Vdex-40

Installation & Administration

Version 1.3f

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Fraudulent Usage Advisory

Although the Vdex-40 system (Vdex) is designed to resist fraudulent usage, including unauthorized access to a long distance network, no product, including Vdex is able to prevent such unauthorized usage. Vdex is likewise unable to prevent such uses as may constitute an invasion of privacy or other tort.

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- ? images depicted in this and related manuals and associated documentation including promotional material may vary from the actual product; and
- ? textual explanations may deviate from that expected by or presented by the product.

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Requirements

Vdex user interface operates under Firefox 2.0.0.11 or higher and Explorer 7.0 or higher. Telcom connection and active account is required for Telco services. Telcom CallerID and active account is required for this service to function in countries supported by this product Internet connection and active account is required for VoIP and other Internet services such as firmware update

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Safety and Compliance

⚠ **Shock Hazard**

Do not use this device and any connected devices such as telephones during periods of thunder and lightening activi ty or during high probability periods such as storms.

Do not place and or operate this device near heat sources and or in direct sunlight.

The mains power supply switch to which this device is connected must be accessible at all times as it is the primary method of disconnecting the power supply to this device.

⚠ **For Indoor Use Only**

Do not operate outside and or near sources of water or other fluids.

⚠ **Compliance**

Altering Country Specific factory defaults from those defined for the country in which the product is in use will negate and relevant compliaces and cuase the product to malfunction.

Important Note About Emergency Usage

IMPORTANT NOTICE RE POWER FAILURE & EMERGENCY USE

⚠ Under power failure conditions this device will NOT OPERATE. Please ensure a separate telephone device not dependant upon local power is available for emergency use.

⚠ The device will NOT FUNCTION for Internet calls when disruption to the Internet and connection to it occurs.

⚠ Many Internet phone service providers do not support emergency service calls such as 911, 000 and 111. Furthermore calls to these services will not function when default configuration of this device is used and or if the device is not configured to specifically support these services.

Chapter 1: Overview

Vdex is a hybrid phone system which interoperates with the telephone and Internet networks to utilize the best of both. Vdex performs as a PBX, voice messaging system and voice over internet (VoIP) gateway, providing access to low-cost calls, call routing over the internet to remote phones and office connected to the Internet virtually anywhere in the world. Vdex operates using a modified, hardened version of Asterisk to run on its multiprocessor platform.

Chapter 2: Physical Connectivity

Remove the Vdex system, power supply, and network cable from the Vdex box. Connect the network cable to the LAN port on Vdex system and to an available port on your network router, switch, or hub. Connect up to four analog telephone lines to FXO ports 1-4 as shown in the diagram below. Power the Vdex system on. Allow up to 5 minutes for the system to startup.

Please Note: You must connect at least one telephone line to FXO port 1 to dial 911 for emergencies by default. This can be changed later as needed (see outbound dialing rules) for details.

Example Network Configurations:

The following example shows two common network configurations. The first shows the Vdex being connected directly to Cable/DSL router. The second shows the Vdex being connected to a hub or switch.
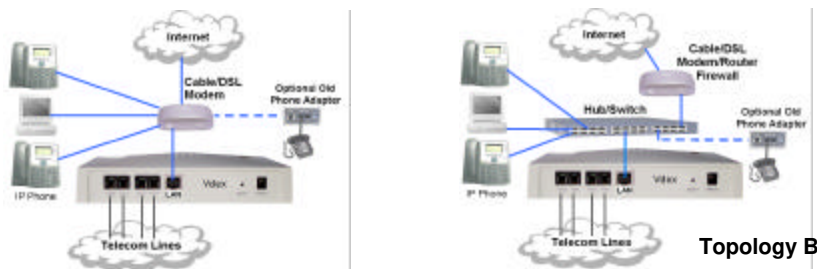
Chapter 3: Assigning a Static IP Address to Vdex

The Vdex40 requires a static IP address. SIP phones can be configured to use DHCP (if you have a DHCP server on your network) or can also be configured to use static IP addresses. By default the Vdex is assigned static IP address **192.168.0.157**. In the event that your network configuration does not use a 192.168.0.x configuration you'll need to change the IP address to support your network configuration.

The easiest way to change the IP address of Vdex is to connect a telephone line to FXO port 1 and call the port. The system will answer after a couple of rings. Once the port answers you'll hear a message stating "**Welcome to your new Vdex Internet telephony system**". When you hear this message, press **654321** to check or change the IP address and network mask of the Vdex system. The Vdex system will speak the current IP address and network mask. When prompted, press 1 to change the IP address and network mask. When entering the new IP address and network mask use the * key for dot and press # when finished.

Please Note: You must enter the new IP address and network mask after hearing each prompt, not during.

If you don't have access to a telephone line at this time, the second way to change the IP address of the Vdex is to run the Vdex wizard. To run the wizard you must have a PC configured for an IP address ranging from 192.168.0.x and 192.168.0.254. You can temporarily reconfigure one of your PC's for an IP address in this range so that you can connect to Vdex via web browser and run the wizard. Using the Vdex wizard you can change the IP address. See chapter 4 on how to run the Vdex wizard.

Chapter 4: Running the Vdex Quick Start Wizard

To run the Vdex Quick Start Wizard, open a web browser and set the web address to the IP address assigned to Vdex. For example, if you've assigned IP address 192.168.1.99, you would enter http://192.168.1.99/wizard into your web browser and press enter to start the wizard. If attempting to connect to the Vdex's default IP address you must make sure your network configuration uses a 192.168.0.x IP scheme and then you would point your browser to http://192.168.0.157/wizard to run the Vdex Wizard.

Using the Vdex Wizard you can change the IP address, network mask, assign the gateway IP address, DNS settings, SMTP server, and email from account to be used by the Vdex system (when sending voicemail to email). You can also change the administrator password. Figures 1, 2, 3 and 4 show the Vdex Wizard screens and what can be done on each screen in more detail.

Step 1: Network Configuration

On the first step (figure 1) set the IP address, subnet mask and gateway IP to be used by the Vdex system. Write them down for future reference and click continue to go onto the next step.

Vdex IP Address   _____

Vdex Network Mask   _____

Gateway   _____

Figure 1

Step 2: DNS Configuration

Set the Hostname, and assign the Primary and Secondary DNS settings to be used by the Vdex system (figure 2). If you're unsure of what to set as the hostname enter **Vdex** for now. If a domain name will be assigned to the Vdex box, enter the domain name as the Hostname. For reference, write down the assigned values. Once defined, click next to move onto step 3.

Hostname   _____

Primary DNS   _____

Secondary DNS   _____

Figure 2

Step 3: Email Configuration

Set the SMTP server value to the name of the SMTP server (or IP address of the SMTP server) to be used to send voicemail to email (see figure 3). Set the Vdex Email value to the name of the email account to be used as the "email from" address when users receive email from the Vdex system. Set the SMTP Username and SMTP Password to the name of the email account and password to be used for authentication. If authentication is not required leave SMTP Username & Password blank. For reference, write down the assigned values and then click continue to move on to step 4.

SMTP Server   _____

Vdex Email   _____

SMTP Username _____

SMTP Password _____

Figure 3

Step 4: Localisation Settings

Select the country where the Vdex40 will be installed. Possible values are United States, Australia, New Zealand, and Singapore.

Step 5: Administrator Password

In this final step you'll be prompted to enter a password and to confirm the password (figure 4) to use the Vdex management interface (Druid). By default the password is not defined. You **must** set a password, and then enter it again to confirm it. For reference, write down the administrator password here and then click finish to complete the Vdex Quick Start Wizard.

Administrator Password _____

Figure 4

Important note: The Vdex administrator password is also used when connecting to Vdex via SSH (see chapter 23 for more information).

Chapter 5: Logging In to the Management Console Druid

The Vdex system is configured and managed via web browser. Suggested web browsers to use for Vdex administration are Mozilla Firefox 2.0 or above or Microsoft Internet Explorer 7.0 or above. Open a web browser and set the web address to the IP address assigned to the Vdex system followed by /**druid**. For example, if the IP address assigned to the Vdex was 192.168.1.99, enter the following web address into your browser:

http://192.168.1.99/druid

Or if Vdex is using the default IP address point your browser to:

http://192.168.0.157/druid

If everything is connected and configured properly, the Vdex management interface (Druid) login screen will be displayed (shown in figure 5). You'll be prompted to enter a username and password. Unless changed in the previous step using the Vdex Wizard, the default user name and password are **admin** and **admin** (these values <u>are</u> case sensitive).

Figure 5

Once logged into the management interface, administrators can create extensions, program incoming and outbound dialing routes, create auto attendants, and complete the system setup.

