Mega PBX + PRI User Manual





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About this manual

This manual describes the Allo product application and explains how to work and use it major features. It serves as a means to describe the user interface and how to use it to accomplish common tasks. This manual also describes the underlying assumptions and users make the underlying data model.

Document Conventions

In this manual, certain words are represented in different fonts, typefaces, sizes, and weights. This highlighting is systematic; different words are represented in the same style to indicate their inclusion in a specific category. Additionally, this document has different strategies to draw User attention to certain pieces of information. In order of how critical the information is to your system, these items are marked as a note, tip, important, caution, or warning.



- **Bold** indicates the name of the menu items, options, dialog boxes, windows and functions.
- The color <u>blue</u> with underline is used to indicate cross-references and hyperlinks.
- Numbered Paragraphs Numbered paragraphs are used to indicate tasks that need to be carried out. Text in paragraphs without numbering represents ordinary information.
- The Courier font indicates a command sequence, file type, URL, Folder/File name
- e.g. <u>www.allo.com</u>

Support Information

Every effort has been made to ensure the accuracy of the document. If you have comments, questions, or ideas regarding the document contact online support: <u>http://support.allo.com</u>



Table of Contents	
About this manual	3
Document Conventions	
Support Information	
1. Product Introduction	7
1.1 Overview	7
2. Getting Started With MegaPBX	8
2.1 Hardware Setup	ç
2.2 Equipment Structure	ç
2.3 Access the web GUI:	
3. Setting up Features	
3.1 System Dashboard	
4. Setup	
4.1Extensions	
4.1.1 SIP Extensions	
4.1.2 SIP Extension Group	16
4.2 Trunks	
4.2.1 SIP Trunks	
4.2.2 PRI Trunks	20
4.3 DID Routing	22
4.4 Dial-out Rules	23
4.5 Time -based Routing	25
5. Features	
5.1 IVR	
5.2 Voice Files	29
5.3 Conference	32

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	5.4 Call Queues	.32
	5.5 Voicemail Groups	.34
	5.6 Directory Entries	.35
6. Ao	dvanced	.37
	6.1 Feature Settings	.37
	6.1.1 Extension	.37
	6.1.2 Voicemail	.38
	6.1.3 Call Park	.39
	6.1.4 Call Back	.40
	6.1.5 FAX	.40
	6.1.6 Conference	.41
	6.1.7 Voice Prompts	.41
	6.2 ISDN PRI Settings	.42
	6.3 SIP Global Settings	.44
	6.3.1 Port Settings	.44
	6.3.2 NAT Settings	.45
	6.3.3 Registration Timer	.46
	6.3.4 Call Recording	.47
	6.4 CDR Settings	.47
	6.4.1 Radius Configuration	.47
	6.4.2 FTP Configuration	.48
	6.4.3 Upload History	.49
7. Sy	/stem	.51
	7.1 Network	.51
	7.2 Date/Time	.52

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	8.1 Diagnostics	54
	8.2 Backup/Restore	55
	8.3 Upgrade Firmware	56
9. S	tatus	57
	9.1 Call Reports	57
	9.2 SIP Station	57
	9.3 SIP Trunks	58
	9.4 PRI Span	59
	9.5 Current calls	60
	9.6 Current Conferences	60
	9.7 Network	60
10.	Administrator	61
	10.1 Reboot	61
	10.2 Call Manager Reload	61
	10.3 Web Settings	61
	10.4 Email Settings	61
	10.5 Logout	62

1. Product Introduction

1.1 Overview

The Allo.com's MegaPBX, a SIP-based, affordable, feature-rich converging communication platform designed to meet the communication requirements for small to medium sized enterprises.

The MegaPBX provides the cutting-edge IP-based communications that businesses demand while leveraging existing infrastructure and providing a smooth transition into IP telephony. Based on open standard SIP, the MegaPBX can easily integrate into and interoperate with other components of your existing communications network while providing a rich set of features to reduce costs and increase productivity.

The MegaPBX also includes a Setup wizard and an intuitive user interface that allows users to quickly configure extensions, Voicemail, fax mail, Voicemail& fax to email, conference bridges and other enhanced features can be turned on with minimal effort via the web configuration interface.

Here, 4 models are available and all supporting up to 200 extensions and 50 simultaneous calls, 30 people conference room with full features (invite, schedule, email, kick, mute...). FXS ports for FAX machines only.

MegaPBX-PRI supports ISDN PRI protocol and adopts standard T1/E1 trunk interface to realize docking with traditional PBX /PSTN.

Model No - aMP1

Equipment Packaging

- 1) One MegaPBX unit
- 2) One 12 Volt power adapter
- 3) One Ethernet cable

2. Getting Started With MegaPBX

Initial Setup of IP/PRI/FXS/BRI PBX



1. Plug one end of the RJ45 Ethernet cable into your Network Switch

2. Plug the other end of the RJ45 Ethernet cable into the WAN port of the MegaPBX

3. Connect a PC to the LAN port of the MegaPBX; Enable the DHCP option in the Network

Settings of the PC

- 4. Plug the Power Adapter included into an available power outlet
- 5. Plug the other end of the Power Adapter into the "DC-IN" port of the MegaPBX
- 6. The MegaPBX will power up (Boot up time takes about 160secs)



Use Straight-through Ethernet cable to connect between the MegaPBX to Router/Switch/PC



2.1 Hardware Setup

2.2 Equipment Structure

MegaPBX-PRI



Figure 2: MegaPBX-PRI

Interface	Description
Power	Connect the power adapter, 12VDC, 3.5A
Reset	Reset button for factory default.
WAN	Standard 10/100BASE-TX Ethernet Interface for WAN
LAN	Standard 10/100BASE-TX Ethernet Interface for LAN
Memory Card	8GB Storage for voice mails and voice files, IVR
PRI 1	E1/T1 ports with line link LED indicators (Orange and Green)

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.



2.3 Access the web GUI:

MegaPBX-PRI Web GUI can be accessed either through WAN or LAN interface. Steps to Access the GUI during the initial setup through LAN interface:

- 1. Make the setup as mentioned in the hardware setup
- 2. Change the Network setting of the PC is set in automatic mode (i.e. DHCP mode). An IP address will be accessed to the PC in manual mode (i.e. Static IP mode). Assign the IP address to the PC in the range of 192.168.113.xxx series (E.g:192.168.113.10), net mask as 255.255.255.0 and gateway & DNS as 192.168.113.1.
- Launch the web browser and enter the URL <u>http://192.168.113.2</u> (Default LAN IP address) to open the login page of MegaPBX-PRI Graphical Interface.

	ALLO.COM'S Mega PBX-PRI
1	admin
•	•••••
	Login



4. Login using the default username & password (Default: Username: admin; Password: admin). 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.

After successful login we get a MegaPBX-PRI Home Dash Board. To guarantee the system safety, When login for the first time, we need to modify Password. For modifying the password go to TOOLS->Admin Account Option



The Recommended web Browser to access GUI is Mozilla Firefox only.

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.

WARRANTY

Hardware Warranty: 1 year

If the MegaPBX PRI was purchased from a Distributor/reseller, please contact the company where the device was purchased for replacement, repair or refund. If the device was purchased directly from Allo.com, contact our Technical Support Team for a RMA (Return Materials Authorization) number before the product is returned. Allo.com reserves the right to remedy warranty policy without prior notification.

Use the power adapter provided with the ALLO MegaPBX PRI. Do not use a different power adapter as this may damage the device. This type of damage is not covered under warranty.



3. Setting up Features

Setting up your browser for working with MegaPBX-PRI is simple. In order to run this application appropriately the following settings are to be configured.

3.1 System Dashboard

ALLO MegaPBX-PRI Dash Board summarizes the MegaPBX-PRI status with a graphical display. Detailed status of an individual entity is available under the Status Tab or it can be directly accessed by clicking on more.

