# VOICE&DATA ROUTER BA

## PRODUCT DOCUMENTATION



# CONTENTS

# 1. PRODUCT OVERVIEW

**Voice&Data Router BA is is device which can be used as BRI/PRI/analog VoIP gateway, VoIP (ISDN) PbX or voice&data router.**

## PRODUCT FEATURES

- based on project Astfin and uClinux
- OS LINUX
- Asterisk SW v 1.4x
- BLACKFIN procesor BF527 600MHz
- 64MB RAM
- 128MB NAND Flash
- 8MB DATA Flash
- optional SD card

## INTERFACES

A variant - (main card) - 8 analog modules

- 0/32 analog FXS interfaces
- 0/16 analog FXO interfaces
- 0/6 analog E&M interfaces
- 0/16 analog LB (local battery) interfaces
- 0/4 BRI S0 TE/NT interfaces with optional power supply 48V
- 0-1xE1
- 1xUSB OTG
- 1-2xETHERNET 10/100BT
- 1xRS232 console
- 0/12 GSM



B variant - (main+extension card) -16 analog modules

- 0/64 analog FXS interfaces
- 0/32 analog FXO interfaces
- 0/12 analog E&M interfaces
- 0/32 analog LB (local battery) interfaces
- 0/8 BRI S0 TE/NT interfaces with optional power supply 48V
- 0-1xE1
- 1xUSB OTG
- 1-2xETHERNET 10/100BT
- 1xRS232 console
- 0/12 GSM

# VARIANTS

## 1U version:

| Voice&Data Router BA ITX 495 02 (stand alone) | |
|---|---|
| *BRI* | 0 / 4xBRI / 8xBRI S0 |
| *FXS/FXO/E&M* | Max. 8 or 16 analog modules |
| | FXS module (quad) |
| | FXO module (dual) |
| | E&M module (dual) |
| *E1* | 0 / 1xE1 120 Ohm (RJ 45) |
| *GSM* | 0 / 1 up to 12xGSM |
| *Power supply* | 230V AC and -48V DC |
| | 230V AC |
| | -48V DC |

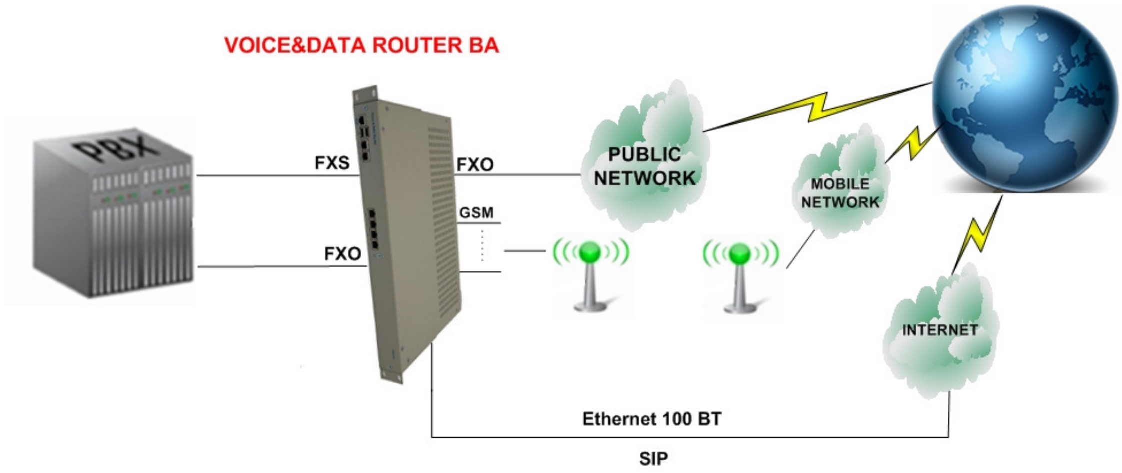
## 19" rack version 6U:

| Voice&Data Router BA ITX 402 42 (rack card) | |
|---|---|
| *BRI* | 0 / 4xBRI |
| *FXS/FXO/E&M* | Max. 8 analog modules |
| | FXS module (quad) |
| | FXO module (dual) |
| | E&M module (dual) |
| *E1* | 0 or 1xE1 120 Ohm (RJ 45) |

*Max. number of ITX 402 42 cards placed to the rack - 8 pcs.*

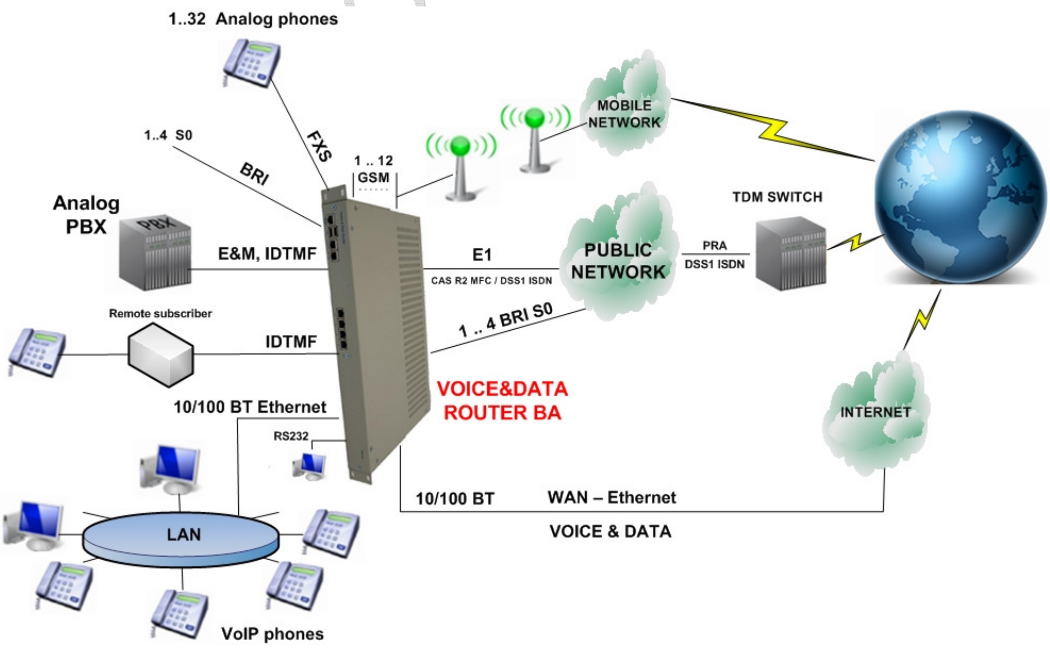
# APPLICATIONS

- **Voice & Data Router**

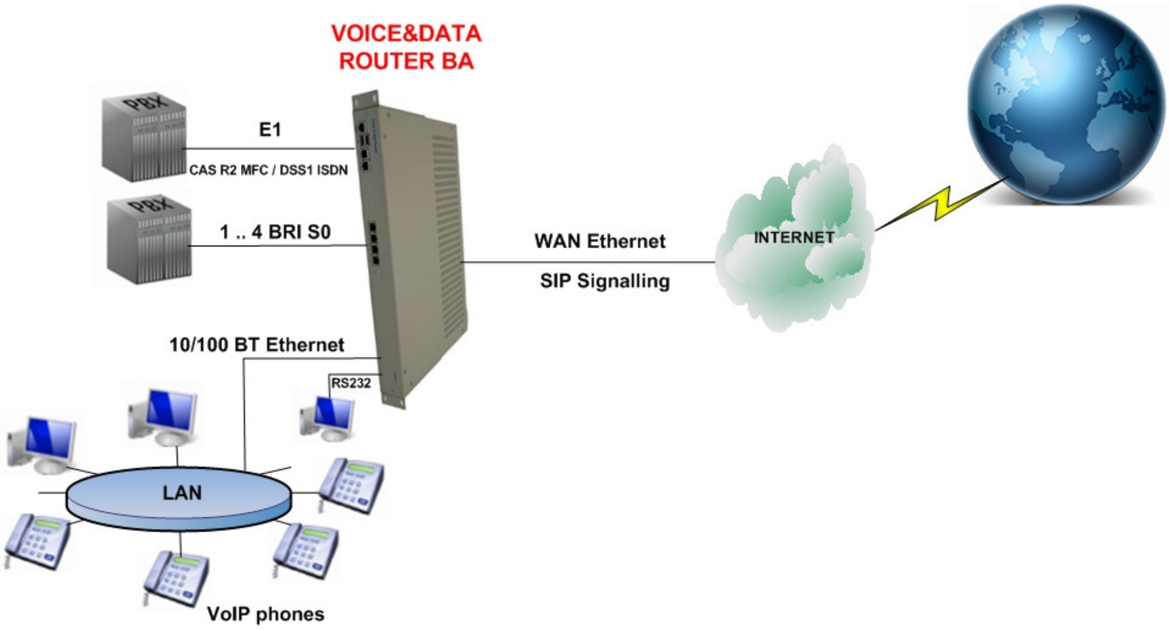



- **VoIP, TDM PBX (SW Asterisk)**

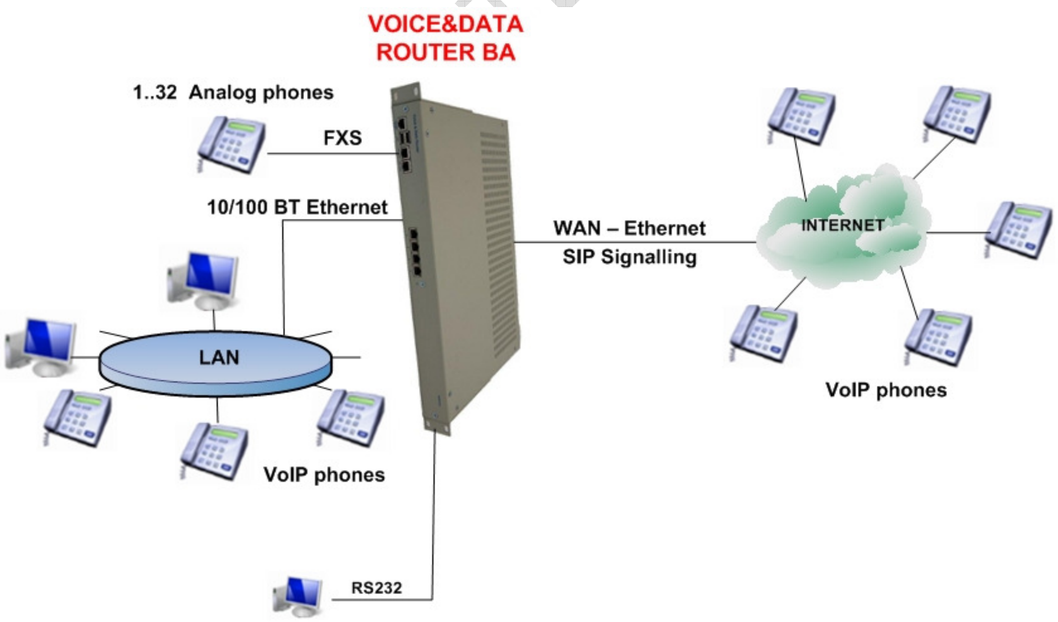
PBX Features: Call Forwarding, Call Hold/Transfer, Music On Hold, Call Parking, Call Pickup, Speed Dial, CLIP/CLIR, Automatic Attendant,...



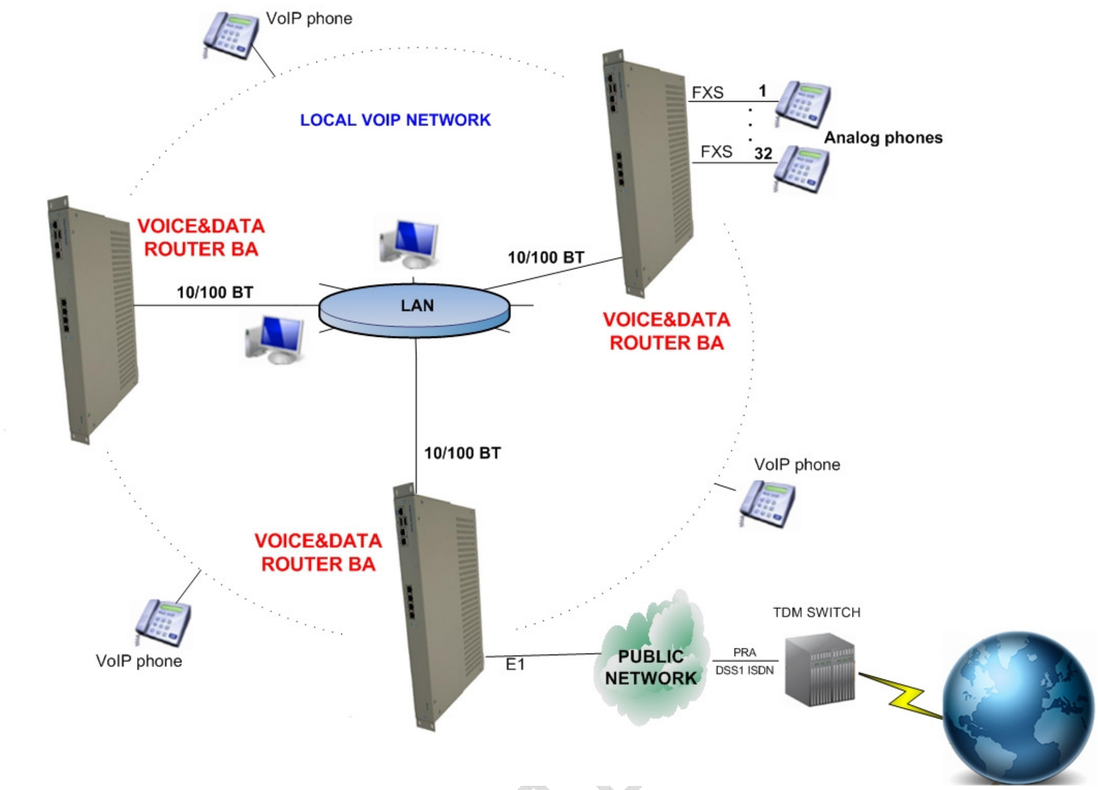
- **TDM – VoIP gateway**

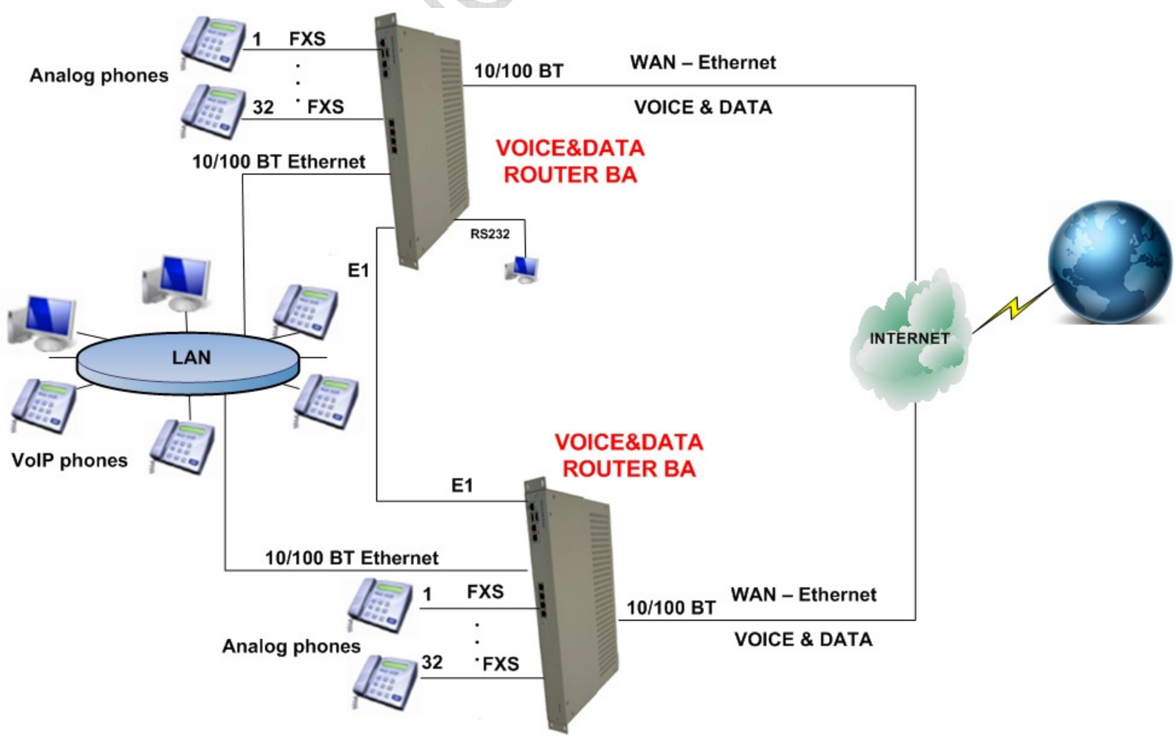


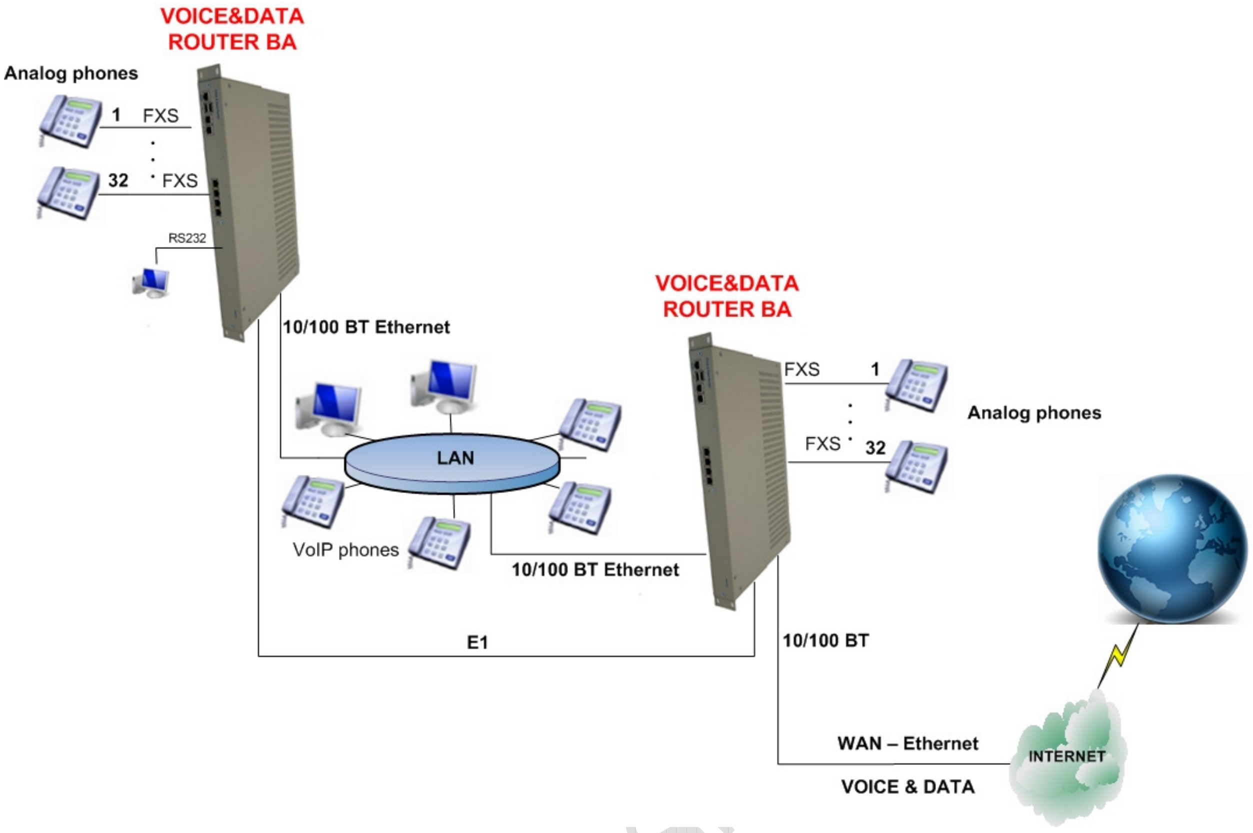

- **SIP PROXY Server**



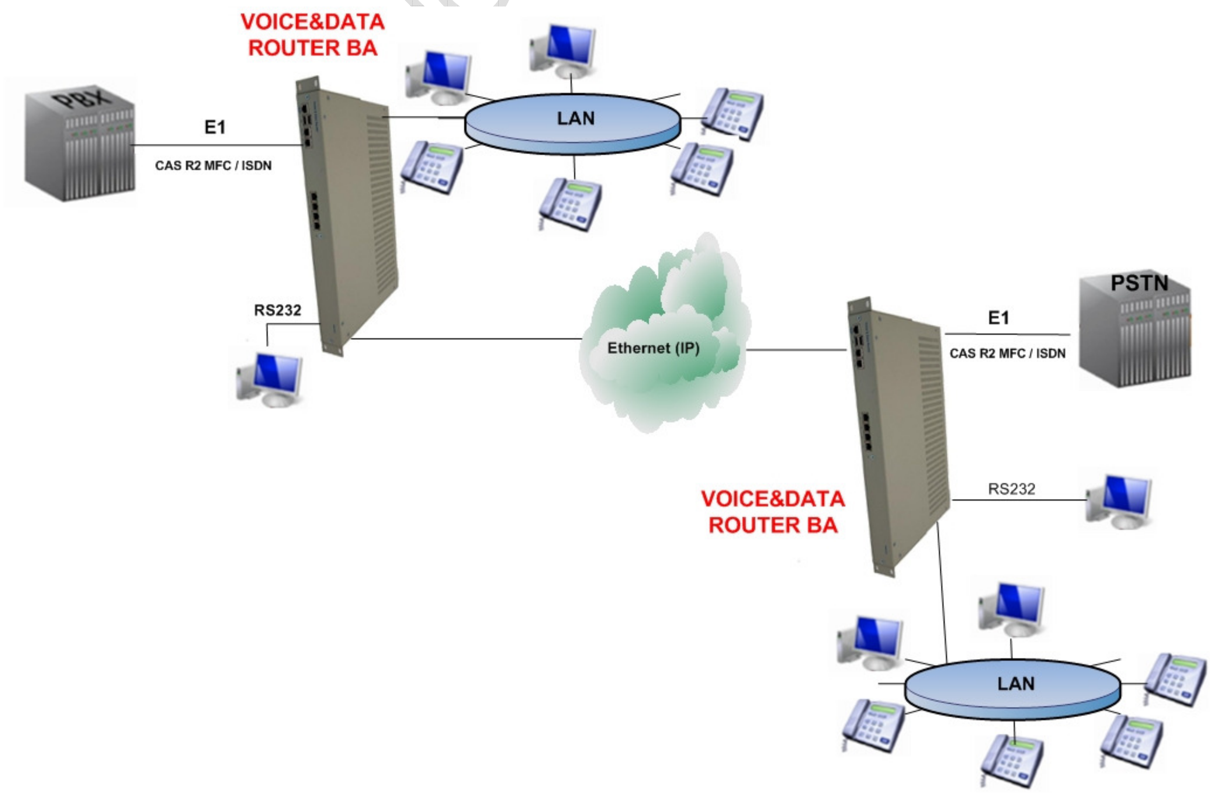
- **Local network, extending of local network**






- **E1 over IP**




***Different applications also available.***

# TECHNICAL SPECIFICATION

## Voice Processing

- GSM
- G.711
- G.726
- G.729 Annex A, Annex B (licence not included)
- Max. number of simultaneous calls (measured at INOTESKA ltd.):
  - Max. 16 calls for calls E1 - SIP or ANALOG - SIP (Echo Canceler - ON and used codecs else than G.711)
  - Calls between ANALOG and E1 are not limited.

Configuration: E1/VoIP

| SW Echo canceller | G.711 | G.729 |
|---|---|---|
| 0 ms (without echo canceller) | 24 calls | 15 calls |
| 4 ms | 16 calls | 13 calls |
| 8 ms | 14 calls | 11 calls |
| 16 ms | 12 calls | 9 calls |
| 32 ms | 10 calls | |

Configuration: E1+BRI/VoIP

| SW echo canceller | G.711 | G.729 |
|---|---|---|
| 0 ms (without echo canceller) | 18 calls | 12 calls |
| 4 ms | 12 calls | 8 calls |
| 16 ms | 8 calls | 7 calls |

**Note: Real data depend on various configurations, used features and codecs!**
(max. cca 10 concurrent calls)

- Timeslot interchange for TDM to TDM traffic processing
- Echo Cancellation
- Jitter buffer
- DTMF generation and detection
- Call progress tone generation
- Tones generation (ringing, busy, ...)
- Caller ID generation and detection
- AOC feature
- Announcement playback

## Generation level - software adjustable

- RTP – RFC 1889, RFC 2833, RFC 3389

### FAX and Modem Support

- G .711 Fax and Data Bypass

## IP Telephony Protocol

- SIP

### Signalling

- E1: ISDN DSS1 –ZAPTEL 1.4
    - 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

- ANALOG: FXS, E&M, OPX, - ZAPTEL 1.4
- BRI: DSS1
- ETH : SIP, - ASTERISK 1.4

### Voice Routing

- Local switching
- Interface hunt groups
- Routing Criteria:
    - Interface
    - Calling / called party number
    - Time of day, day of week, date
- Number manipulation functions
- Replace numbers
- Add / remove digits
- Multiple remote gateways



### Data Routing

- Basic Routing

### QoS Marking

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

### Networking

- DHCP support and capabilities
- Static Routing
- VLSM

## Management

Device is supplied with default configuration. Device can be configured using:

- via Ethernet SSH, FTP,http
- Web manager
  - default password: **inoteska**

**Please contact the producer in case of difficulties!**

## Power Supply

19", 1U rack mount

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

## 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)

# 2. OPERATING INSTRUCTIONS

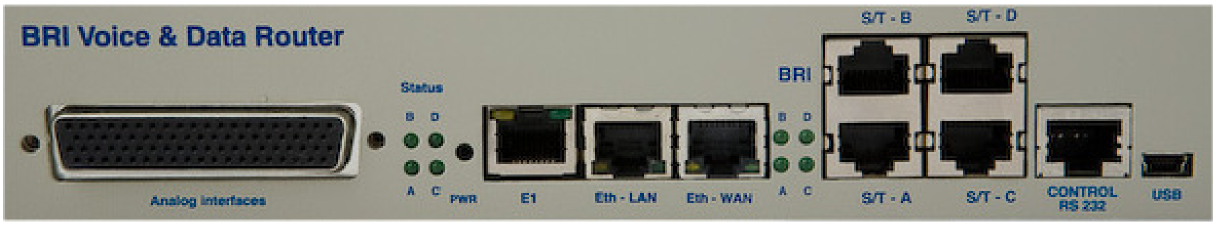
## Operating Environment

Install the device in a place where:

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

## Interfaces

Connect cables to the appropriate connectors.






