VOICE&DATA ROUTER
ITX 495 01

PRODUCT DOCUMENTATION

CONTENTS

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 d e f g h i j – Voice&Data Router

- **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
  - 1 = RJ45
  - 2 = BNC
- **Power Supply**
  - 1 = 230V, 48V
  - 2 = 230V
  - 3 = 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
  - 5 = 8 BRI Uk0
  - 6 = 16 BRI Uk0
  - 7 = 8 BRI S0
  - 8 = 16 BRI S0
  - 9 = 8 BRI Uk0, 8 BRI S0
- **Compression**
  - 0 = no SIP
  - 1 = 16 channels G.729, 48 channels G.711
  - 2 = 32 channels G.729, 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
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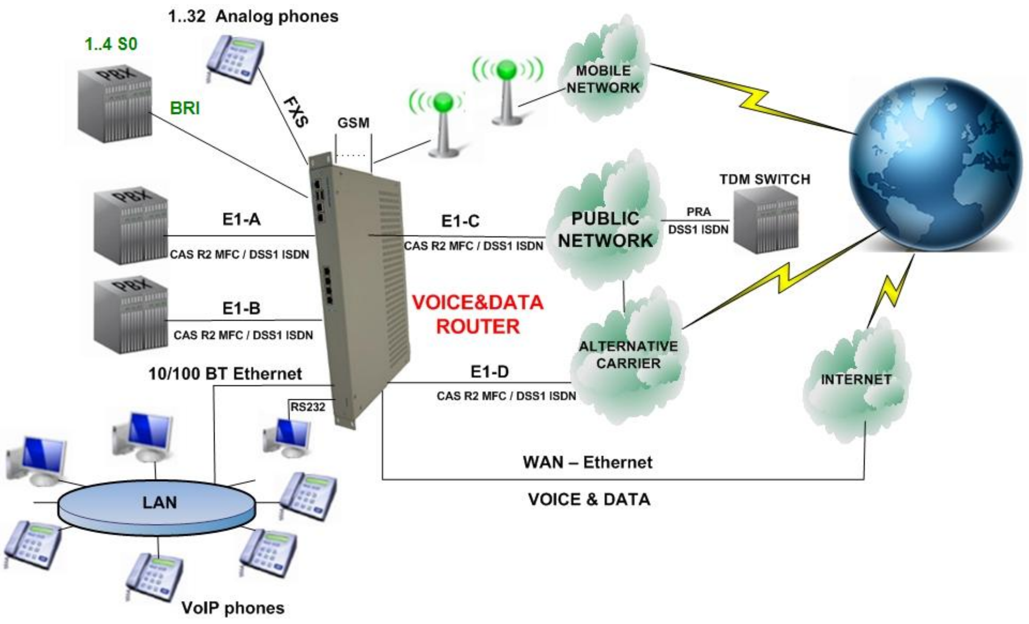
  - 1 = SM (Single Mode)
  - 2 = MM (Multi Mode)
- **Compression**
  - 1 = 16 channels G.729, 48 channels G.711
  - 2 = 32 channels G.729, 64 channels G.711

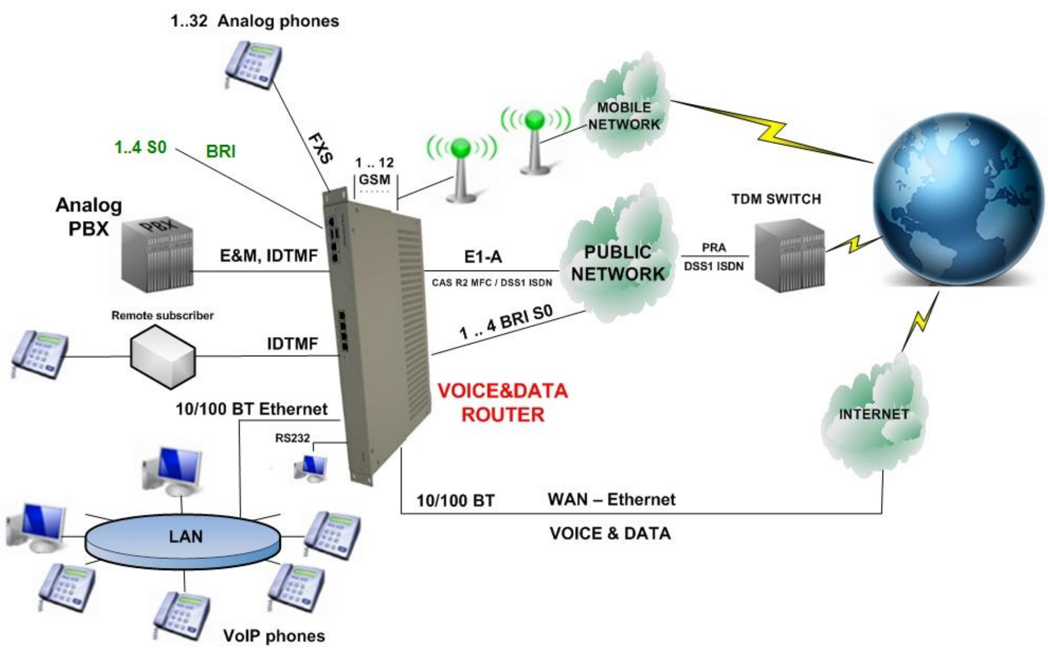
APPLICATIONS

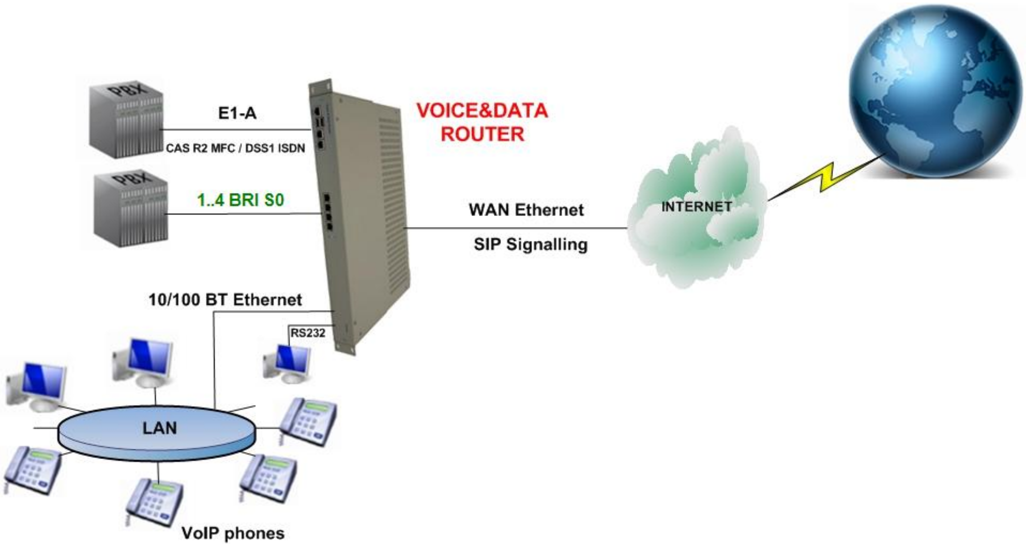
- Voice & Data Router
- VoIP, TDM PBX (SW Asterisk)
- TDM - VoIP Gateway
- Signaling converter R2 MFC / ISDN DSS1 – SIP, Routing to another E1 if there is no more free capacity for simultaneous calls
- Concentration of max. 4xE1 (considering the limitation of max. number of simultaneous calls depending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

Voice&Data Router / VoIP, TDM PBX

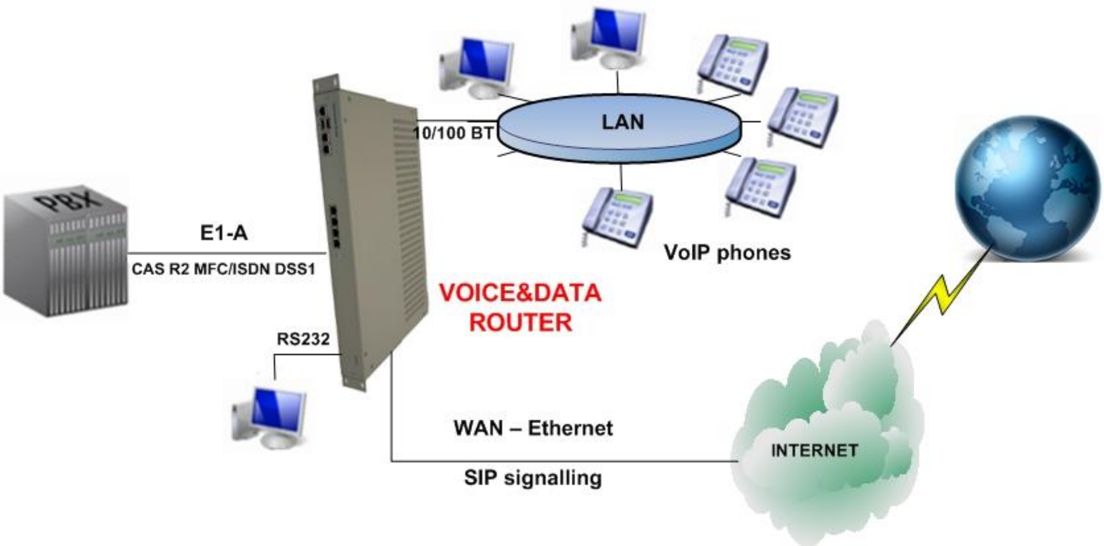


TDM – VoIP Gateway

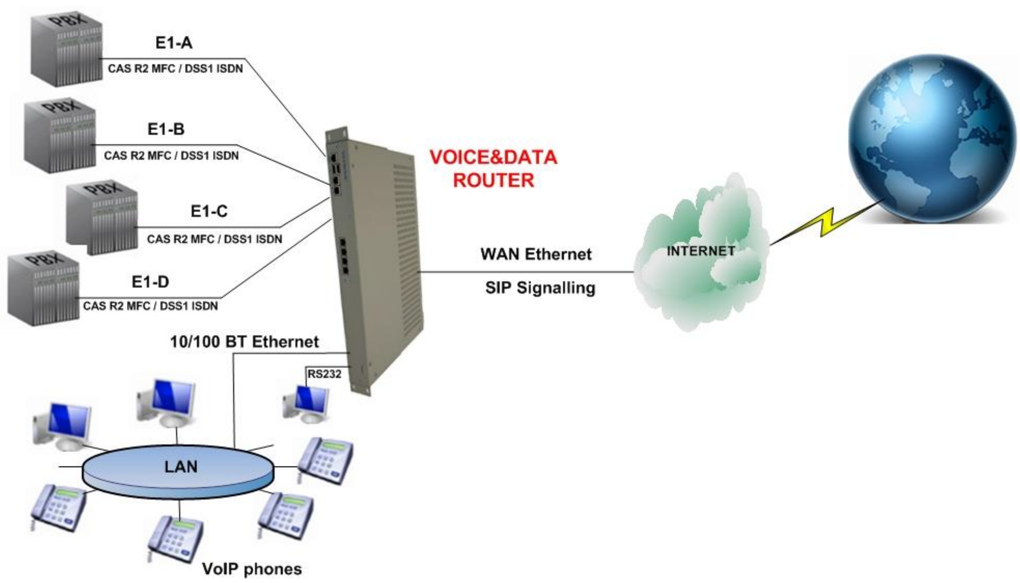




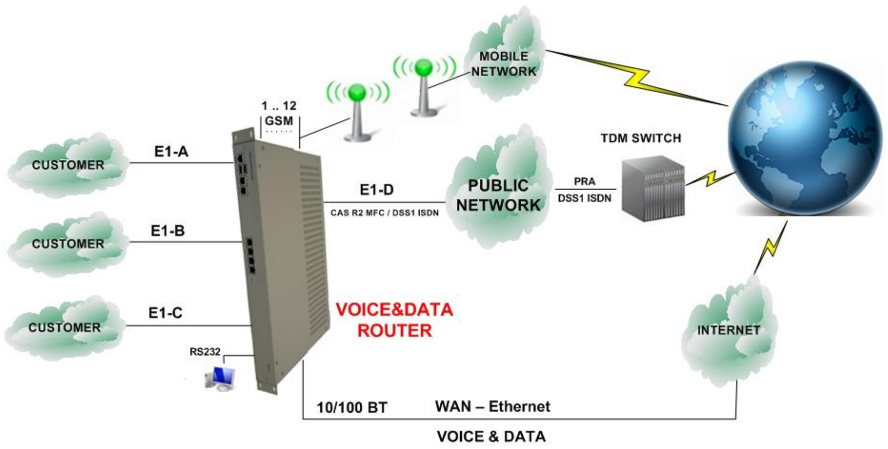
Signaling converter R2 MFC / ISDN DSS1 – SIP



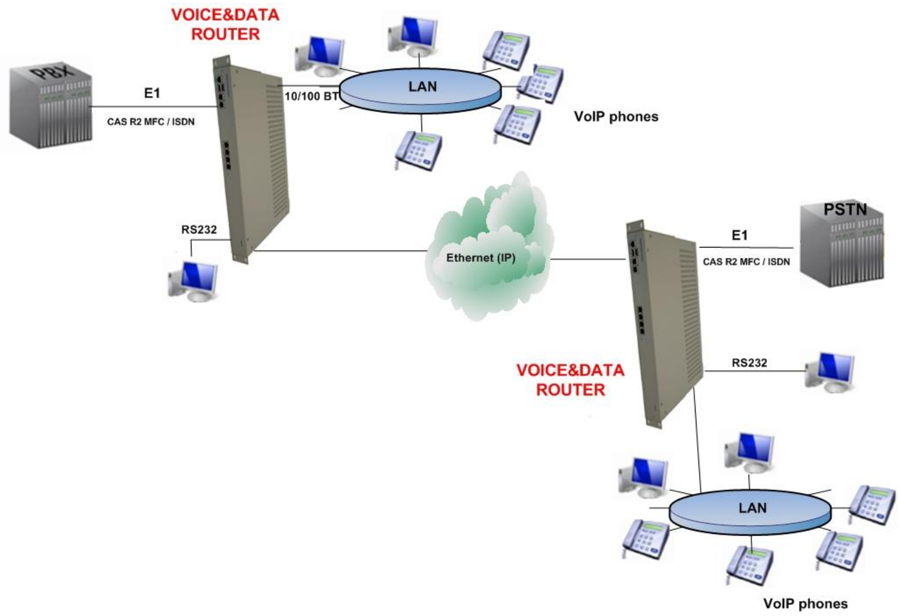
Concentration of max. 4xE1/T1 (considering the limitation of max. number of simultaneous calls depending on the codec used)



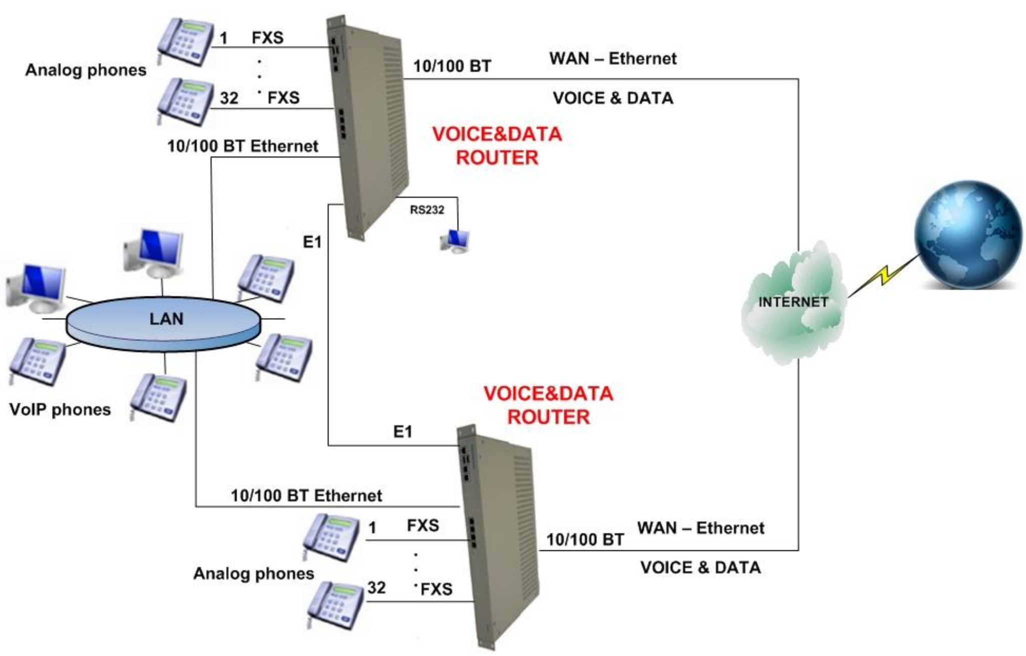
Multi Gateway (possibility to connect several customers and divide them)



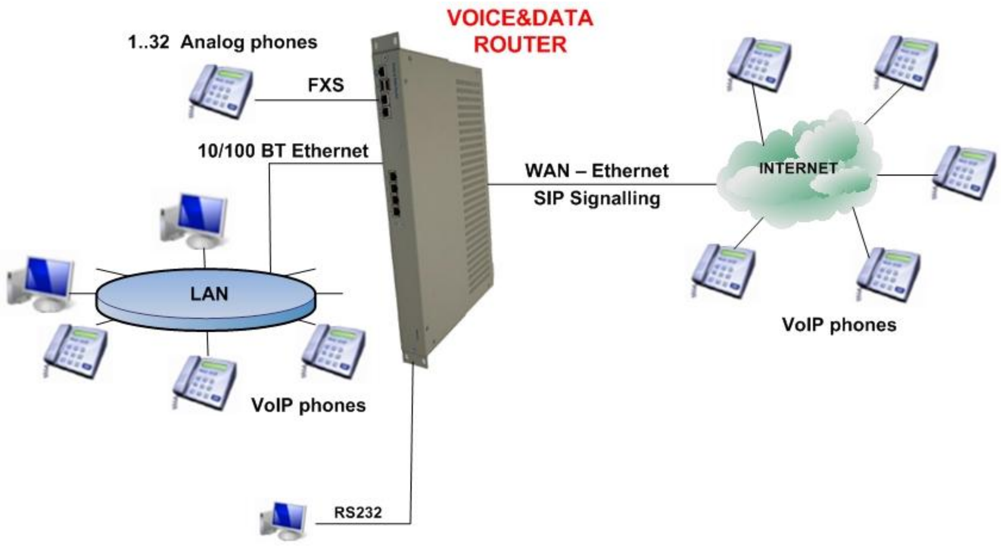
E1/T1 over Ethernet (IP)



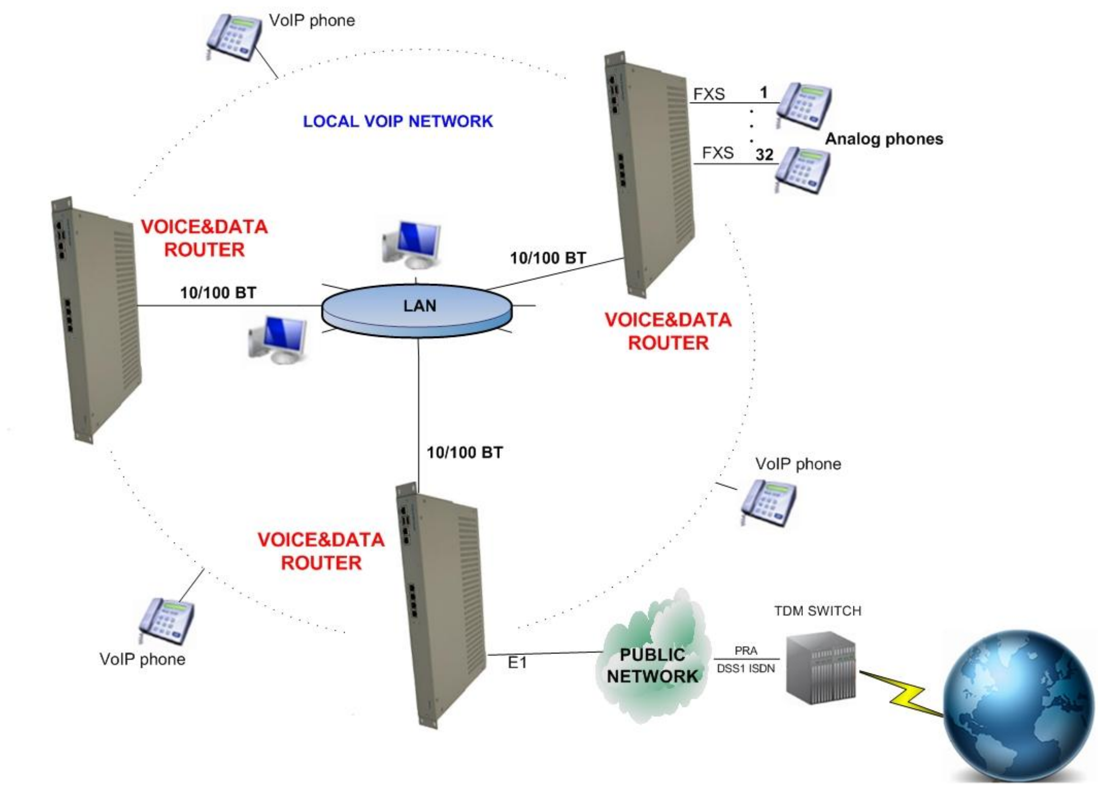
Extending



SIP PROXY Server



Local VoIP network



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
    - 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
    - V.27 2400 / 4800
    - 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
    - G.726
    - G.729 Annex A & B
    - 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 Codecs	Channel Density	
	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
T.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

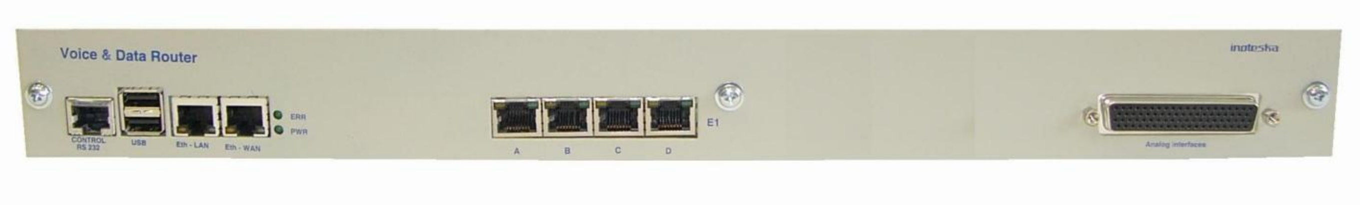
Approx. 3,5kg (real weight depends on device HW configuration)

OPERATING INSTRUCTIONS

Operating Environment
Install the device in a place where:

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

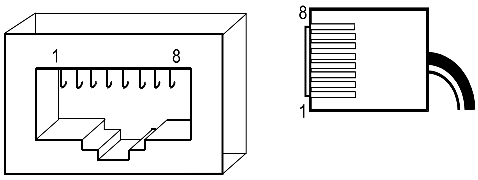
Interfaces



E1/T1 interface

RJ 45 connector

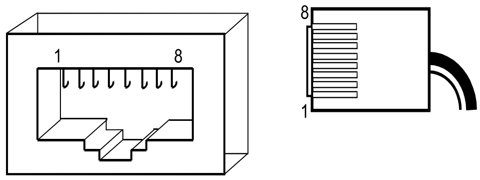
1 – input wire to device Rx –
2 – input wire to device Rx +
3 –
4 – output wire from device Tx –
5 – output wire from device Tx +
6 –
7 –
8 –



Ethernet 10/100Base-T interface

RJ 45 connector

1 – transmit from device Tx +
2 – transmit from device Tx –
3 – receive to device Rx +
4 –
5 –
6 – receive to device Rx –
7 –
8 –



Analog interfaces

Analog extension card can contain 1 up to 8 modules.

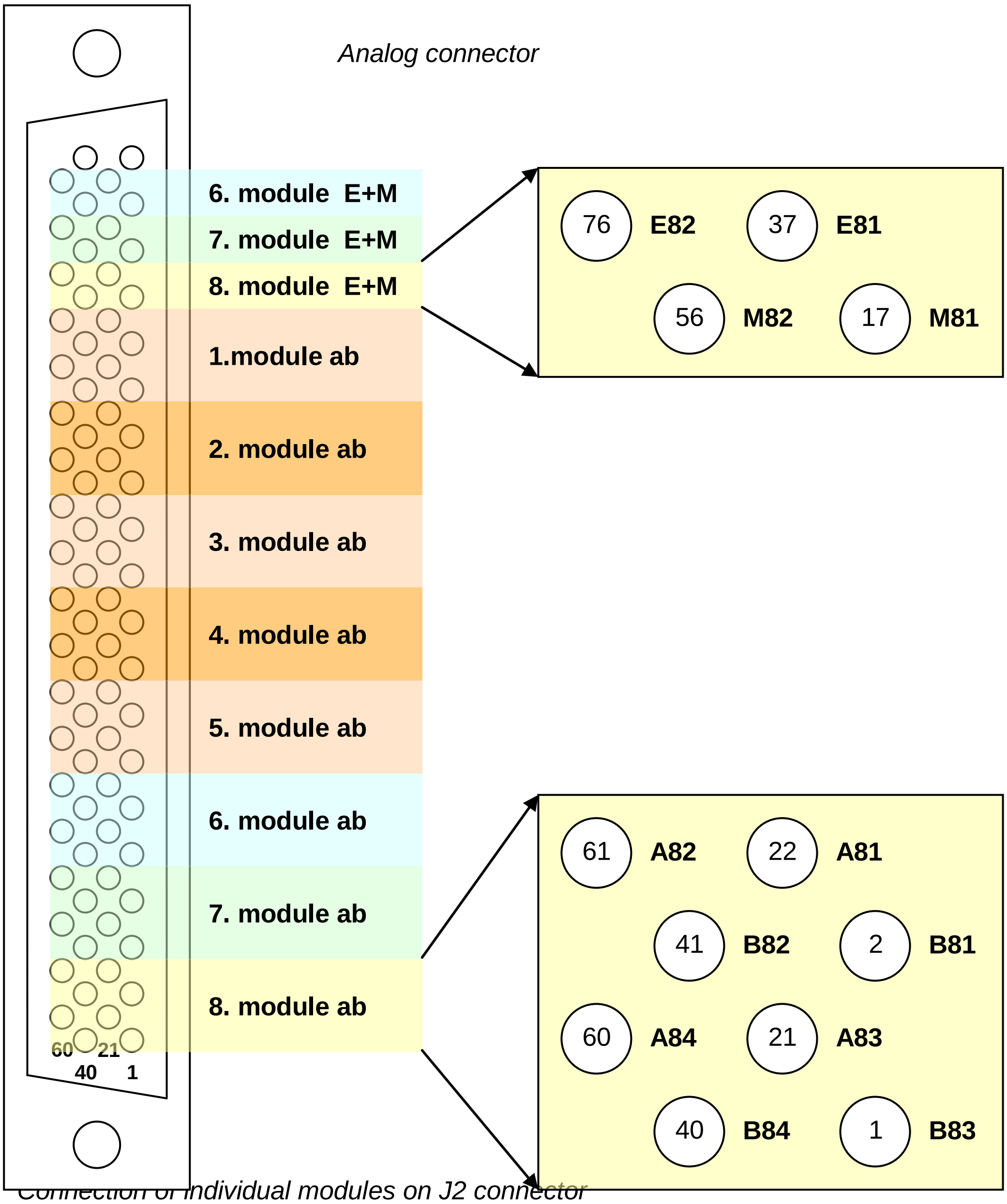
Available modules

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

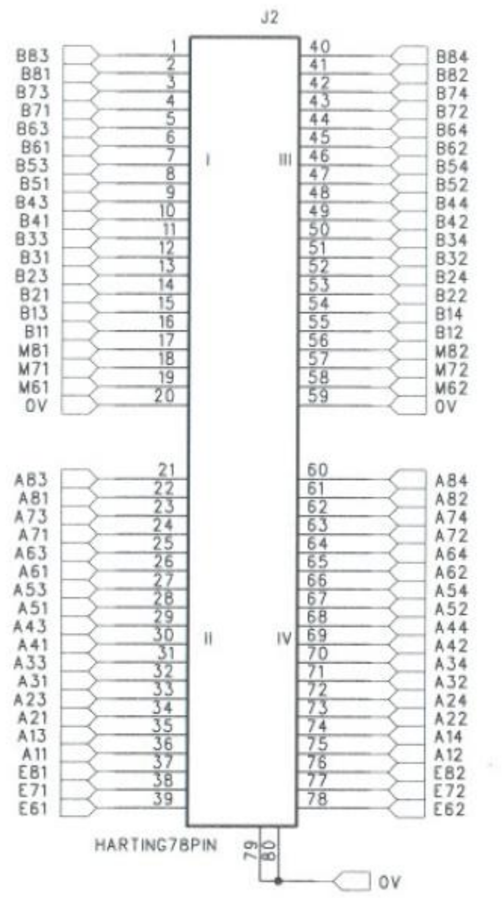
1	2	3	4	5	6	7	8
A11 B11	A21 B21	A31 B31	A41 B41	A51 B51	A61 B61	A71 B71	A81 B81
A12 B12	A22 B22	A32 B32	A42 B42	A52 B52	A62 B62	A72 B72	A82 B82
A13 B13	A23 B23	A33 B33	A43 B43	A53 B53	A63 B63	A73 B73	A83 B83
A14 B14	A24 B24	A34 B34	A44 B44	A54 B54	A64 B64	A74 B74	A84 B84
					E61 M61	E71 M71	E81 M81
					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



Connection of individual modules on J2 connector

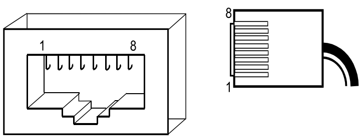


Description of J2 connector

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

Pins are connected as TE in default configuration.

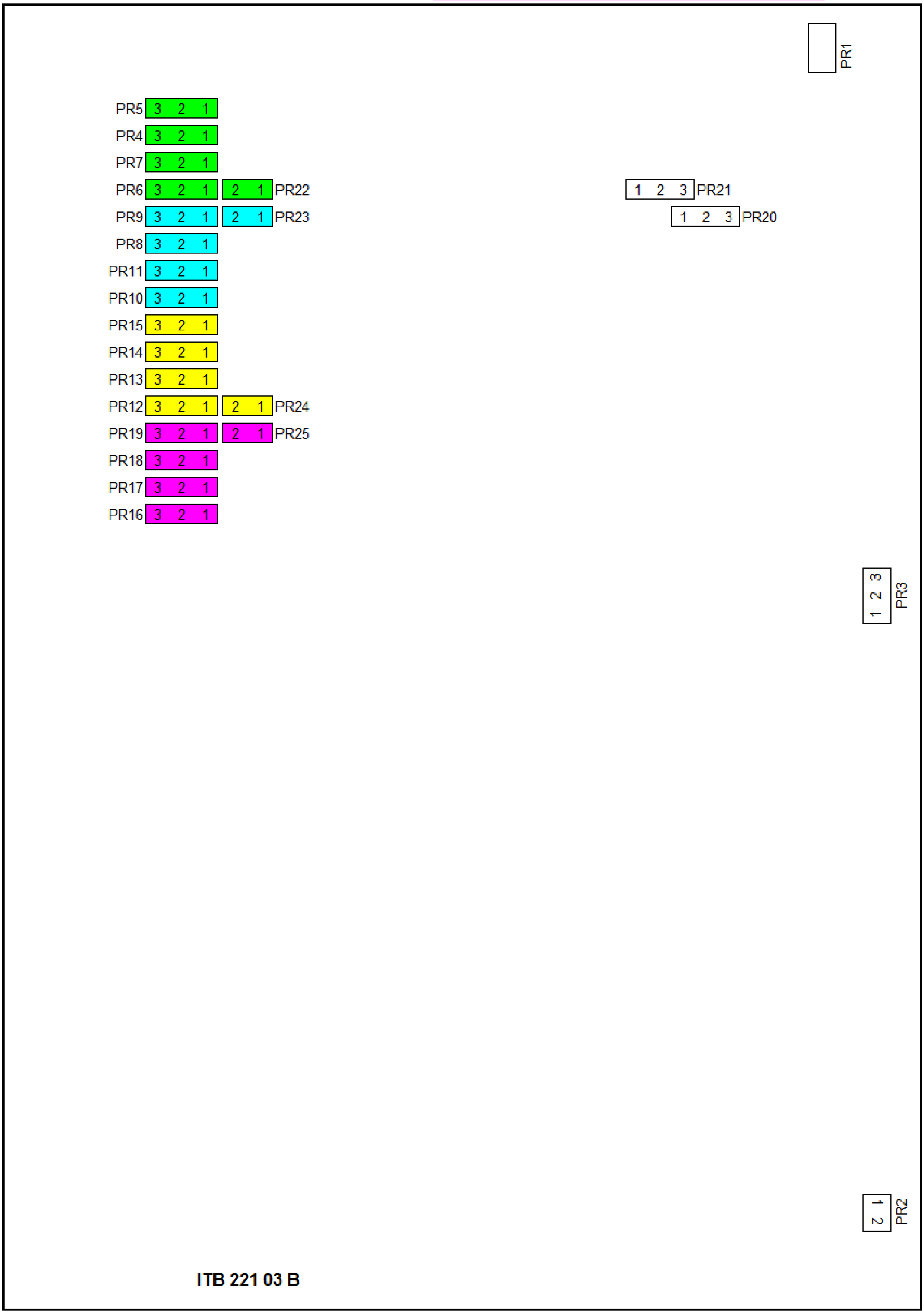


	NT	TE
1 –		
2 –		
3 –	Rx+	Tx+
4 –	Tx+	Rx+
5 –	Tx-	Rx-
6 –	Rx-	Tx-
7 –		
8 –		

4x BRI S0 – jumper settings

Interface	NT+Phantom Power	NT		TE	.
BRIA	PR22-ON	PR4	1-2	PR4	2-3
		PR5	1-2	PR5	2-3
		PR6	1-2	PR6	2-3
		PR7	1-2	PR7	2-3
BRIB	PR23-ON	PR8	1-2	PR8	2-3
		PR9	1-2	PR9	2-3
		PR10	1-2	PR10	2-3
		PR11	1-2	PR11	2-3
BRIC	PR24-ON	PR12	1-2	PR12	2-3
		PR13	1-2	PR13	2-3
		PR14	1-2	PR14	2-3
		PR15	1-2	PR15	2-3

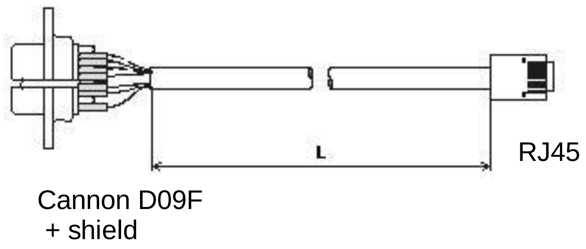
BRID	PR25-ON	PR16 1-2	PR16 2-3
		PR17 1-2	PR17 2-3
		PR18 1-2	PR18 2-3
		PR19 1-2	PR19 2-3



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 green	Led diode yellow	Status
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	OK
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
AIS Alarm Indication Signal – Transmitted signal is constant with data value Log1
LFA Loss of Frame Alignment – Indicates synchronization error in 0th channel
RRA Receive Remote Alarm – Indicates remote device alarm

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.

