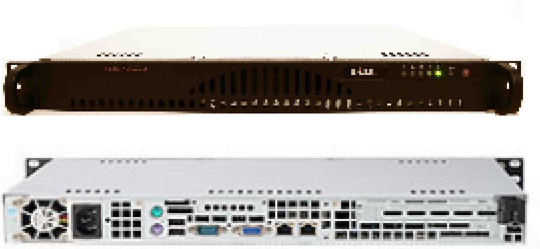
DIAL Systems VX Series IP PBX Administrator Manual

VX-100



VX-200

Table of Contents

Introduction

Thank you for purchasing the VX-100 / VX-200, a SIP based, affordable, feature enriched converging communication platform designed to meet the communication requirements for small to medium sized enterprises.

The VX100/200 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 VX-100/200 can easily be integrated into other components of your existing communications network while providing a rich set of features to reduce costs and increase productivity. Built-in FXO ports enable the VX Series to interface with analog lines and devices while concurrently registering to SIP trunks and SIP based trunk gateways to maximize available communications resources.

The VX-100/200 is easy to setup and has an intuitive user interface that allows users to quickly configure extensions. Voice mail, fax mail, fax-to-email, conference bridges and other enhanced features can be configured with minimal effort via the webGUI.

The VX-100/200’s broad feature set, ease of operation and affordable price range makes it ideal for any SMB and Enterprise level company.

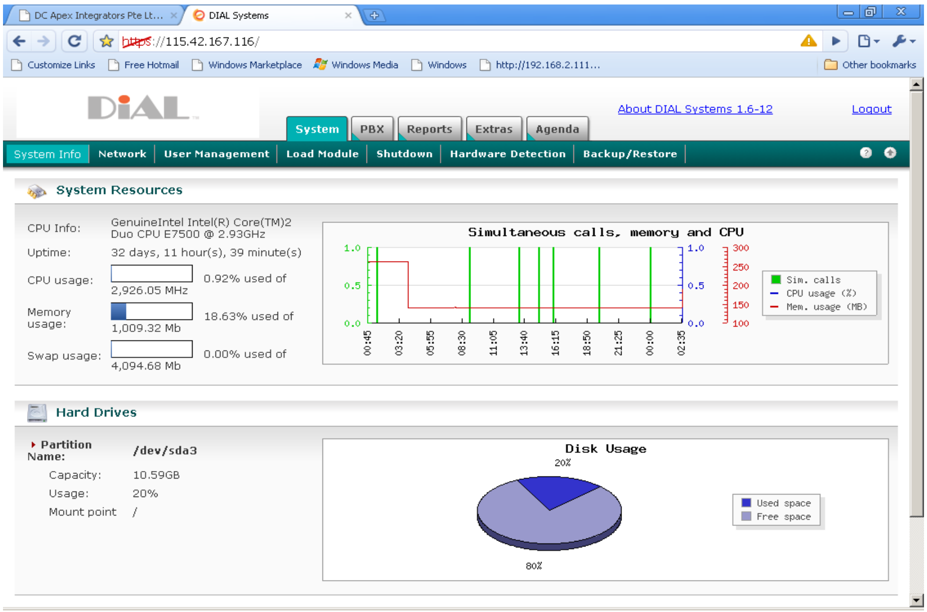
Equipment Packaging

The VX-100/200 IP PBX package should contain:

- 1 x IP PBX unit
- 1 x 230 AC Volt 3 Pin Power Adapter
- 1 x Ethernet Cable
- 1 x CD

1. System

The front end menu of the VX-100/200 web interface is the System Resource Summary.

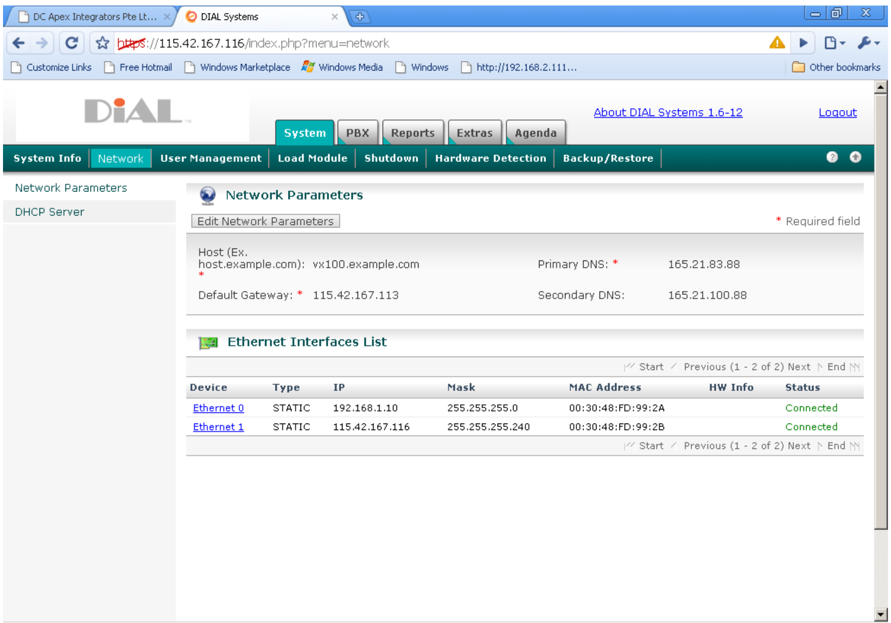


In this screen, you are able to monitor the overall performance of your VX-100/200. We will go through the different tabs/menus in this screen to help you configure and customize your VX-100/200 to your liking.

System Resources Summary

• System Info: Shows you a summary of the usage of your hardware; CPU usage, Memory usage, Swap Usage, Hard drive etc.

Network



The '**Network**' tab allows you to change your network configurations as per your environment.

Network Parameters

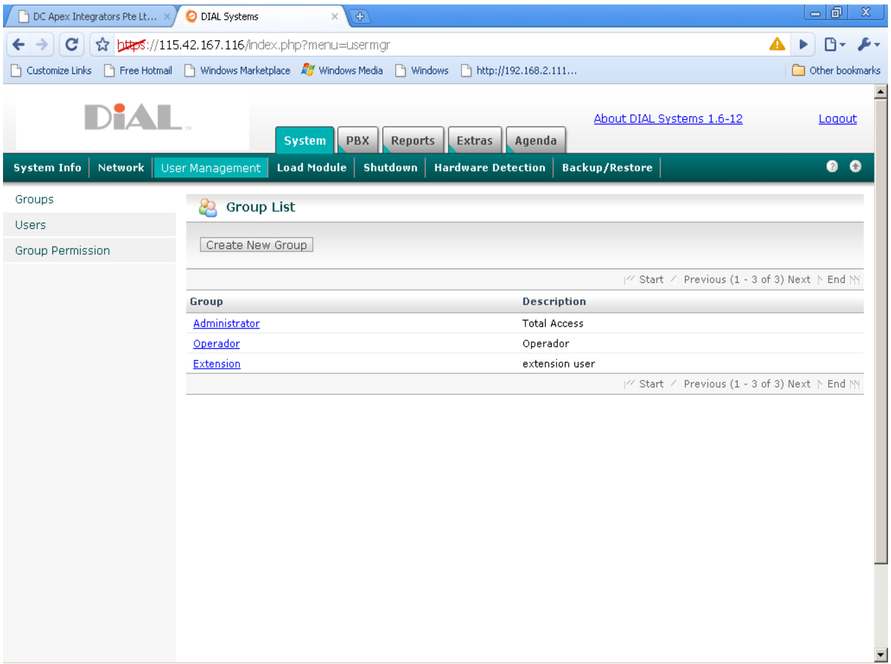
- Here you are able to change your Host, Default Gateway, Primary and Secondary DNS.
- Fill in your customized settings where applicable.
- Click on '**Save**' to keep your settings.

Tips: You may use Eth 0 as LAN and ETH 1 as WAN. Remember to implement security firewall to protect the system from hacking. Use your network creativity to secure it.

User Management

This screen allows you to configure users and administrators allowed on the VX-100/200 and control permissions allocated to different users as well as passwords.

Group List



In this screen, you may add new groups; Administrator, user etc (Recommend to use default setting)

- Click on '**Create New Group**'.
- Fill in the details which follow (*Group Name & Description*).
- Click on the '**Save**' button.

Users

This features allows user to access various functionality such as;

- View users' extensions dash board
- Webmail
- Voicemail with filtering
- Call Monitoring (Voice Logger Module require)
- Address Book with click to dial
- Calender

You may add/edit/delete users in this screen based on their extension number.

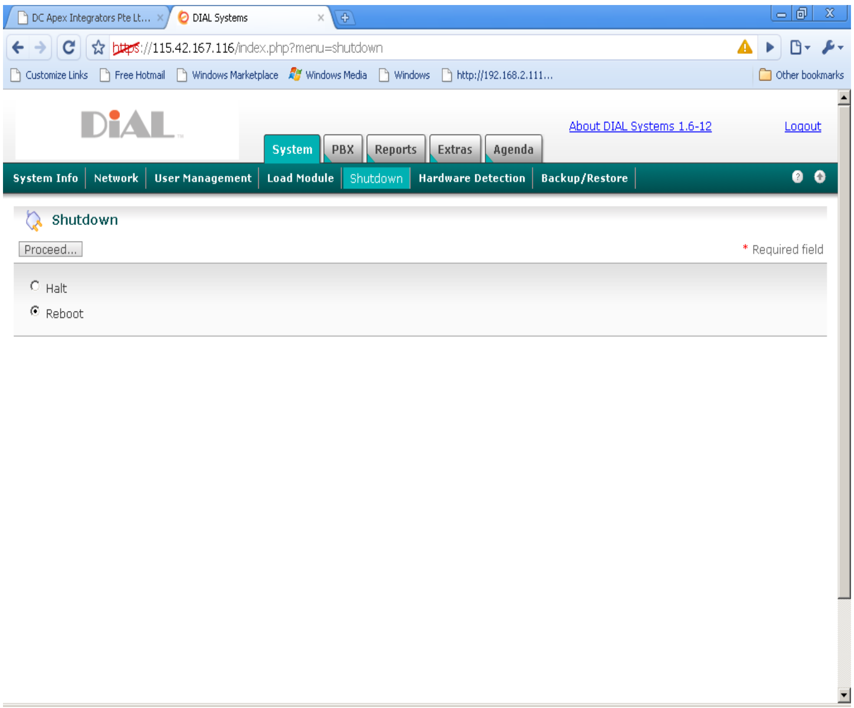
To add a new user:
- Click on '**Create New User**'.
- Fill in the required details of your new user (*Login, Password, Group, Name and Extension*)
- Fill in the Mail Profile section of your new user (*Webmail User, Webmail Domain, Webmail password*)
- Click on the '**Save**' button.

To edit/delete an existing user:
- On the User List screen,

click on the existing user you wish to edit/delete.
- You will be presented with the View User screen.
- Click on the '**Edit**' button to make the necessary changes to your existing user or the '**Delete**' button to delete the selected user.
- Click on the '**Apply Changes**' button to keep your changes.

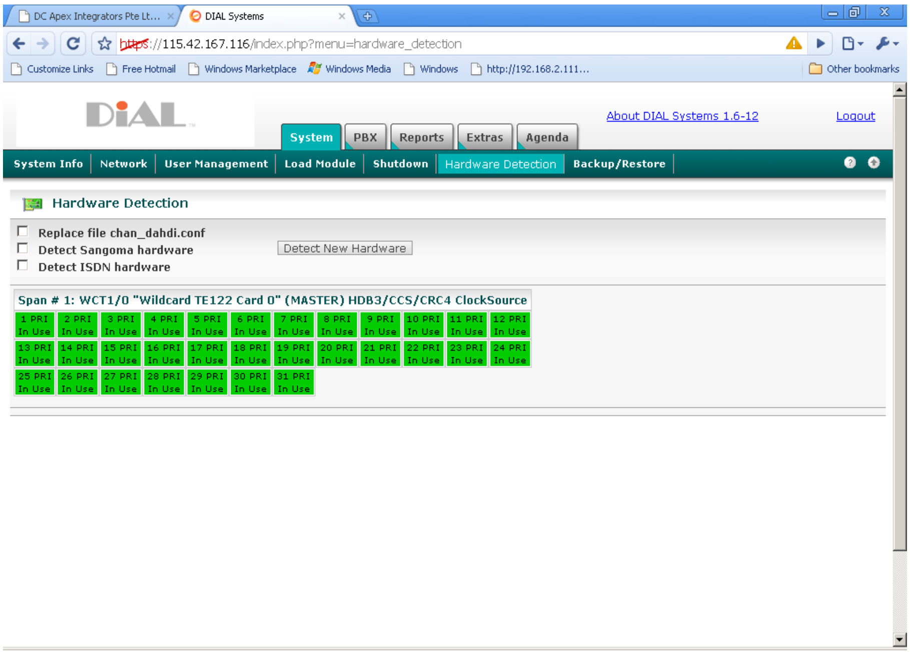
Note: Remember to engage a strong password for security reason

Shutdown



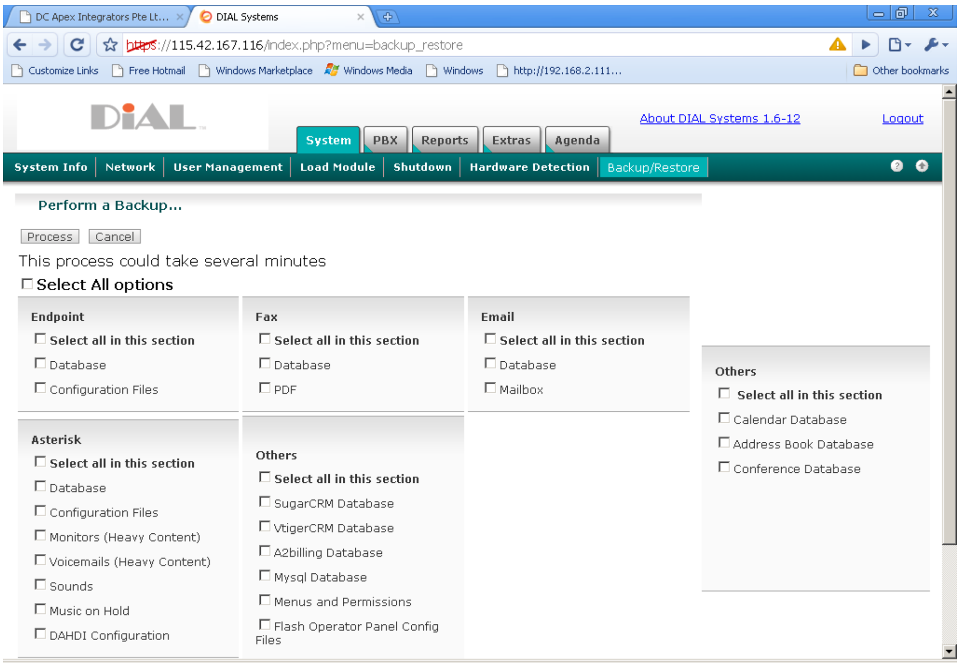
This menu allows you to shutdown or reboot the VX-100/200. Select '**Accept**' when done and the IP PBX will perform the selected task. After reboot, log in again to access the pages.

