

AstA*UI

Asterisk™ Appliance User Interface
by
Firmix Software GmbH



and

Vdex-40

Voice Digital Exchange
by
TechnoCo Pty Ltd

**Installation and Administration Manual**

Revision 1.6.0

valid for FW Version 2.2.38

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IMPORTANT:

From firmware version 2.2.6 onward the call rule pattern has been enhanced. If upgrading from an earlier version you will need to change existing call rules. Refer page 42 for detailed information.

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1 General

1.1 Copyright and Trademark Notices

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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.

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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 reserves 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 3.5 or higher and Internet Explorer 8.0 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 lightning 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

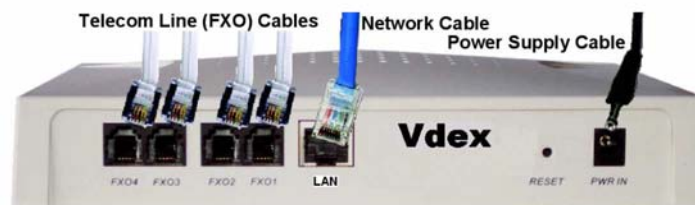
Under power failure conditions this device will NOT OPERATE. Please ensure a separate telephone device not dependant upon local power is available for emergency use. The device will NOT FUNCTION for Internet calls when disruption to the Internet and connection to it occurs.

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.

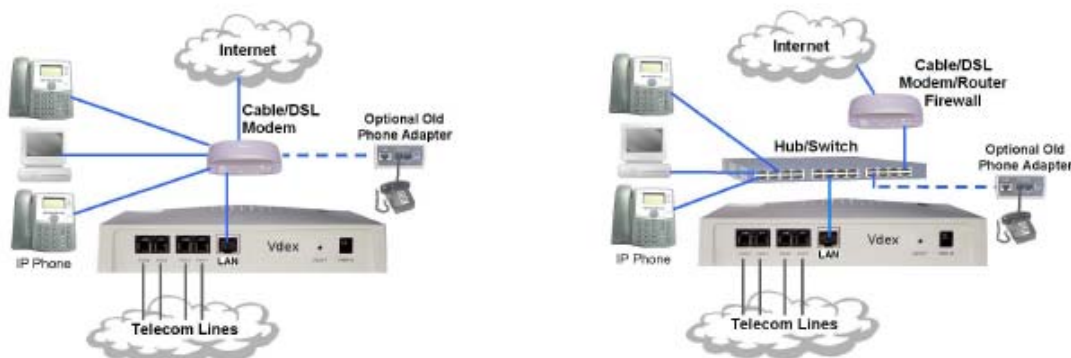
Remove the Vdex system, power supply, and network cable from the Vdex box. Connect the network cable to the LAN port on Vdex system and to an available port on your network router, switch, or hub. Connect up to four analog telephone lines to FXO ports 1- 4 as shown in the diagram:



Please note: You must connect at least one telephone line to FXO port 1 to dial 911 for emergencies and configure an appropriate outgoing callrule - see outgoing dialling 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 dotted-quad 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 servers to be used by the Vdex system. SIP won't really work without a working DNS setup.

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 "Save" to complete the Vdex wizard.

Host Name	<input type="text" value="vdex.firmix.at"/>
	System host name without domain part, e.g. vdex40
Domain Name	<input type="text" value="localdomain"/>
	System domain name, e.g. firmix.at
IP Address	<input type="text" value="10.0.0.170"/> / <input type="text" value="24 (255.255.255.0)"/>
Gateway	<input type="text" value="10.0.0.1"/>
DNS Servers	<input type="text" value="10.0.0.10"/> <input type="text" value="10.0.0.138"/> <input type="text"/>
	IP addresses of the domain name servers to use.
SMTP Server Host	<input type="text" value="smtp.firmix.at"/> : <input type="text"/>
	SMTP server host name or IP address and an optional port (default is 25).
E-Mail "From:" Address	<input type="text" value="vdex40@firmix.at"/>
	Sender address used for email notifications.
System Administrator Password	<input type="text"/> <input type="text"/> (confirmation)
	Change system administrator password.
Enable SSH Access	<input type="text" value="Yes"/>
	Enable SSH access.

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 **x** Delete entry and the related parameter.



Button **+** Add a new entry (e.g. provider, phone)



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 configure the Vdex the subsequent order.

- (1) Configure network interface parameter
- (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 Vdex buddies
- (10) Configure DISA accounts
- (11) Configure external phones
- (12) Define hunt groups and extension profiles
- (13) Create attendants
- (14) 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
System Administrator 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 Tonezone	Austria	Indication tones have to be generated by the Vdex in various 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.
Time Zone	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. The daylight-saving time will be considered automatically.
Time Server (NTP)	3.pool.ntp.org	The Vdex gets the current time and date from any NTP server. You must set a timeserver. Using an non-functional timeserver would cause a malfunction of time dependent callrules and wrong timestamps in all logs. Each night Vdex time and date will be checked against the NTP server and only if the difference is higher than 10 seconds a synchronization will be performed. The last successful synchronization between Vdex and 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). To get the best voice quality, use the one used on your landline (at the FXO interfaces).
G.723 Encoding 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.
FXO 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.
Enable SSH Access	yes	Enable or disable the integrated SSH server for system access. The Vdex incorporates a SSH server. If the SSH server is enabled, you can login as user "root" with the password as

		<p>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.</p> <p>Example: Secure remote connection from a Windows client</p> <p>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.</p>
Allow Call Pickups	yes	If this option is activated, users are able to dial „*8“ (undirected pickup) to pickup any call of the assigned pickup groups (see also call- and pickup group configuration for each SIP phone at page 28 and general pickup info at page 51)

4.2 Network Setup

Parameter	Default Value	Description
Network Device (Mac Address)		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. If the IP address is changed, the web interface automatically redirects to the new IP address.
Gateway		The default gateway IP address must be set.
DNS Servers		The IP address of at least one DNS server must be set. This is necessary to resolve domain names of SIP- and IAX-providers, STUN servers, SMTP servers, etc.. It is possible to define up to 3 different DNS server.
Topology	Public IP Address	<p>This value must be set according to the used network address topology.</p> <p>“Public IP address“ means that the above given Vdex IP address is a routable, static, public IP address.</p> <p>“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".</p> <p>“Use a stun server” means that the Vdex is behind a NAT router, which has a dynamical (or static) 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".</p>

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 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 losing 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 may take several minutes depending on the configuration and soundfile sizes.