20-February-15 10:04:59 ar	m		Firmware Version: 4.0.0		Refresh	Welcome Administrator
Dashboard > D	ashboard	0				
Setup						
Features	* Please config	ure Media IP in the same ser	ies of WAN IP. [More]			
Advanced	SIP Status	PRI Span / Call Status	System Status	Memory Usage		
System						
Tools	SIP Tru	Inks [More]		SIP St	ations [More]	
Status	_					
	Regisi	istered:		Regist Free	tered:	
	Total:	0		Total:	0	

Figure 4: SIP Status

Please change the default administrator password to alphanumeric, to prevent hacking.



4. Setup

4.1Extensions

Here, user can configure Feature (*) Code for different Call Features.

4.1.1 SIP Extensions

Navigate through Setup > Extensions > SIP Extensions

SIP Extensions are the unique number mapped to person that can be reached and be able to place calls.

xtensions	Show	All 🚩 entries					Search:	
→ SIP Extensions		Extn No.	Name	Email	Group	0	Options	
 SIP Exten Group 		3001	3001	. 3001		(user defined)	1	×
unks		3002	3002	3002		DefaultSIP	1	×
ID Routing		3003	3003	3003		DefaultSIP	1	×
ial-out Rules		3004	3004	3004		DefaultSIP	1	×
ime-based Routing		3005	3005	i 3005		DefaultSIP	1	×
Features		3006	3006	3006		DefaultSIP	1	×
		3007	3007	3007		DefaultSIP	1	×
Advanced		3008	3008	3008		DefaultSIP	1	× ×
System	Showi	ng 1 to 300 of 3	00 entries				First Previous	1 Next Last
Tools								

Figure 5: SIP Extensions

Click Add New, to create an Extension



Extn	Number	3001		
First	Name	3001		
Last Name		3001		
Emai	l.			
Pass	word	••••		
Ext	ension Setting	5		
	Enable email alerts on arrival of new voicemail			
	Send voicemail as attachments with alert email			
Message waiting indication				
	Enable Video (Calling		
	Enable T38 Fa:	()		
	Enable Call Re	cording		
	Qualify			
		Group DefaultSIP 💌		

Figure 6: Create an Extension

Extension Range	4000	to 4010	
	Qualify [
	Group Defaul	tSIP 🔽	
Voloomail Cotti			-
	igs		
Enable em	ail alerts on arrival o	f new voicemail	
Send voice	email as attachments	with alert email	
Message \	vaiting indication		

Figure 7: Create Range

Extn Number	Extension Number of endpoint (e.g.: IP phone) will use to
	authenticate with the MegaPBX, eg: 4000.
First Name	A character based first name for the user eg:"Bob".



Last Name	A character based last name for the user eg:"Jones".
Email	E-mail address for the user eg:"Bobjones@xyz.com".
Password	Password used to authenticate your phone and voicemail pin.
Extension Settings	This allows the user to either enable or disable email alerts or
	message waiting indication.
Add Extension Range	Range of Extensions can be added at once with required voicemail
	settings, Eg: from 4000 to 4010.

Show/ Hide Advance Options

Advance options, allows enabling the Features, Codec Configuration and Dialout Settings for a specific user. Default SIP group is disabled when Advanced Options is enabled.

Advanced Options	
DTMF	Set default DTMF mode for sending DTMF digits. Options:
	• INBAND – sent along with audio (requires 64 kbit codec
	- alaw, ulaw)
	 INFO – sent as SIP INFO messages
	 RFC2833 – sent as RTP packets
	• AUTO – System automatic selects the mode. Uses
	RFC2833 if offered, inband otherwise.
	Default: AUTO
Features	Enable or Disable the desired features like voicemail, call queue,
	conference, and call back, call pickup, call park, etc
Codec Configuration	Choose the available Codecs and set priority in the order in which
	Mega PBX should prefer to send and receive audio. Supported
	codecs are alaw, ulaw, G.729, G.722
Dialout Settings	Allow or deny the preferred dial out plans for user.

4.1.2 SIP Extension Group

Navigate through Setup > Extensions > SIP Extension Group

By Default there is an extension group called "Default SIP". You can limit or enable PBX features for extensions by creating Extensions group.

tensions	Show All 🗙 entries		Search:
SIP Extensions	Group Name	Type	Options
SIP Exten Group	DefaultSIP	DEFAULT	
unks			
D Routing			
al-out Rules			
ne-based Routing			
Features			
Advanced			
System	Showing 1 to 1 of 1 entries		First Previous 1 Next Last

Figure 8: SIP Extension Group

	DefaultS	BIP	
Features			
Voicemail	Call Queues	Conference	🔲 Callback
NAT	Subscriber	🔲 Call PickUp	🔲 Call Park
Codec Configu	ration		
Avai	lable	Active	
G722	× ×	G729	

Figure 9: Create Extension Group

Group Name	Descriptive name for extension group
DTMF	Set default DTMF mode for sending DTMF digits. Options:
	 INBAND – sent along with audio (requires 64 kbit codec -
	alaw, ulaw)
	 INFO – sent as SIP INFO messages
	RFC2833 – sent as RTP packets
	 AUTO – System automatic selects the mode. Uses
	RFC2833 if offered, inband otherwise.
	Default: AUTO
Features	Enable or Disable the desired features like voicemail, call queue,
	conference, and callback, call pickup, call park etc
Codec Configuration	Choose the available Codecs and set priority in the order in which
	Mega PBX should prefer to send and receive audio. Supported codecs
	are alaw, ulaw, G.729, G.722.
Dialout Settings	Allow or deny the preferred dial out plans for this extension group.

4.2 Trunks

4.2.1 SIP Trunks

Navigate through Setup >Trunks > SIP Trunks

SIP Trunks provide the interface to any SIP companion such as VoIP service provider, any SIP server or SIP clients. Add different types of interfaces, and configure the signaling & media settings for each trunk.

Show All 💟				and a second	
	entries			Search:	
A	ccount Name	Inbound Route-1	Inbound Route	·2	Options
5	566	Extension - 3001	Did Extension -	9	× ×
				First Dravious	1 Next Last
Showing 1 t	to 1 of 1 entries			First Frevious	T HOLE FORE
		Account Name	Account Name Inbound Route-1 5566 Extension - 3001	Account Name Inbound Route-1 Inbound Route- 5566 Extension - 3001 Did Extension - 3001	Account Name Inbound Route-1 Inbound Route-2 Inbound Route-2 5566 Extension - 3001 Did Extension - 9

Figure 10: SIP Trunks

Account Name	5566		
Username	5566		
Proxy Address	192.168.0.100	: 5060	
Outbound Proxy Address			
Authentication			
Auth. Username	5566		<u>.</u>
Password	••••		
Registration mode			
Registrar Address	192.168.0.100		
Bridge Pin			
Inbound call handling			
Route-1	Extension 💉 3001	~	
Route-2	DID Extensior 💙		~

Figure 11: Create SIP Trunk

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Account Name	Descriptive name for the SIP Trunk for user's reference. e.g.:
	5656.
Username	Username of endpoint (e.g.: IPPBX) will use to authenticate with
	the Mega PBX, e.g.: 5656
Proxy Address	IP address or hostname with port of the endpoint (VOIP Service
	Provider or IPPBX) where the calls will be diverted. Default port
	no.: 5060, e.g.: 192.168.0.100.
Outbound Proxy Address	IP address or hostname with port of the outbound proxy server.
	This ensures that all the SIP packets are sent via specified proxy.
	Specifying the port is not mandatory. Default port no.: 5060
	e.g.: 192.168.0.222:5062 OR 192.168.0.222
Authentication	Enable, if Authentication is required by the End point (VOIP
	Service Provider or IPPBX)
Auth. Username	A username to use only for registration. e.g.: 5656.
Password	Password to authenticate registrations and inbound &
	outbound calls.
Registration Mode	Enable, if Registration to the End point (VOIP Service Provider or
	IPPBX) is required.
Registrar Address	IP address or hostname with port of the Registrar server where
	Mega PBX must register to. Specifying the port is not
	mandatory. Default port no.: 5060
	e.g.: 192.168.0.222:5062 OR 192.168.0.222
Bridge PIN	You can set a PIN for outgoing calls on SIP trunk, thus you will
	set one more level of security. Leave it blank for unsecured
	mode. E.g.: 111.
Inbound Call Handling	Route incoming calls to any destination like extension, queue,
	and voice mail group, IVR, conference, DID Extensions etc.
	E.g.: Route To Extension 2000.