Chapter 6: Administration

Administration starts with the home page (figure 6). The home page contains shortcuts providing quick access to areas within the management interface where you can quickly setup the system. Using the home page you can add extensions, manage voicemail for users, add VoIP trunks, and start and stop the system.
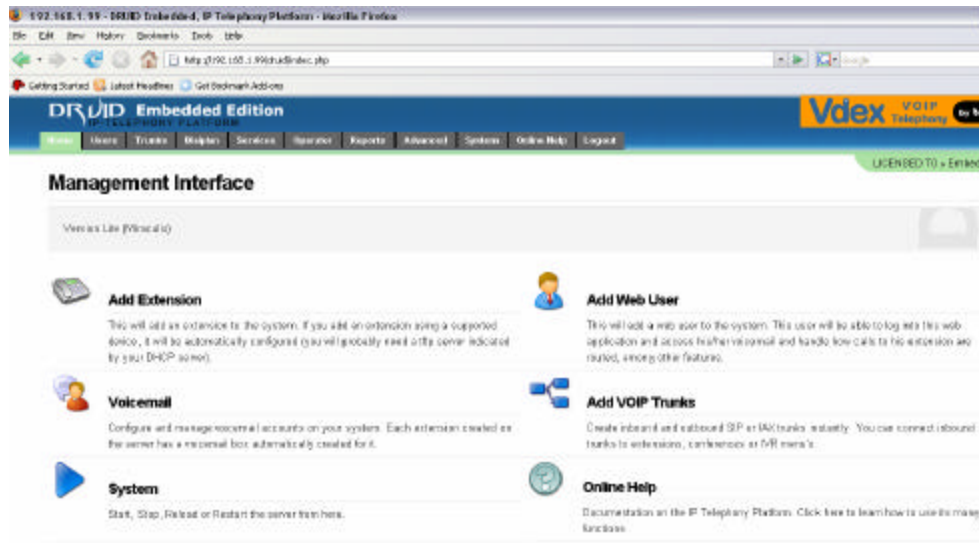

Figure 6

** **Please Note**** the Druid Lite version does not support some features of the Druid Full version. Vdex can be easily upgraded to Druid Full if desired. An upgrade license can be purchased by using the link provided within the Druid Lite interface. Upgrading to Druid Full supports additional features such as web users (allows individual extension users to log in, access voicemail and change call forwarding settings), phone auto-provisioning, dialplan wizard, conference recording, added conference management facilities and Pilot Ring Hunt Groups for inbound calls.

Chapter 7: Adding Extensions

Now that you're connected to the management interface, you'll want to add extensions. Click the **add extensions** button and then choose the type of SIP device that will be using the extension (see figure 7).
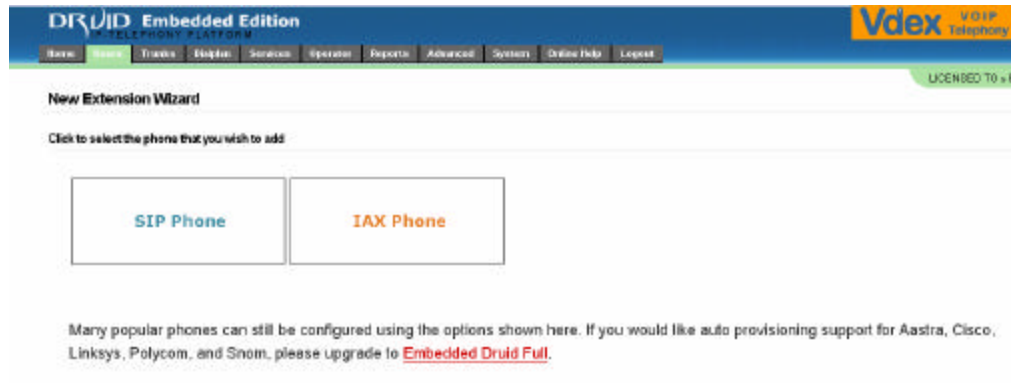

Figure 7

Depending on the type of VoIP telephone being connected to Vdex you may choose SIP or IAX. In most cases you will choose SIP. After choosing the type of extension, you'll be prompted to enter some basic information about the extension (shown in figure 8):

Figure 8

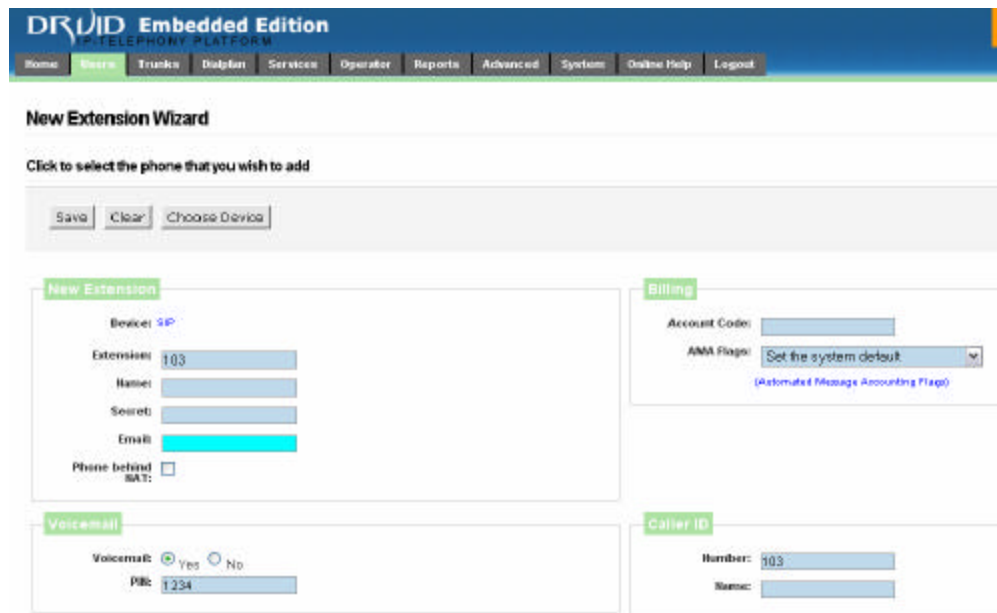When adding an extension you must define the name of the user, a secret password (used for telephone registration), their email address and if their telephone is behind a NAT (network address translator). By default voicemail is enabled for every extension added with a default password of 1234 for all extensions. The system administrator can add up to 35 extensions to Vdex. Repeat this process as needed.

Tip: There is an advantage to using an IAX phone for remote users. IAX phones easily traverse NAT (network address translators) due to only requiring port 80 be used for communication between the IAX phone and Vdex. Port 80 will always be open to configure the Vdex system remotely. Therefore there are no firewall issues to deal with as there are when connecting Vdex using SIP. SIP extensions require ports 5060, and 10,000 to 20,000. See chapter 21 for more information about connecting telephones to Vdex remotely.

** **Please Note**** when using two digit extensions, limit the extension number between 10 and 50 as to avoid conflicts with other dialplan features such as call parking, call pickup, etc. (please see Appendix A: Telephone Features Quick Reference for a full listing).

Customizing SIP/IAX Settings for each Extension:

Once extensions are added to the system, each extension's SIP/IAX settings should be customized for the manufacturer of the telephone. Click on the **Users** tab and then **Manage Extensions**. You'll see a list of all the extensions that have been added to the system (see figure 9).

Figure 9

Choose to **edit** each of the extensions **SIP/IAX Settings** and define the following parameters:

DTMF Mode   Set the DTMF mode to RFC-2833 (this is the preferred DTMF mode with most SIP or IAX telephones). However, if RFC-2833 is not supported by your chosen brand of SIP telephone you will need to change the DTMF mode to the type supported by the telephone. For example when using Linksys SPA 942 telephones you should set the DTMF mode to AUTO.

Tip: Extensions must be added before IP phones can be configured to connect and use the system.

Chapter 8: SIP Phone Configuration

Once extensions are added, it's time to configure the SIP phones to connect to the Vdex server. SIP phones must be manually configured to use the Vdex server. Most SIP phones can be programmed via web browser or using the display on the telephone. You can use whichever is your preferred method of setting up the SIP phones to register with the Vdex server. Log into the management interface for each IP phone either using the display on the phone or web browser and define the following settings:

Registrar or Proxy	Default IP is 192.168.0.157; enter the IP address assigned to the Vdex server.
User ID/Account and Mailbox	Enter the extension the IP phone will be using. Some phones call this the user id; others call it the registration account. The mailbox is also typically the same number as the user id/account.
Display Name	Enter the name of the person using the extension.
Password	Enter the secret password assigned to the ext.
Auth ID	Set the authentication ID to the same number as the User ID/Account and mailbox (enter the extension the IP phone will be using).
NAT Mapping Enable	Set to yes if the phone is behind a NAT.
Use Auth ID	Set to Yes

Keep in mind that some IP phones will be configured for DHCP by default. This is fine as long as you have a DHCP server that will manage the IP addresses used by the SIP phones. If preferred a static IP address can be manually assigned to each telephone. Once the IP phones are connected to the Vdex server, the LEDs on the phone should become green (with some telephones such as Linksys), and or when you press the speaker button or lift the handset you'll hear dial-tone.