To restore a backup, 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 overwrite your existing configuration, so be careful about this step.

Please note, that also the network configuration and the password of the restored file will be activated (which is maybe different to the actual used configuration).

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'.

Factory reset procedure: Hold down the reset button on the rear of the Vdex unit for 12 seconds

After 8 seconds: All LEDs go on for 6 seconds

After 90 seconds: RUN LED is blinking (Vdex is not yet operable)

After appr. 4 minutes: All LEDs go on for 6 seconds

After appr. 5 minutes: RUN LED is blinking. Factory reset finished.
(Vdex-40 is up and operable)

4.6 Reboot

Performing a reboot (press the reset button for a short time or activate reboot from the user interface) 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 Voice Notification

The voice notification feature is designed to inform or alert people with pre-recorded messages. Each voice notification is triggered by a username/password protected HTTP request. The Vdex is calling up to 8 pre-defined phone numbers per voice notification. The called person hears the pre-recorded sound file and has to acknowledge the message by entering a PIN code via DTMF. The voice notification stops after receiving the correct PIN code.

Parameter	Default Value	Description
Name		Name of this voice notification, also used as authentication username for remote voice notification requests.
Authentication Secret		Secret used for authentication of remote voice notification requests.
Soundfile	No Sound file	The scroll down menu offers the selection of an notification sound file. If the desired sound file is not present in the scroll down menu one has to upload the file at the "Sound File Management" section.
Confirmation PIN		Confirmation PIN to be entered by the remote user.
Call Establishment Timeout	40	Call establishment timeout (in seconds, range 5..120). Note: When calling out over an FXO line, the call is considered established after the last digit is dialed. On SIP calls a call is considered established when the remote party answers by going off-hook.
Max. Call Duration	160	Maximum number of seconds (range 0..600) the system waits for DTMF input from the remote user after a voice notification call is established. Note: on FXO lines a call is considered established after the last digit is dialed.
Repeats	5	Number of sound file repeats, an empty value means repeat until end of the call.
Language	English	Language to use for built-in audio prompts (e.g. "notification confirmed, invalid PIN, etc.)
Numbers to Call		Enter the phone numbers to call in this order. The phone number must be matched by at least one outgoing call rule.

5.1.1 Voice Notification Remote Control

Voice notification can be triggered and controlled remotely via http requests. A technical specification of this machine-machine interface can be found at chapter 15 on page 54.

5.2 Voicemail Setup

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 GSM format is also used for soundfiles, which are attached to emails.

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 Voicemail Duration	30 seconds	Specify the maximum duration per voicemail message.
Number of Digits for Voicemail Box PINs	4	Select the number of digits for the voicemail PIN. The PIN length is common for all mailboxes.
SMTP Server Host		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.
Authentication Username		The username can be entered if SMTP authentication is enabled.
Authentication Password		The password can be entered if SMTP authentication is enabled.
Email "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.

5.3 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 uses 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 dialling the handset recording prefix followed by one digit (0 – 9).
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Note: For the remainder of this document, we will assume that the prefixes have their default value and show that in the text and screenshots.

5.4 RTP Settings

Parameter	Default Value	Description
RTP Port Range	10000 - 20000	<p>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.</p> <p>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.</p>
Type of Service	EF (Decimal 184 – Expedited Forward)	<p>Select the Type-of-Service qualifier for the audio and video RTP packets.</p> <p>The numerical value is the content of the Type-of-Service-field (in the IP header). Default is <i>EF (184 - Expedited Forward)</i>. Use the value specified by your SIP/Internet provider.</p>

5.5 SIP Settings

Parameter	Default Value	Description
SIP UDP Port	5060	The UDP port that Vdex uses for SIP communication.
SIP Registration Timers	min: 60 s max: 3600 s default: 120 s	<p>The minimum and maximum of seconds that incoming registrations remain valid.</p> <p>The number of seconds that outgoing registrations (and incoming registration without given expiration time) remain valid.</p>
Disable DNS SRV Record Lookups	enabled	This option allows to disable DNS SRV lookups. It is recommended to leave DNS SRV lookups activated.
Type of Service	EF (Decimal 184 – Expedited Forward)	<p>Select the Type-of-Service qualifier for the SIP packets.</p> <p>The numerical value is the contents of the Type-of-Service-field (in the IP header). Default is <i>EF (184 - Expedited Forward)</i>. Use the value specified by your SIP/Internet provider.</p>

5.6 IAX Settings

Parameter	Default Value	Description
IAX 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.
Type of Service	EF (Decimal 184 – Expedited Forward)	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 <i>EF (184 - Expedited Forward)</i> . Use the value specified by your SIP/Internet provider.

5.7 AMI 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
Access 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.
Account Permissions	unchecked	Allow the use for the selected AMI commands

5.8 GUI Options

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

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

- hide the SIP providers, phones and SIP options and their statuses on the summary page
- hide the IAX providers, phones and IAX options and their statuses on the summary page
- hide the analogue providers

6 Provider Configuration

6.1 SIP Provider

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

Parameter	Default Value	Description
Name		Descriptive name of 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.
Username		Enter the username given by the provider
Secret		Enter the SIP account's password given by the provider.
Authorization User		Enter the separate authorization user necessary for some providers. Leave the field empty when the username and authorization user are identical.
Main Number	none	If Direct-Dial-In is supported the main number (resp. the fixed part of the phone 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.
Main Number Caller ID	none	This caller ID is used for calls from extensions with hidden extension number. If empty, the main number is used.
Enable Direct Dial-In	deactivated	Some SIP providers support Direct-Dial-In. This feature allows external callers to reach an extension directly by dialling the main number added by the extension number. Configure which extensions should be reachable directly at each extension section.
Proxy	empty	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. If the field is empty, the value from SRV records in the DNS is used or - if the SRV records are not found - the default is 5060.
Outbound Proxy	empty	SIP outbound proxy hostname or IP address and optional port. 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. If the field is empty, the value from SRV records in the DNS is used or - if the SRV records are not found - the default is 5060.
Inbound Gateway/Proxy		Most providers deliver calls to their subscribers ONLY from the IP address where the subscriber has registered. But

