AstA*UI

Asterisk™ **A**ppliance **U**ser **I**nterface
by
Firmix Software GmbH

and

Vdex-40

Voice **D**igital **Ex**change
by
TechnoCo Pty Ltd

technoCo
a disruptive company

Installation and Administration Manual

Revision 1.4

valid for FW Version 2.1.14

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Content

1 General

1.1 Copyright and Trademark Notices

1.2 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.
FIRMIX AND TECHNOCO PTY LTD (TECHNOCO) MAKES NO EXPRESS OR IMPLIED WARRANTY AGAINST UNLAWFUL OR UNAUTHORIZED USE OF VDEX OR ITS CAPABILITIES AND HEREBY DISCLAIMS ALL LIABILITY ARISING FROM SUCH USE. YOU AGREE TO INDEMNIFY, DEFEND, AND HOLD FIRMIX AND TECHNOCO HARMLESS FOR ANY UNAUTHORIZED OR FRAUDULENT USE OF VDEX.

1.3 Content, Variation and Notices

Due to ongoing product research and development that may occur in the life of this product:

- 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.

TechnoCo reserve the right to make changes to the product and product literature and promotional material without further notice.

1.4 Requirements

Vdex user interface operates under Firefox 2.0.0.11 or higher and Explorer 6.0 or higher. Telecom connection and active account is required for Telco services. Telecom 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

1.5 Safety and Compliance

Shock Hazard

Do not use this device and any connected devices such as telephones during periods of thunder and lightening activity 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 compliances and cause the product to malfunction.

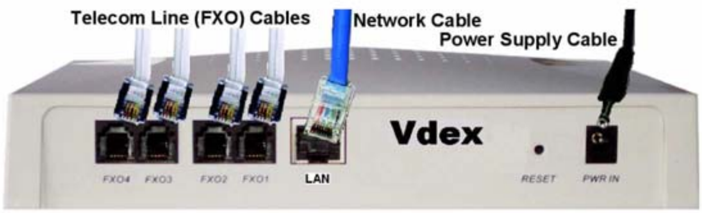
IMPORTANT NOTICE RE POWER FAILURE & EMERGENCY USE

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.

2 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.

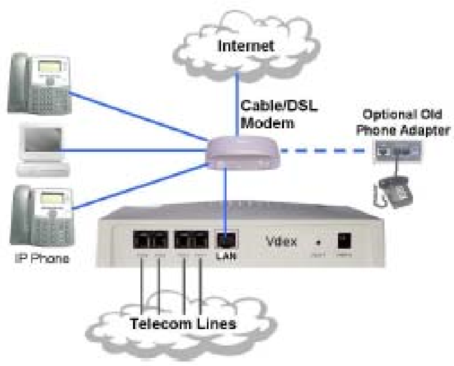
Remove the Vdex system, power supply, and network cable from the Vdex box. Connect the network cable to the LAN port on Vdex system and to an available port on your network router, switch, or hub. Connect up to four analog telephone lines to FXO ports 1- 4 as shown in the diagram:

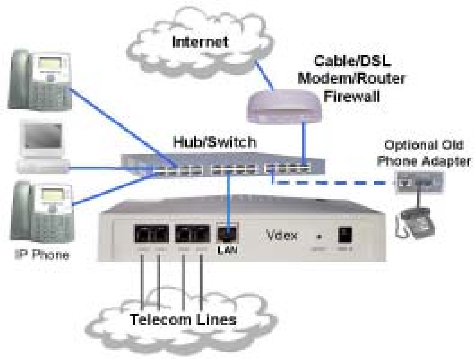


Please Note: You must connect at least one telephone line to FXO port 1 to dial 911 for emergencies by default. This can be changed later as needed (see outbound dialing rules) for details.

2.1 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.





3 Installation

The Vdex requires a static IP address. The Vdex default IP address is 192.168.0.157/24. If the Vdex is used for the first time or after a factory reset the IP address must be set according the target IP network.

One way to change the IP address of Vdex is to connect a telephone line to any FXO port and call the port. Once the port answers a message stating “The current Vdex IP address is 192.168.0.157. The current network mask is 255.255.255.0. Press 1 to change the IP address”. When entering the new IP address and network mask use the * key for dot and press # when finished. Entering the network mask can be done in both formats in quad-dotted decimal representation (e.g. 255.255.255.0) or in CIDR notation (e.g 24).

It is possible to dial again the FXO port and listen if the desired IP address has been entered correctly. After changing the IP address you are able to connect to Vdex via web browser and will get the AstA*UI wizard. Once the save button at the AstA*UI wizard page has been pressed the "Change IP address via FXO dial in" feature is deleted in the Vdex setup.

The second way to change the IP address of the Vdex is to run the Vdex AstA*UI wizard. To run the AstA*UI wizard you must have a PC configured for an IP address ranging from 192.168.0.x and 192.168.0.254 (except 192.168.0.157). 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 AstA*UI wizard. Using the AstA*UI wizard you can change the IP address.

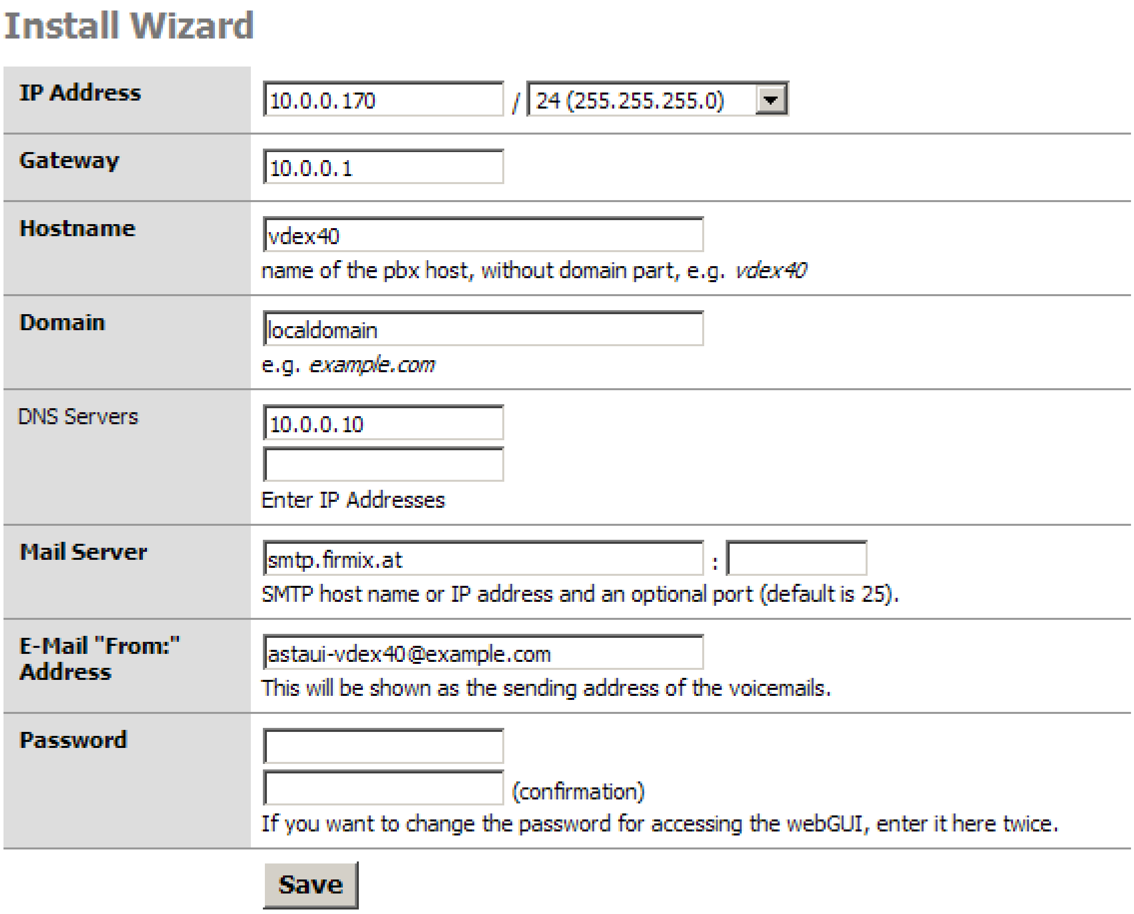
3.1 Vdex AstA*UI Wizard

The AstA*UI wizard comes up with the current settings. You must set all parameters asked at the wizard page.
Set the IP address, subnet mask (or leave it as it is) and gateway IP to be used by the Vdex system. Set the host name and domain name. If you're unsure of what to set as the host- and domainname retain the default values unchanged for now.

Assign the primary and secondary DNS settings to be used by the Vdex system. SIP won't really work without working DNS settings.

Set the SMTP server value to the hostname (or IP address) of the SMTP server to be used to email voicemail and recording files. 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.

Finally enter a password and to confirm the password to use the AstA*UI. 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 wizard.
If the IP address has been changed the browser will be redirected to the new IP address automatically. Of course your browser must be able to reach the new IP address



3.2 Basic Web Interface Functions

Asta*UI offers in many section the possibility to add, delete or edit entries. This is shown by following icons:

Button e	Edit entry. Open new page to edit parameters.
Button +	Add a new entry (e.g. provider, phone)
Button x	Delete entry and the related parameter.
Button speaker:	Upload soundfile to the PC and open it with the appropriate player software.

