# VOICE&DATA ROUTER ITX 495 01

# **PRODUCT DOCUMENTATION**



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# **PRODUCT OVERVIEW**

The device is a multifunction VoIP gateway with application in SIP Trunking, PBX IP Trunking and PSTN access for local IP telephony solutions. There are many usage applications available. It is a perfect solution to decrease costs of all your calls.

### PRODUCT FEATURES

- OS LINUX, SD card, Asterisk SW
- 2 x ETHERNET 10/100 BT or 1 x 10/100 BT and 1 x 10/100 FX\*

\* WAN ETHERNET fiber optic interface (SC DUPLEX) can be single mode or multi mode; for single mode device can use only 1 optical fiber; Auto MDIX option – for both Ethernet interfaces

- 4 x E1/T1 120/75 Ohm PRI DSS1/QSIG or R2 MFC ports software configurable
- Optional interfaces:
  - BRI (8/16 x BRI Uk0 or 4 x BRI S0)
  - GSM (12 x GSM 2G)
  - Analogue (32 x FXS, 16 x FXO, 6 x E&M)
- Console port RS232 (RJ45)
- USB 1.1 host port
- VoIP support: SIP, RTP, UDPTL, T.38
- Codecs: G.711, G.726, G.729a/b, G.723.1
- G.165 / G.168 2004 Echo Cancellation up to 128 ms
- Silence detection / suppression and comfort noise generation (VAD/CNG)
- Simultaneous calls
  - VoIP G.711: 48/64
  - VoIP G.726, G.729 a/b: 16/32
  - VoIP G.723.1, GSM: 16/24
  - TDM not limited
- G.165 / G.168 2004 Echo Cancellation up to 128 ms
- Multiple codec support
- Fax over IP support, including T.38
- Remote management capabilities
- SIP signaling
- SNMP
- LCR
- AOC generation
- Diagnostics of all interfaces
- Alarm indication
- Routing to another E1/T1 if there is no more free capacity for simultaneous calls
- Call Forwarding
- Call Hold/Transfer
- Music On Hold
- Speed Dial
- CLIP/CLIR
- Automatic Attendant
- DTMF generation and detection
- Two step dial (ISDN / MFC R2 and than DTMF) for alternative carrier
- Call progress tone generation
- Carrier tone generation and detection
- Caller ID generation and detection (FSK)
- Silence detection / suppression and comfort noise generation (VAD/CNG)

- Call permissions
- Calling Groups
- Ring Groups
- Call Pickup
- Announcement playback
- Remote management capabilities
- HTTP, SNMP, SSH, SCP, SFTP, TFTP
- Diagnostics functions

### VARIANTS

#### **1U version**

• ITX 495 01 Voice&Data Router – desktop with rack mount option (4 x E1/T1, optional 12 x GSM interface, optional 32 x analogue interface, optional 4 x BRI S0, optional 16 x BRI Uk0)

ITX 495 01	. a b c	def	ghi	j – Voice&Data Router
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	Optic Communication	X = 2 optic fibers A = Type A, 1 optic fiber B = Type B, 1 optic fiber
	Distance	1 = 2km 2 = 15km 3 = 40km 4 = 80km
	Wave Length	1 = 1300nm 2 = 1500nm
	Optic Mode	0 = none 1 = SM (Single Mode) 2 = MM (Multi Mode)
	E1 Interface	0 = none 
	Power Supply	1 = 230V, 48V 
	GSM Interface	0 = none - 1, 2, 3, 4, 5, 6, 7, 8, 9 a = 10, b = 11, c = 12
	Analogue Interface	0 = no 1 = yes
BRI	Interface	0 = no 4 = 4 BRI S0 7 = 8 BRI S0 5 = 8 BRI Uk0 8 = 16 BRI S0 6 = 16 BRI Uk0 9 = 8 BRI Uk0, 8 BRI S0
Compr	ession	0 = no SIP 1 = 16 channels G.729 2 = 32 channels G.729 48 channels G.711 64 channels G.711

#### 19" Rack 6U version

- ITX 402 41 Voice&Data Router 6U rack card (4 x E1/T1)
- ITX 222 05 19" rack 6U
- **ITX 802 05** Converter DC/DC + 5V/20A

#### ITX 402 41 . a b c d e – Voice&Data Router (rack card)

Optic Communication	X = 2 optic fibers A = Type A, 1 optic fiber B = Type B, 1 optic fiber
Distance	1 = 2km 2 = 15km 3 = 40km 4 = 80km
Wave Length	1 = 1300nm 2 = 1500nm
Optic Mode	0 = none 1 = SM (Single Mode) 2 = MM (Multi Mode)
Compression	1 = 16 channels G.729 48 channels G.711 2 = 32 channels G.729 64 channels G.711

### **APPLICATIONS**

- Voice & Data Router
- VoIP, TDM PBX (SW Asterisk)
- TDM VoIP Gateway
- Signaling converter R2 MFC / ISDN DSS1 SIP, Routing to another E1 if there is no more free capacity for simultaneous calls
- Concentration of max. 4xE1 (considering the limitation of max. number of simultaneous calls depending on the codec used)
- Multiple Gateway (possibility to connect several customers and divide them)
- SIP PROXY Server

#### Voice&Data Router / VoIP, TDM PBX







**INDERSTRA** SYSTEMS FOR VOICE AND DATA COMMUNICATION



#### Signaling converter R2 MFC / ISDN DSS1 - SIP



Concentration of max. 4xE1/T1 (considering the limitation of max. number of simultaneous calls depending on the codec used)



Multi Gateway (possibility to connect several customers and divide them)





#### Extending



#### SIP PROXY Server





### **TECHNICAL SPECIFICATION**

#### **IP Telephony Protocols**

- SIP RFC 3261
- SDP RFC 2327
- RTP RFC 1889, RFC 2833, RFC 3389

#### **ISDN Signaling**

- Euro ISDN EDSS 1/ETSI PRI/ NET5
- ETS 300 011 (ISDN PRI UNI)
- ETS 300 012-1 (ITU T I.430)
- ETS 300 402-2 (ITU-T Q I.921)
- ETS 300 403-1/2 (ITU-T Q.931)
- ETS 300 102-2 (ITU-T Q.931)
- ISDN speech, audio and data (Fax Gr4, UDI 64)

#### CAS MFC R2 Signaling

- ITU T from Q.440 to Q.480, Q.490
- ITU T from Q.421 to Q.424
- Basic Technical Requirements for Digital Switching System and PCM Transmission System
- National variants, DTMF dial

#### **GSM** parameters

- Quad-Band 850/900/1800/1900 MHz
- GPRS multi-slot class 10/8
- GPRS mobile station class B
- Compliant to GSM phase 2/2+
  - Class 4 (2W @850/900 MHz)
    - Class 1 (1W @1800/1900 MHz)
  - Voice specification
  - Tricodec

0

- Half rate (HR)
- Full rate (FR)
- Enhanced Full rate (EFR)

AMR

- Half rate (HR)
- Full rate (FR)
- Echo suppression

#### FAX and Modem Support

• FAX Relay T.38

Supported modulations:

- 。 V.21 Ch2
- o V.27 2400 / 4800
- o V.29 7200 / 9600
- V.17 7200 / 9600 / 12000 / 14400)
- FAX Pass-Through transport of fax modulations over G.711 connections
- Modem Pass-Through transport of modem modulations over G.711 connections.

#### Voice Processing

- Supported voice codec standards
  - G.711 (A-law, μ-law)
  - G.723.1 and G.723.1 A
  - o G.726
  - o G.729 Annex A & B
  - o GSM

- G.165 / G.168 2004 Echo Canceller up to 128 ms
- Robust jitter buffer
- DTMF generation and detection
- Call progress tone generation
- Carrier tone generation
- Caller ID generation and detection
- Voice activity detection and comfort noise
- Announcement playback
- Timeslot interchange for TDM to TDM traffic processing (no compression, no delay)

#### Voice Channel Density

Supported Voice	Channel Density			
Codecs	16/48 calls	32/64 calls		
G.711	48	64		
G.726	16	32		
G.729 a/b	16	32		
G.723.1	16	24		
GSM	16	24		
Т.38	16	16		

#### Voice Routing

- Local switching
- Low Cost Routing
- Analyze called party number
- Analyze calling party number
- Called / calling party number modification
- Call permissions
- Multiple fail-over routes
- Two step dialing (DTMF)
- Calling groups
- Ringing groups

#### **Data Routing**

• Basic Routing

#### **QoS Marking**

- QoS pre-router
- TOS /DiffServ
- Supported QoS marking

#### Networking

- DHCP support and capabilities
- Static Routing

#### Management

- Web management GUI
- Terminal access
  - Local control interface RS232
  - Remote LAN/WAN interface (SSH protocol)
- Inoteska UniMan software
- SNMP v1/2c

#### **Power Supply**

19", 1U rack mount

- 85V 260 V AC or -40V to -65V DC
- Frequency: 48Hz to 52 Hz

19", 6U rack

• 48V DC

#### Max.power consumption

Max. 50W

#### Dimensions

19", 1U rack mountable 44 x 282 x 485 mm (h x d x w)

#### Weight

Approx. 3,5kg (real weight depends on device HW configuration)

# **OPERATING INSTRUCTIONS**

#### **Operating Environment**

Install the device in a place where:

- Operating temperature: 0° C to 55° C
- Storage temperature: -10° C to 65° C
- Humidity: up to 80%, non-condensing

#### Interfaces



#### E1/T1 interface

#### **RJ 45 connector**

1 – input wire to device	Rx –
2 – input wire to device	Rx +
4 – output wire from device 5 – output wire from device 6 –	Tx – Tx +

- 7
- 8 –



#### Ethernet 10/100Base-T interface

#### **RJ 45 connector**

1 – transmit from device	1 –	transmit from	device	
--------------------------	-----	---------------	--------	--

- 2 transmit from device
- 3 receive to device
- 4 –
- 5 –

6 - receive to device

7 – 8 –





Tx +

Tx –

Rx +

Rx –

#### **Analog interfaces**

Analog extension card can contain 1 up to 8 modules.

#### Available modules

- FXS module quad
- FXO module dual
- E&M module dual

	1	2	3	1	5	6	7	Q
	1	2	3	4	5	0	1	0
	A11	A21	A31	A41	A51	A61	A71	A81
	B11	B21	B31	B41	B51	B61	B71	B81
	A12	A22	A32	A42	A52	A62	A72	A82
	B12	B22	B32	B42	B52	B62	B72	B82
	A13	A23	A33	A43	A53	A63	A73	A83
	B13	B23	B33	B43	B53	B63	B73	B83
	A14	A24	A34	A44	A54	A64	A74	A84
	B14	B24	B34	B44	B54	B64	B74	B84
						E61 M61	E71 M71	E81 M81
						E62 M62	E72 M72	E82 M82

Location of analog modules on analog extension card Position No. 1 to 5: for FXS/FXO modules Position No. 6 to 8: for FXS/FXO/ E&M modules







Note: In description of connector, e.g. signal A23 means that it is a-wire from second module (first index) of third interface on module (second index).

BRI interface				NT	TE
Pins are connec	cted as TE in default configura	ation.		1 –	•=
				2 – 3 – Rx+ 4 – Tx+ 5 – Tx- 6 – Rx- 7 –	Tx+ Rx+ Rx- Tx-
4x BRI S0 – jur	nper settings			8 –	
Interface	NT+Phantom Power	NT		TE	<u> </u>
BRIA	PR22-ON	PR4 PR5 PR6 PR7	1-2 1-2 1-2 1-2	PR4 PR5 PR6 PR7	2-3 2-3 2-3 2-3
BRIB	PR23-ON	PR8 PR9 PR10 PR11	1-2 1-2 1-2 1-2	PR8 PR9 PR10 PR11	2-3 2-3 ) 2-3   2-3
BRIC	PR24-ON	PR12 PR13 PR14 PR15	1-2 1-2 1-2 1-2	PR12 PR13 PR14 PR15	2 2-3 3 2-3 4 2-3 5 2-3



#### **GSM** interface

Antenna connector: GSC connector coaxial SMT male 50R 6GHz

#### **CONTROL RS-232 connector**



CANNON - Female for D09F cable	RJ - 45
-	1
-	2
-	3
2	4
3	5
-	6
-	7
5	8
-	-

#### LED diodes

Interface	Led diode	Led diode	Status
	green	yellow	
E1/T1	Off	Off	Not enabled
	Off	On	Not connected
	Fast		CRC error or SLIP
	Slow		ISDN – No DLL
		Slow	AIS detected
		Fast	LFA or RRA detected
	On	Off	ОК
Ethernet	Off	Off	Line not connected
	On	(flashes during Reception/ Transmission)	Line active
BRI S0/Uk0	Off		L1 down
	On		L1 up

Off - no light, On - light, Slow - flashes slow (period 1.6sec), Fast - flashes fast (period 0.2sec, 5x/sec)

- CRC Cyclic Redundancy Check error
- No DLL no Data Link Layer active
- Alarm Indication Signal Transmitted signal is constant with data value Log1 Loss of Frame Alignment Indicates synchronization error in 0<sup>th</sup> channel Receive Remote Alarm Indicates remote device alarm AIS
- LFA
- RRA

# MANAGEMENT

Voice&Data Router is supplied with default configuration. Device can be configured using:

- Terminal access
- Inoteska Web Manager (GUI)
- Inoteska UniMan

**Terminal access** allows direct access to the Linux console and Asterisk command line interface. User has full low level control of the whole system. Knowledge of OS Linux, Asterisk SW and configuration files structure is necessary.

Terminal access is available:

- Locally via RS 232 control interface (baud rate 115200 bps, 8 bits, 1 stop bit, no parity)
- Remotely over the Ethernet interface using SSH protocol

Login parameters for terminal access:

- Login name: root
- Default password: inoteska

#### NOTE:

Low level configuration using terminal access is not the scope of this document. Unqualified change of system configuration can make the device non-functional. Please contact the producer in case of difficulties!

**Inoteska Web Manager (GUI)** provides simple and user-friendly interface for device configuration and supervision using web browser.

**Inoteska UniMan** is universal software used for communication with Inoteska equipment which supports TCP / UDP protocol. UniMan operates under OS Windows (Win7/Vista/XP/2000). UniMan provides text or graphical mode for device configuration.

### **WEB MANAGER**

Web Manager is Graphic User Interface for device configuration and supervision. It was tested with following web browsers: **Opera, Google Chrome, Mozilla Firefox**. **MS Internet Explorer** is not fully supported and it is not recommended to use with Voice&Data Router.

Default IP address setting:

- LAN: 192.168.1.100
- WAN: 10.1.1.100

Initial screen serves for user (system administrator) login.

voice adata router	inoteska
Inoteska Configuration Engine Username: admin Password: Login	
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Default login parameters:

- Username: admin
- Password\*: inoteska

\*Change of password can be done in <u>Options</u> menu.

After successful logging in **Home** page with device hardware configuration and identification will be displayed.

Device configuration shows available TDM interfaces (E1/T1 trunks, BRI trunks, GSM trunks, analogue trunks and users) and configured VoIP interfaces (SIP trunks and users). Voice&Data Router is always supplied with 4 E1/T1 interfaces. BRI, analogue and GSM interfaces are optional and their number depends on the hardware configuration. SIP users and SIP trunks shows the number of interfaces configured by device administrator.

Identification provides basic information: device type (product code number), serial number, main board identification and software version.

					_	
Δ.						
- C		 -		<b>U</b> 1		

GUI-version: 1.8 / Board Type: 1

# voice&data router

Users Trunks	Traffic Globals System Diagnostics	
em Home		Uptime : 08:12:56 up 2 min, 0 users, load average: 0.23, 0.14
ice Configuration		
	Available Interfaces	Count
	E1/T1 Trunks	4
	BRI Trunks	4
	GSM Trunks	4
	Analog Trunks	6
	Analog Users	8
	SIP Trunks	2
	SIP Users	5
ntification Hardware		
Device Type	ITX4024102	
Serial Number	402410270008	
Main Board	ITB 166 05 + M82810-12 RevB	
Software		
Operating System	Linux 2.6.11.7-1.08.5.3malindi (#2 Thu Feb 16 12:2	1:06 CET 2012)
MSP Firmware	Test_v2_04_82027_01	
FPGA Design	2.4	

#### How to configure Voice&Data Router

#### Step 1

Make basic system configuration

- IP setting assign IP addresses to connect to LAN and WAN network. Configure DNS servers. In some cases it can be necessary to configure routing table for static IP routing.
- Firewall setting configure firewall eventually NAT to keep Voice&Data Router secure from the Internet attacks.
- Date/Time setting select time zone and actualize time in Voice&Data Router

#### Step 2

Configure users / trunks. This depends on usage application

- Gateway only trunk's configuration is required usually. In some cases also users can be needed.
- PBX both user's and trunk's configuration is required

Depending on the application create the users (SIP or analogue). Configure trunks to connect to PSTN or VoIP provider (usually E1/T1 or SIP), to connect to PBX or other gateway (E1/T1, SIP or analogue trunk), to connect to GSM network (GSM interface), etc. as per device function.

#### Step 3

Configure call traffic - call routing between users and trunks. Call traffic consists of

- calling rules
- dial plans

Calling rules define set of rules for analysis of dialed number eventually together with caller identification number and for selection of destination of the call. Voice&Data Router in function of gateway without users needs to specify only trunk-to-trunk calling rules. PBX device or gateway with users requires also calling rules for user-to-user calls, user-to-trunk calls and trunk-to-user calls.

Dial plan is set of calling rules. Create dial plans from existing calling rules for users and for trunks as needed. When all dial plans are ready go back to the user's or trunk's configuration to assign dial plan for users / trunks.

#### Step 4

Configure all other options offered by Web Manager.

Most of the changes you made are being applied to your Voice&Data Router after click on Apply Changes filed in upper right corner. Some settings (e.g. change of IP settings, routing tables, date/time setting, etc.) are done automatically, without need to Apply Changes.

GULversion :	1.0 (Board	Tyme : 1
001-961 31011	1.07 Doara	Type T

Apply Changes Logout

If you want to end the work with Web Manager click on Logout in the upper right corner.

Web Manager has implemented simple context help system. Question mark icon is displayed everywhere it is available. Moving a mouse pointer over this icon will display tooltip frame with description of selected parameter/setting.

Interface line mode selection:     E1 2.048 Mb/s line with 30 voice channels.     F1 1.544 Mb/s line with 23 voice channels.

#### **Complete structure of Web Manager menu**

#### Home

#### **Users Menu**

- **SIP Users** •
- Analog Users •

#### **Trunks Menu**

- SIP Trunks •
- E1/T1 Trunks •
- **BRI Trunks** •
- Analog Trunks
- **GSM Trunks**

#### **Traffic Menu**

- Calling Rules •
- **Dial Plans**
- AOC Tables Settings

#### **Globals Menu**

- **SIP Settings** •
- **RTP/UDPTL Settings** •
- Call Groups
- **Ring Groups** •
- Carriers •
- Permissions
- Services Codes •
- **CDR Settings** •
- **Announcement Settings** •
- CAS National Variants •

#### System Menu

- **Routing Tables** •
- SNMP •
- **Firewall** •
- **Options** •
- **Backup and Restore Upload** •
- Firmware Update
- Date and Time

#### **Diagnostics Menu**

- Port Status •
- Active Services •
- Identification
- System Logs •
- **CLI Emulator** •
- Debug messages
- System Status
- File Editor

#### **Specials Menu**

AOC Web Update

- IP settings •

#### **USERS MENU**

Users menu offers configuration of user extension lines. Voice&Data Router supports SIP users and analogue FXS users. Analogue users are optional and they depend on hardware configuration.

#### SIP users

SIP users configuration allows creating and managing user accounts for SIP phones. SIP users main screen displays table of configured SIP user extensions. Use button **Create New User** to add new account for SIP user and use button **Edit** to modify existing user account. To remove SIP user click on **Delete** button. User can be removed only if does not exist any reference to this user in calling rules and ringing groups.

GUI-version : 1.0 / Board Type : 1							Logout inoteska
	Home	Users Trunks	: Traffic Globals	System Diagnostics	Specials		
U	Users > SIP Users						
s	IP Us	ers Settings			Create New User	Modify Selected Users	Delete Selected Users
	List of	User Extension	าร				
		Extension	Full Name	Host Address	DialPlan		
		10	Jack Daniel's	dynamic	from_users	Permit/E	Deny Edit Delete
		11	Johnnie Walker	dynamic	from_users	Permit/E	Deny Edit Delete
		12	Jim Beam	dynamic	from_users	Permit/E	Deny Edit Delete

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In addition there also exists possibility to change or remove multiple users in one action. In this case use the checkboxes in the first column to select users you want to modify and click either **Modify Selected Users** or **Delete Selected Users** button.

When you are creating new SIP user or when you are changing parameters of the existing one you are switched to user's details form, with all available SIP user account settings. Here you can configure available parameters and click **Update** button to store new setting.

#### **General Parameters**

#### Extension

Extension number associated with this SIP user line. This number is also used as user name for authentication.

#### Password

Password for the user authentication. If password is not set, authentication will be disabled and Voice&Data Router will never ask for user authentication in this case.

#### Name

A character-based display name used for caller identification (i.e. Bob Jones). Note that it will overwrite display name configured in the user phone.

#### Caller ID

Caller identification number. Extension number is used for the caller identification by default. Use this field only if you need to overwrite default identification.

#### **Dial Plan**

Select dial plan to handle calls coming from user phone to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

VOICE&DATA ROUTER	
GUI-version : 1.0 / Board Type : 1	Logout .
voice&data router	iteska
Home Users Trunks Trattic Clobals System Diagnostics Specials	
Edit User Extension - 10	
General       Extension @ 10       Name @ Jack Daniel's       DialPlan @ from_users v         Password @ heslo       Caller ID @       Image: Caller ID @       Image: Caller ID @         Host Options       Host Options       Host Port @ 5060       Image: Caller ID @         Host Type @ dynamic v       Host Address dynamic       Host Port @ 5060         Enable Voicemail for this User @       Access PIN code @ Mailbox @ 10       Email Address @         Codecs       @       Mailbox @ 10       Email Address @         VoIP Settings       VoIP Settings       Image: Caller ID @       Image: Caller ID @	
Qualify 🗹 😰 👔 NAT 🗹 🎔 Can Reinvite 🗹 🤍 DTMF Mode 🤍 RFC2833 💌 insecure 🆤 no 💌	
Pickup Group <sup>(2)</sup> 1	
User's Services — Call Forward 🗹 🧟 Do Not Disturb 🗹 🧐	
Permissions — User Permission Group 🖗 Without limitation 💌 Call Limit 🧐	
Update Cancel	
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#### **Host Options**

#### Host Type

Voice&Data Router supports two methods how to connect to user SIP phone. Please select host type **dynamic** to use dynamic SIP registration procedure. Extension number and password from general section will be used for registration authentication. **Static** host type is intended for use as permanent static connection to the dedicated IP address. Note that you can restrict user's access for limited IP addresses or networks by **Permit/Deny** setting in main SIP users' screen.