X-Lite Soft Phone (SIP) Configuration Settings

If you would like to use a soft phone, you can download and install X-Lite from Counterpath.

http://www.counterpath.com/xlite-overview.html

It's free version is excellent and a full version can be purchased if desired. There are only a few features that are disabled in the free version. Figure 10 shows an example of how to setup X-Lite to connect to the Vdex server.
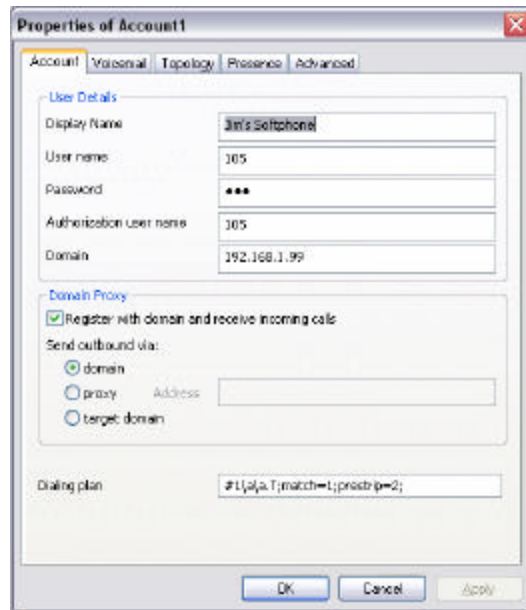

Figure 10

**Tip**: In this example, X-Lite is being used to connect to a Vdex system on the same subnet (network). You can just as easily change the IP address to an external Vdex system. Simply change the domain IP to the external IP address you'll be connecting to.

Zoipher IAX Softphone

Besides using X-Lite, you can download and install ZoIPer from www.zoiper.com. It's a free IAX soft phone that can be used with the Vdex system.
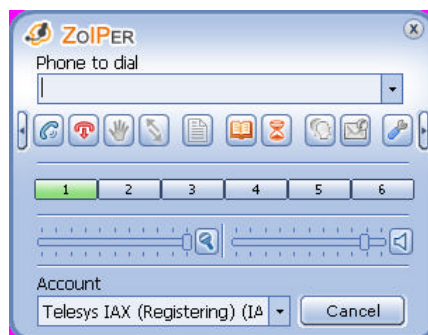
Here is an example configuration for Zoiper using IAX.
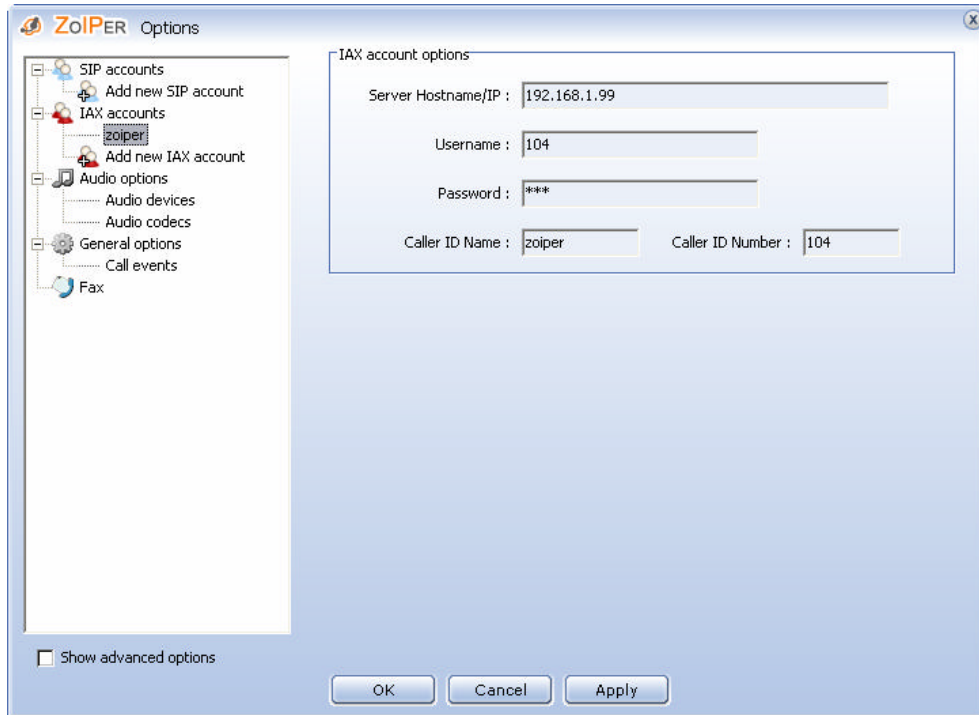
You must define the following settings in order to register the Zoiper extension with the Vdex system:

> **Server Hostname/IP –** Set this to the IP address of the Vdex system if running Zoiper on a PC within the same subnet as the Vdex system, or to the IP address of the router (external IP address) if connecting to Vdex remotely.
>
> **Username –** Enter the IAX extension on the Vdex that will be used by Zoiper.
>
> **Password –** Enter the password for the IAX extension defined by the username. This is also known as the secret.
>
> **Caller ID Name –** Enter the name you would like to show up on the display when you call other Vdex extensions.
>
> **Caller ID Number –** Enter the number you would like to show up on the display of other Vdex users when called using Zoiper.

Chapter 9: Creating IVR Menus (Auto Attendants)

To create an IVR menu click on the **Services** tab, and then click **IVR Management**. An IVR menu is your basic auto attendant. The Vdex can create an unlimited number of auto attendants. Each FXO or SIP trunk can have its own custom IVR menu, or they can all be configured for the same IVR menu. When creating an IVR menu, you can define the following options:

Name	Enter a name for the IVR menu. The name will be used when directing incoming calls to the IVR menu (use Dialplan/Manage Inbound Routes/DID's to direct FXO or SIP trunks to IVR menus).
IVR Message Sounds	Define the sound file(s) to be played when a caller reaches the IVR menu. Sounds can be uploaded or recorded via telephone.
Wait for	The number of seconds the Vdex will wait after the IVR message is played before processing the on timeout goto definition.
On Timeout Goto	When a caller doesn't take any action, their call can be redirected back to the start menu (back to the top of the IVR menu, repeating the menu) or to an extension (such as an operator).
Allow Direct Dialing to local extensions	When enabled, callers can dial extensions from the IVR menu. When disabled, callers can only dial key actions (single digit menus) or wait for the timeout to occur.
Add key action	Used to define single digit menu options. Menus (0-9) can connect to an extension, conference room, directory, or go directly to voicemail for a specific extension.

Recording IVR Message Sounds

There are two methods of recording IVR sounds. You can either use any SIP or IAX extension connected to the Vdex40 system, or use the GUI to upload a sound file to the system.

To record IVR message sounds via telephone:

Using any extension, you can **dial 7900 – 7909** to record IVR message sound files ivr_g0 through ivr_g9 (IVR greeting 0 through IVR greeting 9), up to a maximum defined length of 5 minutes. Once recorded these sound files will be accessible from the **sounds/druid-sounds** folder. After recording an IVR message, you can **dial 7910-7919** to listen to your recordings.

For more information about recording IVR message sounds via telephone see the Vdex40 system telephone features quick reference (Appendix A) at the end of the manual.

To upload an IVR message sound file:

Custom sound files can be uploaded to the Vdex system and be used as IVR message sounds. To upload a custom sound file click **Services => Sounds => Upload file**. When uploading a sound file, it must be in the following format:

WAV 8 KHz, 16 bit, mono

Using the following Sound Upload dialog (figure 11), browse to the sound file you want to use as an IVR Sound Message. Then click ok to upload the sound file.
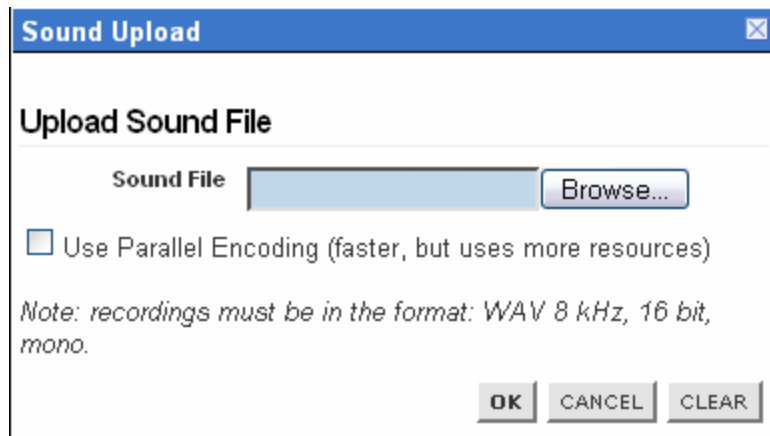

Figure 11

Important Note: The file will be placed in the **druid-sounds** folder on the system. When defining what IVR message sounds to play make sure to look for your uploaded files in the druid-sound folder. Multiple sound files can be played one after the other in IVR menu sounds, just add each sound file to IVR message sounds.