3.3 Step by step configuration recipe

In order to keep the setup expenses as low as possible we recommend to to configure the Vdex the subsequent order. AstA*UI

(1) Configure Network interface parameters

(2) Check if a system update is available

(3) Configure general setup section

(4) Upload music-on-hold and attendant sound files

(5) Configure SIP/IAX/FXO Providers

(6) Define time segments

(7) Create outgoing callrule table

(8) Configure SIP/IAX phones

(9) Configure DISA accounts

(10)Configure external phones

(11)Define hunt groups and user groups

(12)Create attendants

(13)Create incoming callrules

4 System Configuration

4.1 General Setup

Parameter	Default Value	Description
Hostname	Vdex	A hostname must be set. The default hostname can be changed.
Domain	localdomain	A domainname must be set. The default domainname can be changed.
User Name	admin	Enter the AstA*UI username for the administrator login
Password	lb3wn3hg	The password is used for authenticating the AstA*UI administrator: It is strongly recommended to change the default password. This password is also used as root password for the remote access via SSH.
Indications Tones	Austria	Indication tones have to be generated by the Vdex in various tones situations. Every country uses different indication tones with various frequencies and cadences for e.g. ringback-tone, busy-tone, dial-tone etc.. Select from a list of countries the desired indications tonezone.
Timezone	Austria	Select the time zone in which the Vdex is located. This is important for accurate time dependent call routing and proper time and date information of Logs.
Time Server (NTP)		The Vdex must get time and date from any NTP server. A timeserver must be set. Using an non functional timeserver would cause a malfunction of time dependent callrules and wrong timestamps in all logs. Time and date of the last successful synchronization between Vdex and the timeserver is shown at the summary page.
Default G.711 Codec	A-law	Select the default G.711 codec version (a-law or µ-law) Codec. Using phones with a different G711 codec version would work, but would need transcoding resources and would lower the voice quality.
G.723 Encode Rate	high-rate	Select the bit rate for the G.723.1 codec. High-rate means 6,3kb/s. Low-rate means 5,3kb/s.
FX Impedance Setting	TBR21	Select the impedance for all FXO interfaces. The necessary line impedance depends on the country where the Vdex is connected to the analogue telephone lines. TBR21 means a complex line termination according to the standard TBR21 of the European Telecommunication Standard Institute (ETSI). TBR21 must be set when the Vdex is installed in any European country or e.g. Australia. 600 Ohm must be set in e.g. USA.
Allow Call Pickups	yes	If this option is activated, users are able to dial *8xyz (xyz is the number of the ringing extension) or *8 (for pickup any call within the user group) to pickup the currently incoming call from another extension.

4.2 Network Interfaces

Parameter	Default Value	Description
Device		This is the hardware address (MAC address) of the LAN port. The MAC address is hard coded and not changeable.
IP Address	192.168.0.157/24	This is the fixed IP address for the LAN port (WAN port is not used) and the related network mask. The IP address setup changes only after a reboot.
Gateway		The default gateway IP address must be set.
DNS Server		The DNS server IP address must be set. This is necessary to resolve domain names of at least SIP- and IAX-providers, STUN servers, SMTP servers, etc.. It is possible to define up to 3 different DNS server.
Topology	Public IP Address	This value must be set according to the used network address topology. Public IP address means that the above given Vdex IP address is a routable, static, public IP address. NAT+static public IP address means that the Vdex is behind a NAT router, which has a routable, static, public IP address. The above given Vdex IP address is a private one. The public NAT router IP address must be set in the field "Public IP Address". Use a stun server means that the Vdex is behind a NAT router, which has a dynamical public IP address. The above given Vdex IP address is a private one. The public NAT router IP address must be discovered by a STUN server. The STUN server location must be set in the field "STUN Server".

4.3 System Update

Before placing the system into production you should make sure it is running the latest firmware. Start the update process to both to check if the system is already running with the latest firmware and to upgrade to the latest firmware. The system uses a package management system to ensure that only new parts of the firmware will be uploaded and installed. This reduces the upgrade time to a minimum.

Parameter	Default Value	Description
Repository URL		To update the Vdex system enter the web address where the firmware will be obtained and store this address with the "Save" button. By clicking the "Update System" button the firmware upgrade will start. This may take several minutes depending on the speed of the Internet connection and the number of packages that will be updated. Reboot the system when the upgrade completed. Ask your distributor for the appropriate URL.

4.4 Backup/Restore

The Vdex system offers a backup and restore mechanism. In the event of a hardware change the backup file can be easily restored without loosing configuration data. The backup will generate a “.tar.gz” file of the whole configuration including all uploaded sound files. Excluded from backup are voicemail messages. The backup process can take several minutes depending on the configuration and soundfile sizes.

For restore locate an AstA*UI backup file and click the button to restore the configuration. The restore process can take several minutes depending on the configuration and soundfile sizes. The Vdex will reboot automatically after successful restore action. The restoring of the system will first erase your existing configuration, so be careful about this step.

4.5 Factory reset

Performing a factory reset will load a default configuration file and the Vdex will be rebooted. The entire system configuration will be overwritten with the default values. The LAN IP address will be set to 192.168.0.157 and the password will be set to 'lb3wn3hg'. The factory reset will take a while.

4.6 Reboot

Performing a reboot will take up to 2 minutes. The Vdex is ready again when the RUN light is blinking. All active calls will be terminated.

5 Advanced Configuration

5.1 Extension Prefixes

Mailbox, attendant and voicebox extensions use common prefixes for their extensions. The prefix length can be 1 – 4 digits. It is recommended to use prefixes with with the same leading digits since this strategy wastes the smallest numbering space.

Parameter	Default Value	Description
Mailbox Prefix	8889	The mailbox prefix is inherent part of all mailbox extensions. Mailbox extensions consist of a prefix and a freely selectable number for each phone. The prefix length can range up to 4 digits.
Attendant Prefix	8888	The attendant prefix is inherent part of all attendant extensions. Attendant extensions consist of a prefix and a freely selectable number for each attendant. The prefix length can range up to 4 digits.
Personal Conference Room Prefix	8887	Personal conference room prefix is inherent part of all personal conference room extensions. Personal conference room extensions consist of a prefix and a freely selectable number for each phone. The prefix length can range up to 4 digits.
Handset Recording Prefix	8886	10 extensions for handset recording can be reached by dialing the handset recording prefix followed by one digit (0 – 9).

5.2 RTP Settings

Parameter	Default Value	Description
RTP Port Range	10000 - 20000	The RTP ports are used for sending and receiving media data. You must specify a reasonable port range so that you have enough ports for all open calls. Set the port range which the RTP stream will use. The port range can be defined between 1024 and 65535. The default

		port range is 10000 – 20000. The RTP ports for the Vdex must not overlap with any other ports in the system. Changing the RTP ports will drop all current calls.
Type of Service	Decimal 184	Select the Type-of-Service qualifier for the audio and video RTP packets The numerical value is the contents of the Type-of-Service-field (in the IP header). Default is *EF (184 - Expedited Forward)*. Use it unless your most often used provider explicitly tells you something different.

5.3 SIP Settings

Parameter	Default Value	Description
Binding Port	5060	The port that Vdex uses for SIP communication.
Registration Expiration Timeouts	min: 60 s max: 3600 s default: 120 s	The minimum and maximum of seconds that incoming registrations remain valid. The number of seconds that outgoing registrations (and incoming registration without given expiration time) remain valid.
DNS Service Records	enabled	This option allows to disable DNS SRV lookups. It is recommended to leave DNS SRV lookups activated.
Type of Service	Decimal 184	Select the Type-of-Service qualifier for the SIP packets. The numerical value is the contents of the Type-of-Service-field (in the IP header). Default is *EF (184 - Expedited Forward)*. Use it unless your most often used provider explicitly tells you something different.

5.4 IAX Settings

Parameter	Default Value	Description
Binding Port	4569	The port that Vdex uses for IAX communication.
Jitterbuffer	Disabled	The jitterbuffer can be enabled for IAX connections that are terminated by the Vdex (e.g. calls to FXO ports)
Force	Disabled	When the jitterbuffer is enabled and this option is checked, the jitterbuffer is also enabled for IAX calls between 2 IAX clients.

5.5 Accounts

Vdex allows client programs outside of Vdex to communicate with Asterisk via the Asterisk Manager Interface (AMI). Please refer to the Asterisk documentation for more details.

Set the AMI users access permissions and allow AMI clients the use of various commands.

Parameter	Default Value	Description
Username		The username for the AMI client
Secret		The password for this account
Deny Network	/ 0.0.0.0	Enter the network IP address (IP address / network mask) to

		deny the access from this network. Use network 0.0.0.0 / 0.0.0.0 to deny access from all networks.
Permit Network	/ 255.255.255.255	Enter the network IP address (IP address / network mask) to allow the access from this network. Use network 0.0.0.0 / 0.0.0.0 to permit access from all networks.
Permissions	unchecked	Allow the use for the selected AMI commands

5.6 GUI Options

Select with the navigation options if the system-, advanced-, and diagnostics folder in the left main menu should be expanded.

To reduce the shown options in the left main menu ist is possible to

- hide the SIP providers, phones and SIP options
- hide the IAX providers, phones and IAX options
- hide the analogue providers

5.6.1 SSH Server

The Vdex incorporates a SSH server. If the SSH server is enabled, you can login as user "root" with the password as defined in the "General Setup" section. The SSH server can be used for a secure remote access to the Vdex. Once a SSH connection has been established you can use a tunnel to reach the AstA*UI webserver.

Example: Secure remote connection from a Windows client
Use the SSH client PuTTY to reach the remote Vdex via tunnel (configure PuTTY tunnel e.g. source port 10000, destination 127.0.0.1:80). Login to Vdex with user "root". Once the SSH connection has been established navigate at your local browser to http://localhost:10000 and you will reach the AstA*UI.

6 Provider Configuration

6.1.1 SIP Provider

The SIP trunk setup requirements can vary greatly between SIP provider. To ensure a proper configuration please obtain the needed configuration from the provider.

Parameter	Default Value	Description
Name		Descriptive name of the provider
Username		Enter the username given by the provider
Load Provider Defaults		Settings of some SIP providers are stored in the Vdex. If the desired provider can be found in the provider list, then the corresponding parameters of the "Registration Info" section and "Miscellaneous" section can be loaded.
Authorization User		Enter the separate authorization user necessary for some providers. Leave the field empty when the username and authorization user are identical.
Secret		Enter the SIP account's password given by the provider.
Enable Direct Dial-In	deactivated	Some SIP providers support Direct-Dial-In. This feature allows external callers to reach an extension directly by dialing the main number added by the extension number. Configure which extensions should be reachable directly at each extension section.