Show/ Hide Advance Options

Advance options, allows enabling the options like Features, Codec Configuration and Dial out Settings for a specific Trunk.

Advanced Options	
DTMF	Set default DTMF mode for sending DTMF digits. Options:
	 INBAND – sent along with audio (requires 64 kbit
	codec - alaw, ulaw)
	 INFO – sent as SIP INFO messages
	• RFC2833 – sent as RTP packets
	• AUTO – System automatic selects the mode. Uses
	RFC2833 if offered, inband otherwise.
	Default: AUTO
Registration Timeout	You can set the length of time in seconds between registration
	attempts (the default is 20 seconds).
Features	Enable or Disable the desired features like NAT, FAX, DID
	Routing, video calling and call recording.
Codec Configuration	Choose the available Codecs and set priority in the order in
	which gateway should prefer to send and receive
	audio. Supported codecs are alaw, ulaw, G.729, G.722

4.2.2 PRI Trunks

(Only for Mega PBX products with PRI support)

Navigate through Setup > Trunks > PRI Trunks

PRI Trunks provides the interface to any ISDN PRI companion such as PRI service provider or any other ISDN PBX. Create an interface for each span.



	Show	🛛 📶 🔛 entr	ies					Search	(C)	
<s< th=""><th></th><th>Span Name</th><th>٥</th><th>Trunk Name</th><th>٥</th><th>Outbound Caller-ID 🔅</th><th>Inbound Route-1</th><th>Inbound Route-2</th><th>٥</th><th>Options</th></s<>		Span Name	٥	Trunk Name	٥	Outbound Caller-ID 🔅	Inbound Route-1	Inbound Route-2	٥	Options
PRI Trunks		PRI1		PRItrunk1			DID Extension - 1X	None		🖉 🗙
Routing										
out Rules										
-based Routing										
Features										
Advanced										
								_		
Gystem		tion of the of the	f 1 er	ntries				First Prev	ious 1	Next Las
System Tools	Show	'ing 1 to 1 o								

Figure 12: PRI Trunks

Span	Name	PBI1 😒	^
Trunk	Name	PRItrunk1	
Bridge	e Pin		
Outbo	ound Caller-ID		
Caller	-ID Base Number		
Inbo	und call handlin	g	
Rou	ite-1 DID Extension	r 💙	
Pat	ttern 1X	Trim Digits 0	
Rou	ite-2None	~	
Feat	ures		
	🔲 DII	D Routing	~

Figure 13: Create PRI Trunk

Span Name	Select the appropriate PRI Spans.
Trunk Name	Descriptive name for the PRI Trunk for user's reference.
Bridge PIN	You can set a PIN for outgoing call on PRI trunks, thus you can set



	one level of security. Leave it blank for unsecured mode.
Outbound Caller ID	Configure the Caller ID Number that would be applied for
	outbound calls over this trunk.
Inbound Call Handling	Route incoming calls to DID extensions or Dial out rules. Route-1
	and route-2 can be defined. E.g.: Route-1: DID Extension pattern:
	1X. Trim Digits : 0.
DID Routing	Enable or Disable the DID Routing for this trunk

4.3 DID Routing

Navigate through Setup >DID Routing

DID Routing is a feature that enables incoming calls to be routed directly to the selected extension or trunk.

xtensions	Show All	Y entries	Search:		
runks		DID	\$ Routing to	0	Options
ID Routing		1234	Extension - 3001		💉 🗙
ial-out Rules		3000	Extension - 3001		🗡 🗙
 Features Advanced 					
 Advanced System 					
• Tools	Chowing 1	to 2 of 2 optrios		6	ret Dravious 1 Novt Last
	SHOWING 1			ISC FIEVIOUS I WEAT LOST	

Figure 14: DID Routing

DID Number	It is the number provided by the service provider for the
	incoming calls. e.g.: 5023
Route To	Specify where the incoming calls should be routed.
	e.g.: Conference.
ADD DID Range	Here you can specify the range of the DID numbers provided
	by the service provider, e.g.: 5000 to 5010.

MegaPBX PRI User Manual

Start Extension to Map	Here you can specify the starting number of the SIP extension
	which is to be mapped with the DID numbers. e.g.:2000



Figure 15: Add a new DID

)ID Range	3000	To 3010
itart extension to map	3001 💌	
	3000-3001	Map Remove Remove All

Figure 16: Create DID Range

4.4 Dial-out Rules

Navigate through Setup > Dial-Out Rules

Dial out Rules is used to configure the system to judge outgoing calls via trunks and also used to select a least cost routing provider based on the prefix configured.

@ allo.com

ensions	ions 💦 All 🕑 entries						Search:		
nks		Name	\$	Description	\$	Pattern	\$	Options	
Routing		Rule1		Rule		12345		💉 🗙	
il-out Rules									
ne-based Routing									
Features									
Advanced									
System									
Tools									
	Showing	1 to 1 of 1 entries					First F	Previous 1 Next Last	

Figure 17: Dial-Out Rules

Name	Rule1
Description	Rule
Pattern	12345
Enable Seizing Trunk	
Trim Digits	4
Prepend Digits	4
Outbound Call Route	
Available Trunks	Trunks Sequence
	SAVE

Figure 18: Create Dial-Out Rule

@ allo....



Name	Descriptive name for the Dial-Out rule for user's reference.
Description	Provide the description for the Dial-Out rule. (Optional)
Pattern	Specify the pattern to match the dialed string of the incoming call.
	Pattern:
	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.
	e.g.: X. – match at least one digit
	988XXXX – match 988 followed by 4 digits
Enable Ceasing Trunk	Seize the line and dial out.
Trim Digits	Number of digits to trim from the beginning of dialed number. Eg.
	Pattern=52xx , Trim Digits=1, dial out digits = 2xx
Prepend Digits	Entered digits add to the beginning of pattern to be Dialout.
Outbound Call Route	Select the preferred trunks where calls are to be routed for this Dial
	Out rule. Ordering of the trunks in the "Selected" column indicates
	the order in which call flows on failure.

4.5 Time -based Routing

Navigate through Setup > Time-Based Routing Groups

Time-base routing, routes calls to different locations based on the time of day and day of week, when a call is made.

nsions	ns Show All 🝸 entries					Search:			
s		Group Name	Description	0	Rules	¢ Op	tions		
Routing		Test	Testing		<u>Rules</u>	1	×		
out Rules									
-based Routing									
eatures									
dvanced									
System									
,									
ools	Chousing	1 to 1 of 1 options				ect Drouiou			
14	Showing	ar corror relicies			E.	rst previot	12 T NEXC LOS		



Group Na	ame		Test		
Descripti	on		Testing	1	
Assigned Ru	iles				+ ADI
Duration	Route	То	Days	0	ptions
Create/Edit	Rule	1			SAVE CLOS
Route Type	9	Extension	n 💌		
Duration (hh:mm)		00:00]] (from) [23:59](to)
Days		🔲 All	🔲 Sur	Mon	⊡ Tues
		⊘ Wed	⊡ Thur	🔽 Fri	🔲 Sat

Figure 20: Create Time route Groups

Group Name	Descriptive name for the Time-Based Routing Group for user's
	reference.
Description	Provide the proper description for the Time based routing rule.
	(Optional)

@ allo....



