

BR4 - APPLIANCE

USER MANUAL



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1. General overview



What is BR4 - Appliance?

BR4 - Appliance is a Blackfin DSP based **Asterisk** driven, S/T interface **embedded appliance**. It is an **open source** embedded platform based on **uClinux** and **Astfin**.

Basic Rate Interface is [Integrated Services Digital Network](#) (ISDN) configuration that consists of two B channels (64 kbits/s “bearer” channels) used for voice and data. And one D channel (16 kbits/s “data” channel).

The physical interface is called S/T. It uses four wires with a separate pair of wires for the uplink and another pair for the downlink.

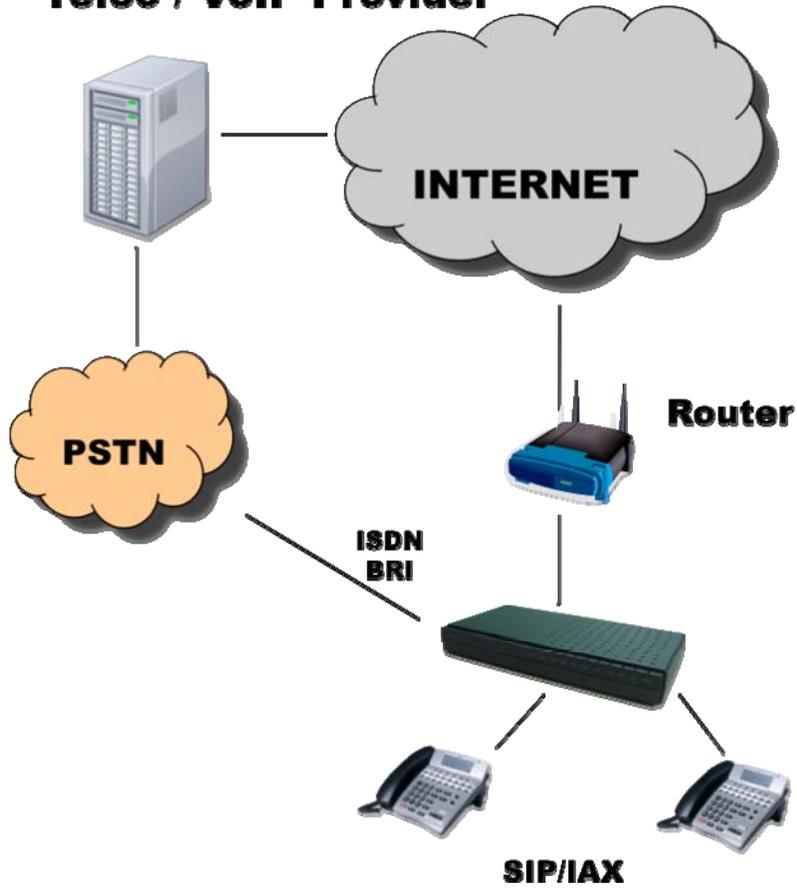
The typical use of the BRI interface is in residential solutions.

The mISDN ports can be configured for PTP or PTMP mode. PTP (Point to Point) is a type of connection restricted to two end points.

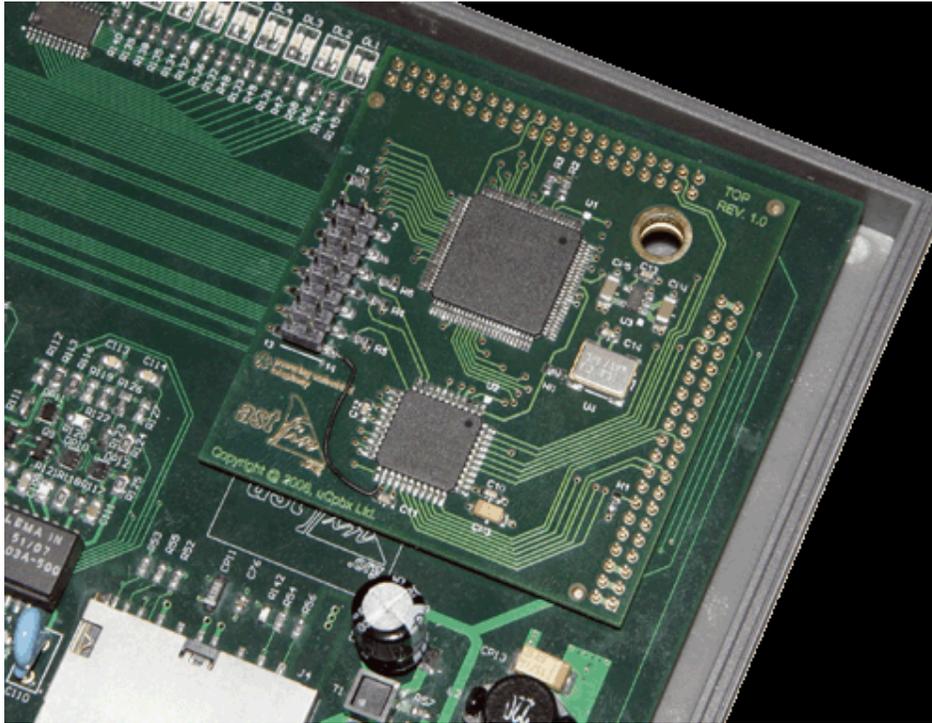
PTMP (Point to Multi Point) is a type of connection, providing multiple paths from a single point to multiple points (locations).



Telco / VoIP Provider



1.1. Echo cancellation module



Echo cancellation module installed in BR4 - Appliance

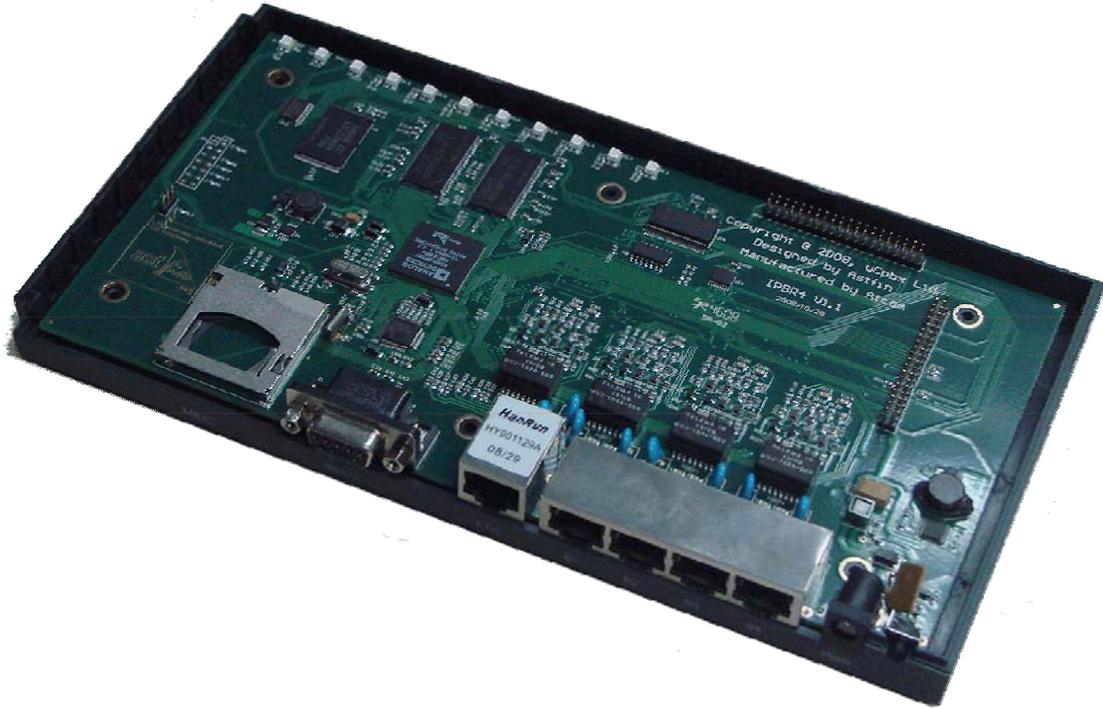
To ensure carrier grade voice quality, BR4 - Appliance can accept an optional line echo cancellation module based on G.168 standard.