**E1/T1 interface**

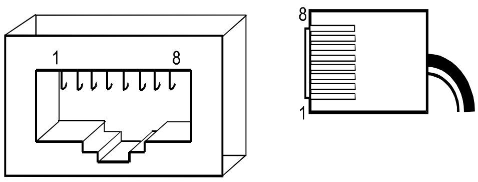
**RJ 45 connector**

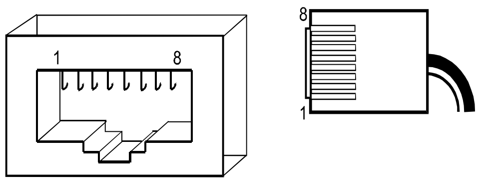
| | | |
|---|---|---|
| 1 – | input wire to device | Rx - |
| 2 – | input wire to device | Rx+ |
| 3 – | | |
| 4 – | output wire from device | Tx - |
| 5 – | output wire from device | Tx+ |
| 6 – | | |
| 7 – | | |
| 8 – | | |

**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 – | | | |




**BRI interface**
- standardly connected as TE

**GSM interface**
- antenna connector: GSC connector coaxial SMT male 50R 6GHz

**Analog interfaces**

Main card can contain 1 up to 8 modules.

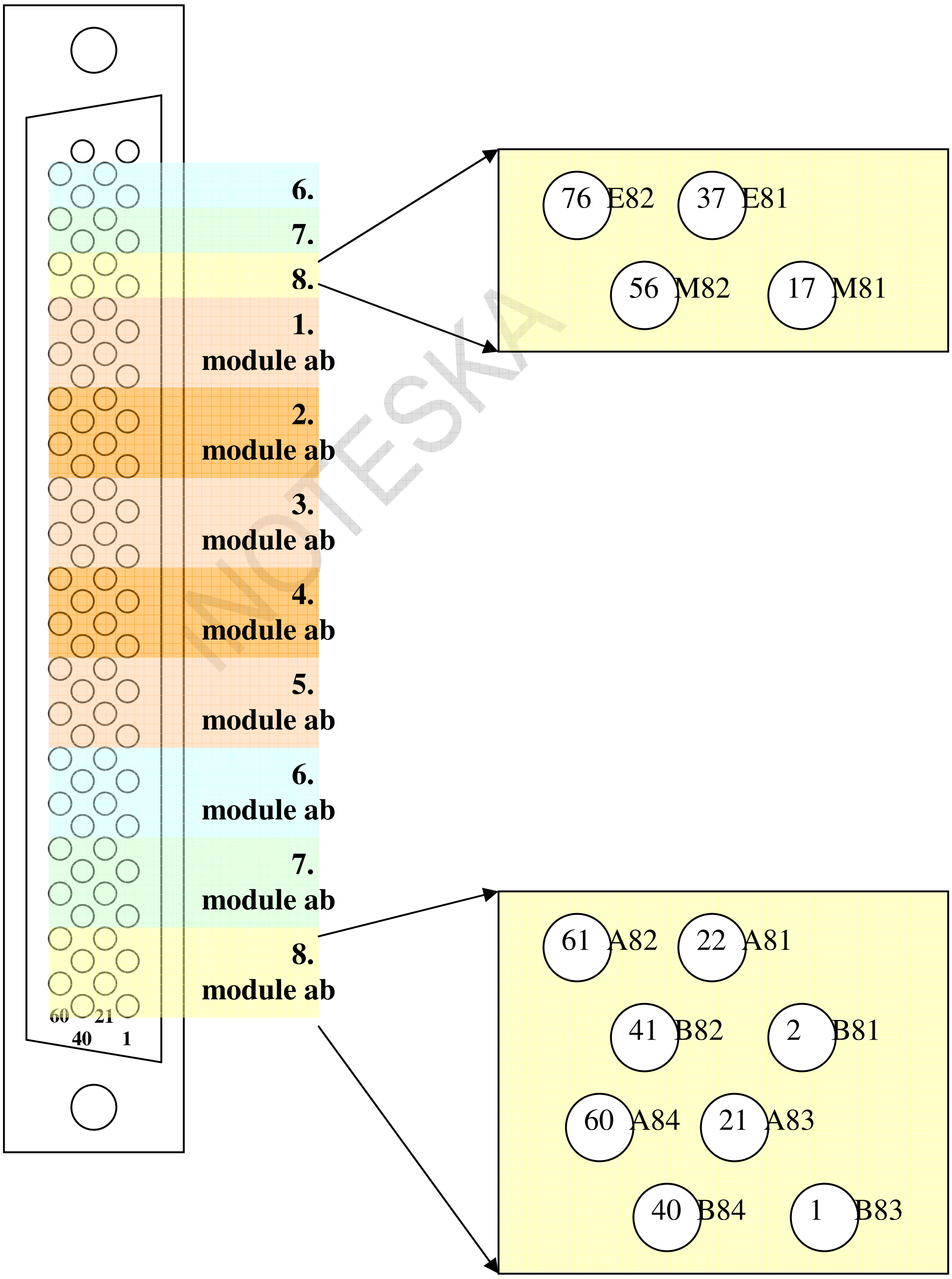
Position no. 1 to 5: for FXS/FXO/LB modules
Position no. 6 to 8: for FXS/FXO/LB/E&M modules

FXS module – quad
LB module – dual

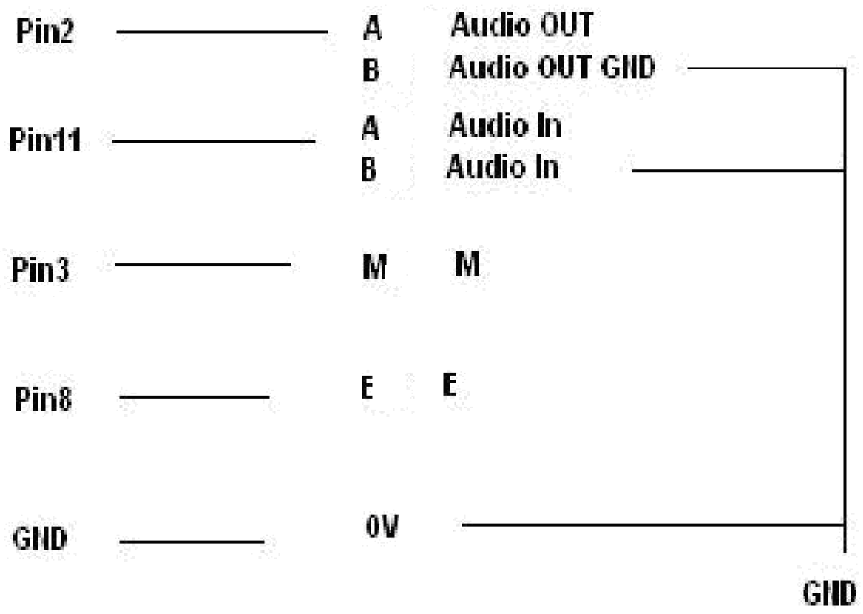
FXO module – dual
E&M module – dual

Analog interfaces can be connected:

**1. using J2 connector** (standard product variant)



*Module ITP 182 31 - Radio connecting*




Description of cable

1. Interface

| Signal | PIN | Meaning |
|---|---|---|
| A61 | 26 | Audio Out 1 |
| B61 | 6 | Audio Out GND 1 |
| A62 | 65 | Audio In 1 |
| B62 | 45 | Audio In GND1 |
| E61 | 39 | E 1 |
| M61 | 19 | M 1 |
| 0V | 20 | GND |

2.Interface

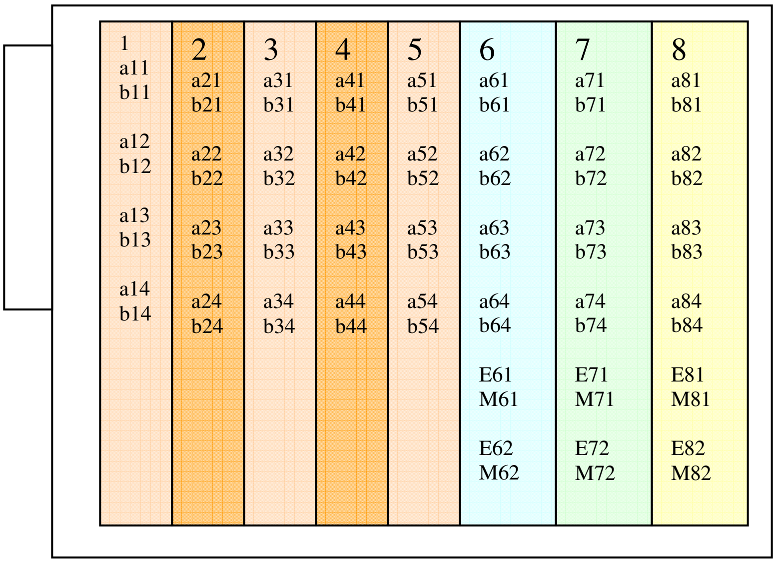
| Signal | PIN | Meaning |
|---|---|---|
| A63 | 25 | Audio Out 2 |
| B63 | 5 | Audio Out GND 2 |
| A64 | 64 | Audio In 2 |
| B64 | 44 | Audio In GND 2 |
| E62 | 78 | E 2 |
| M62 | 58 | M 2 |
| 0V | 59 | GND |

3. Interface

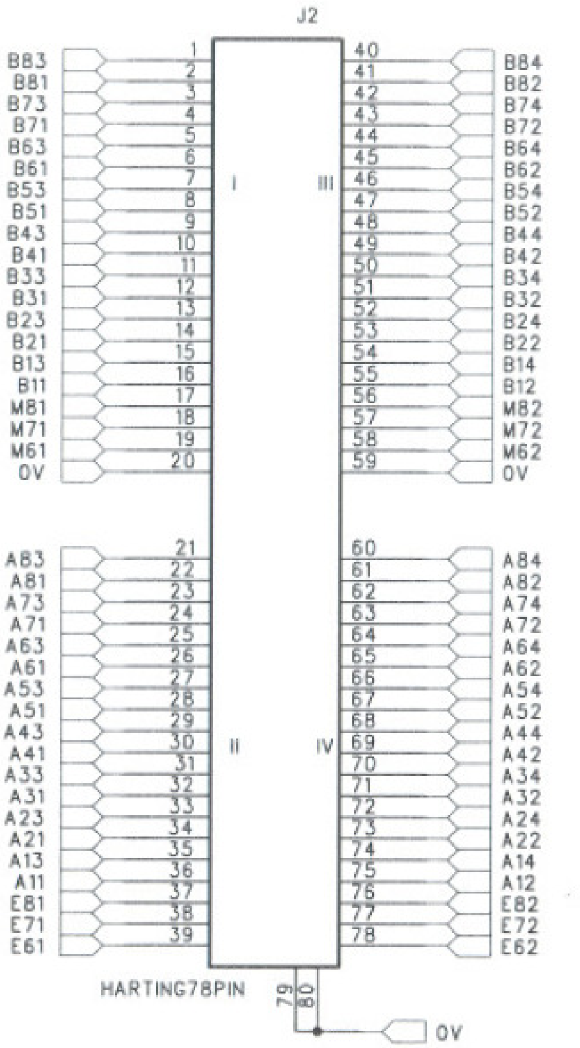
| Signal | PIN | Meaning |
|---|---|---|
| A71 | 24 | Audio Out 3 |
| B71 | 4 | Audio Out GND 3 |
| A72 | 63 | Audio In 3 |
| B72 | 43 | Audio In GND 3 |
| E71 | 38 | E 3 |
| M71 | 18 | M 3 |
| 0V | 20 | GND |

4.Interface

| Signal | PIN | Meaning |
|---|---|---|
| A73 | 23 | Audio Out 4 |
| B73 | 3 | Audio Out GND 4 |
| A74 | 62 | Audio In 4 |
| B74 | 42 | Audio In GND 4 |
| E72 | 77 | E 4 |
| M72 | 57 | M 4 |
| 0V | 59 | GND |




*Location of analog modules on main card*




*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).

**2. using RJ45 connector**

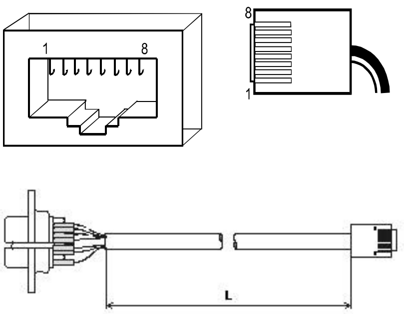
**-** modified product variant for max. 16 interfaces


**Connectors for FXS interface are marked as following:**

| | |
|---|---|
| **1 – 1. interface on 1. module** | **4 – 5** |
| **2 – 2. interface on 1. module** | **4 – 5** |
| **3 – 3. interface on 1. module** | **4 – 5** |
| **4 – 4. interface on 1. module** | **4 – 5** |
| | |
| **5 – 1. interface on 2. module** | **4 – 5** |
| **6 – 2. interface on 2. module** | **4 – 5** |
| **7 – 3. interface on 2. module** | **4 – 5** |
| **8 – 4. interface on 2. module** | **4 – 5** |
| | |
| **9 – 1. interface on 3. module** | **4 – 5** |
| **10 – 2. interface on 3. module** | **4 – 5** |
| **11 – 3. interface on 3. module** | **4 – 5** |
| **12 – 4. interface on 3. module** | **4 – 5** |
| | |
| **13 – 1. interface on 4. module** | **4 – 5** |
| **14 – 2. interface on 4. module** | **4 – 5** |
| **15 – 3. interface on 4. module** | **4 – 5** |
| **16 – 4. interface on 4. module** | **4 – 5** |



**CONTROL connector**




Cannon D09F
+ shield

RJ45

| 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 Receive, Transmit) | **Line active** |

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 datalink layer active

**AIS -** Alarm Indication Signal – Transmitted signal is constant with data value Log1

**LFA -** Loss of Frame Alignment – Indicates synchronisation error in $0^{th}$ channel

**RRA -** Receive Remote Alarm – Indicates remote device alarm

# 3. MANAGEMENT

Device is supplied with default configuration. Device can be configured using:

**A) Terminal access**
- local via RS232 interface (default Baud rate 115.200 kbps, 8 bit, 1 stop bit, none parity)
- remote using SSH
  default password: **inoteska**

***Please contact the producer in case of difficulties!***

**B) Inoteska Web manager - GUI**
GUI provides simple and user-friendly interface for device configuration and supervision using web. More detailed information please see section 3.1 Web manger.

## 3.1 Web manager

GUI for device configuration and supervion. Web manager was tested with following browsers: **Opera, Google Chrome, Mozilla Firefox**.

Initial screen:

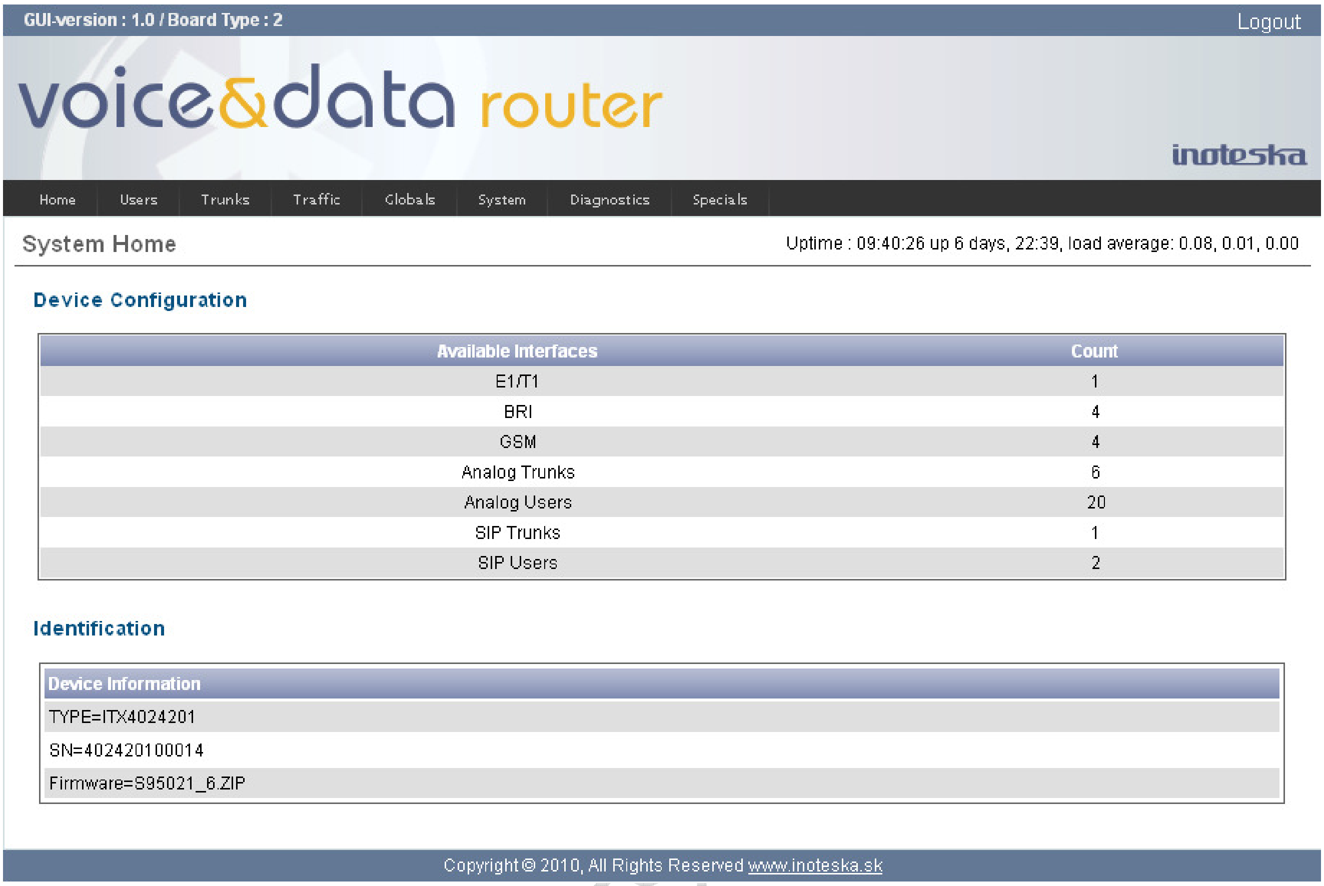

Default login data:

**Username:** admin
**Password*:** inoteska
* Change of password can be done in menu **System – Options**.

Default IP address:
LAN: http://192.168.1.100
WAN: http://10.1.1.100

Then you can see **Available interfaces** and their **count**.

The number of Analogue, GSM, BRI interfaces depends on real HW configuration. SIP users and SIP trunks shows the number of configured interfaces by device user.




## How to configure the device

Configuration of **Users** / **Trunks** depends on usage application.

For Voice&Data Router as Gateway – it is necessary to configure **Trunks**, if there are analog interfaces, configure also **Users**.

For Voice&Data Router as PBX – it is necessary to configure **Users** and **Trunks**.

First define **Users** (***SIP, analog***) and **Trunks** (***SIP, E1/T1, BRI, Analog, GSM interfaces***) – as per your device function. Then set **Traffic** (***Dial Plan, Calling Rules***). Go back to **Users/Trunks** to assign ***Dial Plans*** and ***Calling Rules*** to **Users/Trunks**.

Then configure all other necessary parameters available in Web manager.

Please note that all changes you made are being applied to device after click on **Apply Changes** in upper right corner. Some settings (e.g. change of IP address, restore of saved configuration) are done automatically,without need to **Apply Changes.**

For configuration choose the item from main menu:

**Users** – SIP Users<br>Analog Users

**Trunks** – SIP Trunks<br>E1/T1 trunks<br>BRI Trunks<br>Analog Trunks<br>GSM Trunks

**Traffic** – Calling Rules
Dial plans
AOC Tables Settings

**Globals** – SIP Settings
RTP/UDPTL Settings
Call Groups
Ring Groups
Carriers
Permissions
Services Codes
CDR Settings
Announcement Settings

**System** – IP settings
Routing Tables
Firewall
Options
Backup&Upload
Firmware Update
Date&Time

**Diagnostics** – Port Status
Active Services
Identification
System Logs
CLI Emulator
Debug messages
System Status
File Editor
IP Tables Tests
SMS Manager Tests

**Specials** – AOC Web Update

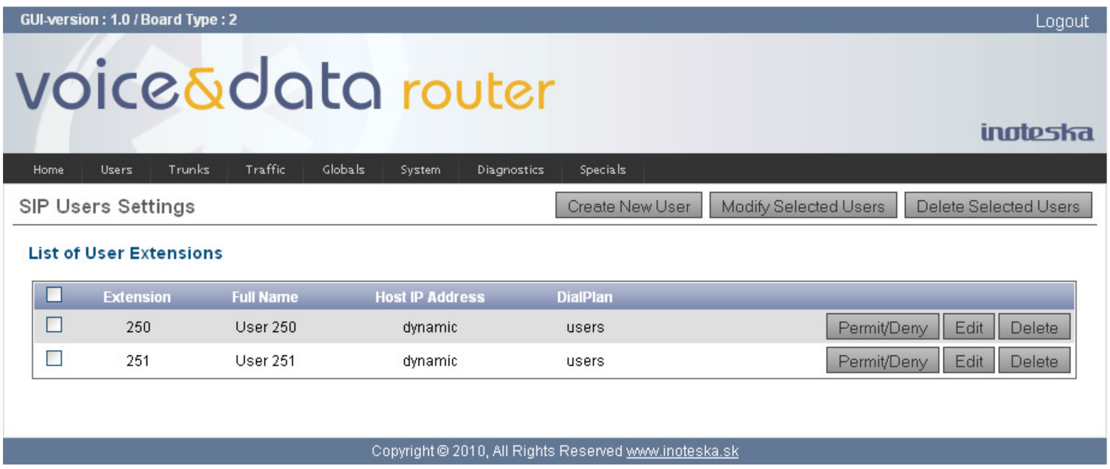
- moving a mouse pointer here you can see help/description of parameter/settings.



### 3.1.1 Users

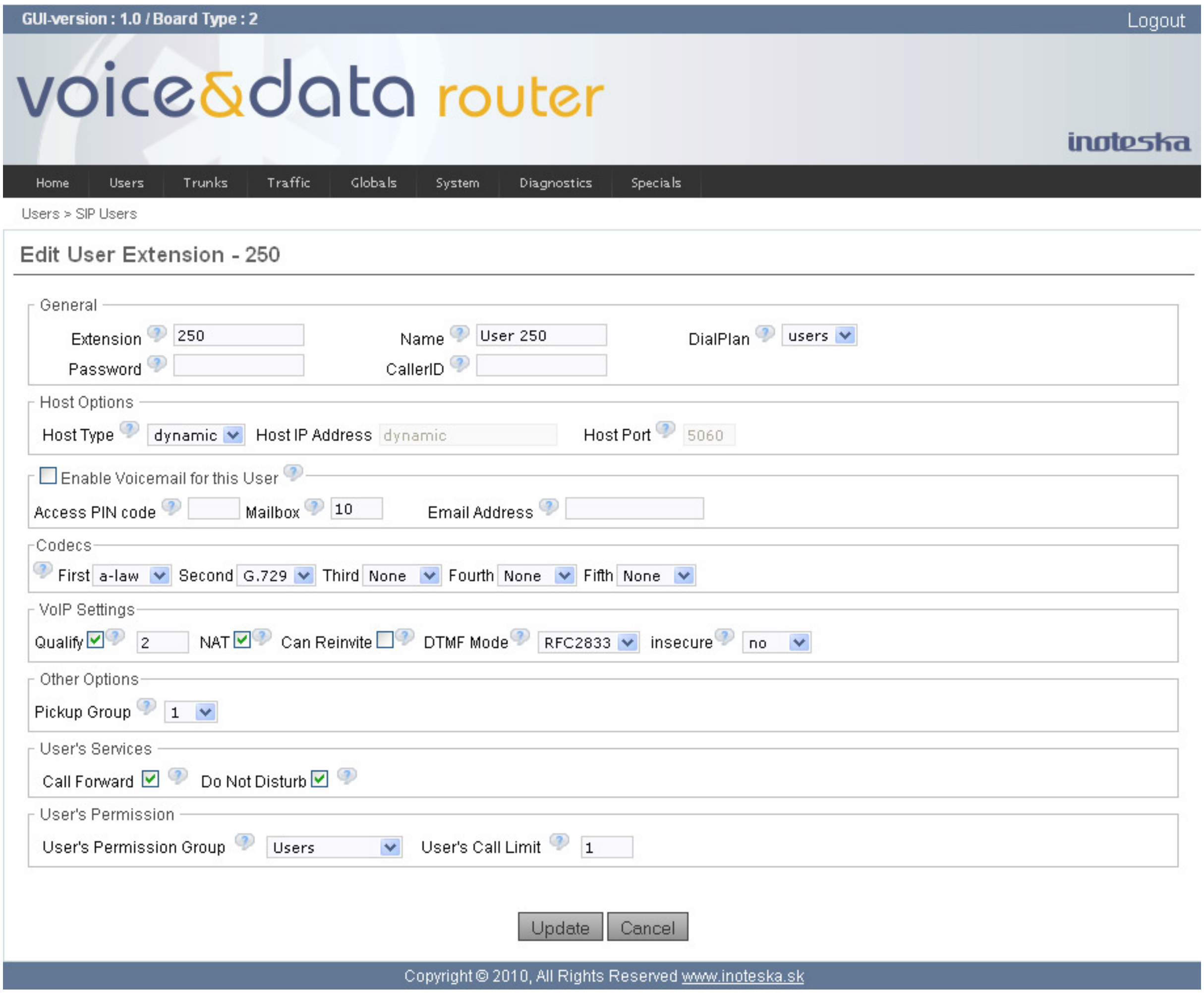
Here you can define **Users – SIP** and **Analog.**

**SIP users**



Click on **Create New User, Modify Selected Users, Delete Selected Users** to manage list of SIP users.