Route Type	Select the destination where the call is routed to on matching
	the time. The destination can be any of these – Extension,
	Trunk, IVR, Queue, Voicemail Group, Fax to E-Mail
Route To	Setting the destination for the incoming route. The destination
	can be any of the created extensions, Trunks, IVRs, Queues
	and so on.
Duration	Specify the time range for which this routing rule will apply.
	Format: hh:mm
Days	Select the day/days during which this routing rule will apply.

Make sure that the current date and time are configured correctly under **System**> **Date/Time Configuration.**



5. Features

5.1 IVR

Navigate through Features >IVR

Interactive voice response is a pre recorded interactive operator that allows an automatic separation of the incoming calls through the sequence of interaction with a multiple choices of menu with telephone callers.

Features 🔉	Show A	II 🔛 entries			Search:
R		Name	Description	Sequence	Options
ice Files		Test	Testing	Sequence	💉 🗙
nference					
l Queues					
cemail Groups					
ectory Service					
Advanced					
System					
Tools	Showing	g 1 to 1 of 1 entrie	s		First Previous 1 Next Last

Figure 21: IVR (Interactive Voice Response)



IVR Name	Test	
Description	Testing	
equence Association		
Seq# 5 💌 Actio	on Hangup	Add
equence List		
eq# Action		Option
Ring	1 💌	×
Allow Dial Extension	n None 🛩	×
Go To	Extension 💌	3001 💌 🔀
Hangup	101	×

Figure 22: Create IVR

IVR Name	A character based name for the IVR.
Description	Detailed description of the IVR.
Sequence Association	Specifying how incoming calling should be handled sequentially.
Sequence List	List of operations to be handled.

5.2 Voice Files

Navigate through Features > Voice Files

Voice files are the audio files that are going to be played during the IVR play back and these files can be uploaded or they can be recorded, the user also has an option to associate key press with his voice file.



Features >	Show	All 🔛 entries								Search:	
ec.		Name	Description	0	Format	٥	Size	٥	Keymaps	0	Options
e Files		vfile1			wav		137 kb		<u>Keymaps</u>		💉 🗙 🛨
ference		vf2			wav		130 kb		<u>Keymaps</u>		💉 🗙 🛨
Queues											
email Groups											
ctory Service											
Advanced											
System											
Tools	Showin	ig 1 to 2 of 2	entries		_	-			First	Previ	ous 1 Next Last

Figure 23: Voice Files

Example 1: Voice file without Key press

"Welcome to Allo.com! If you know your party's extension, please dial it now.

Example 2: Voice file with Key press

Press 1 for Customer Service Press 2 for Sales

Press 0 for Reception



Flienanie	Vo	icefile	
Descriptio	n Te	sting	
ile Upload			
Upload Type	Browse fro	om PC 🛛 💌	
File	Browse	. vf2	
Key Associat	ion		
Associate keyp	resses with	this voice file	No 💌
# 💽 🖌 Go	TO Extension		Add
Associated K	eys		
Key Ac	tion	Option	

Figure 24: Create Voice file

File Name	A character based name for a voice file.
Description	Detailed description for the voice file.
File Upload	Upload the voice file to device.
Key Association	With associated key press option it will route incoming call to
	the specified destination.
Associated Keys	The sequence in which a key and its associated action is given.



Mega PBX Supports".wav" format only



5.3 Conference

Navigate through **Features** > **Conference**

The Mega PBX supports password protected conference bridges that allow up to 50 simultaneous participants from any trunks or Internal Extensions.

Conference Number	Specifies a unique number which can be used to enter the				
	conference room, e.g.: 6000.				
Date	Specifies the date when the conference has to be scheduled,				
	e.g.: 26/2/2014.				
Start Time	Specifies the time to schedule the conference. E.g.: 12:30.				
Duration	The duration during which the conference will be active, e.g.:				
	180 minutes.				
Admin Pin	Specifies the PIN for the admin user to join the conference, e.g.:				
	555.				
Speaker Pin	Specifies the PIN for the speaker and this user can speak and				
	listen, e.g.: 666.				
Listener Pin	Specifies the PIN for the listener and this user can only listen				
	but cannot speak, e.g.: 777.				
Participants	Specifies the privilege to be assigned and it can be Admin,				
	Speaker or Listener.				
Add Dynamic Participants	This option enables the user to add the participants				
	dynamically from the web GUI.				
Outbound Dialing	This option enables the user to add the participants				
	dynamically over the trunks.				
Maximum Participants	Specify the maximum participants to be allowed, e.g.: 10.				

5.4 Call Queues

Navigate through Features > Call Queues

Call Queues are used to distribute calls in the order of arrival to the first available agent. The system answers each call immediately and if necessary holds it in a queue until it can be directed to the next available agent. This feature is used to balance the workload among the agents.



eatures 💦 🗲 🗲	Show 🖌	All ⊻ entries					Search	:
		Queue Name	\$	Strategy	0	Description	0	Options
Files		test		Ringall		testing		× ×
erence								
Queues								
mail Groups								
ory Service								
.dvanced								
ystem								
ools	Showin	g 1 to 1 of 1 entries	_	_	_	_	First Previ	ous 1 Next Last

Figure 25: Call Queues

Queue Name	test
Strategy	Ringall
Description	testing
Queue length	
Queue Members	
Erequency	
Play this sound	None V
Queue timeout	5 (Secs)

Figure 26: Create New Call Queue

Queue Name	A character based unique name for the queue.
Strategy	Calls are distributed among the members handling a queue



	with one of the several strategies like Ring all, Round robin,
	Random etc.,.
Description	Detailed description of the queue.
Queue Length	Used to decide the length of the queue.
Queue Members	Add the members to distribute the incoming calls.
Announcement Options	You can set timeout for queue, member and wrap up.

5.5 Voicemail Groups

Navigate through Features > Voicemail Groups

Voicemail Groups feature allows sending the voice message to multiple people or a group of people.

1 ografob 🖉	Show All	💌 entries		Search:
R		Group Name	Description	Options
ice Files		Voicemail	Testing	
nference				
ll Queues				
cemail Groups				
ectory Service				
Advanced				
System				First Bravious 1 Novt Last
System Tools	Showing	1 to 1 of 1 entries		FIISC FREWINGS I NEAR LOSE

Figure 27: Voicemail Groups



Group Name	Voicemail	
Description	Testing	
Members		
Available	Selected	

Figure 28: Create New Voicemail group

Group Name	A character based unique name for a voicemail Group
Description	Detailed description of the Voicemail Group.
Members	Specifies the members to receive the voicemail from the list of
	available members.

5.6 Directory Entries

Navigate through Features > Directory Service

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 system telephone directory - from which they can search by First or Last Name.

reatures y	Show All	⊻ entries				Search:
	No.	 Extension 	٢	First Name	\$ Last Name	Options
e Files	1	3001		3001	3001	
ference						
Oueues						
email Groups						
tory Service						
Advanced						
System						

Figure 29: Directory Entries

Extension Number	Specify the Extension number to dial for accessing the Name
	Directory, e.g.: 555.
First Name	Allow the caller to enter the first name of a user in the
	directory, e.g.: JOHN.
Last Name	Allow the caller to enter the Last name of a user in the
	directory, e.g.: RAJ.
Import Directory	Import the directory entries by browsing the corresponding
	file. File has to be in .CSV format.
Export Directory	Export the Directory Entries by specifying the filename. The
	File will be exported to the local computer in .CSV format

@ allo....



6. Advanced

6.1 Feature Settings

Navigate through Advanced > Feature Settings

Feature Settings allows to modify the basic call related functionalities like call pickup, Conference, Callback etc...