Echo cancellation is a process of removing the echo from a voice communication in order to improve voice quality on a telephone call. In general there are two sources of echo in the telephony: Acoustic echo and echo from the analog phone hybrids. Speech compression techniques and digital processing delay often make these echoes more severe in telephone networks.

To achieve consistent quality we have selected **hardware** based solution from Zarlink. There is a special connector inside the BR4 - Appliance, where user can plug an optional LEC module, this can be done later, after all the features are configured.



2. Technical information



**picture without LEC module*

2.1. System: uClinux, Astfin

Our PBXs are driven by our own telephony oriented, Open Source uClinux distribution called **Astfin**(<http://blog.astfin.org>). Current version of Astfin provides **Asterisk 1.4.x**, **Zaptel 1.4.x** and **Libpri 1.4.x** together with custom kernel modules to support our hardware. Additional applications such as PPPoE, SMTP forwarder, NTPd and many more are also provided to extend usability of the BR4-Appliance under different scenarios.



2.2. Hardware:

- ADSP - BF537 600MHz CPU. DSP core for the media processing.
- 64MB of SDRAM
- 256KB serial flash for the boot-loader
- 256MB NAND flash for voicemail and prompts.
- SD card interface on a dedicated bus.
- Watchdog timer
- Optional, hardware based, G-168 Line Echo Cancellation.

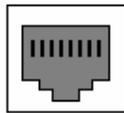
2.3. Interfaces

- QUAD ISDN (2B + 1D) interface
- TE with PTP and PTMP
- EuroISDN (mISDN)
- 10/100Mbps Ethernet port with high performance PHY
- RS232 for console connectivity (115k, 8-N-1)

2.3.1. Interface cables

- For the Ethernet connection you have to use:
- In case you connect to router, switch and etc. Ethernet patch cable
- In case you connect to other PBX device crossover cable
- ISDN ports – the BR4 PBX has 4 ISDN ports:

1 2 3 4 5 6 7 8

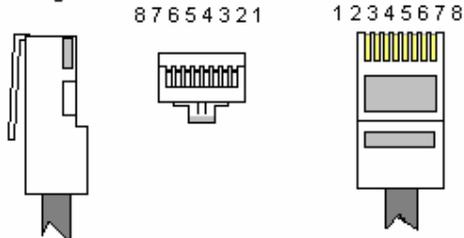


RJ-45
Female

Pin	Description
3	TX+
6	TX-
4	RX+
5	RX-

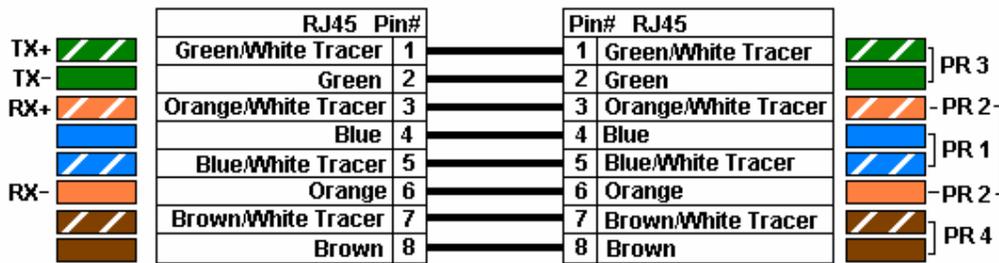
- To connect the BR4-Appliance (TE mode) ISDN port to your telecommunication equipment you should use standard Ethernet “patch” cable. The configuration is:

RJ-45 Male
Plug

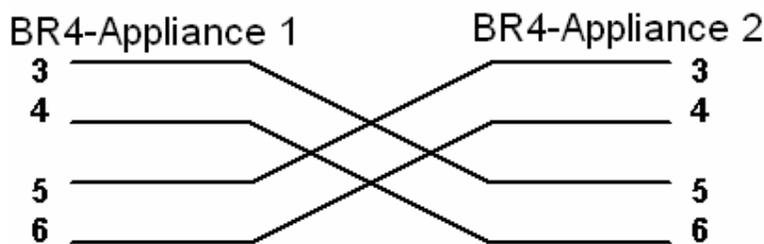


Color Standard
EIA/TIA T568A

Ethernet Patch Cable



Keep in mind that if you want to use BR4-Appliance in NT mode you will need an ISDN crossover cable as shown below:



You will need to use 100Ω external resistive termination (between both TX+,TX- and RX+,RX-) at one of the end of the ISDN connection. BR4-Appliance doesn't provide phantom power on its ISDN ports. If you are connecting ISDN telephone which requires power from the ISDN line you have to implement this externally.

2.4. Applications

- VoIP / TDM Gateways
- PBX / IVR functionality
- VoIP Services
- Conferencing
- Custom platforms
- Voice Routing
- Custom Development

2.5. Additional information

- Power supply 6 - 12VDC
- Current consumption – idle state 150 mA
- Dimensions: 224 x 122 x 30 mm (8" 13/16 x 4" 13/16 x 1" 3/16)
- 12V, 2A power adapter is included



3. Software and Configuration tips.

Working with the GUI



3.1. System Status Menu

After all the interfaces are connected and your BR4-Appliance is powered up you can connect to the GUI through your preferred Web browser. By default all BR4 - Appliances are preconfigured with 192.168.1.100/24 IP address/netmask. Please change the IP address on the computer you will be using to configure the BR4 - Appliance to be a part of 192.168.1.x network (for example: 192.168.1.2). At this point you can connect to the GUI by selecting the following URL:
<http://192.168.1.100>

When the initial page finishes loading you will be prompted to authenticate. Our default user name is **admin** and password is **astfin**

Asterisk™ Configuration Engine	Asterisk™ Configuration Engine
Username: <input type="text"/>	Username: <input type="text" value="admin"/>
Password: <input type="password"/>	Password: <input type="password" value="*****"/>
<input type="button" value="Login"/>	<input type="button" value="Login"/>

After successful login, you will see the system status page.



System Status
Uptime : 08:08:29 up 41 min, load average: 0.00, 0.00, 0.00

Trunks

Status	Trunk	Type	Username	Port/Hostname/IP
	bri2	BRI	Ports: 2	

Agents


 6204
[Login](#)


 6210
[Login](#)


 6250
[Login](#)

Conference Rooms ↕

6300
 Not in use

Extensions

Free
 Busy
 UnAvailable
 Ringing

Extension	Name/Label	Status	Type
<input checked="" type="checkbox"/> 6200	John Brown	Messages : 2/0	SIP User
<input checked="" type="checkbox"/> 6204	Mike Reverouzzi	Messages : 2/0	SIP User
<input checked="" type="checkbox"/> 6210	Sales Department Secretary	Messages : 0/0	SIP User
<input checked="" type="checkbox"/> 6211	Sales Department Head	Messages : 0/0	IAX User
<input checked="" type="checkbox"/> 6250	PR Department Head	Messages : 0/3	SIP User
<input checked="" type="checkbox"/> 6260	Support Department	Messages : 3/0	IAX User
6400	Sofia_All		Ring Group
7000	English_menu		Voice Menu
6221	Check Voicemails		VoiceMailMain
2676	Dial by Names		Directory

The System Status page presents all vital information about the current state of your BR4 - Appliance.