Hardware Detection



The VX-100/200 is able to automatically detect new hardware installed on the IP PBX. Click on the '**Detect New Hardware**' button to scan for new hardware installed on the VX-100/200.

Backup/Restore



This functionality of the VX-100/200 allows you to restore a backup file with all your configuration settings in the event of an emergency.
To do this:

- Check the box next to the backup file you wish to use for the restore process.
- Click the '**Backup**' button to apply.

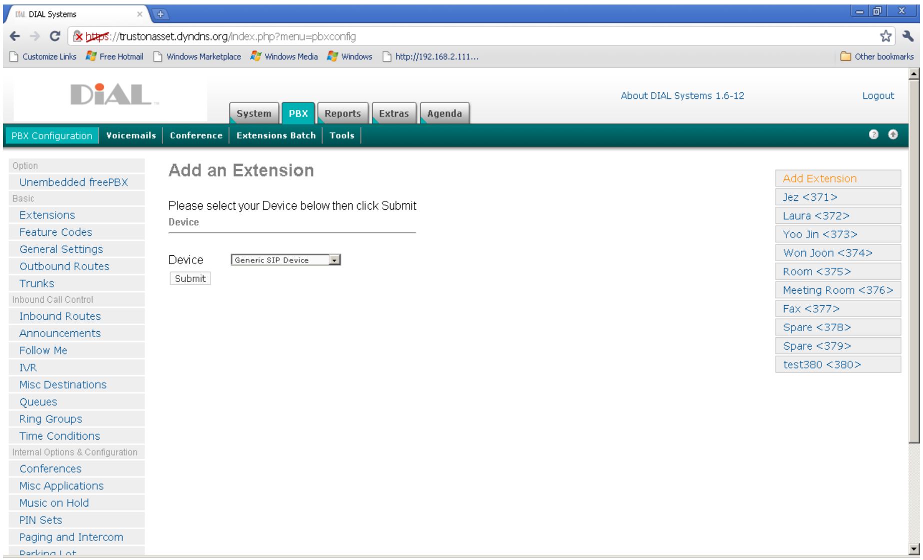
2. PBX Configuration

Extensions

Upon clicking the '**PBX**' tab on the top, you will be presented with the extensions menu. This menu and its several different sub menus allow users to configure and add extensions with ease. The flexibility of the VX- Series allows users to bring their office extensions virtually anywhere that has a broadband Internet connection. This gives users easy access to personnel (home office and road warriors alike) without incurring any PSTN toll charges.

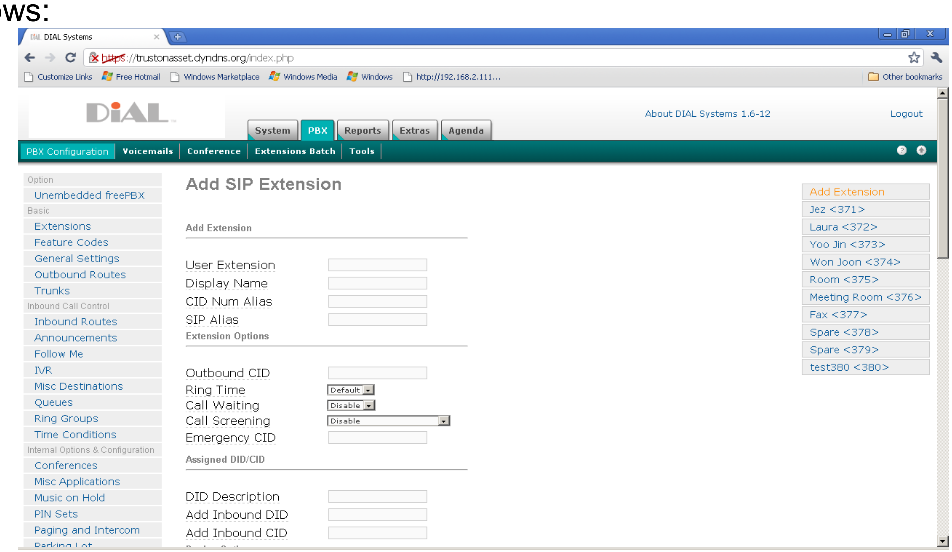
Below is a step by step guide to adding extensions to your VX Series.

- Select the '**PBX**' tab.
- You will be presented with the '**Add an Extension**' screen



- Select '**Generic SIP Device**' and click on the '**Submit**' button.

The next screen will require you to fill out the following details for your new extensions. They are as follows:



Add Extension Settings

- **User Extension:** The extension number.
- **Display Name:** The name of the extension that will appear when you make SIP calls.
- **CID Number Alias:** The Caller ID to use for internal calls, if different from the extension number.
- **SIP Alias:** If you want to support direct sip dialing of users internally or to be use as secondary extension from other phone device, you can supply a friendly name that can be used in addition to the users extension to call them.

Extension Options

- **Outbound CID:** Overrides the Caller ID when dialing out a trunk.
- **Ring Time:** The number of seconds to ring prior to going to voice mail.
- **Call Waiting:** The initial/current Call Waiting state for this extension.
- **Call Screening:** Allow callers to say their name which will allow users to accept or reject the calls.
- **Emergency CID:** This Caller ID will always be set when dialing out an Outbound Route flagged as emergency. This overrides all other Caller ID settings.

Assigned DID/CID

- **DID Description:** Give a unique name for the DID to be assigned.
- **Add Inbound DID:** The direct DID that is associated with this extension. The DID should be in the same format as provided by the provider. (Eg: The DID number allocated from the telco company) Leave it blank if not applicable. 'Inbound route' will be another alternative similar to this.
- **Add Inbound CID:** Add a CID to specify DID + CID Routing. Leave blank if not applicable

Device Options

- **Secret:** Your SIP device's password. (Remember to engage strong password!)
- **DTMF Mode:** Leave as default for IP phones (RFC2833)

Dictation Services

- **Dictation Service:** Select **Enable/Disable**.
- **Dictation Format:** Default is '**Ogg Vorbis**'.
- **Email Address:** The email address that completed dictations are sent to.

Recording Options

- **Record Incoming:** Records all inbound calls received at this extension.
- **Record Outgoing:** Records all outbound calls received at this extension.

Note: 'On Demand' recordings referring to Feature Code *1 for start and stop recording. 'Always' refer to auto record once a new In or Out going call start. Recorded calls can be retrieve from 'Monitoring' page.
Voice Logger Module is require to turn on this feature. Contact your system integrators for more information.

Voicemail & Directory

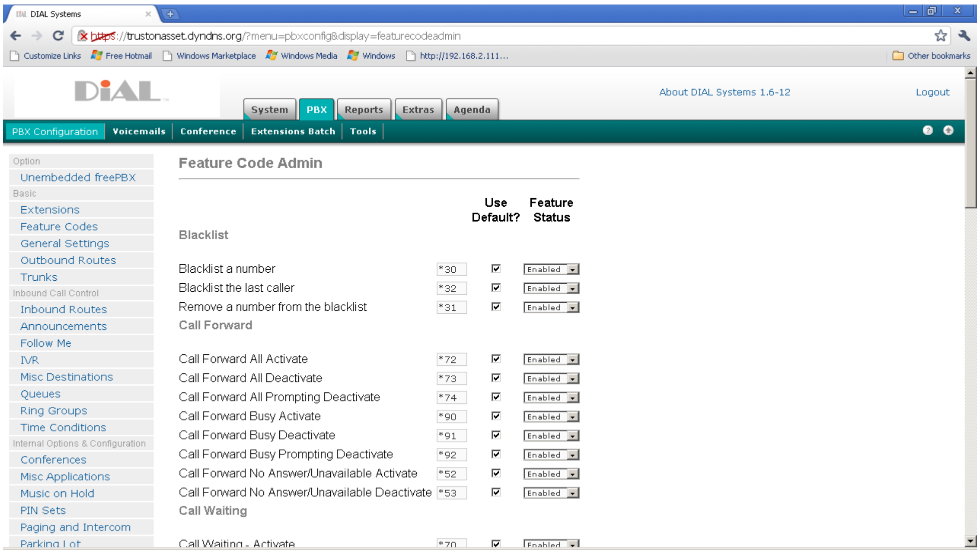
- **Status:** Enable or disable voicemail
- **Voicemail Password:** This is the password used to access the voicemail system. This password can only contain numbers.
- **Email Address:** The email address that voicemails are sent to.
- **Pager Email Address:** Pager/mobile email address that short voicemail notifications are sent to.
- **Email Attachment:** Option to attach voicemails to email.
- **Play CID:** Read back caller's telephone number prior to playing the incoming message, and just after announcing the date and time the message was left.
- **Play Envelope:** Envelope controls whether or not the voicemail system will play the message

envelop (date/time) before playing the voicemail message. This setting does not affect the operation of the envelope option in the advanced voicemail menu.

- **Delete Voicemail:** If set to "**yes**", the message will be deleted from the voicemail via email alone, rather than having the voicemail able to be retrieved from the Webinterface or the Extension handset.
- **VM Options:** Separate options with pipe (|)
- **VM Context:** This is the Voicemail Context which is normally set to default. Do not change unless you understand the implications.
- **VmX Locator:** Enable/Disable the VmX Locator feature for this user. When enabled, all settings are controlled by the user in the User Portal (ARI). Disabling will not delete any existing user settings but will disable access to the feature.

Feature Codes

Change, Enable/Disable the feature codes controlling features such as Call Waiting, Call Forwarding, Recordings, Call Tracing and other builtin features.

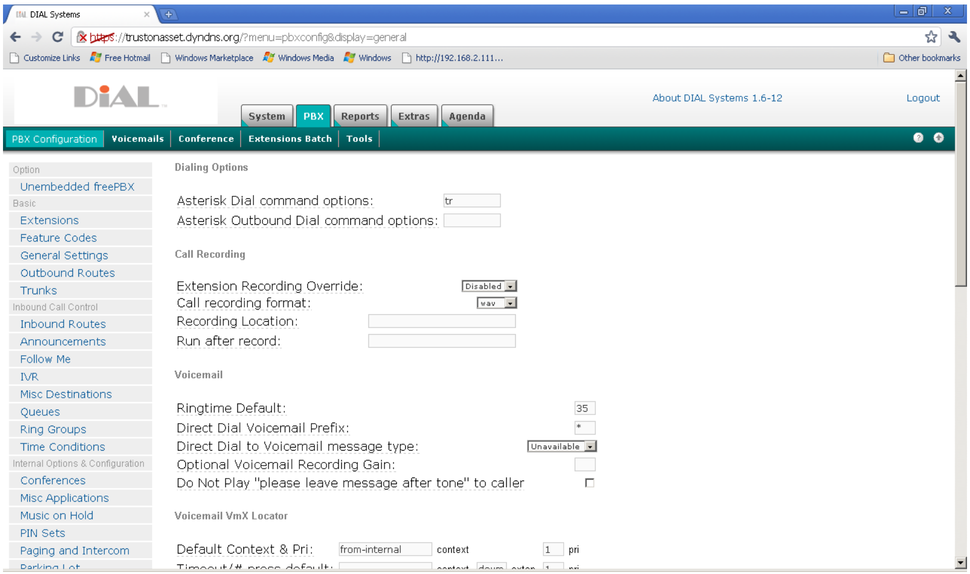


General Settings

Dialing Options

- **Asterisk Dial Command Options:** Input 't' to allow the called user to transfer call by hitting'#', 'T' to allow the calling user to transfer call by hitting '#','r' to generate a ringing tone for the calling party, 'w' to allow the called user to start recording by hitting '*1' and 'W' to allow calling user to start recording by hitting '*1'.
- **Asterisk Outbound Dial Command Options:** 't' to allow the called user to transfer the call by hitting'#', 'T' to allow the calling user to transfer the call with the '#' key, 'w' to allow the called user to start recording upon pressing '*1' and 'W' to allow the calling user to start recording by pressing '*1'.