6.1.1 Extension

Features	Extension	Voicemail	Callpark	Callback	Fax	Conference	Voice Prompts
Advanced >							
eature Settings	Pickup	0		* 9			
SDN PRI Settings	Confe	rence		* 900			
IP Global Settings	Dynar	nic Conference		* 901			
DR Settings	Voicer	nail		* 800			
System	Direct	0.02		* 001			
Tools	Direct	ory		001			
Status							

Figure 30: Feature Settings

Pickup	This feature code is to pick up the ringing extension from
	another extension if the party is not available in the desk. By
	Default Feature code is "*9". For e.g. if *9 is your call pick up
	code, by dialing *91002 from any other extension you can
	attend 1002 phone extension.
Conference	Using this feature code you can enter into the conference
	room. By Default the Feature code is "*900", i.e. Dial *900
	from your phone and It will prompt you for the Conference
	Number and the PIN and it will let you enter into the
	conference room.
Dynamic conference	Using this feature code you can enter into the conference
	room and also add users in to it. By Default the Feature code
	is "*901", i.e. Dial *901 from your phone. It will prompt you



	for the Conference Number and the PIN and it will let you
	enter into the conference room. To add users into the
	conference *0 then followed by extensions.
Voicemail	This is to assign the code for accessing the voice mail. This will
	allow end users to change their personal settings for voice
	mail handling. By dialing to this number, any users who are
	registered to Mega PBX can access the Voice Mail. I.e. dial
	*800 from your phone and follow the instructions.
Directory	Using this Feature code you can dial the other extension with
	first and last name configured in the PBX.

6.1.2 Voicemail

Advanced Enable Voicemail Image: Constraint of the state of the st	Features	Extension voicement campark campack rax connerence voice Prompts
ture Settings Enable Email Alerts on new Voicemail Image: Constraint of the section	Advanced >	Enable Voicemail
N PRI Settings Enable Voicemail Attachments in Email Alerts Image: Constraint of the set	ure Settings	Enable Email Alerts on new Voicemail
Global Settings Enable Voicemail Web Access Image: Constraint of the constrai	N PRI Settings	Enable Voicemail Attachments in Email Alerts 🛛 🔽
R Settings Enable Message Waiting Indication System Customized Voicemail Message Subject New message \${V_M_MSGNUM} in mailbox \${V_M_MAILBDX}	Global Settings	Enable Voicemail Web Access
System Customized Voicemail Message Tools Subject	Settings	Enable Message Waiting Indication
Subject New message \${VM_MSGNUM} in mailbox \${VM_MAILBDX}	System	Customized Voicemail Message
	Tools	Subject New message \${VM_MSGNUM} in mailbox \${VM_MAILBDX}
Body Hi \$(UM_MARL), You have a \$(VM_DUK) long new voicemail Body message (number \$(UM_DESRUM)) in mailbox \$(UM_MAILBOX) from \$(UM_CALLERID), on \$(UM_DATE).	Status	Hi \$(VM_NAME), You have a \$(VM_DUR) long new voicemail message (number \$(VM_MSGNUM)) in mailbox \$(VM_MAILBOX) from \$(VM_CALLERID), on \$(VM_DATE).

Figure 31: Voicemail

Enable Voice mail	With voice mail, callers can leave messages when you are
	busy, unable to answer phone calls, or when the IP phone is
	off-line.
Enable Email alerts on new	Helps to send an email when someone leaves a voice
Voicemail	message.
Enable Voicemail	Helps to attach voicemails in your Emails.
attachments in Email alerts	
Enable Voicemail access	Helps to access your voicemail by using the voicemail access



	number followed by the password.
Enable message waiting	Helps the subscribers to know that a voice message is
indication	waiting.
Customized Voicemail	Allows you to customize the Voicemail message Notifications,
message	which will be sent to the specified Email-ID's.

6.1.3 Call Park

Features	Extension	Voicemail	Callpark	Callbac	k Fax	Conference	Voice Prompts	
- Advanced 🔉					5			
Feature Settings	Parkir	g	*	700	(extension)			- 1
(SDN PRI Settings	Parke	d		701	(extension)			- 1
IP Global Settings	Numb	er of slots		10				- 1
CDR Settings	Timeo	ut		30	(seconds)			- 1
System								- 1
Tools								_

Figure 32: Call Park

Parking	Using this feature code you can park the number of calls. By
	Default the Feature code is "*700". i.e. park a caller by dialing
	*700 by using transfer button on your phone
Parked	Using this feature code you can retrieve the parked calls. By
	Default the Feature code is "*701. To retrieve the call, the
	user can go to any phone in the group and press the feature
	code "*701" and dial the parked extension.
Number of slots	Specifies the number of parking slots.
	For example, to configure six parking slots : (701, 702, 703,
	704, 705, and 706)
Timeout	It is the timeout interval for calls parked at a call-park slot.



6.1.4 Call Back

Features	Extension	Voicemail	Callpark	Callbac	k Fax	Conference	Voice Prompts
- Advanced 🔉					i.		
eature Settings	Callba	ick Prefix	* 4				
SDN PRI Settings	<u>Retry</u>	Options					
IP Global Settings	Attern	ots	10)			
DR Settings	Timeo	ut	10)	(seconds)		
System	Interva	al	3()	(seconds)		
• Tools			12.1				
• Statue							

Figure 33: Callback

Callback prefix	To register callback function, user has to dial destination
	number with this prefix code.
Retry Options	This option helps to callback with number of attempts and
	timeout for the callback and also the time interval between
	callback calls.

6.1.5 FAX

Features	Extension Voicemail (allpark Callback Fax	Conference Voice Prompts						
Advanced >									
eature Settings	Fax to Email Format	PDF 🞽							
SDN PRI Settings	Customized Fax Message			-					
IP Global Settings	Subject	New Fax Received.	New Fax Received.						
DR Settings		You have received the	You have received the fax from $CALLERID$ at $TIME) on (DATE) .$						
System	Body	(DALL).							
Tools		ç							

Figure 34: Fax



Fax to Email format	Specify the format to receive the FAX by email either PDF or
	TIFF format.
Customized Fax message	Allows you to customize the FAX message Notifications, which
	will be sent to the specified Email-ID's.

6.1.6 Conference

Features	Extension	Voicemail	Callpark	Callback	Fax	Conference	Voice Prompts				
Advanced 🔉	Custon	uiza d Courforo									
ature Settings	custon	nizea contere	nce wessage								
DN PRI Settings	Subjec	t	Conference Scheduled								
IP Global Settings			Conference \$ vour pin \${((CONFID) wh: PIN).	ich is o	n \${CDATE} at	<pre>\${CSTIME} using</pre>				
OR Settings	Body	2	54 6 140								
System											
Tools											
- 1/4 Persenan											

Figure 35: Conference

Customized	conference	Allows	you	to	customize	the	Conference	message
message		Notifica	tions, v	which	n will be sent	to the	specified Ema	il-ID's.

6.1.7 Voice Prompts

Dashboard	Feature S	ettings @					
Setup		0					8
Features	Extension	Voicemail	Callpark	Callback	Fax	Conference	Voice Prompts
Advanced 🔉							
ature Settings	Select	language for vo	pice-prompts		English	×	
DN PRI Settings							
P Global Settings							
R Settings	Save Res	set					
System							
Tools							
- Martine and Annual							

Figure 36: Voice Prompts

Select language for Voice-	Allows	you	to	select	the	language	for	the	Voice	prompts.



prompts.	MegaPBX- BRI supports four language voice prompts such as
	English, French, Turkish, and Spanish.

6.2 ISDN PRI Settings

(Only for Mega PBX products with PRI support)

Navigate through Advanced > ISDN PRI Settings

This section provides the ability to modify the PRI settings depending on the carrier (T1/E1), signaling, switch type, etc with respect to the service provider or any other mate.