Here is an example with system status page of a local PBX.

As you can see there is one trunk bri2 in use, the type of it is BRI and its using port 2.

The users with extensions 6204,6210 and 6250 are agents for a call center.

There is Conference room with public extension 6300 and no participants are connected.

Different color lights indicate the state of the users.

The extension 6200 associated with user name “John Brown”, using an SIP telephone and currently have two voicemail messages. The current state of this user is “Available”, which indicates that the endpoint is registered and not in use.

The extension 6204 associated with user name “Mike Reverouzzi”, using an SIP telephone and currently have two voicemail messages. The current state of this user is “Busy”, which indicates that the endpoint is powered and it is on a call.

The extension 6210 associated with user name “Sales Department Secretary”, using an SIP telephone and currently have no voicemail messages. The current state of this user is “UnAvailable”, which indicates that the endpoint is powered down or there is some network connectivity problem.

The extension 6211 associated with user name “Sales Department Head”, using a IAX telephone and currently have no voicemail messages. The current state of



this user is “Available”, which indicates that the endpoint is registered and not in use.

The extension 6250 associated with user name “PR Department Head”, using an SIP telephone and currently have three, already checked voicemail messages. The current state of this user is “Available”. which indicates that the endpoint is powered and not in use.

The extension 6260 associated with user name “Support Department”, using an IAX and currently have three voicemail messages. The current state of this user is “Ringing”, which indicates that the endpoint is powered and it is ringing (receiving a call).

The extension 6400 is designated to Ring Group.

The extension 7000 is designated to the English language voice menu.

The extension 6221 is designated to retrieve voicemail messages.

The extension 2676 is designated to corporate directory service which can be used in one or more IVR menus.

3.2. mISDN Config Menu



BR4-Appliance ISDN interfaces detected !

Card/Port	Mode	
1/1	TE-Mode, PTMP	Edit
1/2	TE-Mode, PTMP	Edit
1/3	TE-Mode, PTMP	Edit
1/4	TE-Mode, PTMP	Edit

Update Cancel

List of mISDN Service providers (trunks)

Trunk Name	Ports	
bri2	2	Delete

Add

Keep in mind that port (s) of your PBX that you want to use, should match configuration of your ISDN device you are connecting to. For example if you have a mISDN phone in PTMP mode, the port of your PBX must be set to PTMP mode as well.

By clicking on the “Edit” button a pop-up menu will appear.

Edit Port X

Card/Port: 1/1

Port Settings: TE-Mode, PTP

Port Options (Optional): Master Clock

Update Cancel



The option here that you can use are :

- Card/Port –Gives the information about how many ports does the appliance have and which one, you are editing.
- Port Settings – You have to select the port for PTP (Point To Point),or PTMP (Point To Multi Point) mode.
- Port Options – you can select the port to be master clock for synchronizing with other devices.

Creating BRI trunks:



You can see the configured trunks of your PBX, you can delete or add new trunks. After creating new trunk you have to click the Reload button in the GUI.

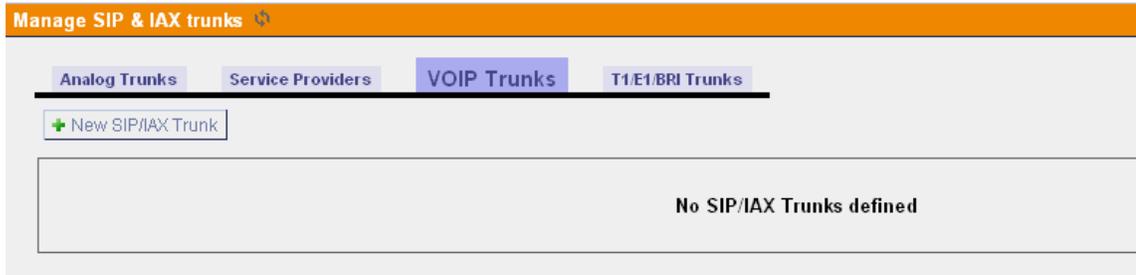
* **Warning** – In order for complete adding of new Trunk , after creating the trunk you must reboot your PBX from Reboot button in to the Option Menu. This should be done before doing any other changes in to the GUI.

3.3. Trunks

Click on the button in to main menu



After loading this page on the system which has not been yet configured the following menu will appear:



If you want to setup VoIP trunk press the tab “New SIP/IAX Trunk”

**Reminder – You can always use the “i” (info) tooltips for additional information*

- First you have to select the type of your provider as shown SIP or IAX
- Provider name – specify your provider name for reference
- Hostname – here you type the default address of your provider for example support.provider.com
- Username – this is you username also given by the provider
- Fromuser – fill this field as per providers instructions
- Fromdomain – fill this field as per providers instructions
- Password – this is you password given by the provider
- Insecure Type – Specifies how to handle connections with peers



Analog Trunks			Service Providers			VOIP Trunks			T1/E1/BRI Trunks		
Trunk			Signalling			Channels					
bri2			BRI			2					

The above example indicated that we have on BRI trunk named “bri2” which contains port number 2 (BRI2).

* **Warning** – In order for complete adding of new Trunk , after creating the trunk you must reboot your PBX from Reboot button in to the Option Menu. This should be done before doing any other changes in to the GUI.

3.4. Outgoing Calling Rules

Click on the button in to main menu



After loading this page on the system which has not been yet configured, you will need to create a “New Calling Rule” The following menu will appear:



New CallingRule [X]

Calling Rule Name ⓘ :

Pattern ⓘ :

Send to Local Destination ⓘ

Destination :

Send this call through trunk:

Use Trunk ⓘ

Strip ⓘ digits from front

and Prepend these digits ⓘ before dialing

Use FailOver Trunk ⓘ

fail over Trunk ⓘ

Strip ⓘ digits from front

and Prepend these digits ⓘ before dialing

**Reminder – You can always use the “i” (info) tooltips for additional information*

- At first you need to assign unique reference name to the new Outgoing rule
- In the second field, “Pattern” standard telephony regular expression patterns needs to be specified:
 - X ... Any Digit from 0-9
 - Z ... Any Digit from 1-9
 - N ... Any Digit from 2-9
 - [12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9
 - Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
 - ! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible

For example, the extension _NXXXXXX would match normal 7 digit numbers, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.
- “Destination” field should be used only if you want to invoke some local application or if the call should be processed in some special way.

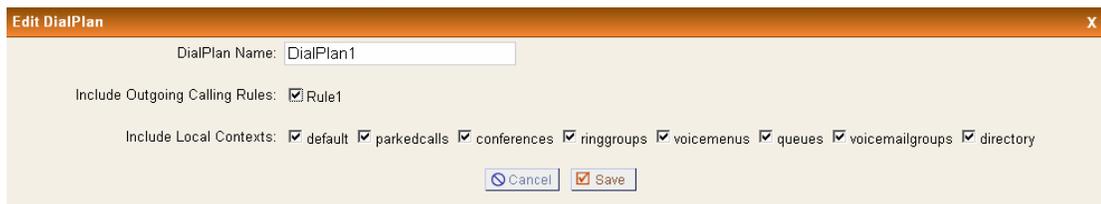


- “Use trunk” indicates which trunk should be used to handle this call (ie: “bri2”).
- “Strip” indicates how many proceeding digits should be removed from a dialstring.
- “Failover trunk” indicates if there is an alternative trunk to be used in case if the primary trunk is not available (in use for example).