Default login parameters:

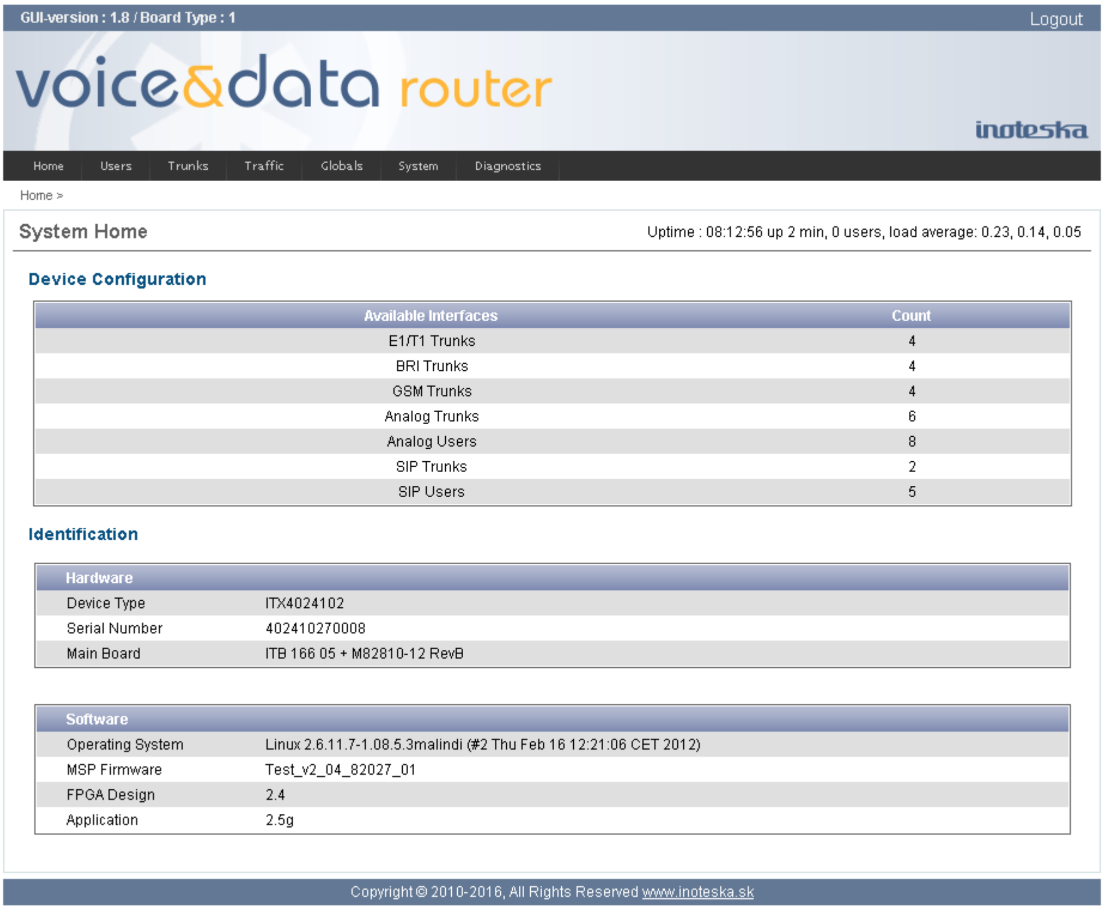
- Username: **admin**
- Password***: inoteska**

**Change of password can be done in Options 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.



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.

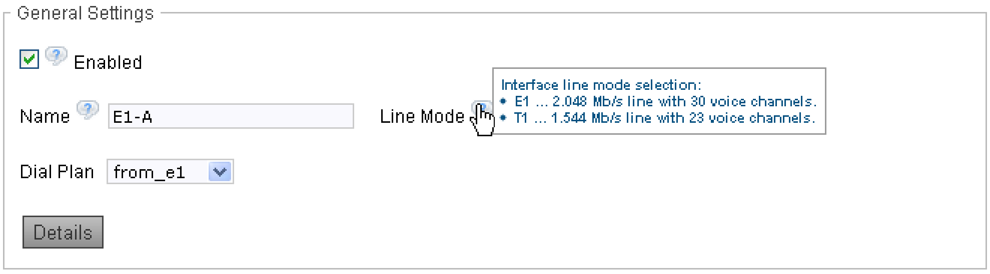
Step 4

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.**

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.



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
- IP settings
- 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

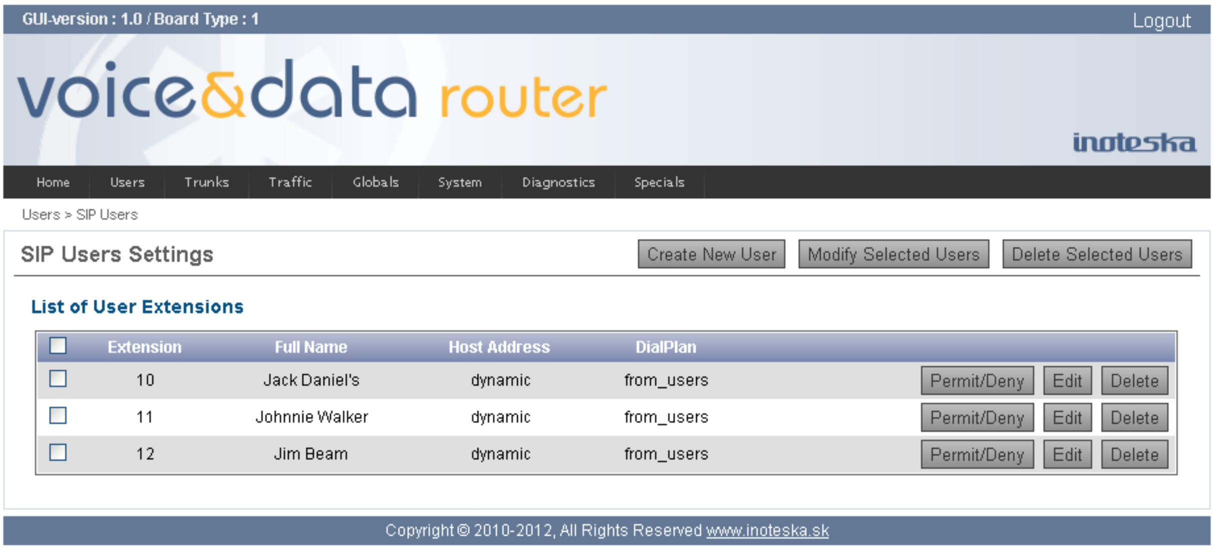
Specials Menu
- AOC Web Update

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.



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

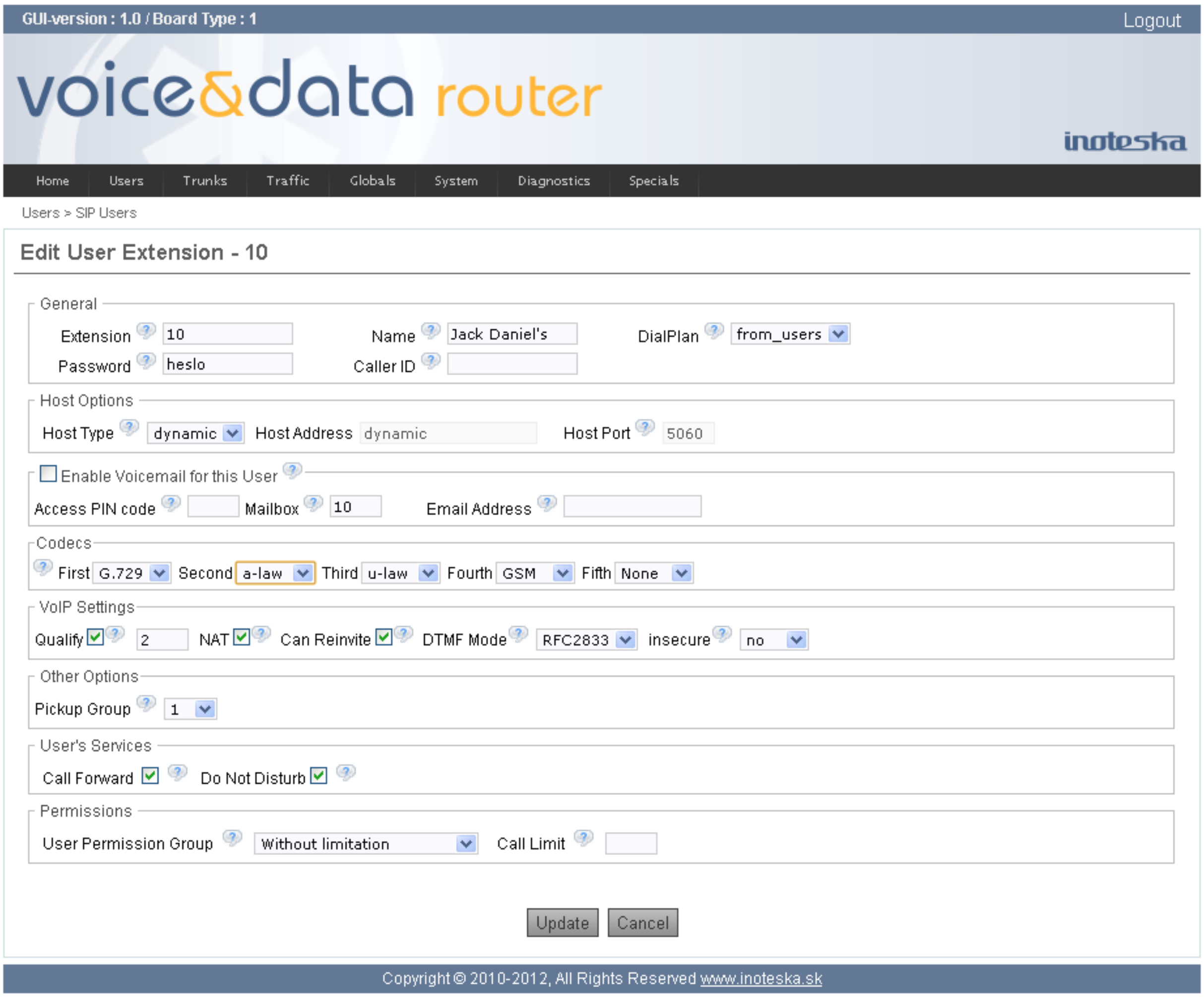
Extension
Extension number associated with this SIP user line. This number is also used as user name for authentication.

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 Traffic Menu for more details about dial plans and calling rules.



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 Services Codes 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 Services Codes 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 Permissions 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.

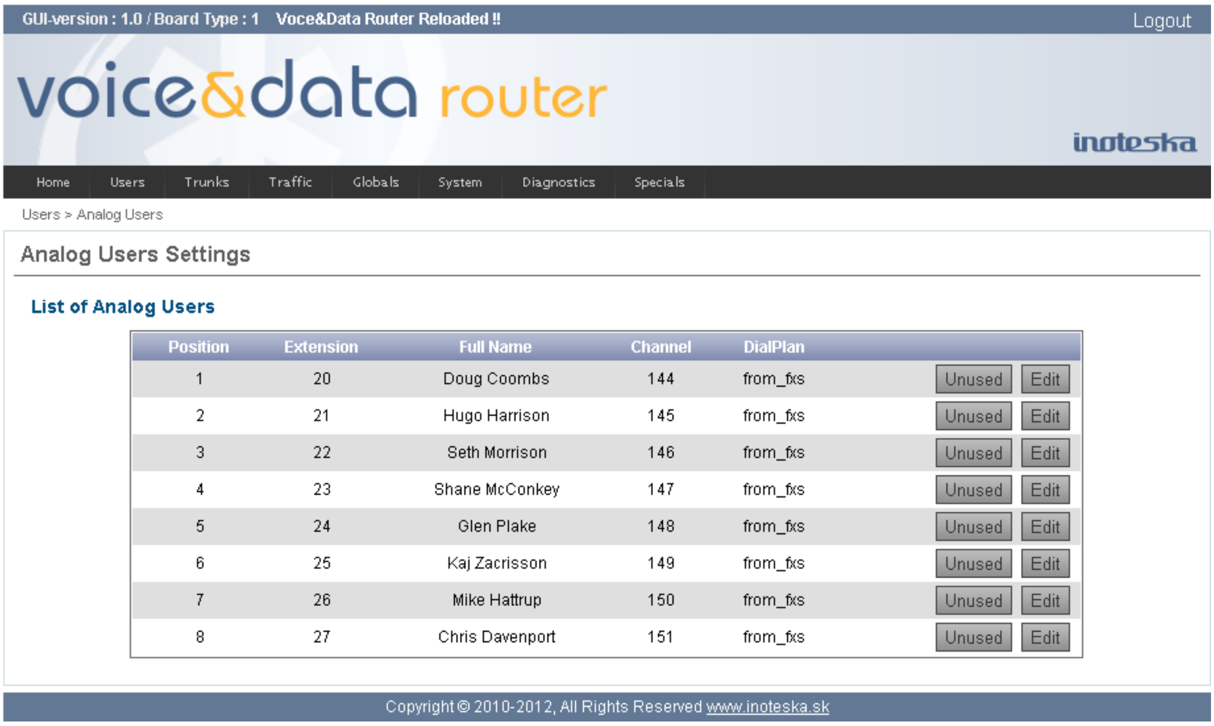
Example:

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

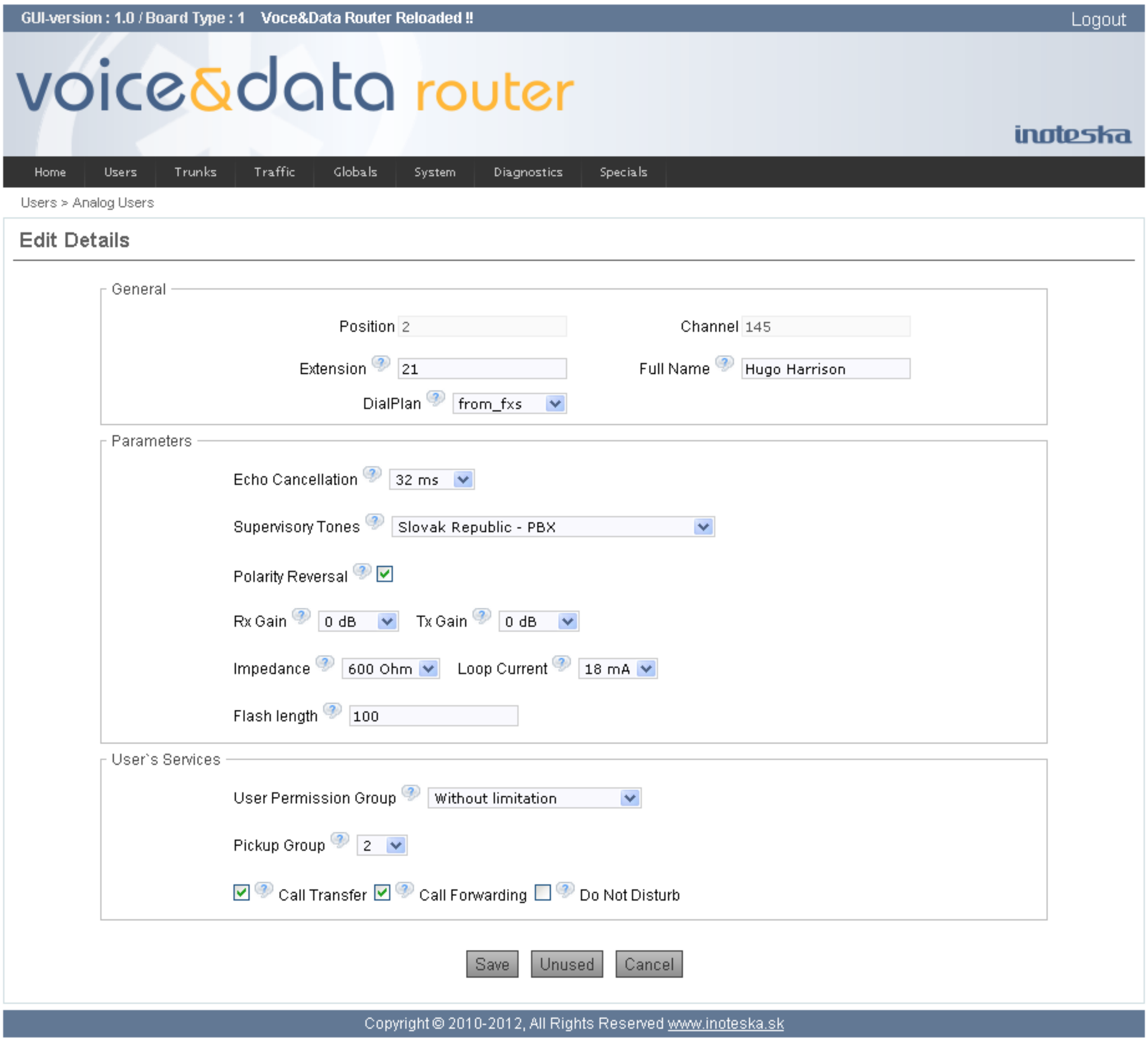
Analog users

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).



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.



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 Traffic Menu 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.

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 Permissions 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 Services Codes 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 Services Codes menu.

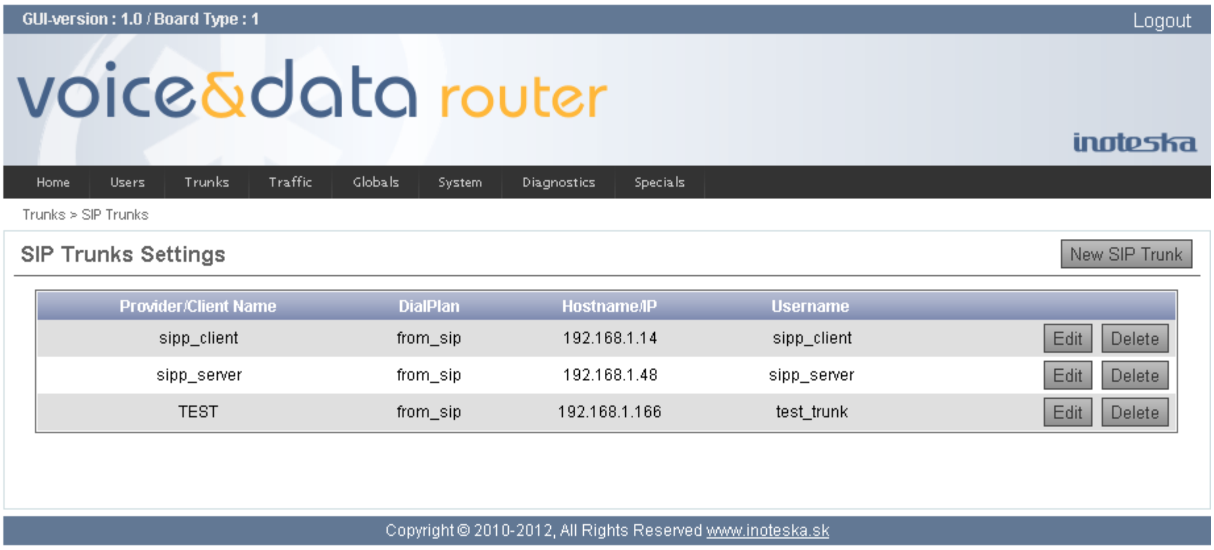
Do Not Disturb
Permit/deny Do Not Disturb service for extension user. If enabled the user is allowed to activate or deactivate this service by service codes defined in Services Codes menu.

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.



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

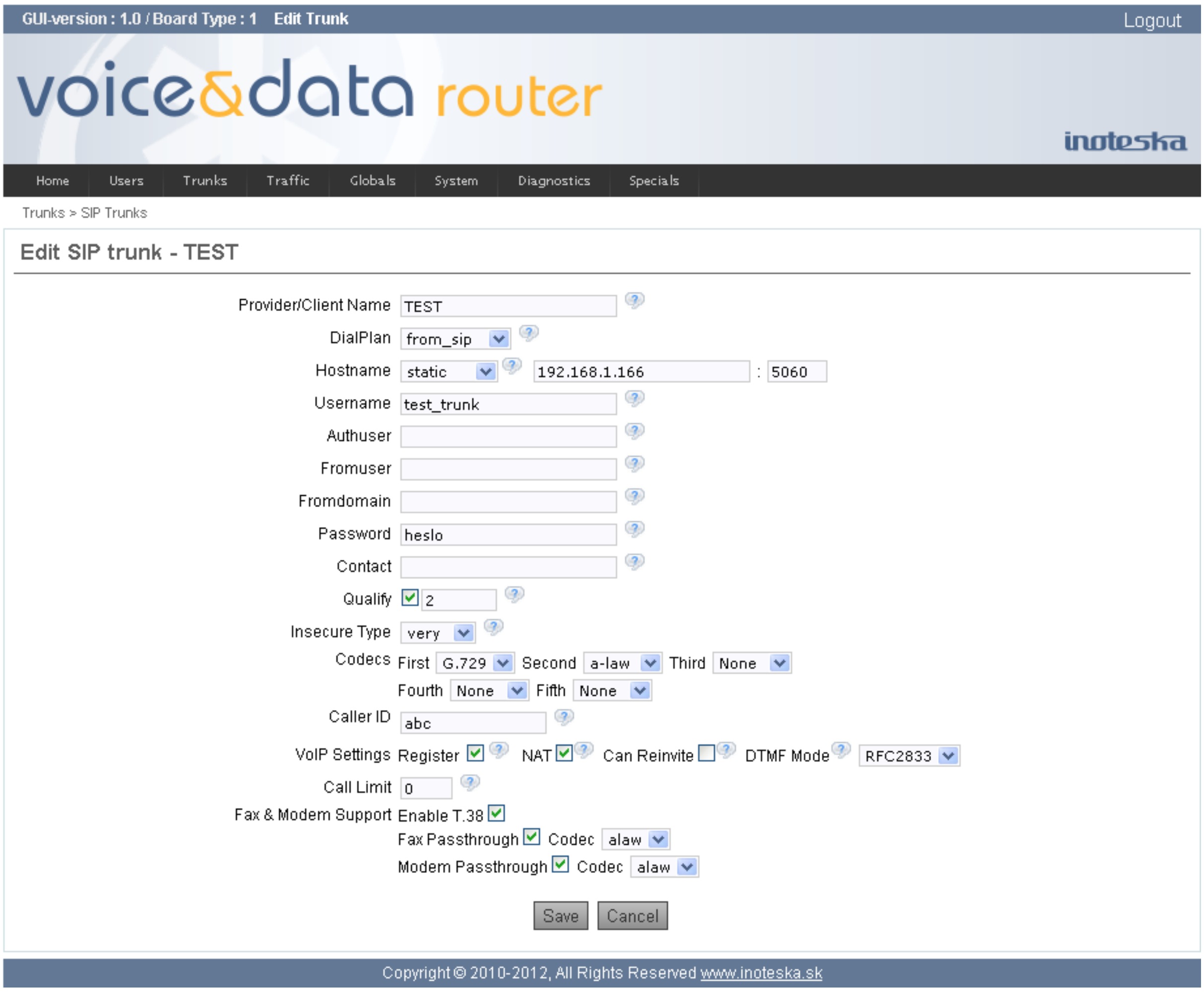
Provider/Client Name
A unique label to help you identify this VoIP trunk. This identifier is used over all of Voice&Data Router configuration where we need reference to this trunk.

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 Traffic Menu 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.



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/ µ-law) to use for Fax Pass-Through.

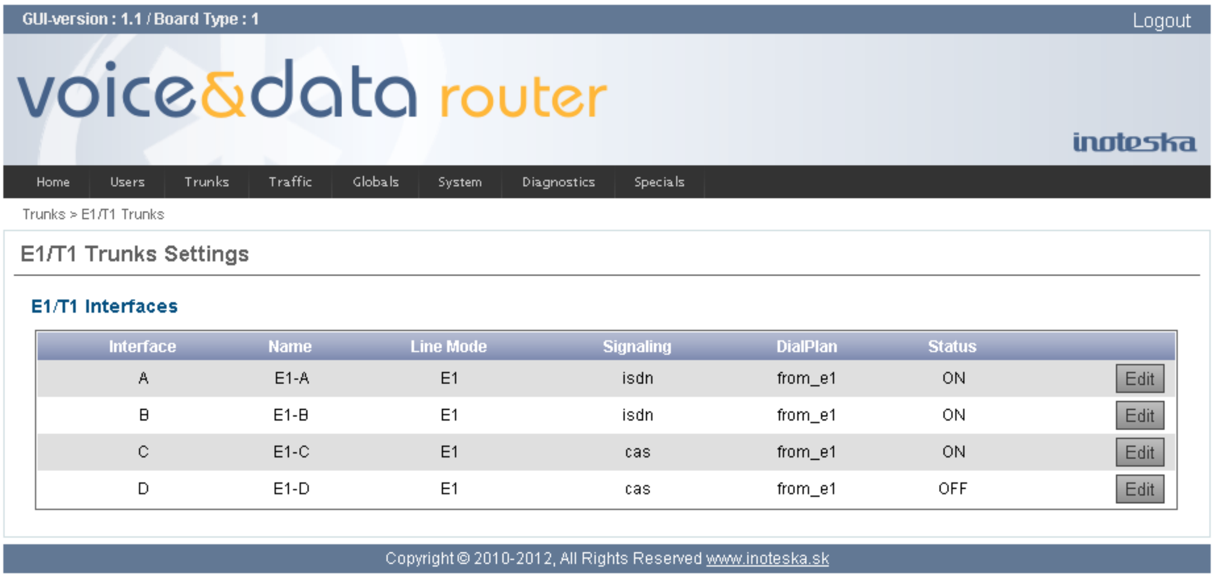
Modem Passthrough

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/ µ-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.



Interface	Name	Line Mode	Signaling	DialPlan	Status	
A	E1-A	E1	isdn	from_e1	ON	Edit
B	E1-B	E1	isdn	from_e1	ON	Edit
C	E1-C	E1	cas	from_e1	ON	Edit
D	E1-D	E1	cas	from_e1	OFF	Edit

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.

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

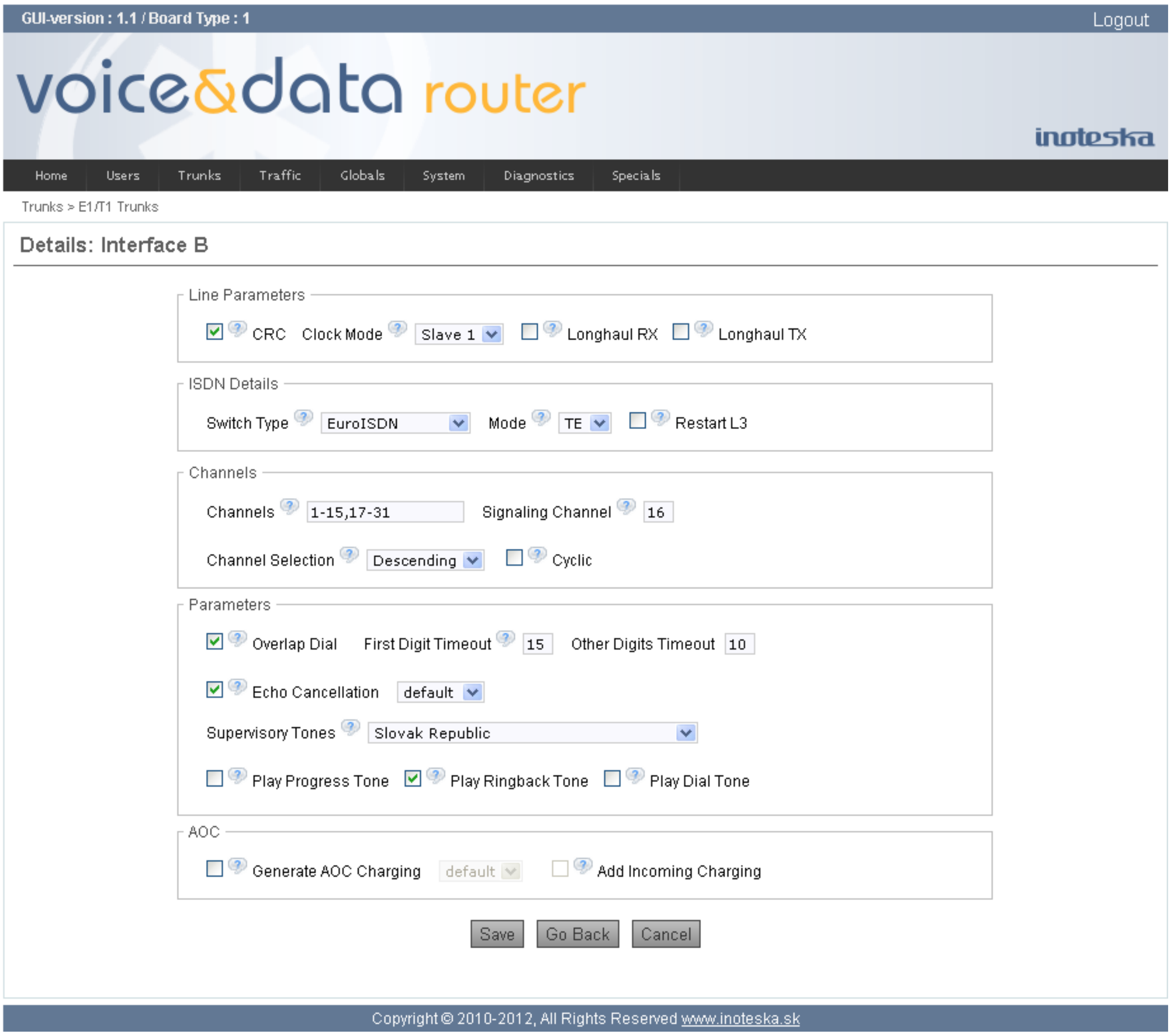
- **E1** 2,048 Mb/s line divided to 32 timeslots
- **T1** 1,544 Mb/s line divided to 24 timeslots.

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 Traffic Menu 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.



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 CAS National Variants 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.

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 16th 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

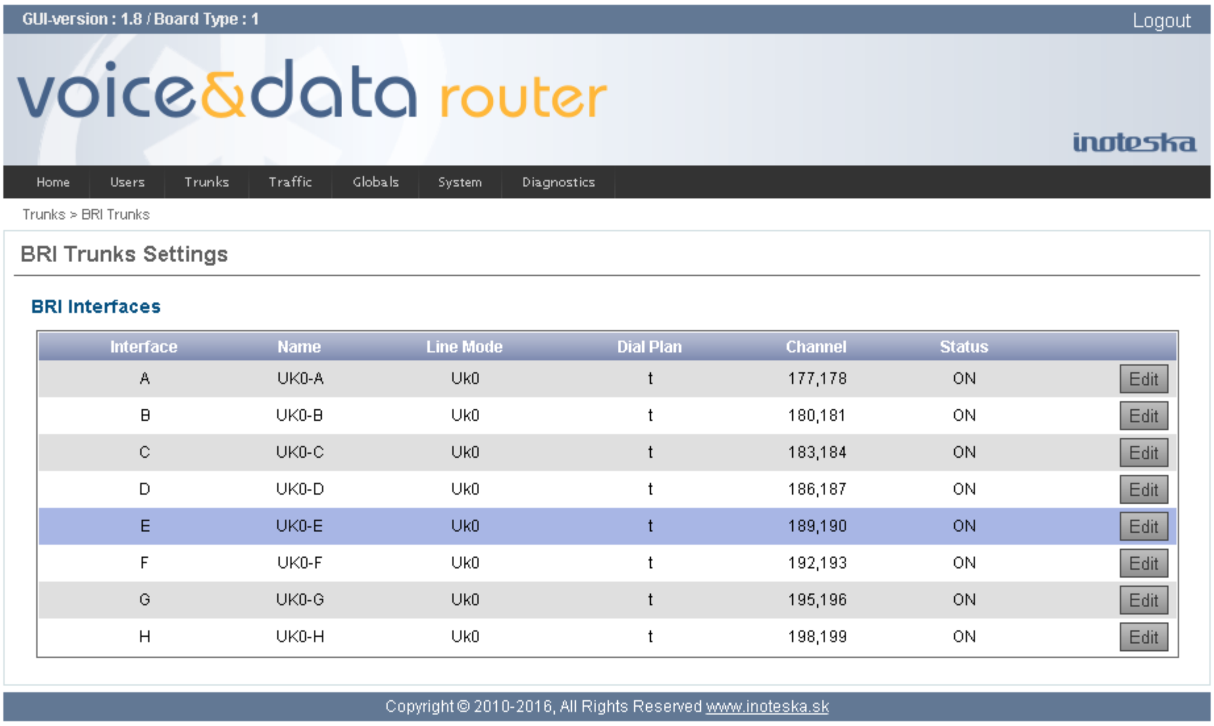
Generate AOC Charging
Enable/disable generation of charging information according selected AOC table. See section AOC Tables Settings for details about configuration of AOC charging generation.

