INVITE messages is not required. • No: Normal IP-based peers matching and authentication of incoming INVITE. The default setting is "No".

Advanced Options	
In Directory	Check this option if the user is to be listed in the system telephone directory
Is Agent	Check this option if this user is a call queue member
Incoming Call record	Record the incoming calls Note: calls with G722 and G729 codecs don't recorded
Outgoing Callrecord	Record the outgoing calls **Note**: calls with G722 and G729 codecs don't recorded

Important Note:

The Incoming/ Outbound calls with G722 and g729 codec won't be recorded

4.2.2. Modify/Delete selected users

Navigation: **Users**: This is where you can edit / delete existing Analog/IP Extensions individually.

On the right side of the page, you can see the list of extensions you have setup. To edit, or delete any of them, simply click the appropriate icon provided to the right of each account/Extension.

Once you click on the Edit button of an Extension then it will display the information of that particular extension. Here you can change the required details and then click on the Update Extension button to save the changes made.

You can delete an extension by clicking on the delete button of the extension from the list of extensions displayed.

You can delete many existing extensions by clicking on the Delete Selected Users button after marking in the check boxes of extensions from the list of extensions displayed. Click ok the popup window to delete the selected users.

Important Note:
Make sure to click the APPLY CHANGES button in the top navigation bar, after adding / editing / deleting any Extension. The APPLY CHANGES tab appears if some changes are made and not saved.

4.3. Trunks

Navigation: **Setup >Trunks > Analog Trunks**: Here you can configure the Analog Trunks.

4.3.1. Analog Trunks

The analog trunk options are listed in the table below.

Analog Trunks	
Channels	Select the channel for the analog trunk.
Trunk Name	Specify a unique label to identify the trunk when listed in outbound rules, incoming rules and etc.

4.3.2. VOIP Trunks

Navigation: **Setup > Trunks > VOIP Trunks**: This is where you setup VoIP Trunk, or manage existing ones.

In this page, fill in the Provider Name, Host name, Host Port, Username, Password and Proxy information given to you by your VoIP provider (known as SIP Credentials). Apply any Codec Settings required. You can prioritize your active codec's by using the drop down buttons.

After you have entered the details, click the Save button at the bottom.

4.3.3. Adding a New VoIP Account Details

Create New SIP trunk	
Context Naming	You can select the context naming based on username/Provider name/Assigned by asterisk GUI By Default: Based on Username
Provider Name	Configure a unique label to identify this trunk when listed in outbound rules, inbound rules and etc.
Host Name	Configure the IP address or URL for the VoIP provider's server of the trunk.

Host Port	
Username	Enter the username to register to the trunk from the provider.
Password	Enter the password to register to the trunk from the provider.
Qualify	If you enable Qualify, asterisk will send the SIP Options command regularly to check that the device is still online

4.3.4. Editing / Deleting an Existing VoIP Account:

On the right side of the page, you can see the list of VoIP trunk you have setup. To edit, or delete any of them, simply click the appropriate icon provided to the right of each trunk.

Once you click on the edit button of a VoIP trunk then it will display the information of that particular VoIP trunk, here you can change the required details and click on the Save button and then click on the apply changes tab to save the changes made.

To ensure successful registration of your VoIP Trunk, you must click the "System Status" tab on the top navigation menu (see Status section for more info)

Important Note:
Make sure to click the APPLY CHANGES tab in the top navigation bar, after adding any new VoIP trunk, or editing / deleting.

4.4. Outgoing Calling Rule

4.4.1. New Calling Rules

Navigation: **Setup** > **Outgoing Calling Rules > New Calling Rule**: This is where you configure Dial out Rules.

Outgoing Calling Rules represent the prefix sequence used to dial when making an outgoing call either through PSTN (Analog) or VoIP. There are two ways to make outgoing calls for the registered extension users:

- VoIP / SIP trunk via ITSP gateway
- LINE / PSTN trunk via FXO port

Calling Rule Name	Configure the name of the calling rule (e.g., local, long-distance, and etc). Letters, digits, _ and - are allowed. Pattern • All patterns are prefixed with the "_".
Pattern	All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. [12345-9]: Any digit from 1 to 9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately.
Caller ID Name	Configure the CallerID Name associated with the FXO trunk

You can even use the Calling rule to route it to local extensions as Destination or Route the calls to Trunks created so as to make the calls successful. There as still many options like striping the number of digits from front, prepend the digits before dialing and filter. There is also another alternative way if the trunk fails to route the call i.e. Failover Trunk an alternative trunk to route calls.

Send this call through trunk	
Use Trunk	Select the trunk for this outbound rule.(FXO/VOIP Trunk)
Strip	Allows the user to specify the number of digits that will be stripped from the beginning of the dialed string before the call is placed via the selected trunk.

Prepend	Specify the digits to be prepended before the call is placed via the trunk. Those digits will be prepended after the dialing number is stripped.
Filter	This option is used to filter out certain characters. The characters listed in the field will be permitted, while all others will be filtered out.

Here we will discuss about how to choose outgoing trunks in two different ways.

The First way is, choose a provider or a trunk based on prefix.

This type of rule will allow user to create a prefix for choosing SIP (VoIP) or LINE (PSTN / FXO) trunk to make an outgoing call.

For Eg 1: If you would like to strip out the first digit from the dialed number follow the example: Add **"_8X."** in pattern & configure **"1"** in strip out field to remove the first digit from the dialed number either analog or SIP users which uses the same dial plan

Second way is, choose provider or trunk based on actual number dialed,

This type of rule will allow the user to choose suitable provider based on Country code. For e.g. If the user want to Dial 44-9872837532, then adding **"_X."** in Dial Pattern and configure "0" in Strip Out field and add "44" in Prepend field & selecting VoIP Trunk/ Analog Trunk in trunk sequence, PBX will allow number dialed from **44+** followed by (any digit from 0 to 9) like 449872837532 will route through VoIP Provider 2. If the ITSP / VoIP provider offer cheaper rates for the region where number starts from 44 users can make use of this rule.

In the same way, user can create a prefix and select a LINE (PSTN / FXO) trunk to make an outgoing call.

4.4.2. Restore Default Calling Rule

Navigation: **Outgoing Calling Rules > Restore Default Calling Rule**: This is where you configure Default Outgoing Calling Rules.

These are the default calling rules where they show us an example of the patterns and which all fields can be used as default.

4.5. Dial plans

Navigation: **SETUP→Dial plans**: This is where you configure Dial plans for the users.