Pool		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.
Do Not Register At Provider		Select if this account should not register at the provider.
Registration by Provider		Select if the provider (which can also be an ISDN gateway) is expected to register at the Vdex.
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 do require a differnt value. If "Main Number with Extension" should be used, "Enable Direct Dial-In" must be configured above.
Audio Codecs	a-law μ-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.
DTMF mode	rfc2833	Set the DTMF transfer mode for this account according the provider supported method. Choose between RFC2833, Inband and SIP Info mode. Please note, that inband DTMF transmission using other codecs than G.711 will not work properly.
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.
Language	English	Set the audio prompt language for this account. Choose between German and English.
Trust Remote-Party-ID	unchecked	Select if the Remote-Party-ID should be accepted.
Enable T.38	No	Must be enabled If the provider AND the fax adapter supports T.38
Propagate Original Caller-ID	unchecked	Propagate the incoming caller ID on forwarded calls. If an internal extension has activated a call forward to any public phone number using this SIP provider, the forwarded call is using the original caller-ID (set in the SIP "P-Preferred-

		Identity" header). Please note that not all SIP providers do support this.
Hide Mainnumber	unchecked	Do not put our main number into the SIP "P-Preferred-Identity" header. Some SIP providers with DDI support expect only the calling extension in the "P-Preferred-Identity" header.
Hide Display Name	unchecked	Do not add a display name to the SIP "P-Preferred-Identity" header. Some SIP providers with DDI support are confused by a display name in the "P-Preferred-Identity" header so we do not send them.

6.1.1 Example Engine (with DDI)

Name	<input type="text" value="Engin Australia"/> Descriptive name of this provider.
Load Provider Defaults	<input type="text" value="All countries"/> <input type="text" value="Select a provider"/> <input type="button" value="Set Defaults"/>
Username	<input type="text" value="031234000"/> Username used for authentication at this SIP provider.
Secret	<input type="text" value="bobisslow"/> Secret used for authentication at this SIP provider
Authorization Username	<input type="text"/> Please note that some SIP providers require a seperate authorization username. Defaults to username entered above.
Main Number	<input type="text" value="03123400"/> Fixed part of the main phone number
Main Number Caller-ID	<input type="text" value="031234000"/> Caller-ID used for calls from extensions with hidden extension numbers (if empty, the value of "Main Number" is used)
Enable Direct Dial-In (DDI)	<input checked="" type="checkbox"/> Enable Direct Dial-In (DDI). if enabled, extensions can be called directly via this provider. Please not that DDI must also be enabled for every extension which should be reachable directly.
Description	<input type="text"/> Short free text description for your reference (not parsed).
Registration Info ▲	
Proxy	<input type="text" value="byo.engin.com.au"/> : <input type="text"/> SIP proxy hostname or IP address and optional port (if empty, the value from SRV records in the DNS is used or - if these are not found - the default is 5060).
Outbound Proxy	<input type="text"/> : <input type="text"/> SIP outbound proxy hostname or IP address and optional port (if empty, the value from SRV records in the DNS is used or - if these are not found - the default is 5060).
Inbound Gateway/Proxy Pool	<input type="button" value="⊕"/> Add all additional hosts or IP addresses this provider uses for incoming calls.
Do Not Register At Provider	<input type="checkbox"/> if enabled, this system will not register at this provider.
Registration by Provider	<input type="checkbox"/> if enabled, this provider is expected to register at this system
Registrar	<input type="text"/> : <input type="text"/> Domain name, hostname or IP address and optional port (default is 5060) where Vdex should register."Proxy" or "Outbound Proxy" are used for registration if this field is left empty.
From Domain	<input type="text"/> Domain name used in the SIP "From:" header field for calls to this provider. 'Proxy' is used if this field is left empty.
From User	<input type="text" value="Main Number with Extension"/> <input type="text"/> The username used in the SIP "From:" header for calls to the provider. "Username" is the recommended option (RFC-compliant). However, some SIP providers do require a different value. If "Main Number with Extension" should be used, "Enable Direct Dial-In" must be configured above.

Miscellaneous ▲		
Audio Codecs	Enabled <i>(drag-and-drop)</i>	Disabled
	<div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.711 A-law</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.711 μ-law</div>	<div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">GSM</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.729A</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">iLBC</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.723.1</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.726 AAL2</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">G.726 RFC3551</div>
Video Codecs	Enabled <i>(drag-and-drop)</i>	Disabled
		<div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">H.261</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">H.263</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">H.263+</div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;">H.264</div>
DTMF Mode	<input type="text" value="rfc2833"/>	
Qualify Timeout	<input type="text" value="2 seconds"/>	
	Timeout after a SIP provider is considered unreachable. Please note that unreachable SIP providers will not be called.	
Language	<input type="text" value="English (US)"/>	
	Audio prompts will be played back in the selected language.	
Trust Remote-Party-ID	<input type="checkbox"/> Trust the Remote-Party-ID. Experts only.	
Enable T.38	<input type="text" value="No"/>	
	Does this provider handle T.38? Please note that the used codecs are G.711 alaw or ulaw	
Propagate Original Caller-ID	<input type="checkbox"/> Propagate the original caller-id on forwarded calls	
	If enabled, all forwarded calls (via this SIP provider) use the original caller-ID in the SIP "P-Preferred-Identity:" header. Please note that not all SIP providers do support this.	
Hide Main Number	<input type="checkbox"/> Do not put the "Main Number" into the SIP "P-Preferred-Identity:" header field	
	Please note that some SIP providers expect only the calling extension in the SIP "P-Preferred-Identity:" header.	
Hide Display Name	<input type="checkbox"/> Do not add a display name to the SIP "P-Preferred-Identity:" header	
	Please note that some SIP providers do not handle a display name in the SIP "P-Preferred-Identity:" header correctly.	