#### Host IP address

Address of the user SIP phone in case of static host type (you can use numeric IP address or the host name).

#### Host Port

SIP protocol port number for this user.

#### **Voice Mail Setting**

#### **Enable Voicemail for this User**

Enable/Disable user voice mail account.

#### Access PIN Code

User's Personal Identification Code to access voice mail mailbox (PIN string can contain only digits).

#### Mailbox

Voice mail mailbox name. It can be different from the extensions number.

#### **Email Address**

The e-mail address to notify about new messages in the mailbox (i.e.bobjones@bobjones.com).

#### Codecs

Here you can define allowed voice compression codecs in order of preference. Voice&Data Router supports following voice codecs:

- G.711 A-Law
- G.711 µ-Law
- G.729 Annex A/B
- G.726-32
- G.723.1
- GSM
- iLBC

Up to five different codecs with defined preference can be configured for every SIP user.

#### **VoIP Settings**

#### Qualify

Enable this feature to check if the phone device is reachable. Qualify will send a SIP OPTIONS command regularly to check that the device is still online. If the device does not answer within the configured period Voice&Data Router considers the device off-line for future calls. Expiration time is configured in seconds (default is 2 seconds). This feature may also be used to keep a UDP session open to the device that is located behind the NAT. This can be used in conjunction with the NAT setting.

#### NAT

This setting allows communication with phone devices behind the NAT. It works properly in conjunction with Qualify setting in order to keep open the connection from Voice&Data Router to the peer behind the NAT.

#### Can Reinvite

Enabling this option causes Voice&Data Router to attempt to route the RTP media stream directly between SIP endpoints, bypassing Voice&Data Router.

#### **DTMF Mode**

Default mode for sending DTMF digits.

#### Insecure

Insecure feature specifies how to handle incoming SIP session and user matching. If it is disabled normal IP-based matching applies and the authentication is required (when password is set for SIP user extension). Set insecure to **port** to allow matching of SIP user by IP address without matching port number. Set insecure to **invite** to remove the requirement for authentication of incoming INVITE messages. And use **very** value to allow both, the matching of user by IP address without matching port number and to remove the requirement for authentication of incoming INVITE messages.

#### **Other Options**

#### Pickup Group

Pickup group allows to pickup fellow workers' calls. Users from the same group can pickup incoming calls each other using code defined in <u>Services Codes</u> menu.

#### **User's Services**

Here you can permit/deny special services for extension user. Any permitted service can be activated and deactivated by service codes defined in <u>Services Codes</u> menu. Voice&Data Router supports following services:

- Call Forward (unrestricted / when busy / when no answer)
- Do Not Disturb

#### **User's Permission**

#### **User's Permission Group**

Select permission group to restrict outgoing calls for this user. User permission group defines a collection of call permission flags. User permission groups are assigned to the users and call permission flags are assigned to the calling rules. User is allowed to make a call only if his permission group includes call permission flag assigned to calling rule for the call. Please see <u>Permissions</u> menu for more details.

#### **User's Call Limit**

Enter maximal number of simultaneous calls permitted for this user.

Voice&Data Router allows controlling SIP user's access according to the IP addresses. There are no restrictions by default. Click **Permit/Deny** button in the SIP users' list to configure range of IP addresses from which the user extension device can connect to Voice&Data Router. There are two types of access rules you can use

- permit
- deny

Each of them can define range of IP addresses (IP address and network mask) to enable or disable access from/to. To create rule for single IP address use network mask 255.255.255.255. You can enter several rules which are applied in the order they were created – order is important.

G	UI-version : 1.	0 / Board Type : 1	Jata rou	ıter			Logout inotesha		
	Home Use	rs Trunks T	raffic Globals System	Diagnostics Specials					
Us	sers > SIP User:	3							
Р	ermit/Den	y list			Add new rule	Delete Selected	Back to Extensions		
	Warning! Ord	der is important							
		Permit/Deny	IP	Network I	/lask				
		Deny	0.0.0.0	0.0.0.	0		Edit Delete		
		Permit	192.168.1.0	255.255.2	255.0		Edit Delete		
	Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>								

Example:

If you want to restrict the user to access Voice&Data Router account only from your local network with address 192.168.1.0/24 then you should create following rules

- 1. Deny All
- 2. Allow 192.168.1.0/255.255.255.0

#### Analog users

Main screen for analogue users displays table of all available analogue FXS user extensions. Analogue users are optional interfaces depending on hardware configuration of Voice&Data Router. Because FXS module has 4 ports we can have from 4 up to 32 analog users in the system (FXS module can occupy all 8 slots available on the analogue extension card).

GUI-version: 1.0 / Board Type : 1 Voce&Data Router Reloaded !! Logout							Logout
voi	ce <mark>o</mark>	data	router				noteska
Home Us	ers Trunks	Traffic Globals	System Diagnostics	Specials			
Users > Analog I	Users						
Analog Us	ers Settings						
List of Ana	log Users						
	Position	Extension	Full Name	Channel	DialPlan		
	1	20	Doug Coombs	144	from_fxs	Unused Edit	
	2	21	Hugo Harrison	145	from_fxs	Unused Edit	
	3	22	Seth Morrison	146	from_fxs	Unused Edit	
	4	23	Shane McConkey	147	from_fxs	Unused Edit	
	5	24	Glen Plake	148	from_fxs	Unused Edit	
	6	25	Kaj Zacrisson	149	from_fxs	Unused Edit	
	7	26	Mike Hattrup	150	from_fxs	Unused Edit	
	8	27	Chris Davenport	151	from_fxs	Unused Edit	
	L						1
		Cn	nvright⊚2010-2012_All Rig	ihts Reserved w	ww.inoteska.sk		

All available FXS ports are always displayed. Unused ports (not configured) are marked as **unset position**. Use **Edit** button to create analogue user extension (configure unused port) or to modify existing one. To remove user extension (set FXO port to unused state) click on **Unused** button. User extension can be removed only if there does not exist any reference to this user in calling rules and ringing groups.

To configure analogue user extension new form with all available settings appears. Here you can configure available parameters and click **Save** button to store new user configuration.

VOICE&DATA ROUTER				
GUI-version : 1.0 / Board Type : 1 Voce&Data Router Reloaded !!	Logout			
voice&data router	nda			
Home Users Trunks Traffic Globals System Diagnostics Specials	Dha			
Users > Analog Users				
Edit Details				
General				
Position 2 Channel 145				
Extension ગ 21 Full Name ગ Hugo Harrison				
DialPlan 🌮 from_fxs 💌				
Parameters				
Echo Cancellation 🧐 32 ms 💌				
Supervisory Tones 🥙 Slovak Republic - PBX 💉				
Polarity Reversal 🎯 🗹				
Rx Gain 🧶 0 dB 💌 Tx Gain 🛞 0 dB 💌				
Impedance 🥙 600 Ohm 💌 Loop Current 🥙 18 mA 💌				
Flash length 🥯 100				
User's Services				
User Permission Group 🥙 🛛 Without limitation 🛛 💌				
Pickup Group 🛞 🛛 💌				
🗹 🥐 Call Transfer 🗹 🧶 Call Forwarding 🗖 🦈 Do Not Disturb				
Save Unused Cancel				
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>				

#### **General Parameters**

#### Position

This is analogue port position (number from 1 up to 32).

#### Channel

Internal channel number.

#### Extension

Extension number associated with this analogue user.

#### Full Name

A character-based display name used for caller identification (i.e. Bob Jones).

#### **Dial Plan**

Select dial plan to handle calls coming from user phone to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

#### **Other Parameters**

#### **Echo Cancelation**

Enable/Disable echo cancelation for VoIP calls and select echo canceller tail length. Voice&Data Router supports echo canceller length from 8 ms up to 128 ms in step of 4 ms. Default echo canceller tail length is 64 milliseconds.

#### **Supervisory Tones**

Select one of predefined set of supervisory (indication) progress tones for this user extension.

inoteska systems for voice and data communication

#### **Polarity Reversal**

Enable/disable FXS line polarity reversal. When enabled every answer and hang-up of the remote party is signalized to the line by changing/switching its polarity.

#### Rx Gain, Tx Gain

Use these parameters for FXS line volume adjustment – transmit and receive gain or attenuation. Rx gain defines the level of voice received by Voice&Data Router from the FXS line and tx gain parameter defines the level of voice transmitted from Voice&Data Router to the FXS Line. You can modify volume level in range from -18 dB up to 6 dB.

#### Impedance

FXS line impedance setting. Two line impedance modes are available

- 600 Ω
- Complex global impedance

#### Loop Current

This setting allows to define FXS line loop current limit. It is available to set the constant loop current in range between 18 mA and 28 mA (default value is 18 mA).

#### **Flash Length**

Maximal length of hook flash in milliseconds. Only hook flash shorter than defined time can be evaluated to hold/unhold active calls.

#### **Services**

#### User's Permission Group

Select permission group to restrict outgoing calls for this user. User permission group defines a collection of call permission flags. User permission groups are assigned to the users and call permission flags are assigned to the calling rules. User is allowed to make a call only if his permission group includes call permission flag assigned to calling rule for the call. Please see <u>Permissions</u> menu for more details.

#### Pickup Group

Pickup group allows to pickup fellow workers' calls. Users from the same group can pickup incoming calls each other using code defined in <u>Services Codes</u> menu.

#### Call Transfer

Permit/deny call transfer service. If enabled the user is allowed to transfer active call to previously held one by hang-up the handset.

#### Call Forwarding

Permit/deny Call Forwarding services for extension user. If enabled the user can activate/deactivate call forwarding service (unrestricted / when busy /when no answer) by service codes defined in <u>Services Codes</u> menu.

#### **Do Not Disturb**

Permit/deny Do Not Disturb service for extension user. If enabled the user is allowed to activate or deactivate this service by service codes defined in <u>Services Codes</u> menu.

#### **TRUNKS MENU**

Trunks menu offers trunks configuration. Voice&Data Router supports SIP trunks, E1/T1 trunks, BRI trunks (S0 or Uk0), analogue trunks (FXO and E&M) and GSM trunks. BRI, analogue and GSM trunks are optional and they depend on hardware configuration.

#### SIP Trunks

SIP trunks main screen displays table of all configured trunks. You can use the **New SIP Trunk** button to create and configure new VoIP SIP trunk. Button **Edit** serves to modify appropriate SIP trunk setting and the **Delete** button is dedicated to remove existing trunk. Trunk can be removed only if there does not exist any reference to this trunk in calling rules.

GUI-ver	GUI-version: 1.0 / Board Type: 1     Logout       VOICESdata router     instanto								
Home	Users Trunks Traffic	Globals System	Diagnostics Specials	_	uwuzsha				
Trunks >	· SIP Trunks								
SIP T	runks Settings				New SIP Trunk				
	Provider/Client Name	DialPlan	Hostname/IP	Username					
	sipp_client	from_sip	192.168.1.14	sipp_client	Edit Delete				
	sipp_server	from_sip	192.168.1.48	sipp_server	Edit Delete				
	TEST	from_sip	192.168.1.166	test_trunk	Edit Delete				
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When you are creating new SIP trunk or when you are changing parameters of the existing one you are switched SIP trunk's details form, with all available trunk settings. Here you can configure available parameters and click **Save** button to store new configuration.

#### SIP trunk parameters

#### **Provider/Client Name**

A unique label to help you identify this VoIP trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

#### Dial Plan

Select dial plan to handle calls coming from SIP trunk to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

#### Hostname

There are two possibilities how to connect to the remote peer (SIP server). **Static** connection defines permanent static connection to the dedicated IP address. It should be used in case when the trunk connects Voice&Data Router to superior system (SIP provider) and the registration with this system is required. Remote peer address should be configured for static connection (you can use numeric IP address or the host name). On the other hand, **dynamic** connection type is selected when remote peer is registering with Voice&Data Router. It is also possible to change standard port number of SIP protocol to listen for SIP signaling.

#### Username

SIP channel user name. This field specifies the user name for the authentication.

#### Authuser

User name used for authentication when Voice&Data Router is registering with the SIP provider or another superior system (see Register option bellow). Use this filed only if you need different name than the username option above.

VOICE&DATA ROUTER			
GUI-version : 1.0 / Board Type : 1 Edit Trunk			Logout
voicesdate	n router		inoteska
Home Users Trunks Trattic Globals Trunks > SIP Trunks	System Diagnostics	Specials	
Edit SIP trunk - TEST			
Provider/Client Name DialPlan	TEST	9	
Hostname	static V 🥙 192.168.1	.166 : 5060	
Username	test_trunk	9	
Authuser		9	
Fromuser		9	
Fromdomain		9	
Password	heslo	9	
Contact		9	
Qualify	2		
Insecure Type	very 💌 🧐		
Codecs	First G.729 💌 Second a-I	aw 💌 Third None 💌	
	Fourth None 💌 Fifth None	e 💙	
Caller ID	abc 🦻		
VoIP Settings	Register 🗹 🥙 NAT 🗹 🤍 C	can Reinvite 🗆 🖗 DTMF Mode 🥙 🛛 RFC2833 💌	
Call Limit	0		
Fax & Modern Support	Enable T.38 🗹 🔄		
	Fax Passthrough 🗹 Codec 📙	alaw 💌	
	Save (	Sancel	

#### Fromuser

This allows you to overwrite the user name in the From and Contact SIP header fields of the INVITE request. It may be required by some providers for authentication.

#### Fromdomain

This allows you to set the domain in the From SIP header field of the INVITE request. It may be required by some providers for authentication

#### Password

Password to use for authentication.

#### Contact

SIP URI contact name (user name in Contact SIP header field) used when Voice&Data Router is registering with the SIP provider or another superior system (see Register option bellow).

#### Qualify

Enable this feature to check if remote device is reachable. Qualify will send a SIP OPTIONS command regularly to check that the remote device is still online. If the device does not answer within the configured period Voice&Data Router considers the remote device off-line for future calls. Expiration time is configured in seconds (default is 2 seconds). This feature may also be used to keep a UDP session open to a device that is located behind a NAT. This can be used in conjunction with the NAT setting.

#### Insecure Type

Insecure option specifies how to handle incoming SIP session and trunk matching. If it is disabled normal IP-based matching applies and the authentication is required (when password is set for user extension). Set insecure to **port** to allow matching of SIP trunk by IP address without matching port number (useful when we have multiple endpoints behind NAT). Set insecure to **invite** to remove the requirement for authentication of incoming INVITE messages. And use **very** value to allow both, the matching of trunk by IP address without matching port number and to remove the requirement for authentication of incoming INVITE messages.

#### Codecs

Here you can define allowed voice compression codecs in order of preference. Voice&Data Router supports following voice codecs:

- G.711 A-Law
- G.711 μ-Law
- G.729 Annex A/B
- G.726-32
- G.723.1
- GSM
- iLBC

Up to five codecs with defined preference can be configured for every SIP trunk.

#### Caller ID

This is the caller identification number that the trunk will try to use when making outbound calls. Set this field if you want to overwrite original calling user identification. It is useful when unique identification for all calls is required.

#### **VoIP Settings**

#### Register

Check this to enable registering at remote server. When enabled Voice&Data Router will register this trunk with the SIP provider or another superior system.

#### NAT

This setting allows communication with remote device behind the NAT. It works properly in conjunction with Qualify setting in order to keep open the connection from Voice&Data Router to the peer behind the NAT.

#### Can Reinvite

Enabling this option causes Voice&Data Router to attempt to route the RTP media stream directly between SIP endpoints, bypassing Voice&Data Router.

#### DTMF Mode

Default mode for sending DTMF digits.

#### Call Limit

Enter maximal number of simultaneous calls permitted for this trunk.

#### Fax & Modem Support

#### Enable T.38

Enable/disable T.38 protocol (real-time Fax over IP) support. When enabled, Voice&Data Router can work as T.38 Fax Relay between SIP and TDM.

#### **Fax Passthrough**

Enable Fax Pass-Through for automatic switch to the codec G.711 to transfer Fax stream over the SIP trunk. If enabled together with T.38 protocol, T.38 Fax Relay will be used as preferred method for FAX transmission and if not acceptable by the peer then G.711 codec will be used instead. Please select G.711 codec version (A-Law/  $\mu$ -law) to use for Fax Pass-Through.

#### **Modem Passthrough**

Enable Modem Pass-Through for automatic switch to the codec G.711 to transfer modem data over the SIP trunk. Please select G.711 codec version (A-Law/ $\mu$ -law) to use for Modem Pass-Through.

#### E1/T1 trunks

Main screen for E1/T1 trunks displays table of available E1/T1 interfaces. There are always four E1/T1 ports in Voice&Data Router and unnecessary ports can be disabled Use **Edit** button to modify configuration of selected E1/T1 interface.

E1/T1 trunk is configured in two steps. First of all some essential settings are done (including selection of line mode and signaling type). After that the configuration of all other details can be completed.

GUI-version: 1.1/Board Type: 1 Voice Solata router							Logout	
Tr	Home Users Trunk unks > E1/T1 Trunks	s Traffic G	lobals System Dia	agnostics Specials			ublezhu	
E	1/T1 Trunks Settin	igs						
E	E1/T1 Interfaces							
	Interface	Name	Line Mode	Signaling	DialPlan	Status		
	A	E1-A	E1	isdn	from_e1	ON	Edit	
	В	E1-B	E1	isdn	from_e1	ON	Edit	
	с	E1-C	E1	cas	from_e1	ON	Edit	
	D	E1-D	E1	cas	from_e1	OFF	Edit	
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When you click the **Edit** button in the trunks table new form with general settings appears. Here you can select basic parameters of E1/T1 interface and click on **Details** to process detailed configuration. Or use **Save** button to store new settings.

GUI-version : 1.1 / Board Type : 1	Logout
voice relate reuter	
VOICEQUALCI router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Trunks > E1/T1 Trunks	
Interface B	
General Settings	
☑ <sup>®</sup> Enabled	
Name 🥙 E1-B Line Mode 🧐 E1 🔍 Signaling 🎱 ISDN 🗸	
DialPlan from_e1 💌	
Details	
Save	
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

#### **General Settings**

#### Enabled

Enable/Disable E1/T1 interface.

#### Name

A unique label to help you identify E1/T1 trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

#### Line Mode

Choose line mode

- E1 2,048 Mb/s line divided to 32 timeslots
- T1 1,544 Mb/s line divided to 24 timeslots.

Note that CAS signaling is only available in E1 mode. It is also important to know that line mode changing resets the entire trunk configuration to default status. You should use **Details** button to modify this default setting where it is necessary.

#### Signaling

Signaling type selection. Voice&Data Router supports **ISDN** and **CAS** (MFC/R2) signaling. Note that CAS signaling is only available in E1 mode. Changing of signaling type resets signaling dependent parameters to their default values. Use **Details** button to check and modify these parameters.

#### Dial Plan

Select dial plan to handle calls coming from E1/T1 line to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

Click **Details** button to continue with detailed configuration. E1/T1 trunk details form can differ in dependence on selected signaling type.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Details: Interface B	
Line Parameters	
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#### **Line Parameters**

#### CRC

Enable CRC4 multiframe format of framing mode.

#### Clock Mode

Choose clock synchronization source

• Master device provides synchronization clock to the line

- **Slave 1** device is synchronized by the clock from the line
- Slave 2 device is synchronized by the clock from the line

Slave 1 has higher priority than Slave 2.

#### Longhaul RX, Longhaul TX

These options enable to increase the device radius by setting the receiving/transmitting more sensitive. Long haul parameter is within G.703 specification and it is also possible to connect a standard device to the device with long haul.

#### **ISDN Details**

This is signaling dependent part is available only for E1/T1 trunk with ISDN signaling.

#### Switch Type

Select ISDN signaling variant.

#### Mode

Select ISDN signaling mode

- NT network side
- **TE** terminal/user (CPE) side

#### Restart L3

Enable automatic restart procedure on 3rd layer for all voice channels. Restart procedure starts after D-channel coming up and repeats every hour.

#### **CAS Details**

This is signaling dependent part is available only for E1 trunk with CAS signaling type.

#### **Register Signaling**

Select register signaling

- MFC/R2 Use MFC/R2 tones to transfer DNIS number, ANI number and call progress status.
- **DTMF** Use DTMF tones to transfer DNIS number. No caller identification is available for this register signaling.
- **MFC/R2+DTMF** Allows combination of MFC/R2 and DTMF signaling when you can select different register signaling in outgoing direction and incoming direction.
  - **Pulse** Use line signaling pulses to transfer DNIS number. No caller identification is available for this register signaling.

#### **CAS** Variant

Select CAS national variant. Voice&Data Router offers predefined national variants you can use. There is also possibility to customize predefined variants for your needs or to create your own national variant. See section <u>CAS National Variants</u> for more details.

#### Max Length of ANI Number

Maximal number of ANI (Automatic Number Identification) digits = caller identification number. This option is valid only for MFC/R2 register signaling.

#### Max Length of DNIS Number

Maximal number of DNIS (Dialed Number Identification Service) digits = called party address.

#### Send ANI after DNIS

Enable this option if you want transfer whole DNIS number before ANI number. This option is valid only for MFC/R2 register signaling.

#### Ask for ANI

Enable caller identification request. Disable this option if remote exchange does not support or you do not require caller identification (ANI number). This option is valid only for MFC/R2 register signaling.

#### Use FORCED RELEASE

Enable Forced Release line signal.

#### Send CLEAR BACK

Enable sending of Clear-Back line signal to remote side.

#### Accept CLEAR BACK

Enable receiving of Clear-Back line signal from remote side.

GUI-version : 1.1 / Board Type : 1	Logout
voices data reuter	
VOICeQUCICI router	instade
Hans Heave Turnin Turffie Clabale Surface Discovertion Spaciale	uwueshd
Trunks > E1/T1 Trunks	
Details: Interface B	
Line Parameters	
🗹 🎱 CRC 🛛 Clock Mode 🧶 🔄 Slave 1 💌 🗖 🖤 Longhaul RX 🔲 🖤 Longhaul TX	
CAS Details	
Register Signaling 🥙 MFC/R2 💽 Send MFC/R2 receive DTMF 😒	
CAS Variant 🎱 Czech-Slovak 💌	
Max.length of DNIS Number 🥯 24 Max.length of ANI Number 🧐 24	
🗌 🏵 Send ANI after DNIS 🗹 🎯 Ask for ANI	
🗹 🦈 Use FORCED RELEASE 🗹 🥗 Send CLEAR BACK 🗹 🦈 Accept CLEAR BACK	
Channels	
Channels 🥐 1-15,17-31	
Channel Selection 🦻 Descending 💌 🗌 🖗 Cyclic	
Parameters	
🗹 🧐 Overlap Dial 🛛 First Digit Timeout 🧐 15 🛛 Other Digits Timeout 10	
🗹 🎱 Echo Cancellation 🛛 default 💌	
Supervisory Tones 🌮 Slovak Republic 💌	
🗖 🎱 Play Progress Tone 🛛 🎱 Play Ringback Tone 🗖 🥙 Play Dial Tone	
AOC	
🗖 🥙 Generate AOC Charging 🛛 default 💌 🗌 🎱 Add Incoming Charging	
Save Go Back Cancel	
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#### Channels

#### Channels

Choose available voice channels.