A Dial plan is a collection of outgoing rules. Dial plan are assigned to users to specify the dialing permissions they have

For Example: you might one Dialplan for local calling that permits the users of that DialPlan to dial local numbers, via the "local" outgoing calling rule. Another user may be permitted to dial long distance numbers, and so would have a DialPlan that includes both the "local" and "log distance" outgoing calling rules.

You have to create the Dialplan first before you create any user accounts to make your call successful. The Dialplan details are as follows.

Dialplan Name: the name user wish to see in that field of Dialplan.

Include Outgoing Calling Rules: when the outgoing Calling rules are created it displays here so that to include it in the dialplan.

Include local contents: here the user can select the features which he wishes to use.

After all the changes, click on the save button. And don't forget to click on apply changes button on top navigation bar immediately after save button.
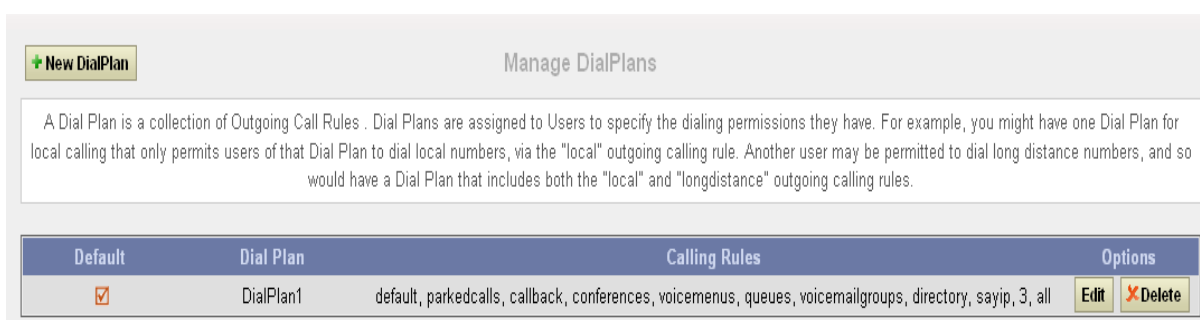
4.6. Time Intervals

Navigation: **Time Intervals**: This is where you can create/edit /delete Time intervals for the scheduling of the incoming calling rules.The time intervals Details are given as:

Time Interval name: the name of the time intervals according to users need.

You can select either by day of week or by days of the monthYou can also select time duration in a day or the whole day.After all the changes are done click on the update button to create the time intervals and click on the apply changes button to confirm the time intervals configured.

4.7. Voicemail Groups

A Voicemail Group is a pre-programmed group of voicemail recipients. All the members of this group will receive the same voicemail message.

4.7.1. Configuring a Voicemail Group

Navigation: **Voicemail Group**: This is where you setup your Voicemail Groups

To create a Voicemail Group, select a Group Name, choose the users who will belong to that Group, and press the Save button.

You can see all existing Extensions in User Mailboxes, which are checked in checkboxes to include in the group. You can also edit or delete groups using edit and delete button simultaneously.

4.7.2. Voicemail Group Details

Voicemail's Group Extension:	The number to be allocated for the Voicemail Group
Label:	Name of the Voicemail Group
User Mailboxes:	The list of available extensions that could be part of this Voicemail Group.

The marked extensions in the list of extensions belong to this Voicemail Group

4.8. Music on Hold (MOH)

Navigation: **SETUP →Music on Hold**: This is where you can upload new MOH based on the classes.

This is a music file which will be played by the NanoPBX when any of the calling party is kept on hold.

All the MOH files which are uploaded will be listed in the MOH selection box under selected MOH Class. Chose any of the MOH class form the drop down box and select from the

Sound files listed. click on Apply Changes tab to save the changes made. You can even delete the MOH classes and create them.

4.9. Call Queues

Queues used to distributes incoming calls in the order of arrival to the first available extension in the queue. The system answers each call immediately and, if necessary, holds it in a queue until it can be directed to the next available extension. This feature is used to balance the workload among group of extensions. Queues will provide the following functions, Incoming calls being placed in the queue Extensions that answer the call in the queue. Option to choose ring type strategy, to handle the calls in the queue and distribute the calls in the queue. Music played while waiting in the queue.

4.9.1. Configuring a Queue

Navigation: Call Queues: This is where you setup your Queues.

To create a Queue, simply fill in all the required details including Extension, Queue Name, Strategy, Queue Length, and select the other options. When done, click the Update button.

Queue Details

- Extension: Extension number to reach the Queue directly
- Queue Name: The name of the Queue
- Strategy Type:
- Ring All: Rings all available extensions
- Round Robin: Takes turns ringing each available extension

- Least Recent: Rings the extension which was least recently called by this queue
- Fewest Calls: Rings the extension with fewest complete calls from this queue
- Random: Rings a random extension
- Round Robin Memory: Performs a Round Robin remembering where we left off with the last ring pass
- Music On Hold : selecting the desired music from the available files in dropdown
- Queue Timeout: Select the required seconds so that the queue will get terminated after the given seconds
- Wrap-Up Time: After a call is finished, the time it takes an extension to become available again to become available in the queue
- Queue Max Len: The maximum number of callers waiting in queue for an available extension.
- Auto fill: The queue will complete as many calls simultaneously to the available agents.
- Auto Pause: Pauses an agent if they fail ti answer the call.
- Report Hold time: Reports the hold time of the agents before the caller is connected to the agents.
- Key Press Events: If a caller presses a key while waiting in the queue , this selects which voice menu should process the key press.

List of Available Members (Agents)

This is the list of available extensions that could be part of this queue by selecting them using checkboxes.

4.10. Greeting Messages

4.10.1. Recording Voice Files

Navigation: **Setup → Greeting Message > Record a new voice menu prompt**

- Specify the Voice file name.
- Select the format (GSM/WAV/ g729) for the IVR prompt file to be recorded.
- Select the extension to receive the call from the Nano2PBX to record the Greeting message.
- Click the "Record" button. A request will be sent to the Nano2PBX. The Nano2PBX will then call the extension for recording the IVR prompt from the phone.
- Pick up the call from the extension and start the recording following the voice prompt.
- The recorded file will be listed in the Greeting Message web page. Users could select to re-record, play or delete the recording.