Chapter 10: Inbound Call Routing

Now that extensions have been added and configured to attach to the Vdex system, it's time to setup incoming call routing. This is done using the **Dialplan** tab. Click dialplan and then **Manage Inbound Routes (DID's)**. A unique inbound route can be configured for each FXO port or SIP trunk. Each FXO port or SIP trunk can be routed to an extension, an auto attendant (IVR menu), to voicemail for a particular user, or to a conference room. Inbound call routes can also be configured to automatically change based on the time of day, day of week, or day of month. For example, calls can be routed to a live operator during the day and to an auto attendant (IVR menu) at night. Figure 12 shows the Dialplan/Manage Inbound Routes DIDs screen. To add an inbound route for an FXO port, click the down arrow next to Add Inbound Route and select the FXO port from the list. From there you can setup call routing for the selected FXO port.

Tip: Keep in mind that you can connect up to four physical telephone lines to the Vdex system using FXO1 – FXO4. Additionally, Vdex can support up to 6 SIP provider accounts and depending on bandwidth and provider a variable number of SIP trunks. SIP trunks use approximately 87k of bandwidth per line when using ulaw as the primary codec.

Adding Inbound Call Routes

To add a new inbound route select the desired FXO port from the drop down list and click the **Add Inbound Route** button (figure 12). This will open the Create/Edit Inbound Routes dialog shown in figure 13.
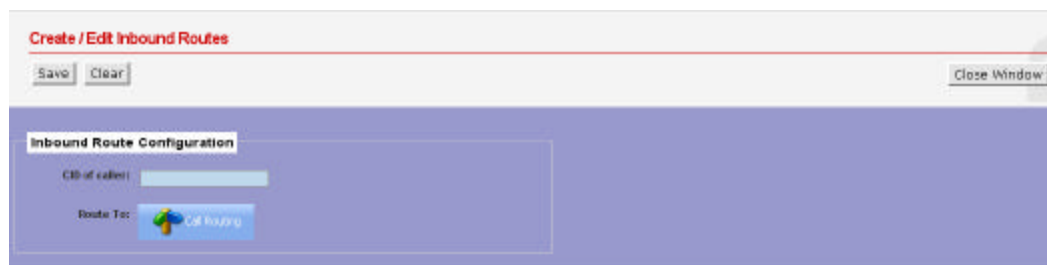

Figure 12

Figure 13

Click the **Call Routing** button to open the scheduled routing dialog where you can define how calls will be routed for that FXO port. Figure 14 shows the FXO port routing calls to extension 101 all the time.
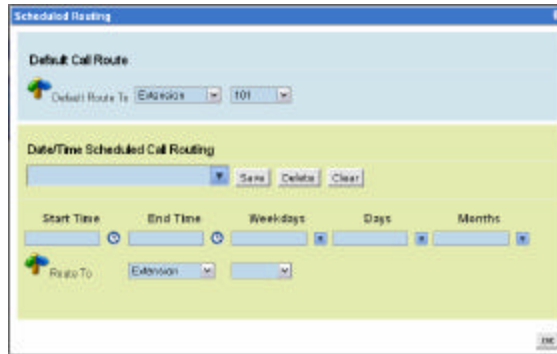

Figure 14

Figure 15 shows calls being routed to extension 101 from 9:00am -5:00pm Monday through Friday, everyday, month, and year.
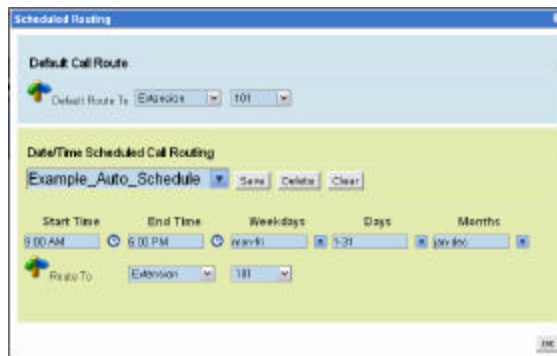

Figure 15

Figure 16 shows calls being routed to an auto attendant (IVR menu) called aa-welcome at night and on weekends from 5:00pm – 9:00am, everyday, month, and year.
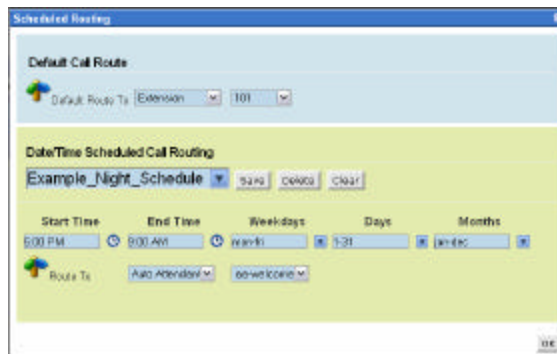

Figure 16

Figures 12 – 16 are good examples of how incoming call routing can be defined for an FXO port. Keep in mind each FXO port must have its own scheduled routes. To schedule

call routing use the mouse pointer and click on the **clock icon** to set the **start time** and **end time**. To set the weekdays, days, or months click the **down arrow** to the right of each of these items and drag it down over the desired days or months. Click ok when finished, and click save to keep the call routing schedule. Repeat this process to add as many schedules as desired for each FXO port (1-4) or SIP trunk.

Chapter 11: Outbound Dialing

To make outbound calls, you must connect at least 1 line to the Vdex system. The system can accommodate up to 4 analog telephone lines, and up to 6 SIP provider accounts. Once the telephone lines are connected to FXO ports users can make outbound calls by dialing:

9 + <7 or 10 digit number> (to make a local call)

9 + 1 + <10 digit telephone number> + (to make a long distance call)

011 + <number> (to make an international call)

These are all default outbound dial rules.

One really nice thing about the Vdex system (and Asterisk) for that matter is how flexible it is. It's very easy to change the default or add additional outbound dialing rules. For example, If you would like to add a new outbound dialing rule changing the default digit to dial out from a 9 to an 8 or some other access code such as a four digit code (making outbound dialing more restrictive) it's easy to do so. Figure 17 shows an example of being able to dial a 1 as a lead digit instead of a 9, and not having the 1 stripped out of the dialed number. This way long distance calls can be made by simply dialing the 11 digit number (1 + the 10 digit number).

To add a new rule, click on **Dialplan => Outbound Dialing Rules => Add Outbound Rule => Pattern Wizard**

Figure 17 shows a new rule defining the ability to dial a 1 + a 10 digit number to make a long distance call.
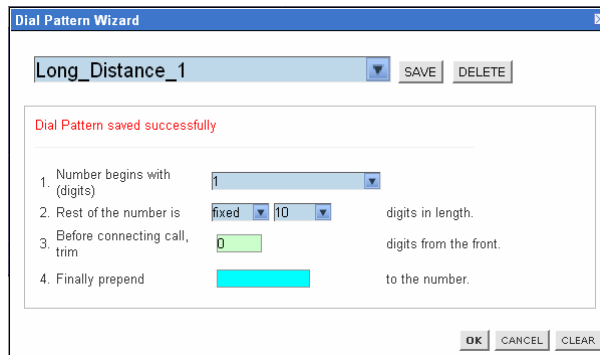

Figure 17

New outbound dialing rules can be added to allow the use of different FXO or SIP trunks based on the first digit dialed. By default when dialing out FXO ports 1-4 may be used to make the outbound calls depending on the lines in use. If you would like to define a new outbound dialing rule so that FXO port 3 is used when callers press 7 + the number, you would select FXO3 as the outbound trunk when creating the dialing rule (see figure 18).

Figure 18

Another example of when to add additional outbound dialing rules is when you are using SIP trunks in addition to FXO trunks. You would dial 9 to access an FXO port, and add a new outbound rule to access the SIP trunks. For example you might create a rule choosing your SIP trunk as the outbound trunk, and configure the rule to use an 8 as the first digit and then have that digit stripped and set the number of digits dialed to fixed 10. Figure 19 shows an example of this.

Figure 19

Important Note about 911 Emergency Calling:

By default users can dial 911 or 9+911 to reach emergency services. Be aware that the default outbound dialing rules use FXO port 1 to make emergency calls. Please make sure to connect at least one telephone line to FXO port 1 or to reconfigure the outbound dialing rules for 911 and 9+911 to use SIP trunks (if supported by your SIP trunk provider).