It is possible to manage each **User** individually by **Permit/Deny** (Permit/Deny traffic from set IP address or all IP adresses – for extensions with ***dynamic*** IP address – more information after click on Permit/Deny)**, Edit , Delete.**




## General

**Extension:** The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.
**Name:** A character-based name for this user, i.e. Bob Jones
**Dial Plan:** Please choose the DialPlan for this user. DialPlans are sets of calling rules and can be managed form the \"Dial Plans\" panel.";
**Password:** The password for the user's sip/iax account , Ex: \"12u3b6\
**Caller ID:** The Caller ID (CID) string used when this user calls another internal user.

## Host Options

**Host Type:** How to find the client - IP # or host name. If you want the phone to register itself, use the keyword dynamic instead of Host IP
**Host IP address**
**Host Port:** SIP port of the klient

## Enable Voicemail for this User

- Check this box if the user should have a voicemail account.
**Access PIN Code:** Voicemail Password for this user, ex: 1234
**Mailbox:** Voicemail Mailbox for this user. It can be different from the extensions numer.
**Email Address:** The e-mail address for this user, i.e.bobjones@bobjones.com

## Codecs

Allow codecs in order of preference. Codec is a compression or decompression algorithm run against voice as it is moved between analog (speaking) and digital (VoIP).

**u-law** - A PSTN standard codec, used in North America, that provides very good voice quality and consumes 64kbit/s for each direction (receiving and transmitting) of a VoIP call. u-law should be supported by all VoIP phones.

**a-law -** A PSTN standard codec, used outside of North America, that provides very good voice quality and consumes 64kbit/s for each direction (receiving and transmitting) of a VoIP call. a-law should be supported by all VoIP phones.

**GSM -** A wireless standard codec, used worldwide, that provides okay voice quality and consumes 13.3kbit/s for each direction (receiving and transmitting) of a VoIP call. GSM is supported by many VoIP phones.

## VoIP Settings

**Qualify:** 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 (or default) period (in sec) Asterisk considers the device off-line for future calls. Default Value '2 sec

**NAT:** Try this setting when Asterisk is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports.

**Can Reinvite:** By default, Asterisk will route the media steams from SIP endpoints through itself. Enabling this option causes asterisk to attempt to negotiate the endpoints to route the media stream directly, bypassing asterisk. It is not always possible for asterisk to negotiate endpoint-to-endpoint media routing.

**DTMF Mode:** Set default dtmf mode for sending DTMF. Default: rfc2833

**Insecure:** Port allows matching of peers by IP address without matching port number. Invite removes the requirement for authentication of incoming INVITE messages. Port,Invite allows both the matching of peers by IP address without matching port number and removes the requirement for authentication of incoming INVITE messages. No requires normal IP-based matching and authenticated INVITES



## Other Options

**Pickup Group:** Group that can pickup fellow workers' calls

## User´s Services:

**Call Forvard:** Call Forvard Service

**Do Not Disturb:** Do Not Disturb Service

## User´s Permission

**User´s Permission Group:** Select User´s Permission Group

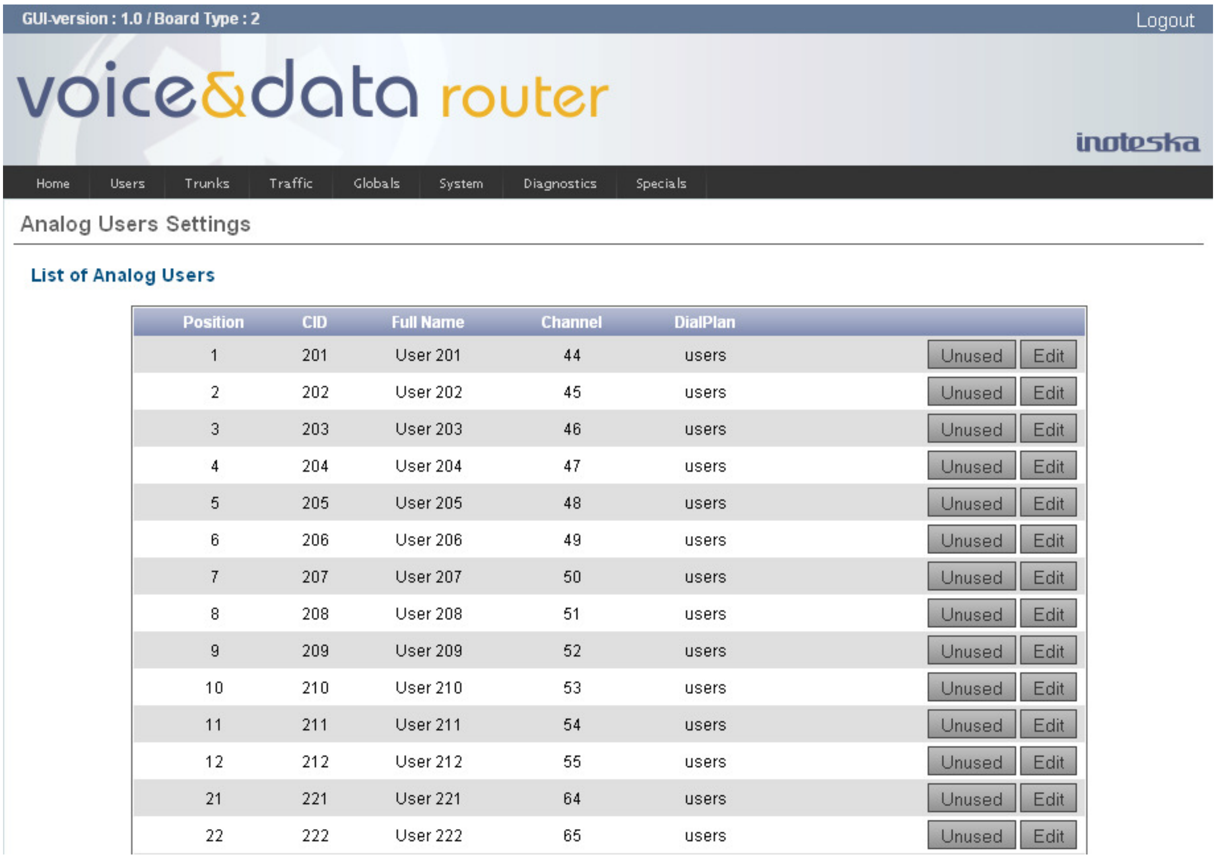
**User´s Call Limit:** Input User´s Call Limit value

### Edit Permit/Deny Option

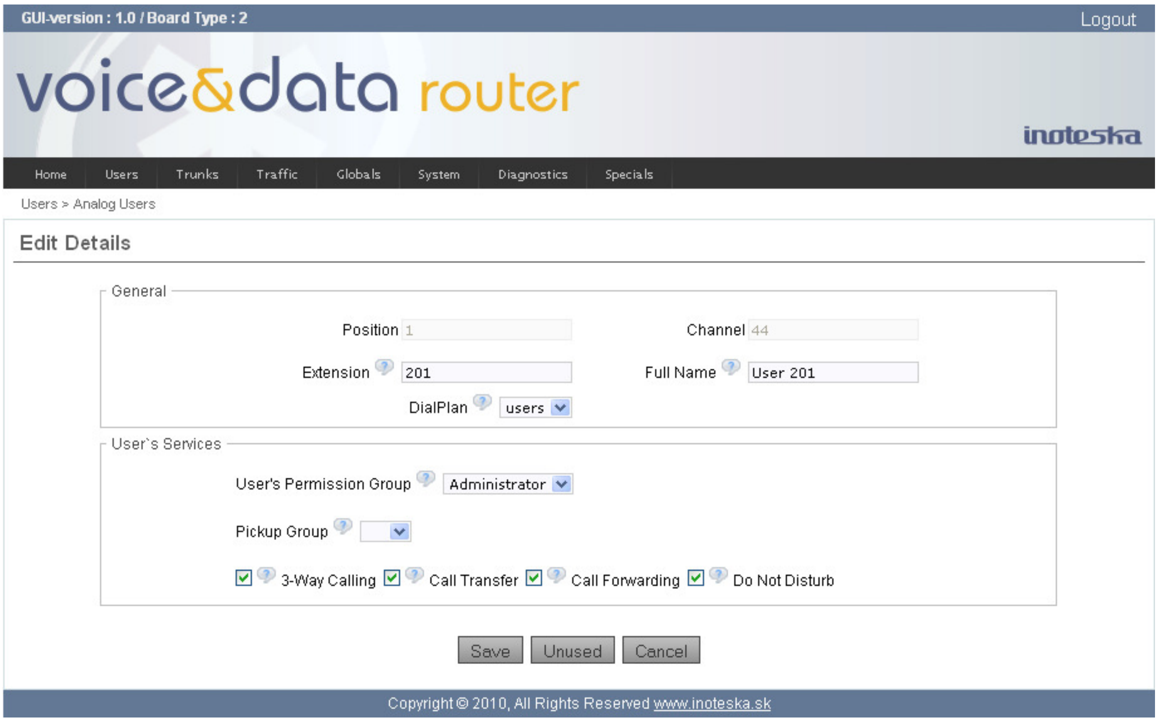
**IP:** Permit or Deny traffic from this IP address, to Deny or Permit all use 0.0.0.0 for the IP and 0.0.0.0 for the Net Mask.
**Network Mask:**To Permit or Deny only the IP set in the IP field use a Net Mask of 255.255.255.255, to specify a range of IP's to Permit or Deny decrease the Net Mask, for example to Permit or Deny IP's from 192.168.1.138 to 192.168.1.142 you should specify 192.168.1.138 as the IP and 255.255.255.248 as the Net Mask

## Analog users

List of all available analog HW positions is displayed. You can manage each analog user individually by **Unused, Edit.**




| Position | CID | Full Name | Channel | DialPlan | |
|---|---|---|---|---|---|
| 1 | 201 | User 201 | 44 | users | Unused Edit |
| 2 | 202 | User 202 | 45 | users | Unused Edit |
| 3 | 203 | User 203 | 46 | users | Unused Edit |
| 4 | 204 | User 204 | 47 | users | Unused Edit |
| 5 | 205 | User 205 | 48 | users | Unused Edit |
| 6 | 206 | User 206 | 49 | users | Unused Edit |
| 7 | 207 | User 207 | 50 | users | Unused Edit |
| 8 | 208 | User 208 | 51 | users | Unused Edit |
| 9 | 209 | User 209 | 52 | users | Unused Edit |
| 10 | 210 | User 210 | 53 | users | Unused Edit |
| 11 | 211 | User 211 | 54 | users | Unused Edit |
| 12 | 212 | User 212 | 55 | users | Unused Edit |
| 21 | 221 | User 221 | 64 | users | Unused Edit |
| 22 | 222 | User 222 | 65 | users | Unused Edit |




**General**
**Extension:** Caller ID
**Full name:** A character-based name for this user, i.e. Bob Jones
**Dial Plan:** Please choose the DialPlan for this user. DialPlans are sets of calling rules and can be managed from the **Dial Plans** panel

**User´s Services**
**User´s Permission Group:** Select User´s Permission Group
**Pickup Group:** Group that can pickup fellow workers' calls
**3-Way Calling:** Check this option if Phone should have 3-Way calling capability
**Call Transfer:** You can transfer the call to another extension
**Call Forwarding:** Call Forward Service
**Do Not Disturb:** Do Not Disturb Service

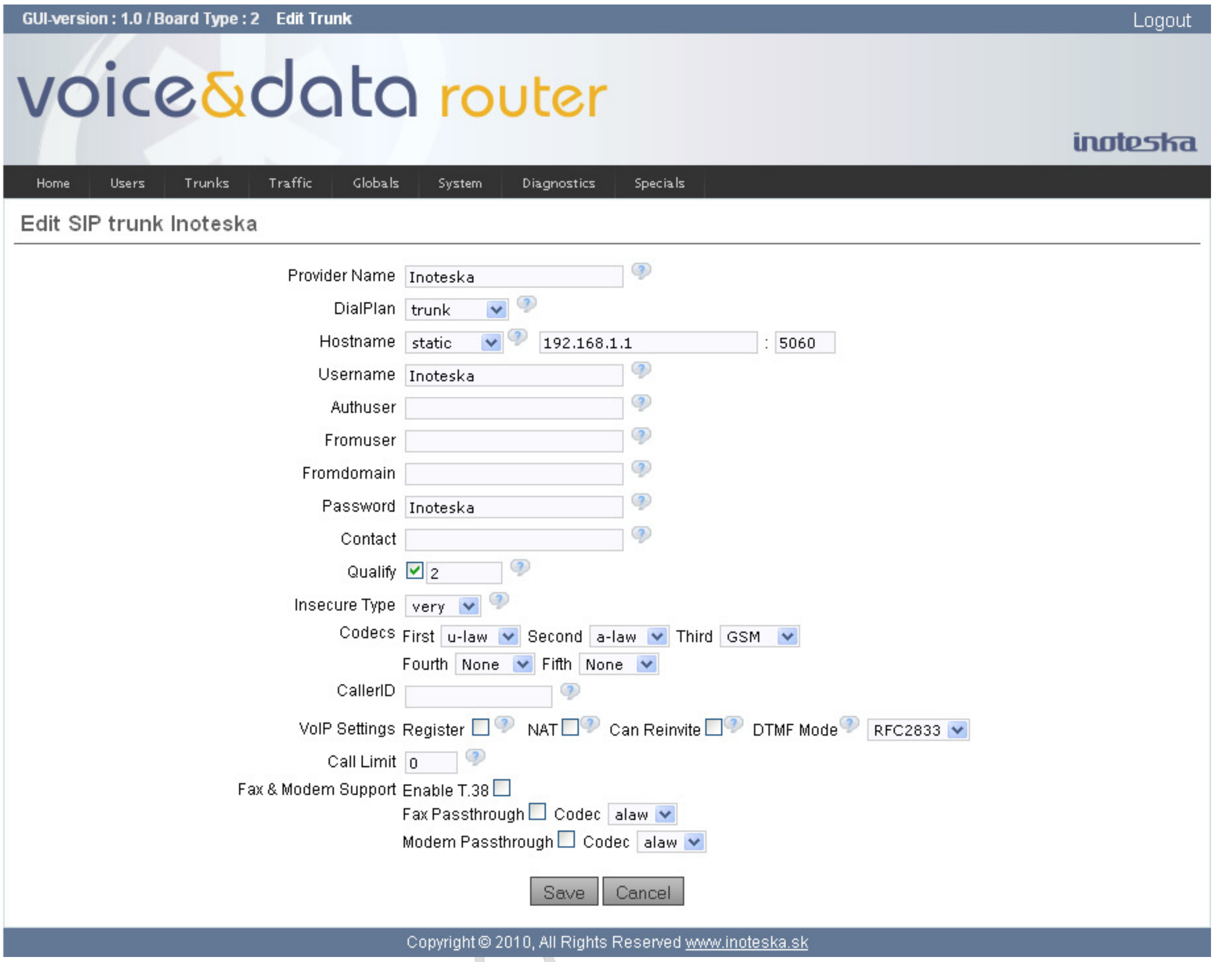


### 3.1.2 Trunks

It is possible to define the parameters´ settings for:

**SIP Trunks**

You can add a **New SIP Trunk**, **Edit** and **Delete** existing ones.

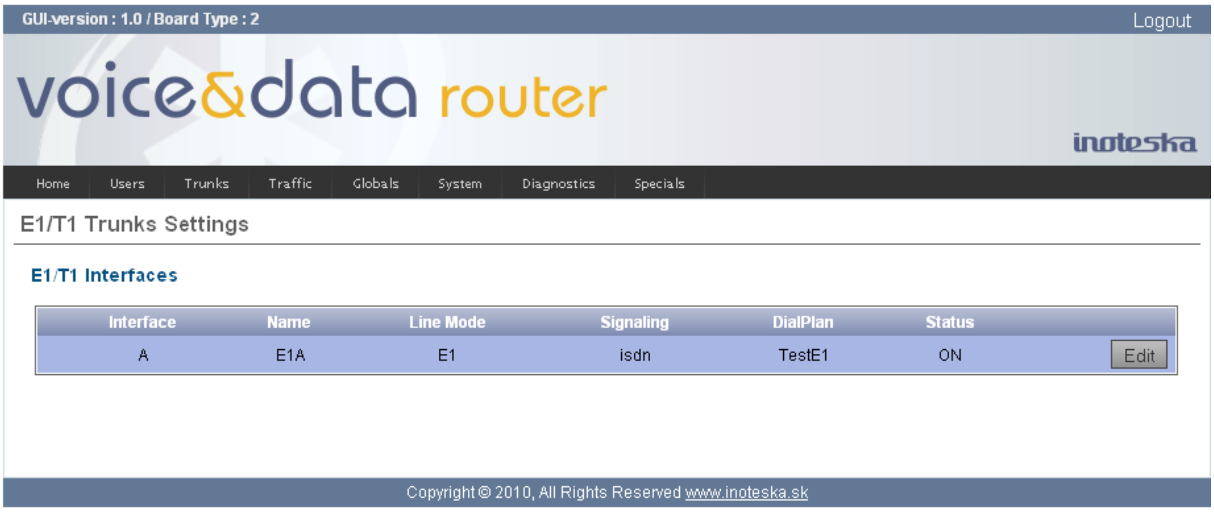



**Provider Name:** A unique label to help you identify this trunk when listed in outbound rules, incoming rules etc.
**Dial Plan:** DialPlan for this Trunk.  DialPlans are sets of calling rules and can be managed form the \"Dial Plans\" panel.
**Hostname:** How to find the SIP server or Provider - IP # or host name. If you want to register itself, use the keyword dynamic instead of Host IP
**Username:** SIP channel username. This field specifies the user name for authentication
**Authuser:** username for autentification, if different from username
**Fromuser:** Name which will be used in SIP header 'From' instead of caller identification. Overrides the callerid
**Fromdomain:** Allows to set IP address or domain which will be used in SIP header 'From'
**Password: Password for autentification**
**Contact:** SIP Url Contact
**Qualify:** 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 (or default) period (in sec) Asterisk considers the device off-line for future calls. Default Value '2 sec
**Insecure Type:** Port allows matching of peers by IP address without matching port number. Invite removes the requirement for authentication of incoming INVITE messages. Port,Invite allows both the matching of peers by IP address without matching port number and removes the requirement for authentication of incoming INVITE messages. No requires normal IP-based matching and authenticated INVITES
**Codecs:** Allow codecs in order of preference.  Codec is a compression or decompression algorithm run against voice as it is moved between analog (speaking) and digital (VoIP).
**CallerID:** This is the number that the trunk will try to use when making outbound calls. For some providers it is not possible to set the CallerID with this option, and this option might be ignored. When making outgoing calls the following rules are used to determine which CallerID will be used, if they exist: The first CallerID used is a CallerID set for the user making the call defined in the 'Users'

tab.The second CallerID is the one that is set in the 'VoIP Trunks' configuration, if applicable.The last CallerID used for outgoing calls is the Global CID defined in the 'Options' tab.
**VoIP Settings Register:** Enables to register SIP channel at remote provider.
**NAT:** Try this setting when Asterisk is on a public IP, communicating with devices hidden behind a NAT device (broadband router). If you have one-way audio problems, you usually have problems with your NAT configuration or your firewall's support of SIP+RTP ports.
**Can Reinvite:** By default, Asterisk will route the media steams from SIP endpoints through itself. Enabling this option causes asterisk to attempt to negotiate the endpoints to route the media stream directly, bypassing asterisk. It is not always possible for asterisk to negotiate endpoint-to-endpoint media routing.
**DTMF Mode:** Set default dtmfmode for sending DTMF.
Default: rfc2833H
Other options:
info : SIP INFO messages
inband : Inband audio (requires 64 kbit codec -alaw, ulaw)
auto : Use rfc2833 if offered, inband otherwise
**Call Limit:** The maximum count of calls.

## E1/T1 trunks

List of available E1/T1 interfaces is displayed.

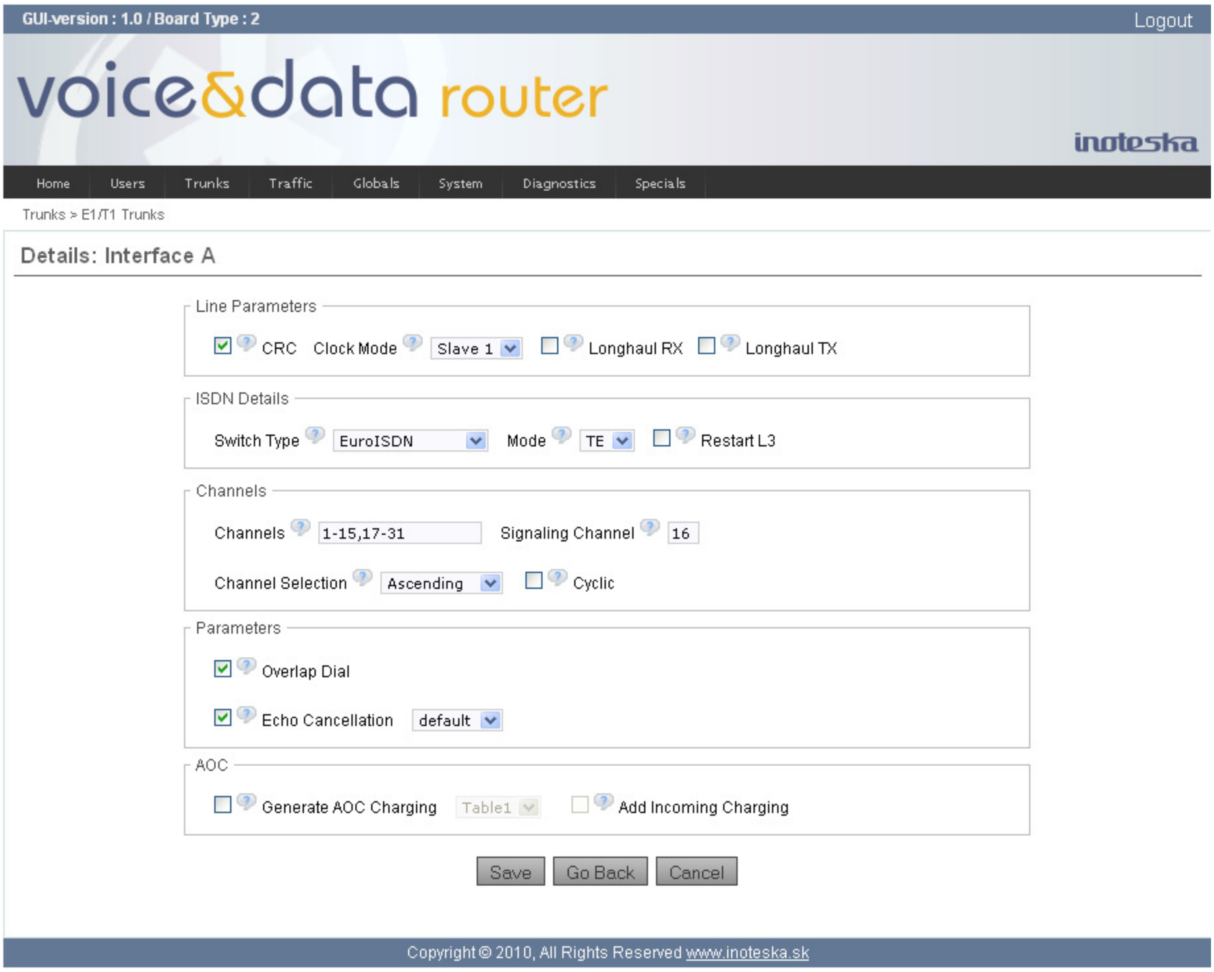



## General Settings

**Enabled:** Enable/Disable E1/T1 interface
**Name:** A unique interface name to help you identify this trunk
**Line Mode:** Interface line mode selection
E1 ... 2.048 Mb/s line with 30 voice channels.
T1 ... 1.544 Mb/s line with 23 voice channels.
**Signaling:** Signaling type selection. Changing of signaling type resets signaling dependent parameters to their default values. Use details button to check/edit these parameters.