4.10.2. Uploading Voice Files

Navigation: **Voice Prompt > Uploading Voice Files**

If the user has a pre-recorded Voice file, click on "Upload IVR Prompt" in Web GUI->**PBX->Internal Options**->**IVR Prompt** page to upload the file to the Nano2PBX. The following are required for the IVR prompt file to be successfully uploaded and used by the Nano2PBX:

- PCM encoded.
- 16 bits,8000Hz mono.
- In GSM, .WAV. g729 format

> **Important Note:**
> File Size Should be under 5 MB

Convert Sound File

Usually, the recorded sound file cannot be directly used by the Nano2PBX. Users would need convert the sound file to make it compatible with the PBX system before uploading, using audio processing software tools. The recommended format for the compatible sound file format is GSM, .WAV, G.729.

For example, the following online converter provided by Digium, Inc can be used to convert the sound file.

http://my.digium.com/en/products/ivr/audio-converter/
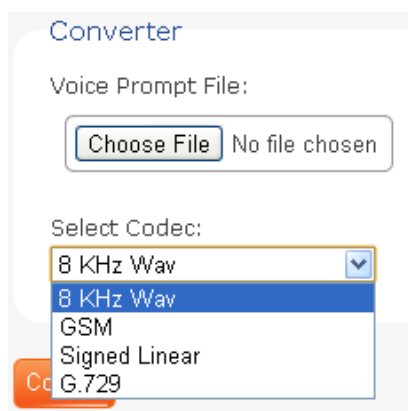
Important Note:

We recommend to use GSM format sound file for uploading as the file size is usually smaller than the other compatible format.

4.11. IVR

IVR (Interactive Voice Response) is a pre-recorded interactive operator defined by a sequence of actions that provides a customer with a better call experience. An IVR can be 'chained' with other IVR's creating a multi-level IVR system.

Example :

"Welcome to ALLO PVT LTD! If you know your party's extension, please dial it now.

Press 1 for Customer Service

Press 2 for Billing

Press 3 for Shipping"

Navigation: **Setup → IVR** , User could Setup, Edit and Delete an IVR.

IVR Configuration Parameters	
Name	Configure the name of the IVR
Extension	Enter the extension number for users to access the IVR.

Allow Dialing Other Extension	If enabled, all callers to the IVR can dial other extensions. The default setting is "No".
Actions	A sequence of actions performed when call enters the IVR
Add new Step	Select the IVR steps as per the requirement
Allow Key press Events	Select the event for each key pressing for 0-9, *Thee event options are: • Extension • Voicemail • Conference Rooms • Voicemail Group • IVR • Queues • IVR • Hang-up

Important Note:
Before you start to Create Voice Menu you need to have Greeting Messages (Navigation: Setup→Greeting Messages).

4.12. Incoming Calling Rules

Navigation: **SETUP →Incoming Calling rule**: This is where you can create/edit /delete Incoming calling rules.

A Incoming Calling Rule is an rule which route the incoming call to phone number. Incoming Calling Rule is a feature that enables incoming calls to be routed directly to selected stations without attendant assistance.

Incoming Calling Rule Details

Incoming Calling Rule Configuration Parameters	
Trunk	Select the trunk to configure the inbound rule.
Time Interval	Select the time interval from the list specify the time for the trunk to use the inbound rule.
Pattern	All patterns are prefixed with the "_". • Special characters: X: Any Digit from 0-9. Z: Any Digit from 1-9. N: Any Digit from 2-9. ".": Wildcard. Match one or more characters. "!": Wildcard. Match zero or more characters immediately
Destination	Select the default destination for the inbound call • Extension • User's Voicemail • Conference Rooms • Voicemail Group • IVR • Queues • IVR • Hang-up

4.13. DID Routing

Navigation: **SETUP →DID Routing**

Direct Inward Dial. A specially configured phone line from the telephone company that allows for dialing inside a company directly without having to go through an attendant. A DID line cannot be used for out dial operation since there is no dial tone offered. However, it can be configured so an outside caller can reach an inside extension with a 7-digit number through the phone company's central office.

DID Configuration parameters	
DID Number	Enter the DID numbers provided by the VOIP Service provider

Destinations	Select the DID destination. Only the selected category can be reached by DID.
	
	• User Extension
	
	• Conference
	
	• Call Queue
	
	    • User's Voice Mail
	
	    • Voicemail Group

5. General Settings

5.1. Voicemail

Navigation: **Voicemail**: This is where you can manage the configurations of the voicemail.

5.1.1. General settings (Accessing, Retrieving & Managing Voice Mail)

The NanoPBX allows users to manage voicemail through voice messages in their phones. This section will summarizes how to access, retrieve and manage voicemail and other settings.

The default feature code for accessing Voicemail can be set by using "Extension for checking messages". After dialing this code, you will enter a basic voice menu with the option to listen or forward messages and configure voicemail options. When prompted, provide the appropriate Voice Mail number and the password, which is same as it was configured in the Extensions i.e. Extension & Password.

While you listen to the recorded voice message you can use the following keys for navigation.

1 Read Voice mail Messages

3 Advanced options

1 Reply

3 Hear Message

5 Leave Message

4 Play previous message

5 Repeat current message

6 Play next message

7 Delete current message

8 Forward message to another mailbox

1 Use Voicemail number

2 Use Voicemail Directory

9 Save message in a folder

0 Save in new Messages

1 Save in old Messages

2 Save in Work Messages

3 Save in Family Messages

4 Save in Friends Messages

* Help; during msg playback: Rewind

 # Exit; during msg playback: Skip forward

2 Change folders

0 Switch to new Messages

1 Switch to old Messages

2 Switch to Work Messages

3 Switch to Family Messages

4 Switch to Friends Messages

3 Advanced options

5 Send Message

1 Use Voicemail number

2 Use Voicemail Directories

0 Mailbox options

1 Record your unavailable message

2 Record your busy message

3 Record your name

4 Record your Temporary Greetings

1 Record your temporary message

2 Erase your temporary message (going back to the standard message)

5 Change your password

* Return to the main menu

* Help; during message playback: Rewind

 # Exit;

There are still many options which are user friendly they can be detailed as:

Direct Voicemail Dial: to enable direct voicemail dial by pressing # followed by extension number

Max greeting (in seconds): Maximum number of seconds for User's voicemail greeting.

Dial '0' for operator: enable callers to enable the voicemail application and connect to an operator extension.