- E1 mode use channel range from 1 to 31.
- T1 mode use channel range from 1 to 24.

Avoid of collision with signaling channel. E1 line with CAS signaling has reserved channel 16 for line signaling transfer.

#### **Signaling Channel**

Signaling channel (ISDN D-channel) selection. It is possible to move ISDN signaling channel to any position when it is necessary. E1 line uses channel 16 and T1 line uses channel 24 for signaling by default. This option is available only for ISDN signaling type. CAS signaling has always reserved 16<sup>th</sup> channel for line signaling transmission and it cannot be changed.

#### **Channel Selection**

Choose method how to select voice channel for outgoing call.

- Ascending search for the lowest available channel number.
- Descending search for the highest available channel number

#### Cyclic

Enable cyclic search of available channel for outgoing call instead of sequential.

#### **Parameters**

#### **Overlap Dial**

Enable/disable overlap dialing in ISDN signaling. If not enabled, complete called number in SETUP is required. E1 line with CAS signaling always uses overlap dialing.

#### **First Digit Timeout**

Enter timeout for the first digit to detect dialing hesitation during overlap dialing. When it expires end of dial is indicated.

#### **Others Digits Timeout**

Enter timeout of others to detect dialing hesitation during overlap dialing. End of dial is indicated after the timeout.

#### **Echo Cancellation**

Enable/Disable echo cancelation for VoIP calls and select echo canceller tail length. Voice&Data Router supports echo canceller length from 8 ms up to 128 ms in step of 4 ms. Default echo canceller tail length is 64 milliseconds.

#### **Supervisory Tones**

Select one of predefined set of supervisory/indication tones generated by this E1/T1 line.

#### **Play Progress Tone**

Enable generation of progress supervisory/indication tones.

#### **Play Ringback Tone**

Enable generation of ring-back tone.

#### **Play Dial tone**

Enable generation of dial tone.

#### AOC

#### **Generate AOC Charging**

Enable/disable generation of charging information according selected AOC table. See section <u>AOC</u> <u>Tables Settings</u> for details about configuration of AOC charging generation.

#### Add Incoming Charging

If enabled charging units received from destination will be added to the charging units generated by Voice&Data Router according AOC table.

Use **Go Back** button to return to general settings or click on **Save** button to confirm changes and store new setting.
# BRI Trunks

Voice&Data Router supports up to 16 BRI Uk0 interfaces or 4 BRI S0 interfaces. Initial screen for BRI trunks displays table of available interfaces according hardware configuration. BRI trunks are configured in two steps. First of all some essential settings are done. After that the configuration of all other details can be completed.

GUI-version: 1.8 / Board Type: 1								
	Home Users Trunks	Traffic Glo	bals System Diagn	ostics				
Tru	unks > BRI Trunks							
В	RI Trunks Settings							
E	RI Interfaces							
	Interface	Name	Line Mode	Dial Plan	Channel	Status		
	A	UK0-A	Uk0	t	177,178	ON	Edit	
	в	UK0-B	Uk0	t	180,181	ON	Edit	
	С	UK0-C	Uk0	t	183,184	ON	Edit	
	D	UK0-D	Uk0	t	186,187	ON	Edit	
	E	UK0-E	Uk0	t	189,190	ON	Edit	
	F	UK0-F	Ukū	t	192,193	ON	Edit	
	G	UK0-G	Ukū	t	195,196	ON	Edit	
	н	UK0-H	Ukū	t	198,199	ON	Edit	

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When you click the **Edit** button in the trunks table new form with general settings appears. Here you can select basic parameters of BRI interface and click on **Details** to process detailed configuration. Or use **Save** button to store new settings.

GUI-version : 1.8 / Board Type : 1	Logout
voice&data router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics	
Trunks > BRI Trunks	
Interface B	
│ General Settings	
🗷 🥯 Enabled	
Name 🧶 UKO-B Dial Plan 🎱 t 🔹 Details	
Save	
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## **General Settings**

### Enabled

Enable/Disable BRI interface.

#### Name

A unique label to help you identify BRI trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

#### **Dial Plan**

Select dial plan to handle calls coming from this BRI line to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules configuration.

Click Details button to continue with detailed configuration.

#### **Line Parameters**

### Clock Mode

Choose clock synchronization source

- **Master** device provides synchronization clock to the line. This is the only option when line is configured in NT mode (S0 interface) or LT mode (Uk0 interface).
- Slave 1 device is synchronized by the clock from the line
- Slave 2 device is synchronized by the clock from the line

Slave 1 has higher priority than Slave 2.

### Line Power Supply

Enable line power supply. Line can be powered only in NT mode (S0 interface) or LT mode (Uk0 interface). This function depends on type of device power supply.

GUI-version : 1.8 / Board Type : 1	Logout
voice Scloto router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics	
Trunks > BRI Trunks	
Interface B	
Line Parameters	
Clock Mode 🧖 🛛 Master 🔻 🖉 🦈 Line Power Supply	
ISDN Details	
Switch Type 🧶 EuroISDN 🔹 Mode 🥙 LT 💌 🖉 🥯 Point-to-Multipoint	
🔲 🦈 L2 Deactivation	
Channels	
Channels 180,181	
Channel Selection 🥗 🔺 🔲 🔍 Cyclic	
Parameters	
🕑 🤍 Overlap Dial 🛛 First Digit Timeout 15 🛛 Other Digits Timeout 7	
🗷 🎯 Echo Cancellation 🛛 default 🔻	
Supervisory Tones 🥙 Slovak Republic - PBX 🔻	
🗷 🥬 Play Progress Tone 🛛 🕐 Play Ringback Tone 🕑 🧐 Play Dial Tone	
AOC	
Generate AOC Charging 🛛 default 🔹 👘 🔍 Add Incoming Charging	
Save Go Back Cancel	
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#### **ISDN Details**

Switch Type Select ISDN signaling variant.

Mode Select ISDN signaling mode.

Modes for S0 interface:

- NT network/provider side
- TE terminal/user (CPE) side

Modes for Uk0 interface:

- LT network/provider side
- NT terminal/user (CPE) side

### **Point-to-Multipoint**

Enable point-to-multipoint mode when Voice&Data Router is attached to an ISDN bus (more than one devices can be used at the CPE side).

### L2 Deactivation

Enable automatic deactivation of L2 data-link connection for line configured in point-to-multipoint mode.

### Channels

### Channels

List of B channels timeslots.

## **Channel Selection**

Choose method how to select B channel for outgoing call.

- Ascending search for the lowest available channel number.
- **Descending** search for the highest available channel number

### Cyclic

Enable cyclic search of available B channel for outgoing call instead of sequential.

### **Parameters**

#### **Overlap Dial**

Enable/disable overlap dialing. If not enabled, complete called number in SETUP is required.

### **First Digit Timeout**

Enter timeout for the first digit to detect dialing hesitation during overlap dialing. When it expires the end of dial is indicated.

### **Others Digits Timeout**

Enter timeout of others digits to detect dialing hesitation during overlap dialing. The end of dial is indicated after the timeout.

### **Echo Cancellation**

Enable/Disable echo cancelation for VoIP calls and select echo canceller tail length. Voice&Data Router supports echo canceller length from 8 ms up to 128 ms in step of 4 ms. Default echo canceller tail length is 64 milliseconds.

### **Supervisory Tones**

Select one of predefined set of supervisory/indication tones generated by this BRI line.

### **Play Progress Tone**

Enable generation of progress supervisory/indication tones.

### **Play Ringback Tone**

Enable generation of ring-back tone.

### Play Dial tone

Enable generation of dial tone.

## AOC

## **Generate AOC Charging**

Enable/disable generation of charging information according selected AOC table. See section <u>AOC</u> <u>Tables Settings</u> for details about configuration of AOC charging generation.

## Add Incoming Charging

If enabled charging units received from destination will be added to the charging units generated by Voice&Data Router according AOC table.

Use **Go Back** button to return to general settings or click on **Save** button to confirm changes and store new setting.

## Analog Trunks

Main screen for analogue trunks displays table of all available FXO interfaces. Analogue trunks are optional interfaces depending on hardware configuration of Voice&Data Router. Because FXO module/card has 2 ports we can have from 2 up to 16 analogue FXO trunks in the system (FXO module can occupy all 8 slots available on the analogue extension card).

All available analogue trunks are always displayed. Unused ports (not configured) are marked as **unset position**. Use **Edit** button to configure selected interface – modify existing setting or configure unused port. To remove some trunk from the configuration click on **Unused** button. Analogue FXO trunk can be removed only if there does not exist any reference to this port in any calling rules or calls groups.

<b>∞</b> da	ta ro	uter			inotesk
runks Traffic	Globals System	Diagnostics	Specials		
ettings					
;					
Name	Trunk Type	DialPlan	CID Number	Channel	
FXO1	FXO	from_fxo		152	Unused Edit
FXO2	FXO	from_fxo		153	Unused Edit
FXO3	FXO	from_fxo		156	Unused Edit
FXO4	FXO	from_fxo	0445567888	157	Unused Edit
FXO5	FXO	from_fxo		160	Unused Edit
FXO6	FXO	from_fxo		161	Unused Edit
unset position	FXO			164	Edit
	51/0			405	
	runks Traffic Traffi	Name   Trunk Type     FX01   FX0     FX02   FX0     FX03   FX0     FX04   FX0     FX05   FX0     FX06   FX0     FX07   FX0     FX08   FX0     FX09   FX00     FX05   FX0     FX06   FX0     TX06   FX0	Name   Trunk Type   Diagnostics     FXO1   FXO   from_fxo     FXO2   FXO   from_fxo     FXO3   FXO   from_fxo     FXO4   FXO   from_fxo     FXO5   FXO   from_fxo     FXO4   FXO   from_fxo     FXO5   FXO   from_fxo     FXO6   FXO   from_fxo     FXO6   FXO   from_fxo     FXO6   FXO   from_fxo	Name Trunk Type Diagnostics Specials   K Name Trunk Type DialPlan CID Number   FX01 FX0 from_fx0 FX02 FX0 from_fx0   FX03 FX0 from_fx0 0445567888 FX05 FX0 from_fx0   FX06 FX0 from_fx0 FX06 FX0 from_fx0	Name   Trunk Type   Diagnostics   Specials     K   Name   Trunk Type   Diagnostics   CID Number   Channel     FX01   FX0   from_fx0   152   153     FX02   FX0   from_fx0   153     FX03   FX0   from_fx0   156     FX04   FX0   from_fx0   145567888     FX05   FX0   from_fx0   160     FX06   FX0   from_fx0   161     unset position   FX0   from_fx0   164

When you click on **Edit** button to configure selected analogue trunk you will continue with new screen with detailed configuration. Here you can configure all parameters and click **Save** button to store new setting.

### **General Settings**

### **Trunk Type**

Analogue trunk type

• FXO FXO interface

### Position

This is analogue port position (number from 1 up to 32).

### Channel

Internal channel number.

### Name

A unique label to help you identify analogue trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

### **Dial Plan**

Select dial plan to handle calls coming from this analogue trunk line to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

### **CID Number**

This is the caller identification number you can use to overwrite original calling user identification.

VOICE&DATA ROUTER	
GUI-version : 1.1 / Board Type : 1	Logout
voice Scloto router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Edit Trunk Settings	
General Settings	
Trunk Type FXO	
Position 11	
Channel 153	
Name 🦻 Fxo2	
DialPlan 🌮 🛛 from_fxo 💌	
CID Number 🦃	
Parameters	
Rx Gain <sup>(2)</sup> 0 dB (2) Tx Gain <sup>(2)</sup> 0 dB (2)	
Impedance 🤗 600 Ohm 💌	
Save	
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### **Parameters**

## **Echo Cancelation**

Enable/Disable echo cancelation for VoIP calls and select echo canceller tail length. Voice&Data Router supports echo canceller length from 8 ms up to 128 ms in step of 4 ms. Default echo canceller tail length is 64 milliseconds.

### **Polarity Reversal**

Enable/disable polarity reversal detection on FXO trunk. When enabled then answer or hang-up of the remote party can be signalized to the Voice&Data by change of polarity on FXO line.

### Pulse Dial

Enable pulse dialing instead of using DTMF tones.

#### Rx Gain, Tx Gain

Use these parameters for line volume adjustment – transmit and receive gain or attenuation. Rx gain defines the level of voice received by Voice&Data Router from the FXO trunk and tx gain parameter defines the level of voice transmitted from Voice&Data Router to the FXO trunk. You can modify volume level in range from -15 dB up to 12 dB.

## Impedance

FXO line AC termination impedance setting. Two modes are available

- 600 Ω
- Global Complex impedance

# GSM Trunks

GSM trunks main screen displays table of available GSM interfaces. GSM trunks are optional interfaces depending on hardware configuration of Voice&Data Router. It is possible to have up to 12 GSM ports in the system.

GUI-version : 1.1 / Board Type : 1	ata ra	outer		Logout inoteska
Home Users Trunks Trat	ffic Globals System	n Diagnostics Specials		
Trunks > GSM Trunks				
GSM Trunks Settings				
GSM Trunks				
Interface	Name	DialPlan	Channel	
1	GSM1	from_gsm	128	Edit
2	GSM2	from_gsm	129	Edit
3	GSM3	from_gsm	130	Edit
4	GSM4	from_gsm	131	Edit
-				

To modify GSM trunk configuration use appropriate Edit button in GSM trunks table.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Trunks > GSM Trunks	
Edit Trunk Settings	
General Settings Name ③ GSM1 DialPlan ④ from_gsm ▼ PIN Number ④ Echo Cancellation ④ Audio Level From GSM ④ 80 To GSM ④ 4	
Save Cancel	
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## **General Settings**

### Name

A unique label to help you identify GSM trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

## **Dial Plan**

Select dial plan to handle calls coming from this GSM interface to Voice&Data Router. Dial plan consists of calling rules and it defines call routing scenario. Please see chapter <u>Traffic Menu</u> for more details about dial plans and calling rules.

### **PIN Number**

Enter valid PIN code to access GSM SIM card.

## **Echo Cancellation**

Enable/disable echo cancellation on GSM module to suppress acoustic echo from GSM network.

## **Audio Level**

## From GSM

Configure speaker volume level in range from 0 to 99. This is voice level received by Voice&Data Router from GSM trunk. Default value is 91.

## To GSM

Configure microphone volume level in range from 0 to15. This is level of the voice transmitted from Voice&Data Router to GSM trunk. Default value is 3.

# **TRAFFIC MENU**

Traffic menu defines call routing between users and trunks. Call routing system of Voice&Data Router can be very easy but it can be also really complex depending on system functionality. It seams to be the most important part of the whole system which designates if Voice&Data Router works correctly or not.

Call routing consists of

- Calling rules
- Dial planes

**Calling rules** define new call destination on the basis of analysis of dialed number eventually dialed number together with caller identification number. Calling rules are divided according type of source and destination to rules for User to User, User to Trunk, Trunk to User and Trunk to Trunk. Calling rules also allow defining modification of dialed number or caller identification number.

**Dial plans** group existing calling rules. Dial plans are assigned to users and trunks. Calling rules included in dial plan then will apply for all inbound calls from the user or trunk.

Configuration of call routing is two-level

- 1. Define calling rules.
- 2. Group existing calling rules to dial plans.

Finally, it is necessary to assign dial plans to users and trunks. Interfaces without dial plan can not serve inbound calls and the calls will be refused.

# Calling Rules

Main screen for calling rules configuration shows list of existing calling rules.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data ra	outer
Home Users Trunks Traffic Globals System	n Diagnostics Specials
Traffic > Calling Rules	
Calling Rules Settings	
Calling Rules Groups	
Calling rules are divided into four groups according to the patterns to be dialed to the local and remote dest Note that this panel manages only individual call rules. See	method of call routing (User-User, User-Trunk, Trunk-User, Trunk-Trunk). This allows different ination. You can optionally set three failover trunks to use when the primary trunk fails. the Dial Plans section to associate multiple calling rules to be used for User and Trunk dialing.
User to User	
Calling Rules: User to User	New Calling Rules Group Name Create
internal	Edit Delete
User to Trunk	
Calling Rules: User to Trunk	New Calling Rules Group Name
e1	Edit Delete
sip	Edit Delete
Trunk to User	
Calling Rules: Trunk to User	New Calling Rules Group Name Create
users	Edit Delete
Trunk to Trunk	
Calling Rules: Trunk to Trunk	New Calling Rules Group Name
out	Edit Delete
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Calling rules are divided into four tables

- User to User
- User to Trunk
- Trunk to User
- Trunk to Trunk

You can create new empty calling rules group when you enter new unique label into the field **New Calling Rules Group Name** of appropriate table and click on the **Create** button. This name is used to identify calling rules group in dial plan configuration. Use the **Edit** button to modify configuration of selected group of calling rules. And use the **Delete** button to remove selected group of calling rules. Calling rules can be removed only if there does not exist any reference to this group of calling rules in any dial plan.

When you click on the **Edit** button for selected group of calling rules new form with list of calling rules appears. Table shows analysis pattern and call destination (the details depend on calling rule type). You can create new calling rule by clicking on the **New Calling Rule** button, you can modify calling rule by clicking on the **Edit** button or you can delete calling rule using the **Delete** button.

GUI-	GUI-version : 1.1 / Board Type : 1 Logout							
V	voice&data router							
Но	me Users	Trunks Traffi	ic Globals System	Diagnostics	Specials		u abtebrita	
Traft	fic ≻ Calling Rule	s						
Cal	lling Rules	Settings "USE	R TO TRUNK"					
	lling Pules	· 61						
00	uning rules	. ei				New Calling Rule	Back to Calling Rules List	
	Pattern	Trunk	Failover Trunks			Clone to Rules Group Name	Clone	
	_01.	E1 :E1-A					Edit Delete	
	_02.	E1 :E1-B					Edit Delete	
	_03.	E1 :E1-C					Edit Delete	
	_04.	E1 :E1-D					Edit Delete	
	_8.	FXO:FXO2					Edit Delete	
	_9.	CallGroup: FXO					Edit Delete	
			Copyright© 201	0-2012, All Right	ts Reserved ;	www.inoteska.sk		

If you need to make copy of calling rules group you can use the **Clone** button. Before you click on the **Clone** button you have to enter unique identifier of new group of calling rules in the field **Clone to Rules Group Name**.

### **Calling Rules Patterns**

Every calling rule has pattern for analysis of incoming dialed number eventually caller identification number. Pattern for caller identification number is added to the dialed number pattern by forward slash character (example: \_NXXXXX/\_04455XXXX). Calling rules patterns may be numbers, letters, or combinations thereof. One pattern can match single specific number or whole group of multiple numbers. If the pattern is used for multiple matching it must be prefixed by underscore symbol (\_). After the underscore, you can use one or more of the following special characters.

- X Matches any single digit from 0 to 9.
- Z Matches any single digit from 1 to 9.
- N Matches any single digit from 2 to 9.
- **[15-7]** Matches a single digit from the range of digits specified. In this case, the pattern matches a single 1, 5, 6, or 7.
- . (period) Wildcard match matches one or more characters, no matter what they are.
- ! (bang) Wildcard match matches zero or more characters, no matter what they are.

There is also special pattern **s** that will match to empty dialed number. This is intended for non-DID lines, where call comes in without any dialed number information (like FXO or GSM trunk).

Examples:

111 matches single extension 111.

VOICE&DATA RO	UTER
_NXXXXXX 1NXXNXXXXXX	matches normal 7 digit number. would represent an area code plus phone number preceded by a one.
_9011.	matches any string of at least five characters that starts with 9011,but it does not match the four-character string 9011 itself.
_9011! _# s	same as above but matches 9011 too. matches a single # key press. matches when dialed number is not available
-	

If you are not careful, wildcard matches can make your call routing do things you are not expecting. You should use the wildcard match in a pattern only after you have matched as many other digits as possible. For example, the pattern match "\_." should probably never be used. Use pattern "\_X." instead, if at all possible.

## **User to User Details**

User to User calling rules defines call routing between local user extensions of Voice&Data Router.

GUI-version : 1.1 / Board Type : 1	Logout
voice Scloto reuter	
VOICeQUCICI router	
ir	oteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Traffic > Calling Rules	
Edit Calling Rule: internal	
Pattern 🥐 _[1234]X	
Call Permission Flag 🌮 Internal 💽	
Send to Destination	
Local Destination User DID	
Save	
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### Pattern

Pattern for analysis of the incoming dialed number from the user. If you need to analyze also caller identification number, you can append caller identification pattern to dialed number pattern using forward slash character (\_XXX/\_111). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

### **Call Permission Flag**

Select call permission flag to allow outgoing calls for only selected users. Each user can be assigned to a user permission group. User permission group defines a collection of call permission flags. The call is allowed only in case when user permission group of calling user incorporates permission flag of the calling rule. Please see <u>Permissions</u> menu for more details.

## **Send to Destination**

### **Local Destination**

Select call local destination. It is possible to select:

- Existing user extension call will be routed directly to the selected user extension.
- Existing ring group call will be routed to the selected ring group (group of user extensions which can ring simultaneously or sequentially, see <u>Ring Groups</u> for details).
- User DID call will be routed to the user extension specified by dialed number. This option is useful with multiple matching patterns like \_XX or \_[234]XX. It can simplify calling rules to only a few lines when we do not need to create separate rules for every user.
- Hangup call will be refused (with busy tone)
- Congestion call will be refused (with congestion tone)

#### **User to Trunk Details**

User to Trunk calling rules defines call routing from local user extensions outside of Voice&Data Router over the trunk interfaces. It can be call to PSTN, call to VoIP provider or call to cooperative gateway/PBX.