## Details

*Extended settings* are available after click on **Details**.




## Line Parameters

**CRC:** Enable CRC4 multiframe format of framing mode.
**Clock Mode:** Synchronization clock source selection.
Master ... device provides synchronization clock to the line.
Slave 1,2 ... device is synchronized by the clock from the line. Slave 1 has higher priority than Slave 2.
**Longhaul RX:** This option enables to increase the device radius by setting the receiving more sensitive. Long haul parameter is within G.703 specification, that means it is also possible to connect a standard device to the device with long haul.
**Longhaul TX:** This option enables to increase the device radius by setting transmitting more intense. Long haul parameter is within G.703 specification, that means it is also possible to connect a standard device to the device with long haul.

### ISDN Details

**Switch Type:** Select ISDN signaling variant.
**Mode:** Select ISDN signaling mode.
NT ... network side
TE ... terminal (CPE) side

**Restart L3:** Enable automatic restart procedure on 3rd layer for all voice channels. Restart procedure starts after D-channel comming up and repeats every hour.

### Channels

**Channels:** Available voice channels.
E1 line ... use channel range from 1 to 31.
T1 line ... use channel range from 1 to 24.
Avoid of collision with signaling channel.
**Signaling Channel:** Selection of signaling (ISDN D-channel) channel number.
**Channel Selection:** Method of the selection of outgoing voice channel.
Ascending ... search for the lowest available channel number.
Descending ... search for the highest available channel number
**Cyclic:** Enable cyclic search of available outgoing channel instead of sequential.

### Parameters

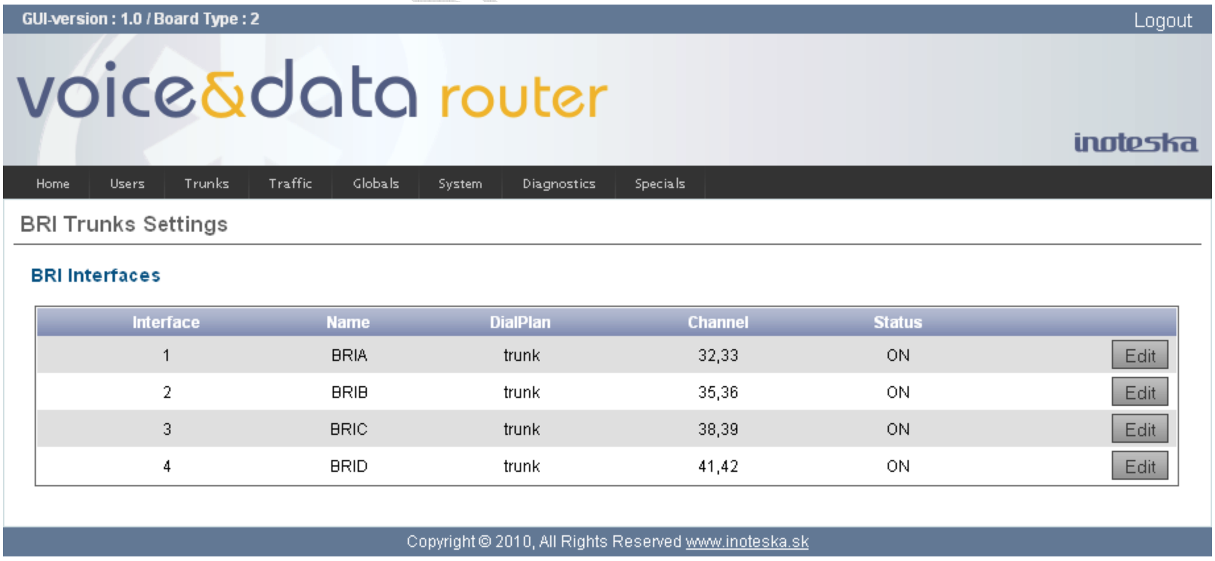
**Overlap Dial:** Enable overlap dialing (consecutive dialing). If not enabled, complete called number in SETUP is required.
**Echo Cancellation:** Enable echo canceller and select echo canceller tail length. Default setting of echo canceller tail length is 64 milliseconds.

### AOC

**Generate AOC Charging:** Enable generation of AOC metering pulses according selected AOC table.
**Add Incoming Charging:** If enabled metering pulses received from destination will be added to metering pulses generated according AOC table.

## BRI Trunks

List of BRI interfaces is displayed.




| Interface | Name | DialPlan | Channel | Status | |
|---|---|---|---|---|---|
| 1 | BRIA | trunk | 32,33 | ON | Edit |
| 2 | BRIB | trunk | 35,36 | ON | Edit |
| 3 | BRIC | trunk | 38,39 | ON | Edit |
| 4 | BRID | trunk | 41,42 | ON | Edit |

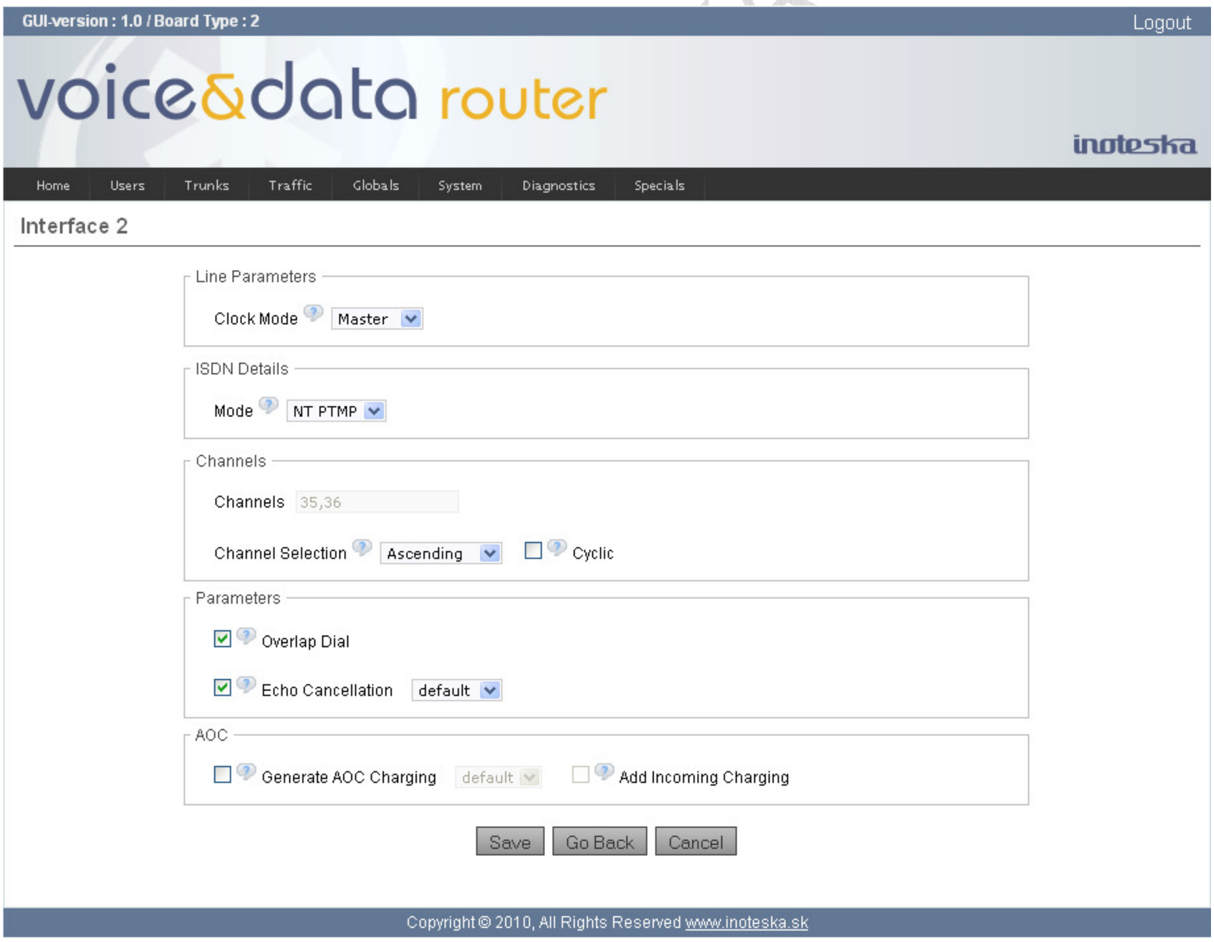
Click **Edit** to set **BRI Trunks** parameters.


## General Settings

**Enabled:** Enable/Disable BRI Interface
**Name:** Interface Name
**DialPlan:** Dialplan for this BRI Trunk

## Details

*Extended settings* are available after click on **Details**.



### Line Parameters

**Clock Mode:** Synchronization clock source selection.
Master ... device provides synchronization clock to the line.
Slave 1,2 ... device is synchronized by the clock from the line. Slave 1 has higher priority than Slave 2.

### ISDN Details

**Mode:** Select ISDN Signaling Mode:
TE - Terminal side (Point to Point) CPE
TEPTMP - Terminal side (Point to Multipoint)
NT - Network side (Point to Point) CO
NTPTMP - Network side (Point to Multipoint)

### Channels

**Channel Selection:** Method of the selection of outgoing voice channel.
Ascending ... search for the lowest available channel number.
Descending ... search for the highest available channel number
**Cyclic:** Enable cyclic search of available outgoing channel instead of sequential.

### Parameters

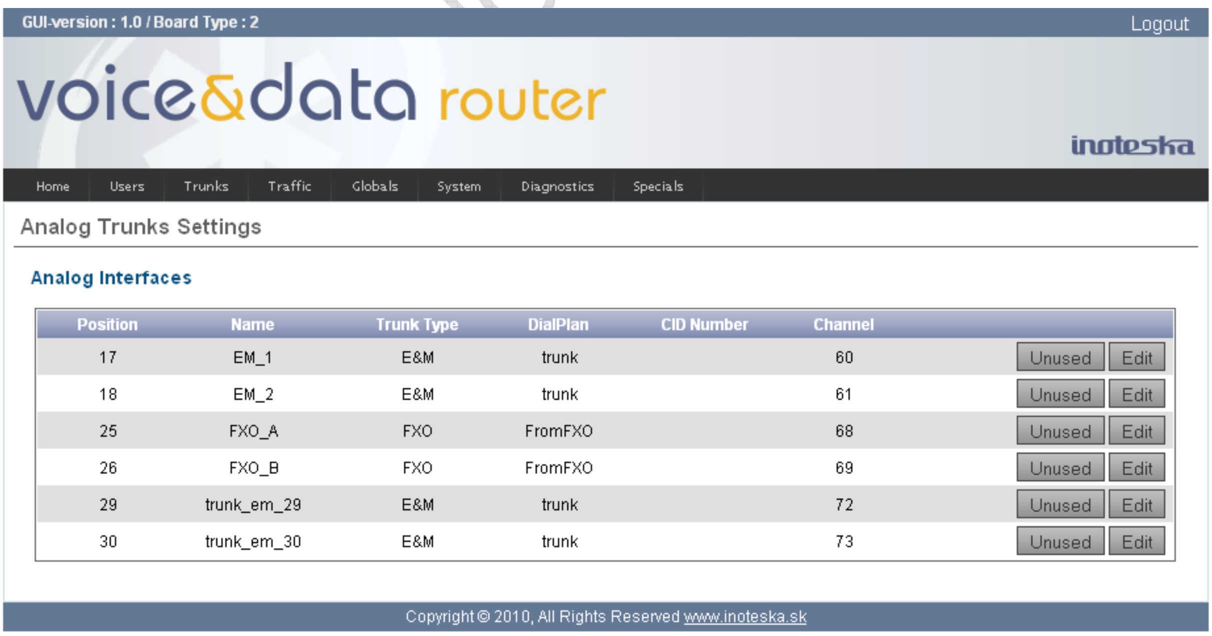
**Overlap Dial:** Enable overlap dialing (consecutive dialing). If not enabled, complete called number in SETUP is required.
**Echo Cancellation:** Enable echo canceller and select echo canceller tail length. Default setting of echo canceller tail length is 64 milliseconds.

### AOC

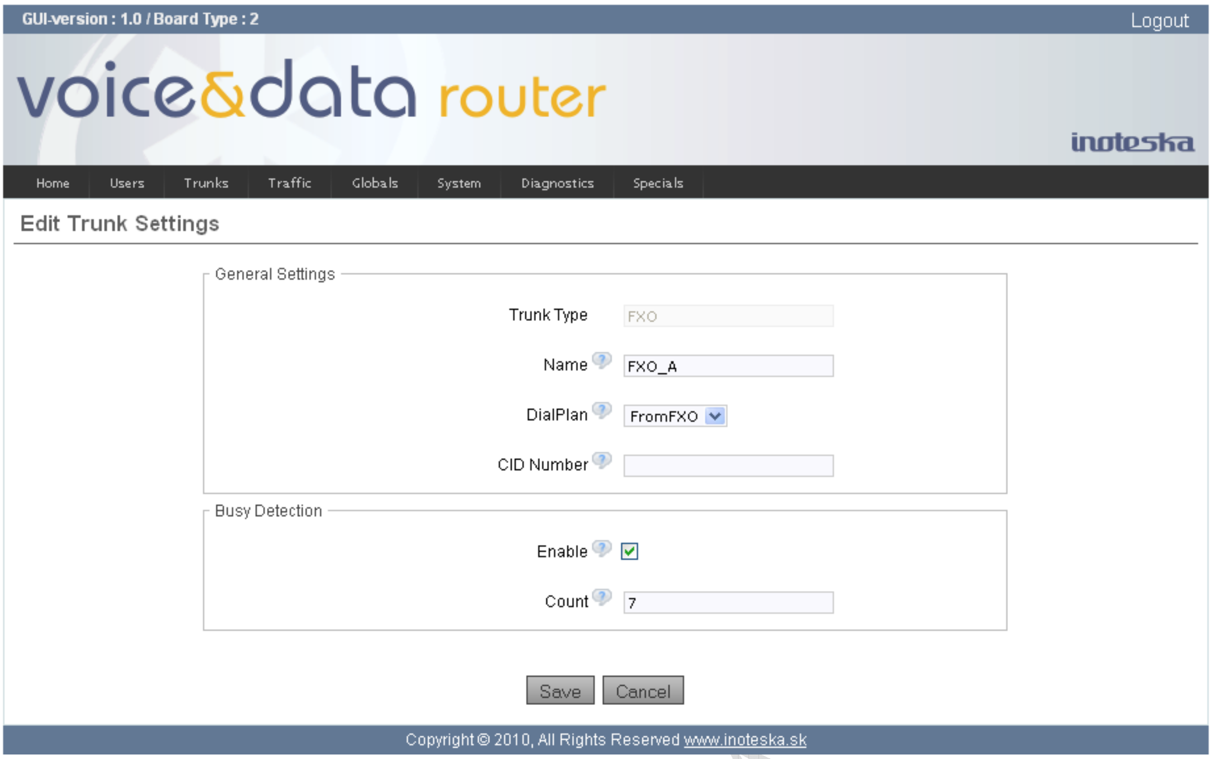
**Generate AOC Charging:** Enable generation of AOC metering pulses according selected AOC table.
**Add Incoming Charging:** If enabled metering pulses received from destination will be added to metering pulses generated according AOC table.

## Analog Trunks

Available analog HW interfaces.




Each interface can be **Edit**ed or set as **Unused**.




### General Settings

**Trunk Type:** FXS, FXO, E&M – in accordance with HW configuration
**Name:** Interface Name
**Dial Plan:** Dial Plan for this Analog Trunk
**CID Number:** Optional. Caller ID for this Trunk.

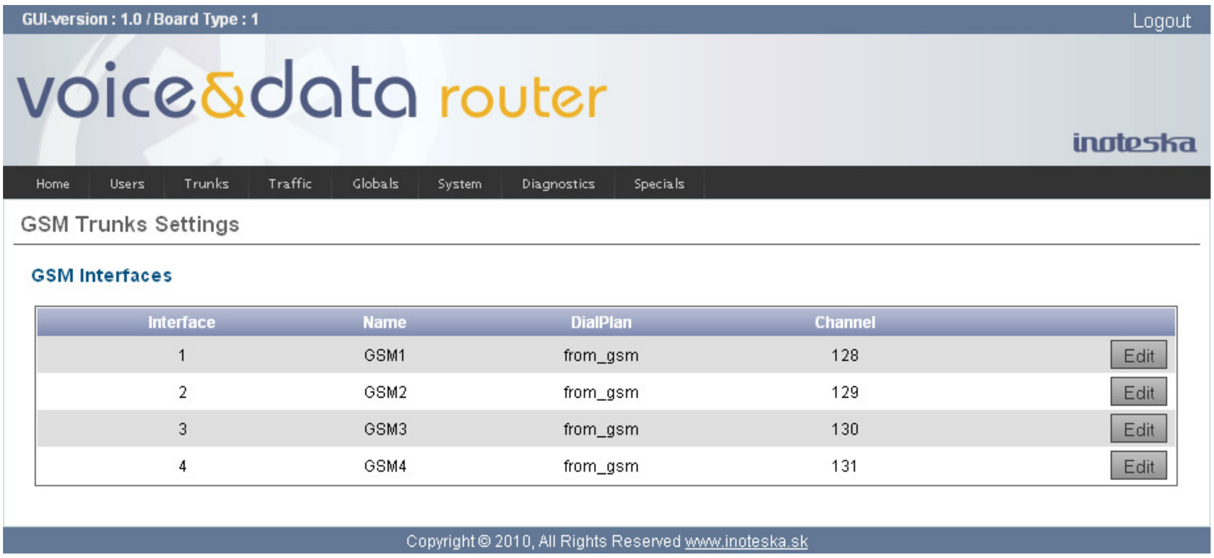
### Busy Detection

**Enable:** Check this if Busy Tone is detected.
**Count:** Enter count of Busy Tone rings to disconnect a call. Range is 3..12. Default value is 6.

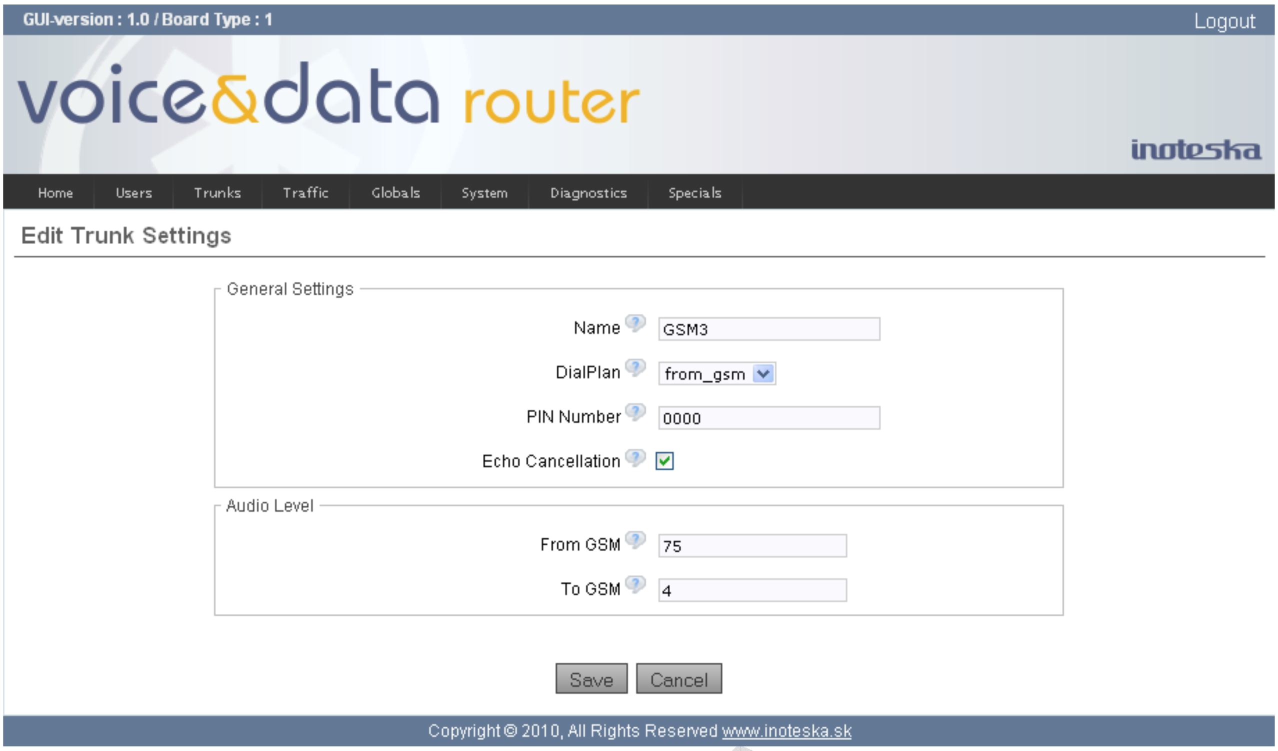


## GSM interfaces

List of available device GSM interfaces.




**Edit** parameters of each GSM interface.




### General Settings

**Name:** A unique interface name to help you identify this trunk
**Dial Plan:** Dial Plan for this GSM Trunk
**PIN Number:** Enter valid PIN code to access GSM SIM card
**Echo Cancellation:** Enable echo cancellation on GSM module

### Audio Level

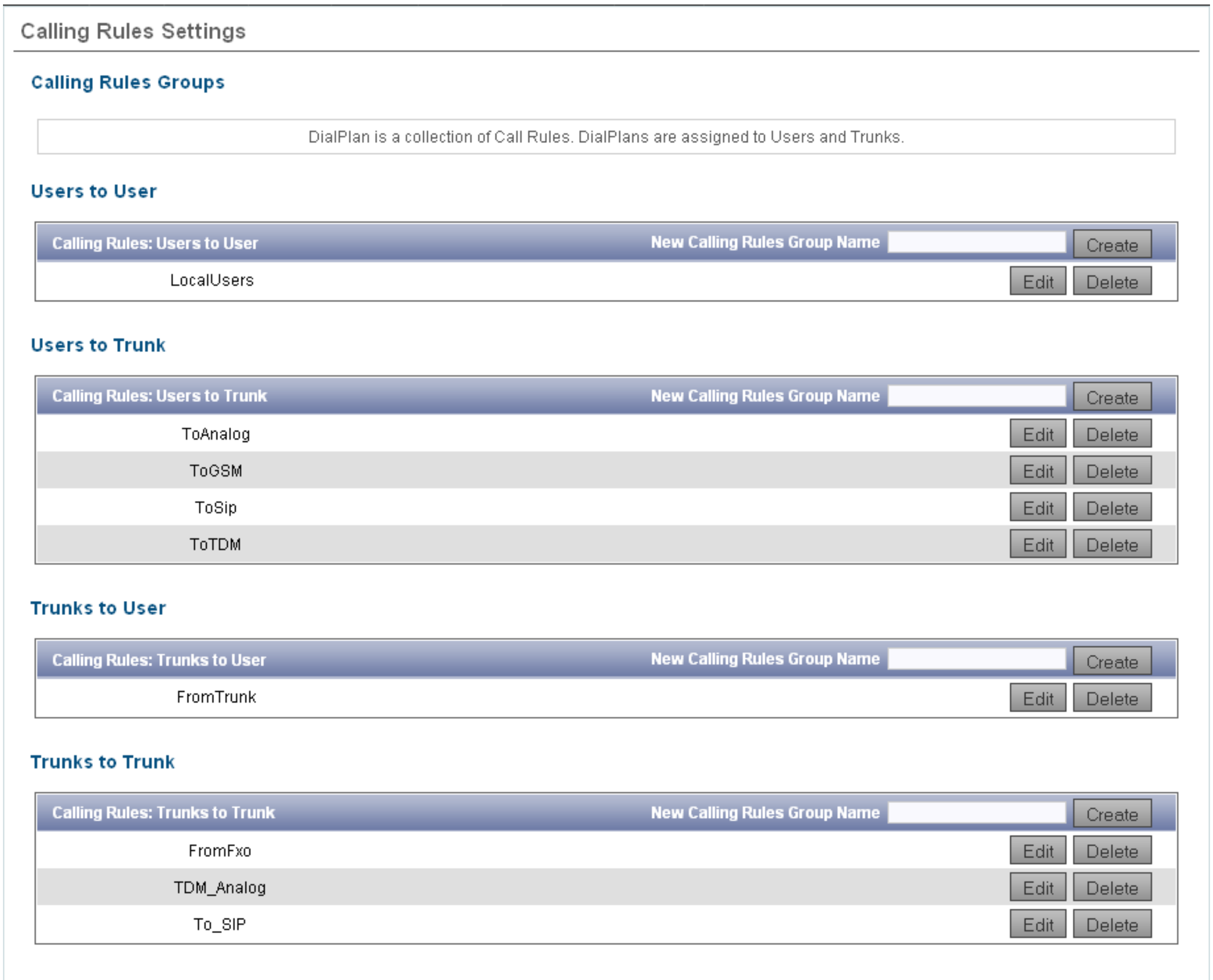
**From GSM:** Speaker level in range 0..99. Default value is 91.
**To GSM:** Microphone level in range 0..15. Default value is 3.



## 3.1.3 Traffic

### Calling Rules

Calling rules are divided into four groups according to the method of call routing (User-User, User-Trunk, Trunk-User, Trunk-Trunk). This allows different patterns to be dialled to users and through different trunks.You can optionally set three failover trunks to use when the primary trunk fails. Note that this panel manages only individual call rules. See **Dial Plans** section to associate multiple calling rules to be used for **User** and **Trunk** dialing.