See below for picture.



Voicemail

Here, you are able to set the the number of seconds to ring before sending callers to voicemail, the extension prefix for dialing direct to voicemail, set the amount of gain(if any at all) to use when voicemail is being recorded and have the option to have the message,
“Please leave your message after the tone” when callers are sent to voicemail.

Voicemail VmX Locator

This optional feature lets users set up a short menu before voicemail takes the actual message.

Company Directory

The Company Directory feature allows callers to search for personnel within your extension list by keying in either their first/last name.
To set this up:

- First select to find users in the Company Directory via first name, last name or first or last name.
- Check or uncheck the box which says 'Play extension number to caller before transferring call' as to per your preference.
- Input your operator extension number so as to direct callers to the operator in the event that the '0' key is pushed.

Fax Machine

Set up your fax settings to have the system handle your fax for you.

International Settings

Select your country and time display format(12/24hr).

Security Settings

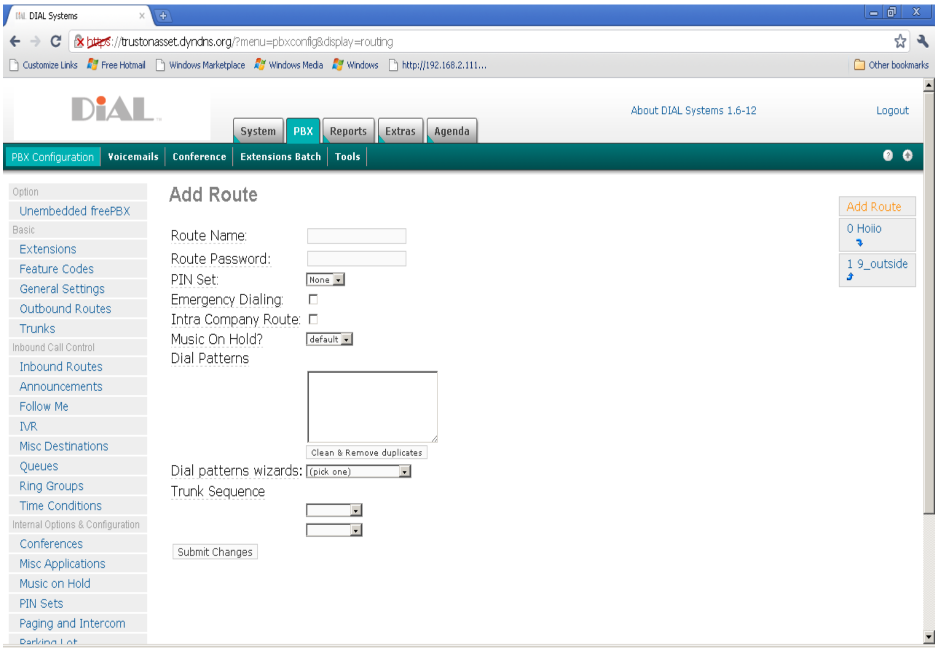
Here you may select to allow anonymous inbound SIP calls. Selecting '**yes**' will potentially allow anybody to call into your VX Series server using the SIP protocol. Only select '**yes**' if you fully understand the impact of allowing anonymous calls into your server.

Online Updates

Select '**yes**' to automatically receive updates. The system checks for updates nightly and the resulting information will be displayed in the dashboard and be sent to the email address you input. This will transmit your FreePBX and Asterisk version numbers along with a unique but random identifier. This is used to provide proper update information and to track version usage to focus development and maintenance efforts. No private information is transmitted.

Outbound Routes

This screen allows you to create an outbound route for local or international calls.



Adding a Route

- **Route Name:** Name your route. This should be used to describe what type of calls this route matches (for example, 'local' or 'longdistance').

- **Route Password:** This is optional. A route can prompt users for a password before allowing calls to progress. This is useful for restricting calls to international destinations. Leave this blank to not prompt for a password.

- **PIN Set:** This is optional. Select a PIN set to use. If not using this option, leave the Route Password field blank.

- **Emergency Dialing:** Selecting this option will enforce the use of a device's Emergency CID setting (if set). Select this option if the set of routes is used for emergency dialing.

- **Intra Company Route:** Selecting this option will treat this route as an intracompany connection, preserving the internal Caller ID information and not the outbound CID of the extension or trunk.

- **Music On Hold:** You can choose which music category to use. For example, choose a type appropriate for a destination country which may have announcements in the appropriate language. Select default to use system setting.

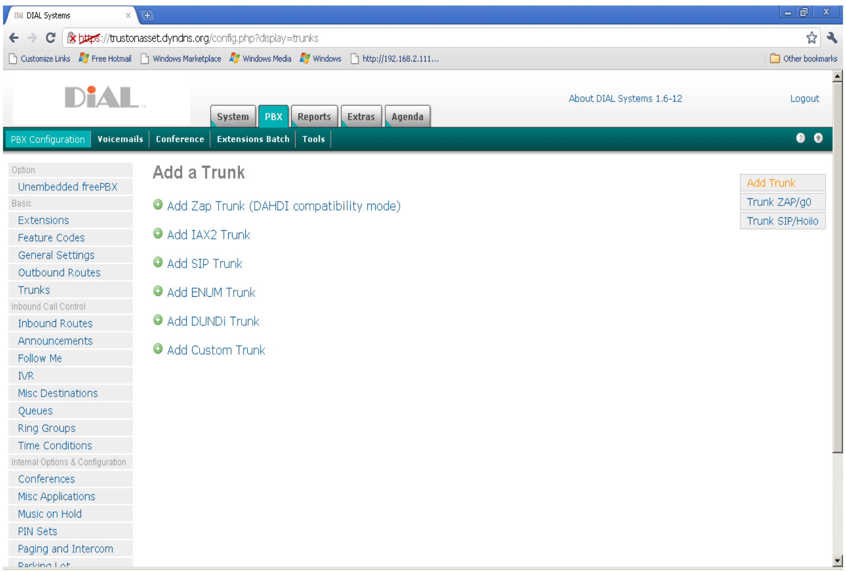
• **Dial Patterns:** A Dial Pattern is a unique set of digits that will select this trunk. Enter one dial pattern per line.
Rules:
X matches any digit from 09
Z matches any digit from 19
N matches any digit from 29
[12379]
matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
. wildcard, matches one or more characters
| separates a dialing prefix from the number (for example, 9|NXXXXXX would match when some dialed "95551234" but would only pass "5551234" to the trunks)

• **Dial Patterns Wizard:** These options provide a quick way to add outbound dialing rules. Follow the prompts for each.

• **Trunk Sequence:** The Trunk Sequence controls the order of trunks that will be used when the above Dial Patterns are matched. For Dial Patterns that match long distance numbers, for example, you would want to pick the cheapest routes for long distance (ie, VoIP trunks first) followed by more expensive routes (POTS lines).

Trunks

This menu allows you to manage your trunks. Add/Edit/Delete ZAP, IAX2, SIP, ENUM, DUNDi or custom trunks here. Default setting are available for ZAP Trunk.



Adding a trunk

Zap Trunk

- **Outbound Caller ID:** Caller ID for calls placed out on this trunk. Format is "caller name" <#######>. You can also use the magic string 'hidden' to hide the CallerID sent out over Digital lines ONLY (E1/T1/J1/BRI/SIP/IAX).

- **Never Override Caller ID:** Some VoIP providers will drop the call if you try to send an invalid CallerID (one you don't own.Check this box to never send a CallerID that you haven't explicitly specified in this trunk or in the outbound Caller ID field of an extension/user. You might notice this problem if you discover that the **FollowMe** or **RingGroups** with external numbers do not work properly. Checking this box has the effect of disabling 'foreign' Caller IDs from going out this trunk. You must define an Outbound Caller ID on this trunk when checking this.

- **Maximum Channels:** Controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. Inbound calls are not counted against the maximum. Leave blank to specify no maximum.

- **Disable Trunk:** Check this to disable this trunk in all routes where it is used.

- **Monitor Trunk Failures:** If checked, supply the name of a custom AGI Script that will be called to report, log, email or otherwise take some action on trunk failures that are not caused by either NOANSWER or CANCEL.

- **Outgoing Dial Rules:** A Dial Rule controls how calls will be dialed on this trunk. It can be used to add or remove prefixes. Numbers that don't match any patterns defined here will be dialed as it is. Note that a pattern without a + or | (to add or remove a prefix) will not make any changes but will create a match. Only the first matched rule will be executed and the remaining rules will not be acted on.

Rules:
X matches any digit from 09
Z matches any digit from 19
N matches any digit from 29
[12379]
matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9) . wildcard, matches one or more characters (not allowed before a | or +) | removes a dialing prefix from the number (for example, 613|NXXXXXX would match when some dialed "6135551234" but would only pass "5551234" to the trunk) + adds a dialing prefix from the number (for example, 1613+NXXXXXX would match when some dialed "5551234" and would pass "16135551234" to the trunk). You can also use both +and |, for example: 01+0|1ZXXXXXXXXX would match "016065551234" and dial it as "0116065551234" Note that the order does not matter, eg. 0|01+1ZXXXXXXXXX does the same thing.

- **Dial Rules Wizards:**

Always dial with prefix is useful for VoIP trunks, where if a number is dialed as "5551234", it can be converted to "16135551234". **Remove prefix from local numbers** is useful for ZAP trunks, where if a local number is dialed as "6135551234", it can be converted to "5551234".
Lookup numbers for local trunk looks up your local number on

www.localcallingguide.com (Naonly), and sets up so you can dial either 7 or 10 digits (regardless of what your PSTN is) on a local trunk (where you have to dial 1+areacode for long distance, but only 5551234 (7digit dialing) or 6135551234 (10digit dialing) for local calls

- **Outbound Dial Prefix:** The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put '9' here to access an outbound line. Another common use is to prefix calls with 'w' on a POTS line that need time to obtain dial tone to avoid eating digits. Most users should leave this option blank.

- **Zap Identifier (trunk name):** ZAP channels are referenced either by a group number or channel number (which is defined in zapata.conf). The default setting is g0 (group zero).

IAX2 Trunk

- **Outbound Caller ID:** Caller ID for calls placed out on this trunk. Format is "caller name" <#######>. You can also use the magic string 'hidden' to hide the CallerID sent out over Digital lines ONLY (E1/T1/J1/BRI/SIP/IAX).

- **Never Override Caller ID:** Some VoIP providers will drop the call if you try to send an invalid CallerID (one you don't own.Check this box to never send a CallerID that you haven't explicitly specified in this trunk or in the outbound Caller ID field of an extension/user. You might notice this problem if you discover that the **FollowMe** or **RingGroups** with external numbers do not work properly. Checking this box has the effect of disabling 'foreign' Caller IDs from going out this trunk. You must define an Outbound Caller ID on this trunk when checking this.

- **Maximum Channels:** This controls the maximum number of outbound channels (simultaneous calls) that can be used on this trunk. To count inbound calls against this maximum, use the autogenerated context: fromtrunk[trunkname] as the inbound trunk's context. (see extensions_additional.conf) Leave blank to specify no maximum.

- **Disable Trunk:** Check this to disable this trunk in all routes where it is used.

- **Monitor Trunk Failures:** If checked, supply the name of a custom AGI Script that will be called to report, log, email or otherwise take some action on trunk failures that are not caused by either NOANSWER or CANCEL.

- **Outgoing Dial Rules:** A Dial Rule controls how calls will be dialed on this trunk. It can be used to add or remove prefixes. Numbers that don't match any patterns defined here will be dialed asis.
Note that a pattern without a + or | (to add or remove a prefix)
will not make any changes but will create a match. Only the first matched rule will be executed and the remaining rules will not be acted on.
Rules:
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N matches any digit from 29
[12379]
matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
. wildcard, matches one or more characters (not allowed before a | or +)

| removes a dialing prefix from the number (for example, 613|NXXXXXX would match when some dialed "6135551234" but would only pass "5551234" to the trunk) + adds a dialing prefix from the number (for example, 1613+NXXXXXX would match when some dialed "5551234" and would pass "16135551234" to the trunk). You can also use both + and |, for example: 01+0|1ZXXXXXXXXX would match "016065551234" and dial it as "0116065551234" Note that the order does not matter, eg. 0|01+1ZXXXXXXXXX does the same thing.

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Always dial with prefix is useful for VoIP trunks, where if a number is dialed as "5551234", it can be converted to "16135551234".

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Lookup numbers for local trunk looks up your local number on www.localcallingguide.com (Naonly), and sets up so you can dial either 7 or 10 digits (regardless of what your PSTN is) on a local trunk (where you have to dial 1+areacode for long distance, but only 5551234 (7digit dialing) or 6135551234 (10digit dialing) for local calls.

• **Outbound Dial Prefix:** The outbound dialing prefix is used to prefix a dialing string to all outbound calls placed on this trunk. For example, if this trunk is behind another PBX or is a Centrex line, then you would put '9' here to access an outbound line. Another common use is to prefix calls with 'w' on a POTS line that need time to obtain dial tone to avoid eating digits. Most users should leave this option blank.

• **Trunk Name:** The name you wish to label this trunk.

• **PEER Details:** Modify the default PEER connection parameters for your VoIP provider. You may need to add to the default lines listed below, depending on your provider. WARNING: Order is important as it will be retained. For example, if you use the "allow/ deny" directives, make sure deny comes first.