3.5. Dial plans



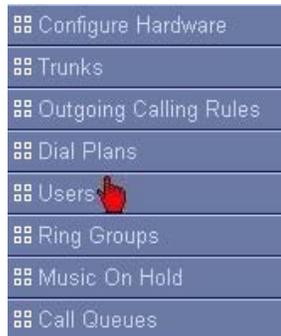
The “DialPlan” page allows you to define different Classes of Service (Dial Plans) and associate them with one or more Outgoing Calling Rules defined in the previous step. Also, predefined local services/application can be associated with each Class of Service.



- Please select a unique identifier for your Dial Plan entry.
- Select all services and rules to be available to users using this Dial Plan entry (Class of Service).

3.6. Users





The “Users” page allows creation of user accounts (Extensions) for VoIP accounts. Voicemail options are also configurable through this page.

Create New User
X

General :

Extension: ⓘ Name: ⓘ DialPlan: ⓘ

CallerID: ⓘ OutBound CallerID: ⓘ

Enable Voicemail for this User ⓘ

VoiceMail Access PIN code: ⓘ Mailbox: ⓘ Email Address: ⓘ

Technology

SIP ⓘ IAX ⓘ Analog Station: ⓘ flash: ⓘ rxflash: ⓘ

Codec Preference : First : ⓘ Second : ⓘ Third : ⓘ Fourth : ⓘ Fifth : ⓘ

VoIP Settings

MAC Address : ⓘ Line Number : ⓘ SIP/IAX Password: ⓘ

NAT: ⓘ Can Reinvite: ⓘ DTMF Mode: ⓘ insecure: ⓘ

Other Options

3-Way Calling ⓘ In Directory ⓘ Call Waiting ⓘ CTI ⓘ Is Agent ⓘ Pickup Group: ⓘ

**Reminder – You can always use the “i” (info) tooltips for additional information*

General

- Extensions – this is the actual number to be dialed to reach the user, it’s an index and it needs to be unique.
- CallerID - specifies the internal caller id number associated with this account. This number will be used to automatically identify the user to the voicemail system. The CallerID does not have to be unique.
- Name – Indicates the Caller ID Name which will be sent to other callers , if the network permits such functionality.
- Outbound CallerID – indicates the public CallerID number to be used for the outbound calls. Depending on the provisioning of your ISDN



line and/or feature set supported by your VoIP trunk providers, this number may or may not be presented to the destination endpoint.

Enable VoiceMail for this User

- VoiceMail Access Pin Code – indicates the passcode required to access this voicemail box.
- Mailbox – this field indicates to which mailbox number Voicemail Indicator will subscribe. (visual indicators and stutter tone)
- Email Address – indicates an email address to be used to send voicemail notifications and actual voicemail attachments (depends on voicemail configuration, please refer to the applicable section below).

Technology

- SIP/IAX/Analog Station - indicates the protocol (technology) to be used for this account. Please note that analog ports are not available on the BR4 - Appliance.
- Flash/RxFlash – Hook flash specific parameters.
- Codec preference – list of available codecs listed in order of preference. Please note that both ends need to agree for connection to be established.

VoIP Settings

- MAC Address – this field is required for Polycom phone provisioning, do not use it at this point.
- Line number – as above
- SIP/IAX password – user password associated with this account
- NAT – indicated if the device will be located behind NAT router in respect to our BR4 -Appliance.
- Can Re-invite – indicates if SIP session can use re-invite to send RTP directly between to endpoints.
- DTMF Mode – DTMF method to be used for VoIP communication. TFC2833 would be the most common choice here.
- Insecure – method of authentication, both ends need to agree.

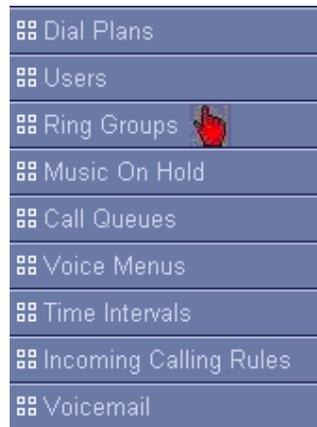
Other Options

- 3 Way Calling – enable/disable 3 way calling functionality, please make sure that your device is capable of handling it before selecting this option.
- In directory - indicates if this user account will be listed in a “directory listing” accessible from one or more IVRs.



- Call Waiting – enable/disable call waiting functionality on the account.
- CTI – Computer Telephony Integration, allows access to 3rd party applications over Asterisk Manager Interface.
- Is Agent – indicates if the user will be available in call queuing application.
- Pickup group – specified the call pickup group, if available

3.7. Ring Groups



This page allows to group several users (extensions) into one Ring Group. Unique extensions number will be associated with each Ring Group to allow easy access from your dial plan.

 A screenshot of a web-based configuration window titled "New RingGroup". The window has an orange header bar with a close button (X) on the right. The main content area is light beige and contains the following fields and controls:

- "RingGroup Name" field with the value "Sales Departament".
- "Extension for this ring group" field with the value "6500".
- Two list boxes: "Ring Group Members" (empty) and "Available Users" (containing "6000(SIP) Peter").
- Navigation buttons between the list boxes: "<<", "←", "→", and ">>".
- "Ring Group Options" section with:
 - "Strategy" dropdown menu set to "Ring in Order".
 - "Seconds to ring each member" field set to "20".
 - "If not answered Goto" dropdown menu set to "Hangup".
- "Cancel" and "Save" buttons at the bottom right.

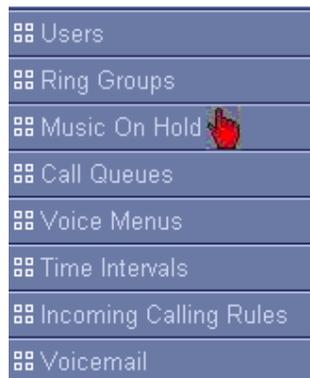


- At first you have to choose the name of the group (you can create several independent groups)
- Type the preferred extension for that group, in the example this is “6500”
- In the left window you can see the users that are members of the specific group, you can add/remove them with button in the middle. The available users are in the right window.

You have several options about the group calls.

- Strategy, when you dial the extension for the group the phones can ring in order (left window) or in sequence for a predefined period of time. The other option is to ring all the devices in the group together at the same time.
- Handling for non answered calls: Call can be redirected to specific Voicemail, another extension, or another ring group.

3.8. Music on Hold



The default Music On Hold (MOH) class is shown below. Please note that you can not upload new MOH files at this point. Direct access to shell is required to scp or ftp new MOH files.

Manage 'Music-on-Hold' Classes - default + New MOH class X Delete

Create New MOH Class

Name : Add Cancel

(Ex: newfile.conf)

Upload an 8 KHz Mono Music file :

Choose file to Upload: Browse...

Upload

List of Sound Files

Sound File	Options
LICENSE-asterisk-moh-freeplay-ulaw	X Delete
fpm-world-mix.ulaw	X Delete
fpm-sunshine.ulaw	X Delete
fpm-calm-river.ulaw	X Delete
LICENSE-asterisk-moh-freeplay-alaw	X Delete
fpm-world-mix.alaw	X Delete
fpm-sunshine.alaw	X Delete
fpm-calm-river.alaw	X Delete

3.9. Call Queues

Call queues allows you to build a call center with one or many trunks.