GUI-version : 1.8 / Board Type : 1	ogout
voice&data router	
inote	ska
Home Users Trunks Traffic Globals System Diagnostics	
Traffic > Calling Rules	
Edit Calling Rule: out	
Pattern 🤍 _01.	
Call Permission Flag 🎱 none 🔻	
-Send to Destination	
Use Trunk 🎱 E1: E1-A 🔹 🔍 🔲 Two Step Dial 🥐 💌	
Strip 🥐 2 digits Strip 🥐 0 digits from Caller ID	
and Prepend these digits 🧐 👘 before dialling 🛛 and Prepend these digits 🎐 👘 to Caller ID	
Presentation 🎱 Original 🔻	
Change TON / NPI 🤎 Change TON / NPI 🤎 of Caller ID	
National   I   ISDN/Telephony   National   I   ISDN/Telephony   I	
🔲 Use Fail-Over Trunk 1 🧐 ——————————————————————————————————	
Fail-Over Trunk 🥐 🔹 🔻 🗌 Two Step Dial 🧶 🔻	
Strip 🥐 📃 digits Strip 🥐 🔤 digits from Caller ID	
and Prepend these digits 🧶 before dialling and Prepend these digits 🧶 to Caller ID	
Presentation <sup>®</sup> Original <b>*</b>	
Change TON / NPI	
Save Cancel	
Conwight © 2010-2016 All Rights Reserved www.insteska.sk	

#### Pattern

Pattern for analysis of incoming dialed number from the user. If you need to analyze also caller identification number, you can append caller identification pattern to dialed number pattern using forward slash character (\_044[5-8]XXXXX/\_111). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

### **Call Permission Flag**

Select call permission flag to allow outgoing calls for only selected users. Each user can be assigned to a user permission group. User permission group defines a collection of call permission flags. The call is allowed only in case when user permission group of calling user incorporates permission flag of the calling rule. Please see <u>Permissions</u> menu for more details.

### **Send to Destination**

This section defines primary routing destination of the call.

#### **Use Trunk**

Select destination trunk to place call through. It is possible to choose directly one of existing trunk or existing call group configured in Voice&Data Router. Call groups can group multiple trunks to one "trunk-group" with defined method of channel selection. Please see <u>Call Groups</u> menu for more details. It is also possible to select Hangup or Congestion options to refuse the call.

### Two Step Dial

Enable two-stage dialing using selected alternative carrier. Two-stage dialing replaces dialed number with carrier access number and on carrier's answer the security code (if set) and the destination number are transmitted in DTMF. See <u>Carriers</u> menu for alternative carrier configuration details.

### Strip digits, and Prepend these digits before dialing

These parameters allow modifying dialed number before the call is placed on the outbound trunk. You are allowed to strip specified number of digits from the front of dialed number and you can enter prefix string which will be prepended to the dialed number.

### Change TON/NPI

Enable to overwrite Type of Number and Numbering Plan Identification of dialed destination number before the call is placed on the selected outbound trunk. This option is applied only on trunk with ISDN signaling.

### Strip digits from Caller ID, and Prepend these digits to Caller ID

These parameters allow modifying caller identification number before the call is placed on the outbound trunk. You are allowed to strip specified number of digits from the front of the caller identification number and you can enter prefix string which will be prepended to this number.

### Presentation

This option allows modifying of caller identification presentation before the call is placed on the outbound trunk.

### **Change TON/NPI of Caller ID**

Enable to overwrite Type of Number and Numbering Plan Identification of caller identification number before the call is placed on the selected outbound trunk. This option is applied only on trunk with ISDN signaling.

### Use Fail-Over trunk 1, 2, 3

This section allows you to define alternative routing of the call if primary destination is busy or down. It is possible to define maximally 3 fail-over alternatives. Each fail-over destination has same options as we define for primary call destination.

## Trunk to User Details

Trunk to User calling rules defines call routing from external interface to the local Voice&Data Router user extensions. It can be call from PSTN or VoIP provider or call from another gateway/PBX.

GUI-version : 1.8 / Board Type : 1 Edit Calling Rule !	igout
voice&data router	
inote	sha
Home Users Trunks Traffic Globals System Diagnostics	
Traffic > Calling Rules	
Edit Calling Rule: in	
Pattern () _1XX Send to Destination Local Destination () User DID () Strip () digits Strip () digits from Caller ID and Prepend these digits () before dialling and Prepend these digits () to Caller ID Presentation () Original () Connect announcement before the call () ()	
Save	
Copyright © 2010-2016, All Rights Reserved <u>www.inoteska.sk</u>	

### Pattern

Pattern to match incoming dialed number from the trunk. If you need to analyze also caller identification number, you can append caller identification pattern to dialed number pattern using forward slash character \_NXXXXX/\_04455XXXX). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

inoteska

## Send to Destination:

## **Local Destination**

Select call local destination. It is possible to select:

- Existing user extension call will be routed directly to the selected user extension.
- Existing ring group call will be routed to the selected ring group (group of user extensions which can ring simultaneously or sequentially, see <u>Ring Groups</u> for details).
- User DID call will be routed to DID (Direct Inward Dialing) user extension. Destination user extension number is extracted from incoming dialed number after applying modifications (strip/prepend). This option is useful with multiple matching patterns like \_55679XX. It can simplify calling rules to only a few lines when we do not need to create separate rules for every user.
- **Hangup** call will be refused (with busy tone)
- **Congestion** call will be refused (with congestion tone)

### Strip digits, and Prepend these digits before dialing

These parameters allow modifying dialed number for "User DID" routing. You are allowed to strip specified number of digits from the front of dialed number and you can enter prefix string which will be prepended to the dialed number.

### Strip digits from Caller ID, and Prepend these digits to Caller ID

These parameters allow modifying caller identification number before the call is routed to the selected local destination. You are allowed to strip specified number of digits from the front of the caller identification number and you can enter prefix string which will be prepended to this number.

### Presentation

This option allows modifying of caller identification presentation before the call is routed to the selected local destination.

### Connect announcement before the call

This option allows to play predefined announcement to the caller before the call will be routed to the selected local destination.

### Trunk to Trunk Details

Trunk to Trunk calling rules defines transit call routing between external interfaces (trunks).

### Pattern

Pattern to match incoming dialed number from the trunk. If you need to analyze also caller identification number, you can append caller identification pattern to dialed number pattern using forward slash character \_NXXXXX/\_04455XXXX). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

### Send to Destination:

This section defines primary destination of the call.

### **Use Trunk**

Select destination trunk to place call through. It is possible to choose directly one of existing trunk or existing call group configured in Voice&Data Router. Call groups can group multiple trunks to one "trunk-group" with defined method of channel selection. Please see <u>Call Groups</u> menu for more details. It is also possible to select Hangup or Congestion options to refuse the call.

### Two Step Dial

Enable two-stage dialing over selected alternative carrier. Two-stage dialing replaces dialed number with carrier access number and on carrier's answer the security code (if set) and the destination number are transmitted in DTMF. See <u>Carriers</u> menu for alternative carrier configuration details.

### Strip digits, and Prepend these digits before dialing

These parameters allow modifying dialed number before the call is placed on the outbound trunk. You are allowed to strip specified number of digits from the front of dialed number and you can enter prefix string which will be prepended to the dialed number.

VOICE&DATA ROUTER	
GUI-version : 1.8 / Board Type : 1	Logout
voice&data router	inoteska
Home Users Frunks Frattic Globals System Diagnostics	
Edit Calling Bule: tt	
Pattern 2 90. Send to Destination Use Trunk Call Group: BRI	•
Unknown ) ) Unknown )<	•
Save Cancel Copyright © 2010-2016, All Rights Reserved www.inoteska.sk	

### Change TON/NPI

Enable to overwrite Type of Number and Numbering Plan Identification of dialed destination number before the call is routed to the selected outbound trunk. This option is applied only on trunk with ISDN signaling.

### Strip digits from Caller ID, and Prepend these digits to Caller ID

These parameters allow modifying caller identification number before the call is placed on the outbound trunk. You are allowed to strip specified number of digits from the front of the caller identification number and you can enter prefix string which will be prepended to this number.

### Presentation

This option allows modifying of caller identification presentation before the call is sent to the outbound trunk.

### Change TON/NPI of Caller ID

Enable to overwrite Type of Number and Numbering Plan Identification of caller identification number before the call is placed on the selected outbound trunk. This option is applied only on trunk with ISDN signaling.

### Use Fail-Over trunk 1, 2, 3

This section allows to define alternative routing of the call if primary destination is busy or down. It is possible to define maximally 3 fail-over alternatives. Each fail-over destination has same options as we define for primary call destination.

## Dial Plans

Main screen for dial plans configuration shows list of existing dial plans. It is divided into two tables

- Users
- Trunks

Users' dial plan can be assigned to the user extensions and trunks' dial plan can be assigned to the trunk interfaces.

GUI-versio	on : 1.1 / Board Type	adata rou	Logout ter inoteska
Home Treffic > Di	Users Trunks	Traffic Globals System D	iagnostics Specials
Dial Pla	ans Settings		
Users		Dialplan is a collection of Calling rules Dialplan are assigned to users a	, which defines dialing string analyze and call routing rules. d trunk to handle incoming calls from these interfaces.
Dia	alPlan: Users	Calling Rules	New DialPlan Name
1	from_users	internal, e1, sip, services	Edit Delete
Trunks	<b>i</b>		
	DialPlan: Trunks	Calling Rules	New DialPlan Name
	from_e1	users, out	Edit Delete
	from_gsm	users	Edit Delete
	from_sip	out, users	Edit Delete
	from_fxo	out, users	Edit Delete
		Convright@2010-20	12 All Rights Reserved www.innteska.sk

You can create new empty dial plan when you enter new unique label into the field **New Dial Plan Name** of appropriate table and click on the button **Create**. This name is used to identify the dial plan over all of Voice&Data Router configuration. Use the **Edit** button to modify setting of selected dial plan. And use the **Delete** button to remove dial plan from the configuration. Dial plan can be removed only if it is not assigned to any user or trunk.

## Users' Dial Plan

Users' dial plan consists of User to User and User to Trunk calling rules and it defines call routing from the user extensions. You can select existing calling rules group in the field **Add Calling Rule** and click on the **Add** button to add calling rules to the dial plan. Only calling rules applicable for users' dial plan are offered for adding. Use the **Delete** button if you want to remove calling rules from the dial plan.

Please note, that the order of calling rules is important for correct functionality of call routing. Same order is used to match dialed number to the calling rule pattern. It is important to have general calling rules (patterns like \_X.) at the end of the calling rules. You can control the order of the calling rules using arrows in the first column of the table.

## Options

## **Use Services Codes**

Enable this option if you want to include users' service codes into this dial plan. Service codes define dial codes that can be used for controlling services (e.g. activation and deactivation) from user's phone. See <u>Services Codes</u> menu for details.

VOIC	E&DATA	ROUTER		
GUI-ver	sion : 1.1 / Boar	d Type : 1		Logout
V	oice	<b>S</b> data	router	inoteska
Home	Users T	runks Traffic Globals	System Diagnostics Specials	
Traffic >	Dial Plans			
Edit D	DialPlan from	n_users		
		Calling Rules	Add Calling	g Rule internal 💌 🗛 🖌
	+	internal		Delete
	<b>+</b> +	e1		Delete
	<b>†</b>	sip		Delete
Optic	ons			
	<sup>IV</sup> Use Services Ignore Pattern	Codes n		
			Back	
		Car	avright © 2010-2012. All Dighte Decement www.inotecka.ek	

### **Ignore Pattern**

This option tells Voice&Data Router to continue to provide a dial tone on an analogue line, even after the caller has dialed the indicated pattern. This will not work with SIP phones, as they usually do not send digits to the system as they are input but they are sent to Voice&Data Router all at once.

## Trunks' Dial Plan

Trunks' dial plan consists of Trunk to User and Trunk to Trunk calling rules and it defines call routing from the trunk interfaces. You can select existing calling rules group in the field **Add Calling Rule** and click on the **Add** button to add these calling rules to the dial plan. Only calling rules applicable for trunks' dial plan are offered for adding. Click on the **Delete** button if you want to remove calling rules from the dial plan.

GU	l-version : 1.1 / Boa	rd Type : 1		Logout
_		e al a ba		
V	<b>OICE</b>	o a coloridado a coloridado a coloridade a c	router	
				inoteska
н	ome Users	Trunks Traffic Globals	System Diagnostics Specials	
Tra	ffic > Dial Plans			
Ec	lit DialPlan fro	om_e1		
		Calling Rules		Add Calling Rule users 💽 🛛 Add
	+	users		Delete
	+	out		Delete
			Back	
		Сору	right©2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

As for calling rules order the same regulations are valid as for users' dial plan. Searching for the valid calling rule which match to the incoming dialed number is executed in the same order as the calling rules are configured in the dial plan. It is important to have general calling rules (patterns like \_X.) at the end of the calling rules. You are allowed to control the order of the calling rules using arrows in the first column of the table.

## AOC Tables Settings

One of useful feature of Voice&Data Router is generation of charging information on E1/T1 lines. E1/T1 trunks with ISDN signaling use AOC supplementary service (AOC-D and AOC-E) and the trunks with CAS signaling uses billing pulses to generate charging information. Different parameters of call charging can be used for working days, weekend eventually for public holidays.

AOC tables setting consist of

- AOC tables
- Time tables
- Holidays definition

AOC tables assign charging rules to call prefixes. AOC table is assigned to E1/T1 trunk with enabled charging generation (see E1/T1 Trunks). Each trunk can use different AOC table than means different charging rules.

Charging rules are defined in time tables and existing time table can be assigned to the call prefix in AOC table. AOC table should define all prefix numbers you want to generate charging information for.

GUI-version : 1	.2 / Board Type : 1	Logout					
voi	ceo	data router					
		inoteska					
Home Use	ers Trunks	Traffic Globals System Diagnostics Specials					
Traffic > AOC Ta	ables Settings						
AOC Table	s Settings	AOC Tables Time Tables Holidays					
AOC Table	s						
	Name	Create New Table Create					
	default	Edit Details Delete Table					
Download AOC File Right Click on the above link and download using the 'Save Link As' option							
		Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>					

To create new AOC table enter new table name in the **Create New Table** field and click on the **Create** button. To edit existing AOC table use **Edit Details** button and to remove AOC table use **Delete Table** button.

AOC table details screen shows list of defined prefixes and associated timetables. To add new item to the AOC table enter call prefix into the **Prefix** field, choose time table to use and click on the **Add** button. The **Edit** button allows you to change time table assignment for existing prefix number. The **Remove** button can be used to delete selected item from the AOC table.

VOICE&DATA ROUTER	
GUI-version : 1.2 / Board Type : 1	Logout
voicosolata	
VOICENDALA router	
	instado
	uwuesha
Home Users Trunks Traffic Globals System Diagnostics Specials	
Traffic > AOC Tables Settings	
AOC Table: default	
Prefix Time Table table1 💌 Add Edit Remove	
00301 table4	
00302 table4	
00305 table4	
003060 table4	
003061 table4	
003062 table4	
003063 table4	
003064 table4	
003065 table4	
003066 table4	
003067 table4	
003068 table4	
0030691 table4	
0030592 table4	
0032 table4	
0033 table4	
00330 table4	
00331 table4	
00332 table4	
00333 table4	
Cancel	

Time tables define charging rules – frequency and number of charging units. Time table can divide the day into zones with different charging rules. Each time table defines charging rules separately for working day, weekend and public holiday. Charging rules allow setting of three time intervals for each call with different parameters.

GUI-version : 1.	.2 / Board Type : *	1 Voce&Data Router F	teloaded !!		Logout
voi	ce <mark>o</mark>	data	router		
					inoteska
Home Use	ers Trunks	Traffic Globals	System Diagnostics	Specials	
Traffic > AOC Ta	bles Settings				
AOC Table	s Settings				AOC Tables Time Tables Holidays
Time Table	s				
	Name			Create New Table	Create
	table1				Edit Details Delete Table
	table2				Edit Details Delete Table
		Соруг	ight © 2010-2012, All Right	ts Reserved <u>www.inoteska.sk</u>	

To create new time table enter new name in the **Create New Table** field and click on the **Create** button. To edit existing time table use the **Edit Details** button and to remove time table use the **Delete Table** button. Time table can be removed only if it is not assigned to any prefix in the AOC tables.

Time table details show charging rules for

- Working day
- Weekend
- Holiday

You can use the **Create** button to define new day time zone and the **Edit Details** button to change charging parameters of existing time zone. The **Remove** button can be used to delete selected time zone.

GUI-ve	ersion : 1.2 / Bo	ard Type :	1								Logout
	dice	5		a F							
V	OICe	50	0	au		ro	UCe	er			
											inoteska
Home	e Users	Trunks	Traffic	: Glob	als 9	iystem	Diagnos	tics :	5pecials		
Time	Teblee: te	aungs									
- Inne	Tables, ta	blea									
	_	_	_	_	V	/orking D	)ay	_	_		Create New Item Create
	START	LTI1	CHI1	CHU1	LTI2	CHI2	CHU2	CHI3	CHU3		
	00:00	60		50				6	5		Edit Details Remove
	07:00	60		100				3	5		Edit Details Remove
	18:00	60		50				6	5		Edit Details Remove
						Weeken	d				Create New Item Create
	START	LTI1	CHI1	CHU1	LTI2	CHI2	CHU2	CHI3	CHU3		
	00:00	60		50				6	5		Edit Details Remove
		_		_		Holiday			_		Create New Item Create
	START	LTI1	CHI1	CHU1	LTI2	CHI2	CHU2	CHI3	CHU3		
	00:00										Edit Details Remove
									Holida	y Table 🛛 💌	Set
								Cancel			

Charging parameters:

- **START** Beginning of day charging zone
- LTI1 Length of the first time interval of the call
- CHI1 Charging interval during the first time interval
- CHU1 Number of charging units for the first time interval
- LTI2 Length of the second time Interval of the call
- CHI2 Charging interval during the second time interval
- CHU2 Number of charging units for the second time interval
- CHI3 Charging interval during the third time interval
- CHU3 Number of charging units for the third time interval

**START** defines beginning of day charging zone. It can be set in one hour step. The first zone must start at 0:00 and the last zone automatically ends at 23:59.

**LTI1**, **CHI1** and **CHU1** define charging generation for the first time interval of the call. LT1 is length of the interval in seconds. CHI1 designates charging rate during this period. It is configured in seconds with precision on tenths of the second (you can use decimal numbers like 1.5). CHU1 specifies number of charging units to generate.

LTI2, CHI2 and CHU2 define charging generation for the second time interval of the call.

CHI3 and CHU3 define charging generation for the rest of the call up to the call end.

It is not necessary to set all parameters and they can be unset (zero).

- If all parameters are not set then no charging is generated (call is free of charge)
- If time interval length is not set then charging interval is ignored.
- If time interval length is not set and no charging units are defined then time interval is skipped and it continues with next one.
- If time interval length is not set and charging units exist then specified charging units are generated once and it continues with next time interval.
- If charging units are not set then no charging is generated for this time interval.

• If charging interval is not set then charging units are generated only once for this time interval.

## Example 1

START	LTI1	CHI1	CHU1	LTI2	CHI2	CHU2	CHI3	CHU3
0:00	0	0	0	0	0	0	10	1

This setting will generate regularly 1 charging unit every 10 seconds during the whole call. It does not matter what time you call.

## Example 2

START	LTI1	CHI1	CHU1	LTI2	CHI2	CHU2	CHI3	CHU3
0:00	0	0	1	60	1	1	10	1
7:00	0	0	5	60	1	2	2	1
19:00	0	0	1	60	1	1	10	1

This setting is more complex. Within hours 7:00 - 19:00 you will get 5 charging units immediately after call answer then 2 charging units per second during the first minute of the call and 1 charging unit every 2 seconds during the rest of the call. At the time 19:00 - 7:00 you will get 1 charging unit after the call answer and 1 charging unit every second during the first minute of the call. And then it will continue with 1 charging unit every 10 seconds up to the end of the call.

Charging rules for holidays are needed only if some holiday table is assigned to the current time table.

Holidays section defines public holidays. Each holiday is defined as a date (day and month). Holiday table can be assigned to the time table to apply special charging rules for these days.

GUI-version : 1.	.2 / Board Type : 1	Jata	router			Logout
					inc	ıteska
Home Use	ers Trunks T	raffic Globals	System Diagnostics	Specials		
Traffic > AOC Ta	bles Settings					
AOC Table	s Settings				AOC Tables Time Tables	Holidays
Holidays						
	Name			Create New Table	Create	
	holidays				Edit Details Delete Table	
		Copyr	ight© 2010-2012, All Righ	ts Reserved <u>www.inoteska.sk</u>		

To create new holiday table enter new name in the **Create New Table** field and click on the **Create** button. To edit existing holiday table use the **Edit Details** button and to remove holiday table use the **Delete Table** button.

Holidays table details show list of public holidays. Use the **Add** and **Remove** buttons to edit the holiday table.

VOICE&DATA ROUTER	
GUI-version : 1.2 / Board Type : 1	Logout
voice&data router	wteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Traffic > AOC Tables Settings	
Holidays: holidays	
Date (Day/Month) 1 Add   Holidays 1.1. 6.1.   6.4. 9.4. 9.4.   9.4. 1.5. 8.5.   5.7. 29.8. 1.9.   1.9. 15.9. V	
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Note that correct functionality of generation of charging information depends on setting of Voice&Data Router system time. See <u>Date and Time</u> setting for more details about system time adjustment.

# **GLOBALS MENU**

# SIP settings

SIP settings define common options of SIP protocol valid for all SIP users and SIP trunks. SIP protocol options are divided into the following groups:

- General Preferences
- TOS
- NAT
- Miscellaneous
- Registrations
- Authentications
- Jitter Buffer
- Codecs

### **General Preferences**

GUI-version : 1.8 / Board Type : 1	Logout
voice&data router	ingtocka
	uwuesha
Home Users Trunks Traffic Globals System Diagnostics	
Globals > SIP Settings	
SIP (Session Initiation Protocol) Configuration Settings	
General Preferences TOS NAT Miscellaneous Registrations Authentications Jitter Buffer	
Realm for Digest Authentication Vol. router	
UDP Port to Bind to 5060	
IP Address to Bind to 0.0.0.0	
Domain	
Overlap Dialling Support 🕢 💿	
Allow Transfers 🗹 💷	
Enable DNS SRV Lookups (on Outbound Calls) 🕢 ગ	
Pedantic 🔲 🥐	
Always Auth Reject 🗹 🖤	
SIP Domain Support Auto Domain 🥅 💿	
Allow External Domains 🖉 💿	
Save Cancel	
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### **Realm for Digest Authentication**

Realm for digest authentication. Realms must be globally unique according to RFC 3261. You can set realm to your host name or domain name.

## UDP Port to Bind to

UDP port that Voice&Data Router will listen on (SIP standard port is 5060).

### IP Address to Bind to

IP address that Voice&Data Router will listen on. The address 0.0.0.0 tells device to listen on all interfaces. Use LAN or WAN IP address to limit SIP connections only for selected interface.

### Domain

Default domain setting. If configured, Voice&Data Router will allow INVITE and REFER messages only to defined domains. REGISTER to non-local domains will be automatically denied if a domain list is configured.

## **Overlap Dialing Support**

Enable/disable overlap dialing support.

### Allow Transfers

Enable/disable call transfers.

## Enable DNS SRV Lookups (on Outbound Calls)

Enable DNS SRV lookups on outbound calls. Voice&Data Router only uses the first host in SRV records. Disabling DNS SRV lookups disables the ability to place SIP calls based on domain names to some other SIP users on the Internet.