Message Options:

Maximum messages per folder: Maximum number of messages that a user can have in any of their folders.
Max message time: Maximum duration of the voicemail message in seconds.
Min message time: Minimum duration of the voicemail message in seconds.
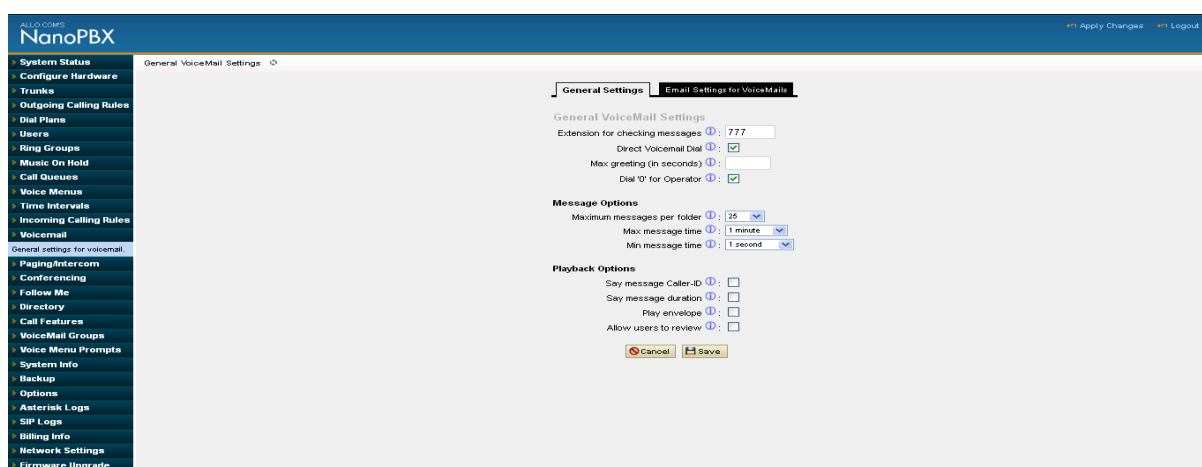
Email Settings for Voicemail

This section will summarize how to configure and notify the user if he/she has received a new voicemail to their email address. The caller after leaving the voicemail, mail will be sent to the user mail address as mentioned while configuring the user extensions.

Voicemail Email Alert Preference	
Send message by Email	If enabled, Customized message can be sent to users email address
Attach Recordings to Email	If enabled, voicemails will be sent to user's Email address as voice recording The default setting is "Yes".
Template for Voice mail Emails	Fill in the "Subject:" and "Message:" content, to be used in the Email when sending to the users. The template variables are: • \t: TAB • ${VM_NAME}: Recipient's first name and last name • ${VM_DUR}: The duration of the voicemail message • ${VM_MAILBOX}: The recipient's extension • ${VM_CALLERID}: The caller ID of the person who has left the message • ${VM_MSGNUM}: The number of messages in the mailbox • ${VM_DATE}: The date and time when the message is left

5.2. Network Settings

Navigation: **Network Settings**: This is where you setup your Networking Configuration

5.2.1. WAN Configuration

Please refer to the following tables for basic network configuration parameters

WAN Configurations	
IP Address Mode	Select DHCP, Static IP. The default setting is DHCP.
IP Address	Enter the IP address for static IP settings.
Subnet mask	Enter the subnet mask address for static IP settings. The default setting is 255.255.255.0
Network Id	your local network segment (Ex: 192.168.0.0)
Broadcast	Enter the broadcast address
Gateway	Enter the gateway IP address for static IP settings.
DNS Primary	Enter the DNS server 1 address for static IP settings
DNS Secondary	Enter the DNS server 2 address for static IP settings

Important Note:
DHCP mode isn't recommended. Or trouble may arise when SIP client need to change registration server address caused by revised IP.

5.2.2. LAN Configuration

Use this setting in the event that you want to use the NanoPBX as your network router and act as a DHCP server.

By default LAN IP address is 192.168.113.1 & you cannot change the ipaddress of LAN interface.

Important Note:
WAN port IP and LAN port IP Address shouldn't be in the same net segment. And we recommend not to change LAN address.

5.2.3. Host Configuration

Host Configuration is used to manage your NanoPBX's Host Name.

Host Name: Used to name the device to identify inside the LAN network. This field is optional but may be required by some Internet Service Providers or system administrators.When you are finished applying Network Settings, click the Update button.

5.3. Feature Settings

Navigation: > **Call Features** is where you can configure the Call feature settings for the extensions.

Feature digit time out, maximum time (ms) between digits for feature activation (default is 1000 ms)

Call Parking Settings	
Extension to Dial to Park a Call	Default Extension: 700. •During an active call, initiate blind transfer and then enter this code to park the call.
Extensions for Parked Calls	Default Extension: 701-720. • These are the extensions where the calls will be parked, i.e., parking lots that the parked calls can be retrieved.
Parked Call Timeout (in secs)	Default setting: 1000. • This is the timeout allowed for a call to be parked. After the timeout, if the call is not picked up, the extension who parks the call will be called back.

PBX Feature Prefixes: This lets the user assign the codes for different features present in the NANOPBX.

Feature Map	
Blind Transfer	Default code: *1 • Enter the code during active call. After hearing "Transfer", you will hear dial tone. Enter the number to transfer to. Then the user will be disconnected and transfer is completed.

Disconnect	Default code: *2
	This is to assign the code for performing call disconnect
Attended Transfer	Default code: *3
	Enter the code during active call. After hearing "Transfer", you will hear the dial tone. Enter the number to transfer to and the user will be connected to this number. Hang up the call to complete the attended transfer.
Call Parking	Default code: *4
	Enter the code during active call to park the call.
Call Pickup	Default code: *8
	This feature code is to pick up the ringing extension from another extension if the party is not available in the desk.
Call Record	Default code: *9
	This is to assign the code for performing Call Record. If the NanoPBX user extension wants to record the particular active call, then he needs to press Call record feature code and that call conversation get recorded.
Call Back	Default code: *7
	Eg: If you want to make call back between your phone and remote party 2002, dial *72002 from your phone then it will hangup automatically. The remote party side he will get the call and once he answer the call, you will get the call back. Suppose If remote party didnt answer the call , it will retry the call with retry count.

5.4. Admin Settings

5.4.1. General Preferences

Navigation: **Admin Settings> General Preferences**: This is where you can configure the General Settings of the Nano2PBX

General Preferences

Global Outbound CID	Configure the global CallerID used for all outbound calls when no other CallerID is defined with higher priority. If no CallerID is defined for extension or trunk, the global outbound CID will be used as CallerID.
Global Outbound CID Name	Configure the global CallerID Name used for all outbound calls. If configured, all outbound calls will have the CallerID Name set to this name. If not, the extension's CallerID Name will be used.
Operator Extension	Specify the operator extension, which will be dialed when users presses 0 to exit voicemail application. The operator extension can also be used in IVR option.
Ring Timeout	Configure the number of seconds to ring an extension before the call goes to the user's voicemail box. The default setting is 20 .