6.2 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“ as used by the provider.
Host		Enter the SIP proxy host URL or IP address of the provider. If the provider is using a different IAX port than 4569 this port can be optional entered in the Host port field.
Registration		Select if this account should not register at the provider (for advanced purposes).
Language	English	Set the audio prompt language for this account. Choose between German and English.
Qualify Timeout	2 sec	If the IAX 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 μ-law	Allow one or more audio codec types for calls to and from this IAX 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 IAX provider. The listing of the enabled codecs (top down) corresponds with the preference order.

6.3 Analog Line (FXO) Provider

Some parameters of this section can be reset to the value of the last stored change (e.g. “reset to 5s”). 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.

Parameter	Default Value	Description
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- <i>n</i>	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 Dialtone	Yes	It is recommended that the dial tone must be detected before dialling.
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 dialling 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 Dialtone“ 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 of the PSTN provider ring

Parameter	Default Value	Description
Minimum On hook Time	0.3 s	Specifies the period of time the line needs to be on hook, before the line can be 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 period 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 calling party control (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 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 than the PSTN silence duration then the call will be terminated.
PSTN Silence Duration	30 s	Period of time with an 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 Buddies

Vdex units can be connected together via the SIP buddy feature. If you run 2 or more Vdex units within one organization, you can configure following features:

- dial extensions at the remote Vdex
- use provider accounts and analogue lines at the remote Vdex
- transfer calls to remote Vdex extensions (blind and attendant transfer)
- forward calls to remote Vdex extensions via
 - attendants
 - timeout action of extension profile
 - busy action of extension profile
 - total timeout action of hunt groups
 - incoming callrules
 - SIP/IAX phone forward feature

Both ends of a SIP buddy connection must use the same username and password for authentication.

Parameter	Default Value	Description
Name		Name of the buddy. This is also the username for authentication. Use the same name at both sides of the buddy connection.
Secret		The password for this account. Use the same password at both sides of the buddy connection.
Proxy		Enter the hostname or IP address of the remote Vdex. 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 remote Vdex is using a different SIP port than 5060 this port can be optional entered in the Host port field. If the field is empty, the value from SRV records in the DNS is used or - if the SRV records are not found - the default is 5060.
Outgoing Callrule Table		Select one of the callrule tables for this buddy. Calls, which comes from the remote Vdex buddy are allowed to establish outgoing calls according the selected callrule. Buddies with permissions for internal calls only (which includes also external phones) must be set to „No outgoing calls allowed“.

Default extension		Calls without target extension, which comes from the remote Vdex buddy will be forwarded to the default extension.
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.
Qualify Timeout	2 seconds	The Vdex sends regularly qualify information to the remote Vdex. If the remote Vdex does not respond within the selected timeout the buddy state will be set to unreachable (see info at the status summery page). Calls to buddy in the unreachable state will receive a busy signal immediately. If the qualify timeout is set to off, the buddy state is unmonitored. Calls to unmonitored buddies will always sent to the buddy, no matter if it the last qualify was ok or not.
Audio Codecs	a-law 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).

A Vdex buddy connection requires also a suitable outgoing callrule table. Please consider, that this callrule table needs entries to reach remote Vdex extensions and lines.

Example:

Dialplan: Outgoing Callrule Tables: Edit

Name
name of this callrule table.

Description

```
- extension 10, 11, 12 via buddy
- extensions starting with 2 via buddy
- prefix 9 remote SIP provider
- prefix 0 local SIP provider
```

 Extensive description of this callrule table.

Enable	Dialling Pattern	Provider/Buddy	Priority	Time Segment	Comment	
<input checked="" type="checkbox"/>	<input type="text" value="0a9 ."/>	<input type="text" value="vienna (SIP)"/>	1	Any Time	<input type="text"/>	✕
<input checked="" type="checkbox"/>	<input type="text" value="0 ."/>	<input type="text" value="Silver Server (SIP)"/>	1	Any Time	<input type="text"/>	✕
<input checked="" type="checkbox"/>	<input type="text" value=" 10"/>	<input type="text" value="vienna (SIP)"/>	1	Any Time	<input type="text"/>	✕
<input checked="" type="checkbox"/>	<input type="text" value=" 11"/>	<input type="text" value="vienna (SIP)"/>	1	Any Time	<input type="text"/>	✕
<input checked="" type="checkbox"/>	<input type="text" value=" 12"/>	<input type="text" value="vienna (SIP)"/>	1	Any Time	<input type="text"/>	✕
<input checked="" type="checkbox"/>	<input type="text" value=" 2."/>	<input type="text" value="vienna (SIP)"/>	1	Any Time	<input type="text"/>	✕

+

- 0a9]. prefix 9 for Vdex buddy "vienna" to reach a remote Vdex SIP Provider. matches e.g. dialstring 97890849 and sends 07890849 to the remote Vdex (the remote Vdex uses an outgoing callrule where 0 is a provider prefix: send 7890849 to the remote SIP provider)
- 0]. prefix 0 for local SIP provider
matches matches e.g. dialstring 07890849 and sends 7890849 to the local SIP provider
- |10 matches dialstring 10 and sends 10 to the remote Vdex via buddy "vienna"
- |11 matches dialstring 11 and sends 11 to the remote Vdex via buddy "vienna"
- |12 matches dialstring 12 and sends 12 to the remote Vdex via buddy "vienna"
- |2. matches dialstrings starting with digit 2 and send these strings to the remote Vdex

Refer page 42 for more information about outgoing call rules.