• **USER Context:** This is most often the account name or number your provider expects. This USER Context will be used to define the below user details.

• **USER Details:** Modify the default USER connection parameters for your VoIP provider. You may need to add to the default lines listed below, depending on your provider.

• **Register String:** Most VoIP providers require your system to REGISTER with theirs. Enter the registration line here.
example:
username: password@switch.voipprovider.com. Many providers will require you to provide a DID number, ex: username:password@switch.voipprovider.com/didnumber in order for any DID matching to work.

SIP Trunk

• **Outbound Caller ID:** Caller ID for calls placed out on this trunk. Format is "caller name" <#######>. You can also use the magic string 'hidden' to hide the CallerID sent out over Digital lines ONLY (E1/T1/J1/BRI/SIP/IAX).

• **Never Override Caller ID:** Some VoIP providers will drop the call if you try to send an invalid Caller ID (one you don't 'own'). Use this to never send a Caller ID that you haven't explicitly specified in this trunk or in the outbound callerid field of an extension/user. You might notice this problem if you discover that FollowMe or RingGroups with external numbers don't work properly. Checking this box has the effect of disabling 'foreign' callerids from going out this trunk. You must define an Outbound Caller ID on the this trunk when checking this.

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• **Disable Trunk:** Check this to disable this trunk in all routes where it is used.

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• **Dial Rules:** A Dial Rule controls how calls will be dialed on this trunk. It can be used to add or remove prefixes. Numbers that don't match any patterns defined here will be dialed asis.
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• **Trunk Name:** The name you wish to label this trunk.

• **PEER Details:** Modify the default PEER connection parameters for your VoIP provider. You may need to add to the default lines listed below, depending on your provider. WARNING: Order is important as it will be retained. For example, if you use the "allow/deny" directives, make sure deny comes first.

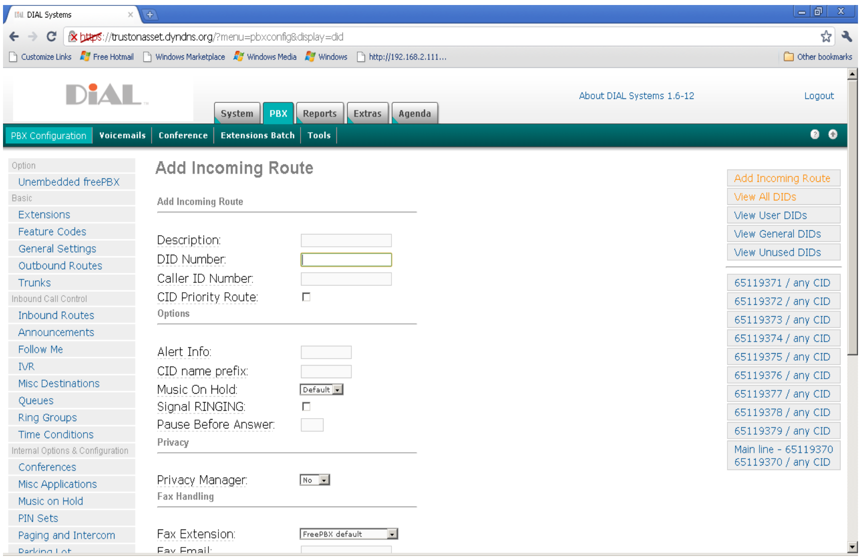
• **USER Context:** This is most often the account name or number your provider expects. This USER Context will be used to define the below user details.

• **USER Details:** Modify the default USER connection parameters for your VoIP provider. You may need to add to the default lines listed below, depending on your provider.

• **Register String:** Most VoIP providers require your system to REGISTER with theirs. Enter the registration line here.
example:
username: password@switch.voipprovider.com. Many providers will require you to provide a DID number, ex: username:password@switch.voipprovider.com/didnumber in order for any DID matching to work.

Inbound Routes

In this menu, you are able to Add/Edit/Delete Inbound Routes to direct the calls where you want them to go such as your IVR, Call Queues, Voicemail, Announcements etc.



Adding a Route

• Description: Provide a meaningful description of what this incoming route is.

- **DID Number:** Define the expected DID Number if your trunk passes DID on incoming calls. Leave this blank to match calls with any or no DID info. You can also use a pattern match (e.g _2[345]X) to match a range of numbers.

- **Caller ID Number:** Define the Caller ID Number to be matched on incoming calls. Leave this field blank to match any or no CID info.

- **Fax Extension:** Select '**system**' to have the system receive and email faxes. The FreePBX default is defined in **General Settings**.

- **Fax Email:** Email address is used if 'system' has been chosen for the fax extension above. Leave this blank to use the FreePBX default in General Settings.

- **Fax Detection Type:** Selecting Zaptel or NVFax will immediately answer the call and play ringing tones to the caller for the number of seconds in Pause below. Use NVFax on SIP or IAX trunks.

- **Pause After Answer:** The number of seconds we should wait after performing an Immediate Answer. The primary purpose of this is to pause and listen for a fax tone before allowing the call to proceed.

- **Privacy Manager:** If no Caller ID is sent, Privacy Manager will ask the caller to enter their 10 digit phone number. The caller is given 3 attempts.

- **Alert Info:** This can be used for distinctive ring with SIP devices.

- **CID Name Prefix:** You can optionally prefix the Caller ID name. e.g: If you prefix with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

- **Music on Hold:** Set the Music on Hold class that will be used for calls that come in on this route. For example, choose a type appropriate for routes coming in from a country which may have announcements in their language.

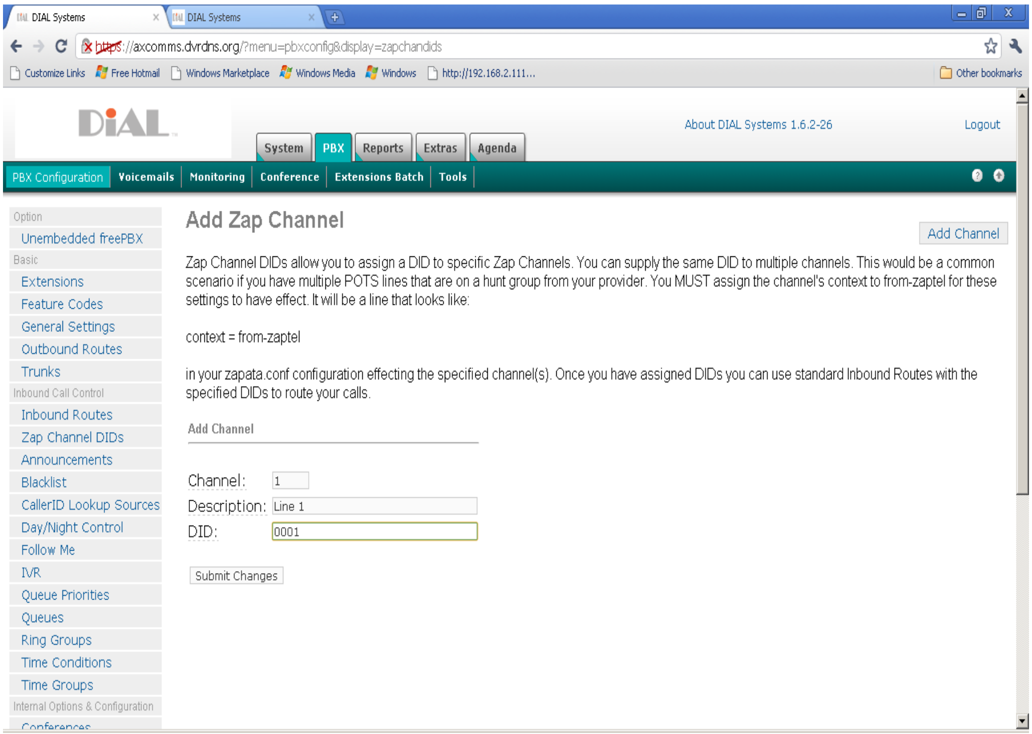
- **Signal Ringing:** Some devices or providers require RINGING to be sent before ANSWER. You'll notice this happening if you can send calls directly to a phone, but if you send it to an IVR, it won't connect the call.

- **CID Lookup Source:** Sources can be added in '**Caller Name Lookup Sources**' section.

- **Set Destination:** Here you may select the destination you wish to route your calls to. This may be your IVR, Conference, Phonebook Directory, Time Conditions, Ring Groups or Queues.

Zap Channels DIDs

Zap Channel DIDs allow you to assign a DID to specific Zap Channels. You can supply the same DID to multiple channels. This would be a common scenario if you have multiple POTS lines that are on a hunt group from your provider.



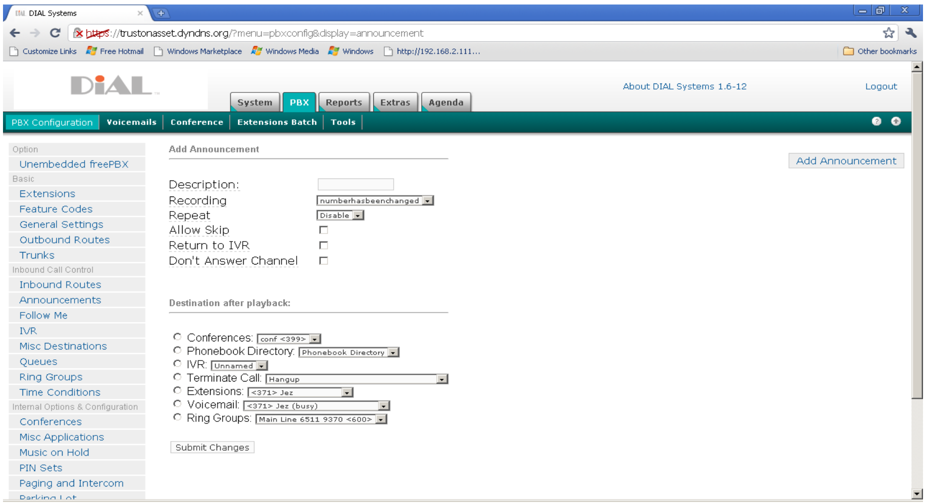
Picture as shown above as an example of creating FXO Channel 1 as line 1. Go to 'Inbound route' to configure DID 001 to the respective destinations such as Ring Group, Extensions, IVR, etc.
Note: Default setting has from Channel 1 '0001' to Channel 8 '0008' has been created.

Announcements

The VX Series is able to record and play announcements to inbound calls. Read on to find out how to add and customize your own announcements to suite your company's needs.

Adding an Announcement

Before you do this, you may want to add a recording to be played back first. For information on how to do this, see '**Recordings**'. When you have done this, just fill out the required details below.



• **Description:** The name of this announcement.

• **Recording:** The message to be played.

• **Repeat:** Key to press that will allow for the message to be replayed. If you choose this option there will be a short delay inserted after the message. If a longer delay is needed it should be incorporated into the recording.

• **Allow Skip:** If the caller is allowed to press a key to skip the message.

• **Return to IVR:** If this announcement came from an IVR and this box is checked, the destination below will be ignored and instead it will return to the calling IVR. Otherwise, the destination below will be taken. Do not check if not using in this mode. The IVR return location will be to the last IVR in the call chain that was called so be careful to only check when needed. For example, if an IVR directs a call to another destination which eventually calls this announcement and this box is checked, it will return to that IVR which may not be the expected behavior.

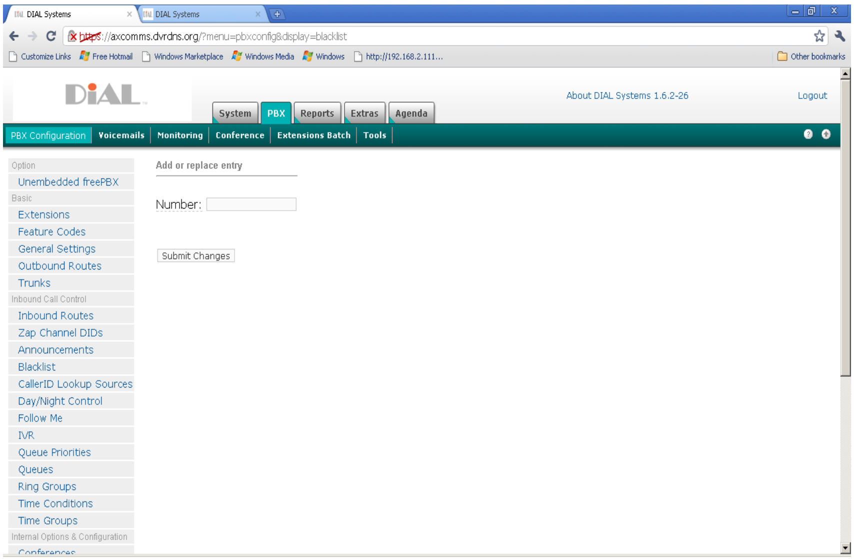
• **Don't Answer Channel:** Check this to keep the channel from explicitly being answered. When checked, the message will be played and if the channel is not already answered it will be delivered as an early media if the channel supports that. When not checked, the channel is answered followed by a 1 second delay. When using an announcement from an IVR or other sources that have already answered the channel, that 1 second delay may not be desired.
When the above details are filled out to your requirements, proceed to select the destination of your choice.

• **Destination After Playback:** Choose the desired destination after announcement played.

Note: You may use Announcement on Ring Group, IVR, Inbound Route, Voicemail or any custom route based on your creativity.