VoIP Phone Digit Map: This option allows the administrator to define a global digit mapping string compatible with RFC 3435. There is no default setting and this option does not sync with the dialplan assigned to an individual user. The following examples should assist in writing an acceptable digit mapping string.

- [2-9]11 - Where calls beginning with digits 2-9 followed by digits 11 are dialed immediately.
- 0T - Where calls beginning with digit 0 followed by a pause equal to the "Digit Timeout" option.
- +011xxx.T - Where calls beginning with the + character, followed by 011 digits and then at least three more digits before any arbitrary number is matched, dialed after Digit Timeout is reached.
- 0[2-9]xxxxxxxxx - Where calls beginning with 0, followed by any digit from 2-9, followed further by 9 more digits are dialed immediately.
- +1[2-9 xxxxxxxx - Where calls beginning with the + character, followed by 1, followed by any digit from 2-9, followed by 8 more digits are dialed immediately.
- [2-9]xxxxxxxxx - Where calls beginning with any digit from 2-9, followed by 9 more digits are dialed immediately.
- [2-9]xxxT - Where calls beginning with any digit from 2-9, followed by three more digits are dialed after Digit Timeout is reached.
- [2-9]11|0T|+011xxx.T|0[2-9]xxxxxxxxx|+1[2-9]xxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT   where each entry is separated by the | character. For more information, please refer to RFC 3435.

VoIP Phone Digit Timeout: The timeout variable is the number of seconds the phone will wait for each segment of a digit map expressed as an integer.

5.4.2. Language

Navigation: **Admin Settings > Language**: This is where you can configure the Language Settings of the NanoPBX.
Language settings allow the user to specify the default language voice prompt. The currently available voice prompts are English, Spanish, Turkey, French and Italy.
By default selected voice prompt language is English.

5.4.3. PBX Settings

Navigation: **Admin Settings > PBX Settings**: This is where you can configure the General Settings of the Nano2PBX

Change Admin Password:

- Enter the new password and retype the new password to confirm. The new password field has to be at least 5 characters
- Click on "Update" and the user will be logged out.
- Once the web page comes back to the login page again, enter the new password to login.

Date &Time Settings: You can set the date and time of the Nano2PBX either through Enabling NTP or by manual entry. If you are enabling NTP, select the time zone according to your country timing and enter the ntp server details.

Enable NTP	Enabling NTP allows you to set the Timezone and Date based on the NTP server provided
Time Zone	Select the proper time zone option so the Nano2PBX can display the correct time accordingly.
NTP Server	Specify the URL or IP Address of the NTP server for the Nano2PBX to synchronize the date and Time. For Eg: pool.ntp.org

Email Settings:

The Email application on the Nano2PBX can be used to send out Emails to users with Fax (e.g., Fax-To-Email), Voicemail (Voicemail-To-Email) and other information as attachment.

Email Settings	
SMTP	The IP Address or domain name of the SMTP server

SMTP PORT	The port number at which SMTP server is running. Generally SMTP port No is 25
AUTH User	Authorized username of the Admin Email ID
AUTH Password	Authorized password of the Admin Email ID
Admin Emailed	Specify the Administrator Email ID.
TLS Support	TLS setting to require mail to be transmitted via a secure connection when users correspond with specific domains and email addresses.

5.4.4. Reboot

Navigation: **Admin Settings > Reboot**: This is where you can configure the General Settings of the Nano2PBX

The administrator of the NanoPBX can remotely reboot the Nano2PBX by pressing the "Reboot" button at the bottom of the System management. Once done, following screen will be displayed to confirm reboot.

The user can re-login to the phone after POWER LED and all the six PHONE LED's turn orange and remain stable on the Front Panel of your NanoPBX

> **Important Note:**
> Nano2PBX will take about 3 minutes time to reboot and keep patience
>
> Rebooting the Nano2PBX results in termination of active calls
>
> IP address of the Nano2PBX after reboot, Please check the IP Address by dialing **# from the Analog Phone connected to FXS port of the Nano2

5.5 SIP Settings

5.5.1. General

Navigation: **SIP settings > General**: This is where you can configure the General sip settings.

General	
Bind UDP Port	Configure the UDP port used for SIP. The default setting is 5060.
Bind IP Address	Configure the IP address to bind to. The default setting is 0.0.0.0, which means binding to all addresses.
Enable DNS SRV lookups	Select to enables DNS SRV lookups on outbound calls from the Nano2PBX. The default setting is "No"

5.5.2. TOS

Navigation: **General Settings** > **SIP Settings > TOS**: This is where you can configure the TOS sip settings.

The Details to be filled are given as:

TOS	
Generate In band Ringing	Configure whether the Nano2PBX should generate inband ringing or not. The default setting is "Never". • Yes: The Nano2PBX will send 180 Ringing followed by 183 Session Progress and in-band audio. • No: The Nano2PBX will send 180 Ringing if 183 Session Progress has not been sent yet. If audio path is established already with 183 then send in-band ringing. • Never: Whenever ringing occurs, the Nano2PBX will send 180 Ringing as long as 200OK has not been set yet. Inband ringing will not be generated even the end point device is not working properly
Server User Agent	Allows you to configure the user agent string for the Nano2PBX
DTMF Mode	Select DTMF mode to send DTMF. The default setting is RFC2833. If "Info" is selected, SIP INFO message will be used. If "Inband" is selected, 64-kbit codec PCMU and PCMA are required. When "Auto" is selected, "RFC2833" will be used if offered, otherwise "Inband" will be used. The default setting is "RFC2833".
Maximum Registration/Subscri	Configure the maximum duration (in seconds) of incoming registration and subscription allowed by the Nano2PBX. The default setting is

ption Time	3600
Min Registration/Subscription Time	Configure the minimum duration (in seconds) of incoming registration and subscription allowed by the Nano2PBX. The default setting is 60.
Default Incoming/Outgoing Registration Time	Configure the default duration (in seconds) of incoming/outgoing registration. The default setting is 120.

5.5.3. NAT

Navigation: **General Setting > SIP settings > NAT**: This is where you can configure the NAT sip settings.