Chapter 12: Creating Conference Rooms

To create a conference room, click on the **services** tab, and then click **conferences**. Click new **conference room** (see figure 20) to open the new conference room dialog (shown in figure 21). Using this dialog, you can quickly and easily create conference rooms by assigning a conference room number, and optionally PIN number for users and administrators.
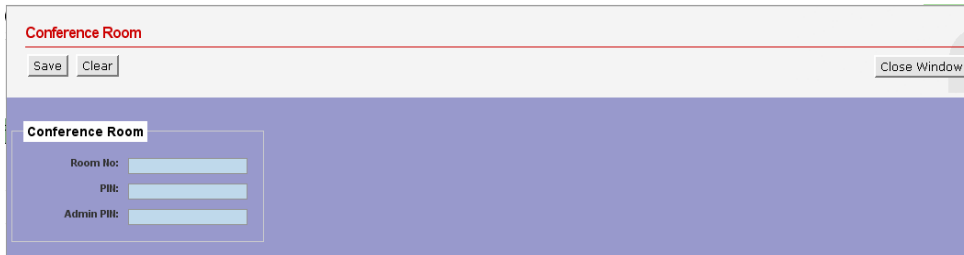

Figure 20

Figure 21

Room No	Enter the number you wish to assign to the conference room.
PIN	Optional, enter a password to assign to the conference. Callers who attempt to enter the conference must enter the PIN number or the will be rejected from the conference.
Admin PIN	Optional, enter a password for the conference administrator. Press * when in the conference for an advanced list of admin options for the conference.

Note: Conference recordings (the ability to record conferences is not supported in the standard Druid Lite version. This feature is available in the Druid Full version.

Chapter 13: Voicemail

When an extension is added to the Vdex system, a voicemail box is automatically created for that extension. The default settings for voicemail users can be modified after the extension is added by clicking on the **voicemail** icon from the home page, or by clicking on the **users** tab and then voicemail. From there you will see all the voicemail boxes on the system, an example of this is shown in figure 22.

Figure 22

Voicemail Parameter Definitions:

Mailbox	Voicemail box number. Mailbox numbers are the same as extension numbers. For example, extension 101's mailbox is 101.
Mailbox Pin	Voicemail box password. The default password for all mailboxes is 1234.
Name	Name of the mailbox user.
Email	Email address used when Vdex copies voicemail to email.
Attach Voicemail to Email	Enable this setting for users who would like to send a copy of their voicemail messages to email.
Announce Caller ID	Enable this setting for users who would like to hear the Caller ID number of the person who recorded them the message when listening to messages via telephone.

Delete Voicemail after sending email	Enable this setting if users would like to automatically delete messages from their mailbox after its been successfully sent to email.

Chapter 14: Call Forwarding

Vdex extensions are automatically forwarded to voicemail after 4 rings. Extension users can also forward calls from their extension to another extension by pressing the cfwd (call forward) button or by **dialing 773** and entering the extension number to forward calls to. Pushing the cfwd button again or **dialing 774** cancels call forwarding. Extensions that are in do not disturb will forward all calls to voicemail immediately. To put an extension in dnd, push the dnd button on the phone. To take an extension out of do not disturb push the dnd button again. For more information call forwarding or do not disturb see appendix a.

773 + <extension>	Call forward calls to designated extension
773	Call forward all calls to the previously designated extension
774	Cancel call forwarding

Chapter 15: Setting the System Time

In order to set the system time to the correct time you must set the time zone where the Vdex system is located. To do this click on the **Advanced** tab and then click on **Security-Network-Codec**. Once there you'll see a number of different tabs. Click on the **Date/Time** tab. You'll see the current system time, and can change the time zone by selecting the correct time zone for your area. Once selected click **Save Date/Time Settings**. Figure 23 shows a Vdex system configured for the US/Eastern time zone. The Vdex does not require setting the date or time of the system. All that is required to correct the system time is to set the system to the correct time zone.
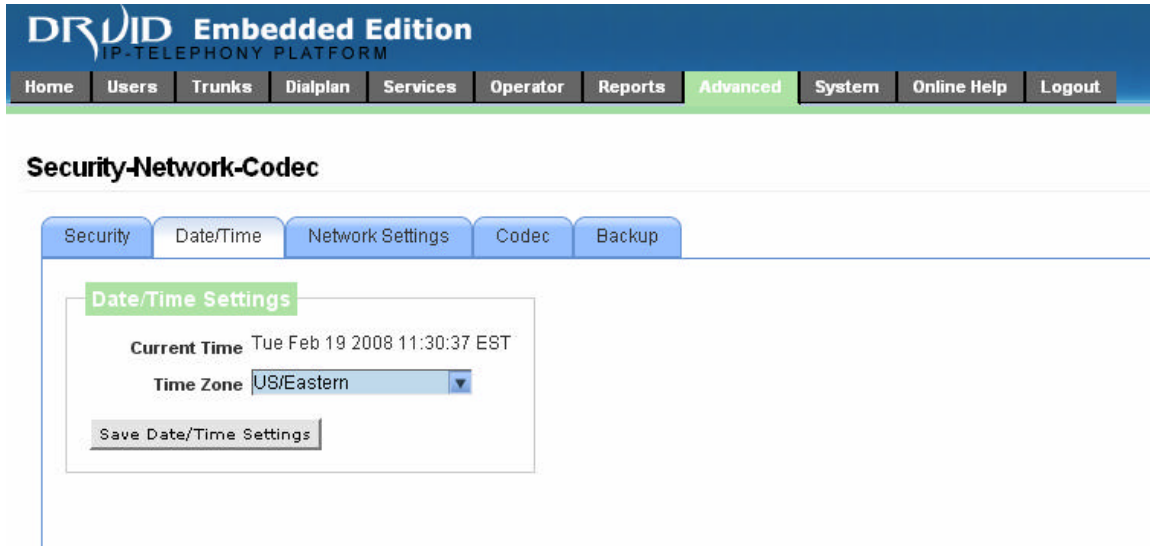
Figure 23

Important Note: In order for the system time to be updated automatically via time server internet connectivity is required. Be sure to configure Vdex to use an appropriate gateway and DNS settings to access the internet. Run the Vdex Wizard to define gateway and other DNS settings if it hasn't been done already.

Chapter 16: How to Create a Hunt Group

Click on the advanced tab and then manual edit configs. In the drop down list choose the **extensions.conf** file and click open. Once the file is open, scroll down until you find the **[druid-users]** context. Add the following lines to the bottom of the context modifying the SIP extension values to the telephone extensions you would like to include in your hunt group. This hunt group is linear and will ring each extension in order. If calls go unanswered to the hunt group the caller is sent to voicemail for extension/mailbox 101 for this example. Save the file and do a reload.

Once the config file has been reloaded you can easily program any or all incoming lines to ring into the hunt group. In this example, hunt group 200 will ring extensions 101, 102, and 103 for 10 seconds each. If the call is not answered the caller will be sent to voicemail for mailbox 101 where they can record a message.

exten=>200,1,Dial(SIP/101,10)

exten=>200,2,Dial(SIP/102,10)

exten=>200,3,Dial(SIP/103,10)

exten=>200,4,VoiceMail(u101)

Looking at the syntax for this example, when someone dials extension 200, first extension 101 will be called. Extension 101 will ring for 10 seconds, and then the call will be routed to extension 102. Extension 102 will ring for 10 seconds, and then the call will be routed to extension 103. Extension 103 will ring for 10 seconds and if the call goes unanswered the caller will be sent to voicemail for extension 101. The caller will hear the unavailable message (u) for extension 101 where they can leave a message. If you would like the caller to hear the busy message of mailbox 101 change the u to a b (for example b101).

If you would like the caller to hear music as opposed to ringing while waiting for their call to be answered, you can modify the Dial sequence like this:

exten=>200,1,Dial(SIP/101,10,m)

Tip: Keep in mind that you must change the values of SIP/ext to the extensions you wish to ring and that you can change the pilot number (dialed number to reach the hunt group) 200 to whatever number you would like to use.

Chapter 17: How to Create a Ring Group

Click on the advanced tab and then manual edit configs. In the drop down list choose the **extensions.conf** file and click open. Once the file is open, scroll down until you find the **[druid-users]** context. Add the following lines to the bottom of the context modifying the SIP extension values to the telephone extensions you would like to ring simultaneously. Save the file and do a reload. Once the ring group extension has been reloaded you can easily program any or all incoming lines to be answered by the ring group. In this example, ring group 888 will ring extensions 201 and 202 at the same time for up to 40 seconds. If the call goes unanswered, then the caller will be sent to the specified auto attendant (for this example aa-vdex_demo_ivr).

exten =>888,1,Playback(please-hold-while-we-connect-you)

exten =>888,2,Dial(SIP/201&SIP/202,40,m)

exten =>888,3,Goto(aa-vdex_demo_ivr,s,1)

Alternatively, you may want to modify the ring group so the caller will be sent to voicemail instead of an auto attendant after 40 seconds if their call is not answered. The following example shows the caller being sent to mailbox 101.

exten =>888,1,Playback(please-hold-while-we-connect-you)

exten =>888,2,Dial(SIP/201&SIP/202,40,m)

exten =>888,3,Voicemail(101)

Tip: Remember to change the SIP extensions to the extensions you would like to ring and add as many as desired using the same format. To reach the ring group dial 888 from any phone internally, or program one or more incoming lines (DID's) to ring into the ring group extension.