Main Number	none	If Direct-Dial-In is supported the main number must be entered. Please ask the provider which number format is used (e.g. 004317890849 or +4317890849 or 017890849 etc.). Numbers attached to the main number will be taken as extension information.
Proxy		Enter the SIP hostname or IP address of the provider. For a reliable voice service we recommend to use the host IP address and not the host name. This ensures service availability also in the case of a DNS problem or the unintentional use of DynDNS hostnames. If the provider is using a different SIP port than 5060 this port can be optional entered in the Host port field.
Outbound Proxy		SIP outbound proxy hostname or IP address and optional port (default is 5060). If this field is set, then all SIP packets are sent to the IP address this field resolves to and incoming SIP packets must originate from this IP address too.
Inbound Gateway/Proxy Pool		Most providers deliver calls to their subscribers ONLY from the IP address where the subscriber has registered. But some providers use a pool of gateways/proxies and an incoming call can come from any one of them. In this case it is necessary to add ALL the IP addresses in this pool to this list to be able to correctly match incoming calls to this provider. When using domain/host names here, take special care that these have a fixed binding to an IP address and do not use rotating DNS or SRV lookup.
Registration		Select if this account should not register at the provider.
Registrar		Domainname, hostname or IP address and optional port (default is 5060). where Vdex is registering. "Proxy" and "Outbound Proxy" are used for registration if this field is left empty. Note that "Outbound Proxy" has no effect on the registration process if this field is set.
From Domain		Domain name used in the "From" header of calls to the provider. 'Proxy' is used if this field is left empty.
From User	Username	The username used in the "From" header of calls to the provider. "Username" is the the recommended option as it is RFC-compliant. However, some SIP providers choose to be more creative so you can send other values in the SIP "From" header. "Main Number with Extension" requires "direct dial-in" enabled providers above.
Fixed Outgoing Caller ID		Enter a valid E.164 telephone number. This number will be used as caller ID. The caller ID will be set to the username when this field is empty.
Language	English	Set the audio prompt language for this account. Choose between German and English.
DTMF mode	rfc2833	Set the DTMF transfer mode for this account according the provider supported method. Choose between RFC2833 and SIP Info mode.
Qualify Timeout	2 sec	If the SIP provider does not response the qualify message within the selected timeout, the SIP provider status will be set to unreachable (see status summary page). If the provider does not support SIP options set the qualify timeout to off.
Incoming Caller ID overrides		If this option is selected, then the caller name will be replaced by the calling number
Audio Codecs	A-law	Allow one or more audio codec types for calls to and from this

		SIP provider. The listing of the enabled codecs (top down) corresponds with the codec preference order.
Video Codecs	none	Allow one or more video codec types for calls to and from this SIP provider. The listing of the enabled codecs (top down) corresponds with the codec preference order.

Example Neotel

Providers: SIP: Edit Account

Name	Neotel mit DDI Descriptive name of this provider.
Load Provider Defaults	All countries / Select the provider / Set Defaults Load the provider's parameters and overwrite all of the values below
Username	+43720123456
Authorization User	Some providers require a seperate authorization username. Defaults to username entered above.
Secret	●●●●●●●●● This account's password.
Enable Direct Dial-In	☑ Activate DDI (and do not longer ignore extensions) Direct Dail-in uses the extension to call a phone directly. You must also enable that feature for each directly reachable extensions.
Main Number	43720123456 Enter the main number so that the called phones can call back. The extension of the phone is appended.

Registration Info	
Proxy	demo.neotel.at : SIP proxy hostname or IP address and optional port (default is 5060).
Outbound Proxy	p01.neotel.at : SIP outbound proxy hostname or IP address and optional port (default is 5060).
Inbound Gateway/Proxy Pool	Accept provider calls also from these hosts and IP addresses
Registration	☐ Do not register with this provider.
Registrar	: Domainname, hostname or IP address and optional port (default is 5060) where Vdex is registering."Proxy" or "Outbound Proxy" are used for registration if this field is left empty.
From Domain	Domain name used in the "From" header of calls to the provider. 'Proxy' is used if this field is left empty.
From User	Username The username used in the "From" header of calls to the provider. "Username" is the the recommended option as it is RFC-compliant. However, some SIP providers choose to be more creative so you can send other values in the SIP "From" header. "Main Number" and "Main Number with Extension" requires "direct dial-in" enabled providers above.

Miscellaneous	
Language	German Audio prompts will be played back in the selected language for this account.
DTMF Mode	rfc2833
Qualify Timeout	4 seconds If activated, we check regularly using the above timeout if the SIP provider is reachable because we do not call known unreachable SIP providers.
Incoming Caller ID Overrides	☐ Replace calling name with calling number.

Example Engin

Providers: SIP: Edit Account

Name	Engin Descriptive name of this provider.
Load Provider Defaults	Australia · Engin · Set Defaults Load the provider's parameters and overwrite all of the values below
Username	0391234567
Authorization User	Some providers require a seperate authorization username. Defaults to username entered above.
Secret	●●●●●●●●●●● This account's password.
Enable Direct Dial-In	☐ Activate DDI (and do not longer ignore extensions) Direct Dail-in uses the extension to call a phone directly. You must also enable that feature for each directly reachable extensions.
Main Number	Enter the main number so that the called phones can call back. The extension of the phone is appended.

Registration Info

Proxy	mel.byo.engin.com.au : SIP proxy hostname or IP address and optional port (default is 5060).
Outbound Proxy	: SIP outbound proxy hostname or IP address and optional port (default is 5060).
Inbound Gateway/Proxy Pool	syd.byo.engin.com.au Accept provider calls also from these hosts and IP addresses
Registration	☐ Do not register with this provider.
Registrar	: Domainname, hostname or IP address and optional port (default is 5060) where Vdex is registering."Proxy" or "Outbound Proxy" are used for registration if this field is left empty.
From Domain	voice.mibroadband.com.au Domain name used in the "From" header of calls to the provider. 'Proxy' is used if this field is left empty.
From User	Username The username used in the "From" header of calls to the provider. "Username" is the the recommended option as it is RFC-compliant. However, some SIP providers choose to be more creative so you can send other values in the SIP "From" header. "Main Number" and "Main Number with Extension" requires "direct dial-in" enabled providers above.

Miscellaneous

Language	English (US) Audio prompts will be played back in the selected language for this account.
DTMF Mode	rfc2833
Qualify Timeout	2 seconds If activated, we check regularly using the above timeout if the SIP provider is reachable because we do not call known unreachable SIP providers.
Incoming Caller ID Overrides	☐ Replace calling name with calling number.

6.1.2 IAX Provider

Parameter	Default Value	Description
Name		Descriptive name of the provider
Username		Enter the username given by the provider
From User		Enter the separate authorization user necessary for some providers. Leave the field empty when the username and authorization user are identical.
Secret	MD5	Select plaintext or MD5 password transmission used by the provider.
Host		Enter the SIP proxy host URL or IP address of the provider. If the provider is using a different SIP port than 4569 this port can be optional entered in the Host port field.
Fixed Outgoing Caller ID		Enter a valid E.164 telephone number. This number will be aller ID used as caller ID. The caller ID will be set to the username when this field is empty.
Language	English	Set the audio prompt language for this account. Choose between German and English.
DTMF mode	rfc2833	et the DTMF transfer mode for this account according the provider supported method. Choose between RFC2833 and SIP Info mode.
Registration	register	Select if this account should not register at the provider (for advanced purposes)
Qualify Timeout	2 sec	If the SIP provider does not response the qualify message within the selected timeout, the SIP provider status will be set to unreachable (see status summary page). If the provider does not support SIP options set the qualify timeout to off.
Incoming Caller ID overrides	unchecked	If the this option is selected, then the caller name will be replaced by the calling number
Audio Codecs	A-law	Allow one or more audio codec types for calls to and from this SIP provider. The listing of the enabled codecs (top down) corresponds with the preference order.
Video Codecs	none	Allow one or more video codec types for calls to and from this SIP provider. The listing of the enabled codecs (top down) corresponds with the preference order.

6.1.3 Analog Line Provider

Some parameters of this section can be reset to the value of the last stored change (e.g. “reset to 5 s”). This feature helps to correct changes to the current value before activation by pressing the „Save“ button.

Parameter	Default Value	Description
Enable this provider	yes	Disabled FXO ports can not be used for any configuration purpose.
Name		Descriptive name of the provider.
Number		Telephone number given by the provider.
Load saved parameter sets		Load a parameter set of the subsequent parameters to reduce the configuration work. Select from parameter sets for USA, Australia or Austria. Alternatively load the already configured parameter set from another port (copy from FXO) or from a user defined parameter set file (user defined #).
Language	English	Set the audio prompt language for this account. Choose between German and English.
Default Incoming Caller ID	FXO-n	If the caller ID of an incoming FXO call is not presented or can not be detected the default caller ID will be used
Wait for Dial tone	Yes	It is recommended that the dial tone must be detected before dialing.
Caller ID Detection	Yes	Enables on hook caller ID detection (off hook caller ID detection is not supported).
Caller ID methode	FSK	Select between FSK and DTMF caller ID detection.
PSTN Gain	0 db	If necessary, adjust the volume to the PSTN network. The volume can be increased or reduced by changing the gain (-xx dB means reduce, xx dB means increase).
VoIP Gain	0 dB	If necessary, adjust the volume from the PSTN network to any Vdex application (e.g. voicemail) and VoIP destination. The volume can be increased or reduced by changing the gain (-xx dB means reduce, xx dB means increase).
Dial Delay	2 s	Outgoing PSTN calls start the dialing process by changing the status to off hook The dial delay specifies a delay between off hook and sending the first dial digit. If the option „Wait-for-Dial tone“ is active, then the dial delay starts with the dial tone recognition.
Dial Digit Length	0.1 s	Specifies the minimum period of time a ring signal must be indicated to be qualified as ring. If the ring signal is shorter than the given period, this ring signal will not qualified as ring.
Dial Digit Pause	0 s	A pause between two dial digits can be set.
Ring Validation Time	0.13 s	Specifies the minimum period of time a ring signal must be indicated to be qualified as ring. If the ring signal is shorter than the given period, this ring signal will not qualified as ring.
Ring Minimum Pause	0.13 s	Specifies the minimum period of time without a ring signal (after a validated ring) which qualifies as ring pause.
Ring Timeout	5 s	Incoming PSTN calls can be cancelled by the remote caller before the call is answered by a Vdex phone or application. In this case the Vdex can indicate this call termination only when the ring pause of the ring exceeds the specified ring timeout. The ring timeout must be longer than the ring pause

Parameter	Default Value	Description
		of the PSTN provider ring
Minimum On hook Time	0.3 s	Specifies the period of time the line needs to be on hook, before the line can used again. If the given value is too small the central office might mistake a fast off hook-on hook-off hook sequence with a hook-flash.
Minimum Off hook Time	0.3 s	Specifies the period of time the line needs to be off hook, before it can go on hook again. If the given value is too small the central office might mistake a fast off hook-on hook-off hook sequence with a hook-flash.