## Users to User – Edit

**Pattern:** Pattern of the DID number, <i>not</I> the pattern for the CID (caller id number). All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:
X ... Any Digit from 0-9
Z ... Any Digit from 1-9
N ... Any Digit from 2-9
[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

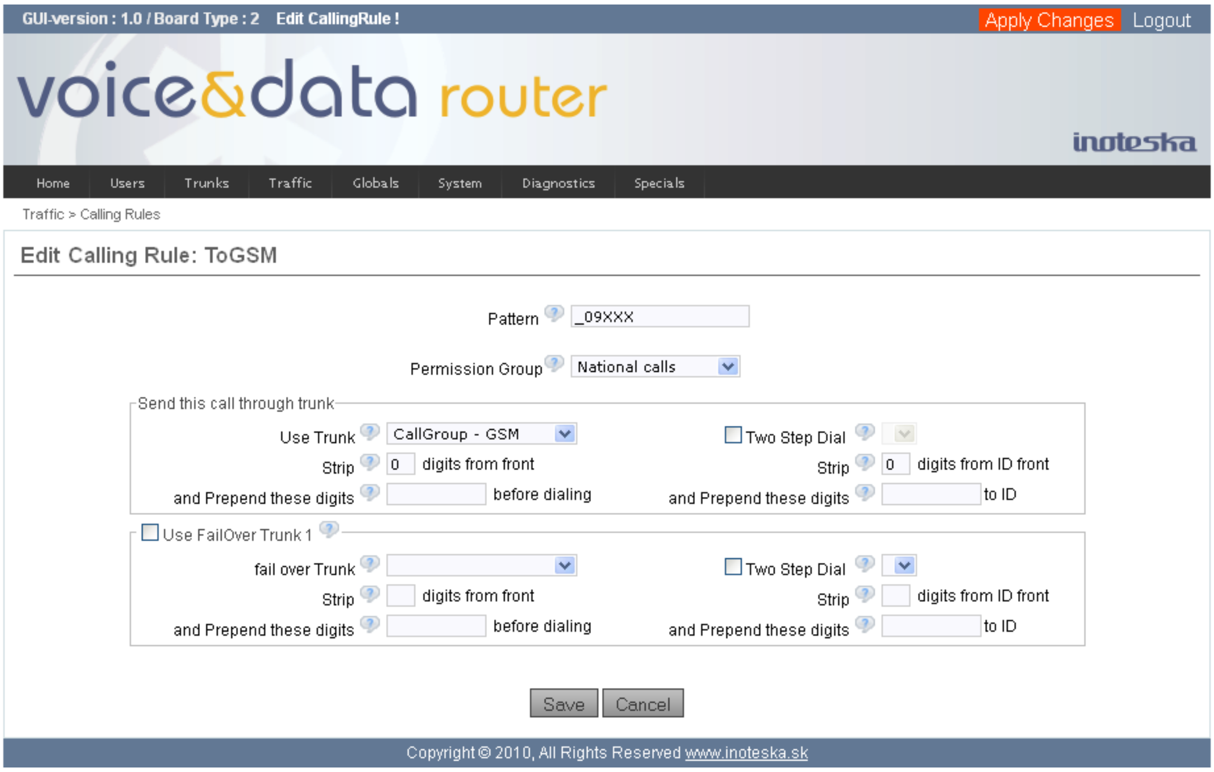
For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one. Note: You may need to add a rule with the pattern "s" (without the quotation marks) for each trunk. This signifies 'catch all', meaning all calls with a DID not matching any other rules will match this.

**Permission Group:** Select Permission Group

## Send to Local Destination

**Local Destination:** Select Local Destination from the list

## Users to Trunk – Edit




**Pattern:** Pattern of the DID number, <i>not</I> the pattern for the CID (caller id number). All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:
X ... Any Digit from 0-9
Z ... Any Digit from 1-9
N ... Any Digit from 2-9
[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
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For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one. Note: You may need to add a rule with the pattern "s" (without the quotation marks) for each trunk. This signifies 'catch all', meaning all calls with a DID not matching any other rules will match this.

**Permission Group:** Select Permission Group

### Send this call through trunk

**Use Trunk -** Defines the Trunk that calls, matching the specified pattern, will be placed through.
**Strip … digits from front -** Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in \"Use Trunk.\" One might; for example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.
**And Prepend these digits … before dialing -** Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing.

**Two Step Dial -** two-step dial
**Strip … digits from ID front -** Allows the user to specify the number of digits that will be stripped from the front of the ID string before the call is placed via the trunk selected in Use Trunk.

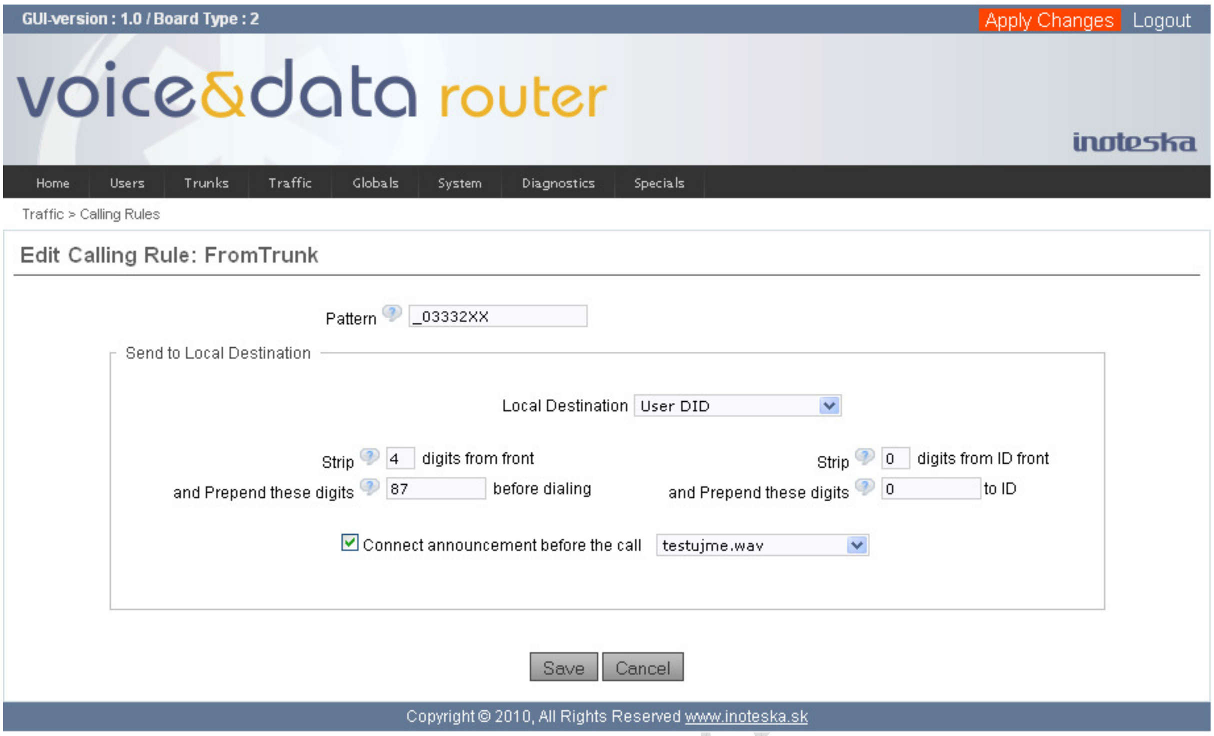
**And Prepend these digits … to ID -** Allows the user to specify digits that are prepended before the user's ID

### Use FallOver Trunk 1

**Fail over Tunk -** Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down If \"Use Failover Trunk\" is checked and \"Failover trunk\" is defined, then calls that cannot be placed via the regular trunk may have a secondary trunk defined. If a user's primary trunk is a VoIP trunk, but one wants calls to use the PSTN when the VoIP trunk isn't available, this option is a good idea.
**Strip … digits from front -** Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in \"Use Trunk.\" One might; for example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.
**And Prepend these digits … before dialing -** Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing.

**Two Step Dial –** two-step dial
**Strip … digits from ID front -** Allows the user to specify the number of digits that will be stripped from the front of the ID string before the call is placed via the trunk selected in Use Trunk.
**And Prepend these digits … to ID -** Allows the user to specify digits that are prepended before the user's ID

## Trunks to User – Edit




**Pattern:** Pattern of the DID number, <i>not</I> the pattern for the CID (caller id number). All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:
X ... Any Digit from 0-9
Z ... Any Digit from 1-9
N ... Any Digit from 2-9
[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible.

For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXX would represent a three digit area code plus phone number, proceeded by a one. Note: You may need to add a rule with the pattern "s" (without the quotation marks) for each trunk. This signifies 'catch all', meaning all calls with a DID not matching any other rules will match this.

### Send to Local Destination

**Local Destination:** Select Local Destination from the list
**Strip … digits from front -** Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in \"Use Trunk.\" One might; for example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.
**And Prepend these digits … before dialing -** Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing.
**Strip … digits from ID front -** Allows the user to specify the number of digits that will be stripped from the front of the ID string before the call is placed via the trunk selected in Use Trunk.
**And Prepend these digits … to ID -** Allows the user to specify digits that are prepended before the user's ID
**Connect announcement before the call** – select audio file from the list

## Trunks to Trunk – Edit

**Pattern:** Pattern of the DID number, <i>not</I> the pattern for the CID (caller id number). All patterns are prefixed by the "_" character. In patterns, some characters have special meanings:
X ... Any Digit from 0-9
Z ... Any Digit from 1-9
N ... Any Digit from 2-9
[12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9)
... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
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### Send this call through trunk

**Use Trunk -** Defines the Trunk that calls, matching the specified pattern, will be placed through.
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**And Prepend these digits … before dialing -** Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing.

**Two Step Dial -** two-step dial
**Strip … digits from ID front -** Allows the user to specify the number of digits that will be stripped from the front of the ID string before the call is placed via the trunk selected in Use Trunk.
**And Prepend these digits … to ID -** Allows the user to specify digits that are prepended before the user's ID

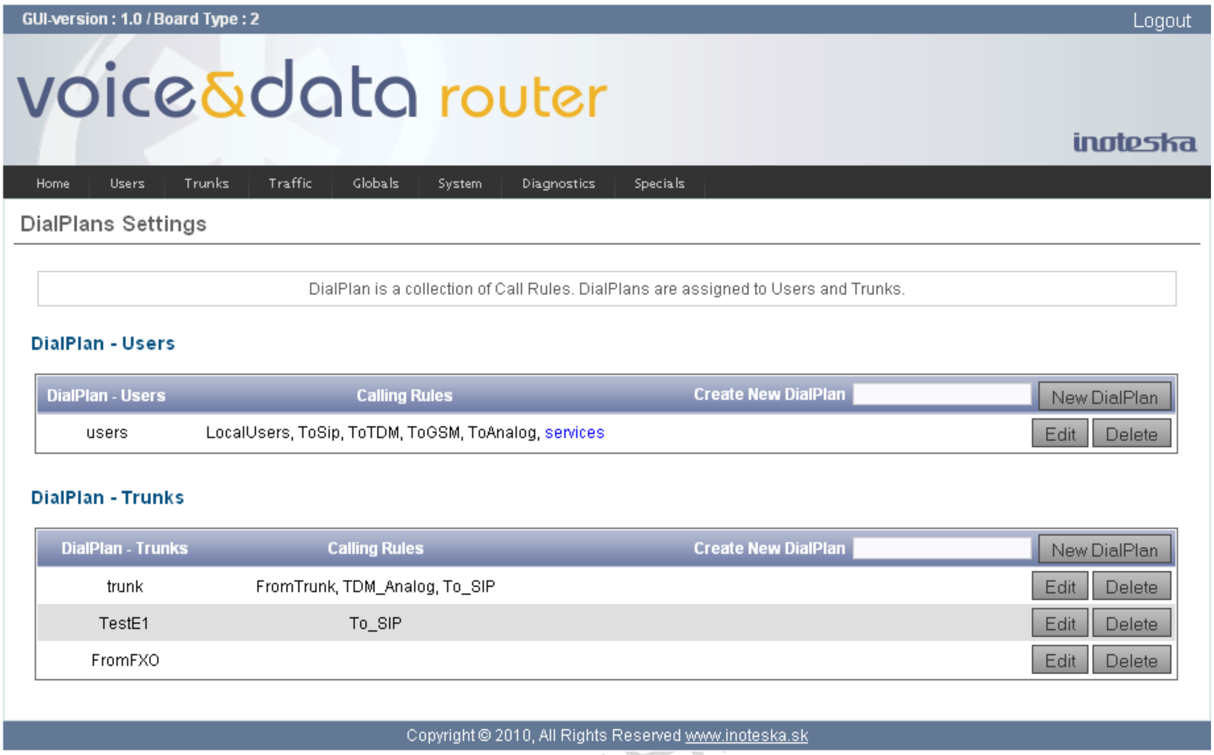
### Use FallOver Trunk 1

**Fail over Tunk -** Failover trunks can be used to make sure that a call goes through an alternate route, when the primary trunk is busy or down If \"Use Failover Trunk\" is checked and \"Failover trunk\" is defined, then calls that cannot be placed via the regular trunk may have a secondary trunk defined. If a user's primary trunk is a VoIP trunk, but one wants calls to use the PSTN when the VoIP trunk isn't available, this option is a good idea.
**Strip … digits from front -** Allows the user to specify the number of digits that will be stripped from the front of the dialing string before the call is placed via the trunk selected in \"Use Trunk.\" One might; for example, want users to dial 9 before their long distance calls; however one does not dial 9 before those calls are placed onto analog lines and the PSTN, so one should strip 1 digit from the front before the call is placed.
**And Prepend these digits … before dialing -** Allows the user to specify digits that are prepended before the call is placed via the trunk. If a user's trunk required 10 digit dialing, but users were more comfortable performing 7 digit dialing, this field could be used to prepend a 3 digit area code to all 7 digit strings before they are placed to the trunk. User may also prepend a 'w' character for analog trunks to provide a slight delay before dialing.

**Two Step Dial –** two-step dial
**Strip … digits from ID front -** Allows the user to specify the number of digits that will be stripped from the front of the ID string before the call is placed via the trunk selected in Use Trunk.
**And Prepend these digits … to ID -** Allows the user to specify digits that are prepended before the user's ID

## Dial Plans

**Dial Plan** is a collection of **Call Rules**. **Dial Plans** are assigned to **Users** and **Trunks**.




## DialPlan – Users - Edit

Add/Delete Calling Rules and use Downward/Upward Arrows to manage Calling Rules – move them up or down.

**User´s Extensions**

**Use Services Codes –** Check this if user can use Services Codes

**Ignore Pattern**

**Ignorepat -** Ignorepat can be used to instruct drivers to not cancel dialtone upon receipt of a particular pattern.

**DialPlan – Trunks - Edit**

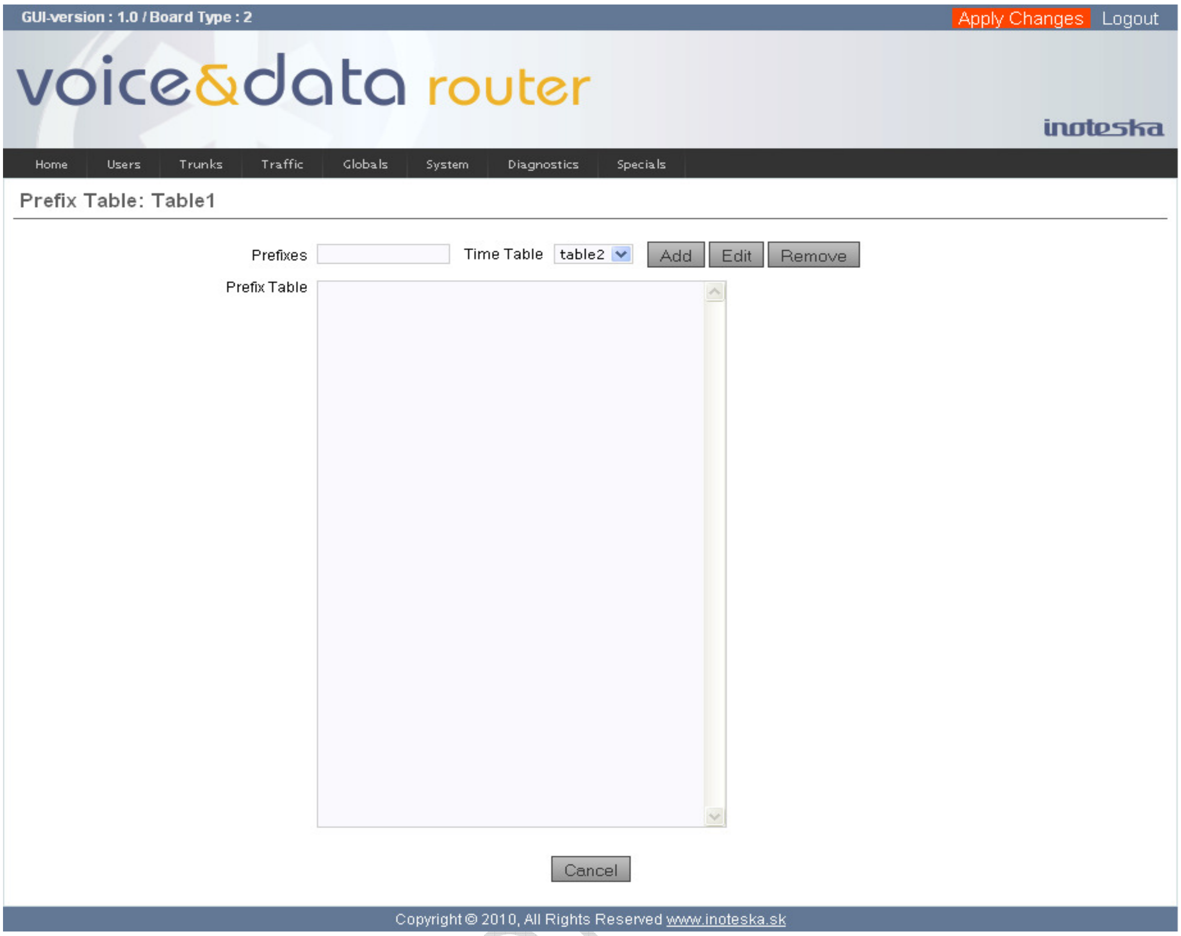

Add/Delete Calling Rules and use Downward/Upward Arrows to manage Calling Rules – move them up or down.

## AOC Tables Settings

### AOC Tabels

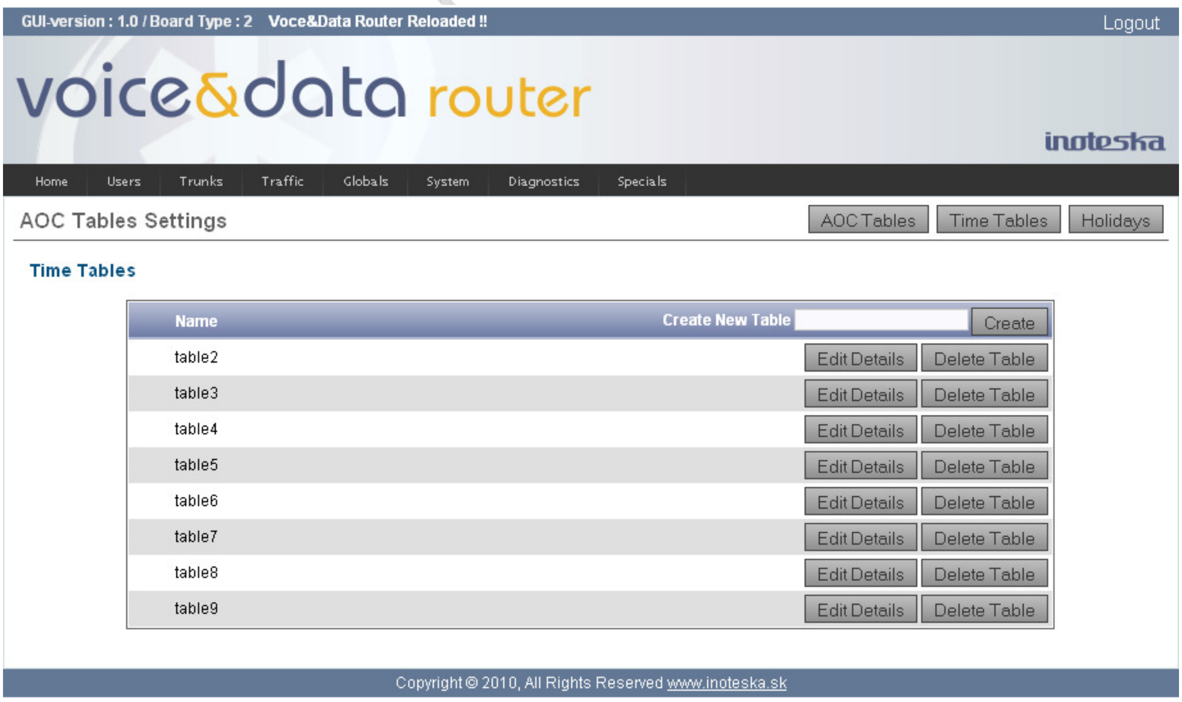
Create/Delete/Edit AOC Tables.

## AOC Tables – Edit

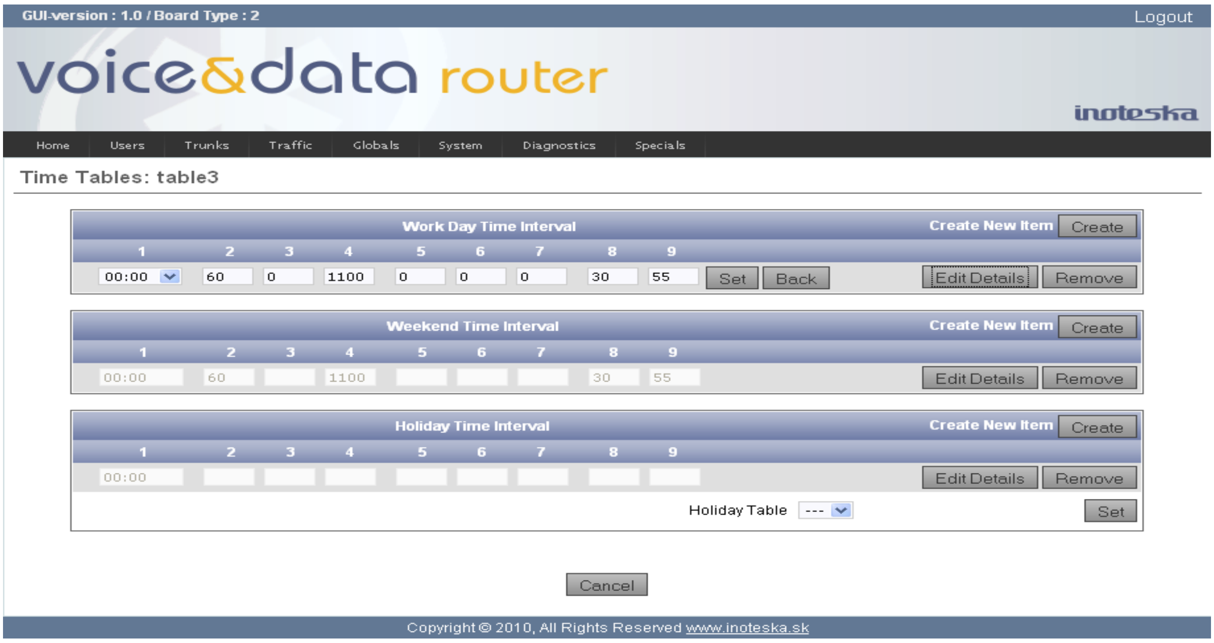



## Time Tables

Create/Delete/Edit Time Tables.

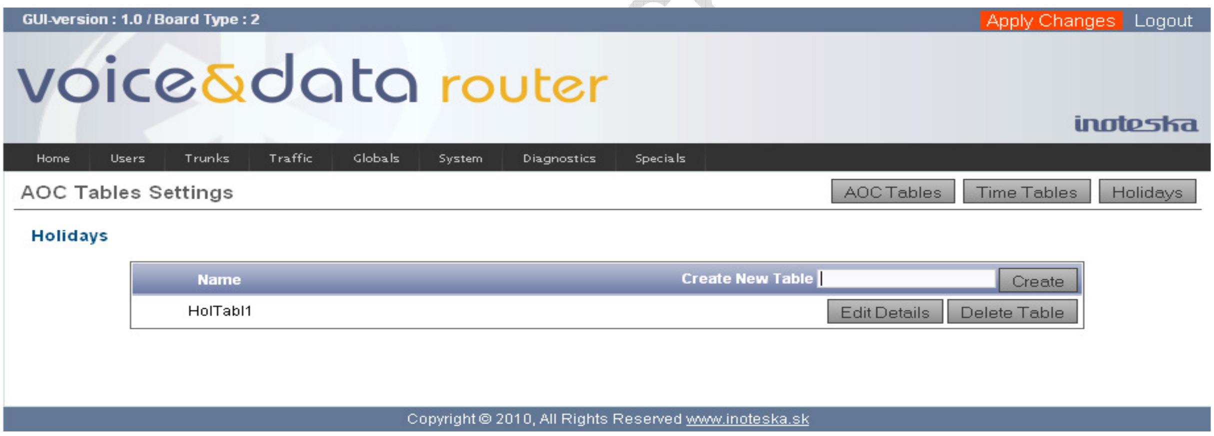


## Time Tables – Edit




## Holidays

Create/Delete/Edit HolidaysTables.