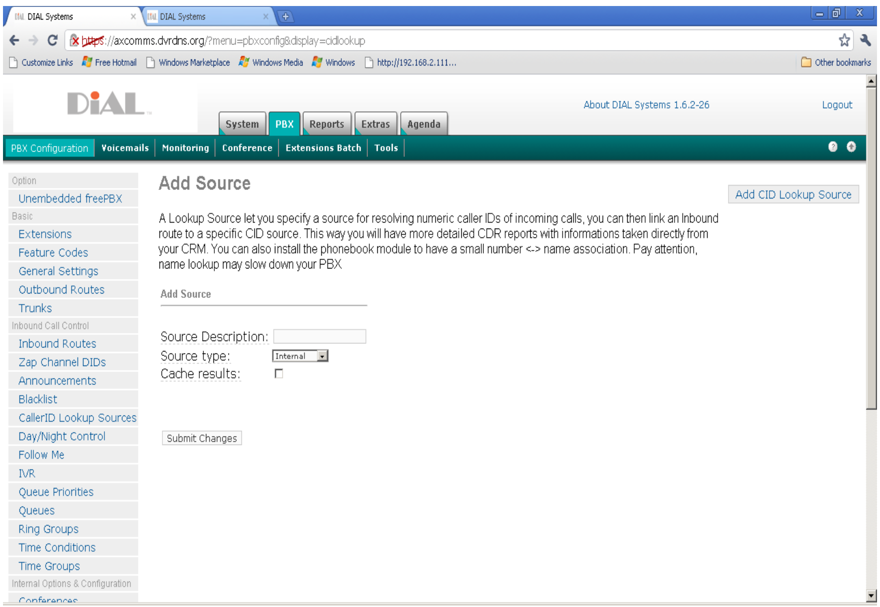
Black List Numbers

To enable certain number to be black listed for outgoing call. Enter the number as shown then click submit. This feature is good for call centre.



Caller ID Lookup Sources

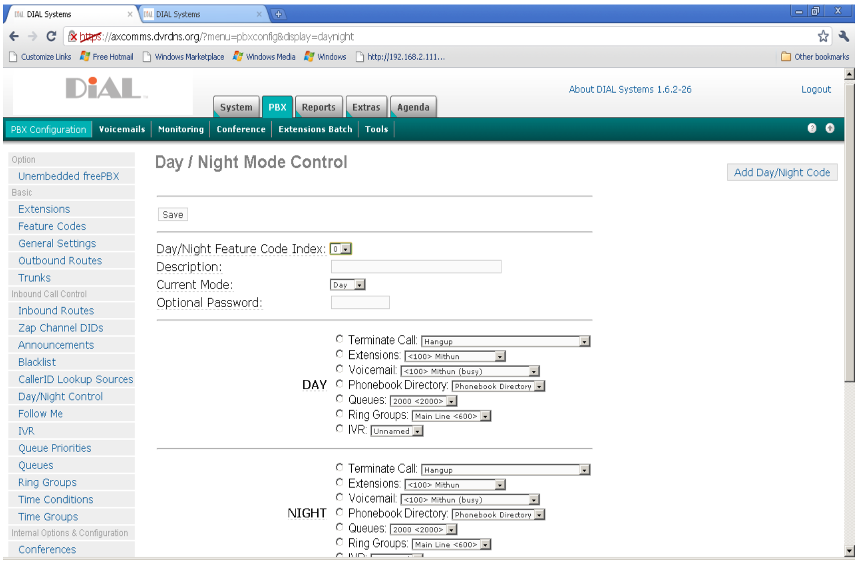
A Lookup Source let you specify a source for resolving numeric caller IDs of incoming calls, you can then link an Inbound route to a specific CID source. This way you will have more detailed CDR reports with informations taken directly from your CRM.



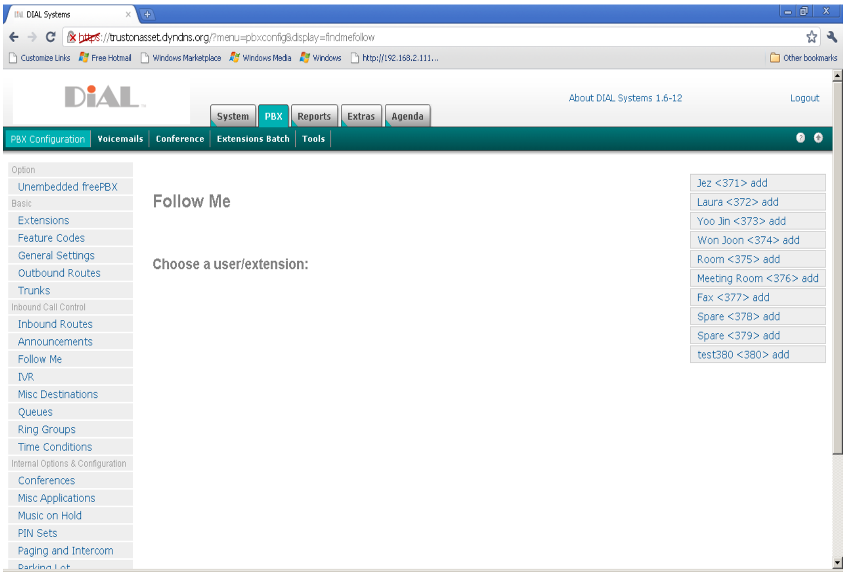
You can also install the phonebook module to have a small number <-> name association. Pay attention, name lookup may slow down your PBX .
Note: Refer to your systems Integrators if you need to know more about this setting.

Day/Night Control Mode

This feature allow you to activate Day or Night mode setting by pressing feature code *280 from your phone. System will playback audible greeting whether it's Day or Night mode.

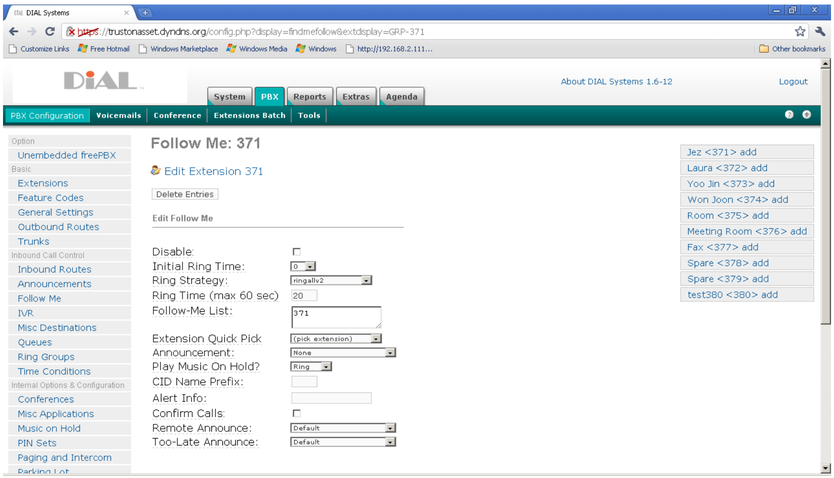


Follow Me



This feature allows your call to find you wherever you may be. You are able to input other extensions/phone numbers so that the calls will be shifted there in the event that the call is not answered. This is how the VX Series reduces the number of dropped calls and ensures that your calls reach you.

To set up your **Follow Me** feature:



• Select the user/extension you wish to customize add a **Follow Me** to.

• Fill out the required details as below.

• **Disable as Default:** This box is not checked by default. Any call to this extension will go to this **Follow Me** instead, including directory calls by name from IVRs. If checked, calls will go only to the extension. However, destinations that specify **Follow Me** will come here. Checking this box is often used in conjunction with the **VmX Locater**.

• **Initial Ring Time:** This is the number of seconds to ring the primary extension prior to proceeding to the **Follow Me** list. The extension can also be included in the **Follow Me** list. A 0 setting will bypass this.

• **Ring Strategy:**
ringallv2: Ring primary extension for initial ring time followed by all additional extensions until one answers.
ringall: Ring all available channels until one answers (default).
hunt: Take turns ringing each available extension.
memoryhunt: Ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list etc.
***prim:**
These modes act as described above. However, if the primary extension (first in list) is occupied, the other extensions will not be rung. If the primary is freepbx DND, it won't be run. If the primary is freepbx CF unconditional, then all will be rung.
firstavailable: Ring only the first available channel.
firstnotonphone: Ring only the first channel which is not off hook.

• **Alert Info:** You can optionally include an **Alert Info** which can create distinctive rings on SIP phones.

• **Confirm Calls:** Enable this if you are calling external numbers that need confirmation. eg, a mobile phone may go to voicemail which will pick up the call. Enabling this requires the remote side to push '1' on their phone before the call is put through. This feature only works with the **ringall/ringallprim ring strategy**.

• **Remote Announce:** Message to be played to the person RECEIVING the call, if '**Confirm Calls**' is enabled. To add additional recordings use the '**System Recordings**' menu.

- **Too Late Announce:** Message to be played to the person RECEIVING the call, if the call has already been accepted before they push '1'. To add additional recordings use the **'System Recordings**' menu.

- **Follow Me List:** List extensions to ring, one per line. You can include an extension on a remote system, or an external number by suffixing a number with a pound (#). ex: 2448089# would dial 2448089 on the appropriate trunk (see Outbound Routing).

- **CID Prefix:** You can optionally prefix the Caller ID name when ringing extensions in this group. For example, if you prefix with "Sales", a call from John Doe would display as "Sales:John Doe" on the extensions that ring.

- **Ring Time:** The number of seconds to ring this extension before transferring it to the **Follow Me** list.

- **Announcement:** Announcement message to be played to the caller before dialing this group. To add additional recordings use the **'System Recordings**' menu.

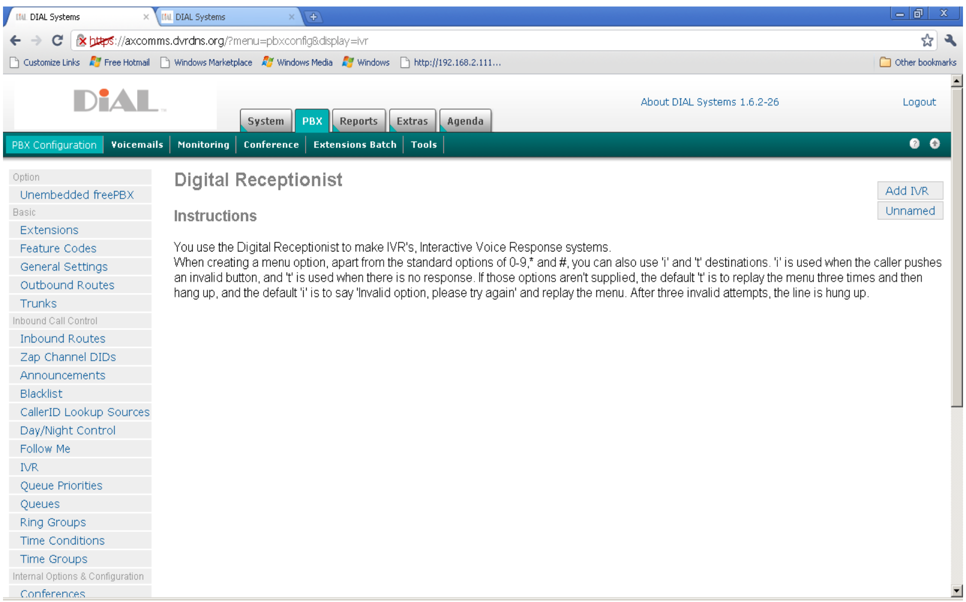
- **Play Music on Hold:** If you select a **Music on Hold** class to play, instead of the default ringing tone, they will hear the selected ringing class while waiting for the call to be answered. shows the destinations you can select for the calls to be routed to should there be no answer on the call(s).

IVR

The DIAL Systems VX Series IP PBX allows you to create your own **IVR**(Interactive Voice Response) menu without hassle. Here you will learn how.

Adding an IVR

Click on the '**Add IVR**' option on the right side of the screen and you will be presented with the '**Digital Receptionist**' menu.



- **Change Name:** This changes the short name of this IVR. The default name is 'Unnamed'.

- **Timeout:** The amount of time (in seconds) before the 't' option, if specified, is used.

- **Enable Directory:** Let callers into the IVR dial '#' to access the directory.

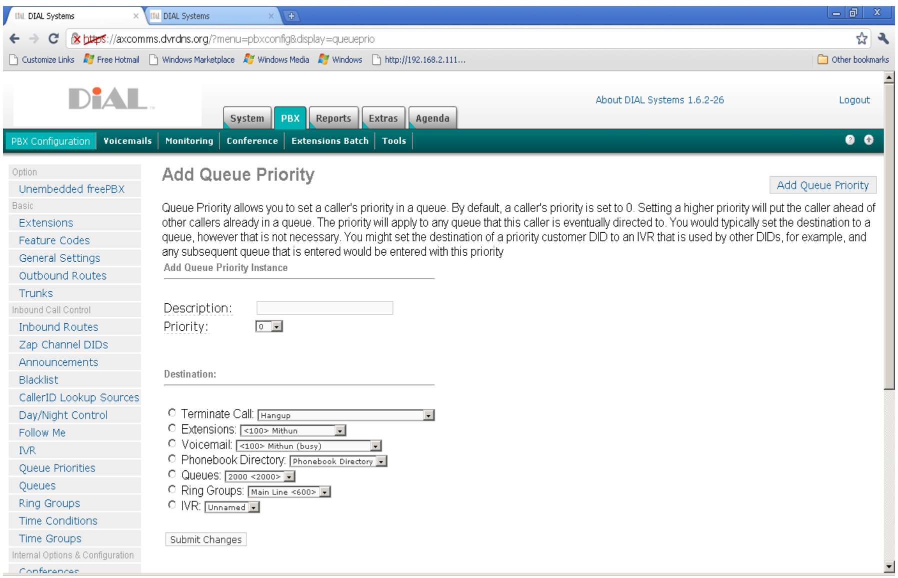
- **Directory Context:** When '#' is selected, this is the voicemail directory context that is used default.

