



Mega PBX

Quick Installation Guide

Quick Installation Guide

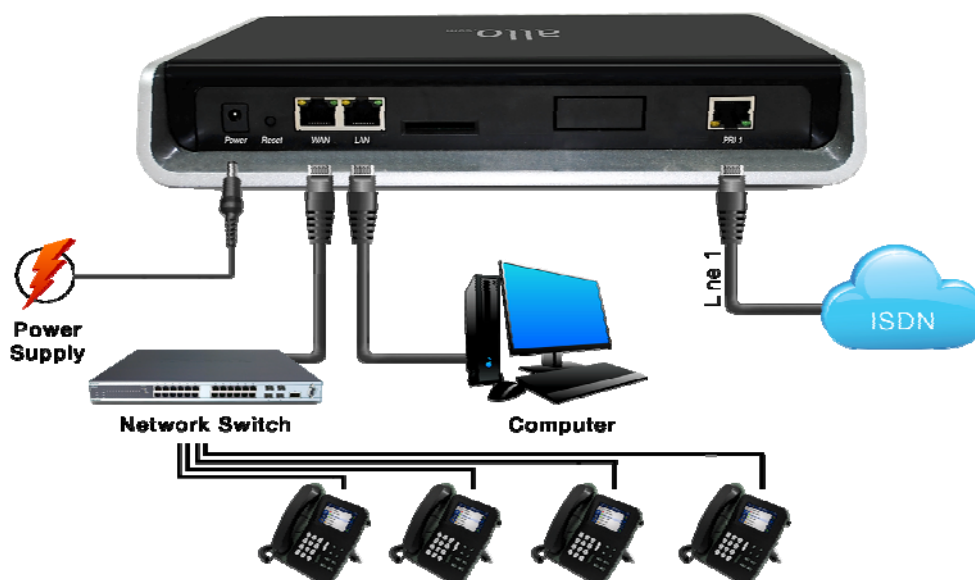
MegaPBX

Version 2.1

Table of Contents

Initial Setup of MegaPBX	4
Notification LEDs (On the Front Panel of the Gateway).....	5
Create SIP Extension	6
Create SIP Trunks	6
Dial Out Rules.....	7

Initial Setup of MegaPBX



1. Unpack the items from the box.
2. Connect the Ethernet cable from your Network switch to the WAN port of MegaPBX. Check for the blinking LEDs of the WAN port.
3. Connect a Ethernet Cable from PC to the LAN port of the MegaPBX , Enable the DHCP option in the Network Settings of the PC
4. Wait until the POWER and all the LAN and WAN LED's turn orange and remain stable on the Front Panel of your MegaPBX.
5. An IP address will be assigned to the PC in the 192.168.113.xxx series (By default DHCP server releases IP address on the LAN interface). Open Firefox browser (recommended) and type: **http://192.168.113.2** to open the login page of MegaPBX Graphical Interface.
6. Factory default login credentials are: Username: **admin** /Password: **admin**
7. Successful login takes you to the Dashboard page. Observe the WAN IP address on the dashboard, this will be used to access the GUI from the WAN interface.

8. If your network is not enabled with DHCP server, configure the WAN port ip address manually in the **SETTINGS > Network Settings** section as per your requirement.

The screenshot displays the MEGA PBX web interface. At the top, it shows 'ALLO.COM'S MEGA PBX' and 'Firmware Version 1.1.1 Tue 03 Sep 2013 11:41:14'. The navigation menu includes 'SETUP', 'ADVANCED', 'TOOLS', 'STATUS', and 'SAVE ALL'. The user is logged in as 'Administrator'. The main content area is titled 'Welcome to Dashboard' and features several status panels:

- PBX Status:**
 - SIP Station Status:** 0% (Registered: 0, Free: 0, Total: 0)
 - SIP Trunk Status:** 0% (Registered: 0, Unregistered: 0, Total: 0)
 - Call Statistics:** Total Active Calls: 0
 - Conference:** Total Active Conference Calls: 0
- System Status:**
 - Uptime:** 7 days, 22:33
 - Network Information:** LAN IP: 192.168.113.1, WAN IP: 192.168.0.125
 - Internal Memory Usage:** 19% (Used: 38 MB, Free: 155 MB, Total: 193 MB)
 - External Memory Usage:** 0% (Used: 0 MB, Free: 0 MB, Total Memory: 0 MB)
 - CPU Usage:** 0%
 - Load Average:**

	1m	5m	15m
	2.00	2.00	2.00
- Setup:** Manage basic system setup. Includes a link to 'Run the quick configuration tool'.