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.



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.

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 Traffic Menu for more details about dial plans and calling rules configuration.

Click **Details** button to continue with detailed configuration.

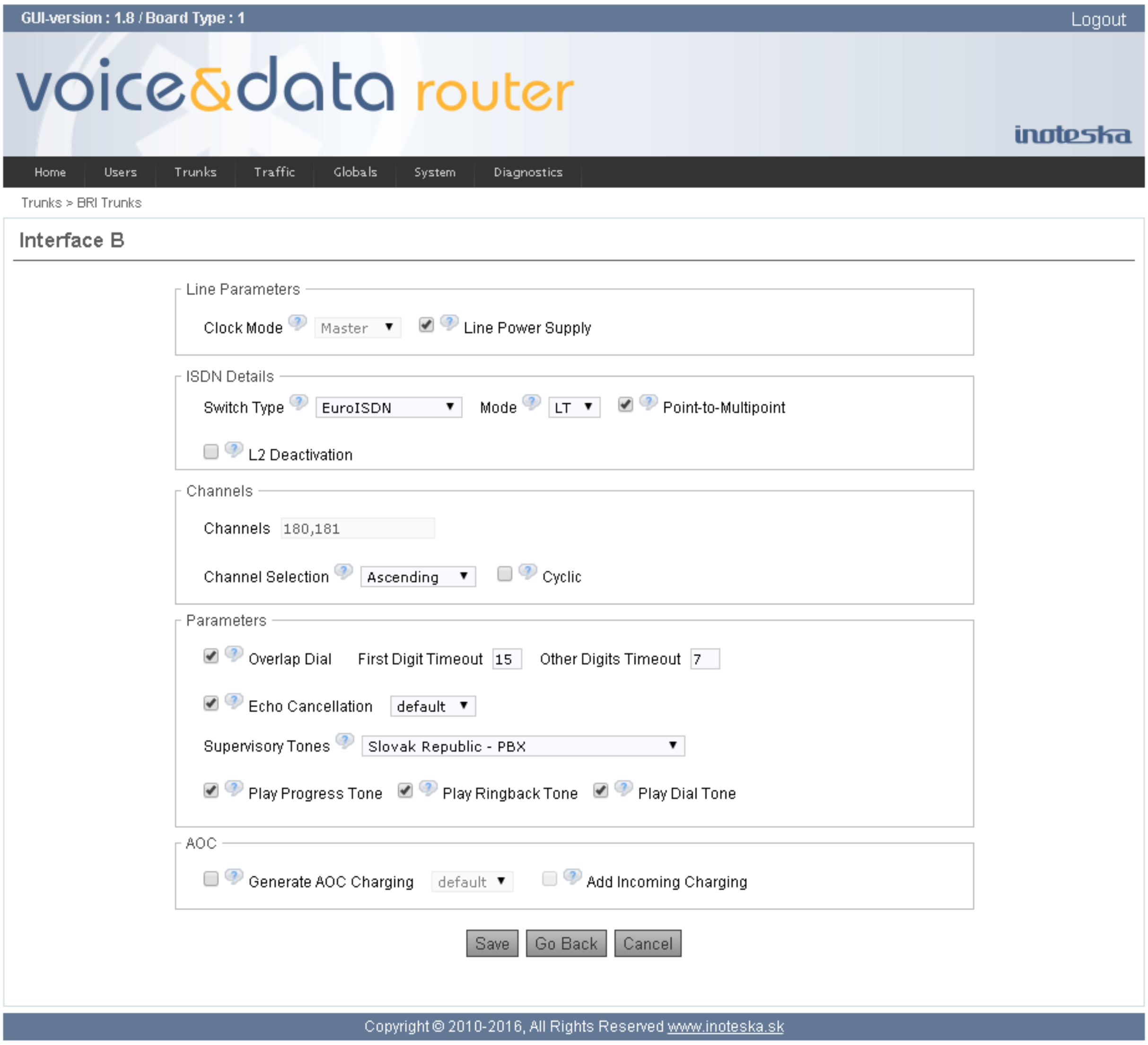
Line Parameters

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.



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 AOC Tables Settings for details about configuration of AOC charging generation.

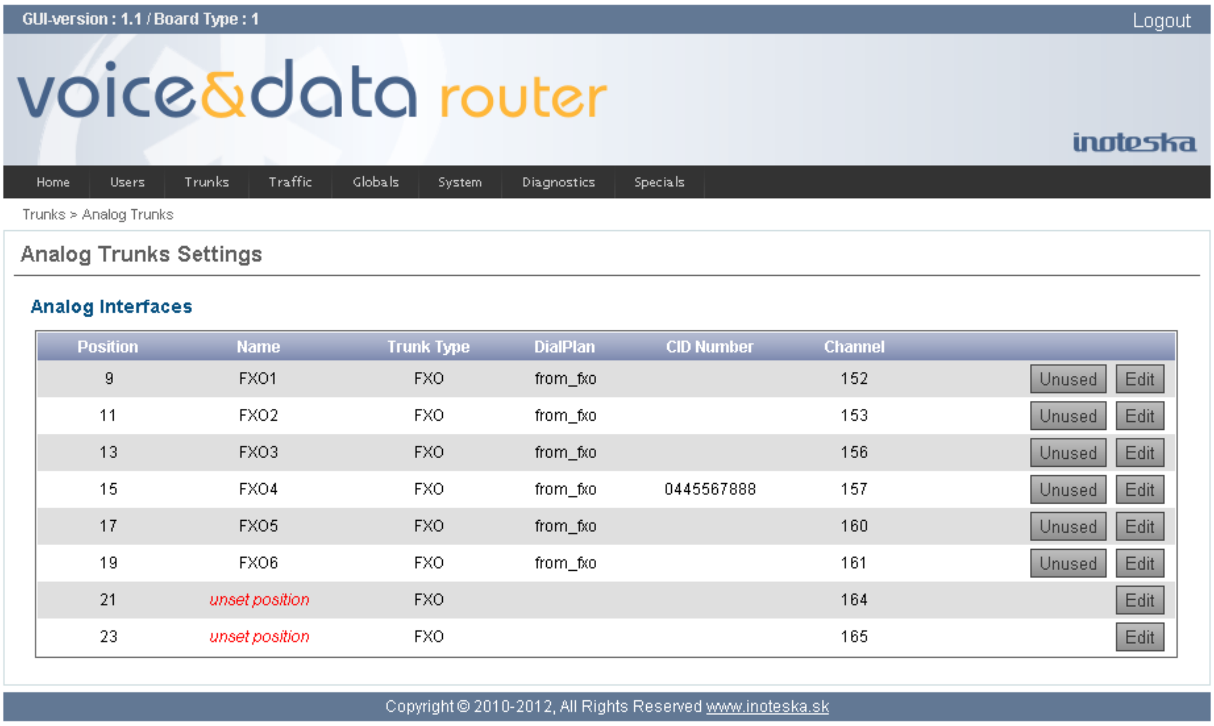
Add Incoming Charging
If enabled charging units received from destination will be added to the charging units generated by Voice&Data Router according AOC table.

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.



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

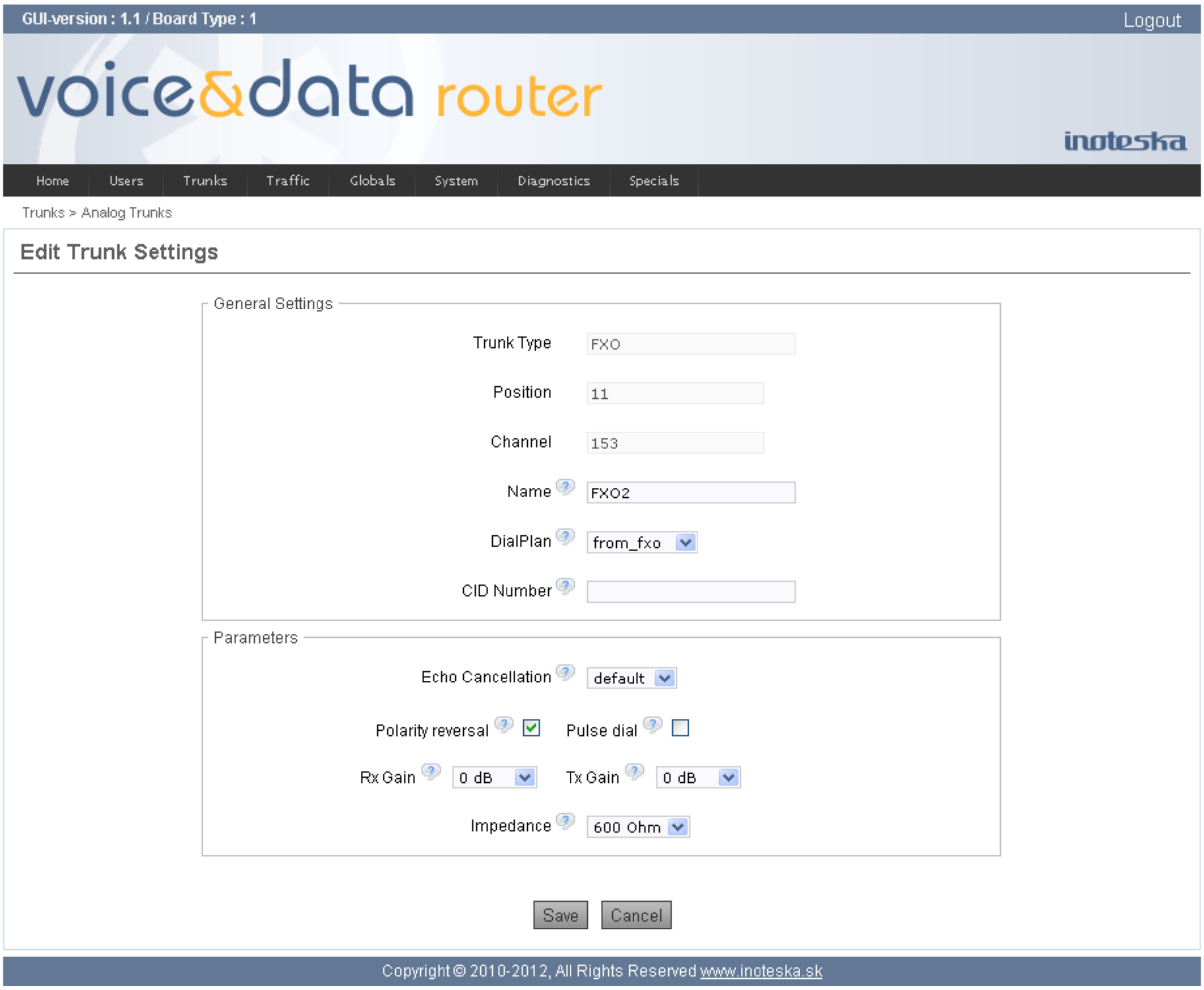
Position
This is analogue port position (number from 1 up to 32).

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 Traffic Menu 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.



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.

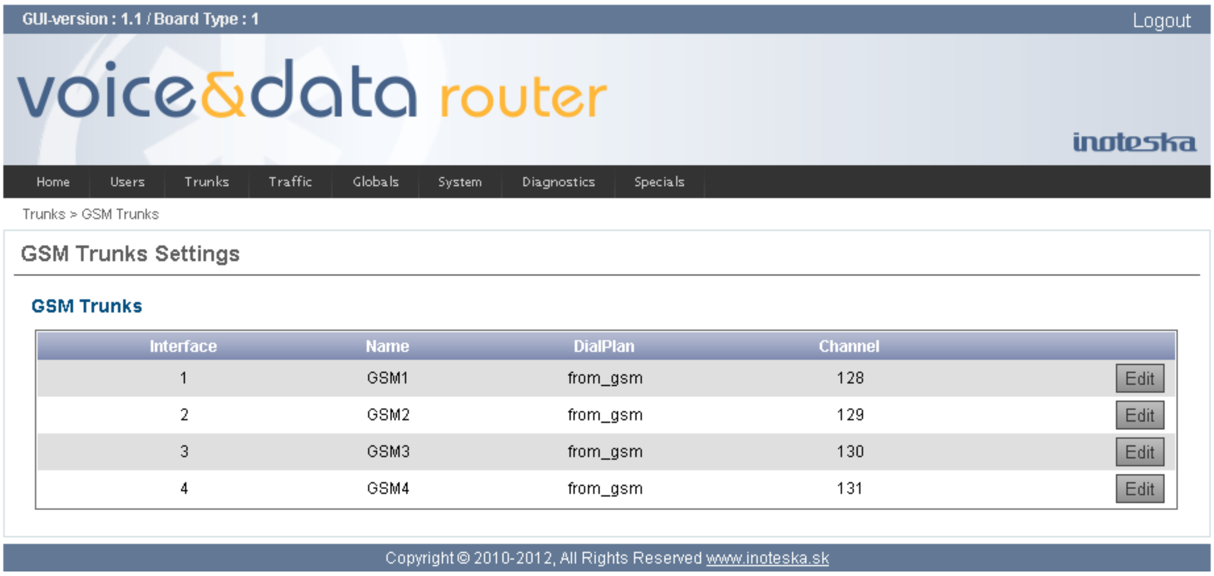
Rx Gain, Tx Gain
Use these parameters for line volume adjustment – transmit and receive gain or attenuation. Rx gain defines the level of voice received by Voice&Data Router from the FXO trunk and tx gain parameter defines the level of voice transmitted from Voice&Data Router to the FXO trunk. You can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

Impedance
FXO line AC termination impedance setting. Two modes are available

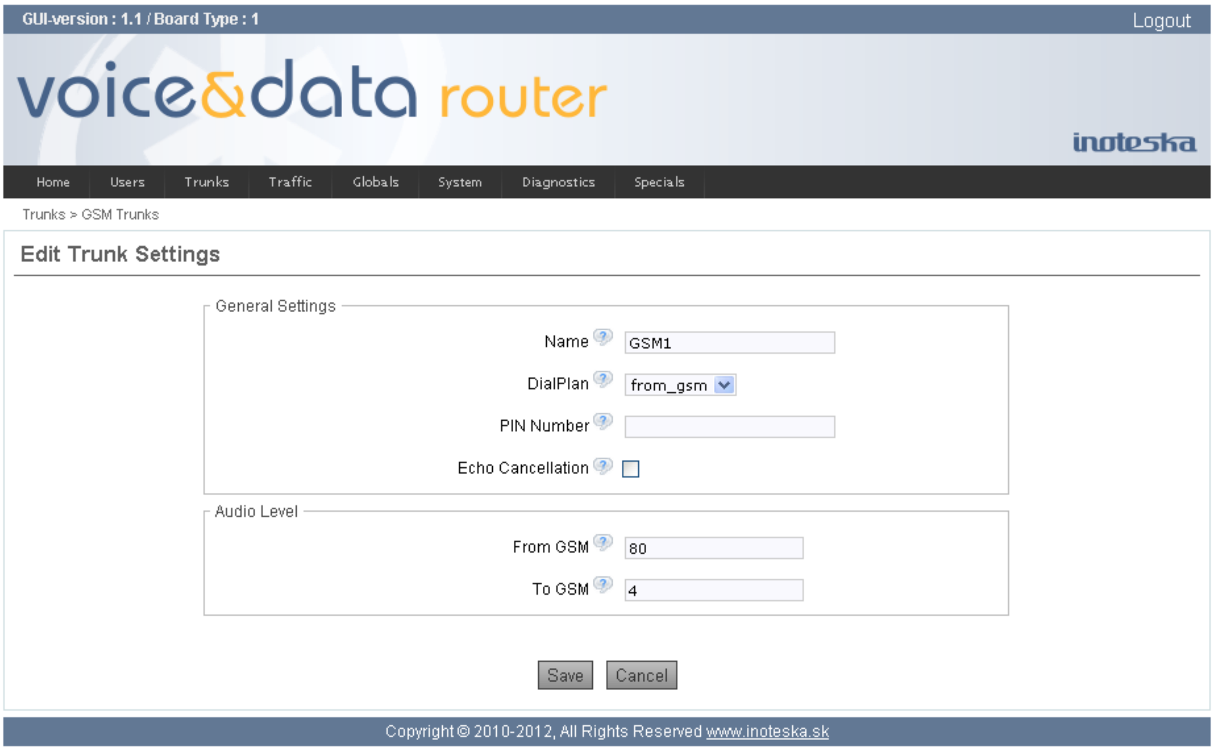
- 600 Ω
- Global Complex impedance

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.



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



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 Traffic Menu 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.

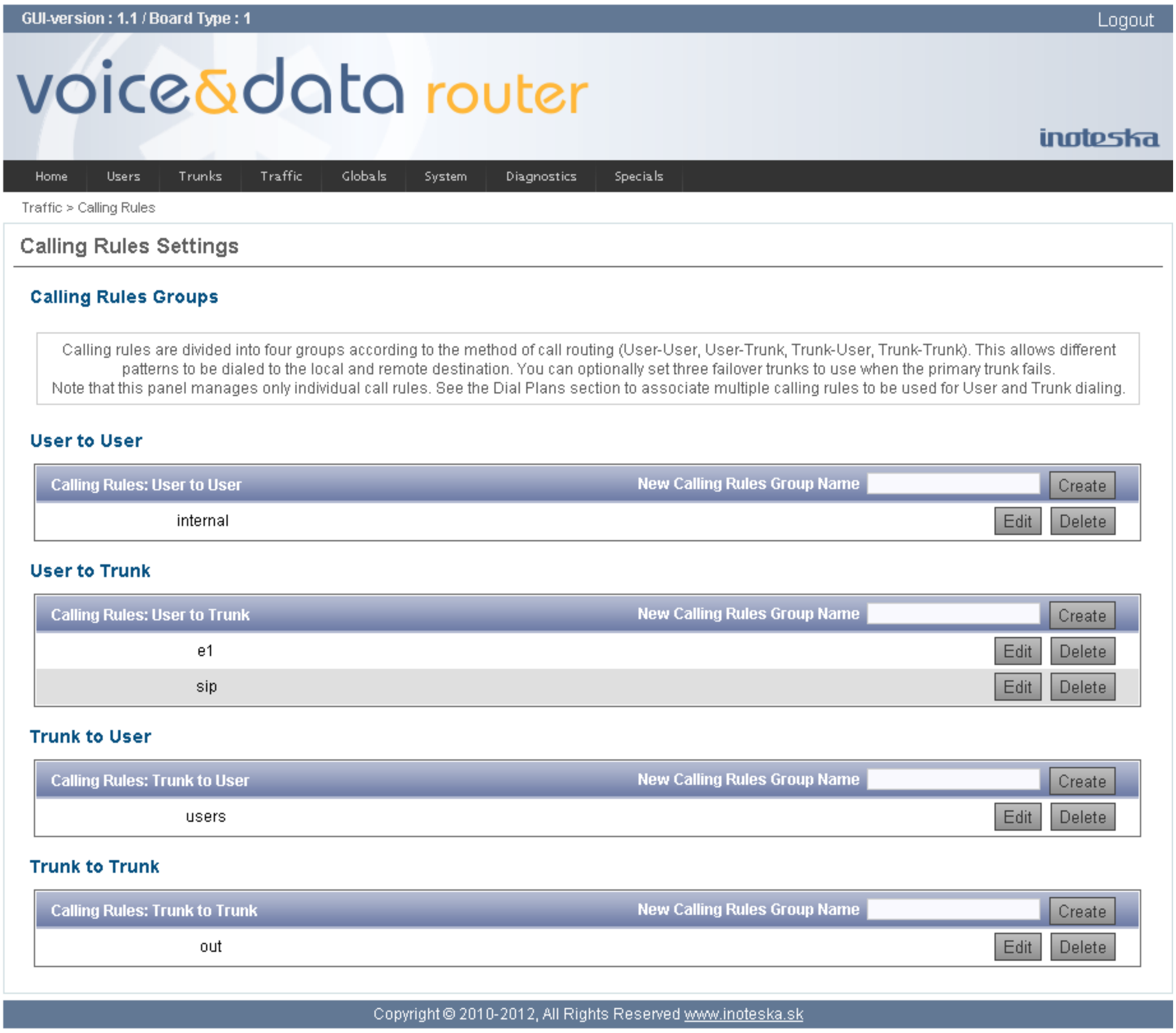
Configuration of call routing is two-level

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.

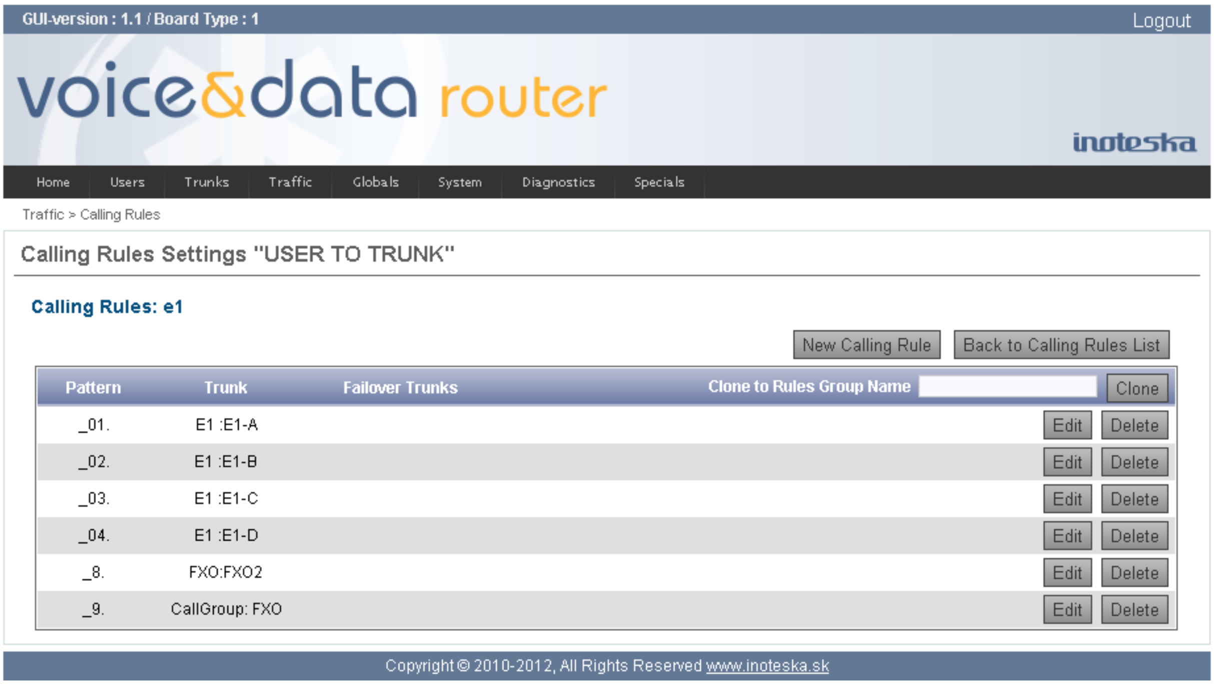


Calling rules are divided into four tables

- 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.



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: _NXXXXXX/_04455XXXXX). 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.

_NXXXXXX	matches normal 7 digit number.
_1NXXNXXXXXX	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!	same as above but matches 9011 too.
_#	matches a single # key press.
s	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.

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 Permissions menu for more details.

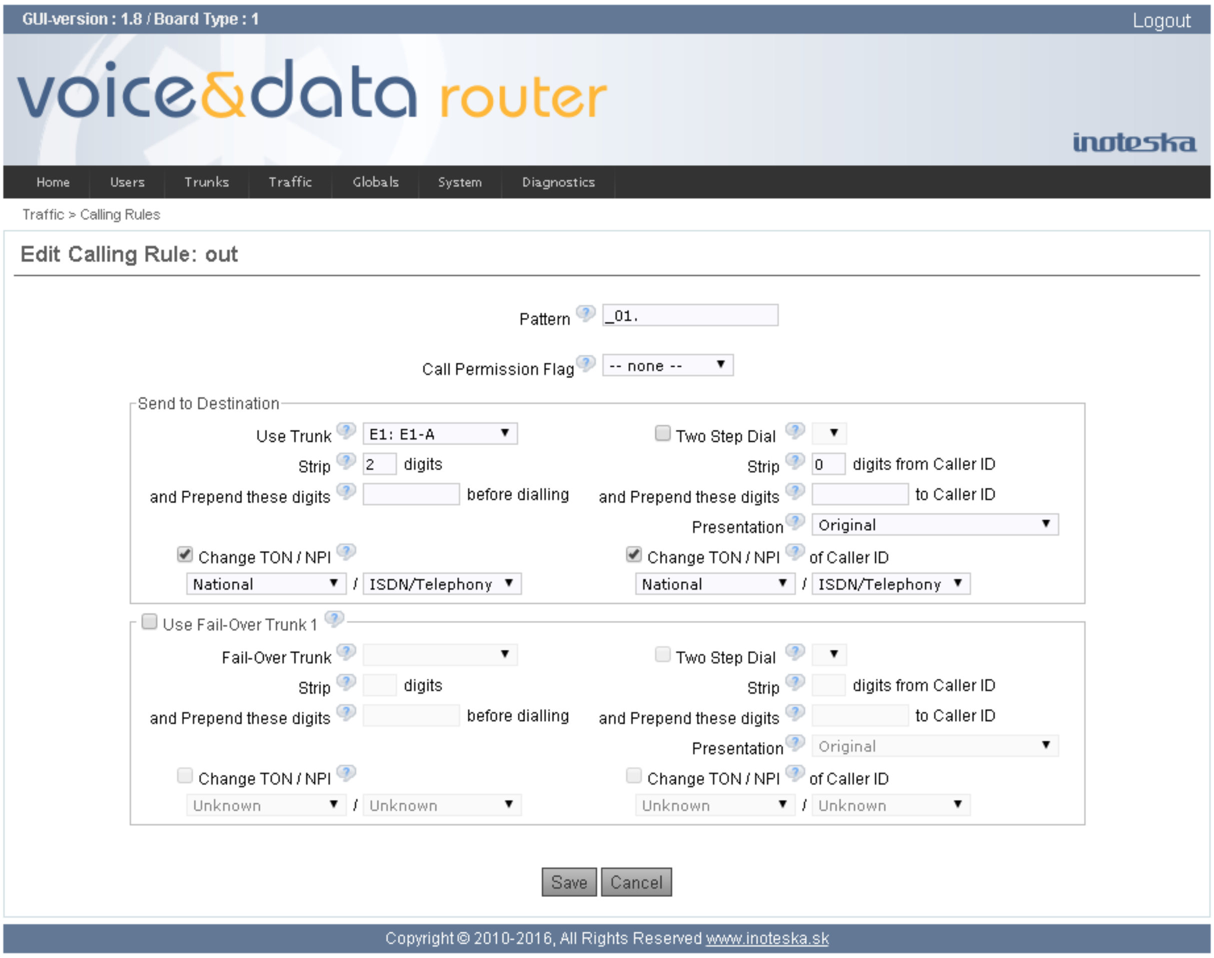
Send to Destination

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 Ring Groups 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.



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]XXXXXX/_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 Permissions 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 Call Groups 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 Carriers 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.

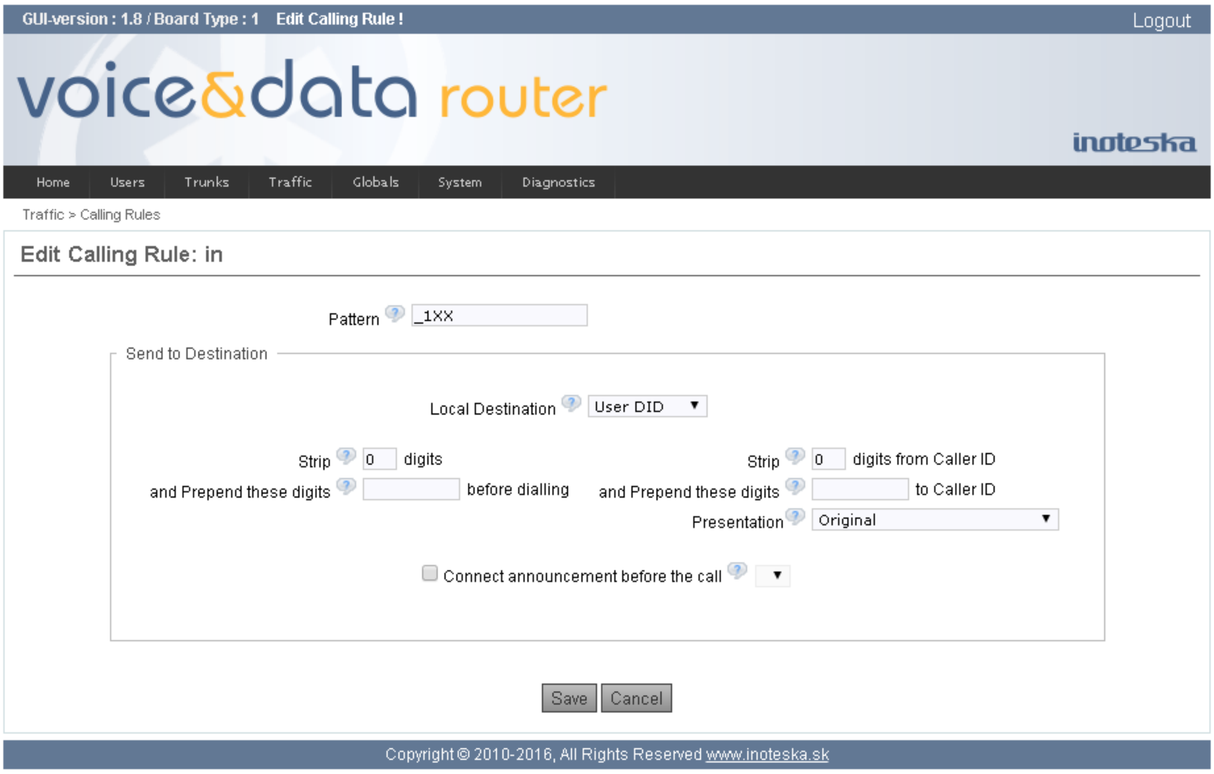
Change TON/NPI of Caller ID
Enable to overwrite Type of Number and Numbering Plan Identification of caller identification number before the call is placed on the selected outbound trunk. This option is applied only on trunk with ISDN signaling.

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.



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 _NXXXXXX/_04455XXXXX). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

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 Ring Groups 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 _NXXXXXX/_04455XXXXX). Only calls that match this pattern will apply this calling rule. See explanation above for details about calling rules pattern syntax.