	-			-		
atures	Span	 Framing/Coding 	Channels	Signaling	Switch	Clock Coptions
lvanced 🔉	PRI1	ccs,hdb3	30/31(E1)	PRI-CPE	EuroISDN	Master 📝
e Settings						
RI Settings	Advanced	U				
bal Settings						
ettings						
stem						
ols						

Figure 37: ISDN PRI Settings

Span Type	Γ	E1 🗸
Enable Progre	ss Indication	
	0.96794.3667.4603.46 X	

Figure 38: Advanced PRI Settings

Framing/Coding	Select Pro	эре	r Frai	min	g & codii	ng by	checki	ng wit	h your	service
	Provider.	If	CRC	is	enabled	from	Telco	side,	Please	select
	CCS/HDB3	S/CF	RC4. If	not	select CC	S/HDB	3.			

	Default for E1: CCS/HDB3
	Default for T1: ESF/B8ZF
Signaling	Select if gateway should work as customer premises equipment
	(CPE) or network device (NET). If Gateway is connecting to PRI
	service provider, select PRI-CPE. If connected with other Digital
	PBX's configured as CPE, then select PRI-NET.
	Default: PRI-CPE
Switch Type	Select the switch type as indicated by the ISDN service provider.
	Default: E1- Euro ISDN & T1- National ISDN2
Sync/Clock Source	Specify a transmit clock source. Master clocking uses the device's
	own system clock. Primary or secondary clocking uses a signal
	received from the T1 or E1 interface.
	Default: Master
Line Build Out	LBO depends on the line length for which attenuation is defined.
	Default: 0db . Check with your service provider for appropriate Line
	build out settings if you face any issue.
Channels	Indicate the number of channels to be used on this span.
Advanced Options	
Dialed Numbering	ISDN-level Type Of Number (TON) or numbering plan, used for the
Plan	dialed number, which is dependant on geographical location.
	Default: unknown
Dialing Numbering	Sets the calling number's numbering plan. Default: National
Plan	
Prefix	Prefixes specified under international, national, private, local, and
	unknown, will be used with dynamic dialing numbering plan, which
	dynamically sets the Type Of Number in the ISDN messages. If the
	called number matches the national prefix specified, it will
	automatically set the ISDN TON of calling number to national.
	Similarly for international & local prefixes. This will also strip off the
	digits in the prefix from the called number as well, but only if
	national or international prefix is matched.



Overlap Dialing

Send overlap digits

Advanced PRI settings					
Span Type		Select the carrier type, E-carrier (E1) or T-carrier (T1) which			
		depended on lines provided in your country. Default: E1			
Enable	Progress	Enabling this will provided call progress tones. Required, if service			
indication		provided fails to provide call progress tones.			

6.3 SIP Global Settings

Navigate through **Advanced** > **SIP Global Settings**

SIP Global settings apply to all VoIP traffic.

6.3.1 Port Settings

• Features	Port Settings	NAT Settings	Registratio	ın Tir	mer	Call Recording
Features	1. 	in a connigo		41310		our recording
 Advanced > 	SIP Bind	Port	5060			
Feature Settings	DTD Devi	Bauna	3000	4 9		
ISDN PRI Settings	(Reboot is r	equired)	16001	8	17000	
SIP Global Set <mark>i</mark> ings	UDPTL Port Range		17001]-[18000	
CDR Settings			S			
System				-		
Tools	Save Reset	D				

Figure 39: Port Settings

SIP Bind Port	Choose a port on which to listen for SIP UDP traffic. Default:
	5060
RTP Port Range	Range of port numbers to be used for RTP traffic.
	Default: 16001- 17000
	Make sure you configure this dynamic range of ports on
	your NAT Router. When the Mega PBX is behind a NAT and



	the NAT is configured to do port forwarding with above
	mentioned port range for UDP ports.
UDPTL Port Range	Port range for T38 Faxing is 17001 to 18000.

6.3.2 NAT Settings

Features	Port Settings	NAT Settings	Registration Timer	Call Recording	
Advanced >			Second Second		
eature Settings	NAT				- 1
(SDN PRI Settings	Туре		Stun Server IP 👻		- 1
SIP Global Set[ings	Stun/Exte	rnal Server IP	3		- 1
CDR Settings	Local Ne	t.			- 1
System					
 Tools 					

Figure 40: NAT Settings

NAT	NAT option is checked, when the Mega PBX is behind the
	Router/Firewall. Select either Stun Server IP or External IP.
	Default: disabled
Stun Server IP	If the Mega PBX is behind a non-symmetric NAT router, it
	may be necessary to use STUN to allow PBX to reliably
	communicate via IP through the router. Enter a STUN
	server IP address or domain name in the STUN Server field.
	For a list of public STUN servers, please Refer to:
	http://www.voip-info.org/wiki/view/STUN
External IP	Enter the NAT Traversal IP address i.e. Public IP Address of
	your internet, to communicate with Public Network when
	PBX is behind the NAT. This IP address will substitute in all
	outgoing SIP messages instead of Local IP address.
Local Netmask	Entering the Net mask of the local network of the Mega
	PBX allows it to identify the hosts falling within the same



network. E.g.: 192.168.2.0/255.255.255.0

6.3.3 Registration Timer

Features	Port Settings	NAT Settings	Registration Tin	ner	Call Recording
Advanced >		1994			
eature Settings	Default R (Reboot is n	egistration Expiry equired)	1	20	(sec)
DN PRI Settings	Minimum	6	0	(sec)	
' Global Setlings	Maximum Registration Expiry		3	600	(sec)
R Settings					100000m
System	Outbou	nd Registration	Timer		
Tools	Registrat	ion Timeout	2	0	(sec)
Status	Registration Attempts		0		

Figure 41: Registration Timer

Default Registration Expiry	Default duration (in seconds) of incoming/outgoing registrations. Default: 120 sec
Min Registration Expiry	Minimum duration (in seconds) of registrations allowed
	by the Mega PBX. Default: 60 sec
Maximum Registration Expiry	Maximum duration (in seconds) of incoming registrations
	allowed by the Mega PBX. Default: 3600 sec
Registration Attempts	Number of registration attempts before giving up with
	registrar (Outbound Registrations only).
	Default: 0 (never give up)
Registration Timeout	Registration attempt will be retried till this duration (in
	seconds), if no response from the Registrar. (Outbound
	Registrations only). Default: 20 sec



6.3.4 Call Recording

Dashboard	SIP Global S	Settings 🛛			
▶ Setup					
 Features 	Port Settings	NAT Settings	Registration Timer	Call Recording	
 Advanced 					
Feature Settings	Enable C	all Recording			
ISDN PRI Settings	Server Address		192.168.0.100		
SIP Global Set[ings					
CDR Settings					
System	Save Reset				
Ter (1996) 100 T					
 Tools 					

Figure 42: Call Recording

Server Address	It enables Call Recording with the given server where the
	ORK audio application is running.

6.4 CDR Settings

Navigate through Advanced > CDR Settings

CDR Settings feature allows managing the call records via radius server and FTP server.

6.4.1 Radius Configuration

Radius Server Configuration allows configuring the radius server information where all the call records will be saved in the form of vendor-specific attributes (VSAs).

- Advanced >	🗹 Enable Primary Radius		Enable Secondary Radius
Feature Settings	Server IP Address	192.168.0.100	Server IP Address
SDN PRI Settings	Server Port	1813	Server Port
SIP Global Settings	Password	••••	Password
 System 	Radius VSA Attrib	CISCO 💌	
 Tools 	Radius Retry Count	3 💌	
Status	Radius Retransmission Interval	2 💌	

Figure 43: Radius Configuration

To upload CDR entries via Radius, we need to install Radius Server on PC and the same has to be configuring it.