Disconnect Detection

A call will be terminated when

- the local phone terminates the call or
- voice inactivity is detected for a periode of time (if activated) or
- the Vdex receives busy tone at the analogue line or
- the Vdex receives a termination indication at the analogue line. This call termination indication varies between countries and providers (CPC or polarity reversal)

Parameter	Default Value	Description
Detect CPC	yes	Enable loop interruption detection. Some PSTN providers interrupt the line power for a specified time when the remote side hangs up the call.
Minimum CPC Duration	0.2 s	Minimum period of time with loop interruption that qualifies as CPC (if CPC detection is enabled).
Detect Polarity Reversal	yes	Enable polarity reversal. Some PSTN providers change the line polarity when the remote side hangs up the call.
Detect Busy Tone	yes	Enable busy tone detection. The busy tone must be configured in the „Tone Detection Duration“ section. When enabled, a call will be terminated by detecting by detecting the busy tone.
Detect PSTN Silence	yes	Enable PSTN silence detection. No voice activity on the PSTN line could mean that no active call is established. The PSTN silence detection feature is used to detect this state and to hang up the call. If the energy line level is lower than the PSTN silence threshold level for longer as the PSTN silence duration than the call will be terminated.
PSTN Silence Duration	30 s	Period of time with a energy line level lower than the configured PSTN silence threshold that qualifies as PSTN silence.
PSTN Silence Threshold	-38dBm0	Energy threshold for PSTN transmit and receive for PSTN silence detection.

Tone Detection Duration (PSTN indication tones)

PSTN tones are a sequence of tones and silence (pauses). For a proper operation the Vdex has to distinguish the different types of PSTN tones by detecting the duration of tone and pause. For each PSTN tone the tone „On“ time range and „Off“ time range (pause) has to be configured. A duration range should be set because provider PSTN tones can not precise. Tone and pause duration can be configured for dial tone, busy tone and unobtainable tone.

Continuous tone: A continuous tone detection is selected when the pause duration is set to 0 seconds.

Tone and pause duration can set by using the sliders

7 Phone Configuration

7.1 SIP / IAX phone accounts

Parameter	Default Value	Description
Extension		The extension number can have up to 4 digits. The extension number is also the SIP username.
Caller ID		Specifies the caller ID name. Caller ID strings are limited to maximal 79 alphanumeric characters.
Secret		The password for this account.
Secret Transmission	MD5	Only for IAX phones. Select plaintext or MD5 password transmission.
User email address		The email address of the person who is the owner of this account. The email address is mandatory when voicemail-to-email and conference recording are used.
Language	English	Vdex audio prompts will be played back in the selected language. Audio prompts are used used in various situation by the Vdex, e.g. voice mail menu, announcements, applications like voicemail, DISA, etc.
DTMF mode	rfc2833	Only for SIP phones. Set the DTMF transfer mode for this phone. Choose between RFC2833 and SIP Info mode.
Qualify Timeout	2 seconds	The Vdex sends regularly qualify information to the phone. If the phone does not respond within the selected timeout the phone state will be set to unreachable (see info at the status summery page). Calls to phones in the unreachable state will reach the voice mailbox (if activated). If the qualify timeout is set to off, the phone state is unmonitored. Calls to unmonitored phones will always sent to the phone, no matter if it the last qualify was ok or not.
Call limit	2 calls	Only for SIP phones. The Vdex can handle the selected maximum number of concurrent incoming and outgoing calls for this phone.
Direct Direct Dial	checked	Allow to reach this extension directly through all SIP providers which support Direct-Dial-In and if it is activated.
Publish this extension	checked	Publish the extension on outgoing calls for all SIP providers which support Direct-Dial-In and if it is activated
Outgoing callrule Table		Select one of the callrule tables for this phone. Phones with permissions for internal calls only (which includes also external phones) must be set to „No outgoing calls allowed“.

Huntgroup Membership	none	Select, if the phone should be a member of one or more hunt groups. Huntgroups are defined in the Dialplan/Huntgroups section.
Usergroup Membership	none	Select, if the phone should be a member in a user group. A phone can be a member only in one user group. User groups are defined in the Dialplan/User group section.
Audio Codecs	µ-law	Enable one or more audio codecs for this phone. The order of the enabled codecs is used for the codec selection in the SIP call establish process (top codec: highest priority).
Video Codecs	none	Enable one or more video codecs for this phone. The order of the enabled codecs will be used for the codec selection in the SIP call establish process (top codec: highest priority).

7.1.1 Voicemail

Parameter	Default Value	Description
Enable Voicemail	no	Activate the voice mailbox for this account.
Extension	8889.	Voicemailbox extension number for this phone. The voicemail extension number starts with a prefix followed by a any number. The prefix is common for all voice mailboxes and can be changed in the System/Extension Prefixes section. The voice mailbox can be reached by any internal phone dialling the defined extension e.g. 8889-100. Voice mailboxes can also be reached over FXO, SIP and IAX provider trunks by using an incoming callrule configuration and/or attendant configuration.
No Answer Timeout		The voicemail timeout defines how long the Vdex will wait until it redirects the call to the mailbox.
Mailbox PIN		It is mandatory to define a mailbox PIN. The mailbox owner can access the voice mailbox from his phone The voicemail application asks the caller for the phone extension and mailbox PIN. The number of the PIN digits is common for all voice mailboxes and can be defined in the Services/Voicemail Section.
Send Voicemails	none	Voice mails can be notified or transmitted via email. Preconditions are a working mail system and a configured user email address. Select between: None: No email will be sent when a new voice mail message has been stored on the mailbox. Notification only: An email will be sent to to the user to inform that a new message has been stored on the mailbox. Soundfile attached: An email will be sent to the user with the message attached as soundfile in wav format. The message retains in the Vdex memory. Soundfile attached and deleted: An email will be sent to the user with the message attached as soundfile in the wav format. The message will be deleted from the Vdex memory. It is recommended to use the setting "Soundfile attached and deleted" since this option assures that voicemail messages do not retain unintentionally on the Vdex over a long period of

Parameter	Default Value	Description
		time.
Public Direct Dial	unchecked	Allow unregistered phones to reach this extension by the SIP/IAX2 URL.

7.1.2 Personal Conference Room

Recorded soundfiles are sent to the phone owner by email. This assures that nobody else than the phone owner can acquire the conference content. Furthermore it is possible that the conference starts only when the phone owner (also called “the leader”) enters the conference room

Parameter	Default Value	Description
Enable	No	Activate the personal conference room for this phone.
Record Conferences		Activate conference recording. Recording soundfiles will be sent to the user email address. Please note that there is no caller announcement that the conference will be recorded.
PIN for the Room		PIN with 1 to 4 digits length can be set. Leave the field empty for no PIN. Callers who reach the conference rooms with activated PIN will be asked for the PIN code. Calls from the phone which is associated to the conference room with not be asked for the PIN code (conference leader).
Wait for you		Activated: Callers who reach the conference room hears the "Music on Hold" soundfiles until the conference owner (the leader) arrives.
Announce joining and leaving users	yes	This option makes the arrival of new users and leaving of users audible by play back of prompts into the conference.
Public Direct Dial	unchecked	Allow unregistered phones to reach the conference room by the SIP/IAX2 URL.

7.2 External Phones

External phones can dial in the Vdex and act like internal phones. Calls from external phones will be authenticated by the caller-ID. After a positive caller-ID verification the external phone the caller hears the internal Vdex dial tone and can establish outgoing calls according to the selected outgoing callrule.

Phones which can be reached by any FXO-, SIP- or IAX provider line can be defined as outside Vdex extension. Calls from external phones must present the caller-ID to the Vdex. The caller-ID check is the only authentication method for the external phones.

If you do not trust the caller-ID, use the DISA feature instead. Additional to the caller-ID, DISA requires a PIN code for authentication.