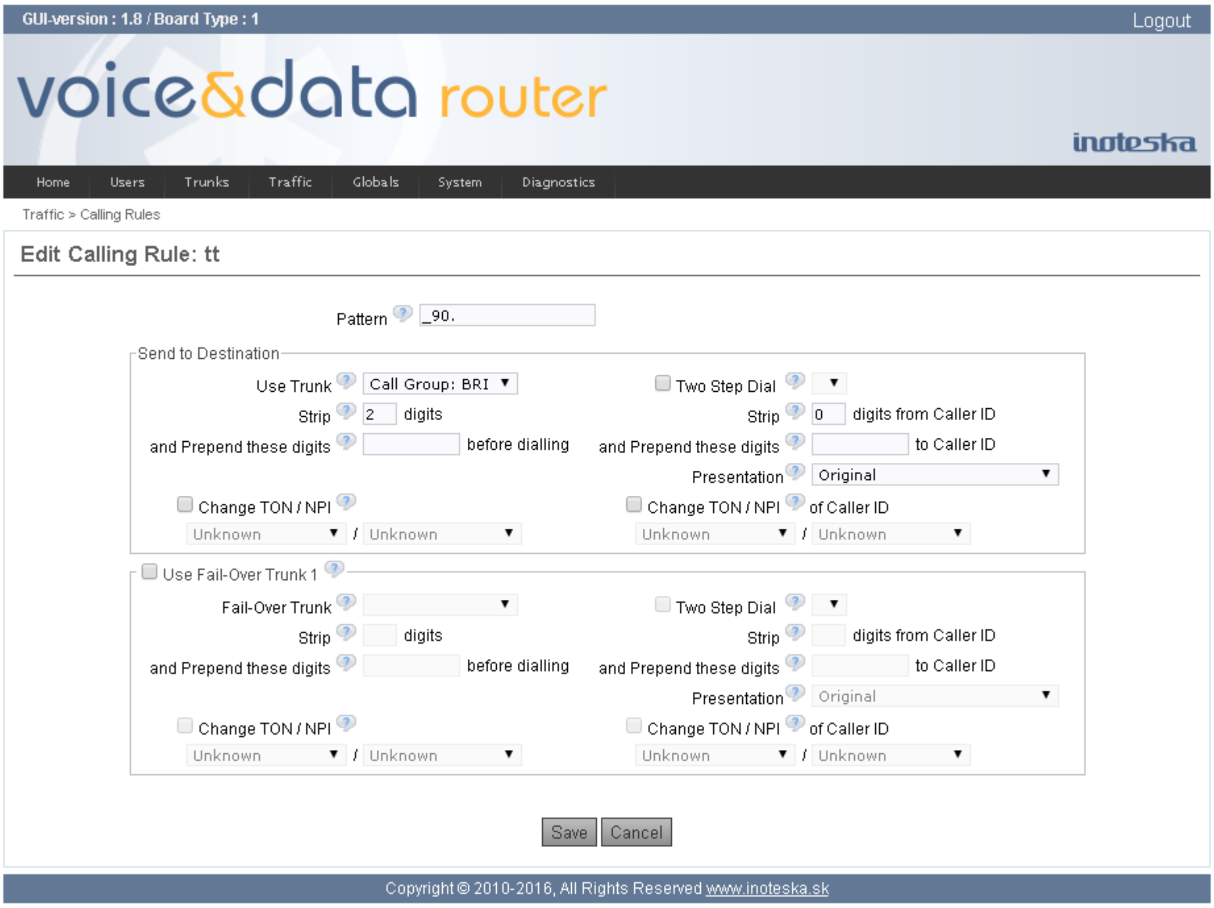
Send to Destination:

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 Call Groups 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 Carriers 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 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

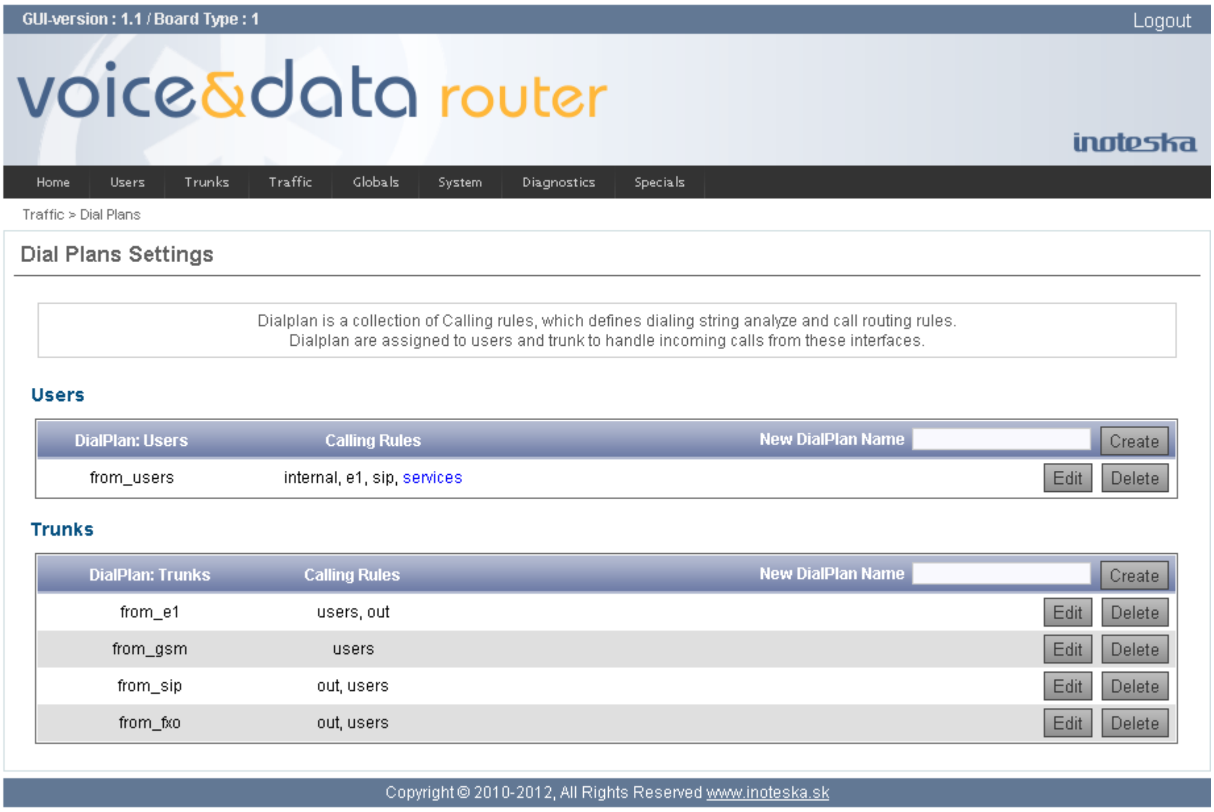
This section allows to define alternative routing of the call if primary destination is busy or down. It is possible to define maximally 3 fail-over alternatives. Each fail-over destination has same options as we define for primary call destination.

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.



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

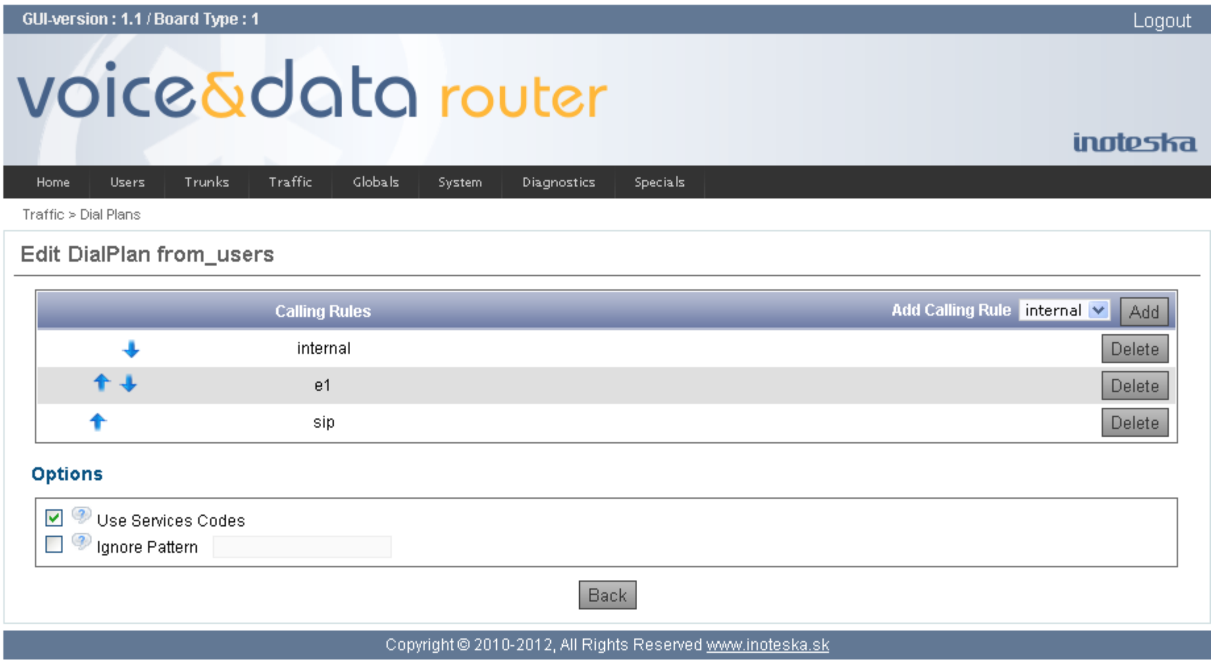
Users' dial plan consists of User to User and User to Trunk calling rules and it defines call routing from the user extensions. You can select existing calling rules group in the field **Add Calling Rule** and click on the **Add** button to add calling rules to the dial plan. Only calling rules applicable 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 Services Codes menu for details.



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.

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

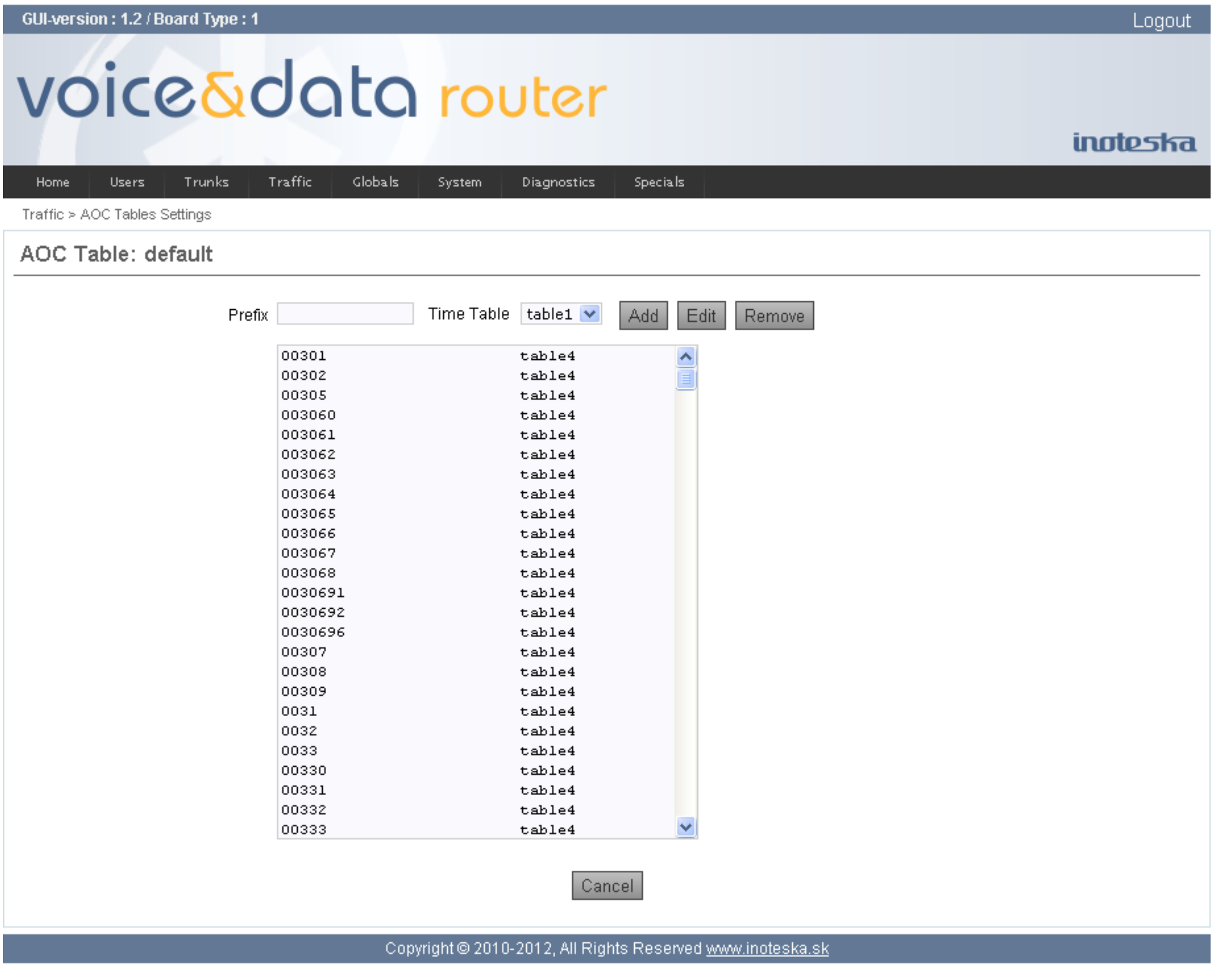
- AOC tables
- Time tables
- Holidays definition

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.

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.



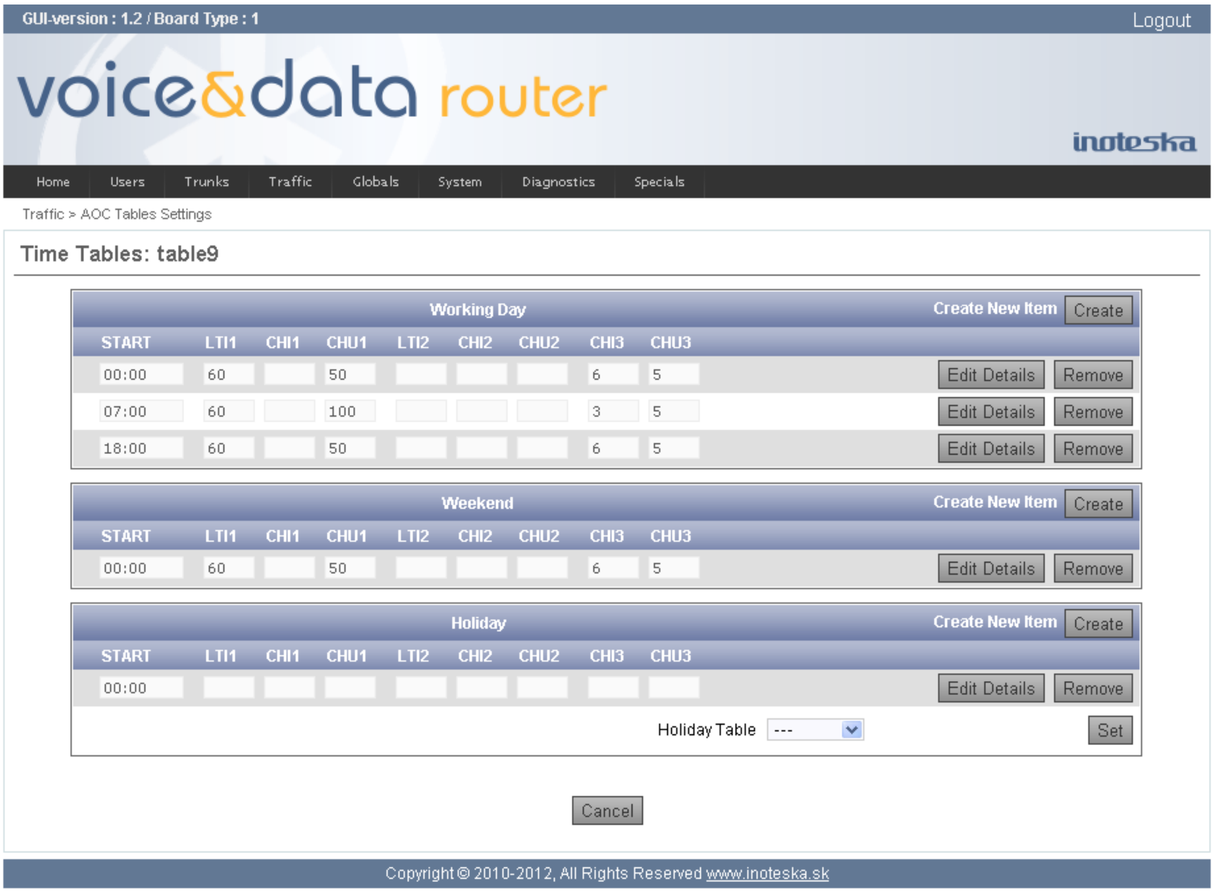
Time tables define charging rules – frequency and number of charging units. Time table can divide the day into zones with different charging rules. Each time table defines charging rules separately for working day, weekend and public holiday. Charging rules allow setting of three time intervals for each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

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.



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

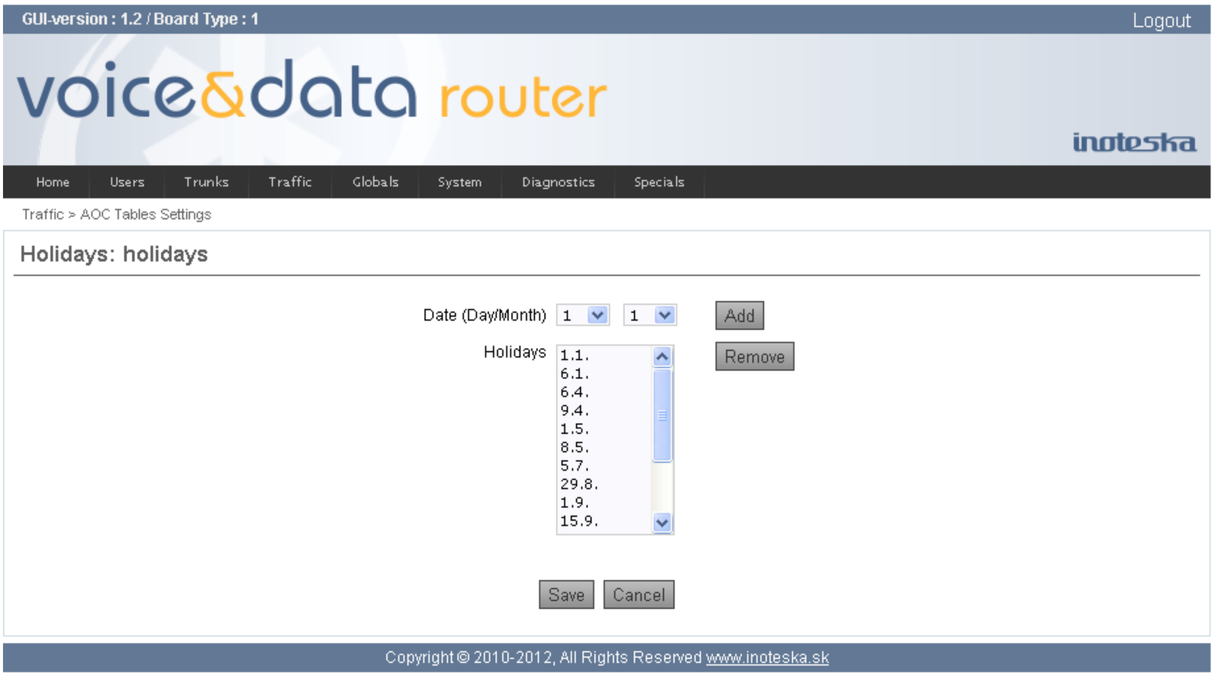
This setting is more complex. Within hours 7:00 – 19:00 you will get 5 charging units immediately after call answer then 2 charging units per second during the first minute of the call and 1 charging unit every 2 seconds during the rest of the call. At the time 19:00 – 7:00 you will get 1 charging unit 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.

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.



Note that correct functionality of generation of charging information depends on setting of Voice&Data Router system time. See Date and Time 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

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.

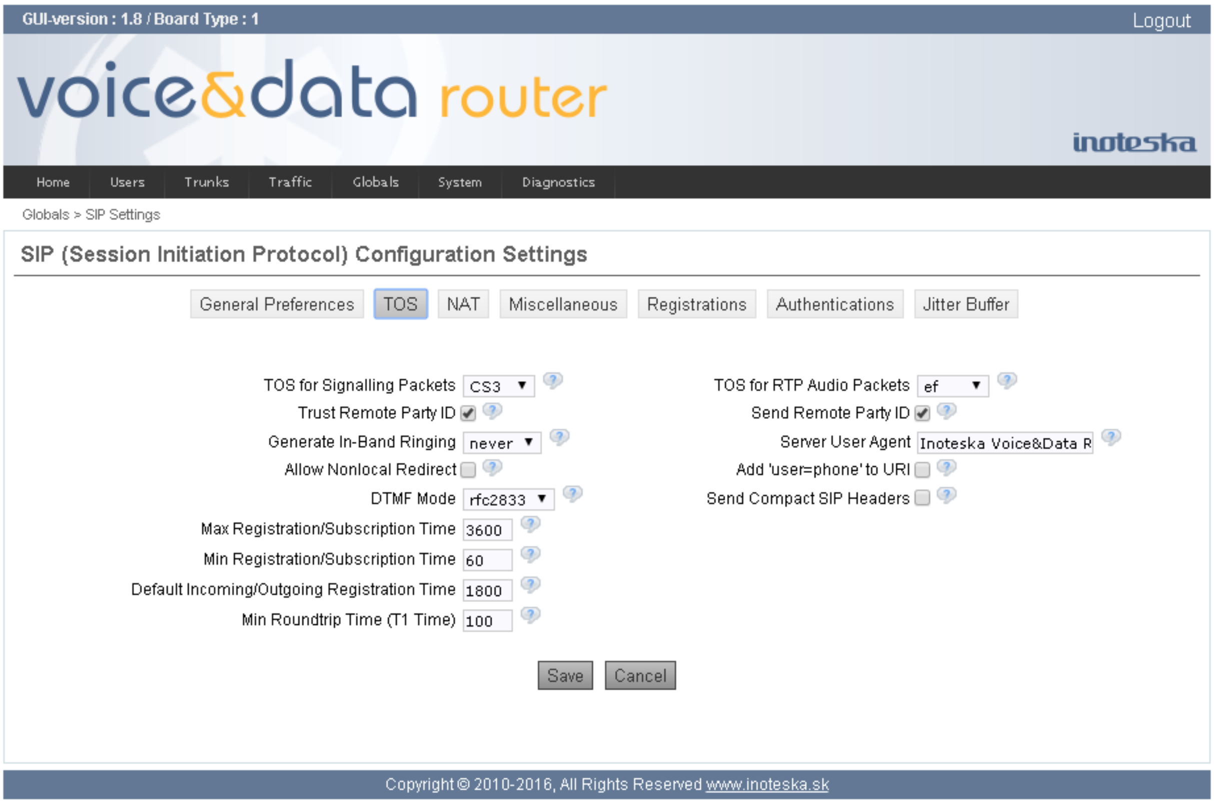
Pedantic
Enable slow, pedantic checking of Call-IDs, multiline SIP headers and URI-encoded headers. This option enables more strict SIP RFC compliancy.

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



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).

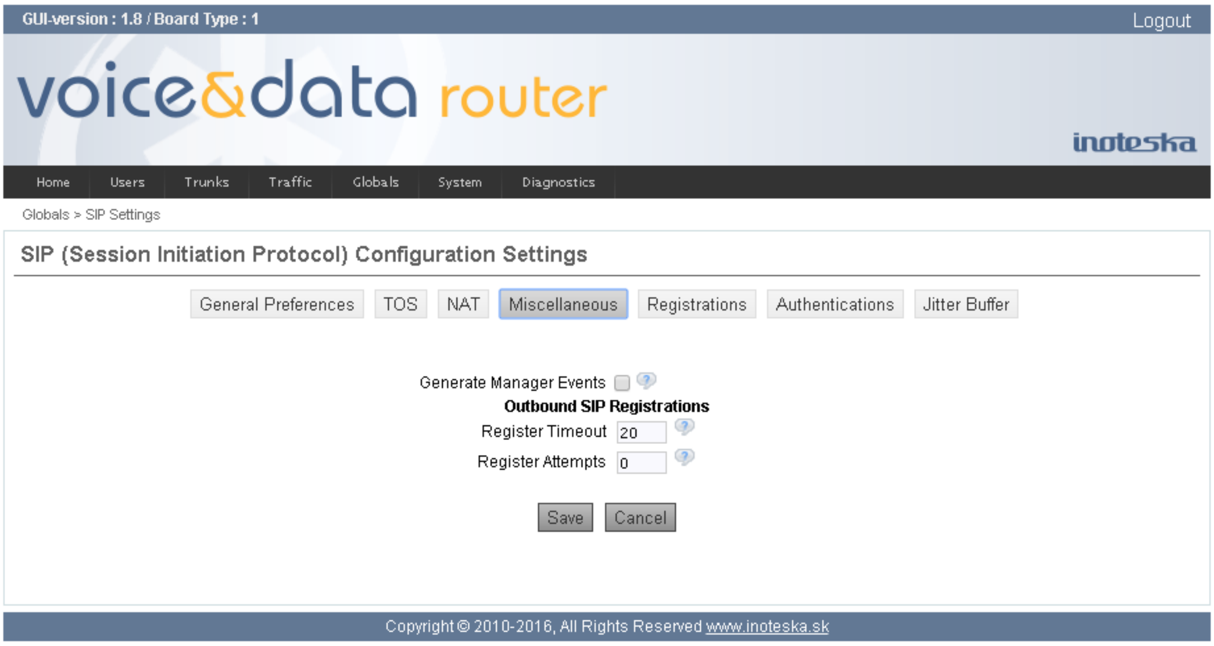
NAT Mode
Select default behavior when Voice&Data is on a public IP and communicating with devices hidden behind a NAT.

- **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

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



Generate Manager Events
Enable/disable generation of Asterisk Manager events.

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

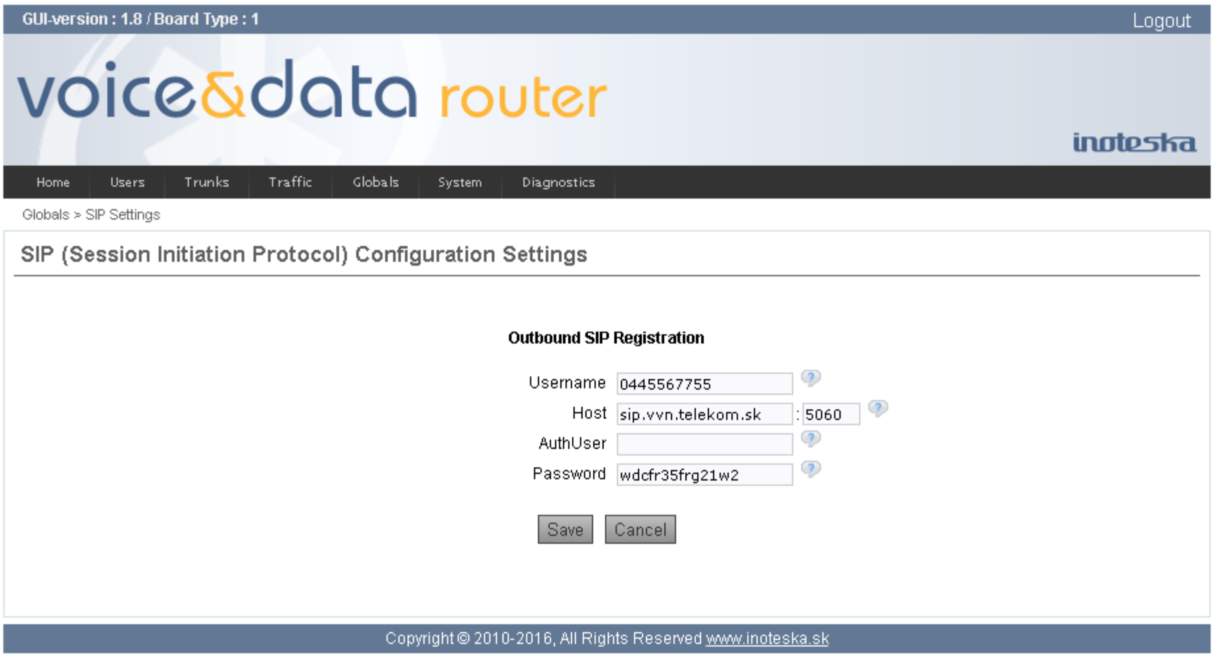
Register Attempts
Number of outbound registration attempts before giving up. Setting to 0 means that Voice&Data Router will retry indefinitely.

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).

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.

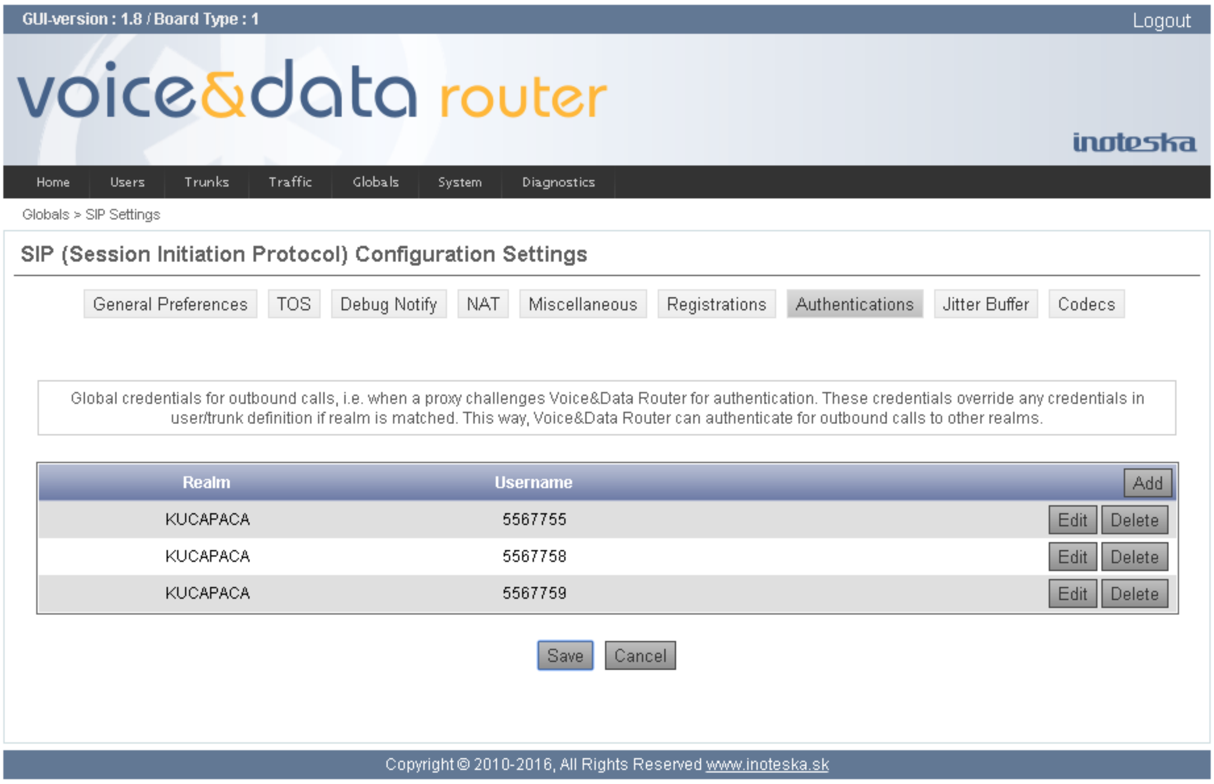
Host
SIP registrar server address (IP address or host name)

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

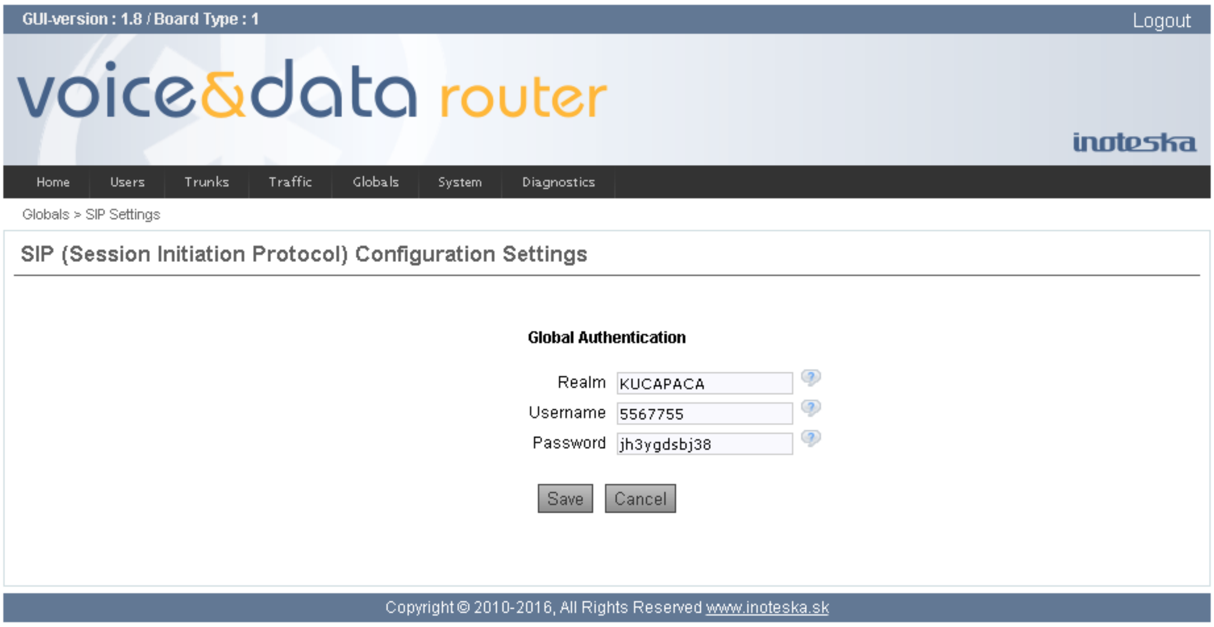
Password
Password used for the authentication.

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.



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.

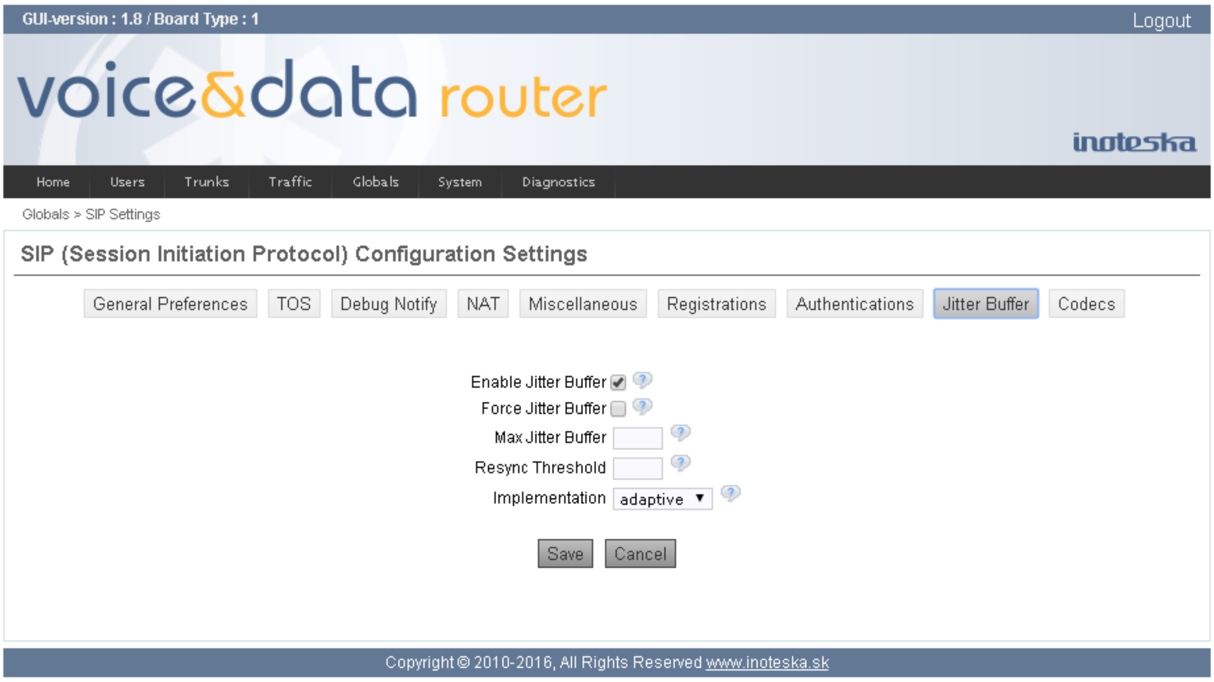


Realm
Authentication realm. If it is matched to realm in proxy challenge, these authentication credentials override any credentials in user/trunk definition.