Server IP Address	User can enter the IP Address of Radius Server. Ex:
	192.168.xxx.xxx
Server Port	It specifies the Port Number on which Radius Server can be
	connected. By default Radius Server works on 1813 Port. It
	can be changed depending upon Radius Server's
	Configuration.
Password	User can enter the Radius Server's password to
	authenticate the Radius configuration.
Radius VSA Attrib	MegaPBX supports two attributes such as CISCO, DIGIUM.
	User can select anyone from the drop down list.
Radius Retry Count	It specifies No. of retries to upload CDR entries to Radius
	Server.
Radius Retransmission Interval	It specifies the time duration between retries.

Note: User can use secondary Radius Server.

6.4.2 FTP Configuration

This will allow you to configure the FTP server information for CDR billing and CDR upload scheduling by setting the Frequency of schedule, Day and Time.

Advanced > acture Settings SDN PRI Settings IP Global Settings DR Settings > System	☑ Enable FTP Upload Server IP Address 192.168.0.100 Server Port 21 Username megapbx Password ••••			Schedule Frequency Daily V Day V Time 4 : 30			
 Toois Status 	Upload How Save Reset						

Figure 44: FTP Configuration



Upload Records
This is a One-Time Upload.
If you want to save these settings, please click the "Save"
button once the upload is finished.
OK Cancel

Figure 45: Upload Records

Server IP address	User can enter the IP Address of Radius Server. Ex:
	192.168.xxx.xxx
Server Port	It specifies the Port Number on which Radius Server can
	be connected. By default Radius Server works on 1813
	Port. It can be changed depending upon Radius Server's
	Configuration.
Username	It specifies the username of endpoint (e.g.: IPPBX) will use
	to authenticate with the Mega PBX, e.g.: 5656
Password	User can enter the Radius Server's password to
	authenticate the FTP Configuration Settings.
Schedule	A user can schedule uploading CDR entries to FTP server.
	There are three frequency options: Daily, Weekly and
	Monthly.

6.4.3 Upload History

It shows the upload history of CDR with the details like time, server address and status of the uploaded CDR.





Features	Radius Configuration FTP	Configuration Opload History		
Advanced >	Show All 😪 entries		Search:	
ature Settings	Time	 Server Address 	Status	0
N PRI Settinas	2015-02-23 08:57:52	192.168.0.100	Upload Failed:No File To Upload	
P Global Settings	2015-02-23 08:58:22	192.168.0.100	Upload Failed:No File To Upload	
R Settings				
Tools				
Status	Showing 1 to 2 of 2 entries		First Previous 1 Next	Last

Figure 46: Upload History



7. System

7.1 Network

Navigate through **System > Network Settings**

Network Settings allows modifying the device network settings.

Advanced	WAN Configuration	n	LAN Configurati	on
System 🔉	OHCP Client	O Static IP	IP Address	192 . 168 . 113 . 2
e/Time	IP Address	192 . 168 . 0 . 145	Media Configura	ation
Tools	Netmask	255 . 255 . 255 . 0	3	
Status	Gateway	192 . 168 . 0 . 254	IP Address	192 . 168 . 0 . 3
	DNS 1	192.168.0.5	Host Configurati	on
	DNS 2	192.168.0.2	Hostname	PRIGW

Figure 47: Network Settings

DHCP	When enabled and a DHCP server is available, the Mega PBX will
	auto configure itself. If DHCP server is not available, select "Static",
	and fill in the Network Configuration.
IP Address	The static IP address corresponding to your WAN configuration.
Net mask	The Net mask corresponding to your WAN configuration.
DNS	The IP address corresponding to a DNS server.

WAN Configuration:

LAN Configuration:

LAN Port is a management port. Mega PBX can be connected back-to-back to a PC or to a LAN network for configuration. It is always recommended to connect back-to-back to a PC. In case, connected to LAN network & if IP series clash is found, IP series can be changed here.





Media Configuration:

It is applicable only for 4. 0.0 Version onwards. If the media IP address is not configured, there won't be voice path in the calls.

Media IP Address needs to be configured manually and it should be in the same network segment of WAN IP address. This IP address is used for Voice Media Configuration.

For E.g.: If the WAN IP address is 192.168.0.145 and media IP address should be in the same Network Segment (available IP address) side. 192.168.0.3



Figure 48: Warning

System will reboot to apply the network changes.

Host Configuration:

Host name: Label or IP of your system identity.

7.2 Date/Time

Navigate through System > Date/Time



Features	Configuration Type 🛛 Manual 🛩			
Advanced	Manual Configuration	NTP Configuration	n	
- System 🔷 🗲		Timezone	Asia/Kolkata	~
Network	Date 23 / 02 / 2015 (dd/mm/yyyy)	NTP Server		Add
)ate/Time	Time 05 : 38 (hh:mm)		3.in.pool.ntp.org	Delete
 Tools 				
 Status 		-		×.

Configuration Type	Date and Time of the Mega PBX can be either set manually (uses
	RTC) or automatically (through NTP). Default: NTP
NTP Configuration	Time Zone: Select the correct time zone for the location where
	the Mega PBX is installed using the Time Zone dropdown box.
	Default: Asia/Kolkata
	NTP Server: URI or IP address of the NTP (Network Time
	Protocol) server, which will be used to synchronize the date and
	time. E.g.: 3.in.pool.ntp.org

Click on "APPLY "button, followed by "SAVE ALL" button to update the configuration changes.



Changing the configuration requires the current calls to be disconnected.



8. Tools

8.1 Diagnostics

Navigate through Tools > Diagnostics

Analyze the functionality of the Mega PBX with some of these diagnostic tools provided.

Features	Ping Test	Traceroute Test	
Advanced System	Host 192.168.0.33	Host 192.168.0.33	
Tools >	Count 1 💌	Hops 3 💌	
gnostics			
:kup/Restore	Ping	Traceroute	
grade Firmware			
Status			

Figure 49: Diagnostics

Ping Test:

It is used to check the packet loss and latency time from your SIP end client like IP Phone/ FXS gateways to check the quality of your network connections.

Diagnostics Result	×
PING 192.168.0.33 (192.168.0.33): 56 data bytes	
64 bytes from 192.168.0.33: icmp_seq=0 ttl=64 time=0.3 ms	
192.168.0.33 ping statistics	
l packets transmitted, l packets received, 0% packet loss round-trip min/avg/max = 0.3/0.3/0.3 ms	
	_

Figure 50: Diagnostics Result

Trace route Test: It is used to determine the route taken by packets across an IP network.



 Diagnostics Result
 X

 traceroute to 192.168.0.33 (192.168.0.33), 3 hops max, 38 byte packets
 1

 1
 www.mail.cemtest.net (192.168.0.33)
 0.268 ms
 0.230 ms
 0.194 ms

Figure 51: Diagnostics Result

Format SD Card / Format USB Drive: This will allow you to completely erase all the contents in the SD card/ USB Drive (MegaPBX-BRI supports USB Drive).

If you are formatting the USB Drive/SD card details, kindly take the back of the MegaPBX configuration.

8.2 Backup/Restore

Navigate through Tools > Backup/Restore

Back Up:

Allow you to take the back up of the System configurations & save it to the local PC.

Restore:

Restoring from a new upload or backup file will destroy all current configurations and require a system reboot. All calls will be dropped and all current configurations will be destroyed.

eatures	Backup	Restore
Advanced System Tools > nostics	(Click the button to download the configuration files) Backup	(Choose the filepath of the restore file) Filename: Browse No file selected. (Filename) Restore
up/Restore		Restore

Figure 52: Backup/Restore



Administrator password will not be restored on restoration. So you should still use same credentials as before restoration.