Parameter	Default Value	Description
Extension		Set the extension to reach the external phone. Calls from the external phone to internal phones are displayed with this extension number.
Name		Descriptive name for this phone.
Provider		Select the provider for outgoing calls to and incoming calls for the external phone. Only one provider can be used to reach an external phone. If external phones are reached via FXO lines then select the

		desired FXO ports. Vdex will use the first free FXO port to reach the external phone (please check if the dial string for all FXO port is the same).
Dialstring		Enter the number to dial the external phone (without any Vdex prefix). Please check at your provider which numbering plan is used (e.g. with/without city code)
Caller ID		Enter the caller ID number which the external phone displays at the Vdex. If you are not sure make a call from the external phone any internal phone and see the caller ID number (e.g. with/without country code, with/without city code, etc.).
Time Segment		The selected time segment defines when the external phone can be reached. The time segment selection does not affect incoming calls from the external phone.
Public Direct Dial	unchecked	Allow unregistered phones to reach this extension by the SIP/IAX2 URL.
Outgoing Callrule Table		External phones which has been authenticated are able to establish calls according the selected callrule table.
User Email Address		The email address of the owner of this account. The email address is mandatory when voicemail-to-email will be used.
Voicemail		
Enable Voicemail	No	Activate the voice mailbox for this account. It is not possible to activate the voice mailbox when the external phone will be reached via FXO port (since FXO ports can not indicate the answer state).
Extension	8889.	The Voicemailbox extension for this phone. The voicemail extension starts with a prefix followed by any number. The prefix is common for all voice mailboxes and can be changed in the System/Extension Prefixes section. The voice mailbox can be reached by any internal phone dialling the defined extension e.g. 8889100. Voice mailboxes can also be reached over FXO, SIP and IAX provider trunks by using incoming callrule configuration and/or attendant configuration.
No Answer Timeout	10	The voicemail timeout defines how long the Vdex will wait until it redirects the call to the mailbox.
Mailbox PIN		A voice mailbox PIN must be defined. The voicemail application asks the caller for the phone extension and mailbox PIN. The number of the PIN digits is common for all voice mailboxes and can be defined in the Services/Voicemail Section.
Send Voicemails	none	Voice mails can be notified or transmitted via email. Preconditions are a working mail system and a configured user email address. Select between: None: No email will be sent when a new voice mail message has been stored on the mailbox. Notification only: An email will be sent to to the user to inform that a new message has been stored on the mailbox. Soundfile attached: An email will be sent to the user with the message attached as soundfile in wav format. The message retains in the Vdex memory. Soundfile attached and deleted: An emaill will be sent to the

		user with the message attached as soundfile in the wav format. The message will be deleted from the Vdex memory. It is recommended to use the setting "Soundfile attached and deleted" since this option assures that voicemail messages do not retain unintentionally on the Vdex over a long period of time.
Public Direct Dial	unchecked	Allow unregistered phones to reach this extension by the SIP/IAX2 URL.

7.3 Call Transfer by External Phones

External phones can transfer calls to other Vdex phones by using DTMF tones.

Attendent call transfer (*5)
Press the buttons * and 5 within 2 seconds. Vdex plays back the announcement "Transfer". The caller will hear the on hold music. You have 4 seconds to enter the extension number. Now you can dial the desired extension and talk to your colleague. When the external phone hangs up the call, the call transfer to the desired extension is performed.

Blind transfer (*4)
Press the buttons * and 5 within 2 seconds. Vdex plays back the announcement "Transfer". The caller will hear the on hold music. You have 4 seconds to enter the extension number. Now you can dial the desired extension. The call will be transferred to this extension immediately.

Precondition for initiating a call transfer at the external phone is a proper DTMF transmission between external phone and Vdex.

7.4 Fax Adapter

Analogue fax machines can communicate with the Vdex by using an analogue terminal adapter (ATA), e.g. Linksys SPA2102. Fax adapter extensions can have only 1 call at the same time. Calls to a busy fax adapter extension will get the busy tone.

Parameter	Default Value	Description
Extension		The extension number can have up to 4 digits. The extension number is also the SIP username.
Caller ID		Specifies the caller ID name. Caller ID strings are limited to maximal 79 alphanumeric characters.
Secret		The password for this account.
Enable T.38	Yes	If you receive fax via analogue provider lines T.38 support must be disabled. If T.38 support is disabled, fax transmission will be done by using the G.711 codec (64kbps). Please ask your SIP provider which method is used.
DTMF mode	rfc2833	Set the DTMF transfer mode for this phone. Choose between RFC2833 and SIP Info mode.
Qualify Timeout	2 seconds	The Vdex sends regularly qualify information to the ATA. If the ATA does not respond within the selected timeout the phone state will be set to unreachable (see info at the status summery page). Calls to ATAs in the unreachable state will get the busy tone. If the qualify timeout is set to off, the phone state is unmonitored. Calls to unmonitored phones will always sent to the ATA, no matter if it the last qualify was ok or not.
Direct Direct Dial	checked	Allow to reach this extension directly through all SIP providers

		which support Direct-Dial-In and if it is activated.
Publish this extension	checked	Publish the extension on outgoing calls for all SIP providers which support Direct-Dial-In and if it is activated
Outgoing callrule Table		Select one of the callrule tables for this ATA . ATAs with permissions for internal calls only (which includes also external phones) must be set to „No outgoing calls allowed“.

8 Dialplan Configuration

8.1 Time Segments

Time segments are defined ranges of time that will be used to route calls appropriately. They allow you to define business hours, weekends, holidays, etc., so that you will be able to set up callrules based on these. There is one default time segment already available on the system for work time which can also be modified or deleted. By using time segments you can do things like send all calls that come in after 7pm on workdays directly to voice mail, route calls to a live operator when they are available and an auto attendant IVR otherwise, or route your extension’s calls to your mobile phone if its during business hours and you’re away from your desk. Time segments are used in the Vdex configuration to specify when

- external phones can be reached
- outgoing callrules should be active
- incoming callrules should be active
- when certain extensions can be reached via attendants

Time ranges can be overlapped by various time segments. When time segments with overlapped time ranges are used then the first listed/used time segment will match the rule.

Example:

Dialplan: Incoming Callrules

Name	Enabled	Time Segment	Caller ID	Provider	Target
FXO in worktime	yes	worktime	Any	BT FXO line 1, BT FXO line 2	Attendant <81> enter extension
FXO in off time	yes	Any Time	Any	BT FXO line 1, BT FXO line 2	Attendant <82> off time message

The check of an incoming call starts with the top entry in the incoming callrule table. If the call arrives during work time it will be forwarded to attendant 81. If the call arrives outside work time then the 2nd callrule will match and the call will be forwarded to attendant 82.

8.2 User Groups

User groups are phones with common actions for incoming calls in the phone states no-answer and busy. In many scenarios calls should be forwarded to a team secretary when the called phone is busy or not answering. Precondition for the user group call forwarding is a deactivated user voice mailbox. It it also possible to forward non-answered or busy calls to an attendant for further call treatment. Phones can be member of only one user group.

Call pickup handling for user group member must be configured at the user phones. Vdex sends BLF (busy lamp field) notifications to all user phones. Phones that support status monitoring via BLF are able to display the phone state (i.e. busy, idle, ringing) of other phones. Usually phones are configured to pickup only calls from their own group.

User group members can perform an undirected pickup calls by dialling *8.This will pickup a call ringing at any extension within the user group. If more than one user group extensions are ringing it is undefined which call will be taken.

Parameter	Default Value	Description
Name		Enter the group name
No-Answer Timeout		Select the number of seconds to wait for answering the call.
Action on No-Answer		Select the forward destination for calls which are not answered.
Action on busy		Select the forward destination when the phone is busy.
Group Member		Select the phones for the group. Phones listed at the non-member side are phones which are free for user group selection. Once a phone is assigned a user group it will not be listed as free non-member phone.

8.3 Hunt Groups

A hunt groups rings all phones in the hunt group sequentially or in parallel to pick up the incoming call. If all extensions should be unavailable or do not answer, the Vdex will move to the total timeout extension. During the ringing state of the hunt group phones, the caller will hear the music-on-hold sound. Hunt groups do also perform queue functions. More calls can reach a hunt group at the same time. These calls are distributed to the hunt group members according the first-in first-out principle. Callers waiting in the hunt group can hear the music on hold sound.

SIP and IAX phones can be selected as hunt group members. Another (second stage) hunt group can be defined if another hunt group will be selected as total timeout extension.

If phones with activated voice mailbox are called via a hunt group the call will not be answered by the phone's voice mailbox. This allows that phones with activated voice mailbox can also be hunt group members without loosing the voice mailbox capability for direct phone calls.

External phones via the FXO interfaces cannot be member of hunt groups as a call via the FXO interface will immediately answer – even if the remote side isn't available.
When the total timeout expires the hunt group forwards the call to the total timeout extension which can be another internal phone, an external phone (e.g. mobile phone), an attendant or a voice mailbox.

Parameter	Default Value	Description
Name		Enter the hunt group name
Extension		The extension number can have up to 4 digits
Hunt method	Sequential	Parallel: All phones of the hunt group ring in parallel Sequential: One phone after the other is ringing. The sequence is according the group member list The no answer timeout per phone can be chosen between 10 seconds and 2 minutes. Random: One phone after the other is ringing in a random order. The no answer timeout per phone can be chosen between 10 seconds and 2 minutes.
Total Timeout	1 minute	Set the time how long the phones should try to ring before taking the default action.
Direct Dial In	unchecked	Allow to reach this extension directly through all SIP providers

		which support Direct-Dial-In and if it is activated.
What hears the caller while waiting	Music on hold	Select what the caller hears while waiting. Selct between "Music on Hold" and "Ringback Tone"
Description		You may enter a description here for your reference.
Group Members		Select the phones for the group. Phones can be member in more than one hunt group.

Using the combination of hunt groups and attendants allows the creation of useful services.

Example 1: Follow Me Service
External callers receive a messages that the system is trying to reach the desired person. Several phones ring in parallel or sequential. If nobody answers the call should be forwarded to a voice mailbox:

Create and upload a soundfile	e.g. "Hello, I will try to get Peter to the phone, please hold the line"
Create hunt group e.g. 4444	Choose the desired SIP/IAX phones which should be dialled in the desired order (parallel, sequential) Total timeout action: e.g. external phone (mobile phone) or voice mailbox (please note: it is not possible to have an external phone within the hunt group, since analogue lines do not indicate call answer events)
Create attendant e.g. 888801	Play the soundfile "Hello, I will try ..." No user input end action 0 retries after timeout Default action: Huntgroup 4444

Use the attendant extension 888801 to reach your follow me service.

Example 2 Follow Me Service with No-Answer Message
External callers receive a messages that the system is trying to reach the desired person. Several phones ring in parallel or sequential. If nobody answers the callers receive a message that the system could not reach the desired person and that he will be forwarded to a voice mailbox:

Create and upload 2 sound files	e.g. "Hello, I will try to get Peter to the phone, please hold the line" and e.g. "I am sorry, but I can not reach Peter but you can leave a message"
Create attendant e.g. 888801	Play desired soundfile "I am sorry ..." No user input end action 0 retries after timeout Default action: Voicemail
Create hunt group e.g.4444:	Choose the desired SIP/IAX phones which should be dialled Total timeout action: attendant 888801
Create attendant e.g. 888802	Play the soundfile "Hello, I will try .." No user input end action 0 retries after timeout Default action: Huntgroup 4444

Use the attendant extension 888802 to reach your follow me service.