New Queue [X]

Extension : ⓘ Name : ⓘ

Strategy : ⓘ Music On Hold : ⓘ

LeaveWhenEmpty : ⓘ JoinEmpty : ⓘ

Queue Options:

TimeOut: ⓘ Wrapup Time: ⓘ Max Len: ⓘ

ⓘ Auto Fill ⓘ Auto Pause ⓘ Report Hold Time

KeyPress Events : ⓘ

Agents: ⓘ

- Mike Reverouzzi (6204)
- Sales Department Secretary (6210)
- PR Department Head (6250)

**Reminder – You can always use the “i” (info) tooltips for additional information*

- At first you need to assign unique extension to your new queue
- Then you need to name it, ie: “sales”
- Select the appropriate strategy for your application:
 - **Ringall** – ring all available agents at the same time.
 - **RoundRobin** – Each all agents in sequence.
 - **Leastrecent** – ring agent which was recently called.
 - **Random** – ring random agent.



- **RRmemory** – RoundRobin with memory, where we left off last ring sequence.
- The appropriate Music on Hold class can be selected from the “Music on hold” menu.
- Leave when empty – this option controls state of users that are in the queue. If **yes** is selected, the callers are pushed out of the queue when no agents are logged in. If **no** is selected, callers will remain in the queue with no agents. If **Strict** is selected, callers are forced out of the queue if no agents are logged in, or if all logged in agents are unavailable.
- Join Empty - If **yes** is selected, callers can join a call queue with no agents or unavailable agents. If **no** is selected, the callers cannot join queue with no agents. If **strict** is selected, the callers cannot join queue with no agents or unavailable agents.
- Queue options –
 - **Timeout** – how many seconds will ring an agent’s phone before the queue tries to ring the next agent.
 - **Wrap-up** time is how many seconds delay has an agent after completing a call, before another call is connect.
 - **Max Len** is how many calls can be queued at once. This includes only calls that have not been yet connected.
 - **Auto Fill** – when multiple calls are in the queue at the same time, to push them to agents simultaneously.
 - **Auto Pause** – this option pauses an agent if they fail to answer a call.
 - **Report Hold Time** – this option reports to the agent the hold time of the caller, before he is connected to an agent.
 - **Key Press Events** – this setting selects which voice menu to be connected if a user waiting in the queue presses a button.
- List of available agents is available at the bottom.
- Agent Login Settings – you have to type the extensions for the agent login and agent callback login



Queues **Agent Login Settings**

Agent Login Settings

Agent Login Extension: ⓘ

Agent Callback Login Extension: ⓘ

Agent Logout:

To logout of **Agent Login** Hangup your phone. To Logout of **Agent Callback Login** Dial the same extension used to login, specify your extension and password when prompted, and hit # when asked for your callback extension. This will successfully log you out of all queues you are a part of.

**Reminder – You can always use the “i” (info) tooltips for additional information*

3.10. Voice Menus



This page allows creation of custom voice menus for an IVR system. Please use the **Voice Menu Prompts** page to record your custom prompts/greeting before creating an IVR menu.

Create New VoiceMenu X

Name: ⓘ

Extension: ⓘ

ⓘ Allow Dialing Other Extensions

Actions ⓘ

Add new Step:

ⓘ Allow KeyPress Events



- Name – indicates a unique identifier assigned to your VoiceMenu. For example “Customer Service” or “Sales Office” .
- Extension - specifies an extension number to invoke your Voice Menu.
- Allow Dialing Other Extensions – controls if extensions which were not explicitly listed are also accessible
- Actions - list all defined steps for this Voice Menu.
- Add new step – List of all available options for Voice Menu. You have to choose and add them one by one.

The screenshot shows a web form titled "Create New VoiceMenu". It has the following elements:

- Name:** A text input field with a blue information icon and an "Advanced Edit" button.
- Extension:** A text input field containing "7000" with a blue information icon.
- Allow Dialing Other Extensions:** A checkbox that is currently unchecked, with a blue information icon.
- Actions:** A large empty rectangular box for listing steps.
- Add new Step:** A dropdown menu is open, showing a list of actions: "-- Select an Option --", "Answer", "Authenticate", "Background" (highlighted), "Busy Tone", "Congestion", "DigitTimeout", "DISA", "ResponseTimeout", "Macro", "Playback", "Ringing", "Set MusicOnHold Class", "SayAlpha", "SayDigits", "SayNumber", "Wait", "WaitExten", "Goto Destination", and "Set Language".
- Buttons:** "Cancel" and "Save" buttons are located at the bottom right of the form.

- Allow Key Press Events – defines specific action to be executed upon pressing specific DTMF codes.



3.11. Time Intervals



New Time Interval [X]

Time Interval Name :

By day of week
 to

By Days of a Month
Date : Month :

Time: Entire Day
Start Time : End Time :

You can define here one more time intervals to allow different call handling during different time of day (week).

This application is very useful, when you want to answer calls only in specific time interval, of the day, week or month.

In the example here the setting has name “Office” and it is set for the regular business hours of the week. You can create several time intervals with different names and later access them under “Incoming Calling Rules” section.



3.12. Incoming Calling Rules



This page defines how to handle incoming calls.

A screenshot of a web form titled 'New Incoming Rule' with a close button (X) in the top right corner. The form contains four fields: 'Trunk' (a dropdown menu), 'Time Interval' (a dropdown menu), 'Pattern' (a text input field with an information icon), and 'Destination' (a dropdown menu). At the bottom of the form are two buttons: 'Cancel' and 'Update' (which has a checked checkbox icon).

- Trunk – indicates incoming trunk to be handled, ie “bri2”.
- Time Interval – indicates time frame when this rule will apply
- Pattern – indicates NDIS number (dialed) number using the same patterns as listed in the “**Outgoing Calling Rules**” section.
- Destination – specifies an extension, IVR Menu or application which will terminate the incoming calls, providing all the rules above apply.



3.13. Voice mail

General Settings **Email Settings for VoiceMails**

General VoiceMail Settings

Extension for checking messages ⓘ :

Direct Voicemail Dial ⓘ :

Max greeting (in seconds) ⓘ :

Dial '0' for Operator ⓘ :

Message Options

Maximum messages per folder ⓘ :

Max message time ⓘ :

Min message time ⓘ :

Playback Options

Say message Caller-ID ⓘ :

Say message duration ⓘ :

Play envelope ⓘ :

Allow users to review ⓘ :