8 Phone Configuration

8.1 SIP / IAX Phone Accounts

Parameter	Default Value	Description
Extension (and Authentication User Name)		The extension number can have up to 4 digits. The extension number is also the SIP username.
Secret		The authentication secret for this account. (Only for IAX phones: select „plaintext“ or „MD5“ password transmission)
Caller ID		Specifies the caller ID name. Caller ID strings are limited to maximal 79 alphanumeric characters.
Audio Codecs	μ-law a-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).
DTMF mode	rfc2833	Only for SIP phones. Set the DTMF transfer mode for this phone. Choose between RFC2833, Inband and SIP Info mode. Please note, that inband DTMF transmission using other codecs than G.711 will not work properly.
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	<p>Only for SIP phones.</p> <p>The Vdex can handle the selected maximum number of concurrent incoming and outgoing calls for this phone.</p>
Auto-Answer	unchecked	<p>Only for SIP phones.</p> <p>Each call to the phone includes the Auto-Answer SIP header. The phone must support the Auto-Answer feature.</p>
Publish extension	checked	<p>If enabled, the extension number will be appended to the main number on outgoing calls.</p>
Direct Dial In (DDI) to this SIP phone	checked	<p>If enabled, external callers can reach this SIP phone directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.</p>
Outgoing Callrule Table	No outgoing calls allowed	<p>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“.</p>
Notification Email Address		<p>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.</p>
Language	English	<p>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.</p>
Call Group	0	<p>Only for SIP phones</p> <p>Call group this SIP phone is a member of. This value is used for enabling undirected pickups (by dialling "*8"). If no value (blank) is given, this SIP phone can not be picked up undirectedly. Valid values are 0..63.</p> <p>A SIP phone can only be member of one call group.</p> <p>e.g. calls to phones of call group 1 can only be picked up undirectedly by phones which have also call group 1 in their pickup group.</p>
Pickup Groups	0	<p>Only for SIP phones</p> <p>Comma seperated list of call groups, whose member phones can be picked up by this SIP-phone undirectedly. If no value (blank) is given, this SIP phone can not pick up calls undirectedly. Valid values are 0..63.</p>
Huntgroup Memberships	none	<p>Select, if the phone should be a member of one or more hunt groups. Huntgroups are defined in the section "Extensions".</p>
Extension Profile	none	<p>Select the extension profile for this phone. Extension profiles are defined in section "Call Routing".</p>

8.1.1 Voicemail

Parameter	Default Value	Description
Enable Voicemail Box	no	Activate the voicemail box for this account.
Voicemail Box Extension	8889.	Voicemail box 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/Extensions 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 and/or Direct-Dial-In.
No Answer Timeout		The voicemail timeout defines how long the Vdex will wait until it redirects the call to the mailbox.
Voicemail Box 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.
Voicemail Notification Option	none	<p>Voice mails can be notified or transmitted via email. Preconditions are a working mail system and a configured user email address. Select between:</p> <p>„None“: No email will be sent when a new voice mail message has been stored on the mailbox.</p> <p>„Notification only“: An email will be sent to to the user to inform that a new message has been stored on the mailbox.</p> <p>„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.</p> <p>„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.</p> <p>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.</p>
Direct Dial In (DDI) to this Voicemail Box	unchecked	If enabled, external callers can reach this voicemail box directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.

8.1.2 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 the Personal Conference Room	no	Activate the personal conference room for this phone.
Personal Conference Room Extension	8888.	Personal conference room box extension number for this phone. This extension number starts with a prefix followed by a any number. The prefix is common for all personal conference rooms and can be changed in the System/Extensions Prefixes section. The personal conference room can be reached by any internal phone dialling the defined extension e.g. 8888-100. Personal conference rooms can also be reached over FXO, SIP and IAX provider trunks by using an incoming callrule configuration and/or attendant configuration and/or Direct-Dial-In.
Record Conferences	yes	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 Personal Conference 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 as that phone is considered the conference leader.
Wait for you	yes	Activated: Callers who reach the conference room hears the "Music on Hold" soundfiles until the conference owner (the leader) arrives.
Announce Joining and Leaving Participants	yes	This option makes the arrival of new participants and leaving of users audible by play back of prompts into the conference.
Direct Dial In (DDI) to this Personal Conference Room	unchecked	If enabled, external callers can reach this personal conference room directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.

External callers can reach a conference room in following ways:

- Direct-Dial-In (dialling the conference room extension)
- Using an attendant
- The conference owner can transfer a caller in the conference room by performing an attendant transfer:
 - put the caller on-hold
 - dial the conference room extension number
 - enter the PIN code
 - press the transfer button

This allows to transfer one participant after the other into the conference room. It is not necessary to publish a PIN code. Depending on the conference room configuration, the participants will hear Music-on-hold until the conference owner enters the room.

8.2 External Phones

External phones can be reached by dialling the associated internal extension number.

External phones can dial into 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 caller hears the internal Vdex dial tone and can establish outgoing calls according to the selected outgoing callrule. Calls initiated by external phones are published with the internal extension number.

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.

External phones can be used in hunt groups only, if a SIP/IAX provider is used to reach this phone. Do not use external phones in hunt groups which are reached via FXO ports (since the call answer event can not be detected via FXO).

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 to 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.
Publish this extension	checked	If enabled, the extension number will be appended to the main number on outgoing calls.
Direct Dial In (DDI) to this External Phone	checked	If enabled, external callers can reach this external phone directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.
Outgoing Callrule Table		External phones which has been authenticated are able to establish calls according the selected callrule table.

Notification Email Address		The email address of the owner of this account. The email address is mandatory when voicemail-to-email is 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.
Voicemail		
Enable Voicemail Box	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).
Voicemail Box 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.
Voicemail Box 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.
Voicemail Notification Option	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 time.
Direct Dial In (DDI) to this Voicemail Box	checked	If enabled, external callers can reach this voicemail box directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.

8.2.1 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.

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 or the ringback tone (configuration depended). 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, IAX and external phones (reached by SIP/IAX providers) 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 losing 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), a Vdex buddy default extension, an attendant or a voice mailbox.

Please note: from firmware version 2.2.21 it is possible to pick-up calls distributed by a hunt group from other phones (via directed call pickup *8xyz).

Parameter	Default Value	Description
Extension		The extension number can have up to 4 digits
Name		Enter the hunt group name
Hunt Method	Sequential	<p>Parallel: All phones of the hunt group ring in parallel</p> <p>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.</p> <p>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.</p>

Total Timeout	1 minute	Set the time how long the phones should try to ring before taking the default action.
Direct Dial In (DDI) to this Hunt Group	unchecked	If enabled, external callers can reach this hunt group directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.
Audio While Caller is Waiting	Music on hold	Select what the caller hears while waiting. Select between "Music on Hold" and "Ringback Tone"
Hunt 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.