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).

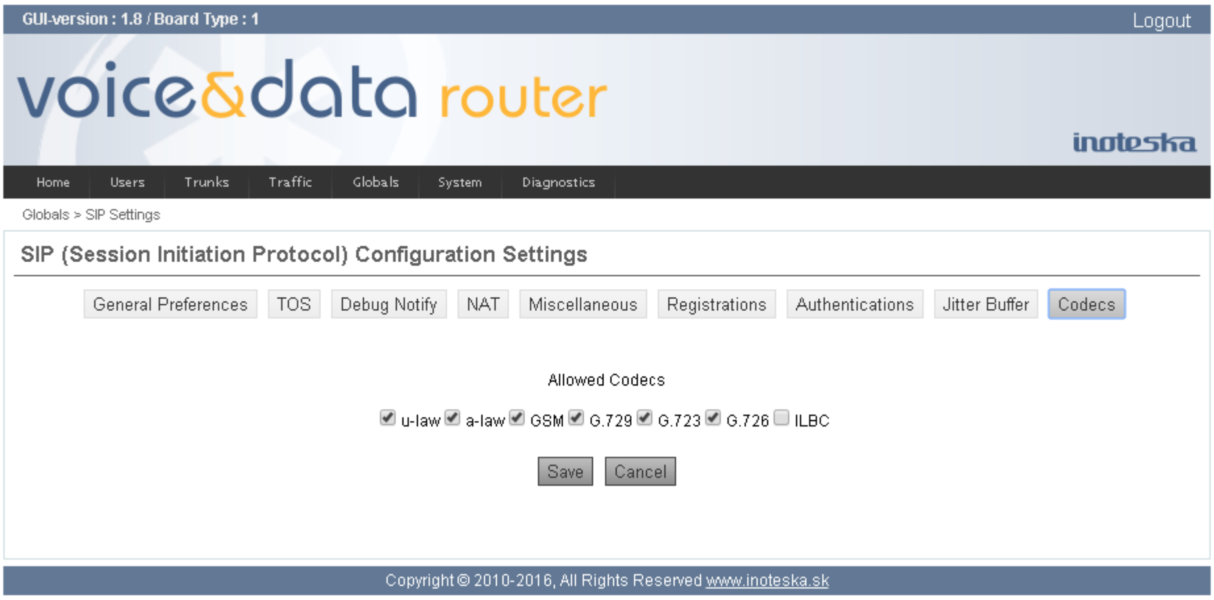
Resync Threshold
Jump in the frame timestamps over which the jitter buffer is resynchronized. It is useful to improve the quality of the voice, with big jumps in/broken timestamps (in milliseconds).

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.


GUI-version : 1.8 / Board Type : 1
Logout
voice&data router
inoteska
Home
Users
Trunks
Traffic
Globals
System
Diagnostics
Globals > SIP Settings
SIP (Session Initiation Protocol) Configuration Settings
General Preferences
TOS
Debug Notify
NAT
Miscellaneous
Registrations
Authentications
Jitter Buffer
Codecs
Allowed Codecs
u-law a-law GSM G.729 G.723 G.726 ILBC
Save
Cancel
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RTP/UDPTL Settings

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

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

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.

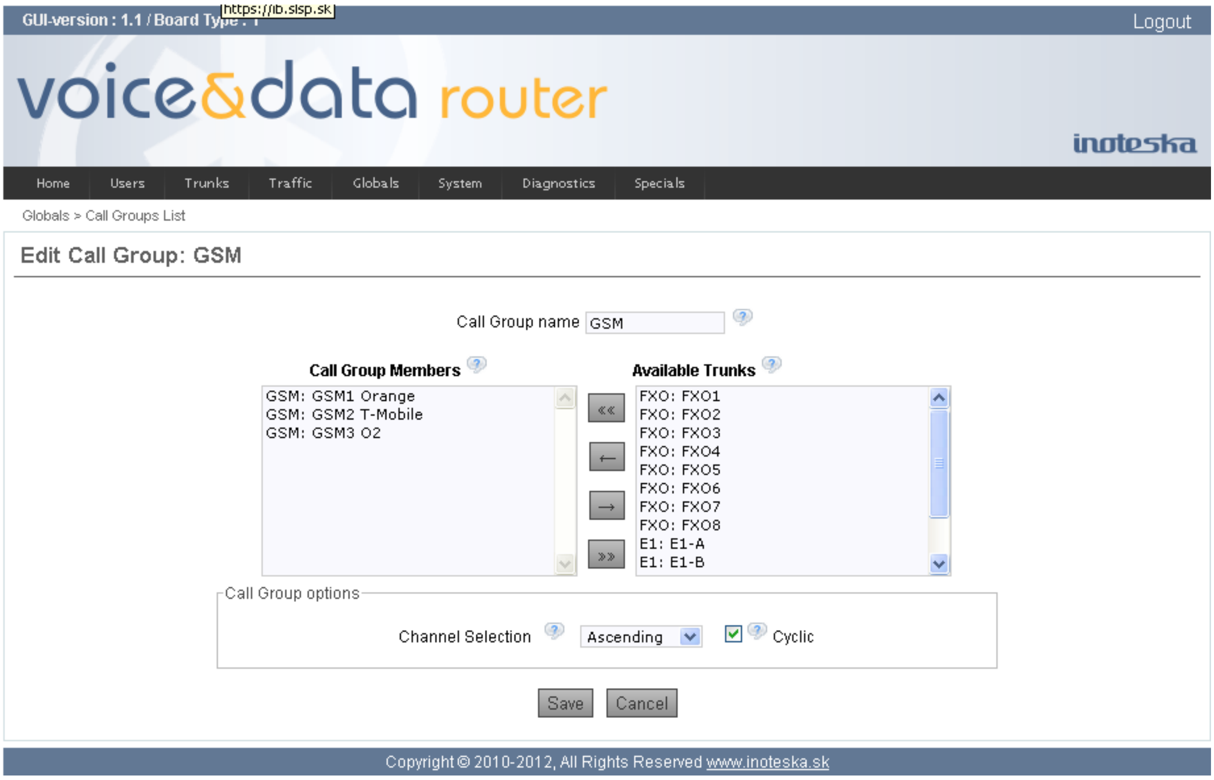
Available Trunks
List of available TDM trunks, which can be added to the call group.

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.


GUI-version : 1.1 / Board Type : 1
https://ib.sisp.sk
Logout
voice&data 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
GSM: GSM2 T-Mobile
GSM: GSM3 O2
FXO: FXO1
FXO: FXO2
FXO: FXO3
FXO: FXO4
FXO: FXO5
FXO: FXO6
FXO: FXO7
FXO: FXO8
E1: E1-A
E1: E1-B
Call Group options
Channel Selection
Ascending
Cyclic
Save
Cancel
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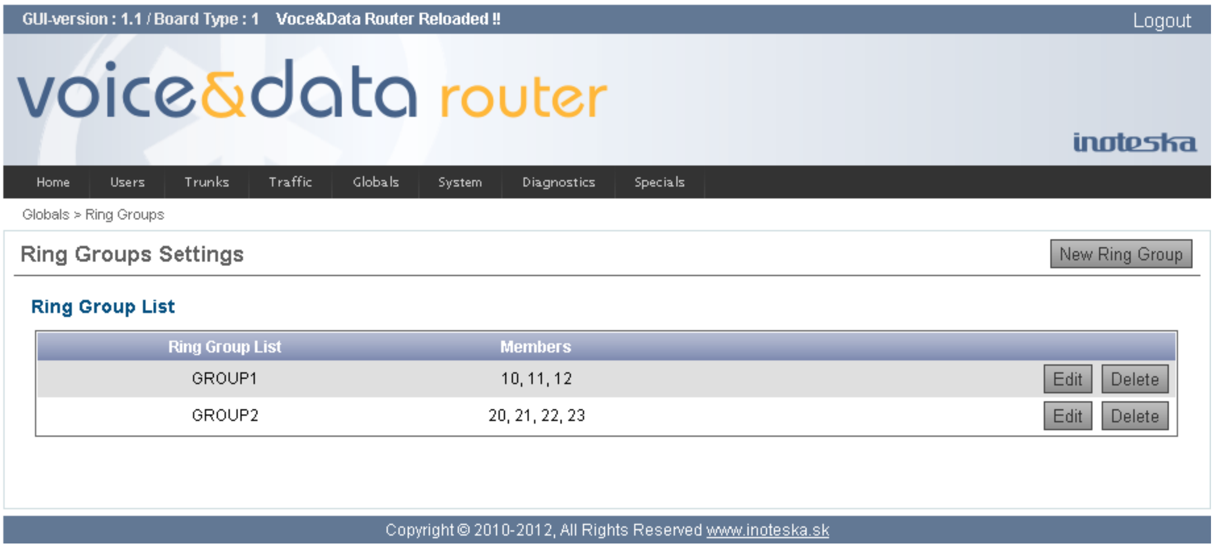
Ring Groups

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.



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.

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.

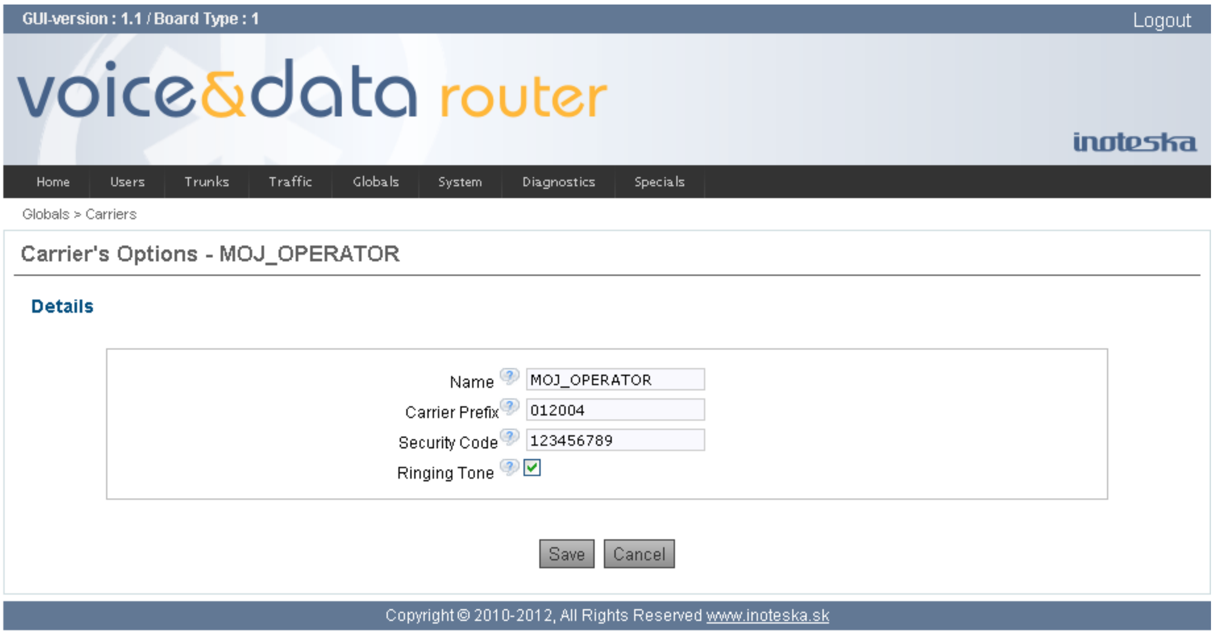
If Not Answered Goto
Define what to do when nobody answers.

- 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. Two-stage 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.

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.



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.

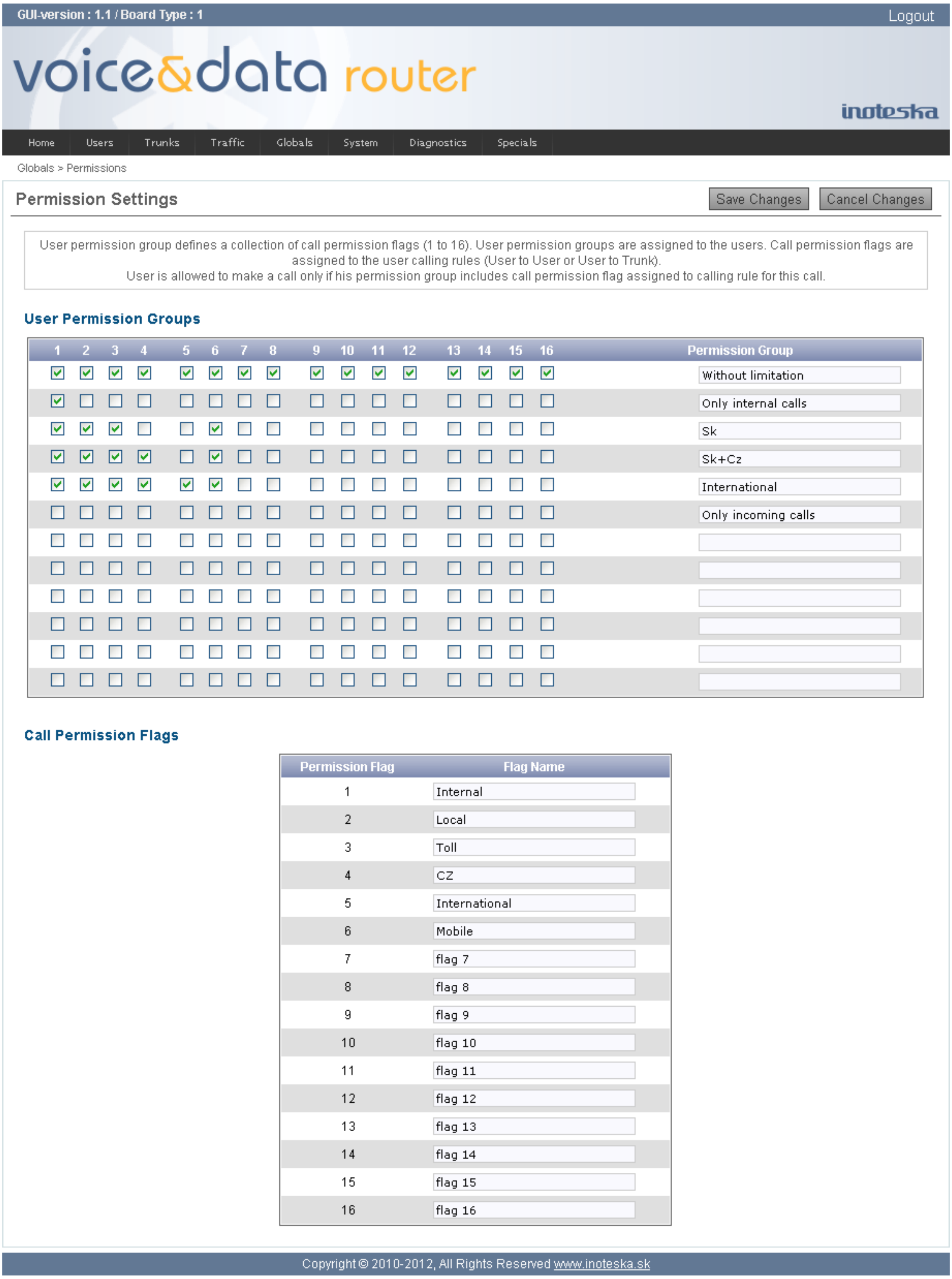
Carrier Prefix
Carrier access number.

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.



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.

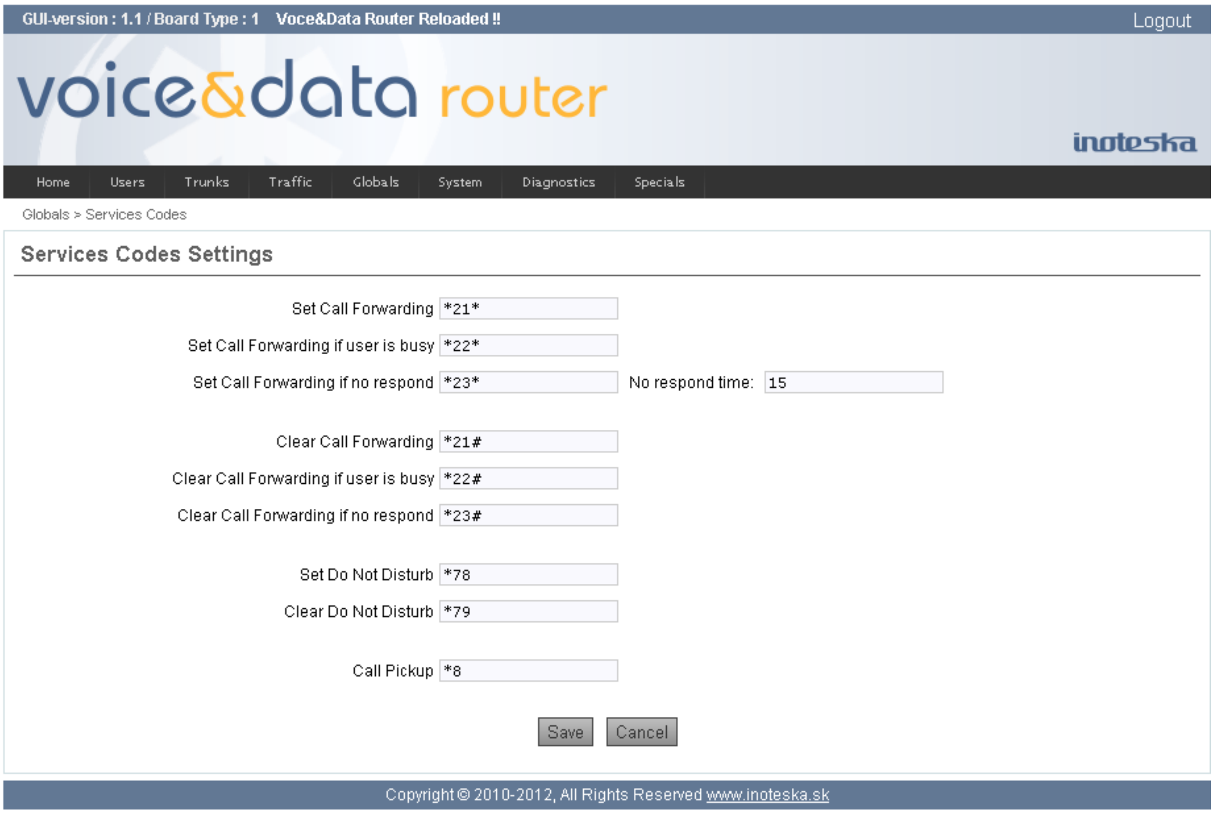
On the other hand user permission group that does not have selected any call permission flag can be limited only for incoming calls. In that case all outgoing calls matching the calling rules with assigned call permission flag will be refused. Only calling rules which don’t use call permission flags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



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.

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.

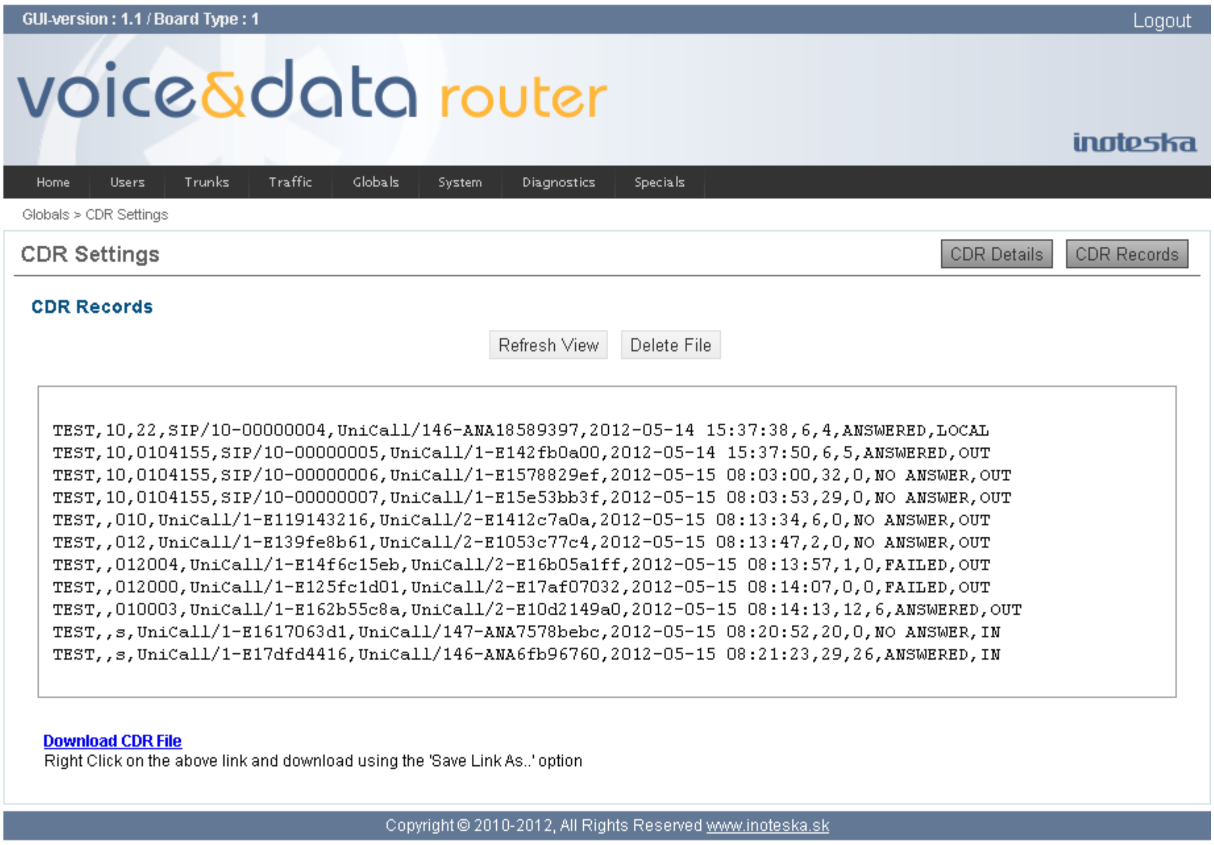
CDR Trunk to User
Enable to log CDR for Voice&Data Router incoming calls.

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)



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

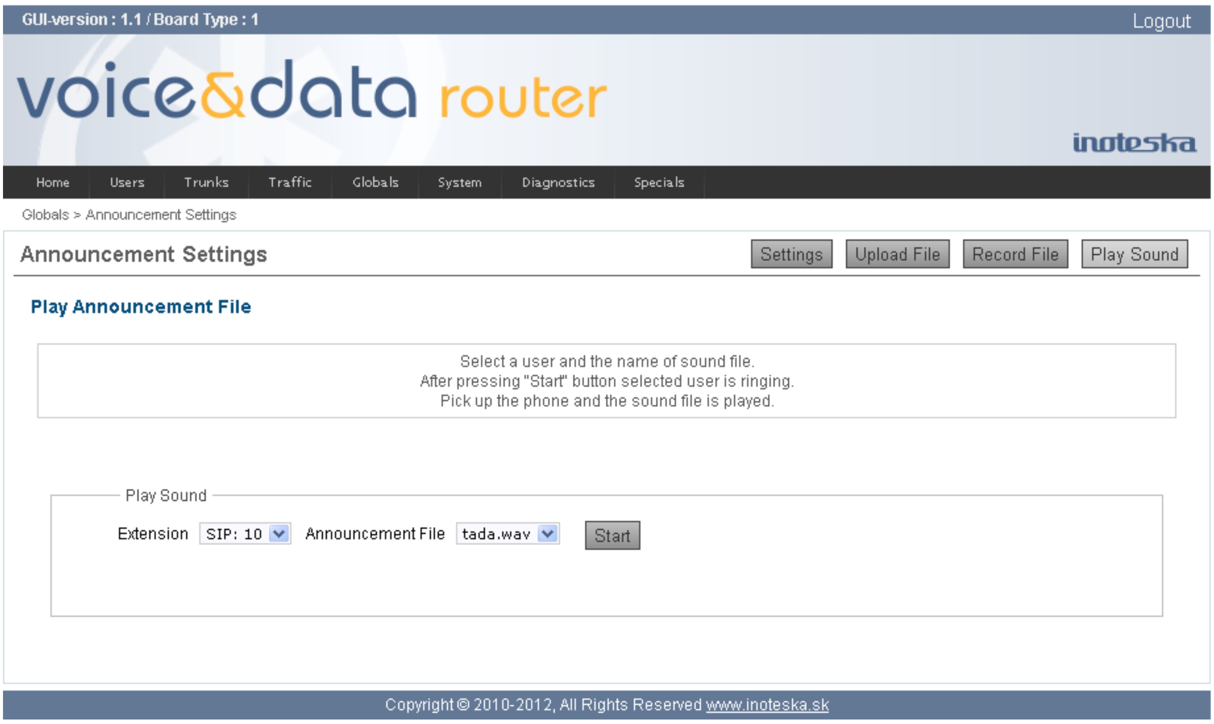
Sample rate	8 kHz	Sample rate	8 kHz
Sample size	16 bit	Sample size	8 bit
Channels	1 (mono)	Channels	1 (mono)
Auto format	PCM	Audio format	CCITT A-Law CCITT µ-Law

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.

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.

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.



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.

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 16th 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.

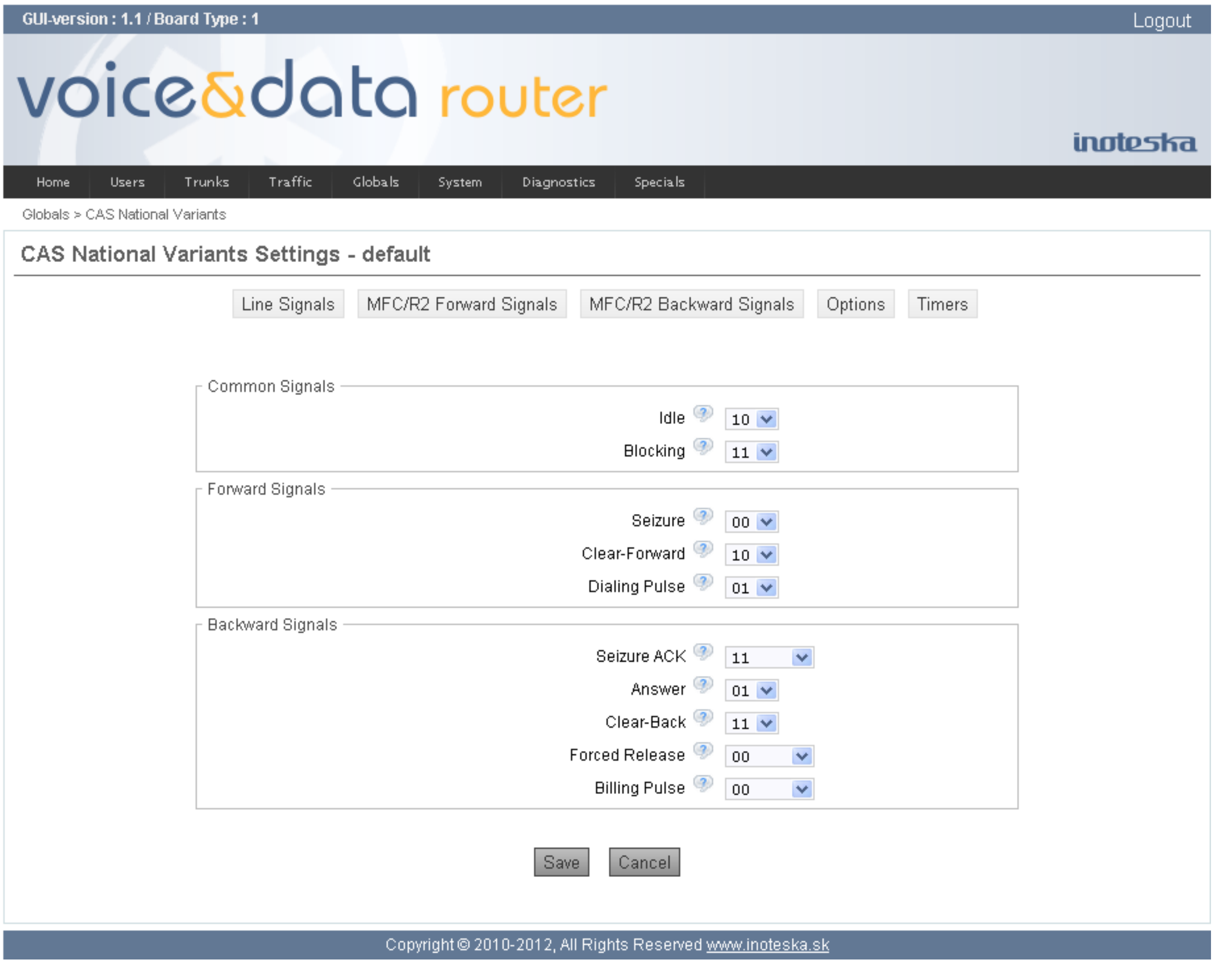
Forward Signals

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.



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.

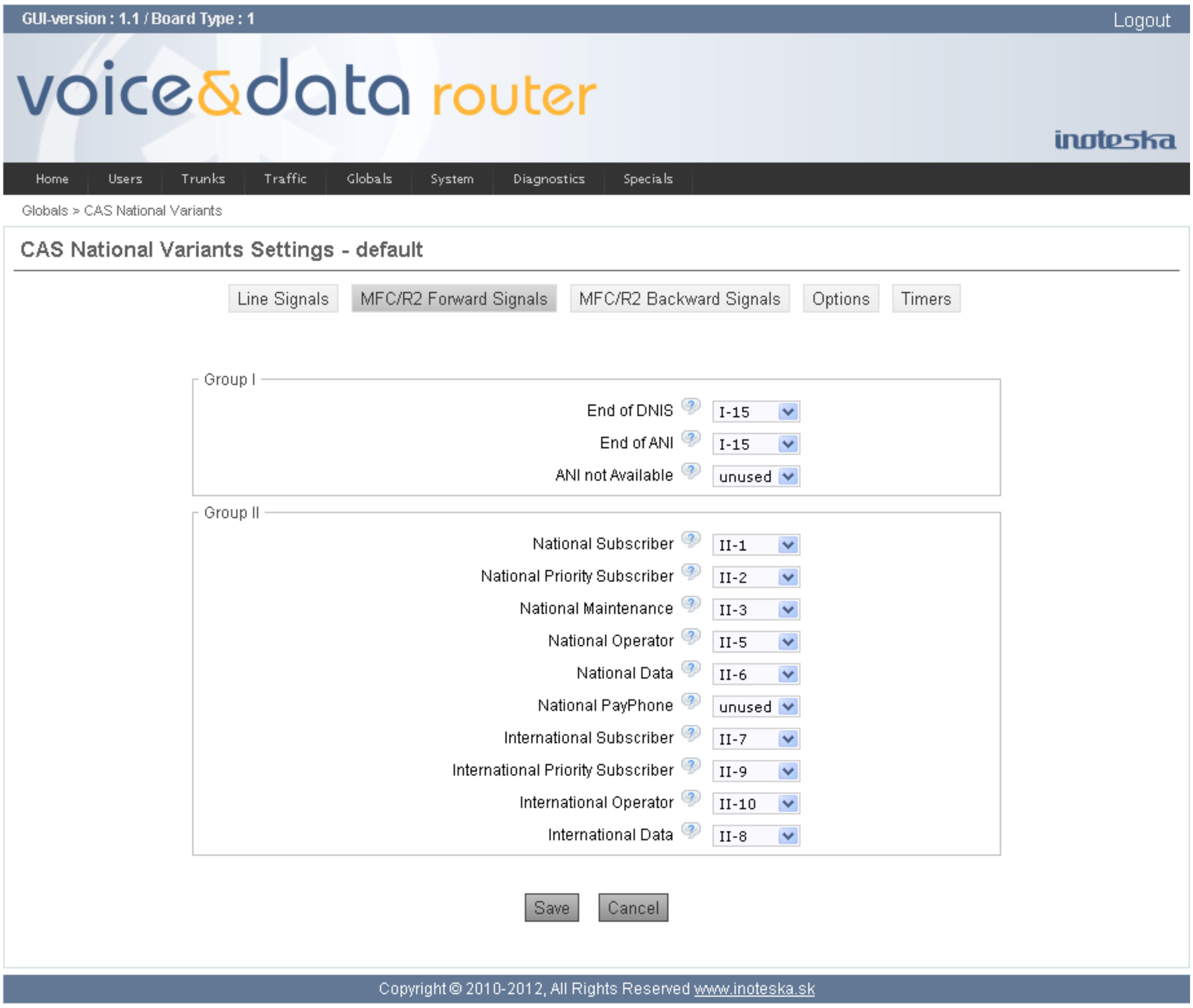
Group I

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.



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.