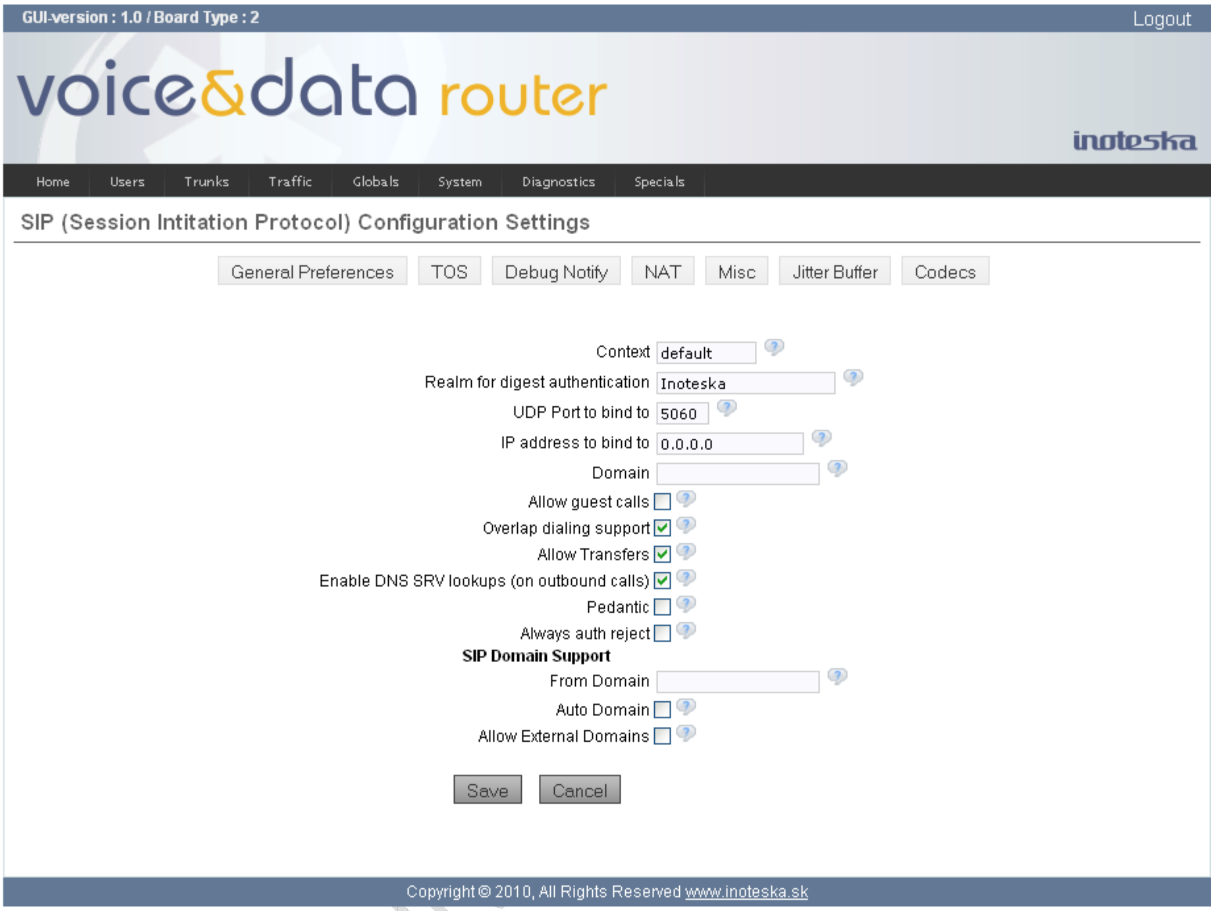



## Holidays – Edit

### 3.1.4 Globals

**SIP settings**

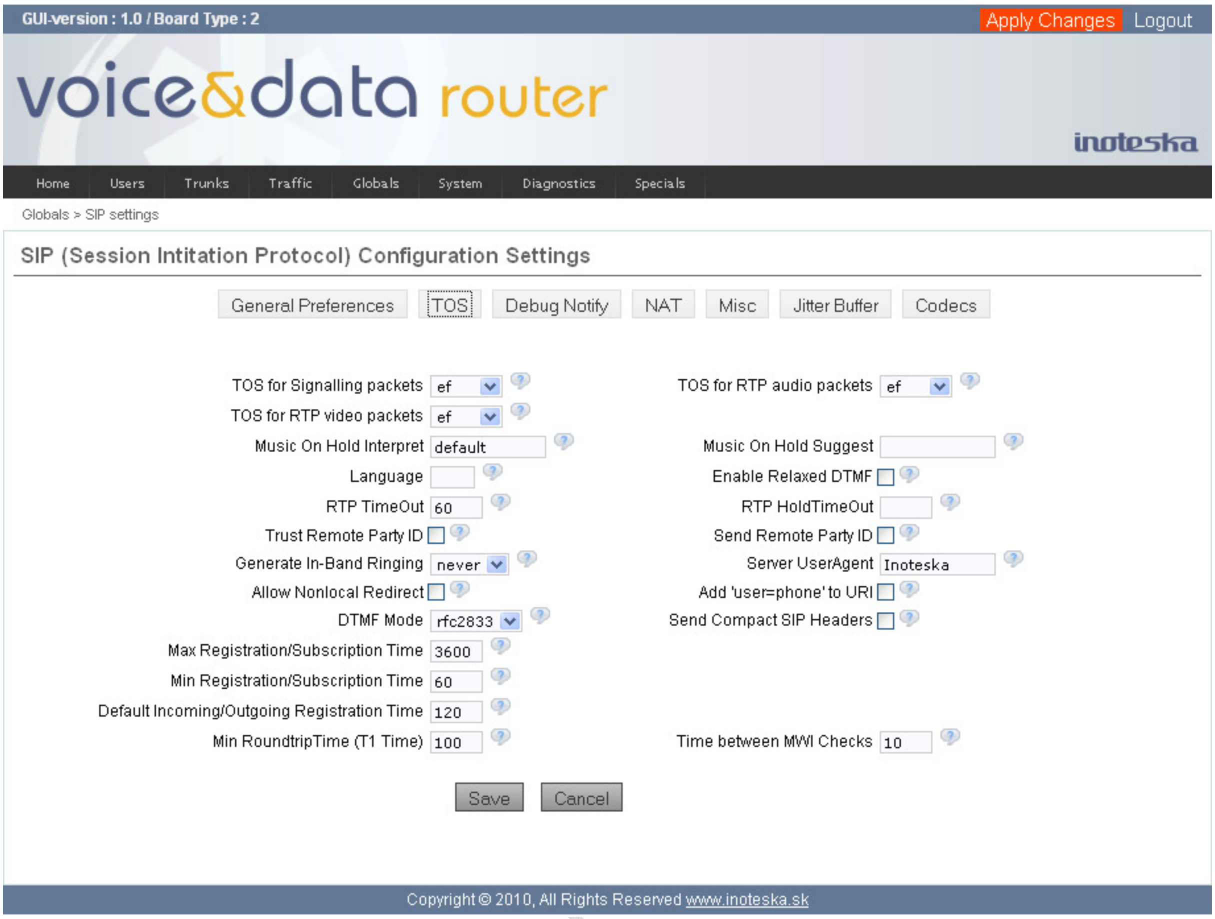
**General Preferences**




**Context –** Deafault Context for incoming calls
**Realm for digest authentication -** Realm for digest authentication.defaults to \'asterisk\'. If you set a system name in asterisk.conf, it defaults to that system name. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name
**UDP Port to bind to -** SIP standard port is 5060
**IP address to bind to -** 0.0.0.0 binds to all
**Domain -** Comma separated list of domains which Asterisk is responsible for
**Allow guest calls –** Enable guest calls
**Overlap dialing support –** Enable dialing support
**Allow Transfers –** Enable transfers
**Enable DNS SRV lookups (on outbound calls) –** Enable DNS SRV lookups on calls
**Pedantic -** Enable slow, pedantic checking of Call-ID:s, multiline SIP headers and URI-encoded headers.
**Always auth reject -** If this option is enabled, whenever Asterisk rejects an INVITE or REGISTER, it will always reject it with a *401 Unauthorized* message instead of letting the caller know whether there was a matching user or peer for their request.

**From Domain -** When making outbound SIP INVITEs to non-peers, use your primary domain \'identity\' for From: headers instead of just your IP address. This is to be polite and it may be a mandatory requirement for some destinations which do not have a prior account relationship with your server.
**Auto Domain -** Turn this on to have Asterisk add local host name and local IP to domain list.
**Allow External Domains -** Allow requests for domains not serviced by this server.

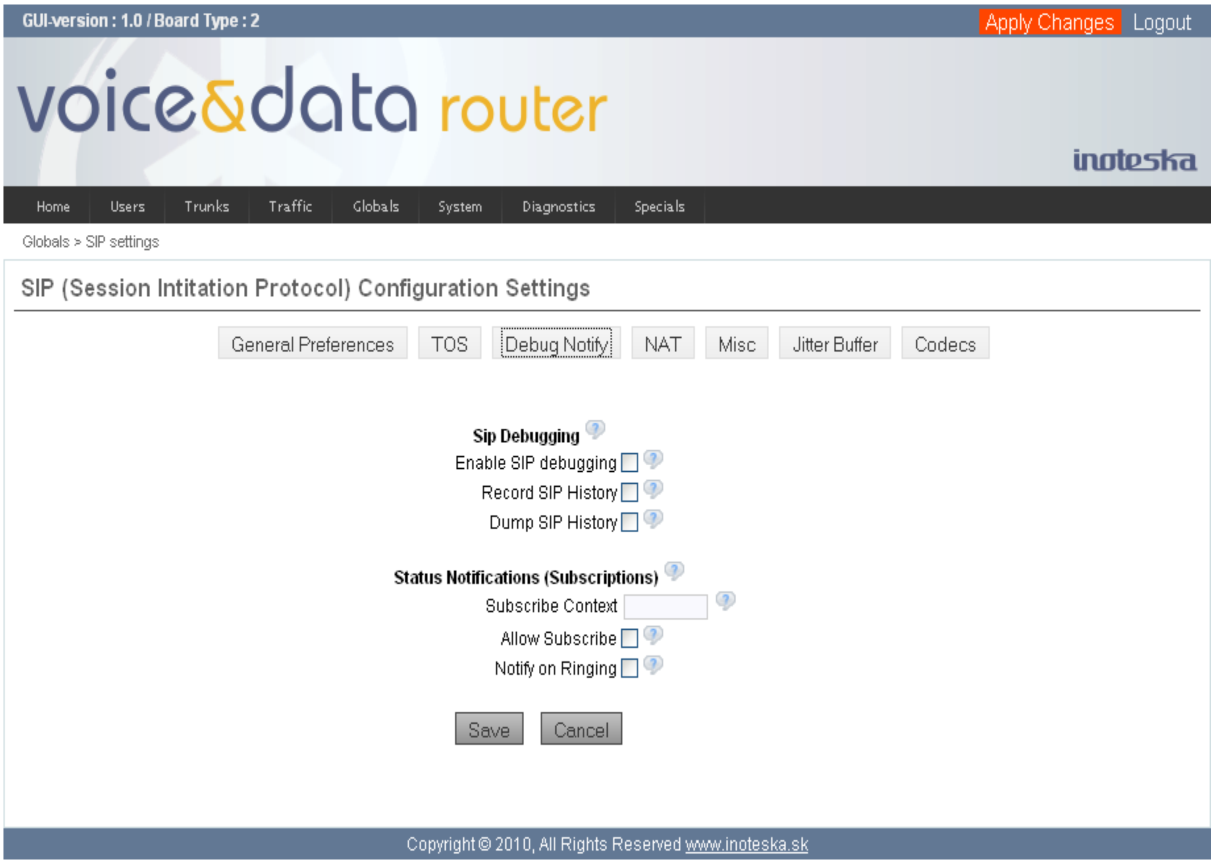
## TOS




**TOS for Signalling packets -** Sets Type of Service for SIP packets.
**TOS for RTP audio packets -** Sets Type of Service for RTP audio packets.
**TOS for RTP video packets -** Sets Type of Service for RTP video packets.
**Music On Hold Interpret -** This option specifies a preference for which music on hold class this channel should listen to when put on hold if the music class has not been set on the channel with Set(CHANNEL(musicclass)=whatever) in the dialplan, and the peer channel putting this one on hold did not suggest a music class.
**Music On Hold Suggest -** This option specifies which music on hold class to suggest to the peer channel when this channel places the peer on hold. It may be specified globally or on a per-user or per-peer basis.
**Language -** Default language setting for all users/peers.
**Enable Relaxed DTMF -** Relax dtmf handling.
**RTP TimeOut -** Terminate call if 60 seconds of no RTP activity when we're not on hold.
**RTP HoldTimeOut -** Terminate call if 300 seconds of no RTP activity when we're on hold (must be > rtptimeout).
**Trust Remote Party ID -** If Remote-Party-ID should be trusted.
**Send Remote Party ID -** If Remote-Party-ID should be sent.
**Generate In-Band Ringing -** If we should generate in-band ringing always use \'never\' to never use in-band signalling, even in cases where some buggy devices might not render it. Default: never.
**Server UserAgent -** Allows you to change the user agent string.
**Allow Nonlocal Redirect -** If checked, allows 302 or REDIR to non-local SIP address Note that promiscredir when redirects are made to the local system will cause loops since Asterisk is incapable of performing a \'hairpin\' call.
**Add ´user=phone´ to Url -** If checked, \'user=phone\' is added to uri that contains a valid phone number.
**DTMF Mode -** Set default dtmfmode for sending DTMF. Default: rfc2833H.
**Send Comapct SIP Headers -** send compact sip headers.

**Max Registration/Subscription Time -** Maximum duration (in seconds) of incoming registration/subscriptions we allow. Default 3600 seconds.
**Min Registration/Subscription Time -** Minimum duration (in seconds) of registrations/subscriptions. Default 60 seconds.
**Default Incoming/Outgoing registration Time -** Default duration (in seconds) of incoming/outoing registration.
**Min Roundtrip Time (T1 Time) -** Minimum roundtrip time for messages to monitored hosts, Defaults to 100 ms.
**Time between MWI Checks -** Default Time between Mailbox checks for peers.

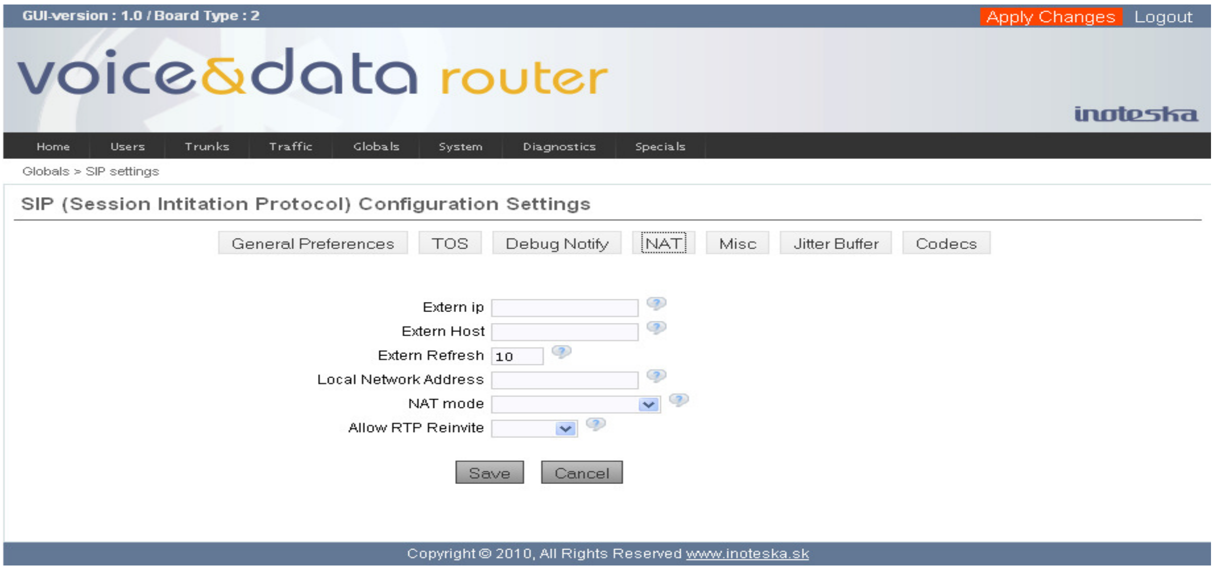
## Debug Notify




**Sip Debugging**
**Enable SIP debugging -** Turn on SIP debugging by default.
**Record SIP History -** Record SIP history by default.
**Dump SIP History -** Dump SIP history at end of SIP dialogue.

**Status Notifications (Subscriptions)**
**Subscribe Context -** Set a specific context for SUBSCRIBE requests. Useful to limit subscriptions to local extensions.
**Allow Subscribe -** Support for subscriptions.
**Notify on Ringing -** Notify subscriptions on RINGING state.

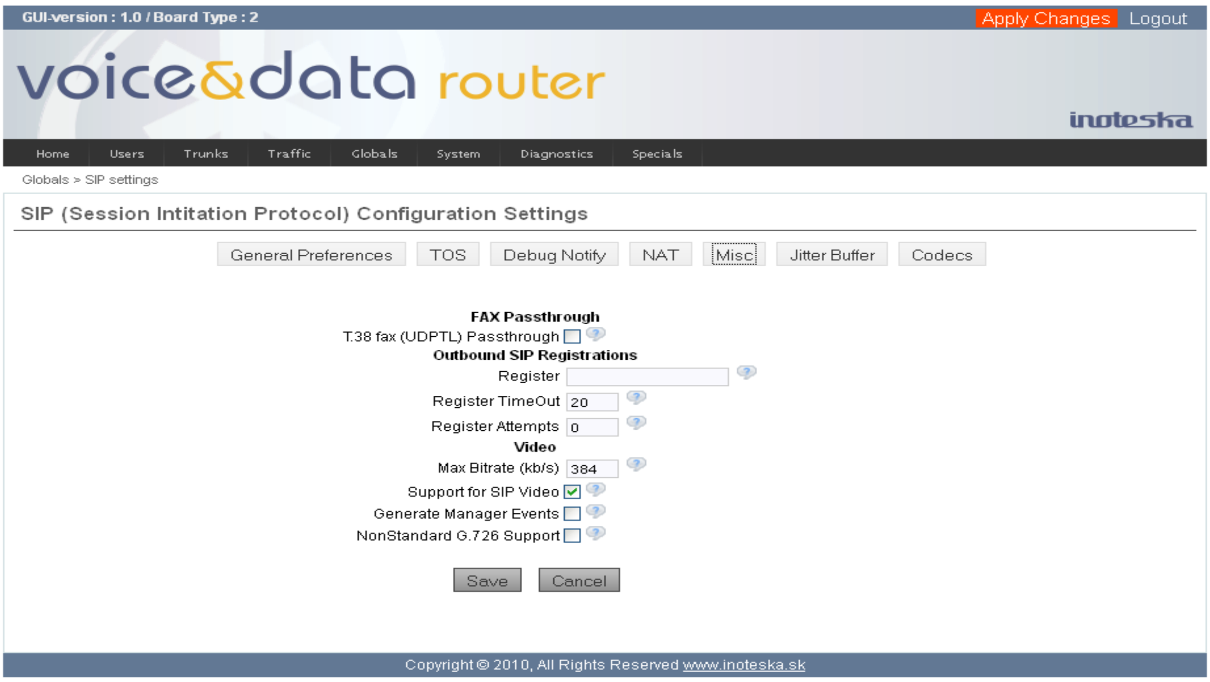
## NAT




**Extern ip -** Address that we're going to put in outbound SIP messages if we're behind a NAT.
**Extern Host -** Alternatively you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use externip instead.
**Extern Refresh -** How often to refresh externhost if used. You may specify a local network in the field below.
**Local Network Address -** \'192.168.0.0/255.255.0.0\' : All RFC 1918 addresses are local networks, \'10.0.0.0/255.0.0.0\' : Also RFC1918, \'172.16.0.0/12\' : Another RFC1918 with CIDR notation, \'169.254.0.0/255.255.0.0\' : Zero conf local network.
**NAT mode -** Global NAT settings (Affects all peers and users); yes = Always ignore info and assume NAT; no = Use NAT mode only according to RFC3581; never = Never attempt NAT mode or RFC3581 support; route = Assume NAT, don't send rport.
**Allow RTP Reinvite -** Asterisk by default tries to redirect the RTP media stream (audio) to go directly from the caller to the callee. Some devices do not support this (especially if one of them is behind a NAT).

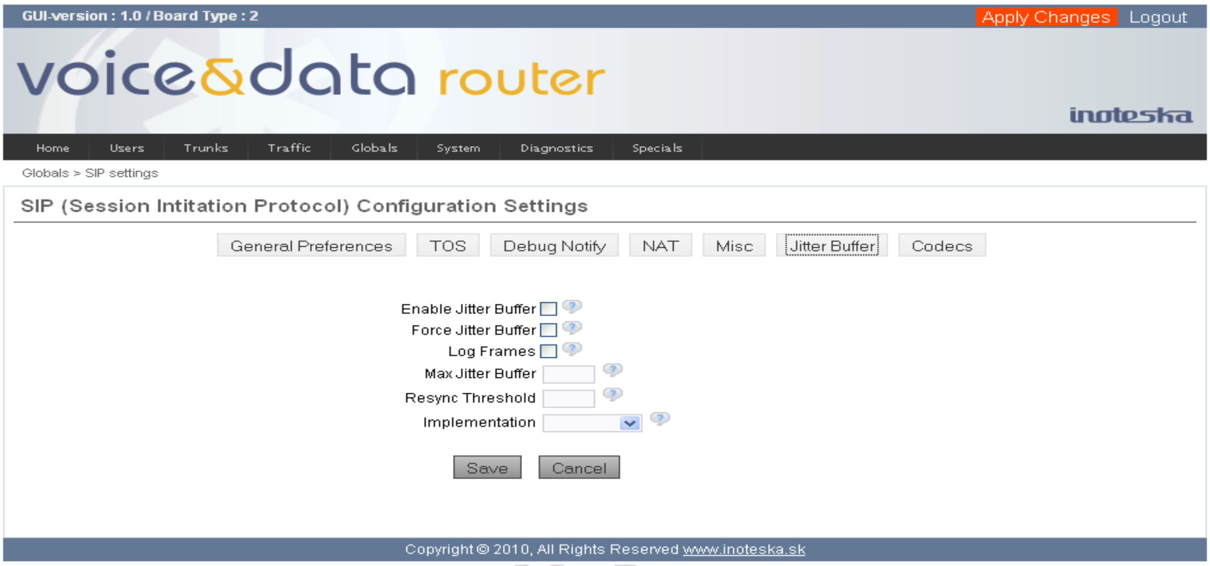


## Misc



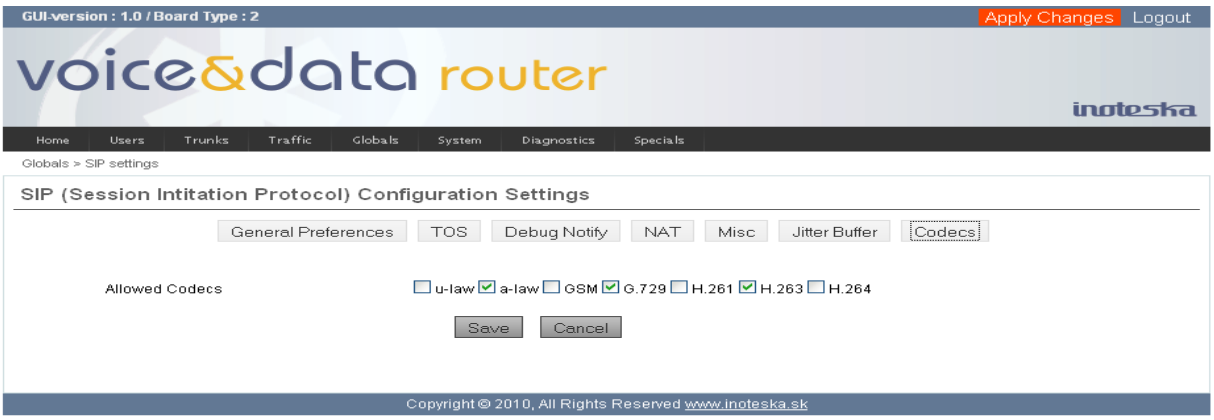
**T.38 fax (UDPTL) Passthrough -** Enables T.38 fax (UDPTL) passthrough on SIP to SIP calls.
**Register -** Register as a SIP user agent to a SIP proxy (provider).
**Register TimeOut -** Retry registration calls at every \'x\' seconds (default 20).
**Register Attempts -** Number of registration attempts before we give up; 0 = continue foreverp.
**Max Bitrate (kb/s) -** Maximum bitrate for video calls (default 384 kb/s).
**Support for SIP Video -** Turn on support for SIP video.
**Generate Manager Events -** Generate manager events when sip ua performs events (e.g. hold).
**NonStandard G.726 Support -** If the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs, among others). This is contrary to the RFC3551 specification, the peer _should_ be negotiating AAL2-G726-32 instead.