#### Pedantic

Enable slow, pedantic checking of Call-IDs, multiline SIP headers and URI-encoded headers. This option enables more strict SIP RFC compliancy.

#### **Always Auth Reject**

When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with an identical response (401 Unauthorized) instead of letting the requester know whether there was a matching SIP account username for their request. This reduces the ability of an attacker to scan for valid SIP usernames.

#### Auto Domain

Enable automatic adding of local host name and local IP to domain list.

## Allow External Domains

Enable/disable INVITE and REFER to non-local domains.

## TOS

GUI-version : 1.8 / Board Type : 1			
voice&data router	inoteska		
Home Users Trunks Traffic Globals System Diagnostics			
Globals > SIP Settings			
SIP (Session Initiation Protocol) Configuration Settings			
General Preferences TOS NAT Miscellaneous	Registrations Authentications Jitter Buffer		
TOS for Signalling Packets CS3 🔻 🥯	TOS for RTP Audio Packets 🛛 🖝 🖤		
Trust Remote Party ID 🕢 🦈	Send Remote Party ID 🕜 ગ		
Generate In-Band Ringing 🛛 never 🔻 🎱	Server User Agent 🛛 Inoteska Voice&Data R 🌻		
Allow Nonlocal Redirect 📃 ગ	Add 'user=phone' to URI 🔲 🕐		
DTMF Mode 🛛 r fc2833 🔻 🎱	Send Compact SIP Headers 🔲 ગ		
Max Registration/Subscription Time 3600			
Min Registration/Subscription Time 🐻 🧶			
Default Incoming/Outgoing Registration Time 1800 🤎			
Min Roundtrip Time (T1 Time) 100			
Save Cancel			
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### **TOS for Signaling Packets**

TOS (Type of Service) for outgoing SIP signaling packets.

### **TOS for RTP Audio Packets**

TOS (Type of Service) for outgoing RTP packets.

### **Trust Remote Party ID**

This option specifies whether or not Voice&Data Router should trust the value in the Remote-Party-ID header.

### Send Remote Party ID

This parameter specifies whether or not Voice&Data Router should send a Remote-Party-ID header.

### **Generate In-Band Ringing**

Configuration of in-band ringing generation.

- **yes** always send 180 Ringing (if it hasn't been sent yet) followed by 183 Session Progress and in-band audio tones.
- **no** send 180 Ringing if 183 has not yet been sent establishing audio path. If audio path is established already (with 183) then send in-band ringing.
- never Whenever ringing occurs, send 180 Ringing as long as 200 OK has not yet been sent.

#### Server User Agent

Customizing User-Agent field in the SIP messages.

#### Allow Nonlocal Redirect

Enable support for 302 Redirects.

#### Add 'user=phone' to URI

This option tells Voice&Data Router to add ";user=phone" to SIP URIs that contain a valid phone number.

### **DTMF Mode**

Set default mode for sending DTMF digits.

### Send Compact SIP Headers

Enable compact format of the SIP headers, which may be required if the size of the SIP header is larger than the maximum transmission unit (MTU), causing the IP packet to be fragmented. Do not use this option unless you know what you are doing

### Max Registration/Subscription Time

Maximum allowed time of incoming registrations and subscriptions (in seconds).

#### **Min Registration/Subscription Time**

Minimum allowed time of incoming registrations and subscriptions (in seconds).

## **Default Incoming/Outgoing Registration Time**

Default expiration time of SIP registration for incoming and outgoing registrations (in seconds). For incoming registration default value you set here is used only if the client does not specify a timeout when it registers. If you are registering to another user agent server (UAS), this is the registration timeout that it will send to the far end.

### Min Roundtrip Time (T1 Time)

Minimum round-trip time for messages to monitored hosts (in milliseconds).

### NAT

### Extern IP

IP address that will replace the private IP address in outbound SIP messages if Voice&Data Router is behind a NAT.

### **Extern Host**

Fully qualified domain name to determine extern IP address. Voice&Data Router performs periodic DNS lookups on this name and replaces the private IP address with the IP address returned from the DNS lookup in case it is behind the NAT.

### **Extern Refresh**

Time period of regular DNS lookups on the extern host (in seconds).

## **Local Network Address**

This is used to tell Voice&Data Router which IP addresses are considered local, so that the address in the SIP header can be translated to that specified by extern IP or extern host. The IP addresses should be specified in CIDR notation (e.g. 192.168.1.0/24).

### NAT Mode

Select default behavior when Voice&Data is on a public IP and communicating with devices hidden behind a NAT.

- yes always ignore info and assume NAT
- Server User Agent use NAT mode only according to RFC3581 (;rport)
- never never attempt NAT mode or RFC3581 support
- route assume NAT, don't send rport

GUI-version : 1.8 / Board Type : 1	Logout
voice t de te	
VOICE QUALA router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics	
Globals > SIP Settings	
SIP (Session Initiation Protocol) Configuration Settings	
General Preferences TOS NAT Miscellaneous Registrations	Authentications Jitter Buffer
Extern IP	9
Extern Host	9
Extern Refresh 10 🤍	
Local Network Address 192.168.1.0/24	?
NAT Mode yes	• 🦻
Allow RTP Reinvite 🛛 yes 🔍 🔍	
Save	
Convirtent@2010.2016_All Picture Reserved www.i	nnteska sk
Copyright @ 2010-2010, All Kights Keselved <u>www.i</u>	

## Allow RTP Reinvite

Configure default media path redirection.

- yes allow direct RTP media stream
- **nonat** allow direct RTP media stream but only when the peer where the media is being sent is known to not be behind a NAT
- update use UPDATE instead of INVITE

## **Miscellaneous**

GUI-version : 1.8 / Board Type : 1	Logout
voice t de la la	
VOICe QUALA router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics	
Globals > SIP Settings	
SIP (Session Initiation Protocol) Configuration Settings	
General Preferences TOS NAT Miscellaneous Registrations Authentications Jitter Buffer	
Generate Manager Events 🥅 🕥	
Outbound SIP Registrations	
Register Timeout 20 🤍	
Register Attempts 🛛 🕐	
Save Cancel	
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### Generate Manager Events

Enable/disable generation of Asterisk Manager events.

### **Register Timeout**

Time interval to retry unsuccessful outbound registration calls (in seconds).

### **Register Attempts**

Number of outbound registration attempts before giving up. Setting to 0 means that Voice&Data Router will retry indefinitely.

### **Enable Jitter Buffer**

Enable/disable the use of an RTP jitter buffer on the receiving side of a SIP channel.

### **Registrations**

These global outbound SIP registrations can be useful for instances when Voice&Data Router needs to register multiple users/extensions to a single SIP proxy (provider).

GUI-version : 1.8 / Board Type : 1	nta router	Logout
		inoteska
Home Users Trunks Traffic	Globals System Diagnostics	
SIP (Session Initiation Protoco	I) Configuration Settings	
General Preferences TOS	Debug Notify NAT Miscellaneous Registrations Authentications Jitter E	Buffer Codecs
Outbound SIP registrations can be usefu	ul for instances when Voice&Data Router needs to register multiple users/extensions to a sir	ngle SIP proxy (provider).
Username	Host	Add
0445567755	sip.wn.telekom.sk:5060	Edit Delete
0445567758	sip.wn.telekom.sk:5060	Edit Delete
	Save	
Copyright © 2010-2016, All Rights Reserved <u>www.inoteska.sk</u>		

Click on **Add** button to create new outbound SIP registration. Use the **Edit** button to modify selected SIP registration. And use the **Delete** button to remove outbound SIP registration from the configuration.



### **Username** Registration user name.

### Host

SIP registrar server address (IP address or host name)

#### AuthUser

User name used for the authentication. Use this setting only if it differs from the registration user name above.

#### Password

Password used for the authentication.

### **Authentications**

Global credentials for outbound calls can be used when a proxy challenges Voice&Data Router for authentication. These credentials override any credentials in user/trunk definition if realm is matched. This way, Voice&Data Router can authenticate for outbound calls to other realms.

GUI-versio	n : 1.8 / Board Type : 1		Logout
vo	iceo	data router	inoteska
Home	Users Trunks	Traffic Globals System Diagnostics	
Globals ≻ Sl	P Settings		
SIP (Se	ssion Initiation P	Protocol) Configuration Settings	
(	General Preferences	TOS   Debug Notify   NAT   Miscellaneous   Registrations   Authentications   Jitter Buffer   0	Codecs
Glo	bal credentials for outb user/trunk de	iound calls, i.e. when a proxy challenges Voice&Data Router for authentication. These credentials override any cr efinition if realm is matched. This way, Voice&Data Router can authenticate for outbound calls to other realms.	edentials in
	Realm	Username	Add
	KUCAPACA	5567755	Edit Delete
	KUCAPACA	5567758	Edit Delete
	KUCAPACA	5567759	Edit Delete
		Save	
		Copyright @ 2010-2016, All Rights Reserved <u>www.inoteska.sk</u>	

Click on **Add** button to create new global authentication. Use the **Edit** button to modify selected global authentication. And use the **Delete** button to remove global authentication from the configuration.

GUI-version: 1.8 / Board Type : 1	Logout
	inoteska
Home Users Trunks Traffic Globals System D	iagnostics
Globals > SIP Settings	
SIP (Session Initiation Protocol) Configuration Se	Global Authentication Realm KUCAPACA ? Username 5567755 ? Password jh3ygdsbj38 ? Save Cancel
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inoteska systems for voice and data communication

### Realm

Authentication realm. If it is matched to realm in proxy challenge, these authentication credentials override any credentials in user/trunk definition.

### Username

Authentication user name for specified realm.

### Password

Authentication password for specified realm.

## **Jitter Buffer**



## Force Jitter Buffer

Force the use of the RTP jitter buffer on the receiving side of a SIP channel.

### Max Jitter Buffer

Maximum length of the jitter buffer (in milliseconds).

### **Resync Threshold**

Jump in the frame timestamps over which the jitter buffer is resynchronized. It is useful to improve the quality of the voice, with big jumps in/broken timestamps (in milliseconds).

### Implementation

Jitter buffer implementation, used on the receiving side of a SIP channel. Two implementations are currently available

- fixed jitter buffer has fixed size that equals to maximum defined length
- adaptive jitter buffer size varies up to the maximum defined length

### Codecs

Global codecs setting.



# **RTP/UDPTL Settings**

RTP and UDPTL protocol configuration. Here you can define port range allocated for voice (RTP) and T.38 fax (UDPTL) transmission.

GUI-version : 1.1 / Board Type : 1	oto router	Logout
Voicodo		inoteska
Home Users Trunks Traf	fic Globals System Diagnostics Specials	
Globals > RTP/UDPTL Settings		
RTP/UDPTL Settings		
RTP/UDPTL Port Range		
	Start port 🧐 Iannn	
	End port 🦻 8099	
	Save	
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### **Start Port**

First port number reserved for RTP/UDPTL protocol.

### End Port

Last port number reserved for RTP/UDPTL protocol.

Note that only odd port numbers (half of the allocated range) can be used for RTP or UDPTL stream. Odd port numbers are reserved for RTCP protocol.

## Call Groups

Call groups can group together outgoing trunks. Call group joins multiple TDM trunks (E1/T1, analogue and GSM) to a single group for outgoing calls (grouping of VoIP trunks is not supported). Such group can be used in the calling rules as the call destination (user to trunk or trunk to trunk). Outgoing call is placed on the first idle channel which is found according method defined in call group configuration.

Call groups have predefined channel order that is fixed. Channel order is determined by arrangement of Voice&Data Router internal TDM bus.

Internal numbering of TDM channels

Channel	Interfaces
0 - 127	E1/T1 – 4 x 32 channels
128 - 143	GSM – 16 channels (only 12 can be used)
144 - 175	Analogue – 32 channels
176 - 223	BRI – 16 x 3 channels

GUI-version : 1.1 / Board Type :	data router	Logout
	Ter file - Clabelle - Constant - Disconstantino - Causia la	inoteska
Globals > Call Groups List	Tramic Globalis System Diagnostics Specialis	
Call Groups Settings		New Call Group
Call Groups List		
Call Group Name	Interfaces	
E1	E1: E1-A, E1: E1-B	Edit Delete
FXO	FXO: FXO1, FXO: FXO2, FXO3	Edit Delete
GSM	GSM: GSM1 Orange, GSM: GSM2 T-Mobile, GSM: GSM3 O2	Edit Delete
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To create new call group click on the **New Call Group** button. Use the **Edit** button if you want to modify existing call group. And use the **Delete** button to remove call group from the configuration. Call group can be removed only if there does not exist any reference to this group in the calling rules.

### **Call Group Name**

A unique label to help you identify call group. This identifier is used over all of Voice&Data Router configuration where we need reference to this call group.

### **Call Group Members**

List of call group members. Use arrow buttons to add group members from available trunks' list or to remove them from the call group.

### **Available Trunks**

List of available TDM trunks, which can be added to the call group.

### **Channel Selection**

Idle channel selection to place outgoing call.

- Ascending search for the lowest available channel number
- Descending search for the highest available channel number

Note that internal fixed channel order is always used for the channel searching, does not matter what is displayed in call group members' list.

### Cyclic

Enable cyclic search of the first available channel instead of sequential.

VOICE&DATA ROUTER	
GUI-version : 1.1 / Board Type - 1	Logout
voicesdata router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > Call Groups List	
Edit Call Group: GSM	
Call Group name GSM    Call Group Members Available Trunks   GSM: GSM1 Orange FX0: FX01   GSM: GSM3 O2 FX0: FX02   FX0: FX04 FX0: FX04   FX0: FX06 FX0: FX06   FX0: FX07 FX0: FX06   FX0: FX08 III: E1-A   Call Group options Channel Selection   Cancel Save	

# Ring Groups

Ring groups join user extensions to the single group that can ring a group of phones simultaneously, stopping when any one of them is picked up. Such group can be used in the calling rules as the call destination (user to user or trunk to user).

Create new ring group by clicking on the **New Ring Group** button. Use the **Edit** button if you want to modify existing ring group. And use the **Delete** button to remove selected ring group from the configuration. Ring group can be removed only if there does not exist any reference to this group in the calling rules.

GUI-version : 1.1 / Board Type : 1 Voce&Data Ro	uter Reloaded !!		Logout
voicesdot			
VOICEQUCIU	CITOULEI		
			inoteska
Home Users Trunks Traffic Glob	als System Diagnostics	Specials	
Globals > Ring Groups			
Ring Groups Settings			New Ring Group
Ring Group List			
Ring Group List	Members		
GROUP1	10, 11, 12		Edit Delete
GROUP2	20, 21, 22, 23		Edit Delete
	Copyright © 2010-2012, All Rig	hts Reserved <u>www.inoteska.sk</u>	

## **Ring Group Name**

A unique label to help you identify ring group. This identifier is used over all of Voice&Data Router configuration where we need reference to this ring group.

### **Ring Group Members**

List of ring group members. Use arrow buttons to add group members from available users' list or to remove them from the ring group.

GUI-version : 1.1 / Board Type : 1	Logout					
voice&data router	inoteska					
Home Users Trunks Traffic Globals System Diagnostics Specials						
Ring Group Name GROUP1						
Ring Group Members Available Users						
10 (SIP) Jack Daniel's 11 (SIP) Johnnie Walker 12 (SIP) Jim Beam 12 (SIP) Jim Beam 12 (Analog) Seth Morrison 23 (Analog) Seth Morrison 23 (Analog) Seth Morrison 23 (Analog) Glen Plake 25 (Analog) Kaj Zacrisson 26 (Analog) Mike Hattrup 27 (Analog) Chris Davenport						
Ring Group Options						
Strategy Ring in Order						
If not answered Goto Hangup						
Save						
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**INDERSTRA** SYSTEMS FOR VOICE AND DATA COMMUNICATION

# Available Users

List of available user extensions, which can be added to the ring group.

# Strategy

Define what ringing strategy is used

- Ring All Simultaneously all phones are ringing at once
- Ring in Order ring group phones are ringing step by step

## Seconds to Ring Each Member

Enter time how long to ring phones in the ring group.

## If Not Answered Goto

Define what to do when nobody answers.

- Hangup call will be refused (with busy tone)
- Congestion call will be refused (with congestion tone)

## Carriers

Alternative carriers supporting two-stage dial service can be used to lower communication costs. Twostage dialing allows you to dial carrier access number first (this is usually low-tariff number). And then after call answer, DTMF tones are used to authenticate and dial destination number. Two step calls over alternative carrier can be enabled in the calling rules (user to trunk or trunk to trunk) and Voice&Data Router makes such calls automatically so it appears like ordinary call.

GUI-version : 1.1 / E		nta router		Logout inoteska
Home Users	Trunks Traffic	Globals System Diagnostics	Specials	
Globals > Carriers				
Carriers Setti	ngs			Create New
List of Carrier	5			
	Name	Carrier Prefix	Ringing Tone	
м	OJ_OPERATOR	012004	Generate	Edit Delete
		Conversets @ 2010-2012 &II Pig	hts Basawad www.instast/s.sk	

To create new carrier, click on the **Create** New button. Use the **Edit** button to modify existing carrier configuration. And use the **Delete** button to remove selected carrier from the configuration. Carrier can be removed only if does not exist any reference to this carrier in the calling rules.

GUI-version	n : 1.1 / Board Type : 1	Logout
vo	ice&data router	
	inota	zska
Home	Users Trunks Traffic Globals System Diagnostics Specials	
Globals > Ca	arriers	
Carrier's	s Options - MOJ_OPERATOR	
Details		
	Name 🧶 MOJ_OPERATOR	
	Carrier Prefix 🛞 012004	
	Security Code 🥬 123456789	
	Ringing Tone 🤍 🗹	
	Save	
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## Name

A unique label to help you identify carrier. This identifier is used over all of Voice&Data Router configuration where we need reference to this carrier.

### **Carrier Prefix**

Carrier access number.

#### **Security Code**

Customer PIN code to authenticate.

#### **Ringing Tone**

Enable false ringing tone generation immediately when call started. Use this feature to prevent long period of silence during connection set-up.
# **Permissions**

Permissions are used to limit outgoing communication of local users. Every user extension has assigned permission group. Permission group is a collection of permission flags which can be assigned to the calling rules (user to user or user to trunk). User is allowed to make a call only if his permission group includes call permission flag assigned to calling rule for dialed number. If there is no permission flag assigned to the calling rule, permissions are not tested and all users are allowed to proceed.

GUI-version : 1.1	1 / Bo	ard T	'ype : 1												Logou
				h		۰.									
VOIC	-6	20	$\overline{\mathbf{N}}$	C	C	١t	0	r		U	e197	-			
															inatoska
Home	-	Term	ske	Tra	ffi-	c	loba le	Sve	000	Dia	apostics		Enacia	le.	u de la ma
Globals > Permissi	ions	Trui	172	114			obais	595			gnostics		эреста	12	
Permission	Set	tine	10												Save Changes Cancel Changes
		cirity	10												Dave onlanges
User permis	sion	grou	p defin	es a	colle	ction	of call p	ermis	sion	lags	(1 to 16) Iling rub	. Use	r perr	nissio Usor	ion groups are assigned to the users. Call permission flags are
	U	seris	allow	ed to	mak	eaca	all only if	f his p	ermis	sion	group in	iclude	serio es call	perm	mission flag assigned to calling rule for this call.
Licor Bormia															
User Permis	2	1.01	oups		-7	0		40		40	42		45	40	Demoission Comm
	ত	4	0 🔽	•		8	9			12	15	14	15	10	Without limitation
															sk
															SkTC3
															International
Call Permiss	sion	Flag	as												
							Perm	nissin	n Flag				Flag	Name	ne
						1		1			Interr	nal	Thug		
								2			Local				
						-1		3			Toll				
								4			cz				
						-1		5			Interr	nation	nal		
								6			Mobile	в			
						-1		7			flag 7				
								8			flag 8				
						-1		9			flag 9				
								10			flag 1	0			
						-1		11			flag 1	1			
								12			flag 1	2			
								13			flag 1	3			
								14			flag 1	4			
								15			flag 1	5			
								16			flag 1	6			
						L									
							Сору	right	9 201	0-201	2, All Ri	ghts F	Reser	ved <u>w</u>	www.inoteska.sk

#### **User Permission Group**

This table defines permission groups which are assigned to the user extensions. Voice&Data Router supports 12 permission groups. Permission group is a set of 16 call permission flags. You can use

check boxes to select call permission flags for each group. Every group can have symbolic name to easier understanding.

User permission group with all call permission flags checked is the most privileged group. There are no restrictions for the user with such group because all permission flags are allowed.

On the other hand user permission group that does not have selected any call permission flag can be limited only for incoming calls. In that case all outgoing calls matching the calling rules with assigned call permission flag will be refused. Only calling rules which don't use call permission flags are allowed for such user.

#### **Call Permission Flags**

This table defines call permission flags. There are 16 permission flags and you can assign each of them symbolic name for better orientation. These identifiers are used in calling rules configuration where call permission flags are associated to dialed number patterns.

# Services Codes

Users' service codes configuration. Here you can define dial codes for activation, deactivation or execution of users' services form the phone. Note that the user is allowed to use configured service codes only if his dial plan includes service codes (see <u>dial plan</u> configuration).

GUI-version : 1.1 / Board Type : 1 Voce&Data Router Reloaded !!	Logout
voice&data rou	ter
	INDLESKa
Home Users Trunks Traffic Globals System D	iagnostics Specials
Globals > Services Codes	
Services Codes Settings	
Set Call Forwarding *21*	
Set Call Forwarding if user is busy *22*	
Set Call Forwarding if no respond *23*	No respond time: 15
Clear Call Forwarding *21#	
Clear Call Forwarding if user is busy *22#	
Clear Call Forwarding if no respond *23#	
Set Do Not Dicturk #79	
Olean De Net Disturb	
Call Pickup *8	
	Save Cancel
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#### Default service codes setting

Service	Activation	Deactivation	Execution
Call forwarding (unconditional)	*21*	*21#	
Call forwarding if busy	*22*	*22#	
Call forwarding if not answered	*23*	*23#	
Do not disturb	*78	*79	
Call pickup			*8

# **CDR Settings**

CDR – Call Detail Records – save information about Voice&Data Router calls. CDR records are stored in the text file as comma separated values.

GUI-versio	n : 1.1 / Board Type : 1	Logout
vo	iceodata router	
		inoteska
Home	Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CD	DR Settings	
CDR Se	ttings CDR Details	CDR Records
CDR De	tails	
	CDR Note 💿 TEST	
	CDR User to User 💷 🔲	
	CDR Trunk to User 💷 📃	
	CDR User to Trunk 💿 🗹	
	Save	
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#### **CDR Note**

Here, you can enter optional string that will be added to every CDR.

#### **CDR User to User**

Enable to log CDR for local calls between Voice&Data Router users.

#### **CDR Trunk to User**

Enable to log CDR for Voice&Data Router incoming calls.