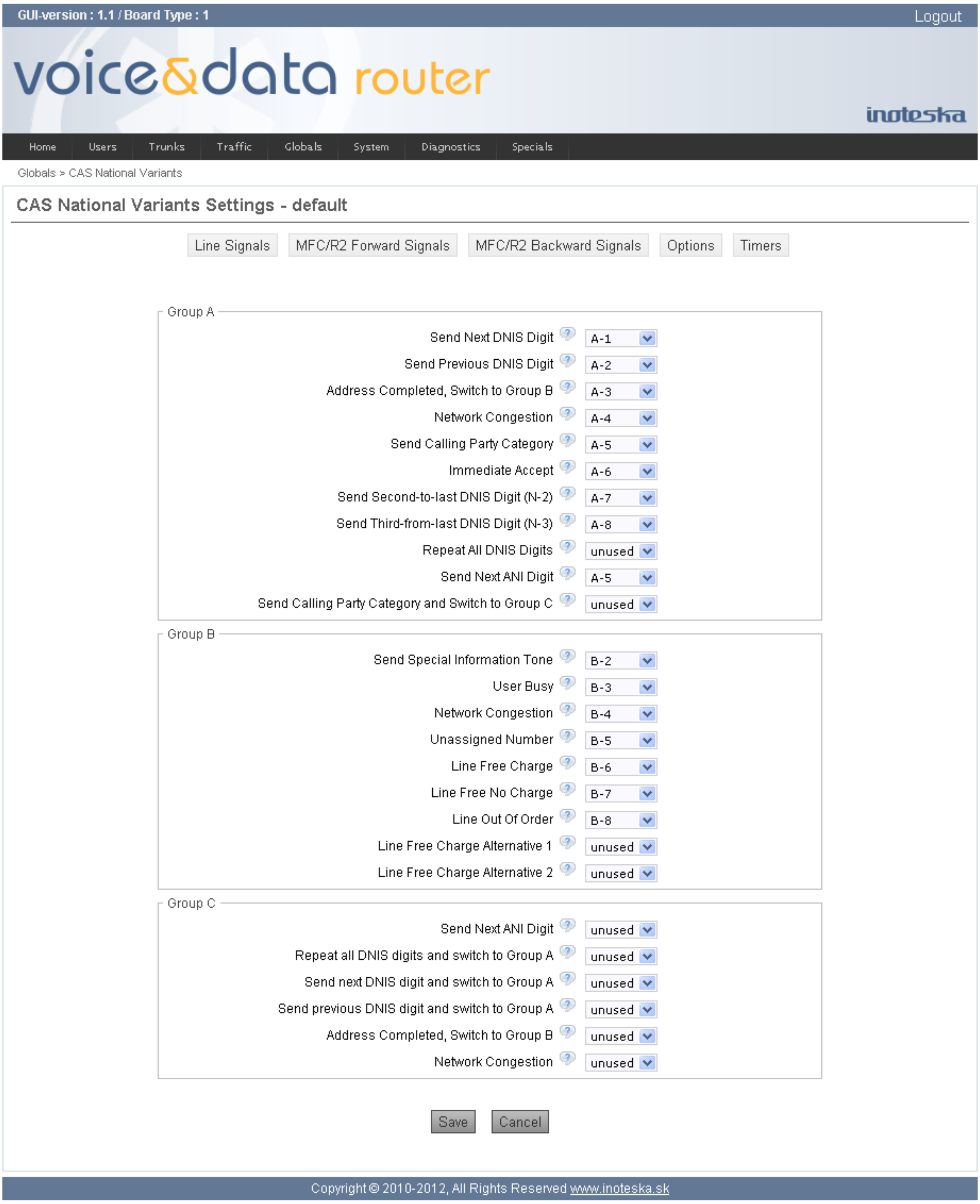
National Data
Calling party is data transmission equipment.

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.



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.

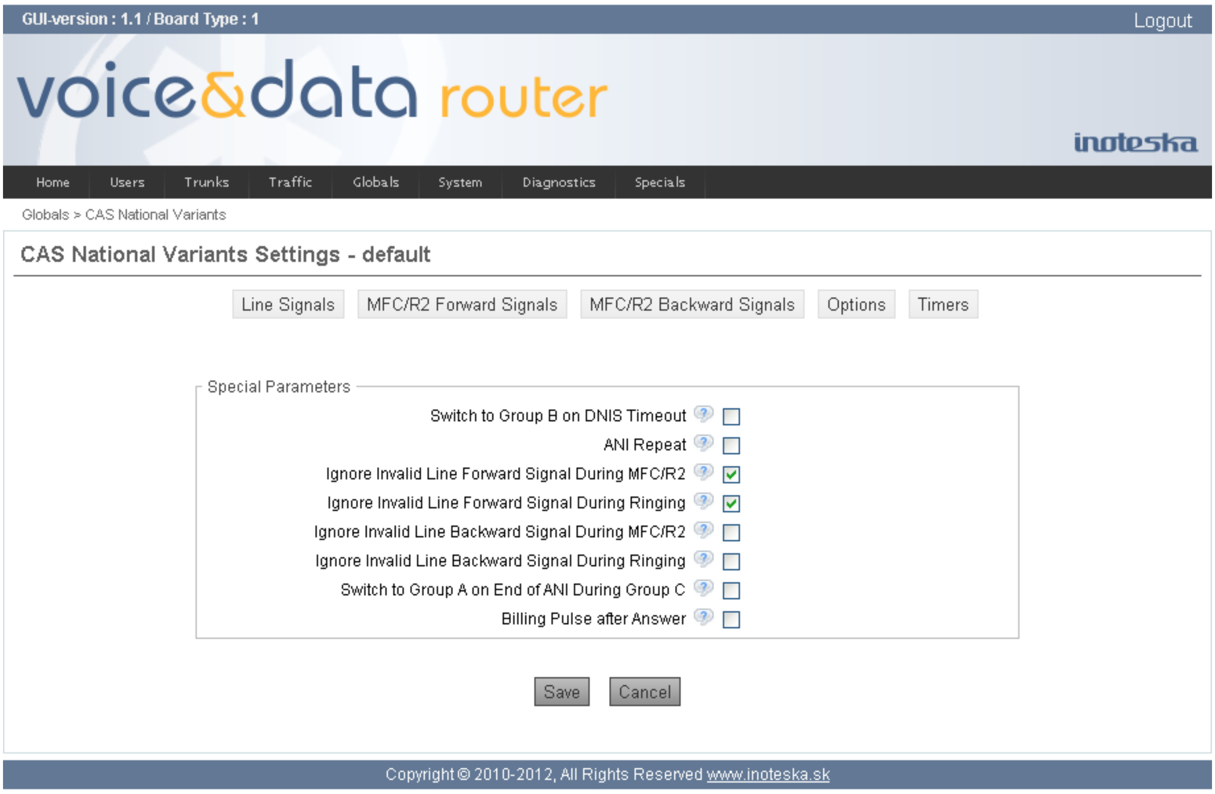
Send Previous DNIS Digit and Switch to Group A
Request to repeat last DNIS digit and switch over to reception of group A signals.

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



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.

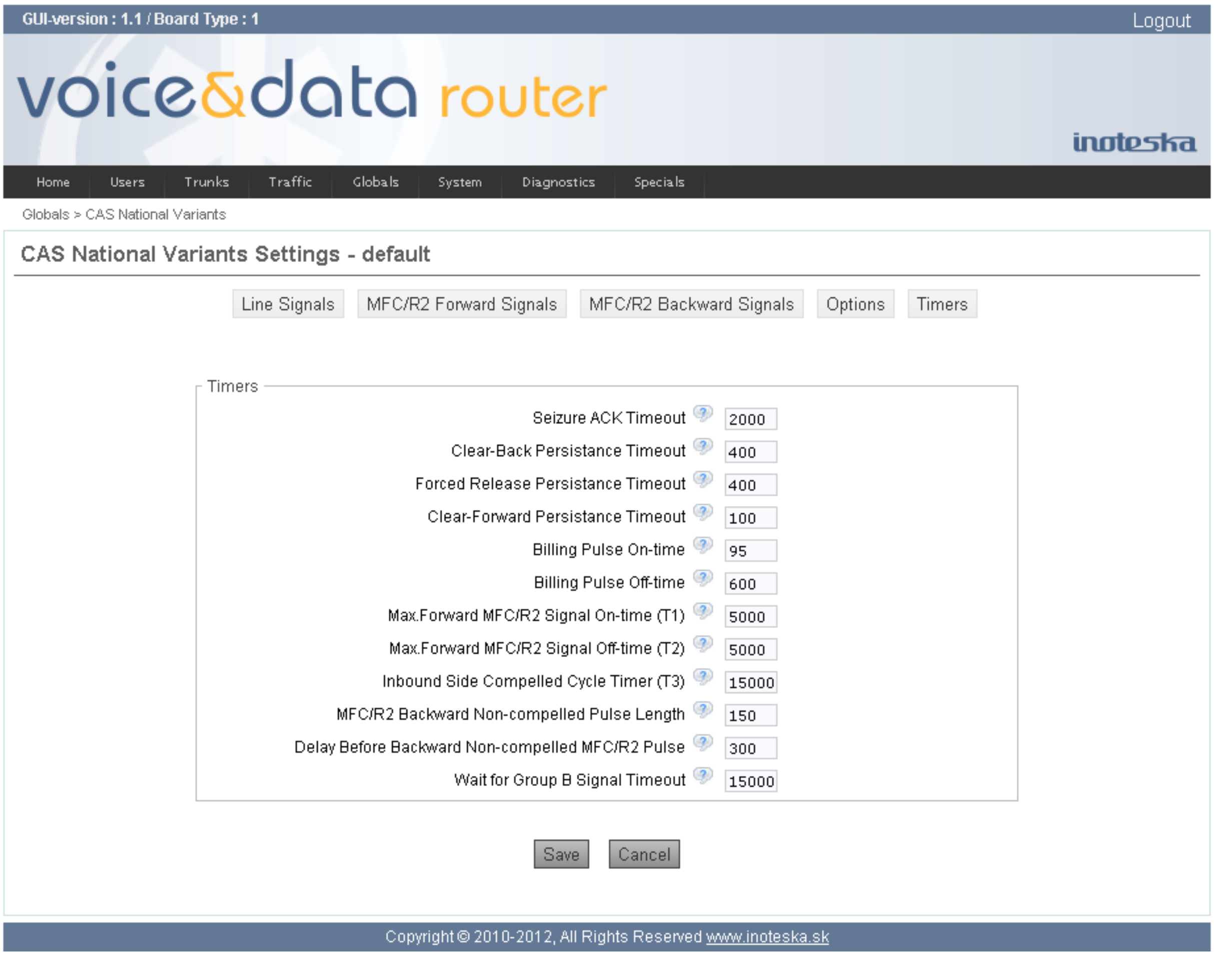
Ignore Invalid Line Forward Signal during Ringing
Ignore Invalid Line Backward Signal during Ringing
When enabled then invalid (unexpected) line forward/backward signals during ringing phase (between end of MFC/R2 register signaling phase and the call answer) are ignored.

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



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.

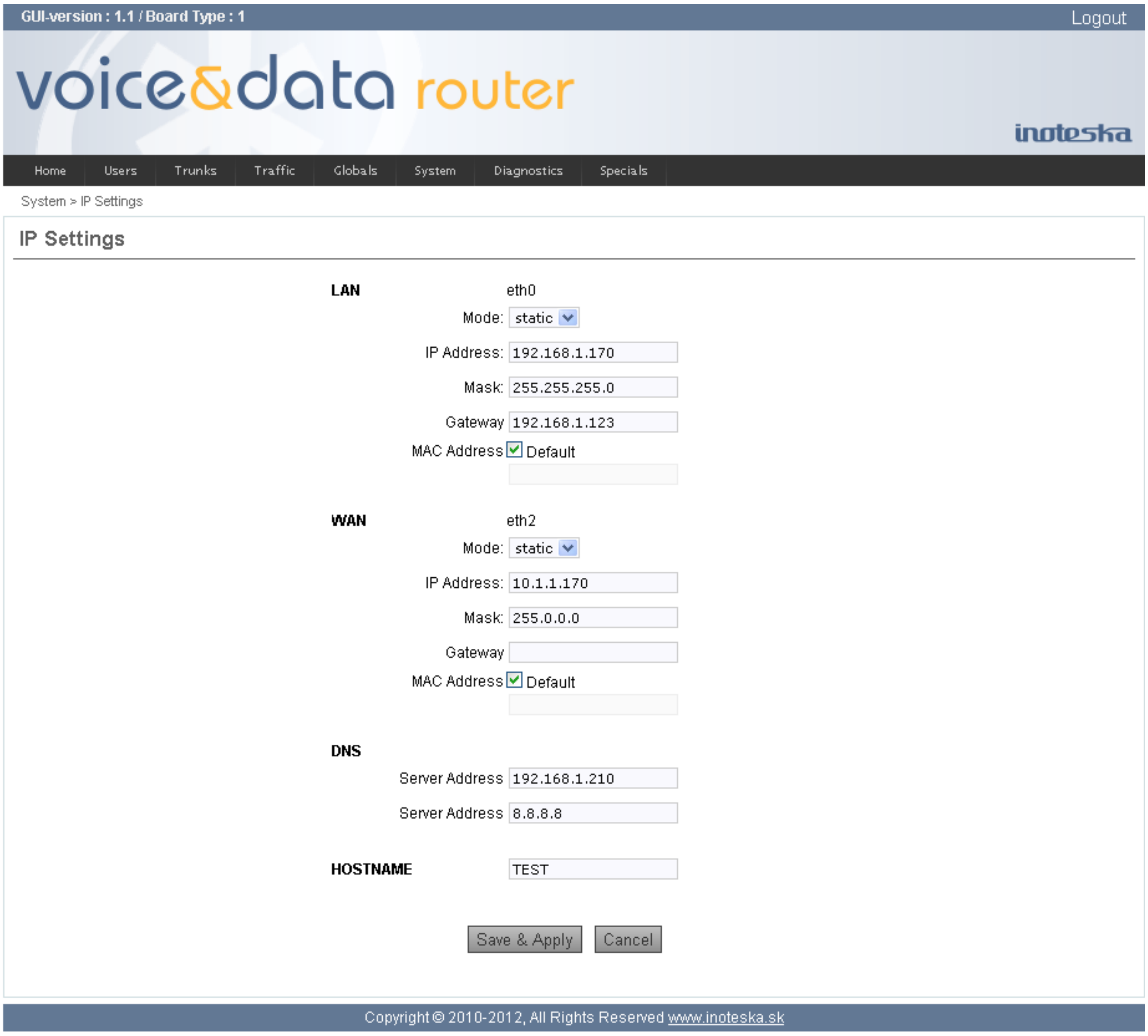
Wait for Group B Signal Timeout
Maximal time of waiting for MFC/R2 Group B signal after provision of the complete addressing information – call acceptance/failure indication.

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



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).

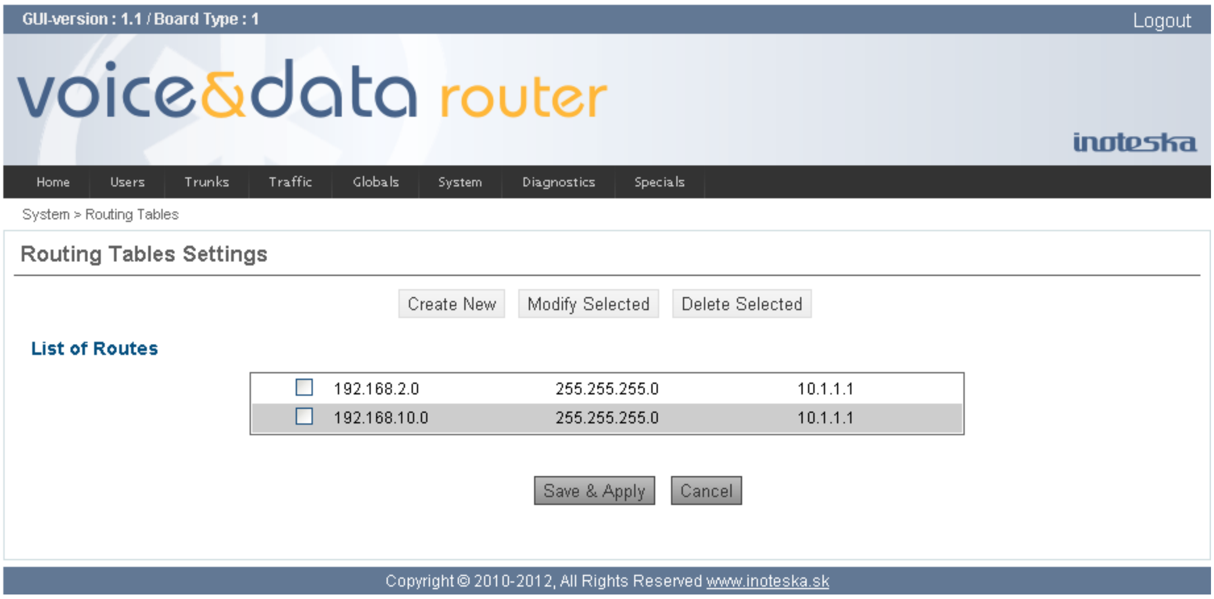
HOSTNAME
Enter Voice&Data Router hostname.

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).



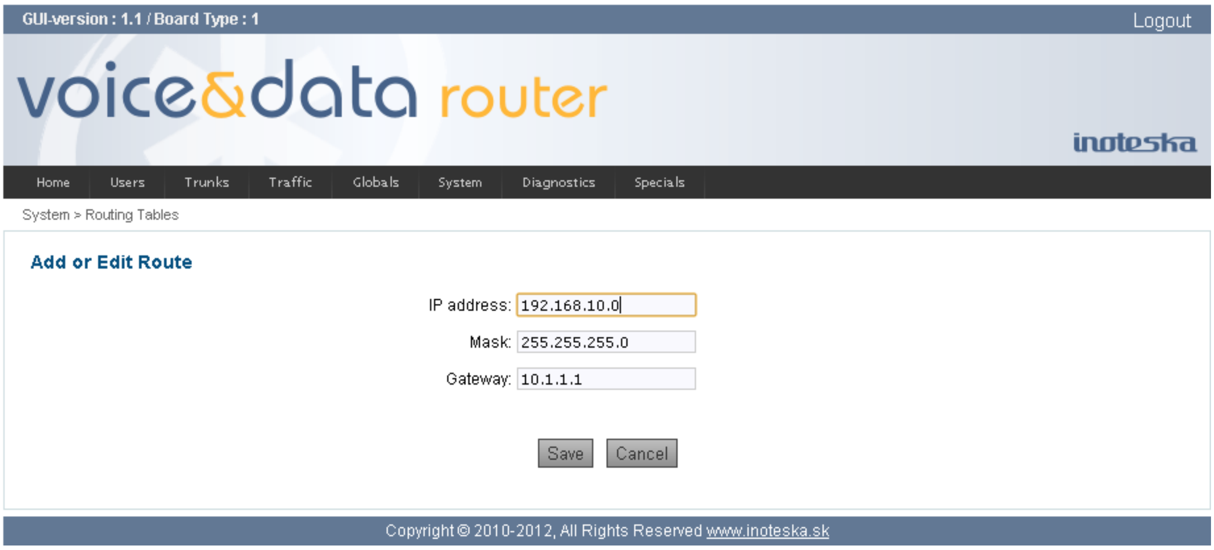
Use the **Create New** button to add new route to the routing table. Check existing route in the table and click on the **Modify Selected** to change route parameters. You can remove selected (checked) routing items when you click on the **Delete Selected** button.

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.



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

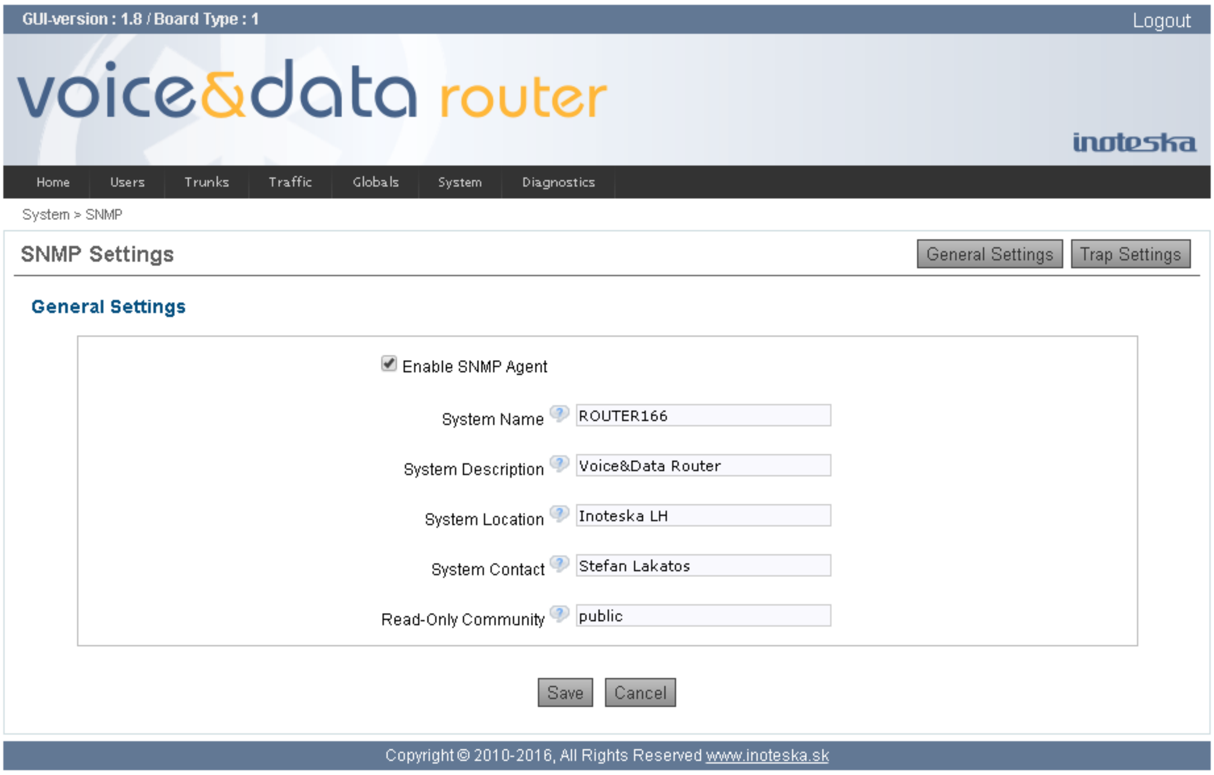
SNMP

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.



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.

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.

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

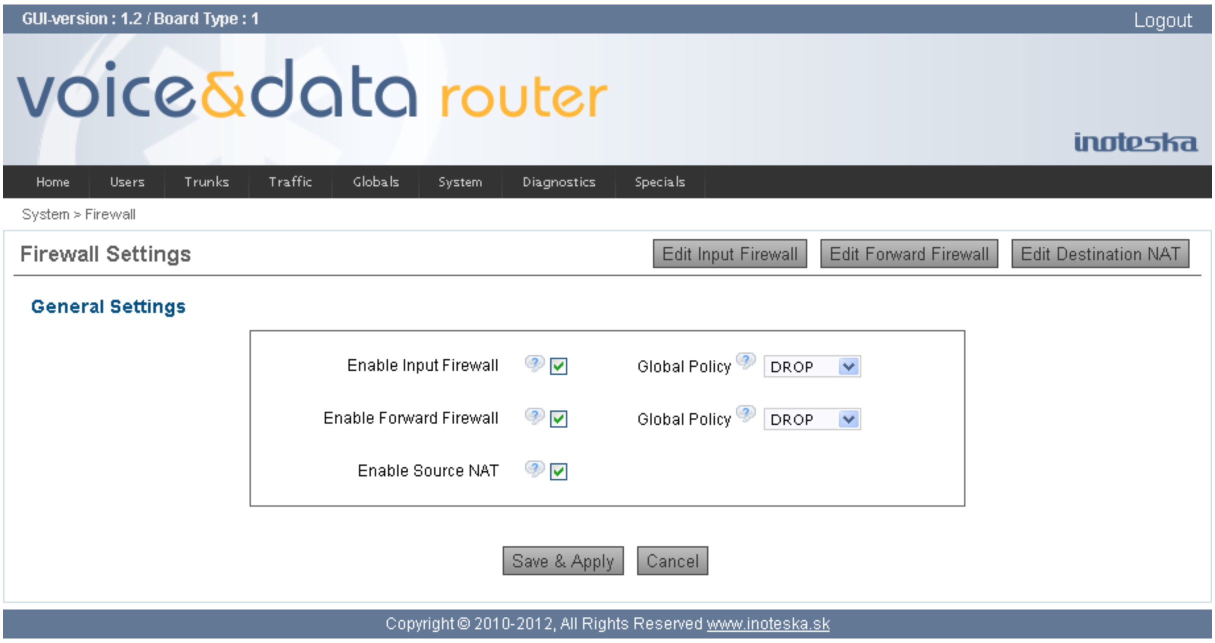
Community
Trap community description (public, private, etc.) for the administrating application to accept the notifications about the certain events on the device. Field may contain some kind of password which should be the same both on the device and on the administrating application for successful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



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.

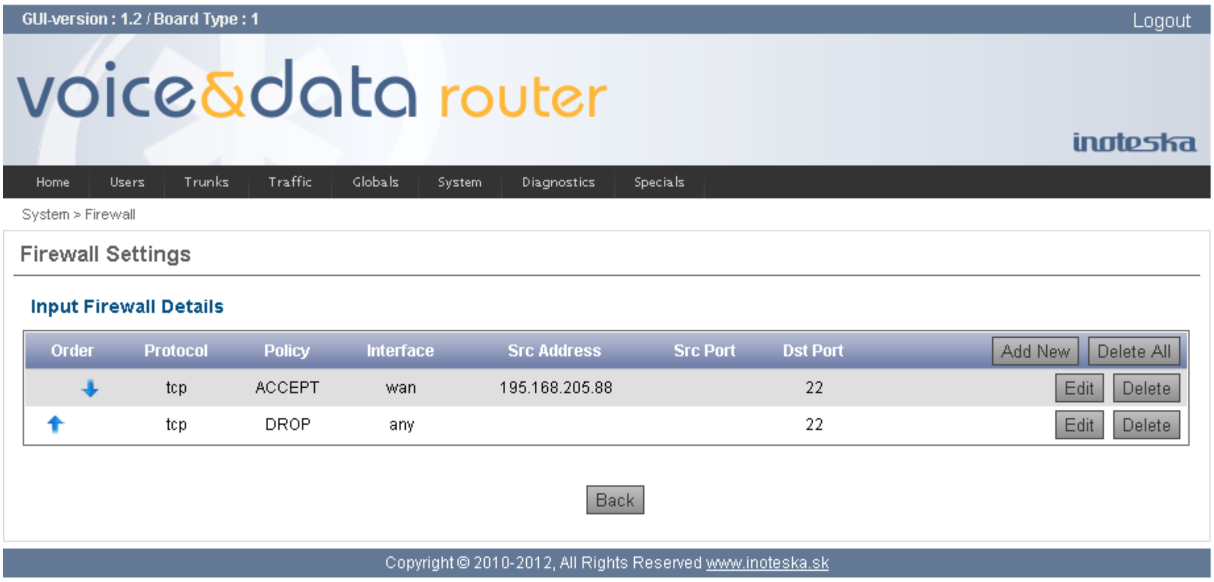
Enable Forward Firewall
Enable/disable forward firewall functionally. It allows you to control packet forwarding between LAN and WAN interfaces. You can select global policy which defines default behavior when packet does not match any firewall rule:

- **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.



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.

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.

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).

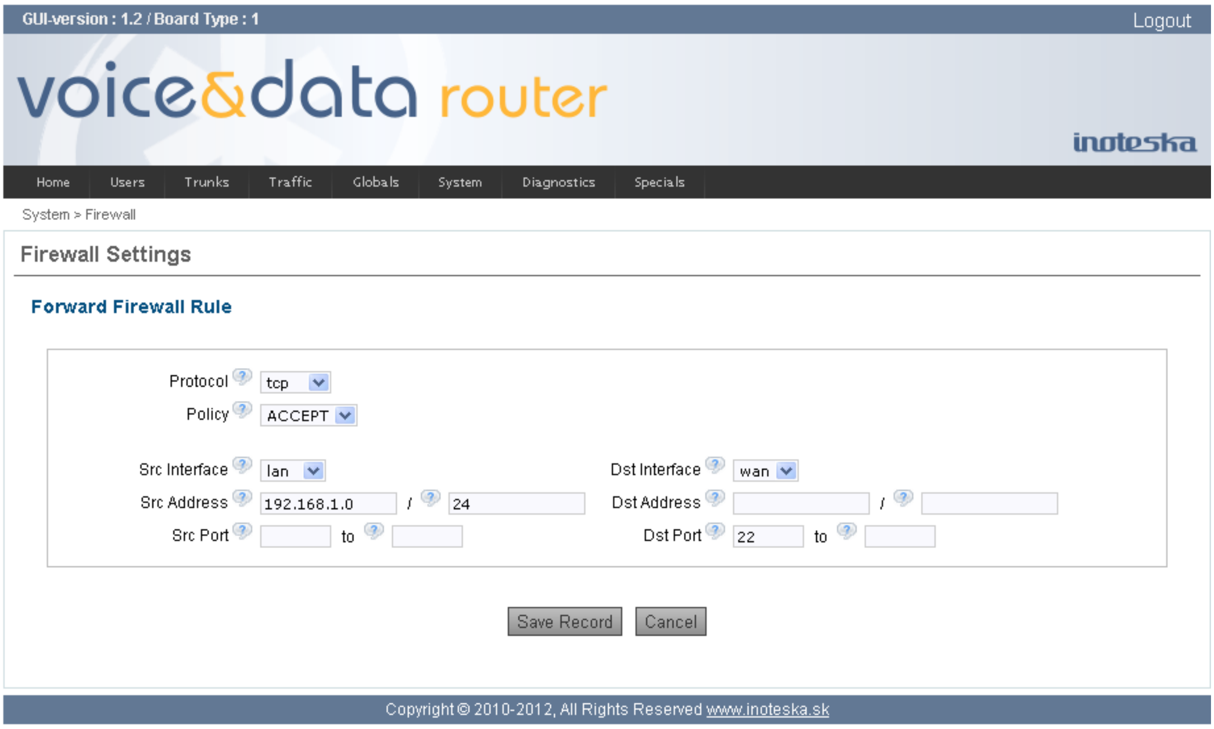
Dst Interface
Select Ethernet interface via which a packet is going to be sent. Together with source interface setting we can define condition for packet direction (from LAN to WAN or vice-versa).

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.

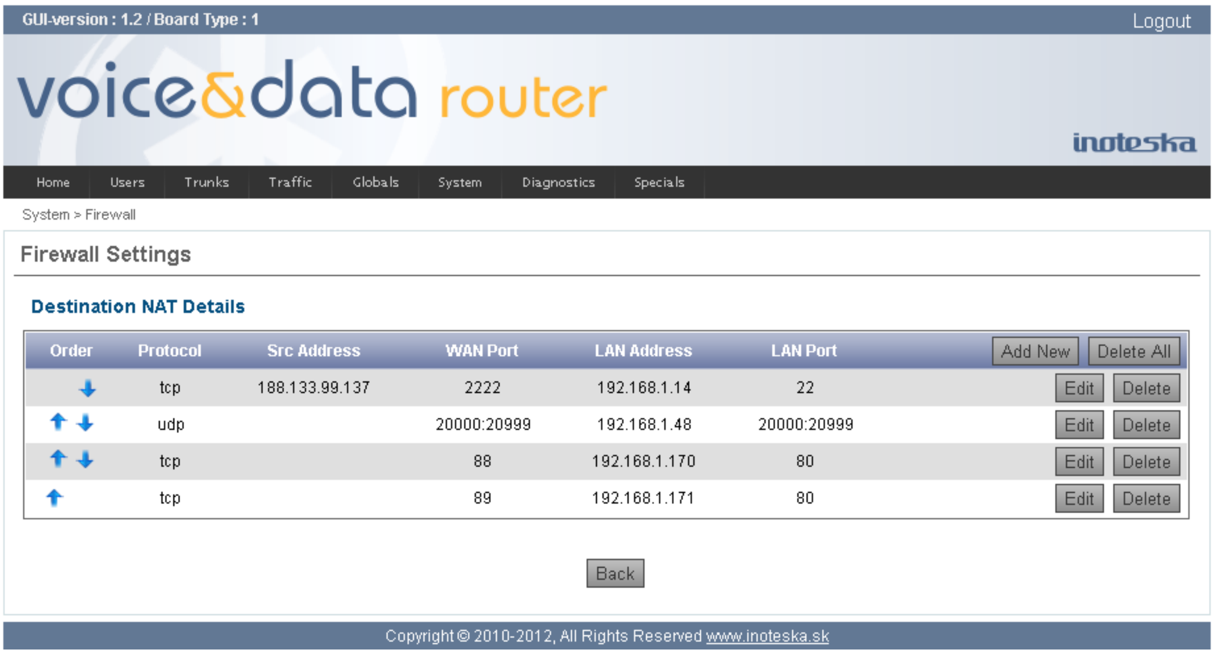


Src Port

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.



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.

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.