8.4 Outgoing Callrule Tables

Callrules are used when an extension dials a number that is not available on the local PBX. You can assign the callrule table per extension. This gives you the possibility to assign different permissions to the extensions. For example, you might want to have a callrule table that allow extensions to handle national calls via analogue provider lines and international calls via cost effective SIP provider.
You can create as many callrule table as necessary.

Create a new callrule table with the add button. Enter the name of the callrule table and if required some comments in the description field.

The callrule table consists of four components:

- The enable check box is used to en- or disable the callrule entry.
- Dialling pattern are matched against the destination of the call. See below for the description of the matching algorithm.
- The provider setting defines which provider is used for the call.
- The priority defines the order of the callrules for call attempts at different providers. The system tries at first the provider with the lowest priority number. The Vdex system can response to dial attempts at provider lines with
  - FXO line is occupied
  - SIP/IAX channel unavailable
  In these cases the system will try the provider with the next priority level.

Dialling Pattern: Enter patterns to define a outgoing call routing. Prefixes can be defined as shown in the following example. A prefix of "9" is equivalent to a pattern of "9|.").

- a - adds a prefix (i.e. "1a555" matches "555" and passes "1555" to the provider)
- | - removes a prefix (i.e. "1|NXX" matches "1555" but only passes "555" to the provider)
- X - matches digits 0-9
- Z - matches digits 1-9 (i.e 0Z. matches numbers like 017890849 but not 00437890849)
- N - matches digits 2-9
- [13-5] - matches any digit in the brackets (here, 1,3,4,5)
- **.** - matches one or more characters (not allowed before | or a)
- ! - matches zero or more characters (not allowed before | or a)

Dialling pattern examples:

- The user dials 21xxxxxxx and the provider should receive 2100xxxxxxx. callrule: 2100+21|**.**
- The user selects all numbers with prefix 1: callrule: 1|**.**
- Route to US numbers without prefix: callrule 001**.**
- Emergency calls in Canada, US: callrule 911
- Emergency calls in Germany with prefix 0: callrule 0|11X

Outgoing callrule example 1

The Vdex system is connected to 4 FXO provider lines. Outgoing calls should try line FXO1. If line FXO1 is busy, then line FXO2 should be used. If line FXO2 is also busy line FXO3 should be used and so on. In this case use the same dialling pattern for the callrules for all FXO lines with different priorities:

Dialplan: Outgoing Callrule Tables: Edit

Name	out FXO name of this time segment.
Description	national calls over FXO lines Extensive description of this call rule table.

Enable	Dialling Pattern	Provider	Priority	Time Segment	Comment
☑	9.	BT FXO line 1 (FXO1)	1	Any Time	
☑	9.	BT FXO line 2 (FXO2)	2	Any Time	
☑	9.	BT FXO line 2 (FXO3)	3	Any Time	
☑	9.	BT FXO line 4 (FXO4)	4	Any Time	

Save

Outgoing callrule example 2

The Vdex is connected to 1 SIP provider and 1 FXO provider line. Calls with numbers of more than 3 digits should try the SIP provider first. If the SIP provider is not reachable the call will be established at the FXO line:

Dialplan: Outgoing Callrule Tables: Edit

Name	out2 name of this callrule table.
Description	Extensive description of this call rule table.

Enable	Dialling Pattern	Provider	Priority	Time Segment	Comment
☑	XXX.	FAHR Telecom (SIP)	1	worktime	
☑	XXX.	BT FXO line 1 (FXO1)	2	worktime	

Save

8.5 Incoming Callrules

Incoming callrules specify the destinations for incoming calls per provider depending on time, date and caller ID. Incoming calls can be routed to phone extensions, external phones, attendants, voice mailboxes or personal conference rooms. At least one incoming callrule must be set. Incoming calls which does not match any callrule are treated as the caller dialled a non-existing number.

Parameter	Default Value	Description
Name		Descriptive name for the callrule.
Enable	yes	Enable or disable the callrule.
Time Segment	worktime	Select the time segment which defines the time when the callrule applies.
Caller ID	Any	Select which caller IDs apply the callrule: any, anonymous or the pattern matched against the caller ID of the call. See below for the description of the matching algorithm.
Callback Prefix		This prefix is prepended to all incoming caller id's so that a callback can be routed with appropriate outgoing callrules. Example: Outgoing call rule prefix for provider Neotel: 90 Callback Prefix: 90 Incoming caller ID: 43123456789 Caller ID received at the phone: 9043123456789 Using the call back function the phone will dial prepended Neotel's prefix
Providers		Apply the provider(s) to the callrule.
Default Extension		Select the destination extension for the callrule.

Incoming calls are checked against the entries in the incoming callrule list beginning with from the top to the bottom entry. The first callrule that matches the call will be executed.

<u>Caller ID Pattern:</u>
Enter a complete caller ID or parts of caller IDs. Please make a test call to a phone with caller id display to check the exactly provider caller ID format. The character + is allowed at the first position of the caller ID string according the E.164 standard.

- X - matches digits 0-9
- Z - matches digits 1-9
- N - matches digits 2-9
- [13-5] - matches any digit in the brackets (here, 1,3,4,5)
- . - matches one or more characters
- ! - matches zero or more characters

<u>Example 1</u>
Incoming calls at worktime should be forwarded to the hunt group. Incoming callrule outside worktime should be forwarded to the attendant.

Dialplan: Incoming Callrules

Name	Enabled	Time Segment	Caller ID	Provider	Target
FXO in worktime	yes	worktime	Any	BT FXO line 1, BT FXO line 2	Huntgroup <60> peters phones
FXO in off time	yes	Any Time	Any	BT FXO line 1, BT FXO line 2	Attendant <82> off time message

The incoming callrules apply from top to bottom until the first one matches the provider and time. If none matches, the call is treated as if the caller dialed an non-existing number.

<u>Example 2</u>
Anonymous call block: Incoming anonymous calls at work time should be forwarded to the attendant 85 (which plays back e.g. "Your call is anonymous. Please call us again with activated caller ID" Incoming calls from Technoco at work time should be forwarded to extension 10. Any call off-time should be forwarded to attendant 82 (which plays a off-time message and forwards calls to a voice mailbox).

Dialplan: Incoming Callrules

Name	Enabled	Time Segment	Caller ID	Provider	Target
FXO in worktime anonymous	yes	worktime	Anonymous	BT FXO line 1, BT FXO line 2	Attendant <85> message for anonymous callers
FXO in Technoco	yes	worktime	+61396505444	BT FXO line 1, BT FXO line 2	Phone <10> 10
fxo in off-time	yes	Any Time	Any	BT FXO line 1, BT FXO line 2	Attendant <82> off time message

The incoming callrules apply from top to bottom until the first one matches the provider and time. If none matches, the call is treated as if the caller dialed an non-existing number.

<u>Example 3</u>
Incoming calls from Germany should be forwarded to a attendant with a German spoken welcome message.
Select at the caller ID section: Calling Number: 0049**.** (please check if the provider send the caller ID in this format; it could also be +49**.**)

8.6 Attendants

An attendant can be seen as a simple receptionist that helps to connect the incoming caller with an extension. The attendant usually plays back a message that asks the caller to enter the destination. For this purpose, it is necessary to record prompts that can be used in this phase (for example, "for sales press xxx"). These prompts have to be uploaded in the "Attendant Soundfile" section.

If the attendant is configured to receive an user input but the input is missing the default action will be performed.

If no user input is selected the default action will be performed after sound file playback. Default action can be a forward to an extension or a call hangup.

If the user input does not match any defined user input sequence then the attendant say "invalid choice" and repeats the sound file playback.

Attendants can not only be used as receptionist but also to play back sound files without any user interaction.