#### CDR User to Trunk

Enable to log CDR for Voice&Data Router outgoing calls. This option also enables transit calls CDR – trunk to trunk.

You can click on the **CDR Records** button to view stored CDR records. Use the **Refresh View** button to update displayed list CDRs or click on the **Delete File** to clear stored CDR records.

CDR record fields:

- CDR note string you configured in CDR Details
- Source number caller identification number
- Destination number dialed number
- Source channel incoming channel internal name
- Destination channel outgoing channel internal name
- Date/Time call timestamp
- Duration total time in system, in seconds
- Billing time total time call is up, in seconds
- Disposition what happened to the call (ANSWERED, NO ANSWER, BUSY, FAILED)
- Call type call type from routing point of view (LOCAL, IN, OUT)

OICE&DATA ROUTER	
GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CDR Settings	
CDR Settings CDR Details	CDR Records
CDR Records	
Refresh View Delete File	
TEST, 10,22,SIP/10-0000004,UniCall/146-ANA18589397,2012-05-14 15:37:38,6,4,ANSWERED,LOCAL TEST, 10,0104155,SIP/10-0000005,UniCall/1-E142fb0a00,2012-05-14 15:37:50,6,5,ANSWERED,OUT TEST,10,0104155,SIP/10-00000007,UniCall/1-E1578829ef,2012-05-15 08:03:00,32,0,NO ANSWER,OUT TEST,10,0104155,SIP/10-00000007,UniCall/1-E15653bb3f,2012-05-15 08:03:53,29,0,NO ANSWER,OUT TEST,010,UniCall/1-E119143216,UniCall/2-E1412c70a,2012-05-15 08:13:34,6,0,NO ANSWER,OUT TEST,012,UniCall/1-E139fe8b61,UniCall/2-E1405ac77c4,2012-05-15 08:13:47,2,0,NO ANSWER,OUT TEST,012004,UniCall/1-E14f6c15eb,UniCall/2-E16505aff,2012-05-15 08:13:57,1,0,FAILED,OUT TEST,012000,UniCall/1-E125fc1d01,UniCall/2-E17af07032,2012-05-15 08:14:07,0,0,FAILED,OUT TEST,010003,UniCall/1-E162b55c8a,UniCall/2-E10d2149a0,2012-05-15 08:14:13,12,6,ANSWERED,OUT TEST,,s,UniCall/1-E1617063d1,UniCall/147-ANA7578bebc,2012-05-15 08:20:52,20,0,NO ANSWER,IN TEST,,s,UniCall/1-E17dfd4416,UniCall/146-ANA6fb96760,2012-05-15 08:21:23,29,26,ANSWERED,IN	
Download CDR File Right Click on the above link and download using the 'Save Link As' option	
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You can use link **Download CDR File** to download CDR file from Voice&Data Router for additional processing.

# Announcement Settings

Announcement management is used for configuration of announcement files for call routing. New announcement file can be recorded from user phone or existing file can be uploaded. Only files of supported format can be uploaded. Voice&Data Router uses Microsoft WAVE files with following format

Sample rate Sample size	8 kHz 16 bit	Sample rate Sample size	8 kHz 8 bit
Channels	1 (mono)	Channels	1 (mono)
Auto format	PCM	Audio format	CCITT A-Law CCITT µ-Law

GUI-version : 1.1 / Board Type : 1	da	ita re	outer					Logout inoteska
Home Users Trunks	Traffic	Globals Syste	m Diagnostics	Specials				
Globals > Announcement Settings								
Announcement Setting	s				Settings	Upload File	Record File	Play Sound
Announcement Files								
	#	Announcement Fi	le Name					
	1	tada.wav			Download File	Delete File		
	2	test.wav			Download File	Delete File		
	h	A	2010 2010 41 5:					

To download announcement file from Voice&Data Router click on the **Download File** button for selected announcement. Download link appears to save file on your local PC. To remove announcement file from Voice&Data Router use the **Delete File** button. File removal is allowed only if there does not exist any reference to this announcement in the calling rules.

Use the **Upload File** button to switch to announcement upload form. You can select WAV file to upload and click on **Upload** button. Only files of supported format be used – see description above.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > Announcement Settings	
Announcement Settings	Settings Upload File Record File Play Sound
Upload Announcement File	
Upload	
<ul> <li>tada.wav (audio/wav) - 31128 bytes</li> </ul>	
Upload	
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Use the **Record File** button to switch to announcement recording form. Voice&Data Router allows you to record your voice message file from your phone. You need to select one user extension, enter new announcement name and click on the **Start** button. Voice&Data Router rings selected user extension. Pick up the phone, wait for beep sound and then say your new voice message. When you finish press # key on your phone or hang-up.

GUI-version : 1.1 / Board Type : 1	Logout
voice Scloto router	
Vorcedociterrooter	inoteska
Home Users Trunks Traffic Globals System Diagnostics	Specials
Globals > Announcement Settings	
Announcement Settings	Settings Upload File Record File Play Sound
Record New Announcement File	
Select a user and enter new name of the audio file with	out an extension (such as.: welcome_message).
Pick up the phone. Wait for the sound tone	e and then say new voice message.
Confirm the end of message by	pressing "#" at the phone.
Record File	
Extension SIP: 10 💌 Announcement File Name test	Start
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Use the **Play Sound** button to test announcement files. Voice&Data Router can play back selected announcement to your phone. In this way you can test ability to play uploaded files or recording voice quality. Select one user extension, select announcement file and click on the **Start** button. Voice&Data Router rings selected user extension and after phone pick-up it plays back selected voice file.

GUI-version : 1.1 / Board Type : 1	Logout
VOICe QUALA router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > Announcement Settings	
Announcement Settings Settings Upload File	Record File Play Sound
Play Announcement File	
Select a user and the name of sound file. After pressing "Start" button selected user is ringing. Pick up the phone and the sound file is played.	
Play Sound	
Extension SIP: 10 🗸 Announcement File tada.wav 🗙 Start	
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# CAS National Variants

There are various country-specific variants of E1 CAS signaling protocol. Voice&Data Router offers set of configurable parameters to adapt CAS signaling to specific requirements of each installation. Device is supplied with several predefined variants of CAS signaling you can use and you are allowed to customize these predefined variants for your needs or to create your own variant. Modified CAS signaling variant can be then assigned to E1 port in E1/T1 Trunks configuration.

To modify existing CAS signaling variant choose one variant from the selection list and click on **Edit Existing** button. To create new CAS signaling variant enter name of the variant and click on the **Create New** button.

voice data router	
inote	eska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CAS National Variants	
CAS National Variants Settings	
default     Image: Second	

CAS signaling variant configuration parameters is divided into five basic categories.

- Line Signals
- MFC/R2 Forward Signals
- MFC/R2 Backward Signals
- Options
- Timers

**Line Signals** define CAS line signaling. Line signaling supports 4 signaling bits (ABCD) in each direction. Only 2 of them (AB) are used and can be configured. Unused bits (CD) have fixed status (01) and they don't change. Signaling bits are transmitted over 16<sup>th</sup> signaling channel. Here you can configure AB signaling bits for particular line signals/states.

#### **Common Signals**

Line signals for both forward and backward direction.

#### Idle

Line idle/release signal.

#### Blocking

Line blocking signal, calls are not allowed.

#### **Forward Signals**

Line signals for forward direction (signals sent by the originating point).

#### Seizure

Line seizure signal – start new call.

#### Clear-Forward

Clear-Forward signal - normal call clearing initiated by calling party.

#### **Dialing Pulse**

Line signal to generate dialing pulses for provision of DNIS number.

VOICE&DATA ROUTER	
GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	
Ü.	noteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CAS National Variants	
CAS National Variants Settings - default	
Line Signals MFC/R2 Forward Signals MFC/R2 Backward Signals Options Timers	
Common Signals	
ldle 🕗 10 💌	
Blocking 🧐 11 💌	
Forward Signals	
Seizure 🧐 💿 💌	
Clear-Forward 🖤 10 💌	
Dialing Pulse 🤍 01 💌	
Backward Signals	
Clear-Back 🧐 11 🗸	
Forced Release 🧖 00	
Billing Pulse 🧐 🛛 💌	
Save	
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#### **Backward Signals**

Line signals for backward direction (signals sent by the terminating/destination point).

#### Seizure ACK

Line seizure acknowledge signal.

#### Answer

Line answer signal.

#### **Clear-Back**

Clear-Back signal – normal call clearing initiated by called party after call answer.

#### **Forced Release**

Forced Release signal – call clearing request before call answer.

#### **Billing Pulse**

Line signal to generate billing pulses.

**MFC/R2** Forward Signals and **MFC/R2** Backward Signals categories define MFC/R2 register signaling parameters. MFCR/R2 register signaling uses MFC/R2 tones (1 - 15) transmitted in voice channels to provide addressing information and call progress status. The forward as well as backward signals have primary and secondary meanings. The forward primary and secondary signals are usually described as group I and group II. The backward primary and secondary signals are described as group A and group B. Some countries extend MFC/R2 register signaling of backward group C. Here you can configure MFC/R2 tones (1 - 15) for register signals/states.

#### **Group I**

Primary forward signals are used to provide DNIS and ANI numbers.

#### End of DNIS

Indication of the end of DNIS (Dialed Number Identification Service) number.

#### End of ANI

Indication of the end of ANI (Automatic Number Identification) number.

#### **ANI not Available**

ANI number is not available - ANI restriction.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	oteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CAS National Variants	
CAS National Variants Settings - default	
Line Signals MFC/R2 Forward Signals MFC/R2 Backward Signals Options Timers	
Group I	
End of DNIS 🧖 I-15 💌	
End of ANI 🧐 I-15 💌	
ANI not Available 🧐 🛛 unused 💌	
Group II	
National Subscriber 🧐 II-1 💌	
National Priority Subscriber 🤎 II-2 🔽	
National Maintenance 🤍 II-3 💌	
National Operator 🌱 II-5 💌	
National Data 🤍 II-6 💌	
International Operator 9 II-10	
International Data 🧐 II-8 💌	
Save Cancel	
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#### **Group II**

Secondary forward signals are used to provide calling party category.

#### **National Subscriber**

Calling party is subscriber without priority.

# National Priority Subscriber

Calling party is subscriber with priority.

#### National Maintenance

Calling party is maintenance equipment.

#### National Operator Calling party is operator.

National Data Calling party is data transmission equipment.

#### **National PayPhone**

Calling party is pay-phone. International Subscriber Calling party is international subscriber without priority.

#### **International Priority Subscriber**

Calling party is international subscriber with priority.

#### **International Operator**

Calling party is international operator.

# **International Data**

Calling party is international data transmission equipment.

sers	Trunks Traffic Globals System Diagnostics Specials	ù
National	Variants	
nal \	√ariants Settings - default	
	Line Signals MFC/R2 Forward Signals MFC/R2 Backward Signals Options Timers	
	г Group А	
	Send Next DNIS Digit 🧐 🗛-1 💌	
	Send Previous DNIS Digit 🥙 🛛 🗛 - 2 🛛 🔽	
	Address Completed, Switch to Group B 🧐 🛛 🗛 💌	
	Network Congestion 🧼 🗛 💌	
	Send Calling Party Category 🎱 🛛 🗛 - 5 🛛 💌	
	Immediate Accept 🧖 🗛 - 6 💌	
	Send Second-to-last DNIS Digit (N-2) 🤗 🛛 🔽	
	Send Third-from-last DNIS Digit (N-3)	
	Repeat All DNIS Digits 🤎 unused 💌	
	Send Next ANI Digit 🤍 A-5 💌	
	Send Calling Party Category and Switch to Group C 🤝 Unused 💌	
	Group B	
	User Busy 🤗 R-3	
	Network Congestion ③ B-4	
	Unassigned Number 🧐 B-5 🔍	
	Line Free Charge 🤗 🛛 💌	
	Line Free No Charge ③ B-7 💌	
	Line Out Of Order 🛞 🖪 💌	
	Line Free Charge Alternative 1 🎱 🛛 unused 💌	
	Line Free Charge Alternative 2 🎱 🛛 unused 💌	
	Group C	
	Send Next ANI Digit 🧐 unused 💌	
	Repeat all DNIS digits and switch to Group A 🥬 unused 💌	
	Send next DNIS digit and switch to Group A 🤍 unused 💌	
	Send previous DNIS digit and switch to Group A 🖤 unused 💌	
	Address Completed, Switch to Group B 🧇 Unused 💌	
	Paula Canadi	

# **Group A**

Primary backward signals are used especially to collect DNIS and ANI digits from the calling party.

Send Next DNIS Digit Request to send next DNIS digit.

Send Previous DNIS Digit Request to repeat last DNIS digit.

#### Address Completed, Switch to Group B

Indication of the complete DNIS number and switch over to reception of group B signals.

**Network Congestion** Call failure, network congestion indication.

**Send Calling Party Category** Request to send calling party category.

#### **Immediate Accept**

Indication of the complete DNIS number and immediate call acceptation.

Send Second-to-last DNIS Digit (N-2) Request to repeat second-to-last DNIS digit.

Send Third-from-last DNIS Digit (N-3) Request to repeat third-from-last DNIS digit.

**Repeat All DNIS Digits** Request to repeat complete DNIS number form the beginning – send the first DNIS digit again.

#### Send Next ANI Digit

Request to send next ANI digit.

**Send Calling Party Category and Switch to Group C** Request to send calling party category and switch over to reception of group C signals.

#### **Group B**

Secondary backward signals are use for indication of status of called party.

Send Special Information Tone Call failure, send special information progress tone.

**User Busy** Call failure, destination user/line is busy.

Network Congestion Call failure, network congestion.

**Unassigned Number** Call failure, called number does not exist.

Line Free Charge Call accepted.

Line Free No Charge Call accepted.

Line Out Of Order Call failure, destination line is out of order.

Line Free Charge Alternative 1, 2 Call accepted. Assign this if there are available more signals for call acceptation.

#### **Group C**

Group C backward signals are used to collect called identification address (ANI number) from the calling party. Group C signals are not in common use, only some countries support them.

Send Next ANI Digit Request to send next ANI digit.

#### Repeat All DNIS Digits and Switch to Group A

Request to repeat complete DNIS number form the beginning and switch over to reception of group A signals.

#### Send Next DNIS Digit and Switch to Group A

Request to send next DNIS digit and switch over to reception of group A signals.

#### Send Previous DNIS Digit and Switch to Group A

Request to repeat last DNIS digit and switch over to reception of group A signals.

#### Address Completed, Switch to Group B

Indication of the complete DNIS number and switch over to reception of group B signals.

#### **Network Congestion**

Call failure, network congestion indication.

**Options** category defines special parameters which can be use to tune the behavior of CAS signaling protocol.

#### **Special Parameters**

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	
vi	wteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CAS National Variants	
CAS National Variants Settings - default	
Line Signals MFC/R2 Forward Signals MFC/R2 Backward Signals Options Timers	
r Special Parameters	
Switch to Group B on DNIS Timeout 🕐 🔲	
ANI Repeat 🧐 🔲	
Ignore Invalid Line Forward Signal During MFC/R2 🧼 🔽	
Ignore Invalid Line Forward Signal During Ringing 🎯 🔽	
Ignore Invalid Line Backward Signal During MFC/R2 🍼 📃	
Ignore Invalid Line Backward Signal During Ringing 🖤 📋	
Switch to Group A on End of ANI During Group C 🖤 📋	
Billing Pulse aller Answer 🧇 📘	
Save Cancel	
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#### Switch to Group B on DNIS Timeout

Enable this option to use expiration of compelled MFC/R2 cycle (timer T3) for DNIS digit request as indication of end of DNIS number. Call set-up goes on with reception of group B signals in this case.

#### **ANI Repeat**

Enable repetition of complete ANI number from the beginning when it is requested after previous end of ANI indication.

#### Ignore Invalid Line Forward Signal during MFC/R2

#### Ignore Invalid Line Backward Signal during MFC/R2

If enabled then invalid (unexpected) line forward/backward signals during MFC/R2 register signaling phase are ignored.

#### Ignore Invalid Line Forward Signal during Ringing Ignore Invalid Line Backward Signal during Ringing

When enabled then invalid (unexpected) line forward/backward signals during ringing phase (between end of MFC/R2 register signaling phase and the call answer) are ignored.

#### Switch to Group A on End of ANI during Group C

Enable switch over to reception of group A signals when end of ANI number indication is sent during reception of group C signals.

#### **Billing Pulse after Answer**

Enable generation of one billing pulse automatically after call answer. Some older exchanges can require this option for correct operation.

Timers category define various protocol timing settings. All timers are configured in milliseconds.

#### **Timers**

GUI-version: 1.1 / Board Type: 1 Voice&data router	Logout
	noteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Globals > CAS National Variants	
Line Signals MFC/R2 Forward Signals MFC/R2 Backward Signals Options Timers	
Timers	
Seizure ACK Timeout 🦻 2000	
Clear-Back Persistance Timeout 🧐 400	
Forced Release Persistance Timeout 🧶 400	
Clear-Forward Persistance Timeout 🧐 100	
Billing Pulse On-time 🧐 95	
Billing Pulse Off-time 🥙 600	
Max.Forward MFC/R2 Signal On-time (T1) 🧐 5000	
Max.Forward MFC/R2 Signal Off-time (T2)  5000	
Inbound Side Compelled Cycle Timer (T3)	
MFC/R2 Backward Non-compelled Pulse Length 🧆 150	
Delay Before Backward Non-compelled MFC/R2 Pulse 🎐 300	
Wait for Group B Signal Timeout 🧐 15000	
Convertent @ 2010-2012 All Rights Reserved www.insteacka.ek	

#### Seizure ACK Timeout

Maximal time of waiting for seizure acknowledge line signal.

#### **Clear-Back Persistence Timeout**

Minimal time to evaluate clear-back line signal as valid signal.

#### **Forced Release Persistence Timeout**

Minimal time to evaluate forced release line signal as valid signal.

#### **Clear-Forward Persistence Timeout**

Minimal time to evaluate clear-forward line signal as valid signal.

#### **Billing Pulse On-time**

Length of billing pulse.

#### **Billing Pulse Off-time**

Minimal length of delay between two subsequent billing pulses.

#### Max.Forward MFC/R2 Signal On-time (T1)

Maximum time a forward MFC/R2 signal can be on, from the outbound perspective.

#### Max.Forward MFC/R2 Signal Off-time (T2)

Maximum time a forward MFC/R2 signal can be off, from the outbound perspective.

#### Inbound Side Compelled Cycle Timer (T3)

Maximum time a whole compelled MFC/R2 cycle can take, from the inbound perspective.

#### MFC/R2 Backward Non-compelled Pulse Length

Duration of non-compelled MFC/R2 signaling pulse in backward direction. Some countries require such pulse, even though the break the compelled signaling scheme, and introduce timeout delays.

#### Delay Before Backward Non-compelled MFC/R2 Pulse

Minimum delay between end of compelled MFC/R2 signaling cycle and non-compelled pulse from destination side.

#### Wait for Group B Signal Timeout

Maximal time of waiting for MFC/R2 Group B signal after provision of the complete addressing information – call acceptance/failure indication.

# SYSTEM MENU

# IP settings

IP settings provide basic configuration of the Ethernet interfaces LAN and WAN (IP address assignment, MAC address modification, DNS setting, etc). Voice&Data Router is supplied with the following default setting:

	LAN	WAN
Mode	static	static
IP Address	192.168.1.100	10.1.1.100
Mask	255.255.255.0	255.0.0.0
Gateway		10.1.1.1
MAC Address	default	default

GUI-version : 1.1 / Board Type : 1

voiceo	data router	
		inoteska
Home Users Trunks	Traffic Globals System Diagnostics Specials	
IP Settings		
iii oottiiigo		
	LAN eth0	
	Mode: static 💌	
	IP Address: 192.168.1.170	
	Mask: 255.255.25.0	
	Gateway 192.168.1.123	
	MAC Address 🗹 Default	
	WAN eth2	
	Mode: static 💌	
	IP Address: 10.1.1.170	
	Mask: 255.0.0.0	
	Gateway	
	MAC Address V Default	
	DVC	
	Server åddress 192 168 1 210	
	Server Address 8.8.8.8	
	HOSTNAME TEST	
	Save & Apply Cancel	
	Dave & Apply   Cancer	
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#### Mode

Select **static** option if you want to configure IP address manually by your own. Select **dhcp** option to obtain IP address automatically from the network.

#### **IP Address**

IP address configuration (static setting).

# Mask

Subnet mask configuration (static setting).

# Gateway

Default gateway (static setting). Note that default gateway can be only one – it should be configured only for one Ethernet interface LAN or WAN.

# **MAC Address**

Check **Default** if you want to use Ethernet hardware address assigned by producer. If you need to overwrite hardware Ethernet address for some reason to suit your preference then uncheck **Default** and enter new address as colon separated hexadecimal numbers (e.g. 00:50:C2:38:76:80). Modified MAC address will be applied only after reboot of Voice&Data Router.

# DNS

Enter IP address of Domain Name Server (primary and secondary).

#### HOSTNAME

Enter Voice&Data Router hostname.

Network configuration changes apply immediately after clicking on the **Save & Apply** button. There is no need to activate standard Apply Changes function.

When you change IP address of the interface you use to communicate with Voice&Data Router then Web Manager automatically tries to switch to the new address (it works only for static setting of IP address).

# **Routing Tables**

Routing tables define rules for static IP routing. In most cases it is enough to configure basic IP settings and no additional rules for IP routing are necessary. Here you can configure additional gateways for routing to the target address (host or network).

GUI-version : 1.1 / Board Type :	1			_	Logout
voice	dal	ta roi	Jter		
					inoteska
Home Users Trunks	Traffic	Globals System	Diagnostics Speci	als	
System > Routing Tables					
Routing Tables Setting	js				
		Create New	Modify Selected	Delete Selected	
List of Routes					
	19:	2.168.2.0	255.255.255.0	10.1.1.1	
	19:	2.168.10.0	255.255.255.0	10.1.1.1	
			Save & Apply	Cancel	
		Copyright @ 2010	-2012, All Rights Rese	erved <u>www.inoteska.sk</u>	

Use the **Create New** button to add new route to the routing table. Check existing route in the table and click on the **Modify Selected** to change route parameters. You can remove selected (checked) routing items when you click on the **Delete Selected** button.

New common form is used for creating and editing item of the routing, where you can enter specific route parameters.

#### **IP Address**

Enter IP address of the destination network or host.

#### Mask

Enter network mask to be used for the network destination. When the destination is host, use value 255.255.255.255 for network mask or leave this field empty.

#### Gateway

Enter address of the gateway to use for routing. The specified gateway must be reachable first.

GUI-version: 1.1 / Board Type: 1 Voice Solata router	Logout
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials System > Routing Tables	
Add or Edit Route	
IP address: [192.168.10.0]	
Mask: 255.255.0	
Gateway: 10.1.1.1	
Save	
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Routing tables changes apply immediately after clicking on the **Save & Apply** button. There is no need to activate standard Apply Changes function.