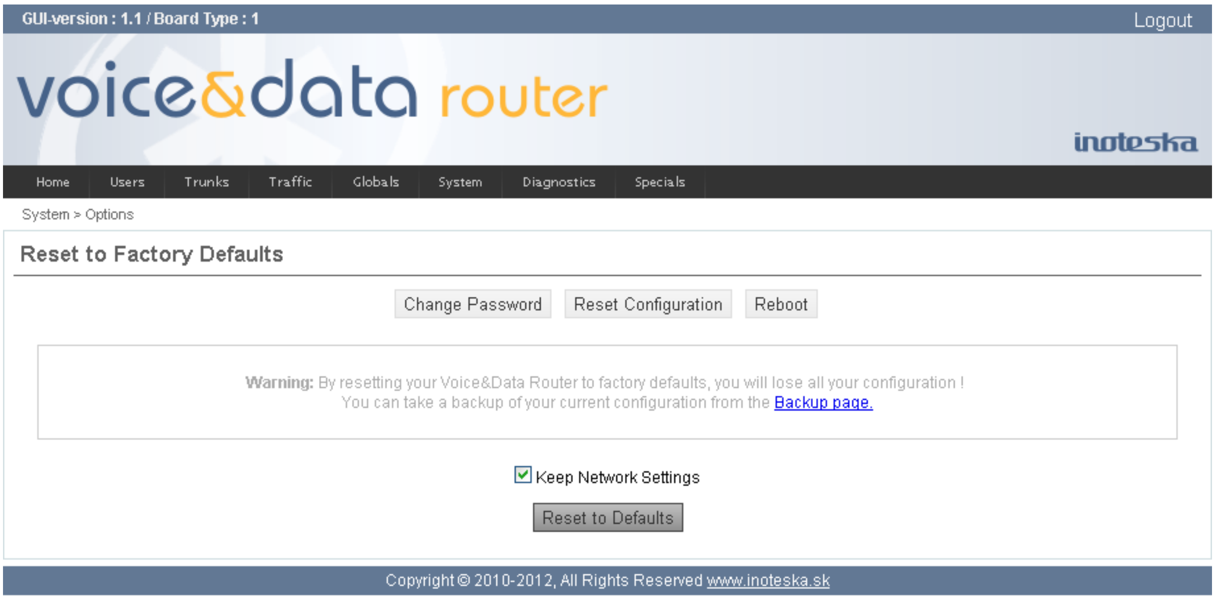
Firewall setting applies immediately after clicking on the **Save & Apply** button on the main screen of firewall setting. There is no need to activate standard Apply Changes function.

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.

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 Backup and restore function can be used before reset to defaults. Voice&Data Router will restart automatically after applying reset to the default setting.



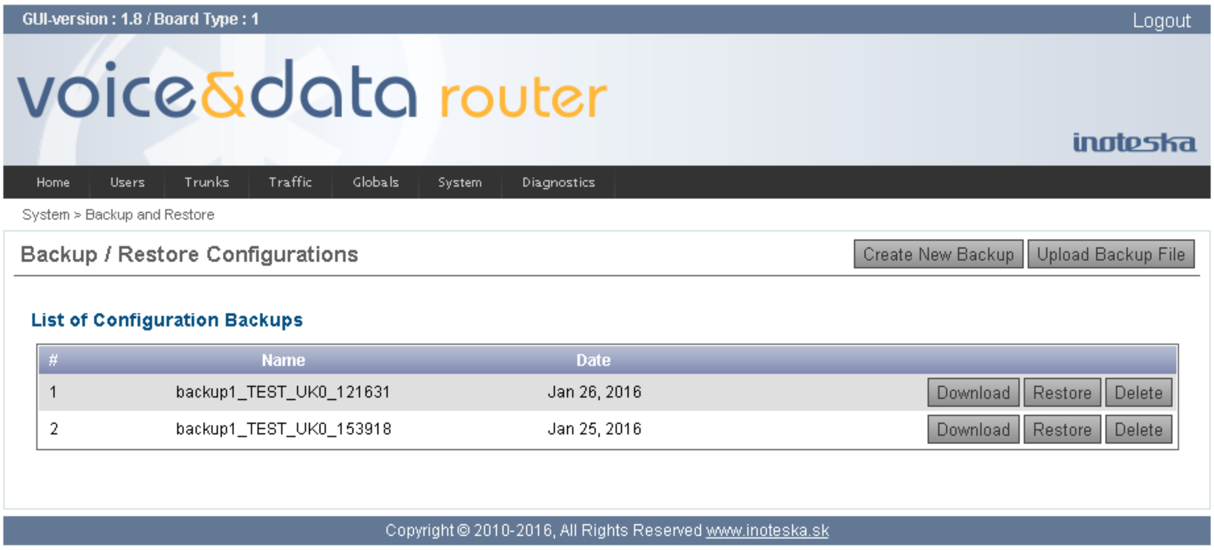
Reboot allows Voice&Data Router restart. System reboot usually takes about 1 minute but it can vary. Web Manager automatically re-connects to Voice&Data Router after minute and half. You can also use browser reload function to try to re-connect before this timeout.


GUI-version : 1.1 / Board Type : 1
Logout
voice&data router
inoteska
Home
Users
Trunks
Traffic
Globals
System
Diagnostics
Specials
System > Options
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

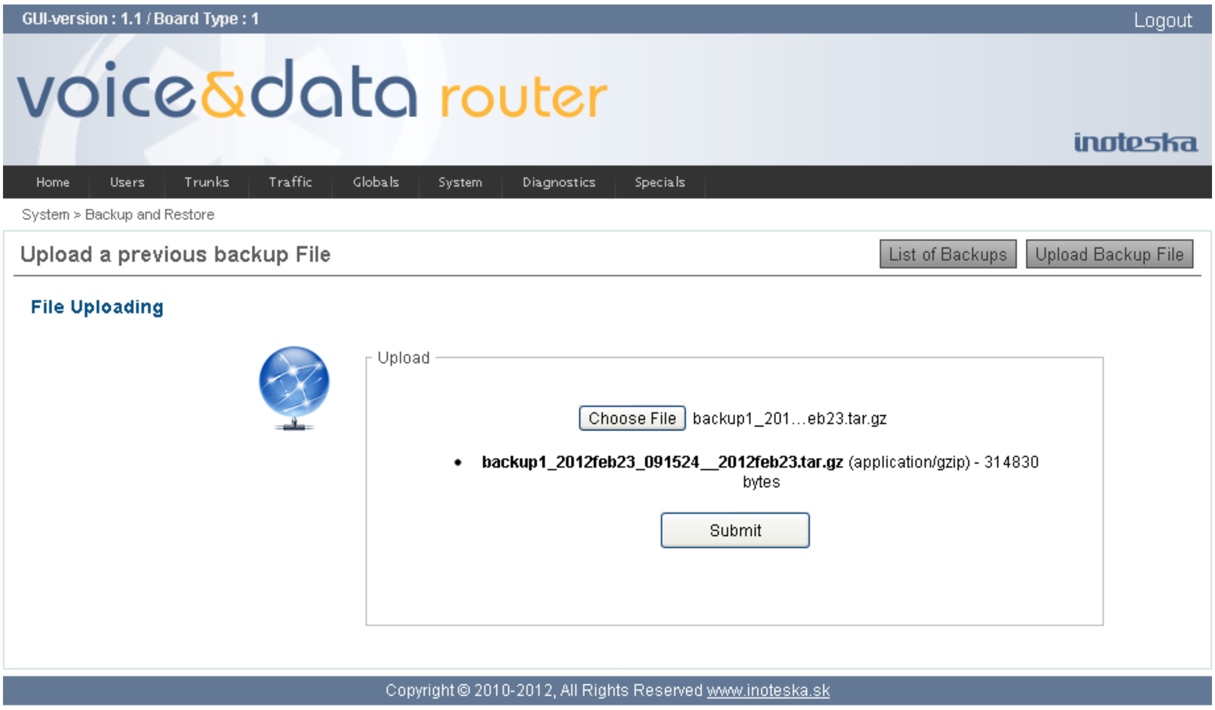
Voice&Data Router allows you to manage configuration data backups. You can create and store backup of actual Voice&Data Router configuration. It is possible to return back to the stored configurations at any moment in the future. You can also download configuration backups to archive them out of the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



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.

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.



Firmware Update

Voice&Data Router supports three possibilities how to update firmware

- TFTP protocol download
- HTTP protocol download
- File upload

Use TFTP Server

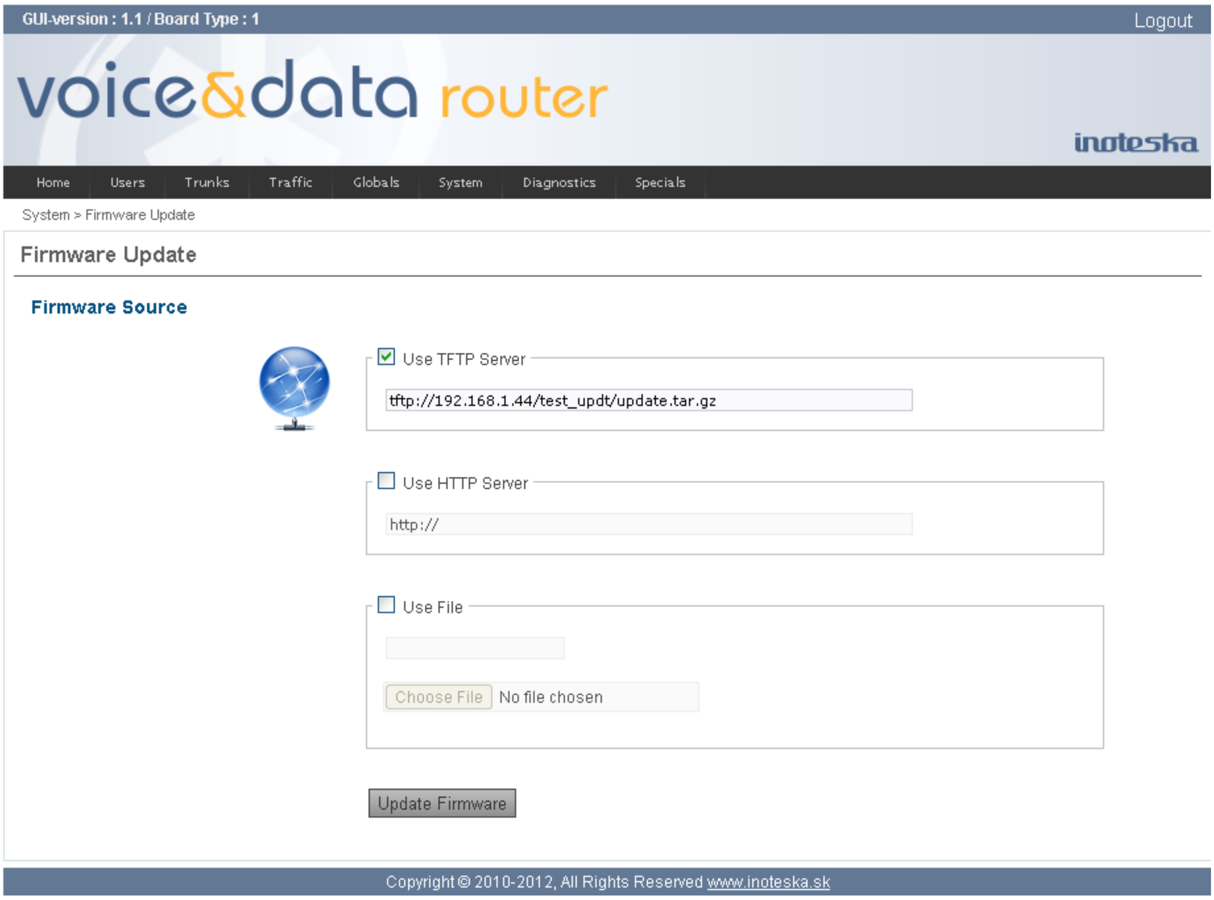
Check this option to use TFTP protocol to download firmware update file. Enter TFTP URL address – server address with path to the update file, into the input field and click on **Update Firmware** button. Voice&Data Router downloads specified file and starts update process. Firmware is checked for validity 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.



System will automatically restart after firmware update.

Date and Time

Voice&Data Router system time adjustment.

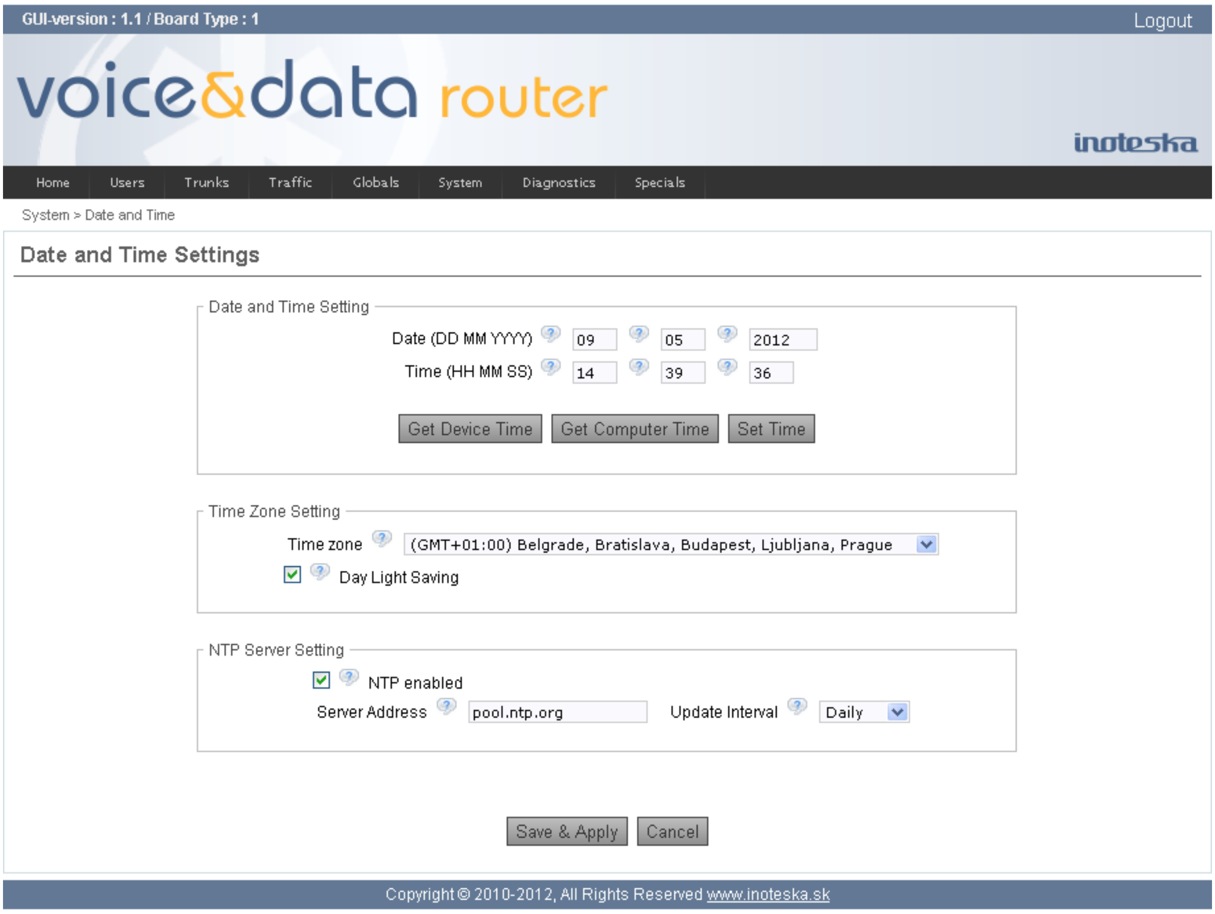
Date and Time Setting

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.



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

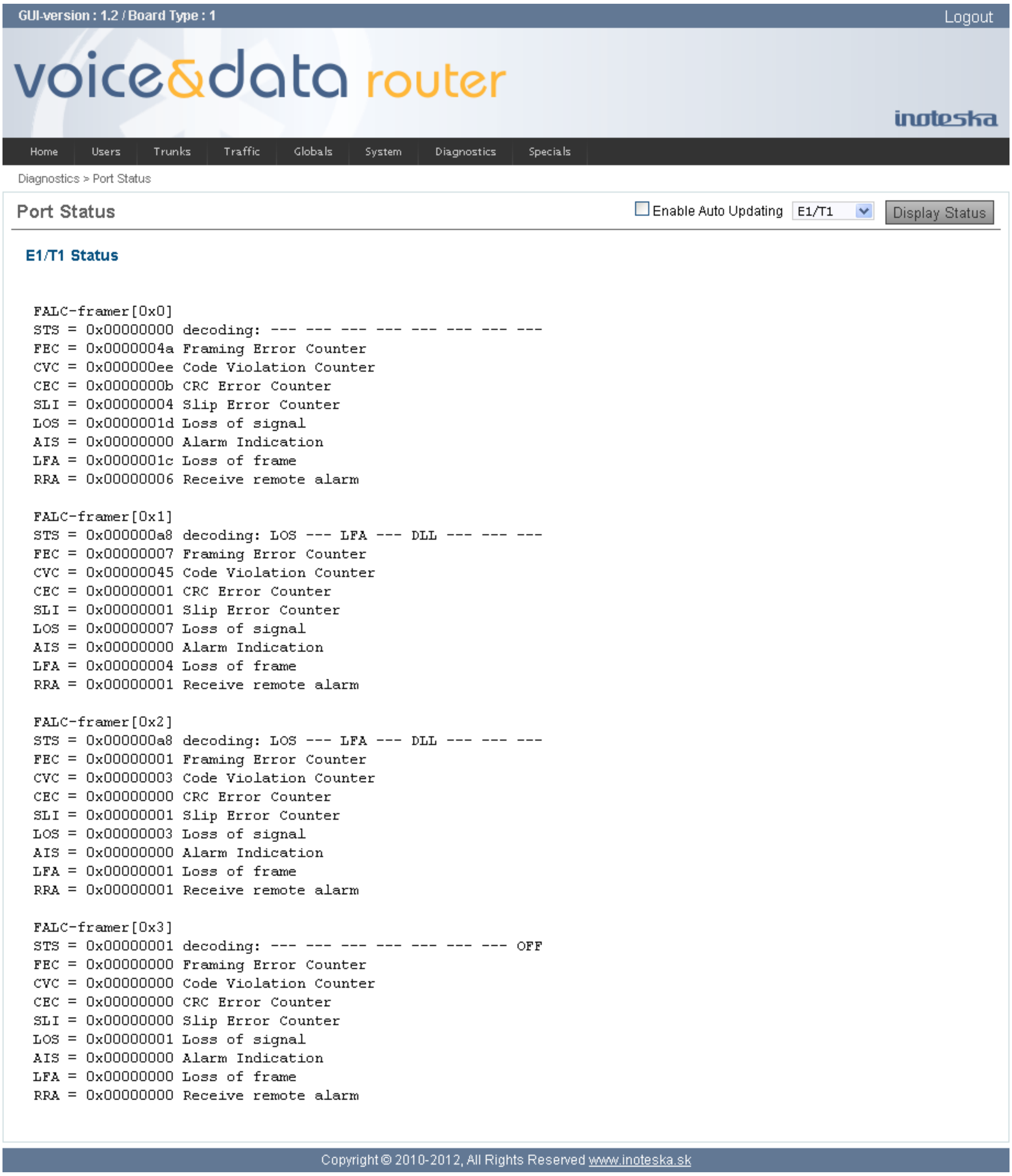
Configuration changes apply immediately after clicking on the **Save & Apply** button. There is no need to activate standard Apply Changes function.

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:

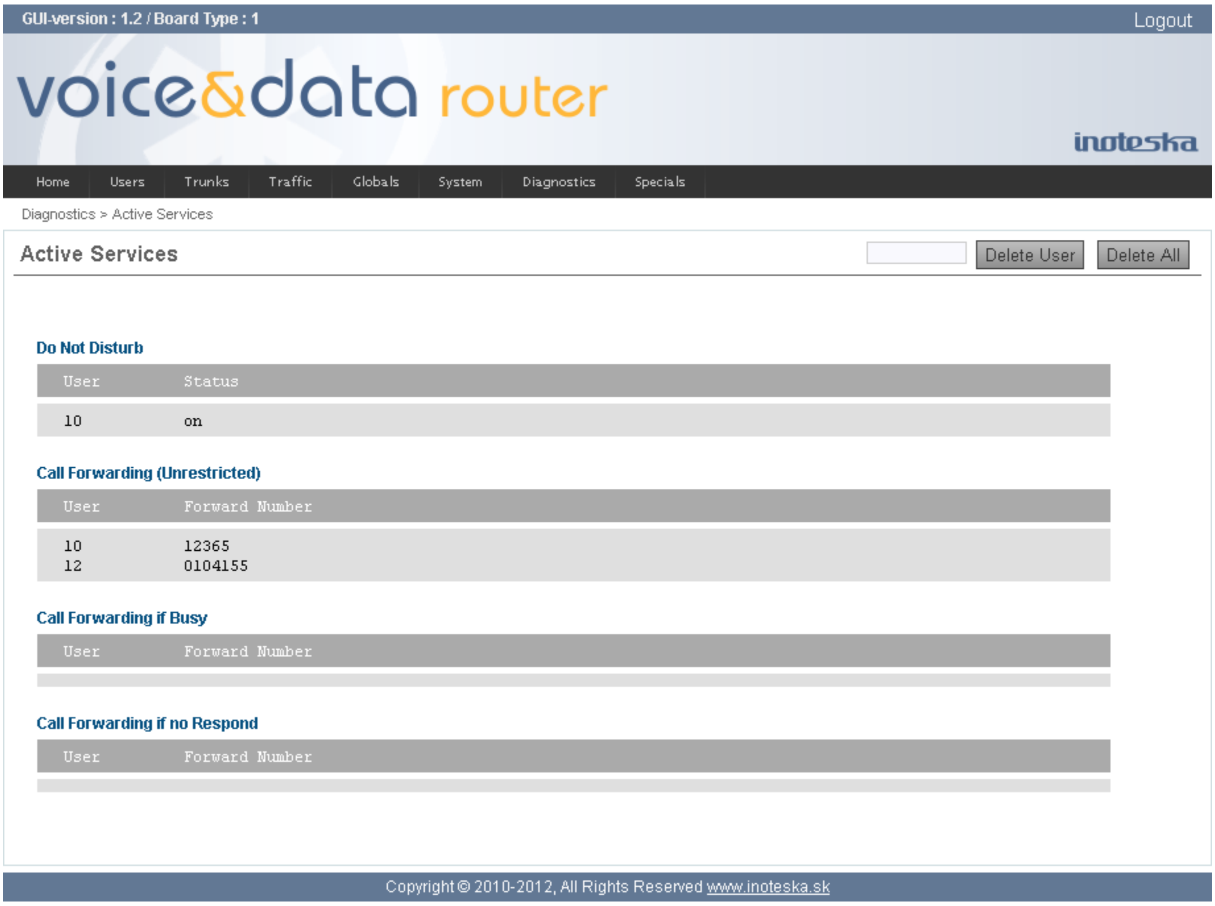
- E1/T1 – actual status of E1/T1 ports
- BRI – actual status of BRI ports
- GSM – actual status of GSM modules (optional)
- Analog –actual status of analog ports (optional)
- SIP – actual status of SIP trunks and users
- Ethernet – actual status of Ethernet ports



Active Services

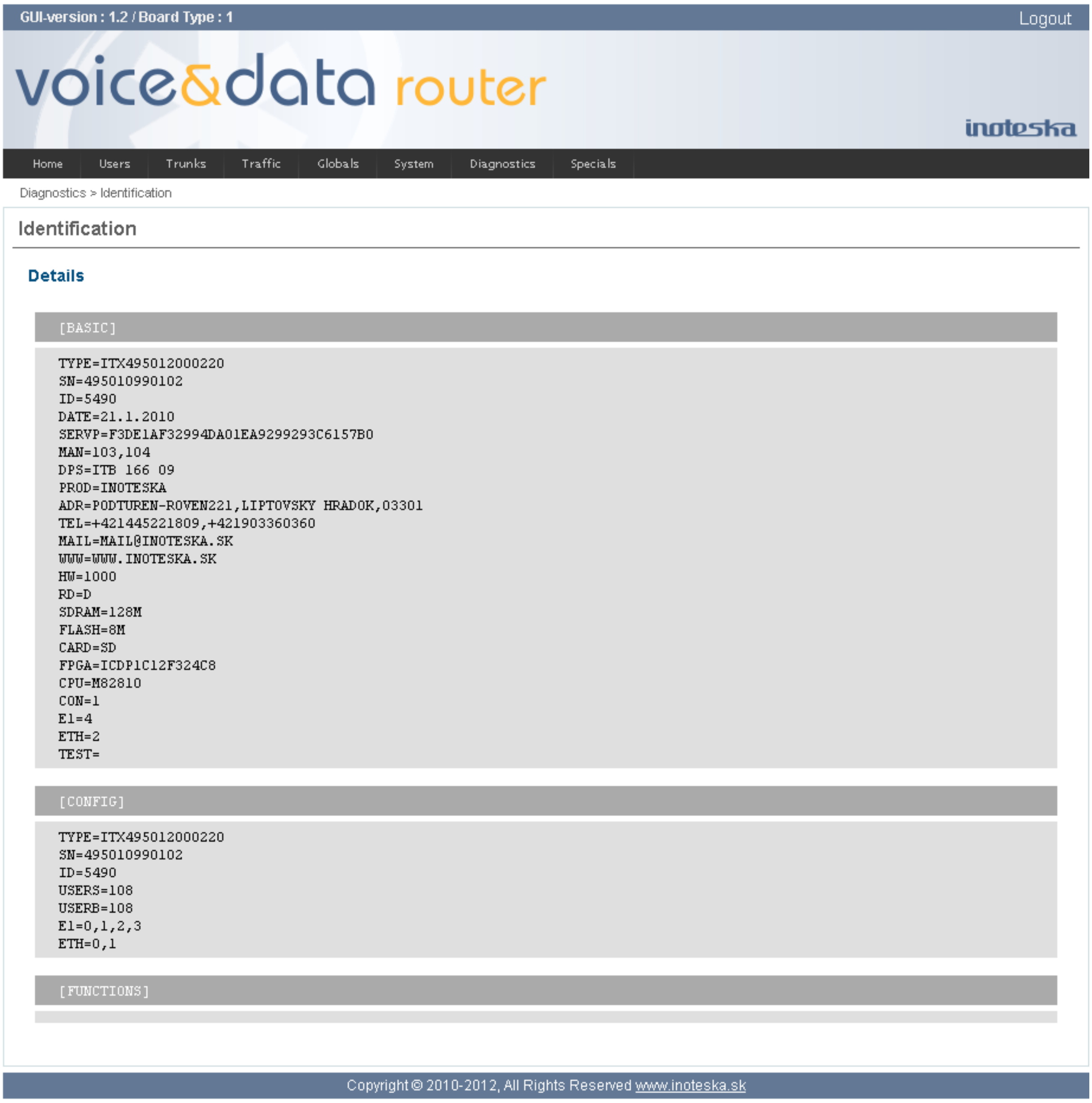
Use active services function to display actual status of users' services. It displays list of all users with active services like Do Not Disturb and Call Forwarding.

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



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.

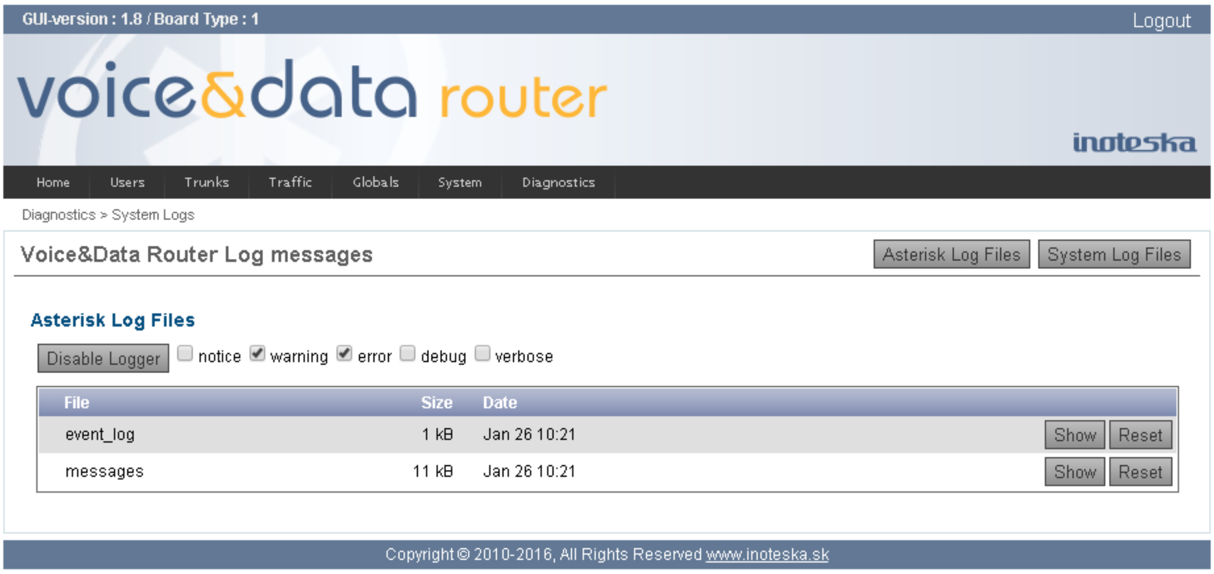


System Logs

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

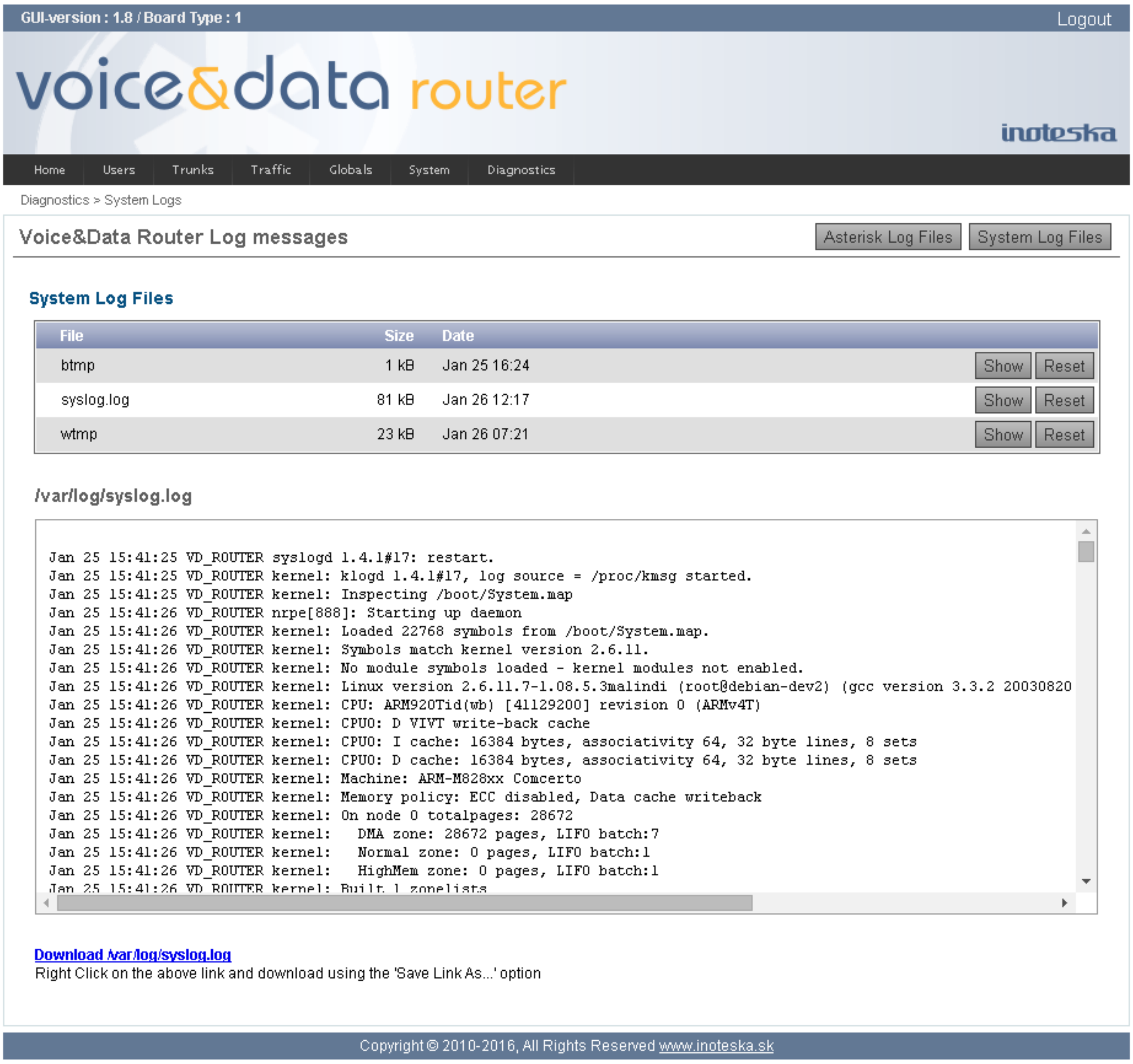
Asterisk Log Files cover log messages from the Asterisk voice communication system application. It is possible to enable/disable Asterisk logging, set logging level and display stored logs.



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
Logout
voice&data router
inoteska
Home Users Trunks Traffic Globals System Diagnostics
Diagnostics > System Logs
Voice&Data Router Log messages
Asterisk Log Files
System Log Files
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 20030820
Jan 25 15:41:26 VD_ROUTER kernel: CPU: ARM920Tid(wb) [41129200] revision 0 (ARMv4T)
Jan 25 15:41:26 VD_ROUTER kernel: CPU0: D VIVT write-back cache
Jan 25 15:41:26 VD_ROUTER kernel: CPU0: I cache: 16384 bytes, associativity 64, 32 byte lines, 8 sets
Jan 25 15:41:26 VD_ROUTER kernel: CPU0: 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 0 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, LIFO batch:1
Jan 25 15:41:26 VD_ROUTER kernel: Built 1 zonelists
Download /var/log/syslog.log
Right Click on the above link and download using the 'Save Link As...' option
Copyright © 2010-2016, All Rights Reserved www.inoteska.sk

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.