Chapter 18: How to Setup a Speed Dial Extension to Call an External Number

In order to setup a speed dial extension, you must edit the dial plan. Click on the advanced tab and then manual edit configs. In the drop down list, open the extensions.conf file. You must add a line that is similar to the following example to the [druid-users] context (modifying the extension number and target phone number appropriately.

exten =>7,1,Dial(MSPD/fxoall/18004102745)

Save the changes to the file and then reload the server. For this example, extension 7 has been added to the system. Extension 7 will grab any available FXO port and dial 18004102745 and then connect the call. Change the extension value from 7 extension value you would like to use to dial the target phone number. Change the target phone number from 18004102745 to the number you would like to dial.

Chapter 19: Upgrading the Vdex Firmware

Before programming your new Vdex system you should check to see if an update to a newer version is available. The Vdex system is constantly evolving and in many cases improvements or fixes to reported problems may already be available in the latest firmware update.

To upgrade the Vdex firmware, click on the **Services** tab. Next to the update firmware button is the web address where the firmware will be obtained. The default value is:

http://downloads.technoco.biz/vdex40/firmware/usa

Change the "usa" to "amanda" to obtain the latest Amanda firmware update for the Vdex 40 system (as seen in figure 24a).

Next, click the **Update Firmware** button. You'll be prompted to please wait… Allow up to 15 minutes for the firmware update to be completed. Figure 24b shows what you'll see after the firmware update has completed successfully. After the upgrade is complete, you **MUST** reboot the Vdex system to finish the upgrade process. Click the ok button to close the information window, and then click the **Reboot** button.

Figure 24a
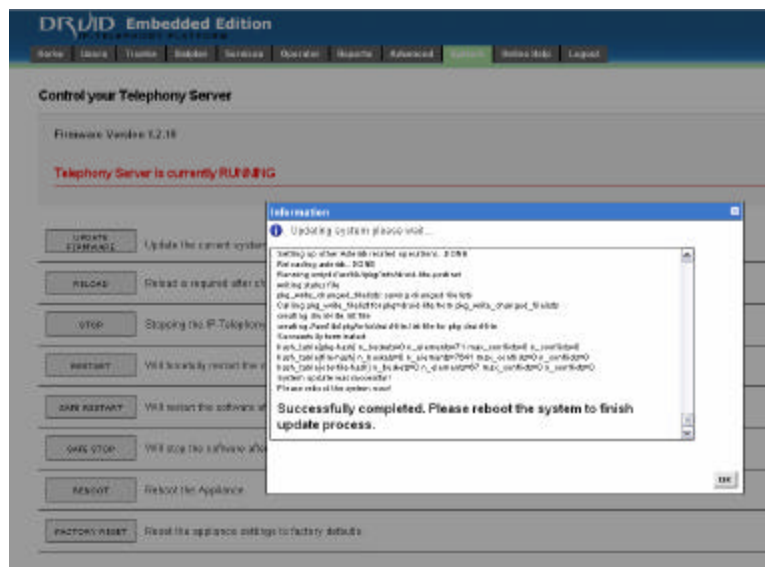
Figure 24b

Note: If you encounter errors upgrading the firmware, make sure you have configured Vdex with valid gateway, netmask, and DNS settings. If so, try doing the firmware upgrade a couple of times and it should eventually work.

Chapter 20: Backing up the Vdex System

Once the system has been configured per user requirements, it's time to backup the system. Vdex backups are done vi a email. To backup the Vdex system, click the **Advanced** tab, and then click **Security Network-Codec**. Within the Security-Network-Codec dialog, click on the **Backup** tab. Enter the email address you wish to send the system backup to, and click **Backup Config to Email** (figure 25). An email will be sent from the Vdex system to the email address provided. In the event of a system failure, the backup can also be restored by clicking **Restore backup from file**.
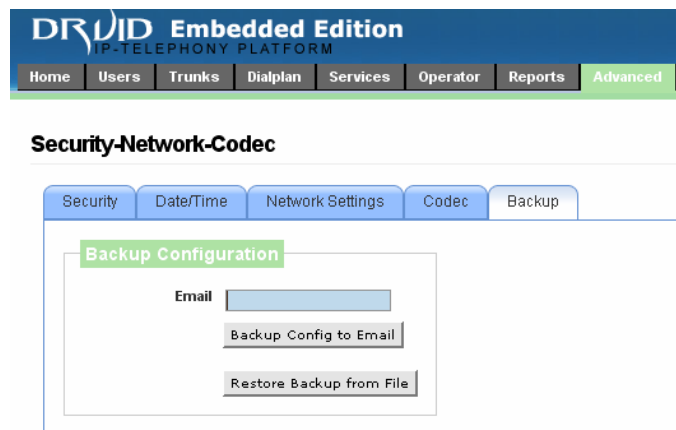
Figure 25

Tip: To restore a backup, you must make sure the file sent to email is located on the local PC used to run the Druid management interface. If needed, copy the backup to some other type of removable media such as a thumb drive or burn it to a CD-ROM to make it easy to access from the local PC. You can also map a network drive to the backup file and access it from there.

Important Note: In order to use the backup to email feature, a valid SMTP server must be defined. Additionally, the SMTP server must allow relay and not require authentication. To check the value for the SMTP server click on Advanced => Security-Network-Codec and then network settings. Scroll down to the SMTP server value under mail settings.

Chapter 21: Remotely Connecting Telephones to Vdex

In order to allow remote telephone users to connect to the Vdex system, the network administrator must forward the following ports within the router to the IP address of the Vdex server.

Port / Protocol	Forward To
80 (TCP/UDP) **for IAX/Web Admin**	<Vdex System Internal IP Address>
4569 (TCP/UDP) **for IAX**	<Vdex System Internal IP Address>
5060 (TCP/UDP) **for SIP**	<Vdex System Internal IP Address>
10,000 – 20,000 **(TCP/UDP)** **for RTP**	<Vdex System Internal IP Address>

Keep in mind that when you configure remote telephones, you must configure the phones to connect to the external IP address of the router (not the internal IP address of the Vdex system). Once port forwarding has been established, the router will forward data packets for ports 5060 and 10,000 – 20,000 to the Vdex server for SIP phones, and port 4659 for IAX phones. Remote users must have some type of broadband connection in order to connect to the Vdex server remotely.

SIP Phone Configuration

SIP phones MUST be configured to use ports 5060 for SIP and ports 10,000-20,000 for RTP. By default SNOM phones are not configured to use these ports and must be defined. Set the SIP & RTP ports within the web interface. Log into the telephone by entering in the IP address of the phone into a web browser. Click on the advanced tab and then SIP/RTP (for SNOM300, 320, and 360 phones).

Setting the NAT external IP address

You must also set the NAT external IP address. This is done by clicking on the advanced tab and then Security-Network-Codec within the Druid interface. Click on the network tab and set the **NAT External IP Address** to the IP address of your router.

Tip: To avoid firewall and NAT related issues it is recommended that IAX phones or soft phones be used when connecting to Vdex remotely. With that said, this is not required; SIP phones can also be used without a problem as long as the appropriate ports are open (forwarded in the router) and that the SIP phones are configured to use those ports.

Chapter 22: Resetting the System to Factory Default

Important Note While it is possible to reset the Vdex system back to factory defaults, be aware that **ALL CONFIGURATION CHANGES WILL BE LOST**. Proceed with caution.

There are two ways to reset the system back to factory default:

Method 1: Hold down the reset button on the rear of the Vdex unit for 12 seconds.

Method 2: Click on the system tab, and then click on the factory reset button. Figure 26 shows the system tab options where factory reset is the last button on the screen.

The reset procedure will reboot the Vdex, and will take approximately 5 minutes to complete. Remember to re-run the Wizard to configure the initial settings of the Vdex.
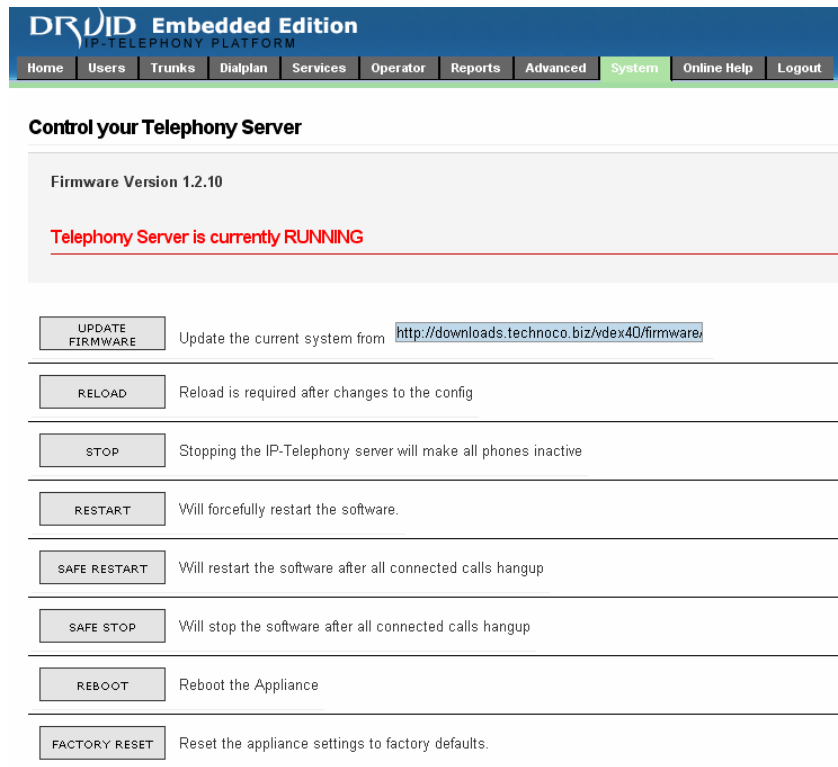
Figure 26

Tip: Before resetting a system back to factory defaults, it is suggested that you perform a backup to email (in case you want to revert back to your previous setup). See chapter 17 for details on how to perform a backup to email. Also, please note that IVR message recordings will not be lost when resetting the system back to factory default.