Email Settings	
Extern IP	Configure a static address and port (optional) that will be used in outbound SIP messages if the Nano2PBX is behind NAT.
Extern Host	you can specify an external host name, and Asterisk will perform DNS queries periodically based on the External Refresh Interval
Extern Refresh	Configure the refresh interval for the external host (if used) The default setting is 10.
Local Network Address	Specify a list of network addresses that are considered inside of the NAT network. Multiple entries are allowed. If not configured, the external IP address will not be set correctly. A sample configuration could be as follows: 192.168.0.0/255.255.0.0
NAT Mode	This is a global NAT setting that will affects all peers and users. The default setting is "YES". • YES = Always ignore info and assume NAT • NO = Use NAT mode only according to RFC3581 • NEVER = Never attempt NAT mode or RFC3581 support • ROUTE = Assume NAT, don't send report.

Allow RTP Reinvite	If enabled, the Nano2PBX will try to redirect the RTP media stream (audio) to go directly from the caller to the callee. The default setting is "No NAT". • Yes • No NAT: Allow media path redirection (Reinvite) but only when the peer is not be behind NAT. The RTP core can detect if the peer is behind NAT or not based on the IP address where the media comes from. • Update: Use UPDATE for media path redirection, instead of INVITE. **Note:**Some devices do not support this (especially if one of them is behind NAT).

5.4.4.1. Misc

Navigation: **General Settings > SIP settings > Misc:** This is where you can configure the miscellaneous sip settings.

The details to be filled are given as:

FAX Settings	
FAX Format	Select the FAX format either in TIFF or PDF
T.38 fax (UDPTL)	Enables T.38 Fax Pass-through Mode.
Fax Email ID	Assign the Email Address to which the FAX attachment has to be sent
Out Bound SIP Registration Settings	
Register	Register as a SIP user agent to a SIP proxy (provider)
Register Time-Out	Configure the register retry timeout (in seconds). The default setting is 20.
Register Attempts	Configure the number of registration attempts before the Nano2PBX gives up. The default setting is 0, which means the Nano2PBX will keep trying until the server side accepts the registration request.

Video Settings	
Max Bit Rate	Configure the maximum bit rate (in kb/s) for video calls. The default setting is 384.
Support for SIP Video	Select to enable video support in SIP calls. The default setting is "Yes".
Generate Manager Events	If enabled, the Nano2PBX will generate manager events when SIP UA performs events (e.g. Hold). The default setting is "No".
Reject Non Matching Invites	If enabled, when rejecting an incoming INVITE or REGISTER request, the Nano2PBX will always reject with "401 Unauthorized" instead of notifying the requester whether there is a matching user or peer for the request. This reduces the ability of an attacker to scan for valid SIP
Nonstandard G.726 Support	If enabled, when the peer negotiates G726-32 audio, the Nano2PBX will use AAL2 packing order instead of RFC3551 packing order (AAL2-G726-32). The default setting is "No".

5.4.4.2. Codecs

Navigation: **SIP settings > Codec's**: The following Audio & Video codec's are supported in Nano2PBX

- G711 u-law
- G711 a-law
- G.726
- G.729
- G.722
- GSM
- H.263
- H.263p

Select the codecs from the list by enabling it.

6. Additional Features

6.1. Conferencing:

The conference bridge configurations can be accessed under Advance **Features >Conference**. In this page, users could create, edit and delete conference bridges.

Nano2PBX supports up to 8 simultaneous PSTN or IP participants irrespective of the number of conference bridges.

Conference Bridge Settings	
Extension	Configure the conference number for the users to dial into the conference.
Play Hold Music For First Caller	If enabled, the Nano2PBX will play Hold music to the first participant in the conference until another user joins in. The default setting is "No".
Announce Callers Count on joining the Conference	If enabled, the caller will be announced to all conference participants when there the caller joins the conference. The default setting is "No".
Quiet Mode	If enabled, if there are users joining or leaving the conference, voice prompt or notification tone won't be played. The default setting is "No".

Working Scenario

By dialing the extension number configured in the above step will allow you to enter into Conference Bridge. During the conference call, users can manage the conference from web GUI.Log in Nano2PBX web GUI during the conference call, the participants in each conference bridge will be listed in the "System Status" Page you can kick out the participant by " Click on to kick" a participant from the conference.

Important Note:
1.NanoPBX will initiate the conference call only with G711u-Law codec with 20ms latency. Please make sure you have enabled the same codec in all the conference IP phones

2. Audio Conference Bridge supports Max 8 users irrespective of the Number of Conference bridge.

3. "Quiet Mode" and "Announce Caller Count" Cannot be enabled at the same time.

6.2. Follow Me

Navigation**: Additional Features > Follow me** Where you can enable the Call Forward for the Users Extensions

6.2.1. Follow-me Preference for Users.

This page is used to Add new follow me number which could be 'Local Extension' or an 'Outside Number' where the call is forwarded to the number added in Destination field after the ring timeout of the user called.

The Details are as follows:

1. Status: Enable/Disable Follow-me for this user.
2. MOH class: MOH class that the caller would listen while tracking the user.
3. Dialplan: DialPlan that would be used for dialing the Follow me numbers. By default this would be the same Dialplan as that of the user.
4. Destinations: List of extensions/Numbers that would be dialed to reach the user during Follow me
5. Clicking on the Add Follow Me Number will expand the window size with many options.
6. New Follow Me Number: Add new follow me number which could be 'Local Extension' or an 'Outside Number'. The selected dialplan should have permissions to dial any outside numbers defined.
7. Dial Order: This is the order in which the Follow Me destinations are dialed to reach the user.

6.2.2. Follow-me Options

Navigation: **Additional Features** > **Follow me > Follow Me Options** is where you can configure the Follow Me settings for extensions.

This is where the options like playing back the status message, Record the caller's name for announcements and playback the unreachable status message.

Important Note:
If you are Selecting "Dial outside Number" as Follow me number follow as per the Outgoing Dialing rules.

For eg: If you want add 9912254123 as Follow me number and if your Outgoing rule has Dialout Prefix as "8", then you need add " 89912254123" as Follow me number in Dial Outside number.

6.3. Directory

Navigation: **Directory** is where you can configure the Directory option for the extensions to search users by their First or last name.

Dialing the 'Directory Extension' would present to the caller, a directory of users listed in the sytem telephone directory - from which they can search by First or Last Name. To add or remove a user from the system telephone directory, edit the 'In Directory' field of the user.

The Directory Details are as follows:

Directory Extension: Specify the Extension number to dial for accessing the Name Directory.

Read the Extension number: Read the extension number to the caller before presenting dialing options.

Use first name instead of last name: Allow the caller to enter the first name of a user in the directory instead of using the last name.

6.4. Call Record

Navigation: **Additional Features** > **Call Record**: The Nano2PBX allows users to record audio during the call. Please follow the instructions below to record the call

6.4.1. Call Record Files

Here the entire Call recordings are listed. You can Download, Play or Delete the recordings

You can play the Call recordings by click on " Play" and you can select the user extension to listen.