- **Enable Direct Dial:** Let callers into the IVR dial an extension directly.

- **Announcement:** Announcement message to be played to the caller before dialing this group. To add additional recordings use the 'System Recordings' menu. To create announcement, refer 'Announcement' setting.

Queue Priorities

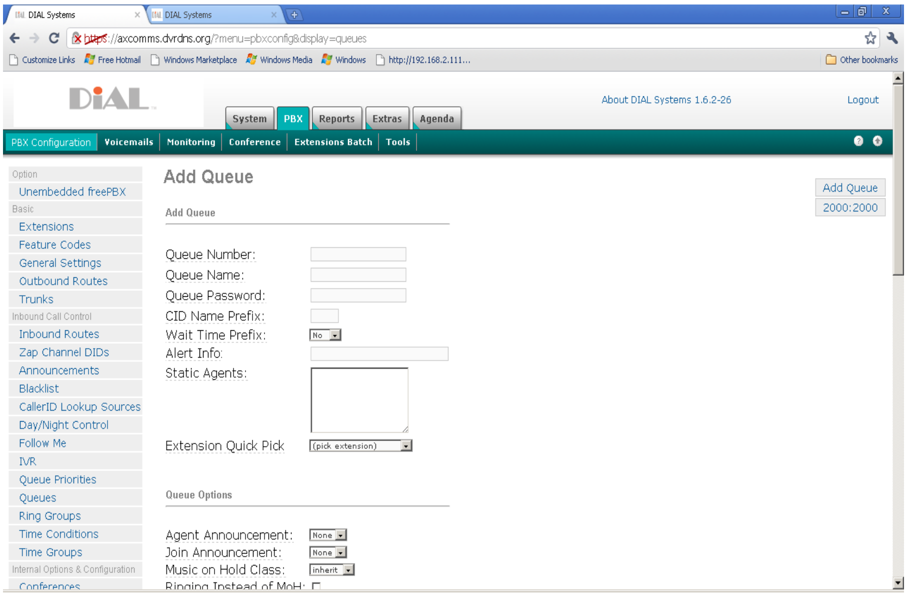
Queue Priority allows you to set a caller's priority in a queue. By default, a caller's priority is set to 0. Setting a higher priority will put the caller ahead of other callers already in a queue. The priority will apply to any queue that this caller is eventually directed to. You would typically set the destination to a queue, however that is not necessary. You might set the destination of a priority customer DID to an IVR that is used by other DIDs, for example, and any subsequent queue that is entered would be entered with this priority



Queues

Call Queues allow you to manage a large number of incoming calls, as you would expect to have in a Call Center. This is very intelligent application, and as such, it has a lot of configuration options.

For Call Centre set up, create new Queues with Static agents assign to it.
Example if Extension 100, the Static Agents Box will be A100. One Extension per line.
Configure Inbound Route with DID Number to allow calls to reach this Queues. It can also be configured with IVR, Ring Group or Time Condition.



• **Queue Number:** This is the number that can be dialed from any extension to be put into the queue. This is also the same number you use when selecting a destination. Agents will dial this queue number plus '*' to log onto the queue, and this queue number plus '**' to log out of the queue.
For example, if the queue number is 123:
123* = log in
123** = log out queue name

• **Queue Name:** A short name for the queue. This is only used in the web interface for ease of identification.

• **Queue Password:** If you are concerned about security, you can put a password on the queue to stop just anyone from logging into it. After an agent tries to log in, he or she will be prompted for the password here. This is optional.

• **Prefix:** As an agent may be logged into more than one queue, it can be useful to have a prefix on the Caller ID seen on the agent's phone, so he or she knows which queue the call is coming from eg, 'Sales:' or 'Tech:

- **Static Agents:** These are devices that are always logged into the queue. This is useful if you have an agent that is not directly connected to the VX Series, but is telecommuting. You can put their number(s) preset with a '#' to signify that it should be treated as an external device. The number will be routed as if it was dialed from a normal extension, so dial rules in **Outbound Routing** and trunks are matched as per normal.

- **Agent Announcement:** This is the announcement that is played to the agent prior to connecting to the caller.

- **Music on Hold Class:** Music (or Commercial) played to the caller while they wait in line for an available agent. Choose "inherit" if you want the Music on Hold Class to be what is currently selected, such as by the inbound route.

- **Ringing Tone instead of Music on Hold:** Enabling this option make callers hear a ringing tone instead of Music on Hold. If this option is enabled, settings of the previous drop down are ignored.

- **Max Wait Time:** The maximum number of seconds a caller can wait in a queue before being pulled out.

- **Max Callers:** Maximum number of people allowed in the Queue. Set as '0' for unlimited.

- **Join Empty:** If you wish to allow callers to join queues that currently have no agents. We recommended 'yes'.

- **Leave When Empty:** If you wish to remove callers from the queue if there are no agents present, set this to 'yes'.

- **Ring Strategy:**

roundrobin: Take turns ringing each available agent.
leastrecent: Ring agent which was least recently called by this queue.
fewestcalls: Ring the agent with fewest completed calls from this queue.
random: Ring random agent.
rrmemory: Round robin with memory, remember where we left off last ring pass.

- **Agent Timeout:** The number of seconds an agent's phone can ring before we consider it a timeout. Unlimited or other timeout values may still be limited by system ring time or individual extension defaults.

- **Retry:** The number of seconds the system will wait before trying all the phones again.

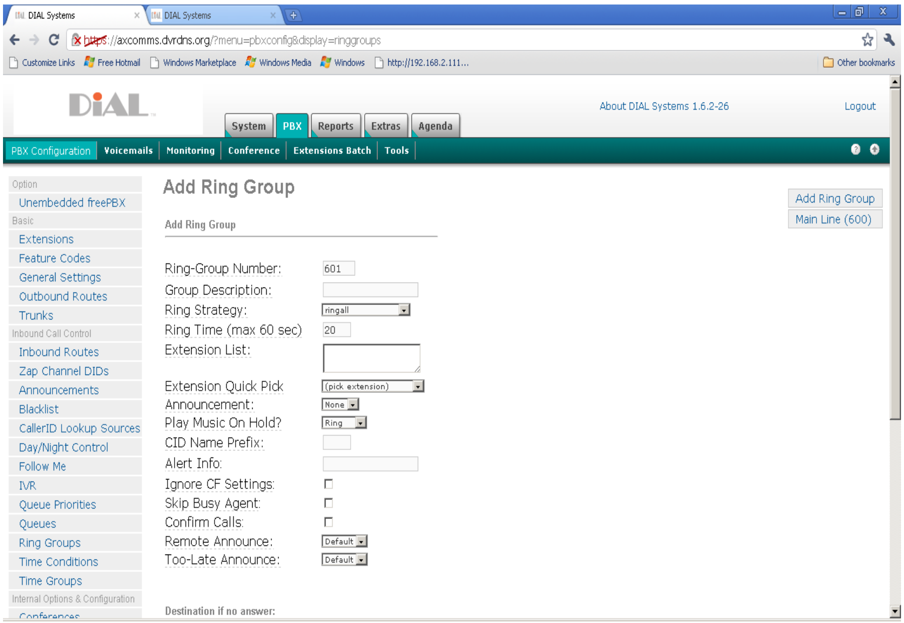
- **Wrapup Time:** After a successful call, how many seconds the system waits before sending a potentially free agent another call (default is 0, or no delay).

Ring Groups

This defines a 'virtual' extension that rings a group of phones simultaneously, stopping when any one of them is picked up.

To set up your Ring Groups, select 'Add Ring Group' then fill out the fields below.

Give a unique name follow by Extension Number in the list. One line per extensions. If Extensions are set with Follow me, insert a # after the digits. Eg: 100#
This is the most commonly used setting for basic routing from Inbound Route.



• **RingGroup Number:** This is the number that is dialled from any extension that will make all of the phones in the group ring.

• **Group Description:** Provide a descriptive title for the Ring Group.

• **Ring Strategy:**
ringall: ring all available channels until one answers (this is the default)
hunt: take turns ringing each available extension
memoryhunt: ring first extension in the list, then ring the 1st and 2nd extension, then ring 1st 2nd and 3rd extension in the list.... etc.

• **Extension List:** List the extensions to ring, one per line. You can include an extension on a remote system, or an external number by suffixing a number with a hash (#). ex: 2448089# would dial 2448089 on the appropriate trunk (see **Outbound Routes**).

• **CID Name Prefix:** You can optionally prefix the Caller ID name when ringing extensions in this group. For example, if you prefix it with "Sales:", a call from John Doe would display as "Sales:John Doe" on the extensions that ring. (Note, you can't use a space here.)

• **Ring Time (max 60sec):** How long the group of phones will ring before deemed as '**failing**' and doing the options specified below. This is not related to the '**hunt**' ring strategy above, but is the total length of time a call will stay in the group before using the '**Destination if no answer**' selection.

- **Announcement:** Announcement message to be played to the caller before dialing this group. To add additional recordings use the 'System Recordings' menu.

- **Play Music On Hold:** If you select a **Music on Hold** class to play, instead of the default ringing tone, they will hear the selected ringing class while waiting for the call to be answered.

- **Alert Info:** You can optionally include an **Alert Info** which can create distinctive rings on SIP phones.

- **Confirm Calls:** Enable this if you're calling external numbers that need confirmation. For example, a mobile phone may go to voicemail. Enabling this requires the remote side push '**1**' on their phone before the call is put through. This feature only works with the **ringall** ring strategy.

- **Remote Announce:** Message to be played to the person receiving the call, if '**Confirm Calls**' is enabled.

- **TooLate Announce:** Message to be played to the person receiving the call if the call has already been accepted before they push '1'.

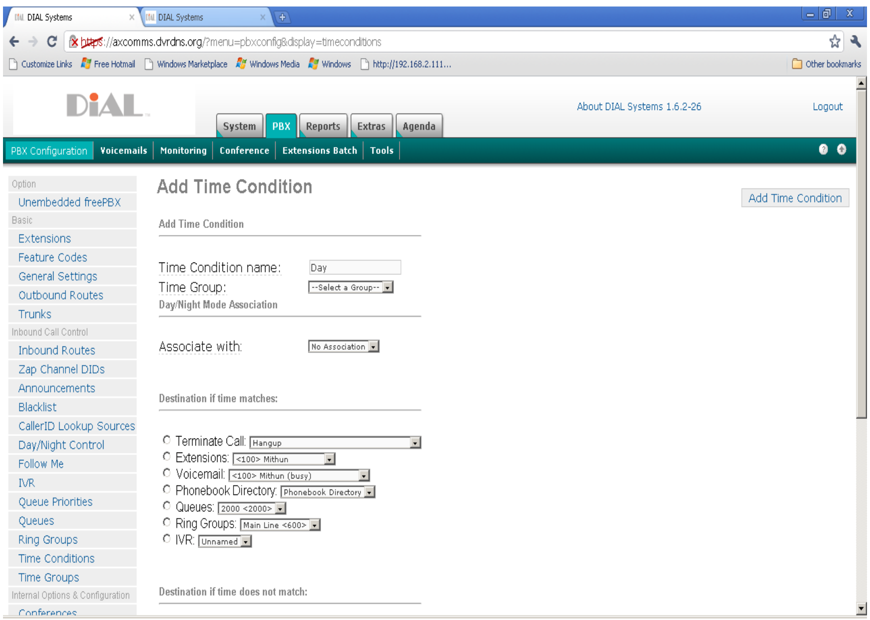
When the above details has been configured to your liking, proceed to select a destination for the call to be routed should there be no answer on the ring groups.

Time Conditions and Groups

Time Conditions are a module that appears as a destination when installed. It allows you to do an 'if' based on the current Time, Weekday, Day of the Month, or Month.

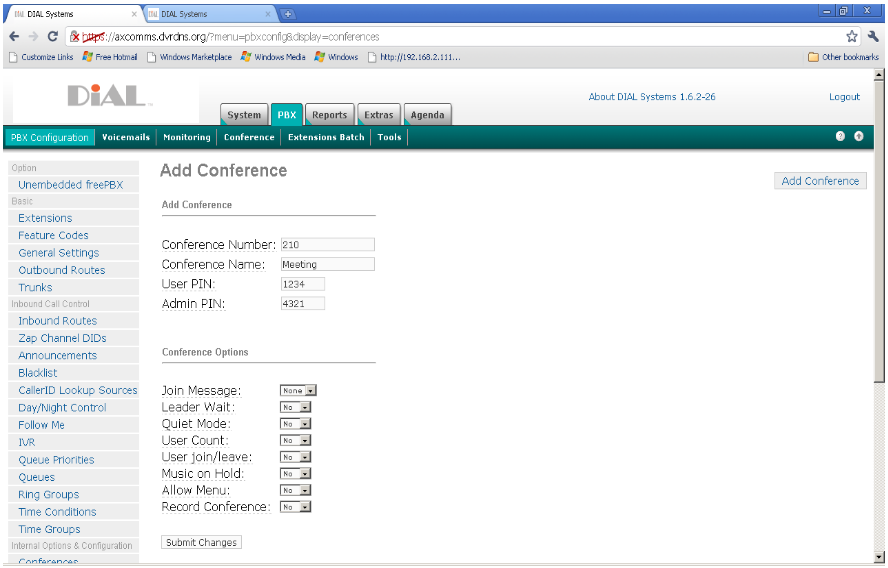
'Time Group' must be pre configured before this.