Note: To reset the Quick Start Wizard to factory defaults, allowing a user to run the Wizard without entering a password first, change the Vdex password to "admin".

Chapter 23: Tracing Problems - Connecting to Vdex using PuTTY (SSH)

Experienced users may wish to access the command line interface of the Vdex using SSH. Any preferred SSH client may be used – for Windows users, we recommend PuTTY. PuTTY is a free, compact SSH client that may be downloaded from the following location:

http://www.chiark.greenend.org.uk/~sgtatham/putty/download.html

Once PuTTY has been downloaded, it can be run by double clicking putty.exe. To connect to the Vdex server, set the host name to the IP address assigned to the Vdex system. If you wish, you can enter a name into Saved Sessions (such as "Vdex") and click Save so that PuTTY remembers the IP address for future use. Click Open to connect to the Vdex system.
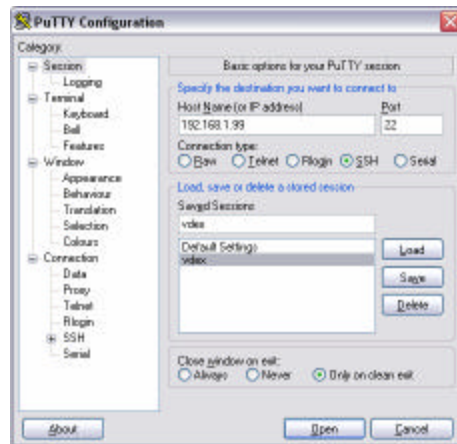
Figure 27

Once connected, you'll be prompted for a login user and password. Enter **root** as the login user, and your Vdex administrator password assigned when using the wizard.

Note: If you wish to enable external access to SSH on the Vdex from the Internet, you must first **forward port 22** to the Vdex IP address in your router.

Figure 28

Once connected, you can use asterisk commands in the CLI (command line interface) such as:

 # asterisk –r
 # set verbose 1000

Connecting using SSH and Putty is a great way to troubleshoot complex problems. If encountering problems with the systems information gathered here can be crucial in finding a quick solution for the problem. After setting verbosity, place some test calls and reproduce the problem. Once reproduced, copy the text from the window and paste into notepad, then save the file. Email the file with a description of the problem to technical support for assistance.

Chapter 24: Manually Editing Configuration Files

Please note that before you edit ANY configuration file you should always perform a backup to email first, and or copy the text from the file to notepad and save a copy of the text prior to editing and reloading an updated configuration file. This is to avoid possible corruption to the file and is an excellent safe practice to insure data is not lost for any reason. To manually edit a Vdex configuration file, click on the **advanced tab**, and then **manual edit configs**. Using the Config file list, select the file you wish to edit, and click open. Make the appropriate changes to the file, save the file and the reload the server for the changes to go into effect. Figure 29 shows an example where the extensions.conf file has been opened for review.

Tip: Remember to backup the text of the Config file using notepad or by doing an email backup prior to making any changes to any Config file.
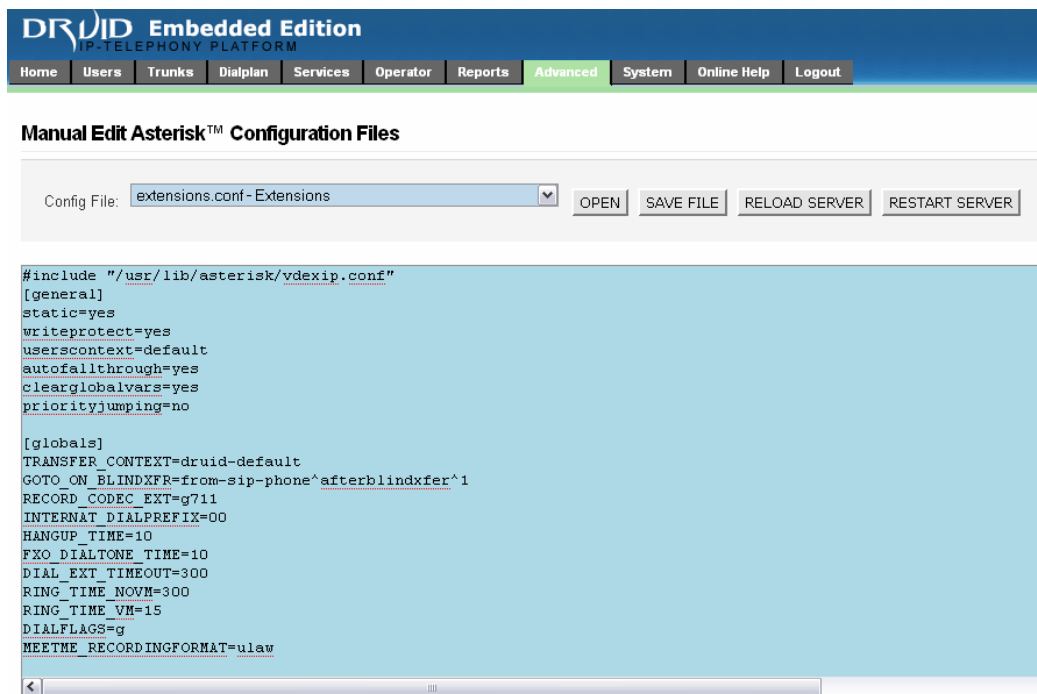
Figure 29

Please Note: Due to the Vdex being an asterisk based system it is possible to modify configuration files to extend its feature set. Experienced Asterisk users will feel very comfortable modifying these files. In most cases the files most commonly modified are sip.conf, and extensions.conf.

Chapter 25: Broadvox SIP Trunk Setup

This chapter outlines how to add SIP Trunks hosted by Broadvox to the Vdex40 system. Broadvox supports static and authenticated SIP trunks. Broadvox supports up to 6 concurrent call sessions per SIP trunk. To obtain information and pricing visit

http://www.broadvox.com

Once the SIP trunks have been provisioned by Broadvox they will send the information needed to configure and use the SIP trunks (usually via email).   This includes the BTN (telephone number), Host (IP address), and Password (if using authentication).

To Setup SIP Trunks:

1) Login to the Druid interface. Click on the **Trunks** tab. Click **Add Trunk Wizard** to add a new SIP trunk.

2) Within the Add Trunk Wizard set values for the following parameters:

Trunk Definition

> **Trunk Type**: - Set to SIP – Session Initiation Protocol
>
> **Trunk Name**: - Enter the name you would like to assign to the SIP trunk. To make managing SIP trunks easy we recommend you use the telephone number for the SIP trunk (BTN) as the name.
>
> **Host**: - Enter the IP address provided by Broadvox. Or as of this writing use either **209. 249.3.59** for static SIP trunks or **209.249.3.100** for authenticated SIP trunks.
>
> **Connection Type: -** Select inbound and outbound if you'll be using the SIP trunk to receive incoming and make outgoing calls.
>
> **Username: -** Leave blank if using a static SIP trunk. When using authentication enter the telephone number assigned to the SIP trunk (BTN) as the username.
>
> **Password: -** Leave blank if not using authentication. Or enter the password assigned to the SIP trunk when authenticating.

Inbound Trunk Settings

> **Register**: - Leave blank (unchecked) when using a static SIP trunk. Check register as a VOIP client when using authentication.
>
> **DID/Inbound Route: -** Set the value to "any" when using a static SIP trunk, or when using authentication set the value to the telephone number assigned to the SIP Trunk (BTN). Click on the Call Routing button. Set the default call route to an extension, conference room, or auto attendant.
>
> **Outbound Trunk Settings** – To define what to dial to use the SIP trunk to make outbound calls, click on the Pattern Wizard button. Assign the new outbound dialing rule a name such as SIP1 and define number begins with, rest of the

number must be set to fixed 10 digits, and before connecting call trim (set to the length of the number begins with value). Save the pattern, click ok and then exit.

Advanced (optional) – Leave these settings blank.