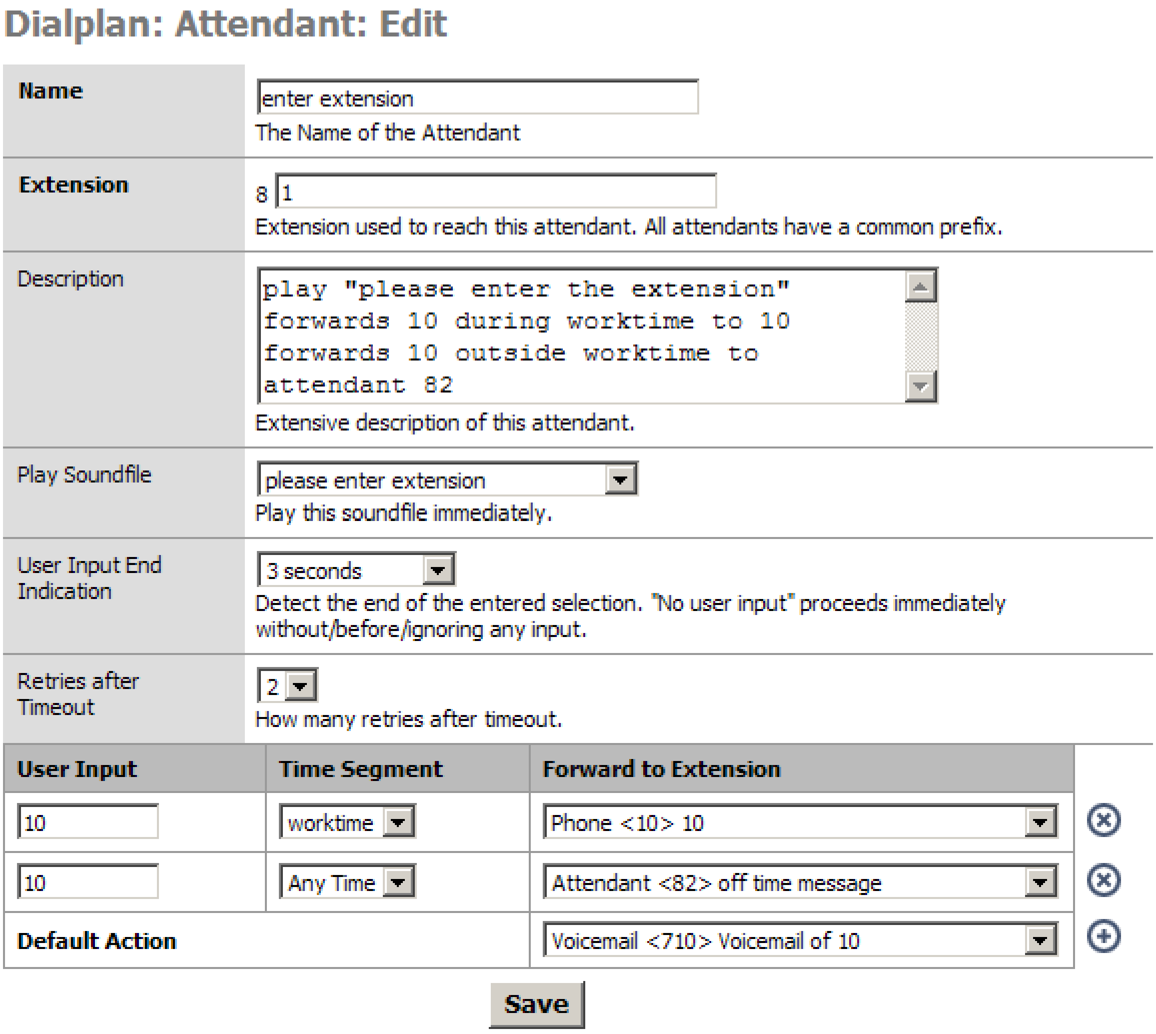
Parameter	Default Value	Description
Name		Enter the attendant name.
Extension		Extensions number for the attendant. The attendant extension number starts with a prefix followed by a any number. The prefix is common for all attendants and can be changed in the System/Extension Prefixes section. The attendant can be reached by any internal phone by dialling the defined extension e.g. 8888100.
Play soundfile		The scroll down menu offers the selection of an attendant sound file. If the desired sound file is not present in the scroll down menu one has to upload the file at the "Attendant Soundfile" section.
User Input End Indication		Select the end indication for the user input. - "no user input": the attendant proceeds immediately and performs the end action. - "3 seconds"/"5 seconds": no input within the given time performs the playback of the soundfile or performs the end action (depending of the "retries after timeout" setting). - "Only pound key": Attendant proceeds only when the caller press the pound key (if the pound key is not pressed within 1 minute the call will be cancelled).
Retries after Timeout		Select how many retries (play soundfile and wait for user input) should be performed when no user input within the selected timeout (user input end indication).
Direct Dial In	unchecked	Allow to reach this extension directly through all SIP providers which support Direct-Dial-In and if it is activated.
Publish this extension	unchecked	Publish the extension on outgoing calls for all SIP providers which support Direct-Dial-In and if it is activated

<u>User Input Table</u>
The user input table apply from top to bottom until the first one entry matches the user input and time. If none entry matches the attendant plays back "invalid choice". This allows call forwards for a certain user input sequence to different extensions at different times.

Parameter	Default Value	Description
User Input		Enter the numerical order of the user input. The maximum number of characters is 10. The user input sequence does not have to be equal to the internal extension number. The user input sequence is also not necessarily an extension number. One can use an attendant also to ask for a password, aera code, post code, age or whatever.
User Input		Select in which time segment the user input will be matched.
Forward to Extension		When the user input matches within the selected time segment the call will be forwarded to the selected extension.
Default Extension		The call will be forwarded to the default extension in the case of - "no user input" or - if the maximum number of retries has been reached or

		- if the user input matches the user input sequence outside the time segment

<u>Example</u>
When someone calls the attendant, the attendant first plays back the message "enter the extension number". If the caller sends the user input sequence 10 the call should be connected to extension 10 during work time. Outside work time the call should be connected to attendant 82 which plays back an off time message. In the case of no user input the call should be forwarded to the voice mailbox 710.



Dialplan: Attendant: Edit

Name	enter extension The Name of the Attendant
Extension	8 1 Extension used to reach this attendant. All attendants have a common prefix.
Description	play "please enter the extension" forwards 10 during worktime to 10 forwards 10 outside worktime to attendant 82 Extensive description of this attendant.
Play Soundfile	please enter extension Play this soundfile immediately.
User Input End Indication	3 seconds Detect the end of the entered selection. "No user input" proceeds immediately without/before/ignoring any input.
Retries after Timeout	2 How many retries after timeout.

User Input	Time Segment	Forward to Extension
10	worktime	Phone <10> 10
10	Any Time	Attendant <82> off time message
Default Action		Voicemail <710> Voicemail of 10

Save

9 Services Configuration

9.1 Voicemail Settings

The general setup for the voicemail service is also valid for conference recording emails. The maximum storage capacity for voicemail messages and conference recording files is 300 MB. Voicemails are stored in GSM format. The total time for all voicemails is appr. 52 hours. The voicemail memory usage is shown at the system information page.

Parameter	Default Value	Description
Maximum Time per Voicemail	30 seconds	Specify how long a voice mailbox message may be.
Number of Digits for the Mailbox PIN	4	Select the number of digits for the voicemail PIN. The PIN length is common for all mailboxes.
Mailserver		Set the hostname or IP address for the SMTP mail server. Enter the port number if another port than 25 is used.
Enable SMTP Authentication	no	Select yes if the SMTP server needs a login with username and password. Authentication methods PLAIN and LOGIN are supported.
Username		The username can be entered if SMTP authentication is enabled.
Password		The password can be entered if SMTP authentication is enabled.
"From:" Address		Enter the email address which will be shown as the sending address of the voicemail.
Language	English	Select the language for the the notification text in the email.

9.2 DISA

DISA means “Direct Inward System Access”. Calls to a DISA extension will be checked against the callers ID. The caller will be asked to enter the PIN code for this DISA account. Authenticated DISA calls will get an internal dial tone and are allowed to act as an internal extension. The caller ID from a DISA account is the internal extension number. Outgoing calls are allowed according to the selected outgoing callrule table.

External calls which should be authenticated for DISA must be forwarded to the respective DISA extension. This can be done via incoming callrules or attendants.

Parameter	Default Value	Description
Name		Descriptive name for the DISA account.
Enable	yes	Enable or disable this account.
Extension		Enter the extension number displayed as caller ID at the callee side.
PIN		Enter the PIN code for user authentication.
Caller ID		Only calls with this caller ID are authorized for using this DISA account.
Language	English	Audio prompts will be played back in the selected language for this account.

Outgoing Callrule Table		The authenticated DISA caller is allowed to establish calls according the selected outgoing callrule.

9.3 Attendant Soundfiles

Before using an attendant the desired soundfiles must be stored at the Vdex system. 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.

9.3.1 Record attendant message sounds via telephone:

You can dial 10 different extension (<HandsetRecordingPrefix>0 – <HandsetRecordingPrefix>9; e.g. 88861 - 88869) to record IVR message sound files up to a maximum defined length of 2 minutes. Follow the instructions at the handset recording extensions ("press 1 to record a new message", "press 9 to delete the message"). Once recorded these sound files will be listed at the Attendant Soundfile section (e.g. "Handset Recording 0" or "Handset Recording 5", etc.; only files with recorded content are listed). After recording an attendant message, you can dial the handset recording extension an press "2" to listen to your recording. Alternatively you can open the recorded file at your browser and play back with the PCs audio player.

The maximum storage capacity for attendant and Music-on-hold soundfiles is 136 MB. This is an equivalent in time of appr. 82 minutes. The memory usage is shown at the system information page.

Please note, that the handset recording extensions can be reached by every Vdex registered phone without any restriction. If you want to make sure that nobody else than the Vdex administrator can change the content of the recorded messages upload the soundfiles via PC.

9.3.2 Record attendant message sounds via PC

Use any sound recorder (e.g. the standard audio recorder of Microsoft Windows) and store the soundfile in the mono WAV format (8 kHz sampling frequency, 8 bit per sample signed format, PCM) with a self explanatory filename (e.g. welcome-message-worktime.wav). Locate the file at your computer by using the "Browse" button. By pressing the "add" button the selected file will be uploaded into the Vdex and automatically converted into several sound formats (e.g. G.729, etc). Depending on the codec used in a call the file in the corresponding format will be played back (this prevents codec conversion for better quality).

Only one file can be uploaded at a time. The system will display the upload finish with an alert box.

The maximum storage capacity for attendant and Music-on-hold soundfiles is 136 MB. The memory usage is shown at the system information page.

9.4 Music on Hold

The Vdex uses "Music on Hold" (MoH) when a call in being put on hold and when a caller is waiting in a hunt group. Vdex uses one or more sound files for MoH. These files plays in an endless loop and in random order.

The MoH files must be in 8 kHz sampling frequency, 8 bit per sample signed format and in mono WAV. Please use any audio file conversion software if you need to convert any sound file into the required format (e.g. free audio conversion tool from NCH Software: http://www.nch.com.au/switch).

Locate the MoH sound file at your computer by using the "Browse" button. By pressing the "add" button the selected file will be uploaded into the system and automatically converted into several sound formats (e.g. G.729, etc). Depending on the codec used in a call the file in the corresponding format will be played back (this prevents codec conversion for better quality).

Please remember that local copyright law also applies to MoH.

9.5 DDI (Direct-Dial-In)

Overview for all configured extensions to specify per extension:

Parameter	Default Value	Description
Allow Dial-Direct-In	checked	Specify if it is allowed to reach the extension directly from outside
Publish extension	checked	Specify if the extension number should be published for outgoing calls

This options can also be set in the each single phone configuration section.
This options are used only when a SIP provider with enable Direct-Dial-In feature is used.

10 Voicemailbox

Global voicemail settings can be changes at the section "Services: Voicemail" (maximum recording time, number of PIN digits, email settings). Each phone can have a dedicated voice mailbox. The language of the voice mailbox audio prompts is the same as the selected language for the phone audio prompts. The voice mailbox settings can be found at each phone account setting. Callers who reach the voice mailbox can leave a message.
Vdex indicates new messages by sending "Message Waiting Indication" (MWI) to the phone. This MWI message forces the phone to indicate a new message with a visual signal (depending on the phone model).