Notification LEDs (On the Front Panel of the Gateway):

LED 8	LED 2	LED 1	WAN	LAN	Power
Blue On-PRI 1 link is up (No Alarm)	Blue On-System Ready Blue Blink-Factory Reset	Blue ON-If the SD Card Mounted properly	Orange Blinks-WAN link is up.	Orange Blinks-LAN link is up.	Orange On-Power is on.

Important Note:

Recommended web browser to access GUI are Mozilla Firefox

We strongly recommend you to change the GUI password to reduce the possibility of a security reach (click on Tools → Admin Account Options)

Create SIP Extension

This will allow you to register SIP extension from the IP Phone/ Softphone/ FXS Gateways.

1. Click on the **SETUP >> SIP Extensions** link in the GUI, and then select **Create extension**.
2. Choose the correct Extension number & Name. Input the required credentials in the password field, click **SAVE**.
3. Click on **Manage Extension Group**, edit **DefaultSIP** group to allow the required Dial-out rule, click **SAVE**.
4. **YOU HAVE CREATED A NEW SIP Extension.**

Create SIP Trunks

This will allow you to make calls using your SIP/VoIP account

1. Click on the **SETUP >> SIP Trunks** link in the GUI, and then select **Add a SIP Provider**.
2. Enter Name your SIP Trunk (Office Line 1 for example), Credentials from VoIP service provider such as username, proxy address, authentication password etc
3. Incoming VoIP calls can be routed to particular extension, IVR or Conference etc
4. Clicks **CREATE**.

Important Note:

Make sure to click the **SAVE ALL** button in the top navigation bar, after adding / editing / deleting any Extension. The **SAVE ALL** tab turns Orange if some changes are made and not saved.