# **SNMP**

SNMP (Simple Network Management Protocol) is an application layer protocol used in network management systems to monitor network-attached devices for conditions that warrant administrative attention. On Voice&Data Router, SNMP agent is running to allow administrators to remotely manage device status and configuration. Voice&Data Router supports SNMP version v1/v2c.

SNMP protocol setting is divided into two parts

- General Settings
- Trap Settings

**General Settings** are used to enable the SNMP agent and to define system options required for its remote administration.

GUI-ver	sion : 1.8 / Board Type : 1	Logout
1	pice Suclata reuter	
V	Dice Qually router	
		INDIESKa
Home System:	Users Trunks Traffic Globals System Diagnostics	
SNMF	> Settings	General Settings Trap Settings
Gene	eral Settings	
	C Enable SNMP Agent	
	System Name 🦈 ROUTER166	
	System Description 🥐 Voice&Data Router	
	System Location 🧇 Inoteska LH	
	System Contact 🦈 Stefan Lakatos	
	Read-Only Community 🦈 public	
	Save	
	Copyright © 2010-2016, All Rights Reserved <u>www.inoteska.sk</u>	

#### Enable SNMP Agent

Enable SNMP agent for remote system monitoring.

#### System Name

System name for identification of Voice&Data Router.

#### **System Description**

Detailed information about managed system.

#### System Location

Information to describe the network where SNMP management is performed.

#### System Contact

Information about the contact person responsible for the SNMP management in the defined network. Field may indicate the point person's name, email address, phone number or other contact information.

#### **Read-Only Community**

SNMP v1/v2c read-only community description (public, private, etc.) for the read-only management. It may contain some kind of password which should be matching both on the device and on the administrating application for successful SNMP management.

**Trap Settings** are used to define the traphosts that should be informed when certain events occur on the Voice&Data Router.

VOICE&DATA ROUTER	
GUI-version : 1.8 / Board Type : 1	Logout
voice&data router	inoteska
Home Users Trunks Traffic Globals System Diagnostics	
System > SNMP	
SNMP Settings	eneral Settings Trap Settings
Trap Settings	
Traphost Community	Add
192.168.1.14 public	Edit Delete

Use the **Add** button to define new SNMP traphost. Click **Edit** button to modify selected SNMP traphost. To remove existing traphost use the **Delete** button.

GUI-version : 1.8 / Board Type : 1	Logout
vaire telete	
VOICE QUALA router	
ü	wteska
Home Users Trunks Traffic Globals System Diagnostics	
System > SNMP	
SNMP Settings	
Edit Traphost	
Traphost 🧼 192.168.1.14	
Community 🧼 public	
Save Cancel	
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#### Traphost

IP address or the host name of the traphost. Administrating application's host address should be inserted here.

#### Community

Trap community description (public, private, etc.) for the administrating application to accept the notifications about the certain events on the device. Field may contain some kind of password which should be the same both on the device and on the administrating application for successful SNMP management.

# Firewall

Voice&Data Router firewall function can protect LAN network and Voice&Data Router itself from unauthorized users of other networks. It is recommended to activate firewall if Voice&Data Router is used in function of access data router or if it is connected directly to the Internet on public address.

Firewall implementation in Voice&Data Router allows:

- Control incoming traffic on both LAN and WAN interfaces
- Control packet forwarding between LAN and WAN interfaces
- Network address translation NAT

GUI-version : 1.2 / Board Type :		iter	Logout
VOICOU	Ocicentot		inoteska
Home Users Trunks	Traffic Globals System	Diagnostics	Specials
System > Firewall			
Firewall Settings			Edit Input Firewall Edit Forward Firewall Edit Destination NAT
General Settings			
	Enable Input Firewall	۹ 📀	Global Policy 🌮 DROP 💌
	Enable Forward Firewall	۷ 📀	Global Policy 🌮 DROP 💌
	Enable Source NAT	الم	
		Save & Apply	Cancel
	Copyright © 2010-	-2012, All Rights	Reserved <u>www.inoteska.sk</u>

#### **Enable Input Firewall**

Enable/disable input firewall functionality. Input firewall controls incoming traffic to Voice&Data Router on both Ethernet interfaces (LAN and WAN). You can select global policy which defines default behavior when packet does not match any input firewall rule:

- ACCEPT let the packet through
- **DROP** drop the packet on the floor

Use the Edit Input Firewall button for configuration of firewall rules to filter incoming packets.

#### **Enable Forward Firewall**

Enable/disable forward firewall functionally. It allows you to control packet forwarding between LAN and WAN interfaces. You can select global policy which defines default behavior when packet does not match any firewall rule:

- ACCEPT let the packet through
- DROP drop the packet on the floor

Use the Edit Forward Firewall button to configure rules for the forward firewall.

#### **Enable Source NAT**

Enable/disable NAT (Network Address Translation) functionality. NAT translates source private local network address to WAN interface (public) address. It allows users from LAN network to connect to the external network (such as the Internet) using WAN IP address. Use the **Edit Destination NAT** button to configure rules for destination NAT, which provides translation of the destination address from WAN to the private local network address.

Input firewall configuration consists of set of the rules to filter incoming packets to the Voice&Data Router. Input firewall rules apply only to packets addressed to Voice&Data Router not to forwarded packets. Click on the **Add New** button if you want to create new input rule. Use the **Edit** button to modify existing rule. Use the **Delete** button to remove selected rule and the **Delete All** button to remove all the rules from the list. Note that the rules are evaluated in the order they are configured and that is why the rules order is important for correct functionality of the firewall. You can control the order of the rules by using arrows in the first column of the table.

VOICE&DATA ROU	TER				
GUI-version : 1.2 / Board Type : *	1				Logout
voice	data	router			inoteska
Home Users Trunks	Traffic Globals	System Diagnostics Sp	pecials		
System > Firewall					
Firewall Settings					
Input Firewall Details					
Order Protocol	Policy Interfac	e Src Address	Src Port	Dst Port	Add New Delete All
🔸 top	ACCEPT wan	195.168.205.88		22	Edit Delete
🕇 tcp	DROP any			22	Edit Delete
	Сари	Back	eserved www.inc	oteska.sk	

Input firewall rules define condition to match incoming packet and the policy. Input packets which satisfy the rule condition are accepted/dropped according rule policy. If packet does not match condition of any rule then global policy is applied.

#### Protocol

Select the IP transport protocol of incoming packet to apply the rule. The following protocol options are available: all, TCP, UDP, ICMP.

#### Policy

Select the policy for input packet matching rule condition. Policy specifies if the packet is allowed or not. You can select following values

- ACCEPT let the packet through
- **DROP** through the packet away

#### Interface

Select Ethernet interface of incoming packets. It allows firewall to filter packets coming only from certain interface.

GUI-version : 1.2 / Board Type : 1	Logout
voice&data router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials System > Firewall	
Firewall Settings	
Input Firewall Rule	
Protocol 🖗 tcp 💌 Policy 🇐 ACCEPT 💌 Interface 🖗 wan 💌	
Src Address        195.168.205.88       /          Src Port        to           Dst Port        22       to	
Save Record Cancel	
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#### Src Address

Enter packet source address. Source address condition can restrict/allow packets from certain IP addresses. You can specify single IP address or network IP address with mask. The mask can be either a network mask or a plain number, specifying the number of 1's at the left side of the network mask. Thus, a mask of 24 is equivalent to 255.255.255.0.

#### Src Port

Enter packet source port or port range. Source port condition is allowed only for TCP and UDP protocol and it filters input packets according source port numbers.

#### Dst Port

Enter packet destination port or port range. Destination port condition is allowed only for TCP and UDP protocol and it can filter input packets by destination port numbers.

Forward firewall configuration is the list of the rules to filter packet forwarding between LAN and WAN interfaces. Click on the **Add New** button if you want to create new forward rule. Use the **Edit** button to modify existing rule. Use the **Delete** button to remove selected rule and the **Delete All** button to remove all the rules from the list. Note that the rules are evaluated in the order they are configured and that is why the rules order is important for correct functionality of the firewall. You can change the rules' order by arrows in the first column of the table.

GUI-versio	on : 1.2 / Boa	rd Type : 1							Logout
vo	ice	200	Jata	) rou	ter				
									inoteska
Home	Users	Trunks 1	Fraffic Globals	System D	iagnostics	Specials			
System > F	rewall								
Firewal	Setting	S							
Forwar	d Firewall	Details							
Order	Protocol	Policy	Src Interface	Src Address	Src Port	Dst Interface	Dst Address	Dst Port	Add New Delete All
	all	ACCEPT	lan	192.168.1.0/24		wan			Edit Delete
					Ba	ck			
				opyright© 2010-20	12, All Right	ts Reserved <u>www</u>	<u>.inoteska.sk</u>		

Forward firewall rules consist of filtering condition and the policy. Forward rules apply to packets coming from local LAN network to WAN network and on the contrary from WAN to LAN. Packets which satisfy the rule condition are accepted or dropped according policy of the rule. If packet does not match condition of any rule then global policy is applied.

#### Protocol

Select the IP transport protocol of filtered packets to apply the rule. The following protocol options are available: all, TCP, UDP, ICMP.

#### Policy

Select the policy for packet matching rule condition. Policy specifies if the packet is allowed or not. Following values are available

- ACCEPT let the packet through
- DROP through the packet away

#### **Src Interface**

Select Ethernet interface via which a packet is received. Together with setting of destination interface we can define condition for packet direction (from LAN to WAN or vice-versa).

#### **Dst Interface**

Select Ethernet interface via which a packet is going to be sent. Together with source interface setting we can define condition for packet direction (from LAN to WAN or vice-versa).

#### Src Address

Enter packet source address. Source address condition can restrict/allow packets from certain IP addresses. You can specify single IP address or network IP address with mask. The mask can be

either a network mask or a plain number, specifying the number of 1's at the left side of the network mask. Thus, a mask of 24 is equivalent to 255.255.255.0.

#### **Dst Address**

Enter packet destination address. Destination address condition can restrict/allow packets addressed to specific IP addresses. You can write single IP address or network IP address with mask. The mask can be either a network mask or a plain number, specifying the number of 1's at the left side of the network mask. Thus, a mask of 24 is equivalent to 255.255.255.0.

GUI-version: 1.2 / Board Type: 1	gout
inotes	ika
Home Users Trunks Traffic Globals System Diagnostics Specials System > Firewall	
Firewall Settings	
Forward Firewall Rule	
Protocol 🌮 tep 💌 Policy 🦈 ACCEPT 💌	
Src Interface        Ian        Dst Interface        wan          Src Address        192.168.1.0       1        Dst Address        1        1          Src Port        to        Dst Port        22       to        1	
Save Record Cancel	
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#### Src Port

Enter packet source port or port range. Source port condition is allowed only for TCP and UDP protocol and it filters packets according source port numbers.

#### **Dst Port**

Enter packet destination port or port range. Destination port condition is allowed only for TCP and UDP protocol and it can filter packets by destination port numbers.

GUI-V		1.2 / Board Typ		n route	er		Logout inoteska
Hon	ne U	Jsers Trunk:	s Traffic Globals	System Diagn	ostics Specials		
Fire	wall S	settings					
De	stinatio	on NAT Deta	ils				
C	order	Protocol	Src Address	WAN Port	LAN Address	LAN Port	Add New Delete All
	+	tcp	188.133.99.137	2222	192.168.1.14	22	Edit Delete
1	+ +	udp		20000:20999	192.168.1.48	20000:20999	Edit Delete
1	++	tcp		88	192.168.1.170	80	Edit Delete
1	-	tcp		89	192.168.1.171	80	Edit Delete
					Back		
			Cc	opyright©2010-2012,	All Rights Reserved <u>ww</u>	<u>w.inoteska.sk</u>	

Destination NAT allows remote users from WAN network (for example, computers on the Internet) to connect to a specific computer or service within our LAN network. This technique is often called port

forwarding or port mapping. You can define set the rules for LAN destination address translation. Click on the **Add New** button if you want to create new destination NAT rule. Use the **Edit** button to modify existing rule. Use the **Delete** button to remove selected rule and the **Delete All** button to remove all the rules from the list. Note that the rules are evaluated in the order they are configured and that is why the rules order is important for correct functionality of the firewall. You can change the rules' order by arrows in the first column of the table.

Destination NAT rules consist of filtering condition and definition of LAN destination. Rules apply to packets coming from external WAN network and if the rule condition is matching packet is redirected to specified LAN destination address.

GUI-version : 1.2 / Board Type : 1	Logout
Voice 200to router	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
System > Firewall	
Destination NAT Rule	
Protocol 🥙 tcp 💌	
Src Address 🧐 👘 👘	
WAN Port 🧐 89 to 🛞	
LAN Address 192.168.1.171 LAN Port 80 to 80	
Save Record Cancel	
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#### Protocol

Select the IP transport protocol of filtered packets to apply the rule. The following protocol options are available: TCP, UDP, ICMP.

#### Src Address

Enter packet source address. Source address condition can limit access only for users from certain IP addresses. You can specify single IP address or network IP address with mask. The mask can be either a network mask or a plain number, specifying the number of 1's at the left side of the network mask. Thus, a mask of 24 is equivalent to 255.255.255.0.

#### WAN Port

Enter destination port or port range of received packet. WAN port condition is allowed only for TCP and UDP protocol and it filters packets according destination port numbers.

#### LAN Address

Enter new destination IP address. Packets matching rule condition are redirected to this address.

#### LAN Port

Enter new destination port or port range. Destination port can be specified only for TCP or UDP protocol.

Firewall setting applies immediately after clicking on the **Save & Apply** button on the main screen of firewall setting. There is no need to activate standard Apply Changes function.

# **Options**

Use **Change password** to change password of active user (admin) for access to Voice&Data Router Web Manager. Password should be at least 4 characters long. For password changing type your new password in the **Enter New Password** text field and confirm it in the **Retype New Password** field. New password is activated after click on the **Update** button and you are automatically redirected to the initial login screen to re-login with new authentication.

GUI-version : 1.1 / Board Type : 1 Password should be at least 4 characters ! Logout
verified to be in
VOICe QUALA router
inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials
System > Options
Change Password
Change Password Reset Configuration Reboot
Enter New Password
Retype New Password
Update
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**Reset Configuration** restores default factory configuration. Default factory setting is created and loaded into the device during production and it is the basic setting the device is supplied with. If the **Keep Network Settings** checkbox is enabled network configuration is not changed and you don't need to change the address in your web browser to re-connect to Voice&Data Router. Use **Reset to Defaults** button to restore factory default configuration.

To avoid loosing of actual configuration <u>Backup and restore</u> function can be used before reset to defaults. Voice&Data Router will restart automatically after applying reset to the default setting.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	
	inoteska
Home Users Trunks Traffic Globals System Diagnostics Specials	
System > Options	
Reset to Factory Defaults	
Change Password Reset Configuration Reboot	
Warning: By resetting your Voice&Data Router to factory defaults, you will lose all your configuration ! You can take a backup of your current configuration from the <u>Backup page.</u>	
Reset to Defaults	
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

**Reboot** allows Voice&Data Router restart. System reboot usually takes about 1 minute but it can vary. Web Manager automatically re-connects to Voice&Data Router after minute and half. You can also use browser reload function to try to re-connect before this timeout.

VOICE&DATA ROUTER	
GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	noteska
Home Users Trunks Traffic Globals System Diagnostics Specials System S Ortions	
Reboot Appliance	
Change Password Reset Configuration Reboot	
Warning: Rebooting the appliance will terminate all active calls.	
Reboot Now	
Copyright © 2010-2012, All Rights Reserved www.inoteska.sk	

# Backup and Restore

Voice&Data Router allows you to manage configuration data backups. You can create and store backup of actual Voice&Data Router configuration. It is possible to return back to the stored configurations at any moment in the future. You can also download configuration backups to archive them out of the device.

To backup actual configuration of Voice&Data Router click on the **Create New Backup** button. New backup file is stored in the file system. You can use **Restore** button to restore device configuration from selected backup file. System will restart automatically after applying of configuration restoring. Note that this operation can change Voice&Data Router network setting and it may be necessary to enter new IP address to your web browser to recover the connection after device restart. Existing configuration backup can be removed by using of **Delete** button.

GUI-version :	1.8/Board Type: 1	router	Logout inoteska
Home U:	sers Trunks Traffic Globals	System Diagnostics	
System > Back	up and Restore		
Backup /	Restore Configurations		Create New Backup Upload Backup File
List of Co	nfiguration Backups		
#	Name	Date	
1	backup1_TEST_UK0_121631	Jan 26, 2016	Download Restore Delete
2	backup1_TEST_UK0_153918	Jan 25, 2016	Download Restore Delete

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If you want to download configuration backup to your PC, click on the **Download** button. Direct link to the backup file appears and can be used to download.

Download File	
Right Click on the above link and download using the 'Save Link As' option	

If you want to upload configuration backup from your PC to Voice&Data Router use **Upload Backup File** button. You need to choose file to upload and click on the **Submit** button. Only files previously downloaded from the device can be successfully uploaded and restored.

GUI-version: 1.1 / Board Type: 1	gout
voice a data router	
	ha
Home Users Trunks Traffic Globals System Diagnostics Specials	
System > Backup and Restore	
Upload a previous backup File List of Backups Upload Backup	File
File Uploading Upload Upload Choose File backup1_201eb23.tar.gz backup1_2012feb23_091524_2012feb23.tar.gz (application/gzip) - 314830 bytes Submit	
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

close

# Firmware Update

Voice&Data Router supports three possibilities how to update firmware

- TFTP protocol download
- HTTP protocol download
- File upload

#### **Use TFTP Server**

Check this option to use TFTP protocol to download firmware update file. Enter TFTP URL address – server address with path to the update file, into the input field and click on **Update Firmware** button. Voice&Data Router downloads specified file and starts update process. Firmware is checked for validity after download and the warning is displayed if it does not suit.

#### **Use HTTP Server**

Check this option to use HTTP protocol to download firmware update file. Enter HTTP URL address – server address with path to the update file, into the input field and click on **Update Firmware** button. Voice&Data Router downloads specified file and launches update process. Firmware is checked for validity after download and the warning is displayed if it does not suit.

#### **Use File**

Check this option to upload firmware update file from your PC. Use the **Choose File** button to select file for upload and then click on the **Upload file** to transfer selected file to Voice&Data Router. Click on the **Update Firmware** button to start update process. Firmware is checked for validity before update is executed and the warning is displayed if it does not suit.

GUI-version : 1.1 / Board Type : 1		Logout
voicesdo	to router	
		inoteska
Home Users Trunks Traffic	Globals System Diagnostics Specials	
System > Firmware Update		
Firmware Update		
Firmware Source		
	┌ 🗹 Use TFTP Server	
- A A A A A A A A A A A A A A A A A A A	tftp://192.168.1.44/test_updt/update.tar.gz	
	Use HTTP Server	
	http://	
	Choose File No file chosen	
	Update Firmware	
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System will automatically restart after firmware update.

# Date and Time

Voice&Data Router system time adjustment.

#### **Date and Time Setting**

#### Date, Time

Current date and time.

#### **Get Device Time**

Click to read current system time from Voice&Data Router and refresh displayed date and time.

#### **Get Computer Time**

Click to get current system time of your PC and refresh displayed date and time (can be used to synchronize Voice&Data Router with your PC).

#### Set Time

Adjust system time of Voice&Data Router. Values in **Date** and **Time** fields are used. Date and time setting applies immediately; there is no need to activate standard Apply Changes function.

GUI-version : 1.1 / Board Type : 1	Logout
voice&data router	
	Juesha
Nome Users Franks Frank Globals System Diagnostics Specials	
Date and Time Settings	
Date and Time Setting Date (DD MM YYYY) ③ 09 ④ 05 ④ 2012 Time (HH MM SS) ④ 14 ④ 39 ④ 36 Get Device Time Get Computer Time Set Time	
Time Zone Setting Time zone 🧇 (GMT+01:00) Belgrade, Bratislava, Budapest, Ljubljana, Prague 💌 🗹 🎱 Day Light Saving	
NTP Server Setting NTP enabled Server Address  pool.ntp.org Update Interval  Daily	
Save & Apply Cancel	
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#### **Time Zone Setting**

Time Zone

Select time zone of your country.

#### **Day Light Saving**

Enable daylight saving time (summer time).

#### **NTP Server Setting**

#### **NTP enabled**

Enable automatic clock synchronization with NTP (Network Time Protocol) server.

#### Server Address

Enter NTP server address (IP address or host name).

# Update interval

Select interval of automatic clock synchronization.

- Hourly every hour
- Daily every day at 2:25 am
- Weekly every Sunday at 2:47 am

Configuration changes apply immediately after clicking on the **Save & Apply** button. There is no need to activate standard Apply Changes function.

# **DIAGNOSTICS MENU**

# Port Status

Port status displays actual status of selected ports. Actual status is read from Voice&Data Router after pressing **Display Status** button. If **Enable Auto Updating** is checked status is updated automatically circa every 3 seconds.

You can display status of the following ports/channels:

- E1/T1 actual status of E1/T1 ports
- BRI actual status of BRI ports
- GSM actual status of GSM modules (optional)
- Analog –actual status of analog ports (optional)
- SIP actual status of SIP trunks and users
- Ethernet actual status of Ethernet ports

GUI-version : 1.2 / Board Type : 1		Logout
voice&data router		in the star
		livoteska
Home Users Trunks Traffic Globals System Diagnostics Specials		
Diagnostics > Port Status		
Port Status	Enable Auto Updating 🛛 E1/T1 🛛 💌	Display Status
E1/T1 Status		
<pre>FALC-framer[0x0] STS = 0x00000000 decoding: FEC = 0x0000004a Framing Error Counter CVC = 0x000000b CCC Error Counter SLI = 0x0000000b CRC Error Counter LOS = 0x00000004 Slip Error Counter LOS = 0x00000001d Loss of signal AIS = 0x00000000 Alarm Indication LFA = 0x0000000 Closs of frame RRA = 0x00000006 Receive remote alarm</pre>		
FALC-framer[0x1] STS = 0x00000008 decoding: LOS LFA DLL FEC = 0x00000007 Framing Error Counter CVC = 0x0000001 CRC Error Counter SLI = 0x00000001 Slip Error Counter LOS = 0x00000007 Loss of signal AIS = 0x00000000 Alarm Indication LFA = 0x00000004 Loss of frame RRA = 0x00000001 Receive remote alarm		
<pre>FALC-framer[0x2] STS = 0x000000a8 decoding: LOS LFA DLL FEC = 0x00000001 Framing Error Counter CVC = 0x00000003 Code Violation Counter CEC = 0x00000000 CRC Error Counter SLI = 0x00000001 Slip Error Counter LOS = 0x00000001 Slip Error Counter LOS = 0x00000001 Sloss of signal AIS = 0x00000000 Alarm Indication LFA = 0x00000001 Loss of frame RRA = 0x00000001 Receive remote alarm</pre>		
<pre>FALC-framer[0x3] STS = 0x00000001 decoding: OFF FEC = 0x00000000 Framing Error Counter CVC = 0x00000000 Code Violation Counter CEC = 0x00000000 CRC Error Counter SLI = 0x00000000 Slip Error Counter LOS = 0x00000000 Slip Error Counter LOS = 0x00000000 Loss of signal AIS = 0x00000000 Alarm Indication LFA = 0x00000000 Loss of frame RRA = 0x00000000 Receive remote alarm</pre>		

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# Active Services

Use active services function to display actual status of users' services. It displays list of all users with active services like Do Not Disturb and Call Forwarding.