## Jitter Buffer




**Enable Jitter Buffer -** Enables the use of a jitterbuffer on the receiving side of a SIP channel.
**Force Jitter Buffer -** Forces the use of a jitterbuffer on the receive side of a SIP channel.
**Log Frames -** Enables jitterbuffer frame logging.
**Max Jitter Buffer -** Max length of the jitterbuffer in milliseconds.
**Resync Threshold -** Jump in the frame timestamps over which the jitterbuffer is resynchronized. Useful to improve the quality of the voice, with big jumps in/broken timestamps, usualy sent from exotic devices and programs. Defaults to 1000.
**Implementation -** Jitterbuffer implementation, used on the receiving side of a SIP channel. Two implementations are currenlty available - \'fixed\' (with size always equals to jbmaxsize) and \'adaptive\' (with variable size, actually the new jb of IAX2).

## Codecs




Select allowed codecs from the list.

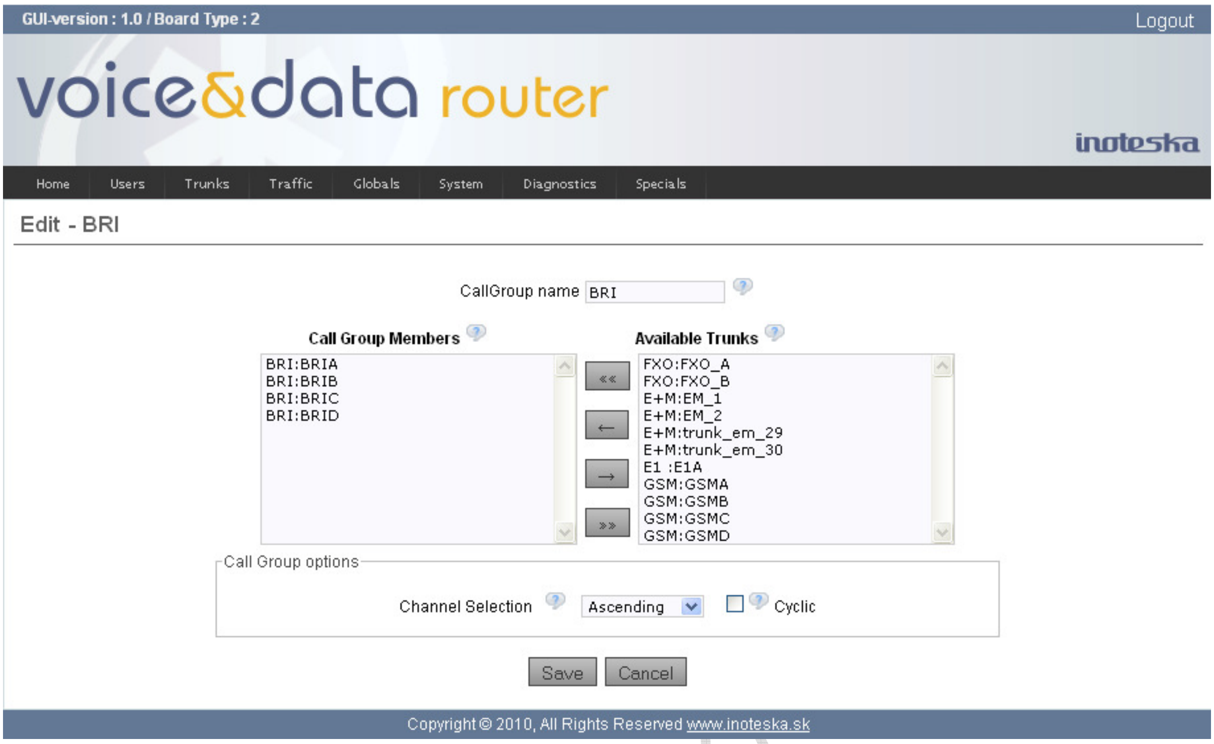
## RTP/UDPTL Settings


**Start port -** The first port of port range allocated for RTP/UDPTL session.
**End port -** The last port of port range allocated for RTP/UDPTL session.

## Call Groups

Create New CallGroup or Edit/Delete the Call Group from the list.

## Call Groups – Edit



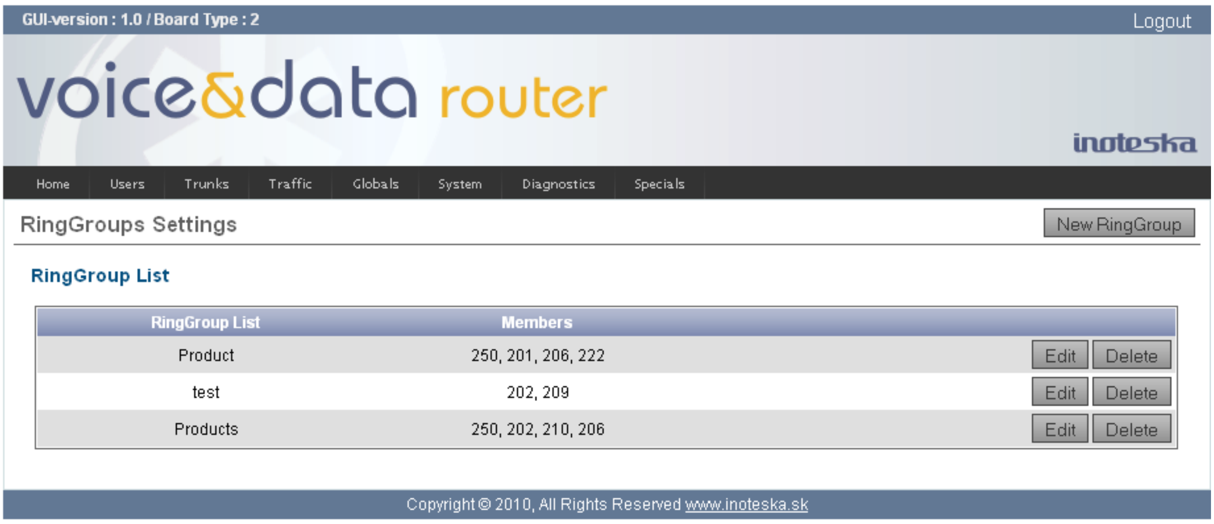

**CallGroup name -** User defined Name for CallGroup. Will be displayed in the list of trunks.
**Call Group Members -** List of trunks in Call Group.
**Available Trunks -** List of all available trunks.

### Call Group options

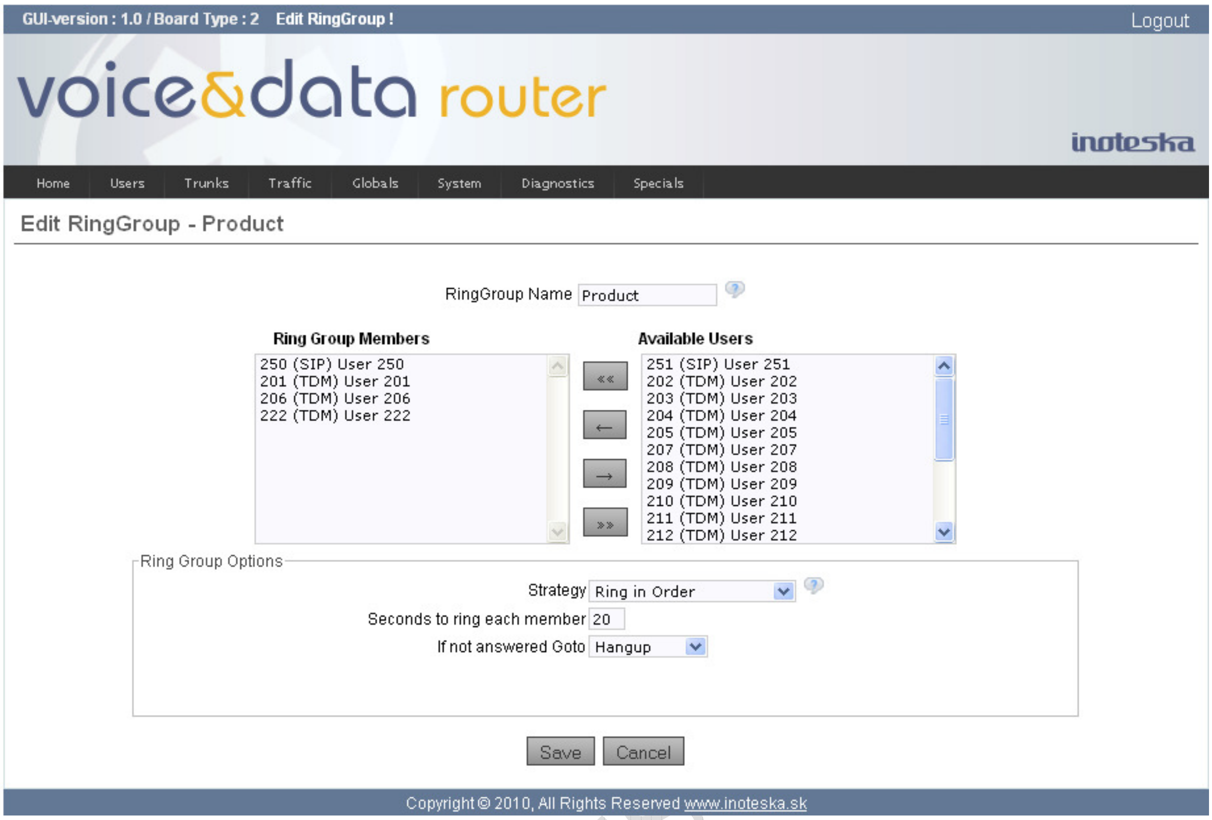
**Channel Selection -** Method of the selection of outgoing voice channel.
Ascending ... search for the lowest available channel number.
Descending ... search for the highest available channel number
**Cyclic -** Enable cyclic search of available outgoing channel instead of sequential.

## Ring Groups

Create New RingGroup or Edit/Delete any RingGroup from the list.



## Ring Groups - Edit




**RingGroup Name -** User defined Name for RingGroup. Will be displayed in the list of users.

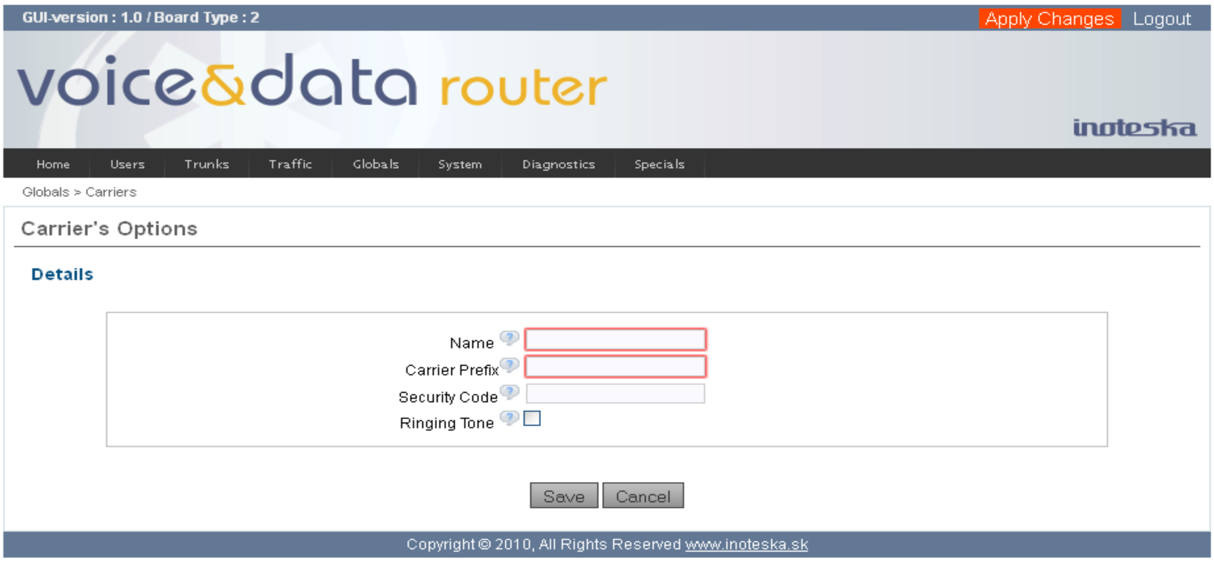
### Ring Group Options

**Strategy –** The strategy used.

### Ring Groups

**Carriers -** list of operators/carriers.

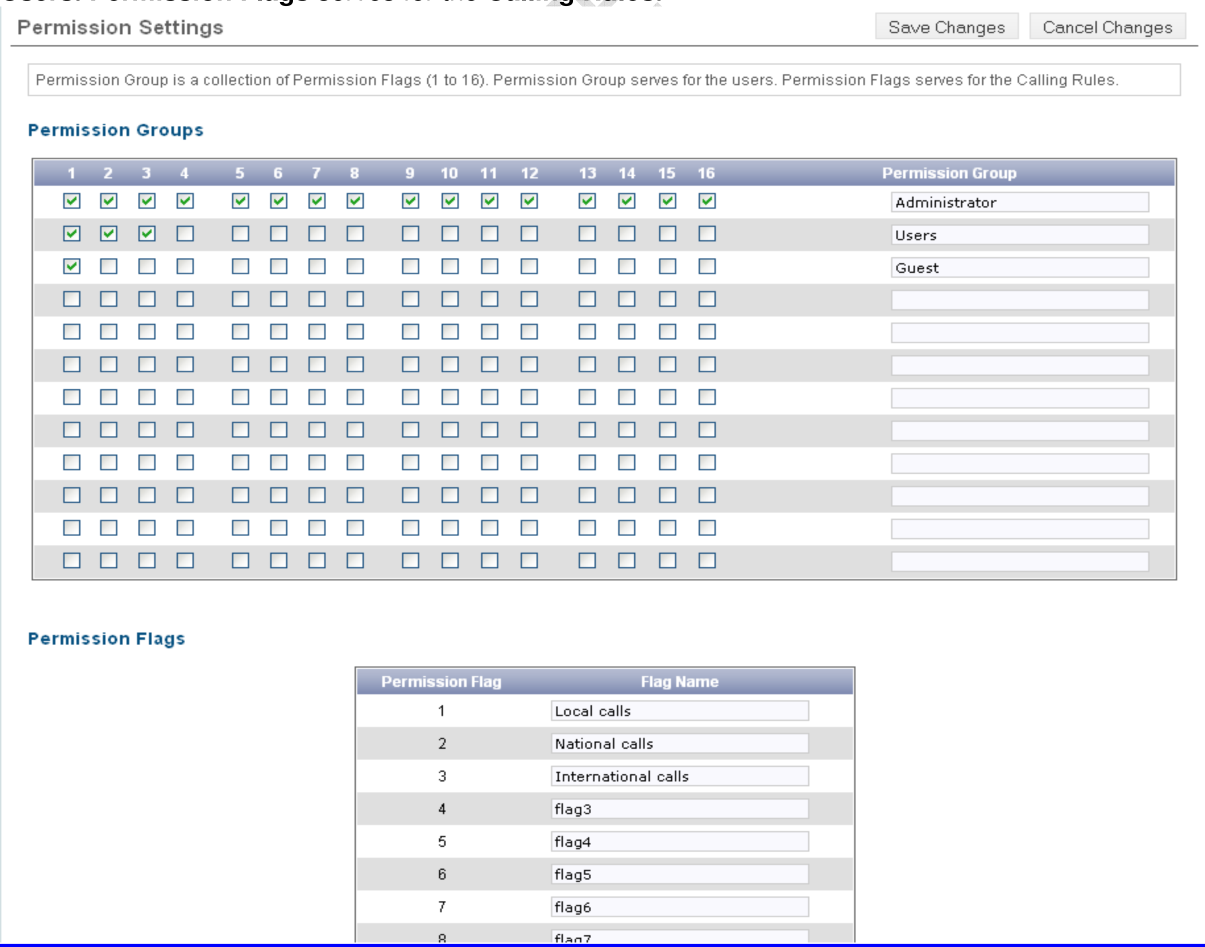
For each operator you can define:



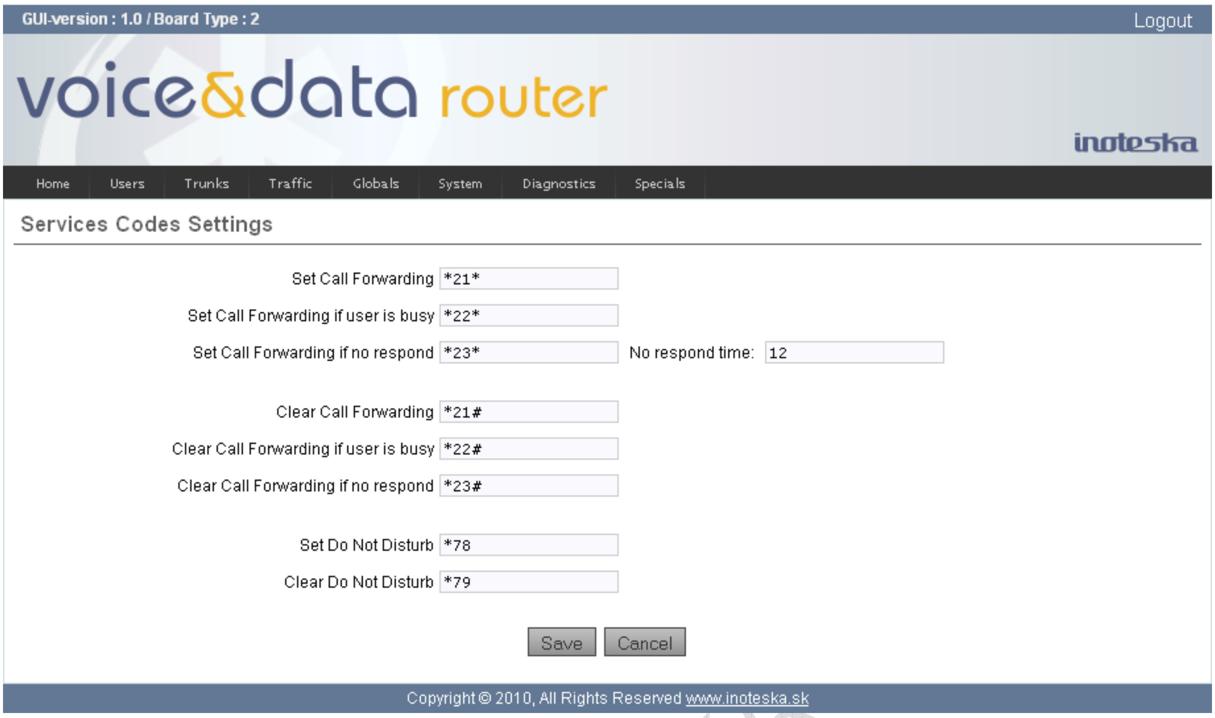

**Name –** Define Name of Operator.
**Carrier Prefix –** Define access code for Carrier.
**Security Code –** Define PIN for using Carrier´s services.
**Ringing Tone –** Check this if Ringing tone is connected.

## Permissions

Permission Group is a collection of Permission Flags (1 to 16). **Permission Group** serves for the **Users**. **Permission Flags** serves for the **Calling Rules.**



## Services Codes




## CDR Settings

### CDR Details


**CDR Note –** User defined string.
**CDR User to User –** CDR records between two users.
**CDR Trunk to User –** CDR records from trunks to users.
**CDR User to Trunk –** CDR records from users to trunks.

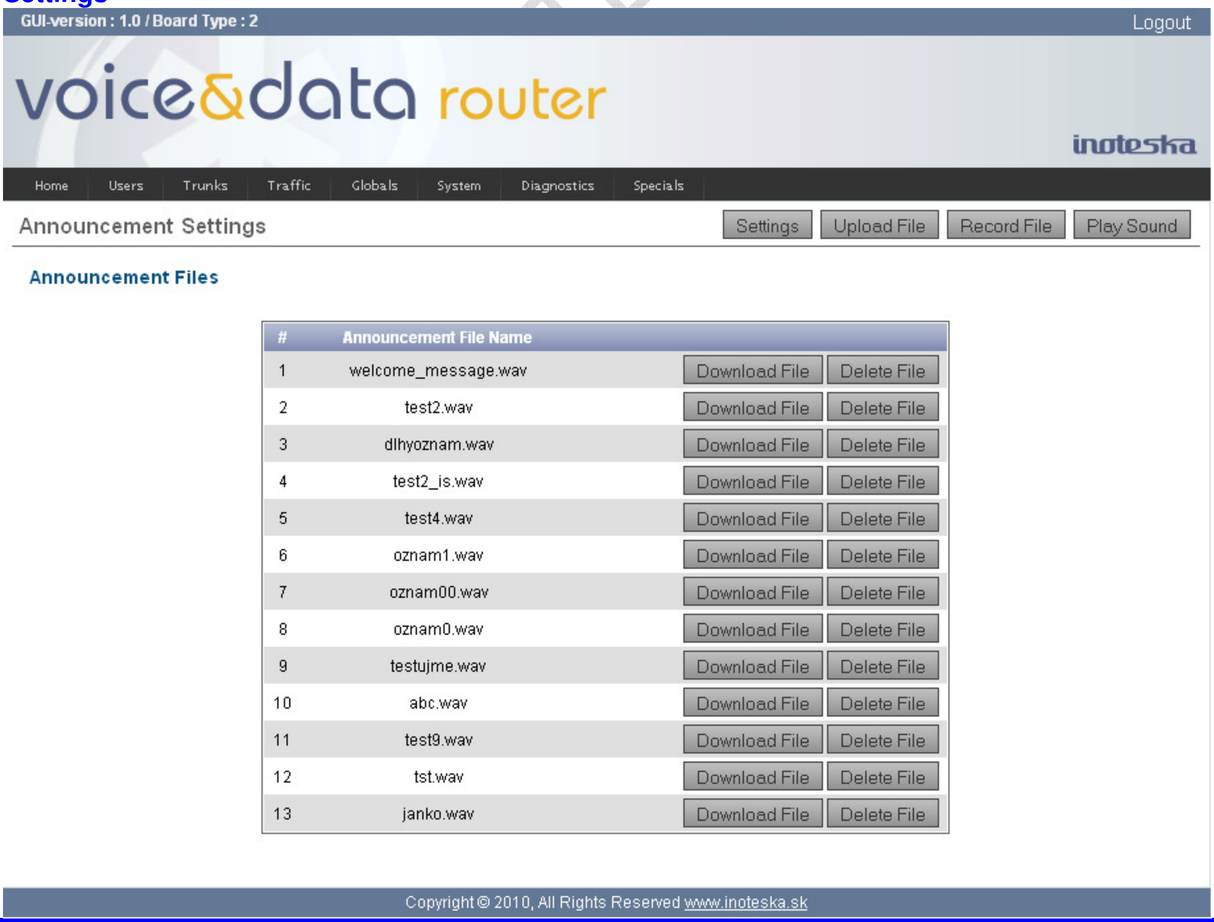
## CDR Records

CDR records in accordance with CDR details set.

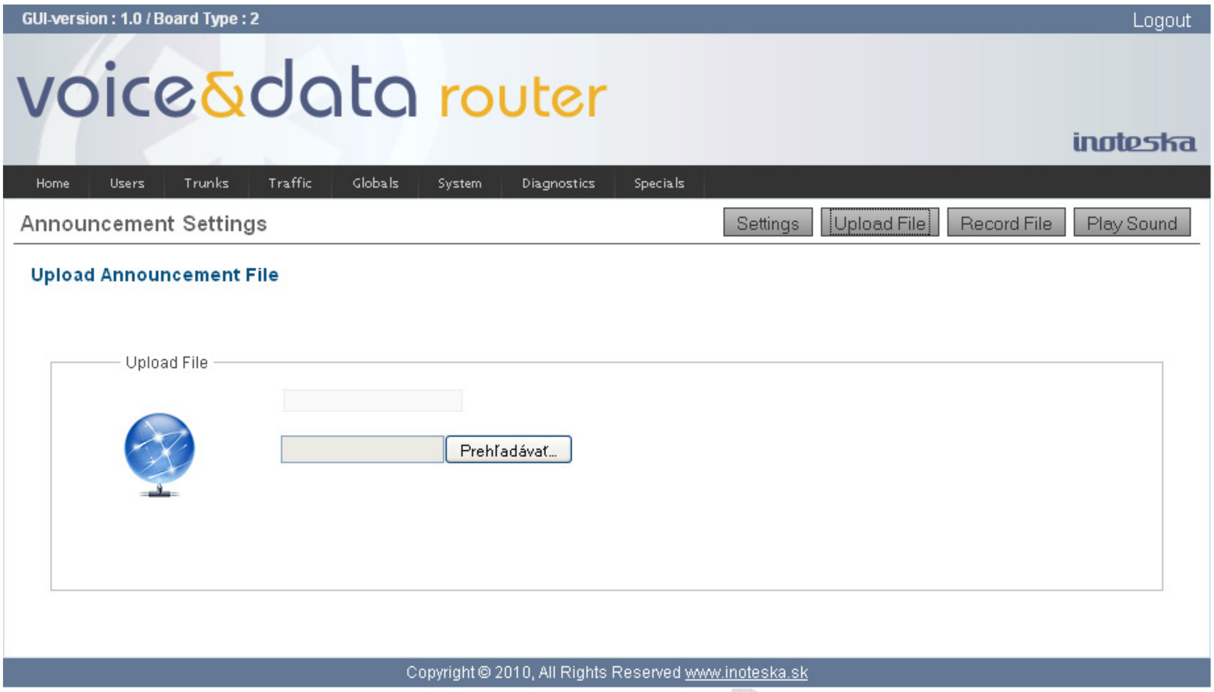

## Announcement Settings

Management of announcement files.