After filling out the **Time Condition** details, choose a destination when the time matches and when it does not. When this is done, click the '**Submit Changes**' button.



Conferences

Conferences is a standard multiparty conferencing facility that is available as a destination.
You are able to set up a conference room in this menu.



- **Conference Number:** This is a number that local users can dial to join the conference.

- **Conference Name:** This is used as an Identifier, along with the number, when picking a conference as a destination.

- **User/Admin PIN:** If either of these options are set, anyone calling into the conference will be prompted for a PIN. If '**User**' is left blank, they can just push '**#**' to enter. The only use of '**Admin**' is to not actually open the conference until the admin user has arrived. If '**Music On Hold**' is enabled, users will be placed on hold with the default **Music On Hold** class.

- **Join Message:** This is a sound that is played to all users upon entering.

- **Leader Wait:** When there is an **Admin PIN** set, the conference will not start until the '**Admin**' user joins.

- **Quiet Mode:** Usually a beep is played when a user enter or leaves the conference, alerting other members to the fact that someone has joined or left. You can disable that by selecting '**Yes**' here.

- **User Count:** When someone joins, a message will be played to inform the users how many callers are in the conference.

- **User Join/Leave:** When someone joins the conference, the system will prompt them to record their name. The conference will then announce their name when they join and leave.

- **Music On Hold:** Totally enables or disables **Music on Hold** in this conference.

- **Allow Menu:** Enables the **User** or **Admin** to enter management mode by pushing '*****'.
The commands whilst in management mode are:

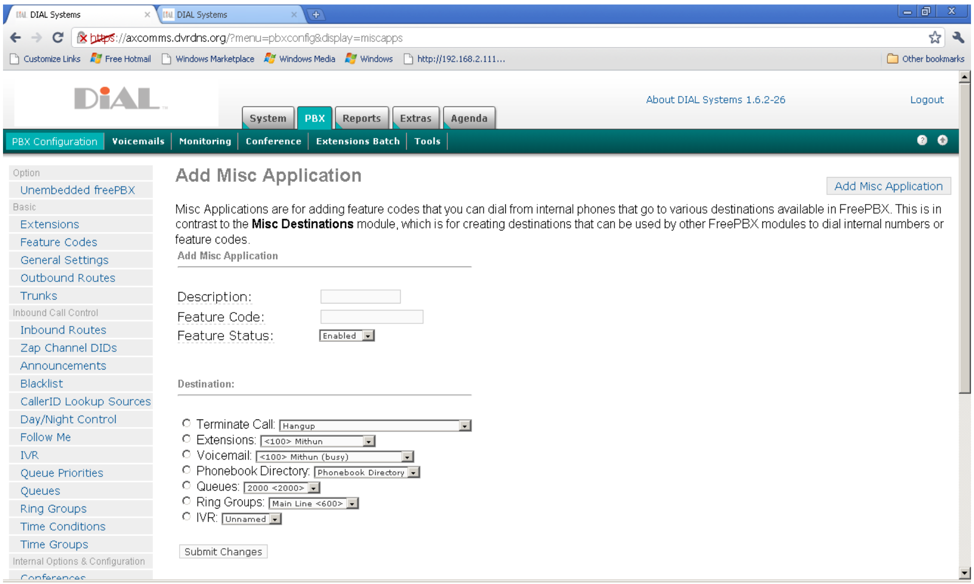
1: Mute yourself 4 or 6: Decrease or Increase the Conference Volume (eg, the sound you hear) 7 or 9: Decrease or Increase your Volume (eg, the sound other people hear) Additionally, **Admin** users have the added features of:

2: Locking or Unlocking the conference

3: Ejecting the last person that called

Miscellaneous Applications

Misc Applications are for adding feature codes which enable you to dial from internal phones to go to various destinations available. This is in contrast to the **Misc Destinations** module, which is for creating destinations that can be used by other FreePBX modules to dial internal numbers or feature codes.



• **Description:** The name of the application you wish to add.

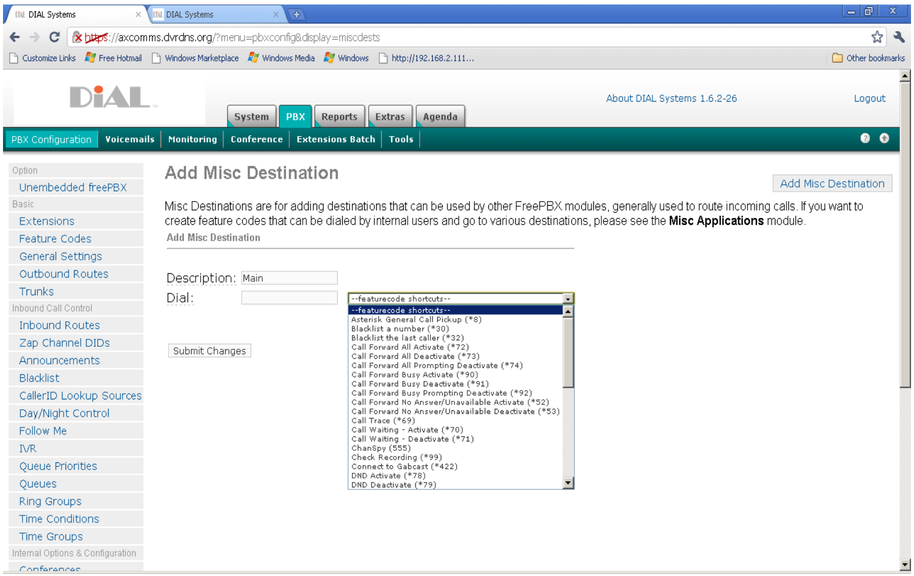
• **Feature Code:** The feature code/extension users can dial to access this application. This can also be modified on the **Feature Codes** page.

• **Feature Status:** Select if you wish for this code to be enabled or disabled.

When the above details have been filled out, select a destination for which you want the call to reach then click the '**Submit Changes'** button.

Miscellaneous Destinations

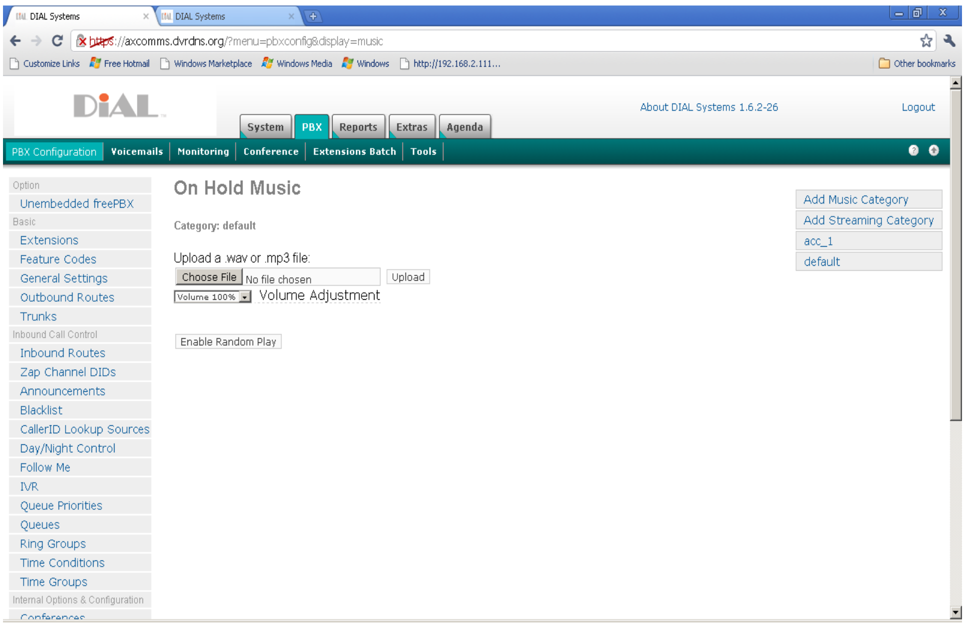
Misc Destinations are for adding destinations that can be used by other FreePBX modules, generally used to route incoming calls. If you want to create feature codes that can be dialed by internal users and go to various destinations, please see the **Misc Applications** module.
Should you choose to add a **Misc Destination**, below the same screen are fields that will require you to fill in.



• **Description:** Give this **Misc Destination** a brief name to help you identify it.

• **Dial:** Enter the number this destination will simulate dialing, exactly as you would dial it from an internal phone. When you route a call to this destination, it will be as if the caller dialed this number from an internal phone.

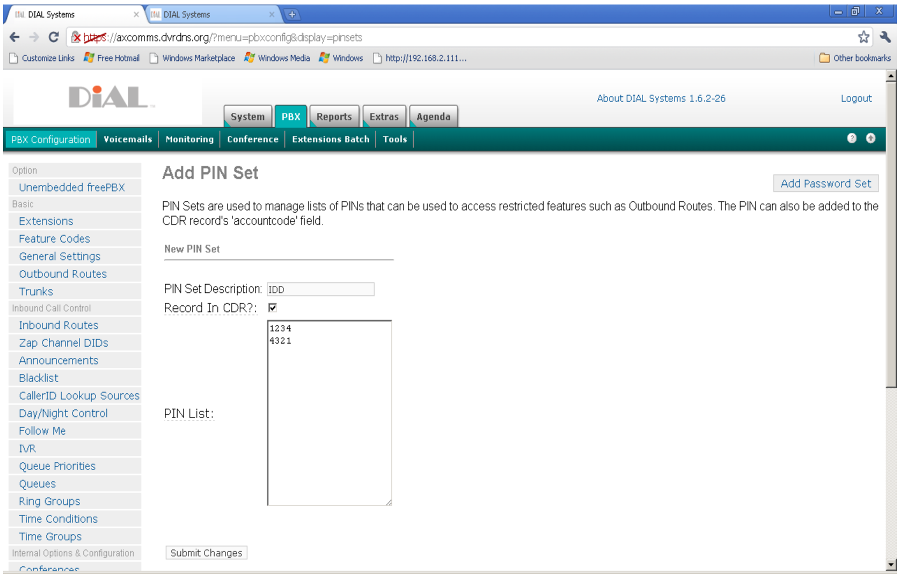
Music On Hold



Here you can configure the **Music On Hold** files that will be played. You can configure various 'Classes' of **Music on Hold** which are used in Queues. You may choose to have the default customized music play in your different queues while your callers are on hold.
Simply select '**browse**' and pick an MP3 file on your system. Then click '**Upload**'. It will appear in the list of MOH files below as seen.

PIN Sets

PIN Sets are a module that allow you to use a range of PINs rather than just one. This is currently only used by Trunks, but it potentially could be used in DISA or anything else that uses PINs for authentication.

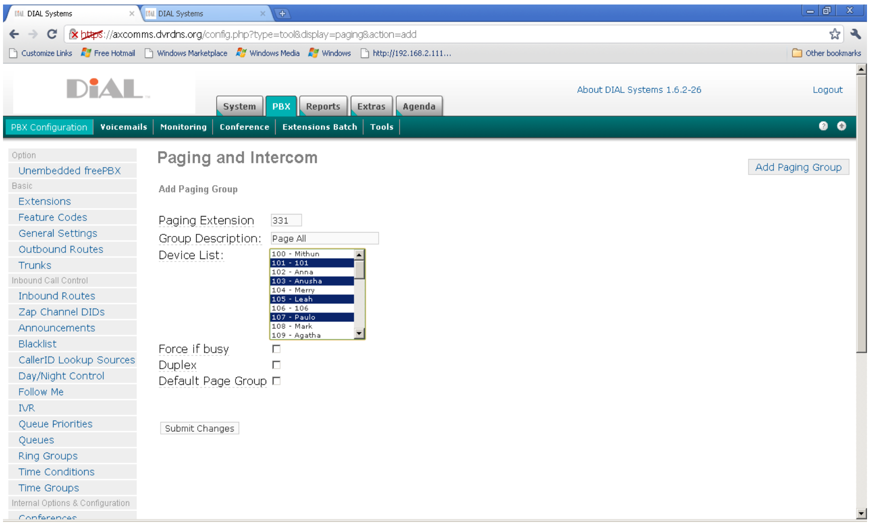


- **PIN Set Description:** Give this **PIN Set** a name.
- **Record in CDR:** Select whether or not to record this **PIN Set** in the call detailed records(**CDR**).
- **PIN List:** Enter a list of one or more PINs. Enter the PINs one PIN per line.

Paging & Intercom

With phones that support '**Paging**', you are able to perform a page simply by dialing a number. All the phones in the group pick up automatically and go into speaker mode. They will then play through their speaker what the caller is saying. This is very useful in a small office environment ("Pizza is at reception!").

Simply click on 'Add Paging Group' and follow enter the paging group number – this is the number that users will dial to perform a page. Proceed to add the relative extensions into the paging group as seen.