Caller ID – Set the caller id name and number to the name and number you would like to show up on the called party's display when using the SIP trunk to make outbound calls.

3) Click Save to keep the settings and add the SIP trunk.

4) If using a static SIP Trunk skip this step. Click on the Trunks tab and Manage Trunks. Edit the settings for the SIP trunk and change the value for **Authentication** / **Insecure** to **Ignore Authentication**. Save the changes.

5) Call the telephone number assigned to the SIP trunk (BTN). It should ring into the designated target extension, auto attendant or conference room. Test the outbound dialing of the SIP trunk to make sure you can make outbound calls using the trunk.

Appendix A: Telephone Features Quick Reference

Calling from extension to extension:	To make a call from one extension to another **enter the extension number and press dial**.
To make a local call: **9** + *<7 or 10 digit #>*	To dial a local number **press 9 and then enter the 7 or 10 digit number and press dial**.
To make a long distance call: **9** + **1** + *<10 digit # >*	To dial a long distance **press 9 + 1 + the 10 digit number and press dial**.
To make an international call: **9** + **0 1 1** + *<number>*	To make an international call **dial 9 + 011 + the number**, and then push dial.
In case of emergency dial: **9 1 1** or **9 9 1 1**	**In case of emergency dial 911 or 9 + 911 on any telephone. Please note that the system will use FXO1 to make the outbound call to 911 by default. You MUST make sure at least one line is connected to FXO1 to use this feature or modify the outbound dialing rule to use a SIP trunk if desired.**
Directory Access: **4 1 1**	**Dial 411** to access the system directory.
Direct send to Voicemail: **7** + *<extension>*	To record a direct message for another extension **dial 7 + the extension number.**
To park a call: **7 0 0**	**Put the caller on hold**. **Dial 700** and press the check button. You will hear the park zone where the caller will be parked. Press the **transfer** button and **hang-up**. To pickup the parked call dial the zone the call was parked in from another extension. Park zones range from 701-709.

Call Forward – Activate **7** (PQRS) **7** (PQRS) **3** (DEF) + **\<extension\>**	**Dial 773 + the extension** you want to forward all calls to. Or you can dial 773 + the extension you want to forward calls to and press dial.
Call Forward – Activate Last Forward Extension **7** (PQRS) **7** (PQRS) **3** (DEF)	**Dial 773** after call forwarding has been activated and de-activated once and forwarding will be re-enabled to the last extension calls had been forwarded to.
Call Forward - De-activate **7** (PQRS) **7** (PQRS) **4** (GHI)	**Dial 774** to de-activate call forwarding. When forwarding is disabled you will hear a message stating "forwarding has been disabled".
Call Pickup **7** (PQRS) **7** (PQRS) **7** (PQRS) + **\<extension\>**	To pickup a call ringing on an extension other than your own, **dial 777 + the extension** number to answer the call.
Single Intercom Page: **7** (PQRS) **8** (TUV) **0** + **\<extension\>**	To announce a message over the intercom of a single extension **press 780 + the extension**. Announce your message to hear it on that user's telephone speaker and then hang-up. Please Note: Auto answer must be enabled on the extension being dialed to use this feature.
To Access Voicemail **7** (PQRS) **8** (TUV) **7** (PQRS)	To retrieve voicemail messages from your telephone extension, press the message button on your telephone or **dial 787 from your telephone**. You will be prompted to enter your password. The default password is 1234.
To reach the message center **7** (PQRS) **8** (TUV) **8** (TUV)	To reach the general message center **dial 788** or **press the envelope button** on your telephone. You'll be prompted to enter your mailbox number and password. **The default password is 1234** for all mailboxes.

To record IVR message 0-9: 7 9 0 0 to 7 9 0 9	IVR messages can be recorded by **dialing 7900-7909** and are used for auto attendant messages (IVR management), up to a maximum defined length of 5 minutes. IVR messages are stored within sounds/druid-sounds. To use an IVR message as an auto attendant greeting, after recording your message, you must choose that IVR message file within sounds/druid-sounds. To record IVR message 0 – 9 (ivr_g0 – ivr_g9) dial one of the following: 7900 - Record IVR message 0 (ivr_g0) 7901 – Record IVR message 1 (ivr_g1) 7902 – Record IVR message 2 (ivr_g2) 7903 – Record IVR message 3 (ivr_g3) 7904 – Record IVR message 4 (ivr_g4) 7905 – Record IVR message 5 (ivr_g5) 7906 – Record IVR message 6 (ivr_g6) 7907 – Record IVR message 7 (ivr_g7) 7908 – Record IVR message 8 (ivr_g8) 7909 – Record IVR message 9 (ivr_g9) .
To play back IVR message 0-9: 7 9 1 0 to 7 9 1 9	To play back IVR messages recorded using 7900-7909 (ivr_g0 – ivr_g9) dial one of the following: 7910 – Playback IVR message 0 (ivr_g0) 7911 – Playback IVR message 1 (ivr_g1) 7912 – Playback IVR message 2 (ivr_g2) 7913 – Playback IVR message 3 (ivr_g3) 7914 – Playback IVR message 4 (ivr_g4) 7915 – Playback IVR message 5 (ivr_g5) 7916 – Playback IVR message 6 (ivr_g6) 7917 – Playback IVR message 7 (ivr_g7) 7918 – Playback IVR message 8 (ivr_g8) 7919 – Playback IVR message 9 (ivr_g9)
Access Default IVR 7 9 9	**Dial 799** from any telephone to access the default IVR. Using the Default IVR you can **check or change the IP address and network mask of the Vdex 40 system by dialing 654321** during the message.
Creating a conference with another user:	While on a call **press the conference button**, **dial the extension** you want to add to the conference and **press the conference button again** to create the conference.
Entering a conference room:	From a local extension, **dial the conference room number** and **press dial** to enter a conference. If needed, enter the conference or admin PIN.

Blind Transfer:	This depends on the type of telephone being used. When using a Linksys phone: While on a call **press the transfer button** and **enter the extension** number, **press the transfer button again** and **hang-up**. When using a SNOM phone: While on a call **put the caller on hold, dial the extension you want to transfer the caller to, press the transfer button, and then hang-up.**
Redial:	The **redial button** can be used to open a list of previously called numbers. Using the **up and down arrows** select the number to be redialed, and **press the dial button** to call that number.
Supervised/Attended Transfer:	While on a call **put the caller on hold, dial the extension you want to transfer the caller to, when the called party answers announce the caller, press the transfer button, and then hang-up.**
Call Forward Busy	Depends on the telephone being used. Enabling and Disabling busy forwarding on is a feature of the phone. Most SIP phones are programmable and come with default codes to enable and disable busy forwarding. In some cases its *90 + the extension to enable and *91 to disable busy forwarding. Please consult the documentation for the telephone for more information.
Auto Answer	Some VoIP phones support auto answer. Check in the phone setup to enable/disable this feature. Auto answer can be enabled on SNOM phones using the web interface for the phone.
Do Not Disturb	Enabling and Disabling do not disturb on your telephone extension is a feature of the telephone. Most SIP phones are programmable and come with default codes to enable and disable do not disturb. In some cases its *78 to enable and *79 to disable do not disturb on your phone. Some telephones have a DND button that you can press to enable do not disturb and press again to disable. Please consult the documentation for your telephone for more information and to confirm the dial codes for do not disturb.

Appendix B: Voicemail Features Quick Reference

Accessing Voicemail

When in the office press the message key on your telephone or press ⑦⑧⑦ + dial to access voicemail. When prompted, enter your mailbox password. The default password is ①②③④ for all users. When away from the office, dial the main number and when the auto attendant answers dial ⑦⑧⑧ to access the message center. When prompted, enter your mailbox number and password.

Message Summary

Once logged in you'll be told how many messages you have. For example you might hear "you have 2 new and 1 old message". If you don't have any messages you'll hear "you have no new messages" and main menu option 1 will be skipped.

Main Menu

Press ① for new/old messages

Press ② to change folders

Press ③ for advanced options

Press ⓪ for mailbox options

Press ＊ for help or ＃ to exit

① Message Folder Options

Press ③ for advanced options

Press ⑤ to repeat message

Press ⑥ to play the next message

Press ⑦ to delete/undelete

Press ⑧ to forward message

Or ⑨ to save this message

Press ＊ for help or ＃ to exit

Advanced Options

Press ① to send a reply ——————

Press ③ to hear the message envelope

Press ⑤ to leave a message

Press ＊ to return to the main menu

Message Envelope
Received at hh:mm, message from phone number …

② Change Folders

Press ⓪ for new messages

① for old messages

② for work messages

③ for family messages

④ for friends messages

or ＃ to cancel

③ Advanced Options

Press ⑤ to leave a message

Press ＊ to return to the main menu.

⓪ Mailbox Options

Press ① to record your unavailable message

Press ② to record your busy message

Press ③ to record your name

Press ④ to record your temporary greeting

Press ⑤ to change your password

Press ＊ to return to the main menu