6.4.2. Call record Settings

You can select either GSM or Wav format in the call record settings

Upload to Server:You can upload the call records to the remote server through ftp and you can schedule the uploading either by time period basis or Call record File size. Once you made the configuration, Save the changes.

Important Note:
Call recording supported by using only G711 codec.

6.5. Bulk Add

Navigation: **Additional Features** > **Bulk Add > Create range of new users**: This is where you can create many user extensions all at a time.

This is the page where we can create many bulk user extensions at a time. The details of the page is as follows:

Create: select the number of users to be created.

Users starting from Extension: select the number range from where the user extension has to start creating.

Click on the Create users followed by apply changes button to update in configuration

> **Important Note:**
> Use the 'Modify Selected Users' button from the Users page to edit any options for the created users.

7. Tools

7.1. Back Up:

The Nano2PBX configuration can be backed up locally or via network. The backup file will be used to restore the configuration on Nano2PBX when necessary.

Users could backup the Nano2PBX configurations for restore purpose by clicking on "Create New Backup" under **Tools > Back Up**.

Once the backup is done, the list of the backups will be displayed with date and time in the web page. Users can download, restore, or delete it from the Nano2PBX internal storage or the external device.

Besides local backup, users could back up the configuration a remote server via TFTP/ HTTP protocol under Web GUI-> **Tools > Back Up**.

RESTORE CONFIGURATION FROM BACKUP FILE

To restore the configuration on the Nano2PBX from a backup file, users could go to Web GUI->**Tools** ->**Backup•** A list of previous configuration backups is displayed on the web page. Users could click on of the desired backup file and it will be restored to the Nano2PBX.

 If users have other backup files on PC to restore on the Nano2PBX, click on "Browse" first and select it from local PC to upload on the Nano2PBX, and then click "Upload".

7.2. Firmware Upgrade

The Firmware Upgrade page allows you to update the NanoPBX with the latest release available, which can contain key updates, added functionalities and bug fixes. When a new firmware release is available, download it and save to your local PC. Then, browse for the file, and click the Upload button. Now your NanoPBX will display a Progress Screen and will prompt you when your NanoPBX is about to reboot. Let your NanoPBX reboot, and wait for the orange LED's to come back on.

You can always find the latest firmware from the web: http://www.allo.com/nanopbx.html

- **Important Note:**
  While upgrading the firmware, please make sure that there won't be power or network disturbances & also make sure to take back-up of configuration if any.

- During the firmware upgrade All the Front LEDs start blinking and LEDs will turn orange after the completion of Firmware upgrade

7.3. File Editor

Navigation: **File Editor**: This is where you can edit the configuration files or verify whether the configuration files are updated.

In the File Editor filed mention the configuration file you want to view. It displays the contents of the file. Here you can view modify the contents of the file. If the user wishes to create new configuration file it can be done using new file.

7.4. Asterisk CLI

Navigation: **Tools >Asterisk CLI**: This page gives an easier access to the user to execute the commands of the CLI. An Example of the "sip show peers" displays all the Nano2 user extensions with the registration status

7.5. Diagnostics

On the Nano2PBX, users could capture traces, ping remote host and traceroute remote host for troubleshooting purpose under Web GUI->**Tools > Diagnostics**.

Ping

Enter the target host in host name or IP address. Then press "Start" button. The output result will dynamically display in the window below.

8. Status

8.1. Call Detail Records

Navigation: **Call Detail Records**: This is where you can create Call Reports

To create a new Report, select the inbound calls, outbound calls, internal calls, and External calls. A list with call details will display in the Call Reports section.

By clicking on the delete button at the bottom Entire call details records (CDR) of NanoPBX are cleared

CDR can also be filtered by selecting inbound calls, outbound calls, internal calls and External calls.

You can select the number of list to shown in the listings by selecting the right dropdown box.By clicking on the Previous and Next button you can see the list pages in next and the previous pages.

You can even download the CDR in CSV format by clicking " Download" button, which makes the Download file to display top of the GUI page. whereby right Click on the " Download File" link and download the CDR using the 'Save Link As..'

8.2. Active Channels

Displays current Active Channels on the PBX, with the options to Hangup or Transfer.

When calls are in progress since there is always Refreshing Active Channels . The Current Active Channels on the PBX are displayed.

8.3. System Info

The NANO2 PBX status can be accessed via Web GUI->**Status**->**System Info**, which displays the following system information.
• **General**
• **Network**
• **Disk Usage**
• **Memory Usage**

General	You will have a summary of your general information, such as **Firmware Version,.Uptime, Asterisk Build, Server Date & TimeZone, Hostname.**
Network	Network Status you will have a summary of your Network information, such as Hostname, WAN IP Address, Subnet Mask, WAN MAC Address and Default Gateway (you may refer to Settings > Network Settings for more info).
Disk Usage	Disk you will have a summary of Disk usage and Disk free space available on the file system.
Memory Usage	Memory Status you will see total memory resources, including RAM usage, Compact Flash usage (to store voice files and voicemail) and Inbox status (for voicemail inbox).

9. APPLY Changes

Navigation: **APPLY CHANGES (On the right top of the web GUI)**

This is the button which you must press after adding / editing / deleting such things as Extensions, Voice Files, IVR's and modifying settings such as General Settings, VoIP Accounts, Network Settings, DID Routing, Firmware Upgrades, and other System Settings.

10. Managing & Handling Nano2 PBX Features

10.1. Attended Transfer

This type call transfer occurs, when before making the transfer- a user first call to the third party to inform that a transferred call is coming their way.

Follow the steps below to perform an attended or supervised call transfer:

With an active call in progress, press the *9 button. This puts the original caller on hold and gives you a dial tone on a second line. Dial the party that you wish to transfer to.

Inform the third party that they are about to receive a call

Once transferor hangs up the call, the original caller and the party you transferred to are now connected

10.2. Blind Transfer

This type of call transfer occurs, when the person receives a call, and transfers the caller to another person or call without any consultation or announcement from the transferor party.

Follow the steps below to perform a Blind or unattended call transfer:

With an active call in progress, press the *8 button. This puts the original caller on hold and gives you a dial tone on a second line. Dial the party that you wish to transfer to.

Once you transferred the call, the transferor call will get hang up.Only the original caller and the party you transferred to are now connected

> **Important Note:**
> When you do Attendant Transfer, you can retrieve the transferred call by dialing "*0" before third party picks up the call.
>
> When you want to transfer a transferred call through Blind Transfer, you can able to make either Blind or Attendant transfer. But once you started with Attendant transfer, you cannot make any more call transfer.