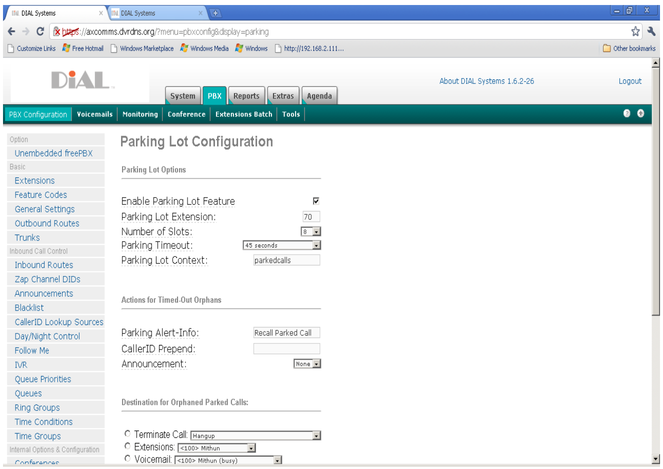


Click Submit follow by Apply apply changes to takes effect.

Do take note that this feature may not applicable to IP Phones other than with Cisco SPA500 series and more.

Parking Lots

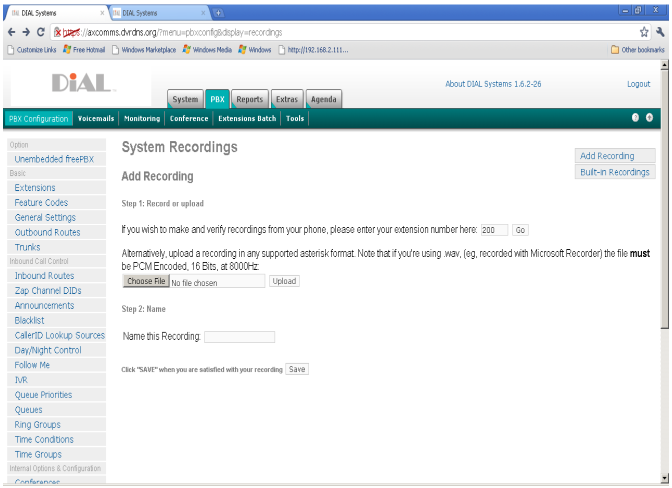
Enable this feature for Calls to be Park with feature code '70'.
Transfer the call to '70', which allows system to response with a Lot Number, eg: 71 or 72.
After that, you may go to any IP Phone by dialling the given Parking Lot Number to retrieve the calls.
Refer to User Manual for more usage guide.



Click Submit follow by Apply apply changes to takes effect.

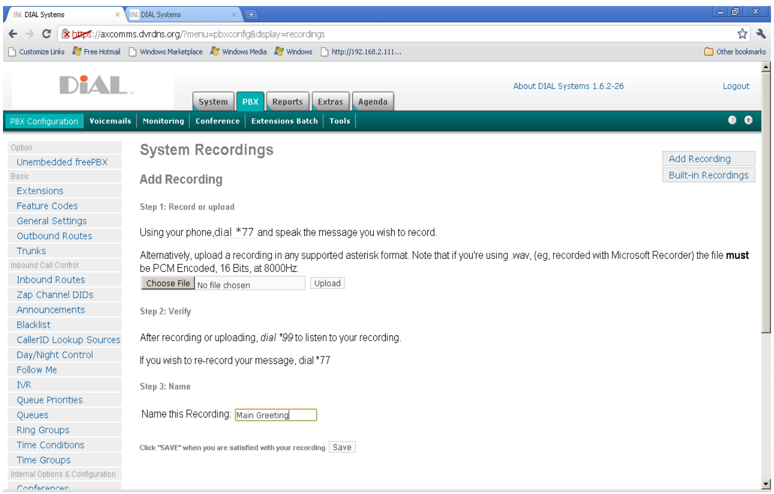
System Recordings

This menu allows you to upload or record new recordings for your announcement or IVR.
To record:



- Enter the extension number of the phone you wish to record your announcement from.
- Click on '**go**' when done.
- Dial '***77**' and record your message.
- Hit the '**#**' key when done.
- Dial '***99**' to play back your recording.

Alternatively, you may choose to upload a previously recorded file. This however, needs to be in an asterisk supported format and be PCM Encoded, 16 Bits, at 8000Hz.



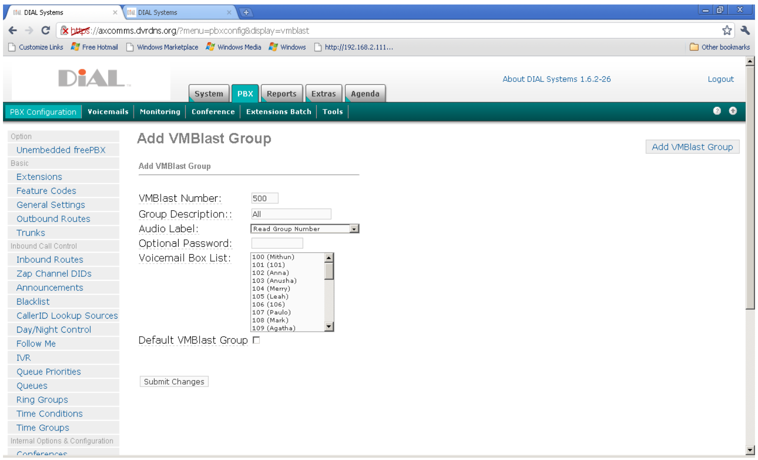
To do this:

- Click on the '**Browse**' button to search for the file on your computer.
- Select the file you wish to upload then click the '**Upload**' button.

When either of these steps are done, name your recording then click the '**Save**' button when you are finished.

Voice Mail Blasting

To send a group voice mail message to selected extensions.



Create a voice mail number with a unique name and selected extensions.

Callback

A callback will hang up on the caller and then call them back, directing them to the selected destination. This is useful for reducing mobile phone charges if your incoming calls are free on your mobile plan.

First, click on the '**Add Callback**' button on the right. Then proceed to fill out the relative fields.

- **Callback Description:** Enter a description for this callback.

- **Callback Number:** Enter the number to dial for the callback. Leave this blank to just dial the incoming Caller ID Number. This field is optional.

- **Delay Before Callback:** Enter the number of seconds the system should wait before calling back.

When this is done, select a destination after the callback.

Tip: Select DISA if you just wish for the system to provide you a dial tone to dial out again if you are performing a callback from your mobile.

DISA

DISA (Direct Inward System Access) allows you to provide an internal dial tone to external callers. When you configure a **DISA** destination, you can use it as a menu destination within a **Digital Receptionist**, so that you can get an internal Asterisk dial tone. This means you could call into your Asterisk system and dial out as if you were using an extension connected to the Asterisk box itself.

- **DISA Name:** A name to help you identify the DISA destination.
- **PIN:** A code required to be input by the remote user to access the dial tone. You should always have a PIN for security purposes. You can, if you must, leave it blank. The voice prompt will still ask you for a password, but you won't need to enter anything.

- **Response Timeout:** The maximum amount of time it will wait before hanging up if the user has dialed an incomplete or invalid extension.

- **Digit Timeout:** The maximum time delay between digits as they are input.

- **Require Confirmation:** Before prompting for a password, the system will say 'Press 1' every 3 seconds until a digit is pressed. It will then proceed to ask for a password. This is mainly useful if dialing out through a device (eg, X100P, TDM400, or a poorly configured VSP) that does not indicate when the call is actually answered.

- **Caller ID:** You can change the outgoing caller ID for this DISA by setting an override value.

- **Context:** By default, internal context (from-internal) is used. You could provide a custom context to limit the access for this DISA, but only if you are fully aware of what you are doing.

IMPORTANT: This function will impose a security threat if not properly implemented. Refer to your Systems Integrators before engaging this function.

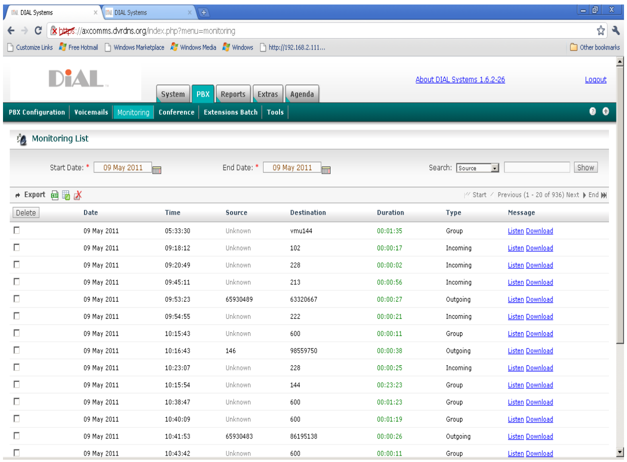
Voice Mail

The Voice Mail List function allows you to view and listen to all voicemails that has come in on the VX Series. You are able to filter the results according to date. You may also delete them in this menu.

Monitoring

With this feature, you are able to monitor users on your DIAL Systems VX Series. The system is able to perform **OnDemand** call recordings as well as perpetual recordings for individual extensions. In this menu, you are able to view all your recordings and filter them according to date.

See Below picture as shown on next page.



Extension setting to turn on the Recording Incoming or outgoing as 'On Demand' or Always.

To turn on this feature, Voice Logger Module must be purchased before system can perform this. Voice Logger modules includes logger application, Storage Hard Disk and Memory Increase. Refer to your systems integrator for more information.

Note: Also refer to User Guide to information on how to filter recorded calls.

Extension Batch

Create a list of extensions by using CSV Format. Download the CSV file from page link. When extensions list created, upload it to the system. This is the fastest way to create many extensions at once.

Tools

For more advanced users, you are able to enter Asterisk CLI (command line interface) in this screen. Simply type your command in the field provided and hit the '**Execute**' button.
If you click on the '**File Editor**' button on the left hand side, you will enter a different menu.

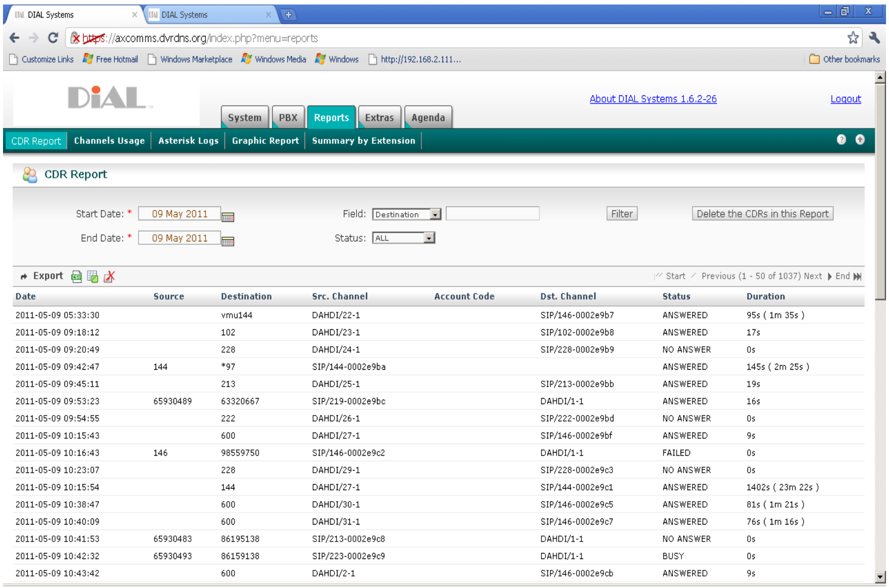
Here you are able to edit the already existing .conf (configuration files) files or create new

Alternatively, you can create your own configuration files. Click on the '**New File**' button to create a new configuration file.

3. Reports

CDR Report

The DIAL Systems VX Series IP PBX is able to provide you with a **CDR**(call detail reporting) of all your incoming and outgoing calls. You are able to filter the search results according to source, destination, PIN sets, channel, dates and status of calls.



Channels Usage

In this screen, you are able to monitor your channel usage for your phone calls be it **SIP**, **ZAP**, **IAX**, **H323** etc.

Asterisk Logs

In this screen, you are able to monitor your system major logs. This is where you can see system errors.

Graphic Report

A Graphical Report for calls being made per extension with Date Filtering.

Summary Of Extensions

A general report of how many In/Outgoing calls of being made per extensions with total durations and Date Filtering.

4. Extras

Text to wave

To create text to wave file, type the text you would like the system to automate an audible speech for your announcements for any inbound calls related purposes.

Click 'Generate Audio File', you will be prompted to download the file.

You may go to 'System Recording' to upload the file as such.

Note: This is a high privilege features and we offer it for free. As such we do not support high quality audio speech.

Address Book

The VX Series IP PBX comes with a built-in address book. You are able to add contacts and perform **clicktodial** functions. The extensions which you have configured in your system, will automatically be added in your contact database. You are able to filter and search for contacts in accordance to their names, phone number, internal and external numbers When adding a new contact, simply fill in the fields provided.

- **Name:** Your contact's first name.
- **Last Name:** Your contact's last name.
- **Phone Number:** Your contact's phone number.
- **Email:** Your contact's email address.

Note: This feature will work in hand with User Login with all contacts as created. Take note, this is a shared contact as general.

6. Conclusion

We have come to the end of the DIAL Systems VX-100/200 Administrator Manual.
DC Apex integrators wishes you have an amazing experience with the DIAL Systems IP PBX and an enjoyable journey with IP Communication. Should you face any difficulties that this manual could not assist you with, you may contact DC Apex Integrators (S) Pte Ltd for additional technical support.

7. Contact Us

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