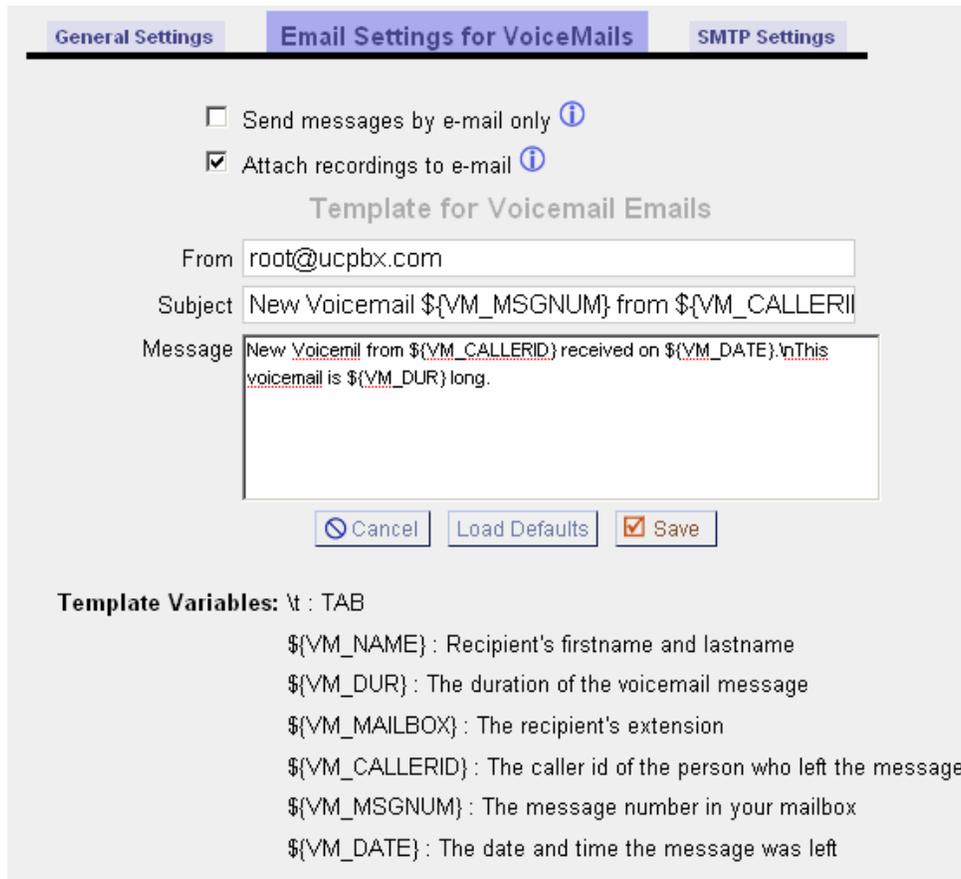
**Reminder – You can always use the “i” (info) tooltips for additional information*

The General Settings Menu:

- Extension for checking messages – you have to print the number, you want to use for checking your Voicemail. For example 111
- Direct Voice mail
- Max Greeting – is the length of the greeting in seconds
- Dial “0” for operator
- Message Options
- Maximum messages per folder – you have choice from 10 to 1000
- Max time – you have to select what is the maximum length of recorded message
- Main time – you have to select what is the minimum length of recorded message
- Playback Options - they are simple and very useful



Email Settings for Voicemail:



General Settings | **Email Settings for VoiceMails** | SMTP Settings

Send messages by e-mail only ⓘ
 Attach recordings to e-mail ⓘ

Template for Voicemail Emails

From: root@ucpbx.com
Subject: New Voicemail \${VM_MSGNUM} from \${VM_CALLERID}
Message: New Voicemail from \${VM_CALLERID} received on \${VM_DATE}. This voicemail is \${VM_DUR} long.

Template Variables: \t : TAB

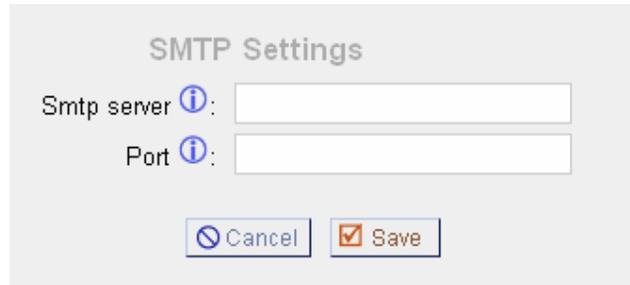
- `\${VM_NAME}` : Recipient's firstname and lastname
- `\${VM_DUR}` : The duration of the voicemail message
- `\${VM_MAILBOX}` : The recipient's extension
- `\${VM_CALLERID}` : The caller id of the person who left the message
- `\${VM_MSGNUM}` : The message number in your mailbox
- `\${VM_DATE}` : The date and time the message was left

**Reminder – You can always use the “i” (info) tooltips for additional information*

- Send messages – if this option is set, the voicemails will not be kept on the server. They will be sent directly to the e-mail.
- Attach recordings – this option defines whether or not to attach recordings to voicemail attachments. Note: You need SMTP server set for using this option.
- There are a couple options about the emails. You can set the PBX to send e-mail with information about the caller ID, message number, recipient's names and etc.



SMTP Settings



**Reminder – You can always use the “i” (info) tooltips for additional information*

- STMP server – it is the IP/host name of the outgoing mail server that your BR4 will connect and send e-mails with **voicemail** notifications
- Port – The port number on which the SMTP server is running

3.14. Conferencing



After loading this page on the system which has not been yet configured, you will need to create a “New Conferencing Bridge”. The following menu will appear:



New Conference Bridge X

Extension : 6300 i Marked/Admin user Extension : 6400 i

Password Options:

Pin Code: 1234 i Admin PinCode: 5678 i

Conference Room Options:

i Play hold music for first caller i Close conference when last marked user exits

i Enable caller menu i Announce callers

i Quiet Mode i Wait for marked user

**Reminder – You can always use the “i” (info) tooltips for additional information*

The Conference Bridge can be used for creating a conference call with several participants. The bridge has public extension for all users and admin extension for the administrator (if applicable). Access to the conference call could be protected with PIN codes for both admin and public.

- Extension – public extension for entering the bridge
- Marked/Admin user Extension – administrative extension
- Pin Code – optional Pin codes associated with public and admin extensions.
- Conference room options
- Play hold music – this option causes Asterisk to play Hold Music to the first user in the conference, until another user joins the same conference
- Caller Menu – checking this option allows the user to access the Conference menu by pressing “ * ” on his numpad
- Quiet Mode – do not play enter/leave sounds
- Close conference when last marked user exits – when the last user logouts from the bridge, close it.
- Announce callers – when checked the participants of the bridge are announced when another participant is joining the bridge.
- Wait for marked users – prevent conference participants from hearing each other, until marked user has joined

Note that the conference can starts after entrance of the admin. If the admin is not available all users hear the default music on hold.



3.15. Follow me



- The follow me option allows to specify sophisticated call handling for selected extensions. Calls can be routed to other applications, local numbers and external numbers. By default “FollowMe” options are disabled for all users.

A configuration window titled 'Follow Me' with a close button 'X' in the top right corner. The window contains the following settings:

- Status: Enable Disable
- 'Music On Hold' Class: default
- DialPlan: DialPlan1
- Destinations: A list box containing two entries: '+18001000 (30 seconds)' and '+3591234 (30 seconds)'. Each entry has three small icons to its right: a downward arrow, an upward arrow, and an 'X'.
- New FollowMe Number: Dial Local Extension Dial Outside Number
- Below the radio buttons: a dropdown menu showing '6001 David' and a text box containing '30', followed by the word 'Seconds'.
- Dial Order: Ring after Trying previous extension/number Ring along with previous extension/number
- Buttons: 'Cancel' and 'Add'.

**Reminder – You can always use the “i” (info) tooltips for additional information*

- Status – Select to enable or disable it on per user basis.
- Music on hold class – select the appropriate MOH class.
- DialPlan – Select Class of Service to control access to outbound trunks.
- Destination – list of numbers in priority sequence



- Add Number – you can add local numbers or outside numbers.
- Dial Order – method of handling the call

For some cases you may find the additional follow me options useful.



- Playback the incoming status messages prior to starting the follow-me step(s) – by checking this option there will be a message for the caller, before starting the follow me steps
- Record the caller’s name –
- Playback the unreachable status message – if this option is checked there will be status message for the caller when we’re out of follow me steps or the callee wants to be not reachable.

3.16. Directory



- Directory extension provides searchable list of users for IVR purposes. User will be listed in the Directory, if the field “In Directory” is selected under the “Users” settings window. The directory can be based on the users first or last name.



Directory Settings

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. To add or remove a user from the system telephone directory, edit the 'In Directory' field of the user.