10.3. Conferencing

This is type of conference will allow eight parties in the bridge. It's a conference bridge where all the callers will enter into the room.

> **Important Note:**
> NanoPBX will initiate the conference call only with G711u-Law codec with 20ms latency. Please make sure you have enabled the same codec in all the conference IP phones.

10.4. Call Forwarding

Call forwarding / Diverting is a telephony feature. If your phone is unreachable / out of service area or you do not wish to receive calls, the incoming calls can be diverted to other phone numbers. The Call Forward feature enables incoming calls to be redirected to any other telephone in the country. Outgoing calls can still be made even after you have activated Call Forward. Only incoming calls are redirected to another number. (Please refer Section 2.1 to activate and operation of this feature).

10.5. Call Parking

Enables a user to hold a call and to retrieve it from another station within the group. To park a call, dial the Blind Transfer prefix code followed by Call Parking feature code i.e. *8 7000. The call is parked and the caller is held. And it will announce the parked extension (i.e. 7005).

To retrieve the call, the user can go to any phone in the group and dial the parked extension (i.e. 7005). Then call will be retrieved and connects to the retrieved user.

Use the following steps to park and retrieve the call:

1. While you are on a call, press the transfer button on your phone
2. Dial extension 7000 (default)
3. PBX announces the parked extension (e.g. 7001, 7002, etc.)
4. Hang-up the phone
5. To retrieve the parked call, dial the previously announced parked extension (7001, 7002, etc.)

10.6. Call Hold and Retrieve

Enables users to automatically hold and retrieve incoming calls without requiring the use of feature access codes. This feature is especially useful for attendants managing a large volume of incoming calls.

Any user can put a call on hold by pressing the Flash key. The extension on hold will start getting Music on hold. The same call can be retrieved back by pressing the flash key again.

10.7. BLF (Busy Lamp Field) Support

NanoPBX supports BLF, which sends information about other extensions to a phone connected to the same PBX to inform the status.

By default BLF Support is enabled. This feature is used by a receptionist or secretary for routing incoming calls.

10.8. Hard Reset

A reset button is provided in the back panel of Nano2PBX. This will hard reset the Nano2PBX. Press and hold the red button for 5-6 seconds, so that the Nano2PBX will reset to factory settings. Hard reset is for factory default, you will lose your configuration.

10.9. USB Drive

A **USB flash drive** is a data storage device that includes flash memory with an integrated Universal Serial Bus (USB) interface.USB flash drives are typically removable and rewritable. 8 GB USB drive available along with NanoPBX v2 for storing MoH, Voicemail, and Call Recording.

10.10. NPA Dialing

The North American Numbering Plan (NANP) is an integrated telephone numbering plan of 24 countries and territories: the United States and its territories, Canada, Bermuda, and 16 of the Caribbean countries.

For E.g.: If this is your phone number, 001-2127773456, then you can configure as NPA in the Dial out Rules as a 001212+XXXXXXX where 001 is a Country code and the 212 as a Area code and X is a number of digits to dial out. In this above case you have to dial only 7773456 from any of the NanoPBX Extensions. Once you dial, it will compare the number of digits which is configured in the Dial out Rules configured for NPA.

Appendix-A: Glossary of Terms

ATA

Analog Telephone Adapter: Used to connect a standard telephone to a high-speed modem to facilitate VoIP and/or fax calls over the Internet.

DHCP

Short for Dynamic Host Configuration Protocol, a protocol for assigning dynamic IP addresses to devices on a network. With dynamic addressing, a device can have a different IP address every time it connects to the network. DHCP also supports a mix of static and dynamic IP addresses.

DID

Direct Inward Dial. A specially configured phone line from the telephone company that allows for dialing inside a company directly without having to go through an attendant. A DID line cannot be used for outdial operation since there is no dial tone offered. However, it can be configured so an outside caller can reach an inside extension with a 7-digit number through the phone company's central office.

DNS

The Domain Name System is the system that translates Internet domain names into IP numbers. A "DNS Server" is a server that performs this kind of translation.

FXO

In telecommunications, a Foreign Exchange Office, or FXO, is a telephone signaling interface that receives POTS, or "plain old telephone service".

FXS

Foreign Exchange Station is the interface on a VoIP device for connecting directly to phones, faxes, and CO ports on PBXs or key telephone systems.

GATEWAY

A network point that acts as an entrance to another network.

IP ADDRESS

Every machine that is on a network (a local network, or the network of the Internet) has a unique IP number [four sets of numbers divided by period with up to three numbers in each set. (i.e. 192.168.0.100)] - If a machine does not have an IP address it cannot be on a network.

IVR

Interactive Voice Response: A system to automatically manage incoming calls, IVR can link phone callers (voice and/or touchtone) with a computer database. It can accept a question, access the company's database and provide a caller with the information they are seeking. It

can also take information from the caller, convert it to data and input that data to the database.

LAN

Local Area Network: A LAN is a group of computers and associated devices that share a common communications line or wireless link and typically share the resources of a single processor or server within a small geographic area (for example, within an office building).

NETMASK

Used by the TCP/IP protocol to decide how the network is broken up into sub-networks (ex: 255.255.255.0).

PBX

Private Branch Exchange: An in-house telephone switching system that interconnects telephone extensions to each other, as well as to the outside telephone network.

PROXY

A server that receives requests intended for another server and that acts on the behalf of the client behalf (as the client proxy) to obtain the requested service. A proxy server is often used when the client and the server are incompatible for direct connection. For example, the client is unable to meet the security authentication requirements of the server but should be permitted some

SIP

Session Initiation Protocol: An application-layer control protocol, a Signaling protocol for Internet Telephony. SIP can establish sessions for features such as audio/videoconferencing, interactive gaming, and call forwarding to be deployed over IP networks thus enabling service providers to integrate basic IP telephony services with Web, e-mail, and chat services. In addition to user authentication, redirect and registration services, SIP Server supports traditional telephony features such as personal mobility, time-of-day routing and call forwarding based on the geographical location of the person being called.

PSTN

Public Switched Telephone Network: This is defined as the regular telephone network services.

VoIP

Voice over Internet Protocol. The technology used to transmit voice conversations over a data network using the Internet Protocol. Such data network may be the Internet or a corporate Intranet.

WAN

Wide Area Network. A computer network that spans a relatively large geographical area. Typically, a WAN consists of two or more local-area networks (LANs).