8.3.1 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.

8.3.2 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 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 (and Authentication User Name)		The extension number can have up to 4 digits. The extension number is also the SIP username.
Secret		The authentication secret for this account.
Caller ID		Specifies the caller ID name. Caller ID strings are limited to maximal 79 alphanumeric characters.
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; inband 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 summary page). Calls to ATAs in the unreachable state will get the busy tone. If the qualify timeout is set to off, the ATA state is unmonitored. Calls to unmonitored ATAs will always sent to the ATA, no matter if the last qualify was ok or not.
Publish Extension	checked	If enabled, the extension number will be appended to the main number on outgoing calls.
Direct Dial In (DDI) to this Fax Adapter	checked	If enabled, external callers can reach this fax adapter directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.

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“.
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8.5 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 "Sound File Management" 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
Extension		Extension 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. 8887100.
Name		Enter the attendant name.
Sound File		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 "Sound File Management" section.
DTMF Inter-Digit Timeout		Select the end indication for the user input. <ul style="list-style-type: none"> - "no caller 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 default action (depending of the "retries after timeout" setting). - "Long timeout": as long as the Attendant can not decide if the given input will not match any defined user input string the attendant waits appr. 1 minute for the input. (no input performs the default action)
Retries if DTMF Inter-Digit Timeout Occurs		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 (DDI) to this Attendant	unchecked	If enabled, external callers can reach this attendant directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.
Publish Extension	unchecked	If enabled, the extension number will be appended to the main number on outgoing calls.

User Input Table

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
Caller Input		Enter the numerical order of the caller input. The maximum number of characters is 10. The caller input sequence does not have to be equal to the internal extension number. The caller input sequence is also not necessarily an extension number. One can use an attendant also to ask for a password, area code, post code, age or whatever.
Time Segment		Select in which time segment the caller input will be matched.
Forward to Extension		When the caller input matches within the selected time segment the call will be forwarded to the selected extension. If a buddy is selected as extension, the caller input is used as remote extension number.
Default action		The call will perform the default action in the case of - "no caller 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

Example

When someone calls the attendant, the attendant first plays back the message "enter the extension number". If the caller sends the user input sequence 100 the call should be connected to extension 100 during work time. Outside work time the call should be connected to attendant 72 which plays back an off time message. In the case of no user input the call should be forwarded to the voice mailbox of extension 100.

Extension	7 1 <input type="text"/>	Extension of this attendant. Please note that all attendants have a common prefix.
Name	<input type="text" value="enter extension"/>	Descriptive name of this attendant
Sound File	<input type="text" value="No sound file"/>	Sound file to be played back.
DTMF Inter-Digit Timeout	<input type="text" value="3 seconds"/>	DTMF inter-digit timeout. if "No caller input" is selected, the configured "Default Action" will be taken immediately.
Retries if DTMF Inter-Digit Timeout Occurs	<input type="text" value="2"/>	Number of retries before proceeding to the configured "Default Action", if a DTMF inter-digit timeout occurs.
Publish Extension	<input checked="" type="checkbox"/>	if enabled, the extension number will be appended to the main number on outgoing calls.
Direct Dial In (DDI) to this Attendant	<input checked="" type="checkbox"/>	if enabled, external callers can reach this attendant directly through all providers by appending the extension number to the main number. Please note that direct dial-in must be supported by the respective provider and enabled in the provider configuration.
Description	<pre>play "please enter extension" forward 100 during worktime to 100 forward 100 outside worktime to attendant 72 forward all other inputs to voicemail of 100</pre>	
	Free text description for your reference (not parsed).	

Action Table		
Caller Input	Time Segment	Forward to Extension
<input type="text" value="100"/>	<input type="text" value="Any Time"/>	<input type="text" value="Phone (SIP) <100> 100"/>
<input type="text" value="100"/>	<input type="text" value="Any Time"/>	<input type="text" value="Attendant <72> off time message"/>
Default Action		<input type="text" value="Voicemail <611> Voicemail of 100"/>

8.6 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 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 Direct-Dial-In, incoming callrules or attendants.

Parameter	Default Value	Description
Extension		Extension of this direct inward system access.
Name		Descriptive name of this direct inward system access.
Enable	yes	Enable this direct inward system access
PIN		PIN the caller has to enter to use this direct inward system access.
Caller ID		Only calls with this caller ID are authorized for using this DISA account. See below for the description of the matching algorithm.
Publish this extension	checked	If enabled, the extension number will be appended to the main number on outgoing calls.
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.

Caller ID Pattern:

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

9 Call Routing

9.1 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 at extension 81. If the call arrives outside work time then the 2nd callrule will match and the call will be forwarded to attendant at extension 82.

9.2 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 dialed 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 callrule 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.

Change call rule order: use the  symbol to drag and drop the call rule to the desired position. After dropping the entry a short symbol jitter indicates that the operation is in process. Please wait for the page reload and check the call rule order.

Caller ID Pattern:






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

9.2.1 Incoming Callrule Example 1

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.

9.2.2 Incoming Callrule Example 2

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 a non-existing number.

9.2.3 Incoming Callrule Example 3

Incoming calls from Germany should be forwarded to an 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.)

9.3 Outgoing Callrules

Outgoing 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.

IMPORTANT:

From firmware version 2.2.6 onward the call rule pattern has been enhanced. If upgrading from an earlier version you will need to change existing call rules.

You can create as many callrule tables 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 matching rule at the right side of |(i.e. "1|NXX" matches "1555" but only passes "555" to the provider.
- |XXX. - matches all numbers with at least 4 digits and passes the complete number to the the provider)
- a| - adds and removes a prefix (i.e. "2a1|NXX" matches "1555" but passes "2555" 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 00xxxxxxx. 21|00.
- The user selects all numbers with prefix 1: 1|.
- Route to US numbers without prefix: |001.
- Emergency calls in Canada, US: |911
- Emergency calls in Germany with prefix 0: 0|11X
- Route all numbers (except extensions) with more than 4 digits: |XXX.
- Add country- (0043) and city-code (1) to numbers starting with 1-9 and more than 5 digits 00431aZXXX.
- Route the buddy extension 30: |30
- Route buddy extensions starting with digit 3: |3.

see also the "Buddies" section on page 25.