Directory Extension ⓘ :

Also read the extension number ⓘ :

Use first name instead of last name ⓘ :

- Directory extension – extension to dial for accessing the name directory
- Also read extension – in addition to the name also read the extension number to the caller before presenting the dialing options
- Use first or last name – when this option is checked the caller is allowed to enter the first name in to the directory instead of using the last name.

3.17. Call Features



Features Codes

Blind Transfer (default is #)

Disconnect (default is *)

Attended transfer

Call Parking



- Feature Codes section specifies DTMF sequence to answer specific services (features).

- Call Parking - section allows to configure functionality that allows the user to put a call on hold and pick it up at a different phone. Single call parking extensions can be defined here and a range of call parking “spots” where calls will be parked in sequence.

Enabled	Feature Name	Digits	Active On/By	App Name	Arguments	
<input type="checkbox"/>			self			Delete

- Application Map section - allows definition of a key-sequence (DTMF digit sequence), an application and the party on which this application is executed when the sequence is pressed.
- Dial Options section allows to specify how the Feature Codes applies to the calling party and called party.



Feature Codes Call Parking Application Map **Dial Options**

Dial Options

(t-Option) Allow the called party to transfer the calling party by sending the DTMF sequence defined on the Feature Codes page.
 (T-Option) Allow the calling party to transfer the called party by sending the DTMF sequence defined on the Feature Codes page.
 (h-Option) Allow the called party to hang up by sending the DTMF sequence defined on the Feature Codes page.
 (H-Option) Allow the calling party to hang up by sending the DTMF sequence defined on the Feature Codes page.
 (k-Option) Allow the called party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page.
 (K-Option) Allow the calling party to enable parking of the call by sending the DTMF sequence defined on the Feature Codes page.

3.18. VoiceMail Groups



VoiceMail Groups allows to create a virtual mailbox allowing to distribute a message to several mailboxes at once. In the example below, mailboxes 6204, 6210 and 6250 are grouped under virtual mailbox 6600. Any messages recorded for mailbox number 6600 will appear in mailboxes 200, 204 and 207. This is useful to for teams such as **support** or **sales**.

Edit Voice Mail Group - 6600
X

VoiceMail Group's Extension:

Label:

User MailBoxes: 6200 6204 6210 6250 6260



- **Extension** to access the voicemail of the group.
- **Label** used for reference
- **MailBoxes** distribution list

3.19. Voice Menu Prompts



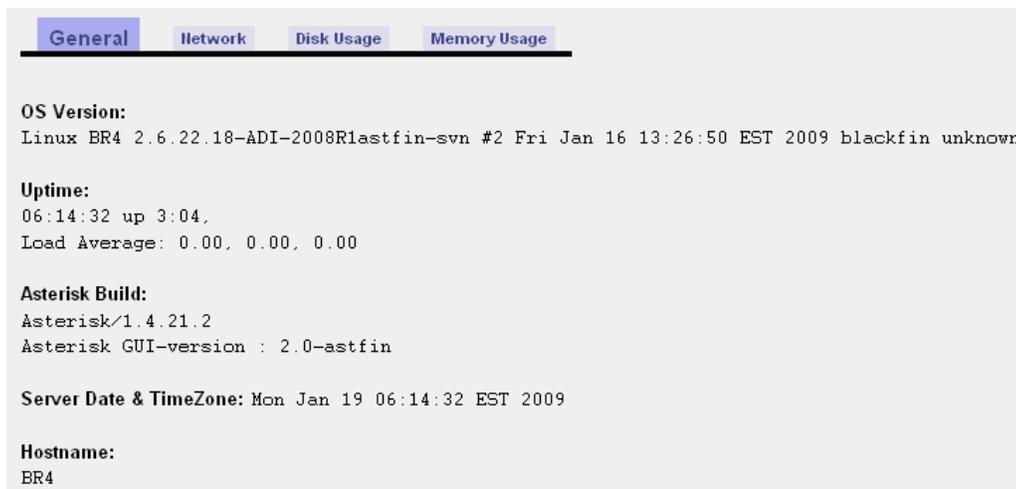
- You can record custom voice menu prompt for IVR purposes. This could be a greeting or instructions in foreign languages. You need to specify a valid extension to perform recording procedure. Please note that at this time we do not support uploads of your custom prompts through the GUI. Shell access and ftp, scp will need to be used.

List of Custom Voice Menu Prompts

#	Name	Options
1	main_english.gsm	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Delete"/>
2	main_bulgarian.gsm	<input type="button" value="Record Again"/> <input type="button" value="Play"/> <input type="button" value="Delete"/>



3.20. System Info



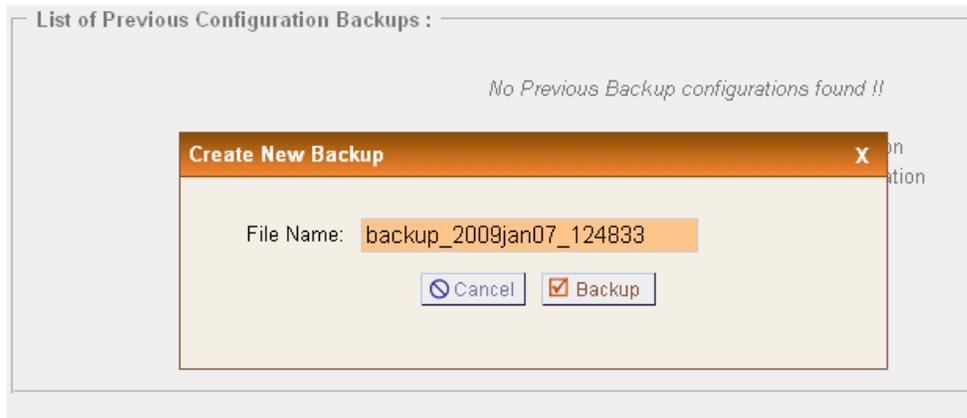
System info page provides information about vital components of your system.

3.21. Back up

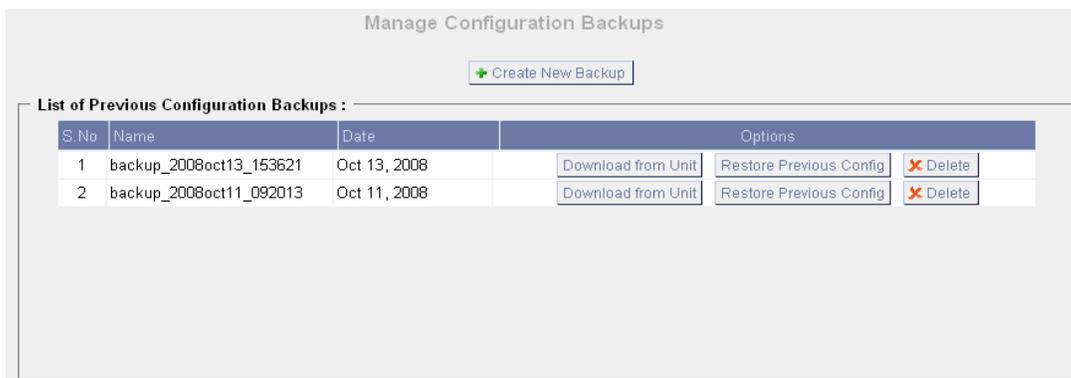


With this option you can make a back up of your system configuration, saving all the settings of the PBX. All the previous back up files are visible in to window.

- If you press **Create New Backup** button all the PBX configurations will be stored in a single backup file stored in the PBX itself. Please note that at this point we will only backup the content of /etc/asterisk. Your voicemail, custom recordings and networking setup will not be saved.