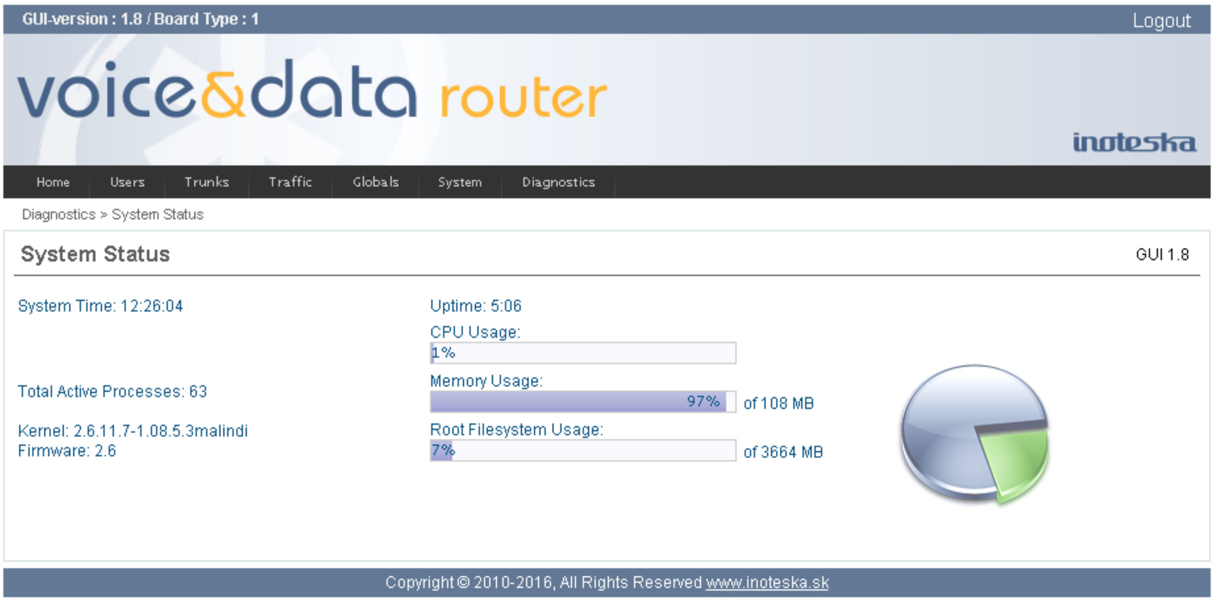
Use command “help” to display all available CLI commands. To display brief command description you can use “help” plus complete command (e.g. “help core show channels”).

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

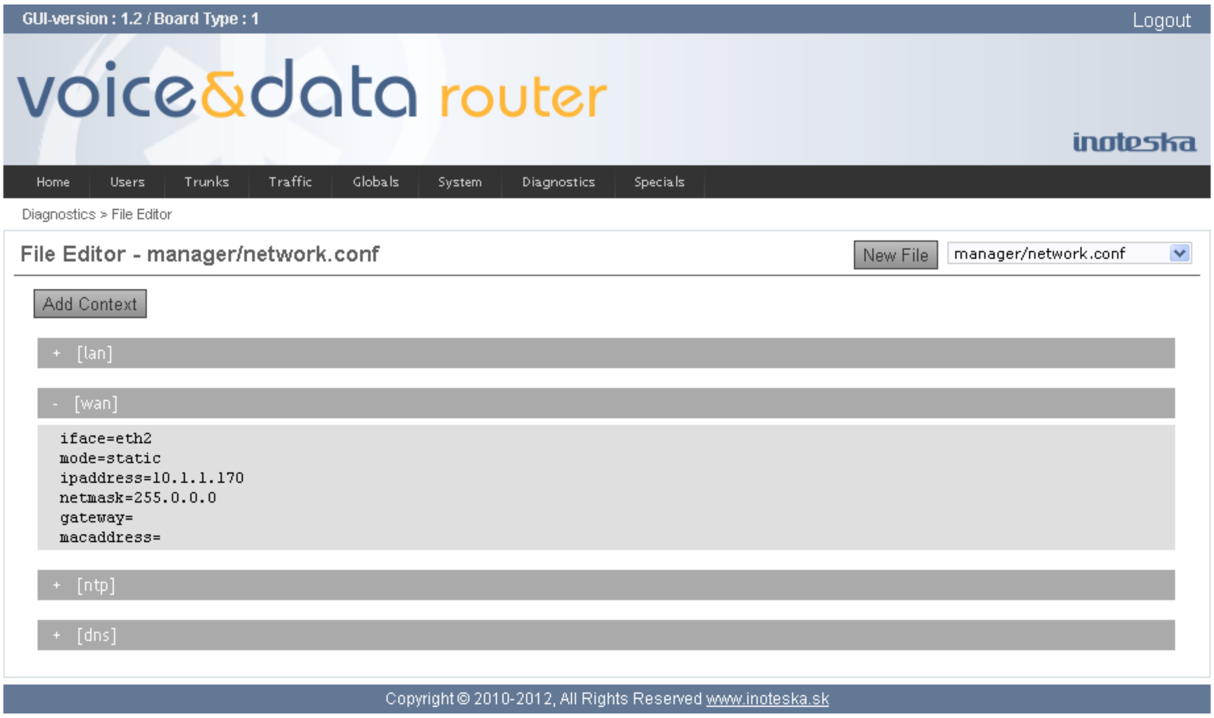
System status shows actual status of Voice&Data Router. It involves actual system time and device running time, usage of hardware resources (CPU, memory, file system) and information about software version.



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.

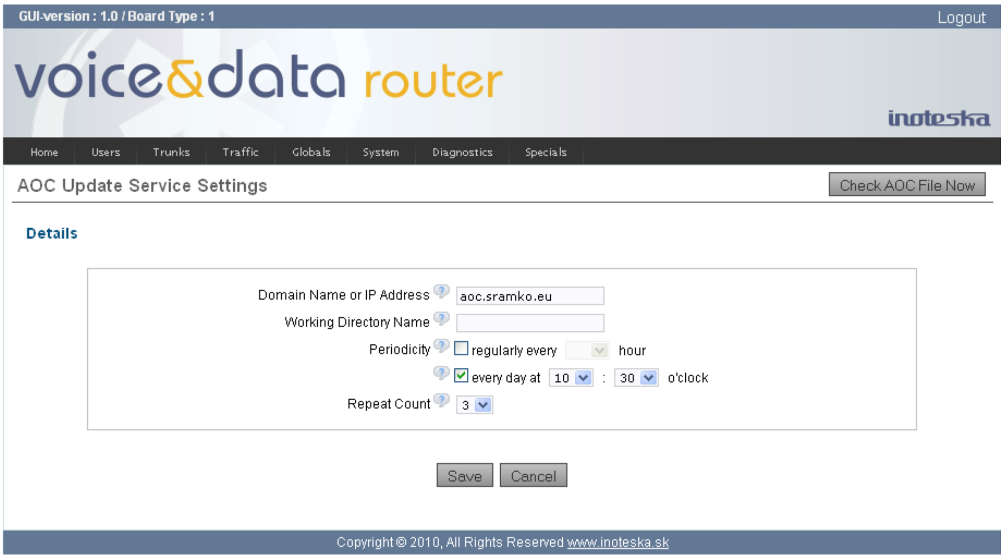
Files you can access are listed in the **Config Files** list on the right side. Selected file is displayed on the screen. File editor can handle only files divided into the contexts/sections. All the comments are filtered out – not displayed. The comments are all lines starting with semi-colon ';' and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



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.

AOC Web Update



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

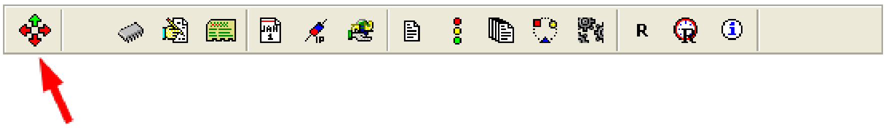
UniMan is proprietary management application running on Windows platform. It can be used for the communication with various Inoteska equipments.

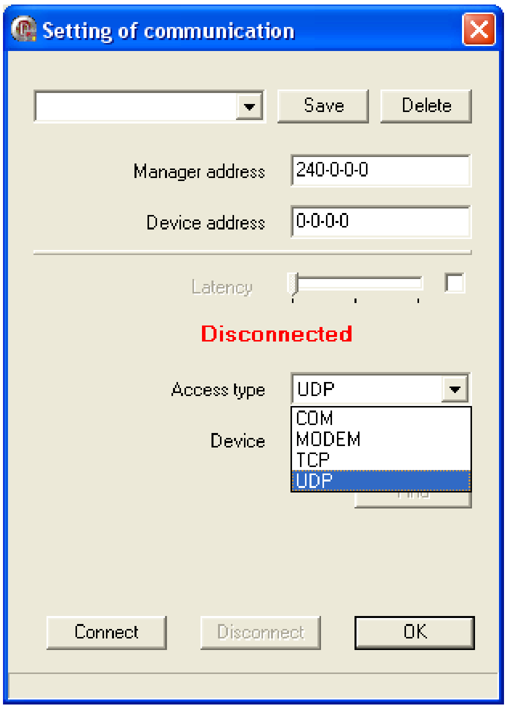
UniMan is not preferred management tool for Voice&Data Router and it does not offer complete device configuration. UniMan support in Voice&Data Router is limited only for network configuration and diagnostics.

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**.





Manager address
Select manager address which is used by UniMan protocol. First number can be from interval 240–254, other three numbers from interval 0–255. Default value (240-0-0-0) can be used for communication with Voice&Data Router.

Device address
Select device address used by UniMan protocol. First number is from interval 0–239, other three numbers from interval 0–255. Use always default setting (0-0-0-0) for Voice&Data Router.

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:

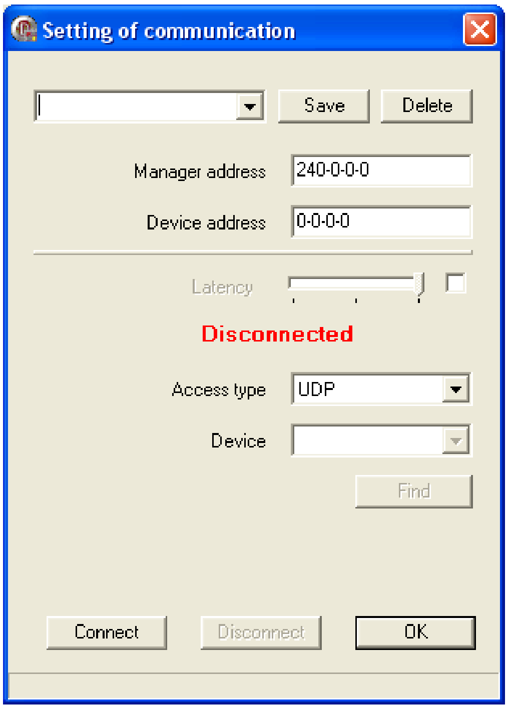
Device is connected in network

- 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



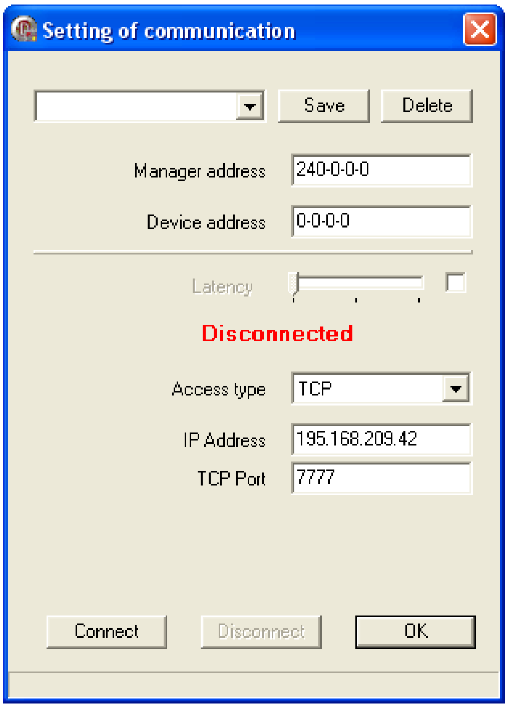
1. Set **Access type** to **UDP**.
2. Click on **Connect.** All the devices connected in network will be displayed.
3. Select the device you wish to configure. If it is connected, then **Connected** is displayed.
4. Click **OK**.

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



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**.

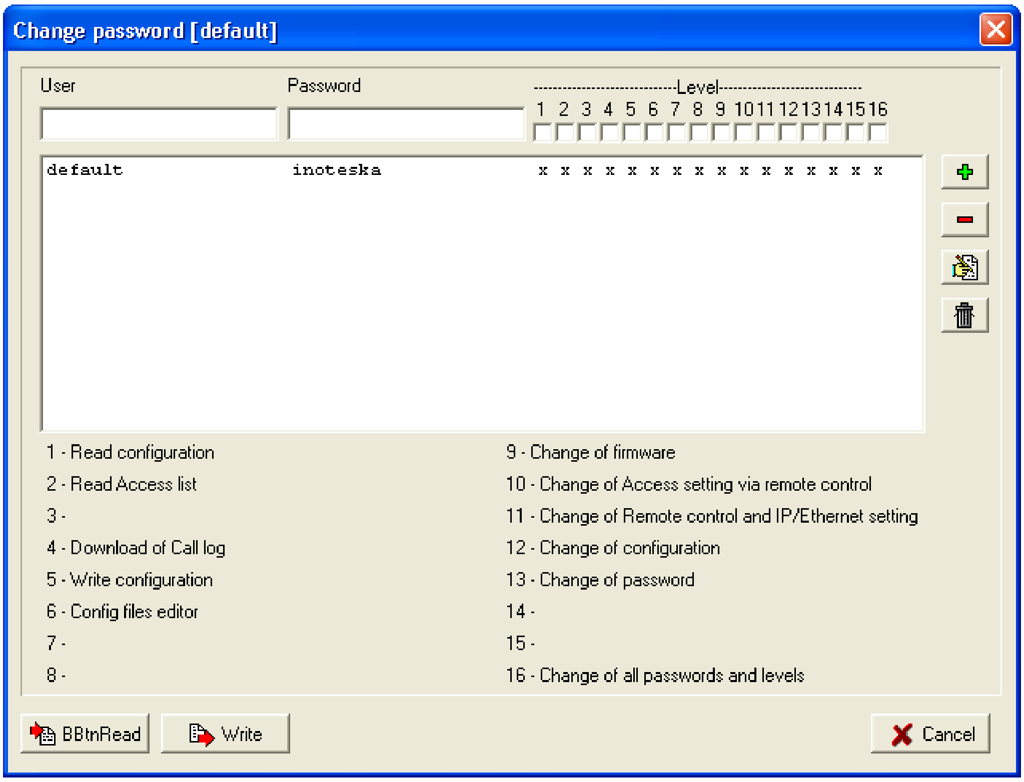
SETTING OF PASSWORD

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.

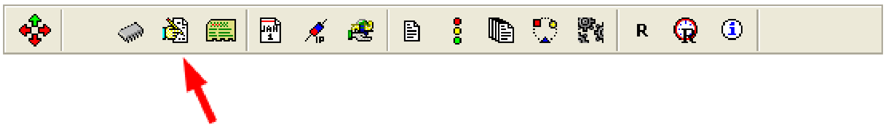
Change password of device
Change of default password. It is possible to authorize the different levels and passwords for different users.



CONFIGURATION FILES EDITOR

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

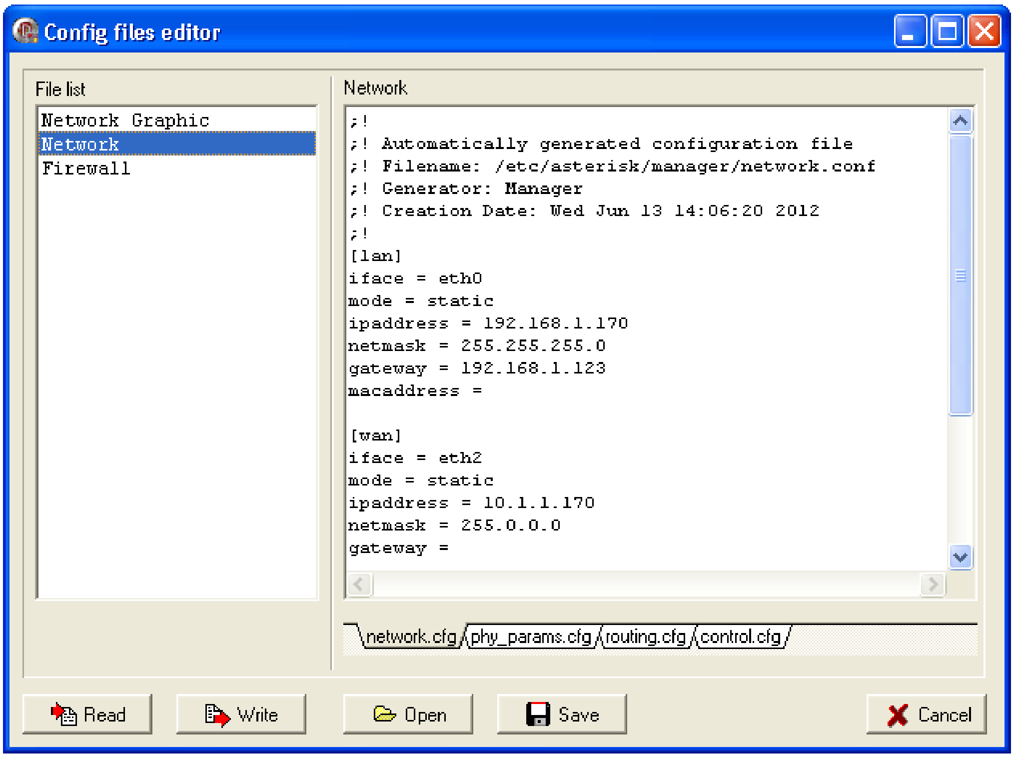
To open configuration editor click on the speed button **Config files editor** or use main menu **Communication/Config files editor**.



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

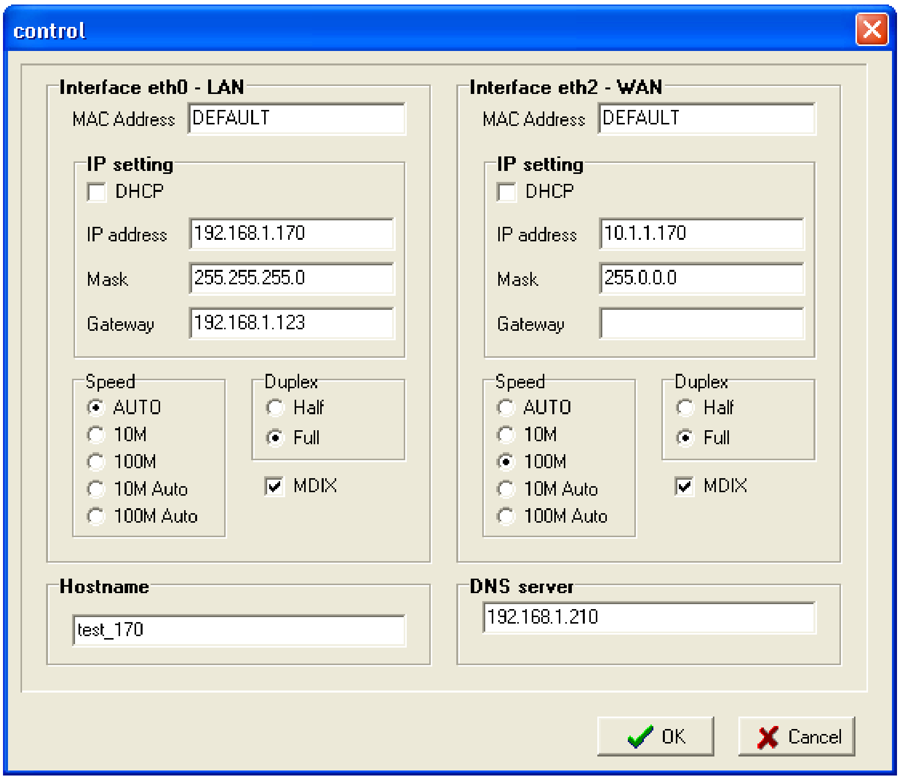


Double click on the selected item in the file list reads configuration files from the device and eventually activates GUI interface. In the text mode you can directly edit configuration files in the right side of the window.

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**.



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.

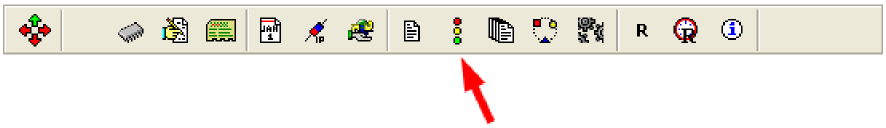
MDIX
Enable/disable MDIX (Medium Dependent Interface Crossover) option for Ethernet interface.

Hostname
Enter Voice&Data Router hostname.

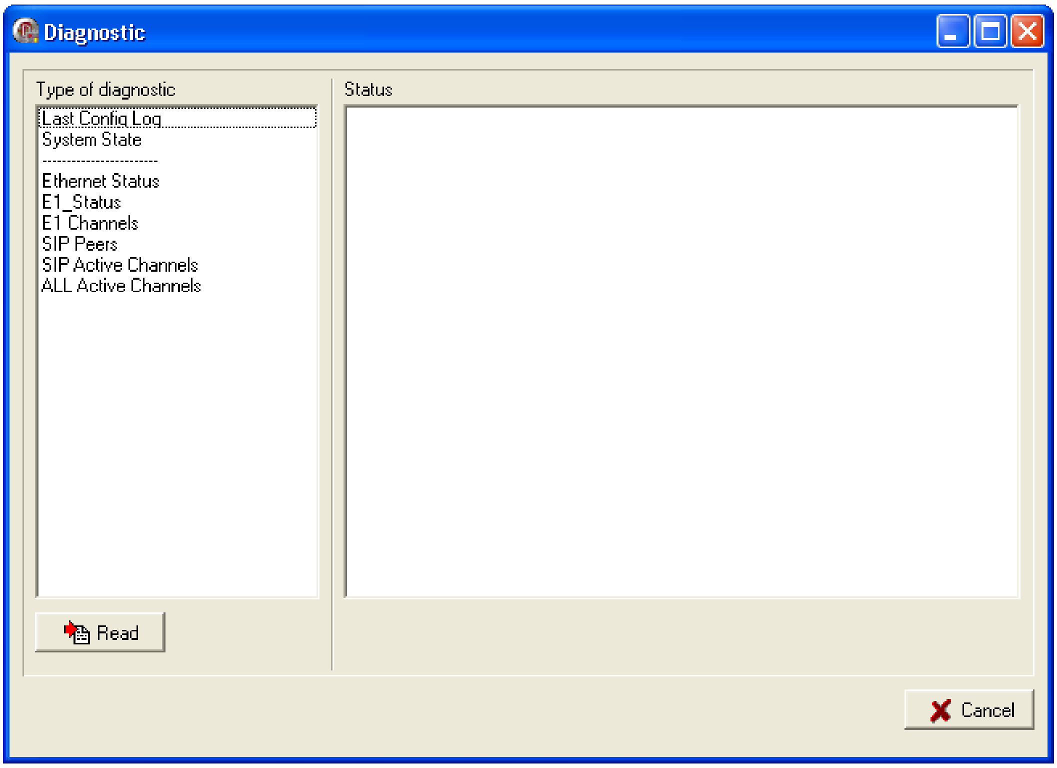
DNS Server
Enter IP address of Domain Name Server (only primary).

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**.



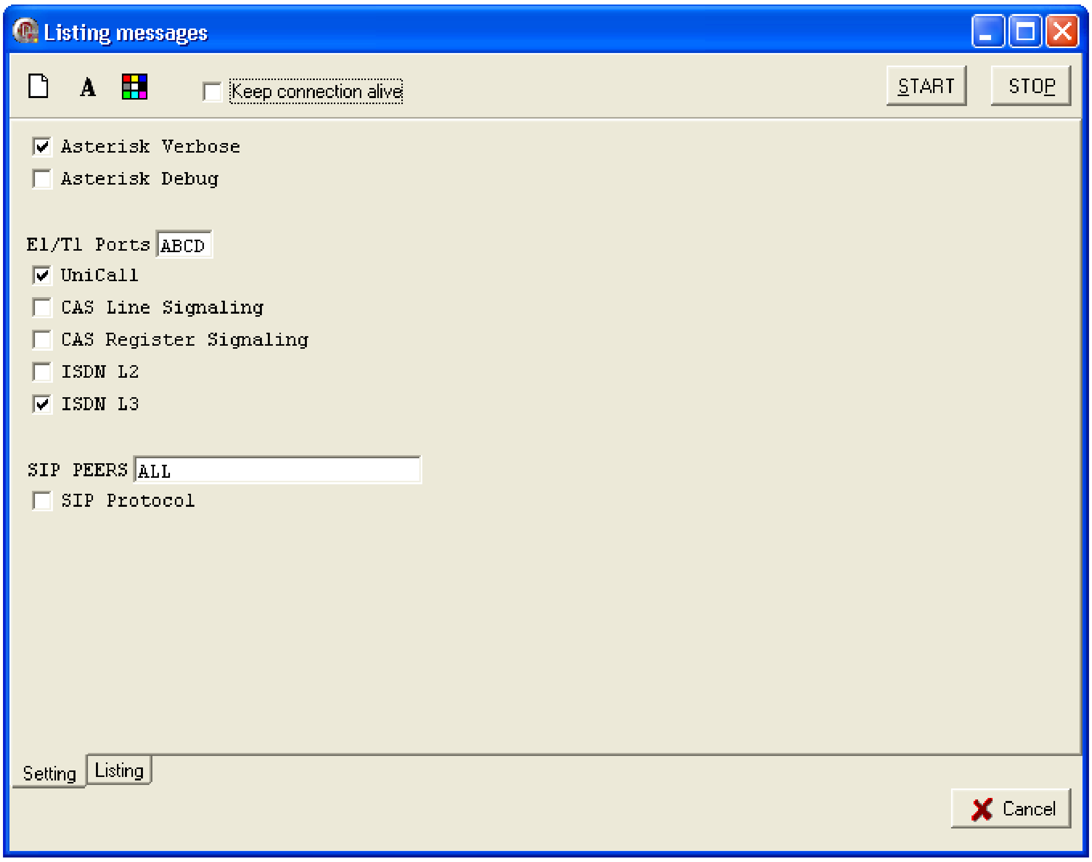
Type of diagnostic list on the left side of the diagnostics shows index of items that can be monitored. Double click on the selected item will display information of actual status. In the most cases is the status automatically updated.



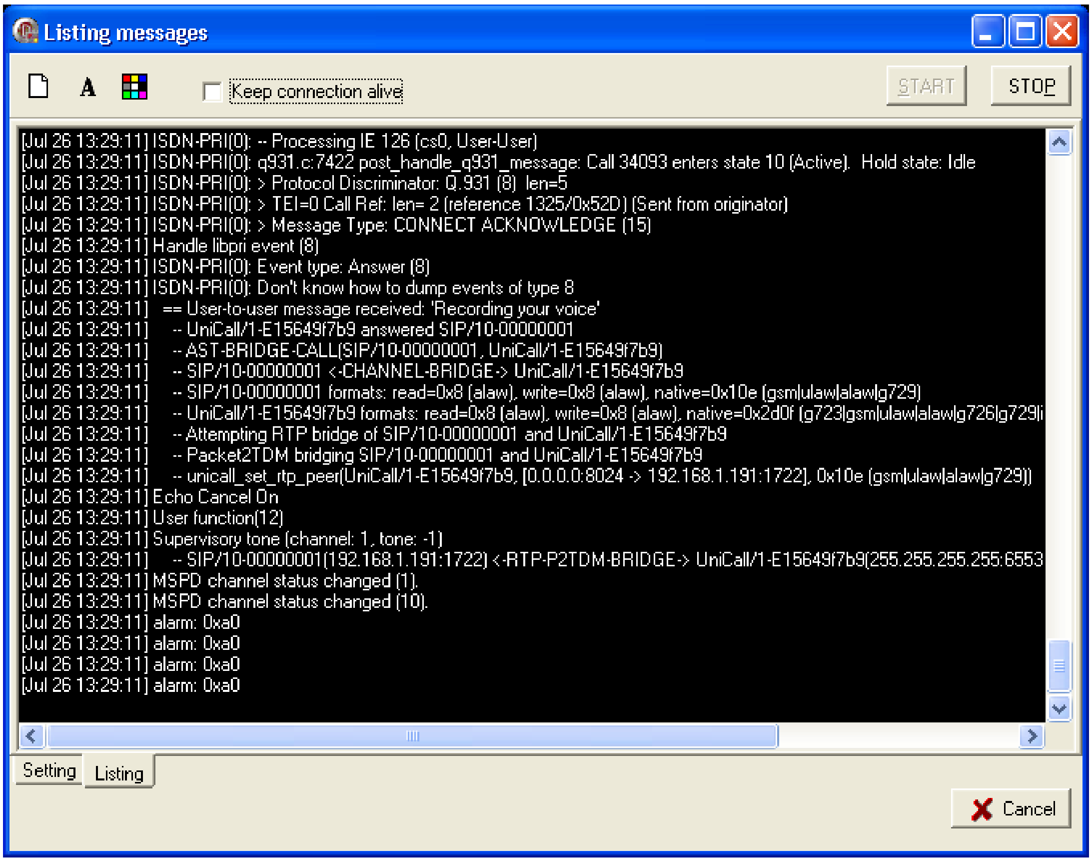
LISTING MESSAGES

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.

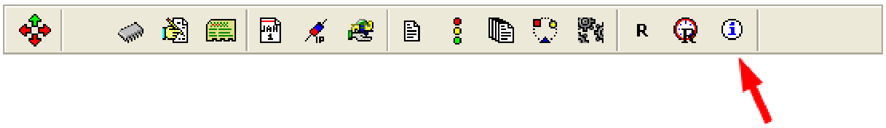


Click **START** button to begin capturing of the logs from the device. Received logs are displayed in real-time on the **Listing** card. Use **STOP** button to finish device monitoring and store captured logs in the text file on the local PC.

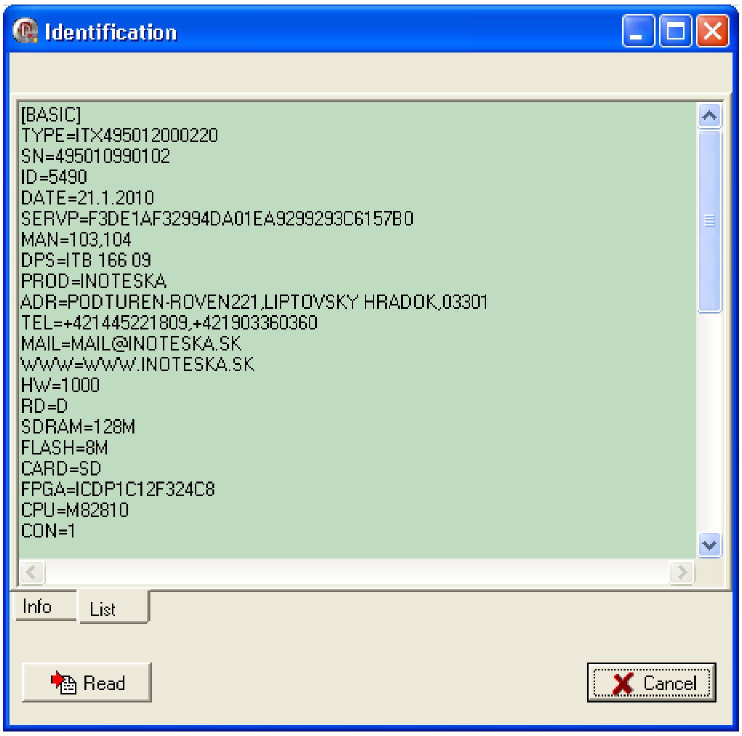

Listing messages
Keep connection alive
START
STOP
[Jul 26 13:29:11] ISDN-PRI(0): -- Processing IE 126 (cs0, User-User)
[Jul 26 13:29:11] ISDN-PRI(0): q931.c:7422 post_handle_q931_message: Call 34093 enters state 10 (Active). Hold state: Idle
[Jul 26 13:29:11] ISDN-PRI(0): > Protocol Discriminator: Q.931 (8) len=5
[Jul 26 13:29:11] ISDN-PRI(0): > TEI=0 Call Ref: len= 2 (reference 1325/0x52D) (Sent from originator)
[Jul 26 13:29:11] ISDN-PRI(0): > Message Type: CONNECT ACKNOWLEDGE (15)
[Jul 26 13:29:11] Handle libpri event (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] -- AST-BRIDGE-CALL(SIP/10-00000001, UniCall/1-E15649f7b9)
[Jul 26 13:29:11] -- SIP/10-00000001 <-CHANNEL-BRIDGE-> UniCall/1-E15649f7b9
[Jul 26 13:29:11] -- SIP/10-00000001 formats: read=0x8 (alaw), write=0x8 (alaw), native=0x10e (gsm|ulaw|alaw|g729)
[Jul 26 13:29:11] -- UniCall/1-E15649f7b9 formats: read=0x8 (alaw), write=0x8 (alaw), native=0x2d0f (g723|gsm|ulaw|alaw|g726|g729|i
[Jul 26 13:29:11] -- Attempting RTP bridge of SIP/10-00000001 and UniCall/1-E15649f7b9
[Jul 26 13:29:11] -- Packet2TDM bridging SIP/10-00000001 and UniCall/1-E15649f7b9
[Jul 26 13:29:11] -- unicall_set_rtp_peer(UniCall/1-E15649f7b9, [0.0.0.0:8024 -> 192.168.1.191:1722], 0x10e (gsm|ulaw|alaw|g729))
[Jul 26 13:29:11] Echo Cancel On
[Jul 26 13:29:11] User function(12)
[Jul 26 13:29:11] Supervisory tone (channel: 1, tone: -1)
[Jul 26 13:29:11] -- SIP/10-00000001(192.168.1.191:1722) <-RTP-P2TDM-BRIDGE-> UniCall/1-E15649f7b9(255.255.255.255:6553
[Jul 26 13:29:11] MSPD channel status changed (1).
[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
Setting
Listing
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.



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:

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