9.3.1 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:

Name	<input type="text" value="out FXO"/> <small>name of this callrule table.</small>					
Description	<div style="border: 1px solid black; padding: 5px;"> national and emergency calls over FXO lines </div> <small>Extensive description of this callrule table.</small>					
Enable	Dialling Pattern	Provider/Buddy	Priority	Time Segment	Comment	
<input checked="" type="checkbox"/>	<input type="text" value=" 02."/>	Telstra Line 1 (FXO1)	1	Any Time	<input type="text" value="national"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 02."/>	Telstra Line 2 (FXO2)	2	Any Time	<input type="text" value="national"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 02."/>	Telstra Line 3 (FXO3)	3	Any Time	<input type="text" value="national"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 02."/>	Telstra Line 4 (FXO4)	4	Any Time	<input type="text" value="national"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 000"/>	Telstra Line 1 (FXO1)	1	Any Time	<input type="text" value="emergency"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 000"/>	Telstra Line 2 (FXO2)	2	Any Time	<input type="text" value="emergency"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 000"/>	Telstra Line 3 (FXO3)	3	Any Time	<input type="text" value="emergency"/>	⊗
<input checked="" type="checkbox"/>	<input type="text" value=" 000"/>	Telstra Line 4 (FXO4)	4	Any Time	<input type="text" value="emergency"/>	⊗
						⊕

9.3.2 Outgoing Callrule Example 2

The Vdex is connected to 1 SIP provider and 1 FXO provider line. Calls with numbers of more than 4 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	<input type="text" value="out 2"/> name of this callrule table.				
Description	<pre>primary provider: Sipcall backup: FXO line</pre> Extensive description of this callrule table.				
Enable	Dialling Pattern	Provider/Buddy	Priority	Time Segment	Comment
<input checked="" type="checkbox"/>	<input type="text" value=" XXX."/>	<input type="text" value="SIPCALL (SIP)"/>	<input type="text" value="1"/>	<input type="text" value="Any Time"/>	<input type="text" value="national"/>
<input checked="" type="checkbox"/>	<input type="text" value=" XXX."/>	<input type="text" value="Telstra Line 1 (FXO1)"/>	<input type="text" value="2"/>	<input type="text" value="Any Time"/>	<input type="text" value="national"/>

9.4 Extension Profiles

Extension profiles define 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 extension profile call forwarding is that the voice mailbox no-answer-timeout is longer than the extension profile no-answer-timeout. It is also possible to forward non-answered or busy calls to an attendant for further call treatment. Phones can be member of only one extension profile. Each phone can also be assigned its own extension profile.

Parameter	Default Value	Description
Name		Descriptive name of the extension profile
No-Answer Timeout	10	Number of seconds a call is ringing before proceeding with the no-answer action.
Action on No-Answer	Hangup	Action taken after a "No-Answer Timeout" occurs.
Action on Busy	Hangup	Action taken if the called extension is busy.
Extension Profile Members		Select the phones for the profile. Phones listed at the nonmember side are phones which are free for extension profile selection. Once a phone is assigned an extension profile it will not be listed as free non-member phone.

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 Soundfiles

10.1 Sound File Management

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.

10.1.1 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 Sound File Management 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 summary 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.

10.1.2 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 summary page.

10.2 Music on Hold

The Vdex uses "Music on Hold" (MoH) when a call is 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 play 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.

11 Voicemailbox

Global voicemail settings can be changed at the section "Advanced: Voicemail Setup" (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).

Voice mail to email: Vdex offers the feature to email the voice message as an 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.

11.1 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 displayed 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.

11.2 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.

12 Conference Rooms

Each 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.

13 Diagnostics

13.1 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.

System Logs

The operating system of the Vdex is Linux. The log messages from `/var/log/messages` are shown. Please refer to any basic Linux introduction.

PBX logs

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

Call logs

The call detail records of Asterisk are shown:

- Start: Start of call (date/time)
- Src: Caller ID number
- Dst: Dialed 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.

13.2 Ping/Traceroute

Ping

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.

Traceroute

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.

13.3 ARP Table

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

13.4 Commandline Interface

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

14 Examples and tips

14.1 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

14.2 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

14.3 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 callrule

attendant options:

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

14.4 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 pickup of calls, if the call group of the ringing extension is listed in the pickup group of the phone which performs the call pickup:

- dialling *8

The phone assignment to a call group and the pickup group definition can be done at each SIP phone configuration.

14.5 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.

Example:

The BLF configuration of the **Linksys** SPA942 can be done in the "Phone" section.

Use the settings of Line Key 1 to 4 for BLF support:

Extension: Disabled
 Short Name: monitored extension number or descriptive name
 Share Call Appearance: shared
 Extended Function: fnc=blf+sd+cp;sub=extension@host;ext=extension@host;
 extension: number of the monitored extension
 host: hostname or IP address of the Vdex

Line Key Extended Function:

Server Type: Asterisk

SPA932 Call Pickup Code: *8#

SPA942 supports 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.

Screenshot (monitor extension 20 at Vdex with IP address 10.0.0.170 and SIP Port 5061)

Line Key 4			
Extension:	Disabled	Short Name:	Thomas
Share Call Appearance:	shared		
Extended Function:	fnc=blf+sd+cp;sub=20@10.0.0.170:5061;ext=20@10.0.0.170:5061;		
Line Key Extended Function			
Subscribe Expires:	1800	Subscribe Retry Interval:	30
Subscribe Delay:	1	Server Type:	Asterisk
SPA932 Call Pickup Code:	*8#		

14.6 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.