- The backup file will be tagged with the current date and the backup will be displayed in the list of backups as it is shown below



- You can download the backup file from the PBX to your local PC for even safer storage. Do this by pressing **Download from Unit** button
- If you want to restore the configurations which is backed up on the given date just press the corresponding button **Restore Previous Config**.

Please note that we do not provide upload functionality at this point.



3.22. Options



This is the general view of the Options menu

**Reminder – You can always use the “i” (info) tooltips for additional information*

The first tab in the menu is General preferences.

- Global Outbound CID – Caller ID to be used for outbound calls, if no specific callerID is defined in caller’s profile.
- Operator Extension – extension that will be dialed when a user presses “0” in VoiceMail menu.
- Ring Timeout – default global timeout.
- Extension Preferences – ranges of extensions for specific features of the BR4 - Appliance



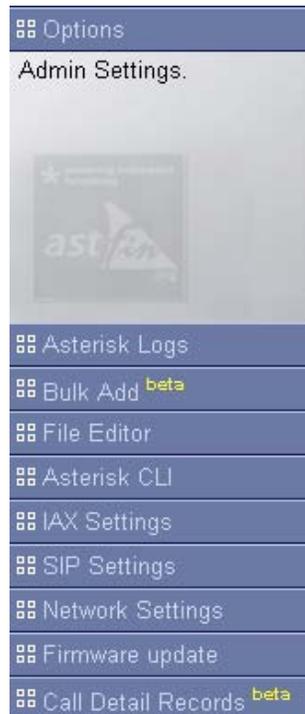
- Language – You can select the GUI language here. Only English is supported at this time.
- Password – Change your admin password
- Reboot – reboot the appliance.

3.23. Advanced Options

To enable the advanced option, click on the “Options” button, select the “Advanced options” tab and then press the “Show advanced options” button.



You will see the following at the bottom left of your GUI main menu.



Network settings

eth0 Interface

DHCP:

Hostname:

Domain:

IP address:

Subnet mask:

Gateway:

DNS:

NTP:

VLAN Interface for Eth0

VLAN:

Vlan number:

Vlan IP address:

Vlan Subnet mask:

Vlan Gateway:

System TimeZone

TimeZone:



- DHCP - selection determines if the PBX will use static IP or it will obtain its IP from the DHCP server of the network BR4 -Appliance is connected to.
- Hostname - is the hostname of the PBX. This is the name which will be used in any log and cdr files.
- IP address - is the Internet Protocol (IP) address of the BR4 - Appliance. Please note that this field is only editable if static IP address is selected (no DHCP client)
- Subnet mask – Defines size of your LAN, please use xxx.xxx.xxx.xxx notification to specify your subnet.
- Gateway – Indicates IP address of the default router on your network.
- DNS – Indicates Domain Name Server, to be used to resolve names to IP addresses.
- IP/host name of your preferred NTP server. If unsure specify pool.ntp.org

Call detail records – provides you with information about all the calls made thru your PBX. From this record we can see the following information: source and destination of the call, start time, duration, disposition etc.

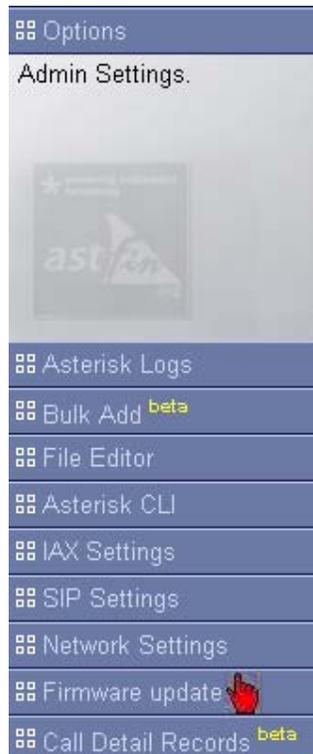
Account Code	Source	Destination	Dest_Context	Caller_ID	Channel	Dest_Channel	Last_app.	Last_data	Start_time	Answer Time	End Time	Duration	Billable seconds	Disposition	AMA flag
1	6200	6221	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01326584		VoiceMailMain	6200@default	2009-01-12 06:38:03	2009-01-12 06:38:03	2009-01-12 06:38:47	44	44	ANSWERED	DOCUM
2	6200	6221	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01326584		VoiceMailMain	6200@default	2009-01-12 06:37:40	2009-01-12 06:37:40	2009-01-12 06:37:59	19	19	ANSWERED	DOCUM
3	6200	6600	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01350004		VoiceMail	6204@default86210@default86250@default	2009-01-12 06:37:15	2009-01-12 06:37:15	2009-01-12 06:37:32	17	17	ANSWERED	DOCUM
4	6200	6600	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-0103e004		VoiceMail	6204@default86210@default86250@default	2009-01-12 06:37:09	2009-01-12 06:37:09	2009-01-12 06:37:12	3	3	ANSWERED	DOCUM
5	6210	6211	DLPN_DialPlan1	""Sales Department Secretary"" <6210>	SIP/6210-01350004	IA/2/6211-343	Dial	IA/2/6211	2009-01-12 06:15:41		2009-01-12 06:16:41	90	0	NO ANSWER	DOCUM
6	6200	6260	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-0103e004	IA/2/6260-650	Dial	IA/2/6260	2009-01-12 06:15:36	2009-01-12 06:15:41	2009-01-12 06:16:38	62	67	ANSWERED	DOCUM
7	6200	6260	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01350004	IA/2/6260-519	Dial	IA/2/6260	2009-01-12 06:05:09	2009-01-12 06:05:13	2009-01-12 06:06:23	74	70	ANSWERED	DOCUM
8	6210	6211	DLPN_DialPlan1	""Sales Department Secretary"" <6210>	SIP/6210-0103e004	IA/2/6211-320	Dial	IA/2/6211	2009-01-12 06:03:47		2009-01-12 06:06:21	164	0	NO ANSWER	DOCUM
9	6210	6211	DLPN_DialPlan1	""Sales Department Secretary"" <6210>	SIP/6210-0103e004	IA/2/6211-731	Dial	IA/2/6211	2009-01-12 06:03:42		2009-01-12 06:03:43	1	0	NO ANSWER	DOCUM
10		6200	DLPN_DialPlan1		IA/2/6260-704	SIP/6200-0103e004	Dial	SIP/6200	2009-01-12 06:02:50	2009-01-12 06:03:05	2009-01-12 06:03:28	98	23	ANSWERED	DOCUM



3.23.1 Updating your BR4 firmware

Your BR4-Appliance can be updated over the network.

Here you have to select “Firmware update”



- You can use your LAN (preferred) or WAN and either TFTP server or HTTP server to distribute your new uImage. After compiling your new image in Astfin, you will see that uImage and uImage-md5 files are available under build_br4/image_br4/ folder. Please use uImage-md5 for all updates through the GUI.
- You have to type the following information, and click the + **Go** button. If you have a local TFTP server, you have to type its IP address and type the name of the uImage in the field **File Name**.

Download image from a :

HTTP URL TFTP Server

TFTP Server :

File Name ⓘ :

Reset Configs ⓘ :

**Reminder – You can always use the “i” (info) tooltips for additional information*



You need to use the image with the included md5 check sum which is automatically generated when you compile **Astfin**. The image you will find in the normal image directory. Please don't forget to put the image in the TFTP directory of your server. Alternatively, HTTP URL link can be used instead.

Important* – In order to complete the upgrade of the unit you have to **reboot the system.