8.3 Upgrade Firmware

Navigate through Tools > Upgrade Firmware

Update Mega PBX with the latest release available, which can contain key updates, added functionalities and bug fixes. When a new release is available, download it and save to your local PC. Then, browse for the file, and click the Upload button. Now your Mega PBX will display a Progress Screen and will prompt when it is about to reboot. Reboot and wait for blue LED's turn ON.

	Opgrade Firmware
• Setup	
 Features 	Current Firmware Version: 4.0.0
Advanced	
 System 	(Choose the filepath of the new firmware)
• Tools >	Filename: Browse No file selected. (Need Reboot)
Diagnostics	
Backup/Restore	
Upgrade Firmware	Upgrade
Status	

Figure 53: Upgrade Firmware

During firmware upgrade, there should not be any power or network disturbances, which may leads to Mega PBX board faulty. Firmware up-gradation process will take few minutes.



9. Status

This section generates the various status of the MegaPBX-PRI are explained below.

9.1 Call Reports

Navigate through Status > Call Reports

Call Reports displays a detailed list of calls pass through the MegaPBX-PRI. The list can be generated on the bases of date range, CDR count, latest 50 entries or all entries. Generated report can also be exported to local PC as CSV file or directed to a printer.

To create a new Report, select the Extension Range or Date range, and click the **"Generate"** Report button. A list with call details will display in the Call Reports section.

You can either export to your local PC or Print the Call reports.

Features	Generat	е Туре	Default (Late:	st 50 entries) 💌			C	ienerate	🗷 Export 🖶 Pr	int
Advanced										
System	Show	10 💌 entries						Searc	h:	
Tools	No. *	Start o Time	Caller ᅌ	Callee	○ Duration ○	Status 🗢	Owner 🗘	Hangup Cause	CLINK ID	\$
Status >	1	2015-02-19 17:26:58	2323	5656	1297	Answered	2323	Normal Cleaning	1424366818.10	^
O Stations	2	2015-02-19 15:12:26	2323	5656	5	Answered	2323	Normal Cleaning	1424358746.8	
P Trunks I Span	з	2015-02-19 14:56:22	2323	5656	5	Answered	2323	Normal Cleaning	1424357782.6	
rrent Calls	4	2015-02-19 12:36:13	2323	5656	4	Answered	2323	Normal Cleaning	1424349373.4	
urr. Conferences	5	2015-02-19	2323	5656	15	Answered	2323	Normal	1424347809.2	~

Figure 54: Call Reports

9.2 SIP Station

Navigate through Status > Call Reports

SIP Station Status page displays detailed status of each SIP Extensions available on the MegaPBX-PRI.

E.g.:

No.	Extension	Host	Status
1	1001	Unspecified	Registration Failed
2	1002	192.168.0.127	Registered

Dashboard	SIP Stat	ion Status 🛛		
Setup				
Features	Show A	II 💌 entries		Search:
Advanced	No.	* Extension	\$ Host	\$ Status
	1	2323	192.168.0.172	🔽 Registered
System	2	4455	(Unspecified)	😢 Registration failed
Tools	З	4545	(Unspecified)	🔀 Registration failed
Status >	4	5656	(Unspecified)	🔀 Registration failed
Reports	5	7575	(Unspecified)	🔀 Registration failed
Stations				
Trunks				
Span				
rent Calls	Chowing	1 to E of E optrios		First Browleys 1 Mart Las
r. Conferences	Showing	y 1 to 3 or 3 entries		First Previous 1 Next Las
work				

Figure 55: SIP Station Status

9.3 SIP Trunks

Navigation: Status > SIP Trunks

SIP Trunk Status page displays detailed status of each SIP trunks available on the MegaPBX-PRI.

• Dashboard	SIP Tru	ınks Statı	IS (2							
• Setup											
Features	Show	All ⊻ entries						Se	arch		
Advanced	No.	Ac. Name	٥	Username	\$ Host	٥	Port	\$ Status	\$	Reg. State	٥
	1	5566		5566	192.168.0.100		5060	ОК		💟 Registered	
 System 	2	8989		8989	192.168.0.165		5060	UNREACHABLE			
Tools											
Status 🔹 🗲											
all Reports											
(P Stations											
IP Trunks											
RI Span											
urrent Calls	Chowin	og 1 to 0 of 0 c	untrin	-				First	Drow	aug 1 Blaut La	et
urr. Conferences	SHOWI	ng 1 to 2 01 2 6	andre	5 2				Pirse	F4 B VI	DOD 1 WEXT 19	ac
letwork											

Figure 56: SIP Trunks Status

Status	Reg. State	Description	
ОК	Registered	Configured, Registered &	-
		reachable	
ОК	-	Configured & Reachable, but	-

@ allo....



		no Registration	
ОК	Request	Configured, but Host not	Check Registrar
	Sent	responding or unreachable	Address
ОК	Rejected	Configured & reachable, but	Check
		Registration failure	Authentication
UNREACHABLE	Registered	Configured, Registered, but	Check Proxy Address
		not reachable	
UNREACHABLE	-	Configured, but not reachable	Check Proxy Address
UNKNOWN	_	Not Registered	Client not registered

Dynamic: Host IP is obtained dynamically on registration.

9.4 PRI Span

(Only for Mega PBX products with PRI support)

Navigate through Status > PRI Span

PRI Span Status page displays detail status of each E1/T1 Port with individual channel info, available on the PRI Gateway.

eatures	Port Status: E1
dvanced	Port Port 1
ystem	Status [+]
ools	
tatus 🔹 🗲	
eports	Channel Status: E1
ations	Channel 1 2 3 4 5 6 7 8 9 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31
unks	Pot 1
an	⊖ Busy ⊖ Idle ⊖ Disabled
nt Calls	

Figure 57: PRI Span Status



E1/T1 Port Status	Description
	Appliance is not seeing far end, circuit is not up, or cable is bad.
	Appliance is synchronizing or is receiving a yellow alarm from the far end.
	PRI Link is Active. Appliance is in-sync with the far end.
	T1/E1 driver is not initialized or device undetected.

Channel Status	Description
-	Channel is Busy
	Channel is Idle and ready to receive or make calls
	Channel is not active

9.5 Current calls

Navigate through Status > Current Calls

Current Calls page displays detailed status of the real time calls available on Mega PBX.

9.6 Current Conferences

Navigate through Status > Current Conferences

Current Conferences page displays detailed status of the real time conference available on Mega PBX.

9.7 Network

Navigate through Status > Network Status

Network Status page displays detailed status of the network configuration on Mega PBX.



10. Administrator

ALLO COM'S Mega PBX-PRI			12 APPLY
23-February-15 10:00:14 am	Firmware Version: 4.0.0	Refresh	Welcome Administrator
			Reboot
			Call Manager Reload
			Web Settings
			Email Settings
			Locout

Figure 58: Administrator

10.1 Reboot

Navigate through 💌 > Reboot

Using this option administrator can reboot (SOFT reboot) MegaPBX System remotely

10.2 Call Manager Reload

Navigate through 👛 > Call Manager Reload

Reloading Call Manager will restart call manager and drop all current calls.

10.3 Web Settings

Navigate through 😻 > Web Settings

Session Timeout	Duration after which current web login session expires. Default:
	3600 sec
Pagination	Number of entries in a table per page to be displayed.
Change Password	Modify Administrator password here.

10.4 Email Settings

Navigate through 🙆 > Email Settings

To configure the Mega PBX to send out voicemail/FAX via email, the related SMTP setting must be configured.



Mail Server	Enter the domain name of the Email Server address of the particular authorized email client account
Email ID	Enter the email ID of the particular authorized email client account.
Username	Email ID given by the Mail Server administrator.
Password	Password of the Email ID
TLS Support	To secure the server to server transfer of emails, the provider needs to enable a technology called Transport Layer Security (TLS).

10.5 Logout

Navigate through 🙆 > Logout

Administrator Logout option after use.

Thank you for choosing

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