14.7 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:

Dialog

Line DN: SIP/10

Line Prefix: AST

Caller ID Name: VDEX

Context

Outgoing: internal

Agent:

Asterisk Connection

Host IP: 10.0.0.170

Port: 5038

User: active

Password: 6gge9d0d

Enable Logging to file

Activa TAPI Service Provider (Activa TSP) V1.4.6.1015 \$Revision: 440 \$
see <http://activa.sourceforge.net>

OK Cancel

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

15 Voice Notification Remote Control Interface Specification

15.1 General

Configured voice notifications can be remotely controlled via authenticated http requests. Currently the following request are available:

- **start request**
triggers (starts) a voice notification
- **status request**
retrieves the status of a notification. The status can be one of *pending*, *ok*, *failed* or *unknown*.
- **log request**
retrieves the status and log file of a voice notification.
- **clear request**
deletes all log and status information of a voice notification.

15.2 Http Authentication and Http Request Syntax

15.2.1 Http Authentication

In order to send http requests to the Vdex system *HTTP Basic Authentication* must be used. There are two ways authentication credentials can be supplied:

- authentication as administrator ('admin')
- authentication with the voice notification's name

If authentication as administrator is used, all configured voice notifications can be accessed. If the voice notification's name is used, only this voice notification can be accessed.

15.2.2 Http Request Syntax

The URL to access the voice notification remote control interface is:

`http://<vdex-system-ip-address-or-hostname>/astai/voicenotification-remote.php`

The following http query parameter can or must be used:

- **"vn"**
voice notification name, is mandatory if authentication as administrator is used
- **"action"**
request type, mandatory, can be one of
 - **"start"**
 - **"status"**
 - **"log"**
 - **"clear"**
- **"tid"**
transaction id (foreign key) of the request. Is mandatory for all requests but "start".
The value must be alphanumeric and 1 to 16 characters long.

15.2.3 Http Request Examples

The following examples demonstrate how to issue remote voice notification requests. It is assumed that the Vdex system has the IP address 1.2.3.4.

- start a voice notification (with name "test" and transaction id "1234")
http://1.2.3.4/astauti/voicenotification-remote.php?action=start&vn=test&tid=1234
- query the status of a voice notification (with name "test" and transaction id "1234")
http://1.2.3.4/astauti/voicenotification-remote.php?action=status&vn=test&tid=1234
- query the status of a voice notification with log file contents (with name "test" and transaction id "1234")
http://1.2.3.4/astauti/voicenotification-remote.php?action=log&vn=test&tid=1234
- clear the status of a voice notification (with name "test" and transaction id "1234")
http://1.2.3.4/astauti/voicenotification-remote.php?action=clear&vn=test&tid=1234

15.3 Http Response Syntax

After issuing a remote voice notification request the system answers with the following responses in the body of a html page. Generally there are 2 types of responses:

15.3.1 Success Responses

If the Vdex system has processed a given request correctly, it responds with an easy to parse success response.

Syntax:

VNS-REQUEST-OK: vn=<vn>;action=<action>; tid=<tid>; status=<status>

The value of "status" can be one of:

- **'VNS-STATUS-UNKNOWN'**
The system does not have any status information of the given voice notification.
- **'VNS-STATUS-PENDING'**
The given voice notification is currently active.
- **'VNS-STATUS-OK'**
The given voice notification was completed successfully, i.e. one of the callees has entered the correct confirmation pin code.
- **'VNS-STATUS-FAILED'**
The given voice notification was not completed successfully.

Note: It is guaranteed, that character ';' (semi-colon) is only used as key value pair delimiter ';' in the response status line - i.e. no value will contain a semi-colon.

15.3.2 Error Responses

If the Vdex system cannot handle a given request, it responds with an error response

Syntax:

VNS-REQUEST-ERROR: <descriptive-error-msg-what-went-wrong>

15.3.3 The "log" Success Response

When a "log" voice notification request was issued, the system appends the current contents of the given voice notification's log file to the success status line . This is done within a html <pre> tag in the body of the html response.

15.4 Response Examples

15.4.1 Response of a start request (with name "test" and transaction id "1234")

```
<!DOCTYPE html PUBLIC "-//W3C//DTD HTML 4.01 Transitional//EN" "http://www.w3.org/TR/html4/transitional.dtd">
<html>
<head>
<title>VNS-REQUEST-OK</title>
</head>
<body>
VNS-REQUEST-OK: vn=test; action=start; tid=1234; status=VNS-STATUS-PENDING
</body>
</html>
```

15.4.2 Response of a status request (with name "test" and transaction id "1234")

```
<!DOCTYPE html PUBLIC "-//W3C//DTD HTML 4.01 Transitional//EN" "http://www.w3.org/TR/html4/transitional.dtd">
<html>
<head>
<title>VNS-REQUEST-OK</title>
</head>
<body>
VNS-REQUEST-OK: vn=test; action=status; tid=1234; status=VNS-STATUS-FAILED
</body>
</html>
```

15.4.3 Response of a log request (with name "test" and transaction id "1234")

```
[header skipped]
<body>
VNS-REQUEST-OK: vn=test; action=log; tid=1234; status=VNS-STATUS-FAILED
<pre>
2011.08.30-18:23:43 INFO initiated by test@1.2.3.4 tid="1234"
2011.08.30-18:23:44 INFO voice notification client running
2011.08.30-18:23:44 INFO call to 123 starting
2011.08.30-18:23:44 INFO call to 123 SIP-INVITE was sent
2011.08.30-18:23:44 INFO call to 123 proceeding
2011.08.30-18:23:57 INFO call to 123 is now active
2011.08.30-18:23:57 INFO call to 123 playing 'short-soundfile.alaw'
2011.08.30-18:23:58 INFO call to 123 was hung up by remote side
2011.08.30-18:23:58 INFO call to 123 now hung up
2011.08.30-18:23:58 INFO voice notification not confirmed
2011.08.30-18:23:59 INFO voice notification client terminating
2011.08.30-18:23:59 INFO VNS_RESULT_ERROR voice notification not confirmed
</pre>
</body>
</html>
```

15.4.4 Error Response of a start request (with name "test" and transaction id "1234")

[header skipped]

<body>

VNS-REQUEST-ERROR: bad request, duplicate transaction-id "1234" for voice notification "test"

</body>

</html>

15.5 Demonstration Perl Script + Perl Module

An demonstration perl script and module can be found at:

<http://www.vdex.at/files/voice-notification-contrib.tar.bz2>

Please note that this software is used for demonstration and test purposes only and is provided AS IS.