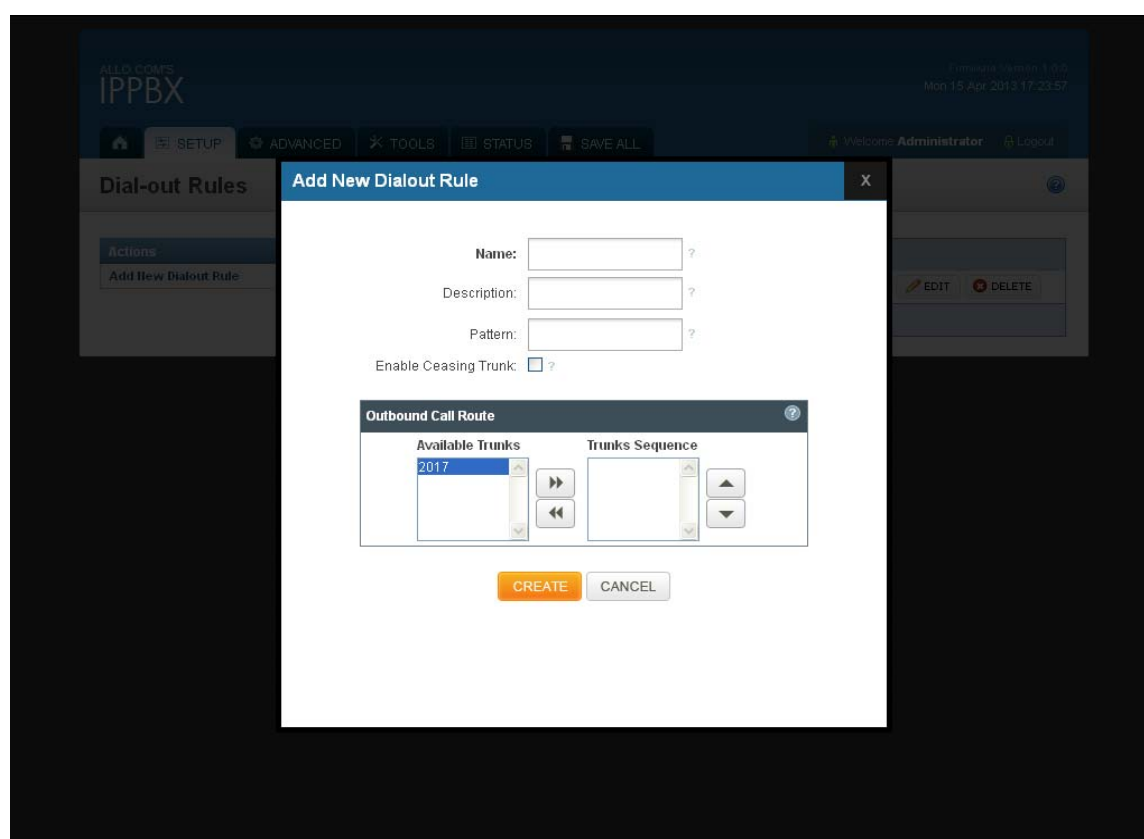
Dial Out Rules

Depending on the number you dial from an extension, the MEGAPBX will select the configured ANALOG/PRI Trunk or SIP/VOIP provider.

Add New Dial Out Rules:

One of the tricky concepts while configuring MegPBX is learning how to create good outbound routing rules. A primary usage of outbound routing rules that has been common for many years is least-cost routing. With traditional PBX systems, we might have used least-cost routing to pick one provider for local calls, another for in-state calls, and yet another for out-of-state calls. In some cases, this is still a valid usage of outbound routing rules. With a modern PBX such as Asterisk, there are many reasons to set up different outbound rules:

- Send 999/911/611 calls to an analog line to ensure that emergency calls go out through the analog lines.
- Use a different provider for international calls for cost savings. We can use VoIP for our international calls and go through an ITSP for substantial cost savings.
- Send toll-free calls through a specific provider.
- Use different rules if users dial 9 first, for example to use 9 for an outside line.



An outbound routing rule is made up of several components:

Name: Out Bound Calling Rule Name

Description: Provide the description for the Calling rule

Dialplan Patterns:

We should take care to create valid dialplans so as to ensure that our calls are handled via the appropriate trunks.

Many people get confused about dialplan pattern construction; but it doesn't have to be that difficult. The following table lists the special dialplan pattern characters and their significance:

Special Dialplan Pattern Characters	Significance
X	Matches any digit from 0-9
Z	Matches any digit from 1-9
N	Matches any digit from 2-9
[1237-9]	Matches any digit or letter in the brackets (in This Example 1,2,3,7,8,9)
.	Wildcard; matches one or more characters
	Separates dialing prefix from number

For e.g. adding '01|X.' in the Dial Pattern and selected a "VoIP Provider1" in Trunk Sequence, will allow user to dial 014567807890 numbers, so that 01 will be stripped off from the dialed digits and only 4567807890 are dialed through VoIP Provider1. (Where as a VoIP Provider1 is an account which is registered to any Internet Telephony Service Provider)

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 actual number. For e.g. adding '44X.' in Dial Pattern and selected VoIP Provider 2 in trunk sequence, PBX will allow number dialed from 44 + (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.

Trunk Sequence:

The final phase of creating the dialplan is setting the trunk sequence. Not all dialplans have multiple trunks available for use, such as the remote extensions example. In other cases, such as long-distance dialing, our preferred carrier would be first, followed by a backup provider in case we max out the available channels, or if the system was unable to contact the primary provider. We can choose our trunk sequence from the list of available trunks with priority.

**YOU ARE NOW ABLE TO MAKE/RECEIVE BASIC CALLS THROUGH THE
MEGAPBX**

**THE FULL VERSION MANUAL WILL PROVIDE YOU DETAILED INSTRUCTIONS
ABOUT THE FULL FEATURE SET OF THE MEGAPBX.**

THANK YOU!