Voicemail to email: Vdex offers the feature to email the voice message as attached WAV file to the phone user email account. The user has the possibility to listen the message by using any standard player at the desktop, to archive the emails and to listen the voice messages also by downloading the sound file in webmail. It is recommended to select the option "soundfile attached and delete" since this saves memory space at the Vdex.

Retrieve messages via phone: Vdex offers the phone owner various ways to play back the messages:

- Calling the voice mailbox extension from the own phone extension: The caller will be asked for the PIN code to enter the voice mailbox menu.
- Calling the voice mailbox via a provider trunk or any other extension: The caller hears the standard voice mailbox welcome message. Pressing the star button during this message will lead the caller to the PIN code request message to enter the voice mailbox menu.

10.1 PIN assignment

It is mandatory to set a PIN code which is needed to enter the voice mailbox menu (there is no default Pin code). Once the PIN code is set by the Vdex administrator the PIN code will not be display at the AstA*UI. The user has the possibility to change the PIN code in the voice mail menu to another value. The necessary number of digits for the user PIN code input is not stipulated as it is at the AstA*UI. In the case that the user has forgotten the PIN code the Vdex administrator has to set a new PIN code.

10.2 Voice mail menu structure

- **1** Read voice mail messages
    - **3** Advanced options
        - **1** Reply a message to the voice mailbox of the caller (only internal Vdex caller)
        - **3** Say message envelope information (time, date, number)
        - **5** Forward message to another Vdex user
    - **5** Repeat current message
    - **6** Play next message
    - **7** Delete current message
    - **8** Forward message to another user
        - **1** Prepend a message to forwarded message
        - **2** Forward message without pre-pending
    - **9** Save message in a folder
        - **0** Save in new Messages
        - **1** Save in old Messages
        - **2** Save in Work Messages
        - **3** Save in Family Messages
        - **4** Save in Friends Messages
    - ***** Help
    - **#** Cancel / Exit to main menu
- **2** Change folders
    - **0** Switch to new Messages
    - **1** Switch to old Messages
    - **2** Switch to Work Messages
    - **3** Switch to Family Messages
    - **4** Switch to Friends Messages
- **3** Advanced Options
    - **5** Send Message to another Vdex user
- **0** Mailbox options
    - **1** Record your unavailable message
    - **2** Record your busy message
    - **3** Record your name
    - **4** Record your temporary greeting
        - **1** Record your temporary greeting
        - **2** Erase your temporary greeting
    - **5** Change your password
    - ***** Return to the main menu
- ***** Help
- **#** Exit

Backup: Voice mail messages are not included in the backup process. Please follow the recommendation to use the voice mail to email feature.

11 Conference Rooms

Every SIP and IAX phone user can be the owner of a personal conference room. A personal conference room allows conference recording. Recorded soundfiles are sent to the phone owner by email. This assures that nobody else than the phone owner can acquire the conference content. Furthermore it is possible that the conference starts only when the phone owner (the leader) enters the conference room. Personal conference rooms can be activated and configured at each SIP and IAX phone section.

12 Diagnostics

12.1 Logs

The Vdex keeps 4 different log message tables in the memory. The maximum numbers of entries per table can be selected in the settings tab. The messages are stored in ring buffers which stores the selected number of the latest messages. At the end of each log table can be cleared by using the "Clear Log" button at the bottom of each log table.

<u>System Logs</u>
The operating system of the Vdex is Linux. The log messages from /var/log/messages are shown. Please refer to any basic Linux introduction.

<u>PBX logs</u>
Vdex uses the telephony application Asterisk. The log messages from var/log/asterisk are shown. For more information please refer to any Asterisk documentation.

<u>Call logs</u>
The call detail records of Asterisk are shown:

- Start: Start of call (date/time)
- Src: Caller ID number
- Dst: Dialled destination
- Channels: Channel used (channel type / channel ID)
- Last App: Last application if appropriate
- Total: Total time call is in use, in seconds
- Call Up: Time call is up, in seconds
- disposition: What happened to the call: answered, no answer, busy, failed

The call logs can be uploaded as a CSV file by using the "download" button at the end of the log table.

12.2 Ping/Traceroute

<u>Ping</u>
You can use the network tool Ping whether a particular host is reachable across an IP network. Enter the target host name or IP address and select how many packets should be sent.

<u>Traceroute</u>
You can use the network tool traceroute to determine the route taken by packets across the network to the target host. Enter the target host name or IP address and select the maximum number of hops in the outgoing probe packets.

12.3 ARP Table

The ARP table shows the IP and MAC addresses for all computers/devices/host that have transferred data to the Vdex.

12.4 Commandline Interface

This feature allows to enter Asterisk specific commands. For more information please refer to any Asterisk documentation.

13 Examples and tips

13.1 Call recording announcement

AstA*UI offers the possibility to record personal conference. The recorded sound files will be sent to the conference owners email account. If the conference participants should hear an announcement that the following conference will be recorded, then use an attendant:

create and upload a soundfile e.g. "this conference will be recorded"
create a new attendant extension, which should be published as the new conference extension

attendant options:	play uploaded soundfile No user input end action 0 retries after timeout default action: personal conference room

13.2 Automated operator

Callers can dial the desired extension after a welcome message with DTMF sequences.

Create and upload a soundfile e.g. "please dial the desired extension or press zero for the operator"

Create attendant:	play uploaded soundfile		
	User input end action: e.g.5 seconds		
	Retries after timeout: e.g. 2 retries		
	User Input 0	All Day	Forward to Extension 100
	User Input 101	All Day	Forward to Extension 101
	User Input 102	All Day	Forward to Extension 102
	User Input 103	All Day	Forward to Extension 103
	User Input 104	All Day	Forward to Extension 104
	User Input 702	All Day	Forward to Personal Conference Room 888702
	User Input 45885	All Day	Forward to Voicemail 888902
	Default Action	All Day	Forward to Extension 100

13.3 Off-Time and vacation messages

Route incoming calls during off-time and vacation time segment to an attendant. This attendant plays the message e.g. "You reach us outside business hours ....." and hangs up the call.

create and upload a soundfile e.g. "You reach us outside business hours .."
create a new attendant extension, which should be used by the incoming call rule

attendant options:	play uploaded soundfile No user input end action 0 retries after timeout default action: hangup

13.4 Call Pick-Up

The Vdex supports directed call pick-up of calls from any other Vdex extensions by

- dialling *8xxx (xxx : number of the ringing extension)
- using the BLF feature of phones (described in the section "Extension Monitoring")

The Vdex supports undirected call pick-up of calls from extensions in the same user group by

- dialling *8

13.5 Extension Monitoring

Extension monitoring indicates the user the status (idle, ringing, busy, not registered) of other phones. Vdex supports extension monitoring by provisioning of the dialog state via the busy lamp field (BLF) feature. Some devices are able to monitor and display the dialog state of other extension. In order to do this, you need to specify at the BLF aware phone the "List of extensions to watch" in the user mode for an extension.

Example:
Snom BLF configuration can be done in the Functions Keys section. Use one of the programmable buttons and select the type "BLF" and enter following string in the number field:

<sip:extension@host;user=phone>|*8 extension: number of the monitored extension
host: hostname or IP address of the Vdex
e.g. <sip:40@192.168.0.157;user=phone>|*8

Snom phones support not only extension monitoring with the BLF feature but also call pick-up from and speed dial to the monitored extension by pressing the assigned button.

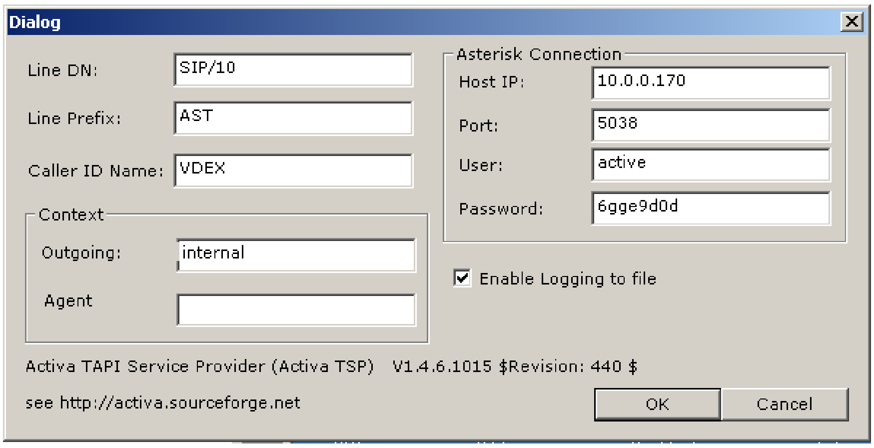
13.6 Call Queue Support

Simple call queues can be realized by using hunt groups. Calls to a hunt group will be queued and distributed according the first-in first-out principle. The callers will hear the music on hold sound until a hunt group member pickups the call.
It is also possible that each phone has its own call queue. Configure a hunt group for each phone with only the target phone as hunt group member. Publish the hunt group extensions as phone extensions.

13.7 Computer Telephony Interface

Various third party Computer Telephony Integration software is available to integrate Asterisk with desktop productivity applications such as Microsoft Outlook through the use of a TAPI driver. This enables users of TAPI compatible software to dial phone numbers from their computer. Users can select a contact in their software address book, click a dial button, and the call is initiated by the Vdex.

Example ActivaTSP
The free available tool Asterisk ActivaTSP for Microsoft Windows . For download, installation and use of ActivaTSP please refer to activa.sourceforge.net.
ActivaTSP must be connected to the Vdex Asterisk Manager Interface (AMI). Please configure an Asterisk Manager Account (see section Accounts) at the Vdex. Use this account data for the ActivaTSP configuration:



Line DN:	SIP/<Extension Number> (e.g. SIP/10), extension that is related to the computer.
Line Prefix:	AST
Caller ID Name:	Caller ID name of the phone
Host IP:	the Vdex IP address
Port:	5038
User:	Username of the Vdex AMI account
Password:	Password of the Vdex AMI account
Context Outgoing:	internal
Agent:	empty