You can use **Delete User** and **Delete All** buttons to deactivate services for selected user eventually for all users.

GUI-version : 1.2 / Bo	ard Type : 1	Logout
voice	eadata router	
		inoteska
Home Users	Trunks Traffic Globals System Diagnostics Specials	
Diagnostics > Active Se	rvices	
Active Services	5	Delete User Delete All
Do Not Disturb		
User	Status	
10	on	
Call Forwarding (U	nrestricted)	
User	Forward Number	
10 12	12365 0104155	
Call Forwarding if	Busy	
User	Forward Number	
Call Forwarding if	no Respond	
User	Forward Number	
	Copyright @ 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

# **Identification**

Identification provides basic hardware information about Voice&Data Router. BASIC section information is read from permanent memory which is programmed during device production. Here you can find information such as device type – product code number, serial number, date of production, hardware interfaces, etc. CONFIG and FUNCTIONS parts are stored in the file and they can be used to enable/disable optional interfaces and functionalities.

GUI-version : 1.2 / Board Type : 1	Logout
voice&data router	inateska
Home Users Trunks Traffic Globals System Diagnostics Specials	
Diagnostics > Identification	
Identification	
Details	
[BASIC]	
TYPE=TTX495012000220 SN=495010990102 ID=5490 DATE=21.1.2010 SERVP=F3DE1AF32994DA01EA9299293C6157B0 MAN=103,104 DPS=TTB 166 09 PROD=INOTESKA ADR=PODTUREN-ROVEN221,LIPTOVSKY HRADOK,03301 TE1=+421445221809,+42190350360 MAIL=NA1EUNOTESKA.SK WWW=WWW.INOTESKA.SK WWW=WWW.INOTESKA.SK HW=1000 RD=D SDRAM=128M FLASH=8M CARD=SD FPGA=ICDP1C12F324C8 CPU=M82810 CON=1 E1=4 ETT=2 TEST=	
[CONFIG]	
TYP=11/495012000220 SN=495010990102 ID=5490 USERS=108 USERB=108 E1=0,1,2,3 ETH=0,1	
[FUNCTIONS]	
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>	

# System Logs

Voice&Data Router stores operation logs in the file system. Voice&Data Router logs management is divided into two parts:

- Asterisk Log Files
- System Log Files

Asterisk Log Files cover log messages from the Asterisk voice communication system application. It is possible to enable/disable Asterisk logging, set logging level and display stored logs.

GUI-version : 1.8 / Board Type : 1					Logout
voiceod	ata r	outer			
					inoteska
Home Users Trunks Traff	ic Globals Syste	em Diagnostics			
Diagnostics > System Logs					
Voice&Data Router Log me	ssages			Asterisk Log Files	System Log Files
Asterisk Log Files	ning 💌 error 🔲 debug	verbose			
File	Size	Date			
event_log	1 kB	Jan 26 10:21			Show Reset
messages	11 kB	Jan 26 10:21			Show Reset
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There is **Disable Logger** or **Enable Logger** button to switch on and off Asterisk logging. When logging is enabled you can use check boxes (notice, warning, error, debug and verbose) to set logging level.

Recorded log files are listed in the table. You can use **Show** button to display content of select log file and you can clear selected log file with the **Reset** button.

System Log Files allows

GUI-version : 1.8 / Board Type : 1

# voice&data router

Diagnostics > System Logs Asterisk Log Files System Log Files Voice&Data Router Log messages System Log Files

File	Size	Date	
btmp	1 kB	Jan 25 16:24	Show Reset
syslog.log	81 kB	Jan 26 12:17	Show Reset
wtmp	23 kB	Jan 26 07:21	Show Reset

#### /var/log/syslog.log

	-
Jan 25 15:41:25 VD ROUTER syslogd 1.4.1#17: restart.	
Jan 25 15:41:25 VD_ROUTER kernel: klogd 1.4.1#17, log source = /proc/kmsg started.	
Jan 25 15:41:25 VD_ROUTER kernel: Inspecting /boot/System.map	
Jan 25 15:41:26 VD_ROUTER nrpe[888]: Starting up daemon	
Jan 25 15:41:26 VD ROUTER kernel: Loaded 22768 symbols from /boot/System.map.	
Jan 25 15:41:26 VD_ROUTER kernel: Symbols match kernel version 2.6.11.	
Jan 25 15:41:26 VD_ROUTER kernel: No module symbols loaded - kernel modules not enabled.	
Jan 25 15:41:26 VD_ROUTER kernel: Linux version 2.6.11.7-1.08.5.3malindi (root@debian-dev2) (gcc version 3.3.2 2003082	:0
Jan 25 15:41:26 VD_ROUTER kernel: CPU: ARM920Tid(wb) [41129200] revision 0 (ARMv4T)	
Jan 25 15:41:26 VD_ROUTER kernel: CPUO: D VIVT write-back cache	
Jan 25 15:41:26 VD_ROUTER kernel: CPUO: I cache: 16384 bytes, associativity 64, 32 byte lines, 8 sets	
Jan 25 15:41:26 VD_ROUTER kernel: CPUO: D cache: 16384 bytes, associativity 64, 32 byte lines, 8 sets	
Jan 25 15:41:26 VD_ROUTER kernel: Machine: ARM-M828xx Comcerto	
Jan 25 15:41:26 VD_ROUTER kernel: Memory policy: ECC disabled, Data cache writeback	
Jan 25 15:41:26 VD_ROUTER kernel: On node O totalpages: 28672	
Jan 25 15:41:26 VD_ROUTER kernel: DMA zone: 28672 pages, LIFO batch:7	
Jan 25 15:41:26 VD_ROUTER kernel: Normal zone: 0 pages, LIFO batch:1	
Jan 25 15:41:26 VD_ROUTER kernel: HighMem zone: 0 pages, LIF0 batch:1	
Jan 25 15:41:26 VD ROHTER kernel: Built 1 zonelists	-
	b.

Download /var/log/syslog.log Right Click on the above link and download using the 'Save Link As...' option

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inoteska
## **CLI Emulator**

CLI emulator provides you access to Command Line Interface of internal Asterisk. It allows you to execute CLI commands you enter in the **CLI Command** field.



Use command "help" to display all available CLI commands. To display brief command description you can use "help" plus complete command (e.g. "help core show channels").

## Debug messages

Debug messages can log GUI manager functionality. Use the **Start DEBUG** and **Stop DEBUG** buttons to control Voice&Data Router GUI manager debugging.



# System Status

System status shows actual status of Voice&Data Router. It involves actual system time and device running time, usage of hardware resources (CPU, memory, file system) and information about software version.

GUI-version : 1.8 / Board Type : 1		Logout
	ha i	
VOICeodo	ta router	
		inoteska
Home Users Trunks Traffic	Globals System Diagnostics	
Diagnostics > System Status		
System Status		GUI 1.8
System Time: 12:26:04	Uptime: 5:06	
	CPU Usage:	
	L%	
Total Active Processes: 63	97% of 108 MB	
Kernel: 2.6.11.7-1.08.5.3malindi	Root Filesystem Usage:	
Firmware: 2.6	7% of 3664 MB	
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## File Editor

File editor provide direct access to selected configuration files of Voice&Data Router. In this way you are able to configure nonstandard functions or functions not supported by Web Manager.

Files you can access are listed in the **Config Files** list on the right side. Selected file is displayed on the screen. File editor can handle only files divided into the contexts/sections. All the comments are filtered out – not displayed. The comments are all lines starting with semi-colon ';' and also multi-line blocks between comment markers ";--" and "--;".

GUI-version : 1.2 / Board Type : 1		Lo	igout
voice&data router			
		inotes	ika
Home Users Trunks Traffic Globals System Diagnostics Specials			
Diagnostics > File Editor			
File Editor - manager/network.conf	New File	manager/network.conf	*
Add Context			
+ [lan]			
- [wan]			
iface=eth2 mode=static ipaddress=10.1.1.170 netmask=255.0.0.0 gateway= macaddress=			
+ [ntp]			
+ [dns]			
Copyright © 2010-2012, All Rights Reserved <u>www.inoteska.sk</u>			

Clicking on the selected context it is switched to the edit mode. You can use **Add Context** button to create new file context.

Note that direct changing of configuration files is not recommended for people without detailed knowledge of the system. . Unqualified change of the configuration can make the device non-functional.

## **SPECIALS MENU**

# AOC Web Update

GUI-versio	n : 1.0 / Board Type : 1	Logout
vo	ice&data router	
		inoteska
Home	Users Trunks Traffic Globals System Diagnostics Specials	
AOC Up	date Service Settings	Check AOC File Now
Details		
	Domain Name or IP Address 🖤 🛛 aoc.sramko.eu	
	Working Directory Name 🖤	
	Periodicity 🧼 🗖 regularly every 👘 📝 hour	
	🔍 🗹 every day at 🛛 10 💌 🗄 30 💌 oʻclock	
	Repeat Count 🖤 😗 💌	
	Save	
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## **Domain Name or IP Address**

Name or address of web server where AOC files are saved

### Working Directory Name

Directory name on server where AOC files are saved (item can be blank if files are in server main directory)

#### Periodicity

Periodicity of AOC files correctness control, it is possible to choose either time interval (every 1, 4, 6 or 12 hours) or hour when control will be done

#### **Repeat Count**

Number of repeated files control (used when control is executed with error)

#### **Check AOC File Now**

Runs immediate control of AOC file. After successful completion, name of currently used AOC file is displayed.

# **UNIMAN**

UniMan is proprietary management application running on Windows platform. It can be used for the communication with various Inoteska equipments.

UniMan is not preferred management tool for Voice&Data Router and it does not offer complete device configuration. UniMan support in Voice&Data Router is limited only for network configuration and diagnostics.

# **CONNECTION TO THE DEVICE**

UniMan communicates with Voice&Data Router over Ethernet using TCP or UDP protocol. You can access the device directly connected to the PC or you can access the device co in your local network or you can also connect to the remote device over the Internet.

To open window for communication setting click on the speed button **Setting of communication** or use menu item **Communication/Setting of communication**.



# UDP Connection

UDP connection can be used to communicate with local devices. You do not need to know device IP address – UniMan is able to search for available Inoteska equipments.

This access type can be used only if the conditions stated below are met:

Device is connected in network

- Device and PC must be connected in the same local network
- Network must transmit broadcast
- PC must have IP address allocated
- Device must have default gateway configured
- Device must be attached by default Ethernet interface

Device is connected to PC locally

- PC must have arbitrary IP address allocated (it is necessary to disable DHCP and set static IP address, e.g. 192.168.1.2)
- Receive/Transmit of *broadcast* packets must be enabled on PC

- UDP port 3864 must be enabled on PC
- Device must have default gateway configured
- Device must be attached by default Ethernet interface

Restring of communication	on 🛛 🔀
	Save Delete
Manager address	240-0-0-0
Device address	0-0-0-0
Latency	j =
Discon	nected
Access type	UDP 💌
Device	-
	Find
Connect Disconn	iect OK

- 1. Set Access type to UDP.
- 2. Click on **Connect.** All the devices connected in network will be displayed.
- Select the device you wish to configure. If it is connected, then Connected is displayed.
- 4. Click **OK**.

## **TCP Connection**

TCP connection can be used if we know IP address of the device we want to connect to. There are no restrictions and you can connect both local device in your LAN and remote device over the Internet.

Default IP address setting:

- LAN: 192.168.1.100
- WAN: 10.1.1.100

Setting of communication	on 🛛 🔀
<b>_</b>	Save Delete
Manager address	240-0-0-0
Device address	0-0-0-0
Latency	,
Discon	nected
Access type	TCP
IP Address	195.168.209.42
TCP Port	7777
Connect Discont	nect OK

- 1. Set Access type to TCP.
- 2. Set IP address and TCP Port (by default UniMan communicate on port 7777).
- 3. Click on **Connect**. If the device is connected, then **Connected** is displayed.
- 4. Click OK.

# **SETTING OF PASSWORD**

To have access to the Voice&Data Router configuration, password authentication is required. Default password is **inoteska**.

You can manage UniMan password using main menu Options/Password.

#### New login

New login with new password.

### Change password of device

Change of default password. It is possible to authorize the different levels and passwords for different users.

Chang	e password [default]			×
User		Password		
def	ault	inoteska	* * * * * * * * * * * * * * * * * * *	+ - ③
1 - F	Read configuration		9 - Change of firmware	
2 · F	Read Access list		10 - Change of Access setting via remote control	
3.			11 - Change of Remote control and IP/Ethernet setting	
4 - [	ownload of Call log		12 - Change of configuration	
5-\	Vrite configuration		13 - Change of password	
6-0	Config files editor		14 -	
7.			15-	
8.			16 - Change of all passwords and levels	
h Bl	BtnRead 🕞 Write	]	×	Cancel

## **CONFIGURATION FILES EDITOR**

Configuration files editor provides access to the device configuration. Just now there is available only network configuration for Voice&Data Router.

To open configuration editor click on the speed button **Config files editor** or use main menu **Communication/Config files editor**.



**File list** on the left side of the configuration editor shows list of configurable items. It can work in text mode or in the graphic (GUI) mode. Text mode offers only direct access to the text configuration files. Graphic mode provide comfort GUI interface.

Configurable items in Voice&Data Router

- Network Graphic GUI interface for the network interface configuration.
- **Network** text mode of the network interface configuration
- Firewall text mode of the firewall configuration

Config files editor		
File list	Network	
Network Graphic Network Firewall	<pre>;! ;! Automatically generated configuration file ;! Filename: /etc/asterisk/manager/network.conf ;! Generator: Manager ;! Creation Date: Wed Jun 13 14:06:20 2012 ;! [lan] iface = eth0 mode = static ipaddress = 192.168.1.170 netmask = 255.255.255.0 gateway = 192.168.1.123 macaddress =</pre>	
	<pre>[wan] iface = eth2 mode = static ipaddress = 10.1.1.170 netmask = 255.0.0.0 gateway = </pre>	
🐴 Read 🛛 🕒 Write	🗁 Open 🔄 🖬 Save	🗙 Cancel

Double click on the selected item in the file list reads configuration files from the device and eventually activates GUI interface. In the text mode you can directly edit configuration files in the right side of the window.

Configuration changes will be applied only after writing configuration files to the device using the **Write** button. You have to write changed configuration also in case of using GUI interface. Buttons **Save** and **Open** can be used to store/load active configuration files to/from disk on your PC.

## Network Setting – GUI

GUI interface for network setting is available from the configuration file editor – **Network Graphic** or directly form main window using speed button **Remote control & IP/Ethernet setting** or from main menu **Communication/Remote control IP/Ethernet setting**.

control	
Interface eth0 - LAN MAC Address DEFAULT	Interface eth2 - WAN MAC Address DEFAULT
IP setting         □ DHCP         IP address       192.168.1.170         Mask       255.255.255.0         Gateway       192.168.1.123         Speed       □ Duplex         ○ AUTO       ○ Half         ○ 100M       ☑ MDIX         ○ 100M Auto       ☑ MDIX	IP setting         □ DHCP         IP address       10.1.1.170         Mask       255.0.0.0         Gateway
Hostname test_170	DNS server 192.168.1.210 V OK Cancel

#### **MAC Address**

Enter 'DEFAULT' if you want to use Ethernet hardware address assigned by producer. If you need to overwrite hardware Ethernet address for some reason to suit your preference enter new address as colon separated hexadecimal numbers (e.g. 00:50:C2:38:76:80). Modified MAC address will be applied only after reboot of Voice&Data Router.

#### DHCP

Check this option to obtain IP address automatically from the network.

#### **IP** address

IP address configuration.

#### Mask

Subnet mask configuration.

#### Gateway

Default gateway. Note that default gateway can be only one – it should be configured only for one Ethernet interface LAN or WAN.

#### Speed, Duplex

Ethernet interface speed and duplex configuration.

#### MDIX

Enable/disable MDIX (Medium Dependent Interface Crossover) option for Ethernet interface.

### Hostname

Enter Voice&Data Router hostname.

### **DNS Server**

Enter IP address of Domain Name Server (only primary).

## DIAGNOSTICS

Use diagnostic window to monitor system status and status of each interfaces. Diagnostic is activated by the speed button **Diagnostic** or from the main menu command – **Communication/Diagnostic**.



**Type of diagnostic** list on the left side of the diagnostics shows index of items that can be monitored. Double click on the selected item will display information of actual status. In the most cases is the status automatically updated.



## LISTING MESSAGES

This function allows making online logs of Voice&Data Router call switching. You can access listing window by clicking on the speed button **Listing messages** or by using main menu commond – **Communication/Listing messages**.



On the **Setting** card you can configure logging filter – selection of interfaces and protocols to monitor and also monitoring level.

A Listing messages	
🗅 A 🔢 🗖 Keep connection alive	START STOP
🔽 Asterisk Verbose	
Asterisk Debug	
E1/T1 Ports ABCD	
✓ UniCall CAS Line Signaling	
CAS Register Signaling	
ISDN L2	
SIP PEERS ALL	
Setting Listing	
	🗙 Cancel

Click **START** button to begin capturing of the logs from the device. Received logs are displayed in real-time on the **Listing** card. Use **STOP** button to finish device monitoring and store captured logs in the text file on the local PC.

Listing messages	
🗅 A 🖪 🦵 Keep connection alive	START STOP
[Jul 26 13:29:11] ISDN-PRI(0): Processing IE 126 (cs0, User-User) [Jul 26 13:29:11] ISDN-PRI(0): q931.c:7422 post_handle_q931_message: Call 34093 enters state 10 (Active). H [Jul 26 13:29:11] ISDN-PRI(0): > Protocol Discriminator: Q.931 (8) len=5 [Jul 26 13:29:11] ISDN-PRI(0): > Message Type: CONNECT ACKNOWLEDGE (15) [Jul 26 13:29:11] ISDN-PRI(0): > Message Type: CONNECT ACKNOWLEDGE (15) [Jul 26 13:29:11] ISDN-PRI(0): Event type: Answer (8) [Jul 26 13:29:11] ISDN-PRI(0): Event type: Answer (8) [Jul 26 13:29:11] ISDN-PRI(0): Don't know how to dump events of type 8 [Jul 26 13:29:11] = User-to-user message received: "Recording your voice" [Jul 26 13:29:11] - UniCall/1-E15649f7b9 answered SIP/10-00000001 [Jul 26 13:29:11] - User-to-user message received: "Recording your voice" [Jul 26 13:29:11] - VIniCall/1-E15649f7b9 answered SIP/10-000000001 [Jul 26 13:29:11] - SIP/10-00000001 <-CHANNEL-8RIDGE > UniCall/1-E15649f7b9 [Jul 26 13:29:11] - SIP/10-00000001 <-CHANNEL-8RIDGE > UniCall/1-E15649f7b9 [Jul 26 13:29:11] - SIP/10-00000001 <-CHANNEL-8RIDGE > UniCall/1-E15649f7b9 [Jul 26 13:29:11] - UniCall/1-E15649f7b9 formats: read=0x8 (alaw), write=0x8 (alaw), native=0x2d0f (g723]gsm [Jul 26 13:29:11] - UniCall/1-E15649f7b9 formats: read=0x8 (alaw), write=0x8 (alaw), native=0x2d0f (g723]gsm [Jul 26 13:29:11] - unicall set_rtp_peer(UniCall/1-E15649f7b9 [0.0.0.0.8024 -> 192.168.1.191:1722], 0x10e ( [Jul 26 13:29:11] - unicall set_rtp_peer(UniCall/1-E15649f7b9 [0.0.0.0.8024 -> 192.168.1.191:1722], 0x10e ( [Jul 26 13:29:11] User function(12) [Jul 26 13:29:11] User function(12) [Jul 26 13:29:11] User function(12) [Jul 26 13:29:11] MSPD channel status changed (10). [Jul 26 13:29:11] MSPD channel status changed (10). [Jul 26 13:29:11] alarm: 0xa0 [Jul 26 13:29:11] alarm: 0xa0 [Jul 26 13:29:11] alarm: 0xa0 [Jul 26 13:29:11] alarm: 0xa0 [Jul 26 13:29:11] alarm: 0xa0	old state: Idle  g729)  ulaw alaw g726 g729 i gsm ulaw alaw g729)) 255.255.255.255:6553
	🗶 Cancel

## **IDENTIFICATION**

Identification provides basic hardware information about Voice&Data Router. To open identification window use speed button **Identification** or main menu command – **Communication/Identification**.



BASIC section information is read from permanent memory which is programmed during device production. Here you can find information such as device type – product code number, serial number, date of production, hardware interfaces, etc. CONFIG and FUNCTIONS parts are stored in the file and they can be used to enable/disable optional interfaces and functionalities.

@ Identification	
[BASIC] TYPE=ITX495012000220 SN=455010990102 ID=5490 DATE=211 2010	
SERVP=F3DE1AF32994DA01EA9299293C6157B0 MAN=103,104 DPS=ITB 166 09 PR0D=INOTESKA ADD=DDTUBEN POVEN231 LIBTOVSKY HEADOK 03201	≣
ADR=PUD10HEN-RUVEN221,LIP10VSKY HRADUK,03301 TEL=+421445221809,+421903360360 MAIL=MAIL@INDTESKA.SK WWW=WWW.INDTESKA.SK HW=1000 RD=D SDRAM=128M FLASH=8M CARD=SD FPGA=ICDP1C12F324C8 CPU=M82810	
	►
Info List	Cancel

# **TERMS OF SALE**

## Warranty:

Product warranty period is 24 months from the date of delivery or installation. Warranty does not apply in case of an accident, handling by a non-professional or improper use or force majeur.

#### **Delivery:**

Standard delivery time is max. 6 weeks from the signing of the purchase order or after mutual agreement.

#### Contact:

Inoteska s.r.o.	Tel.:	+ 421 44 5567 911
Podtureň-Roveň 221	Fax:	+ 421 44 5221 519
Liptovský Hrádok	Hotline:	+ 421 902 774 538
033 01		
Slovenská Republika	Web:	www.inoteska.sk
-	E-mail:	email@inoteska.sk

VAT no.: SK2020428300 Bank information: Všeobecná úverová banka a.s. Account no.: 616243342/0200 SWIFT code: SUBASKBX IBAN: SK34020000000616243342

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