## Settings

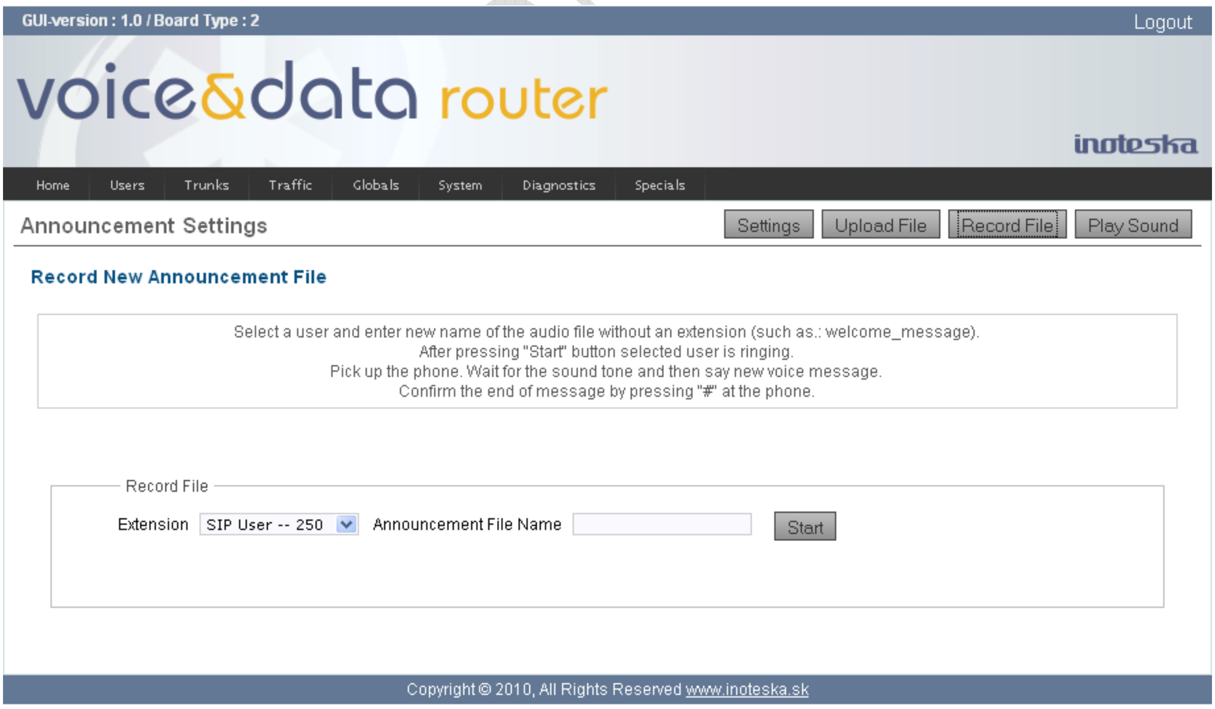


## Upload File

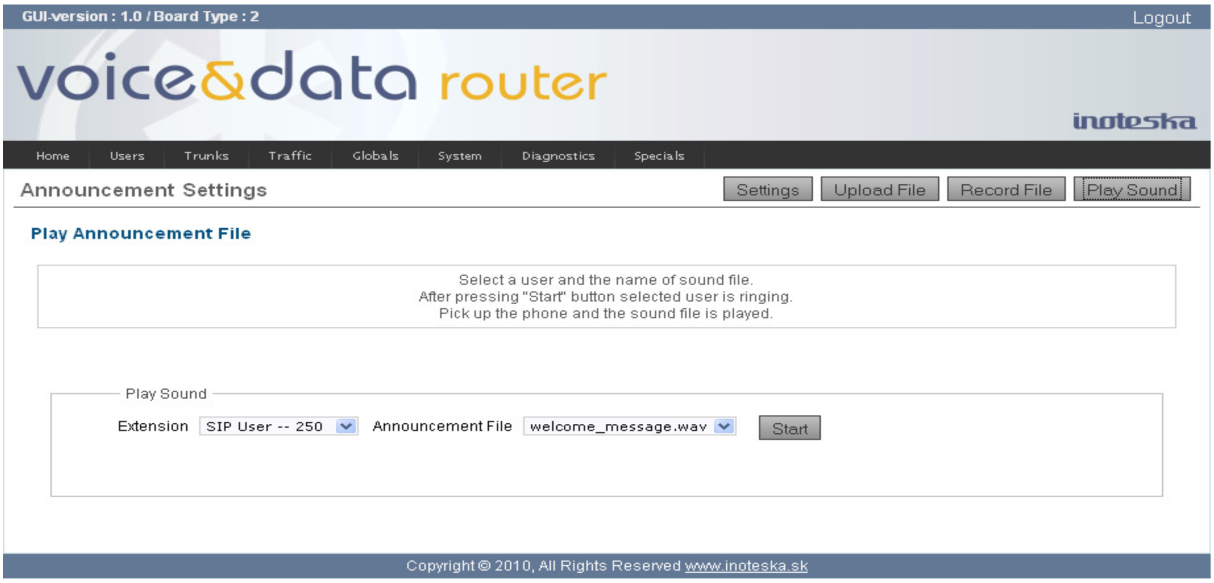



## Record file

Select a user and enter new name of the audio file without an extension (such as.: welcome_message). After pressing "Start" button selected user is ringing. Pick up the phone. Wait for the sound tone and then say new voice message. Confirm the end of message by pressing "#" at the phone.



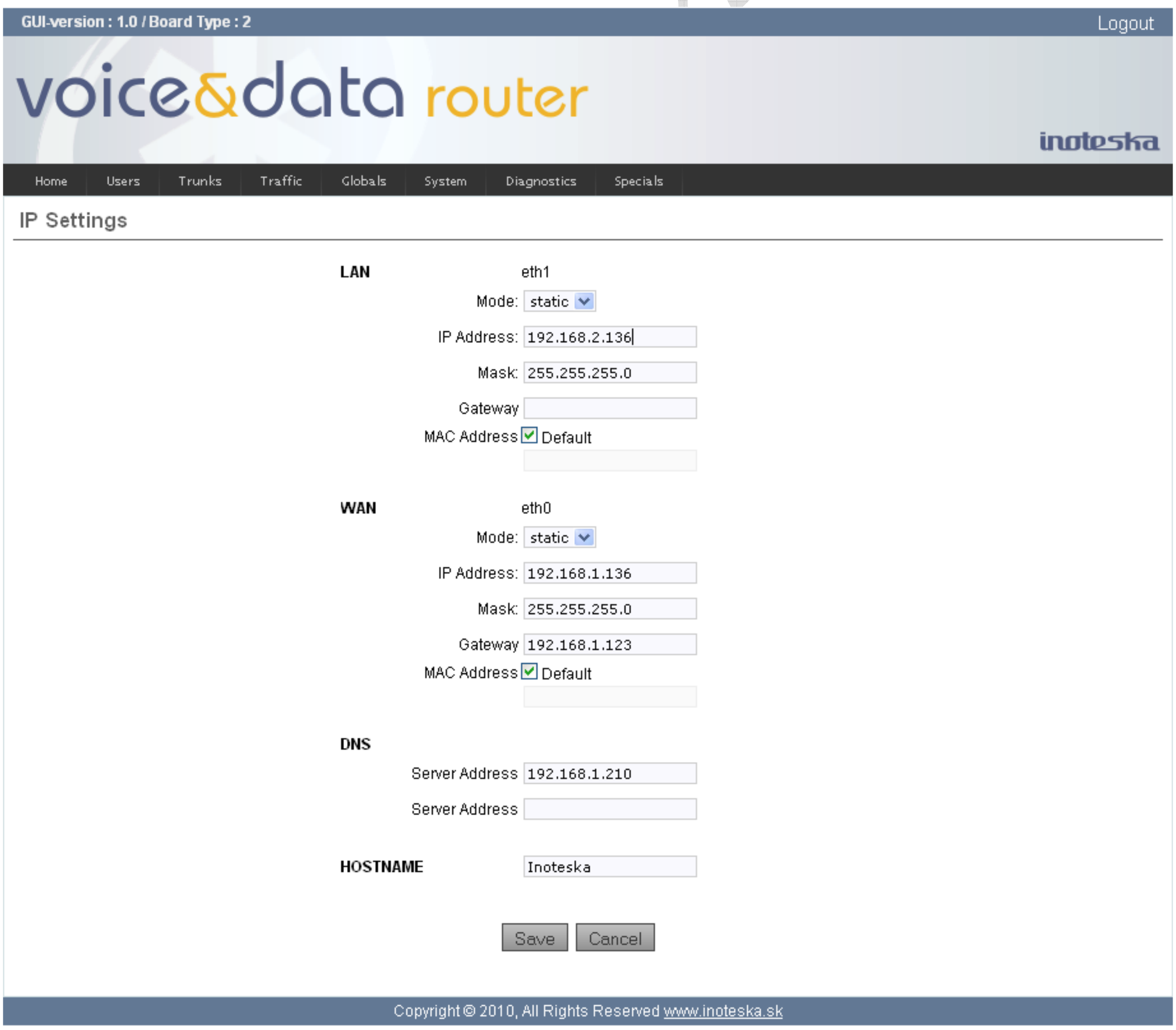

## Play sound




## 3.1.5 System

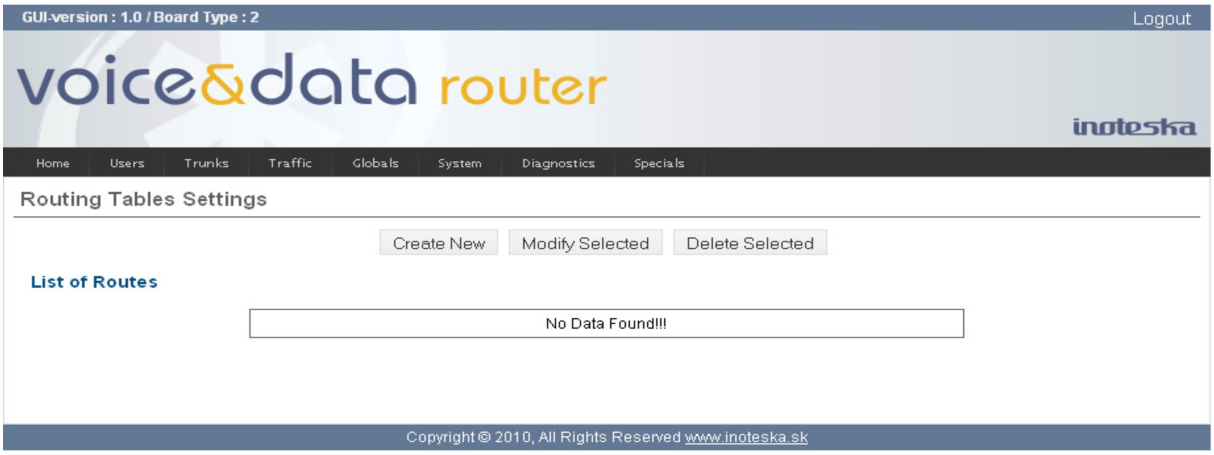
**IP settings**

– settings for **LAN,WAN, DNS, HOSTNAME**



## RoutingTables

**Create, Modify,Delete** of list of Routing tables, you can set **IP address,Mask, Gateway** for each **Route**



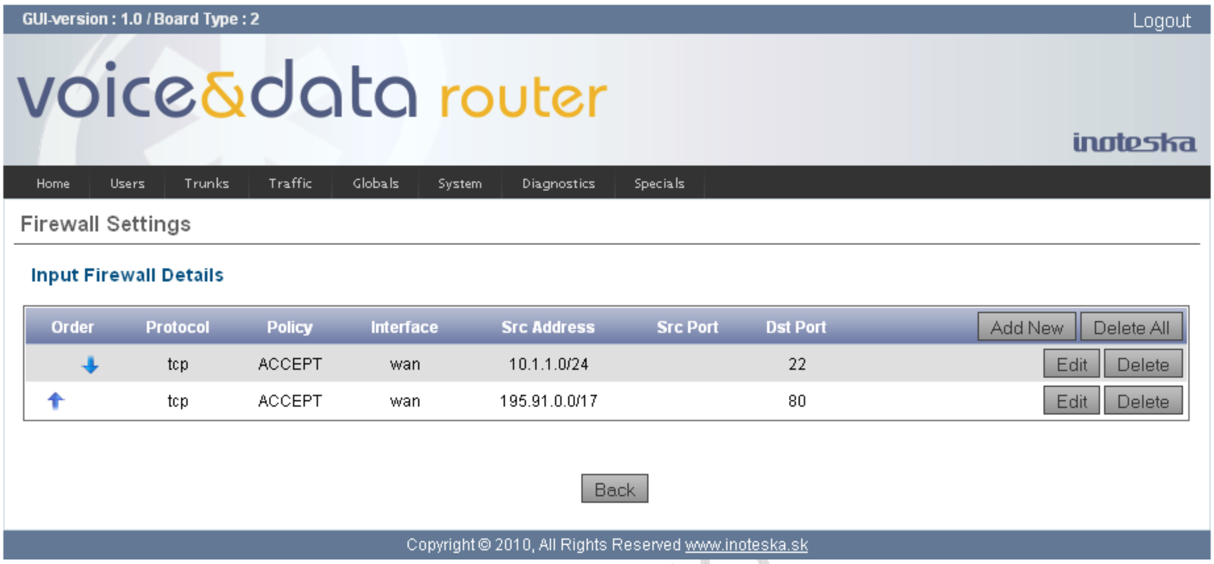


## Firewall

### General Settings

**Enable Input Firewall -** Enables/Disables Firewall functions.
**Global Policy -** Select Accept for enalble or Drop for disable as a default policy.
**Enable Forward Firewall -** Enables/Disables Route functions.
**Global Policy -** Select Accept for enalble or Drop for disable as a default policy.
**Enable Source NAT -** Enables/Disables NAT functions for LAN to WAN traffic.

## Input Firewall Details




Add New/Delete all/Edit Firewall Rules. Use Downward/Upward Arrows to manage Calling Rules order – move them up or down.

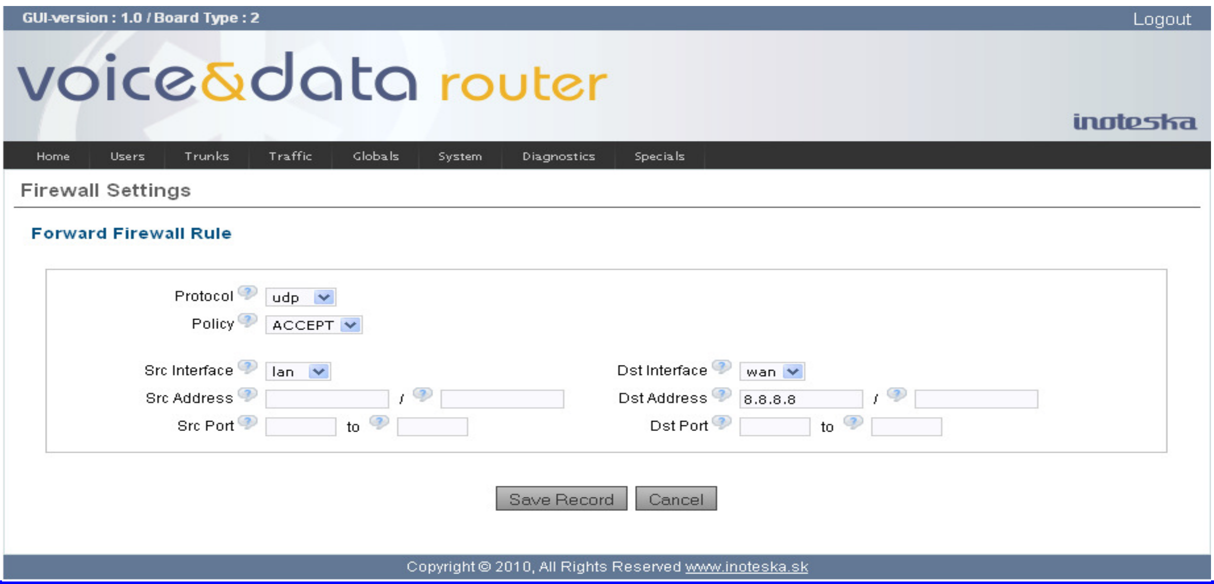
## Input Firewall Rule


**Protocol -** The rule is applicated for selected protocol only.
**Policy -** Select Accept for enalble or Drop for disable this rule.
**Interface -** The rule is applicated for selected interface only.

**Src Address/… -** The address can be entered in this forms: anywhere, a.b.c.d (192.168.1.15), a.b.c.d/e (192.1.168.0/24), a,b.c.d/m1.m2.m3.m4 (192.168.1.0/255.255.255.0) / Network mask.
**Src Port … to … -** The port can be entered in this form:
all,p 80:81 HTTP/TCP
5060 SIP/UDP
20:21 FTP/TCP
22 SSH/TCP
25 SMTP /TCP
53 DNS/UDP
123 NTP/UDP
161:162 SNMP/UDP
**Dst Port … to … -** The port can be entered in this form:
all,p 80:81 HTTP/TCP
5060 SIP/UDP
20:21 FTP/TCP
22 SSH/TCP
25 SMTP /TCP
53 DNS/UDP
123 NTP/UDP
161:162 SNMP/UDP

## Forward Firewall Details


## Forward Firewall Details - Edit



**Protocol**
**Policy**
**Src Interface**
**Src Address**
**Src Port**

**Dst Interface**
**Dst Address**
**Dst Port**

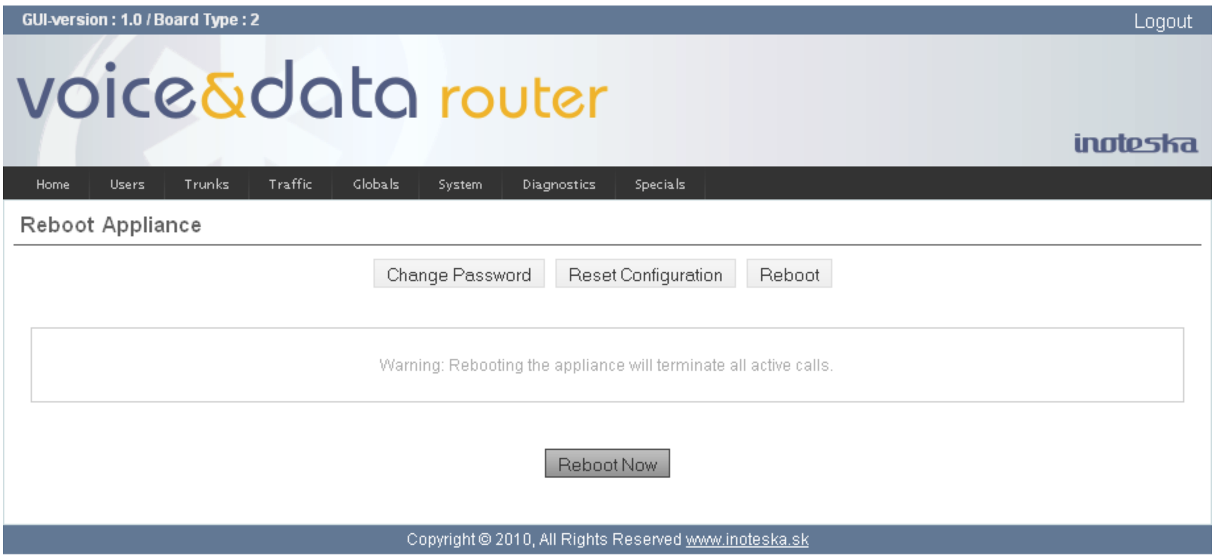
## DNAT Details


## DNAT Details - Edit

## Options

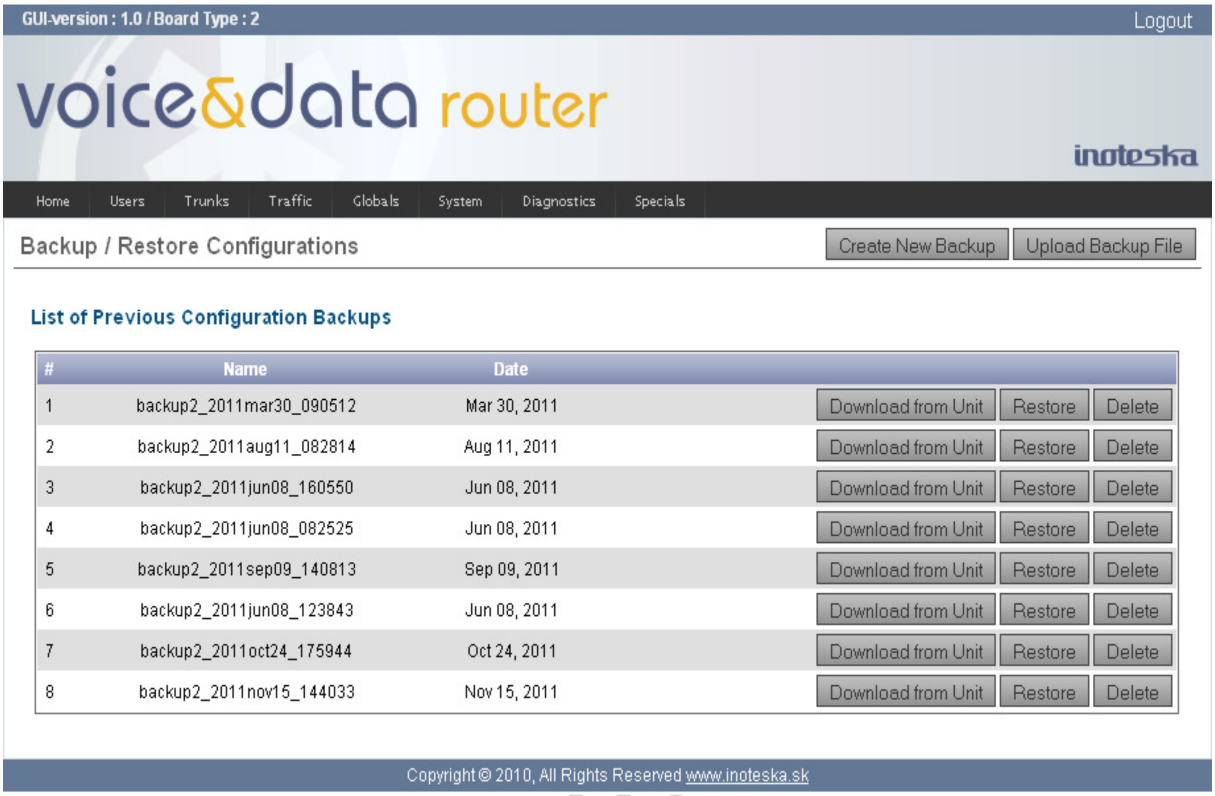
– settings of General **Preferences, Change Password, Reset Configuration, Reboot, DHCP server**



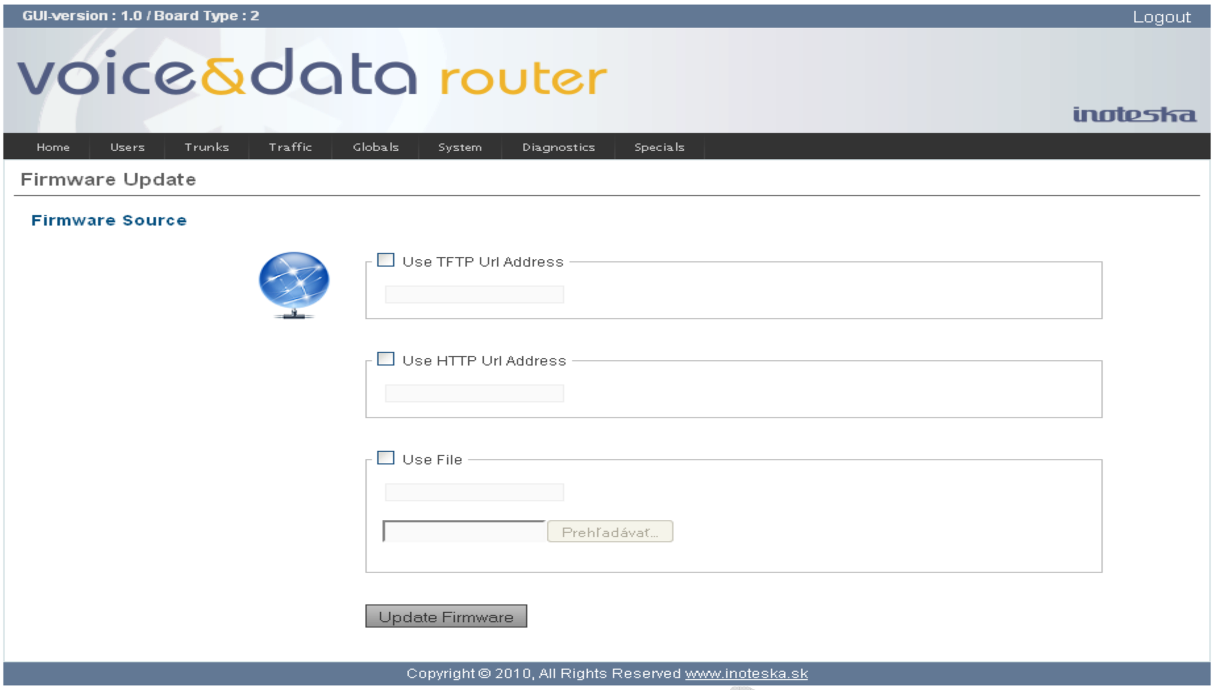


## Backup&Upload

– **Backup** of actual configuration of **Restore/Upload** of older one from the list.

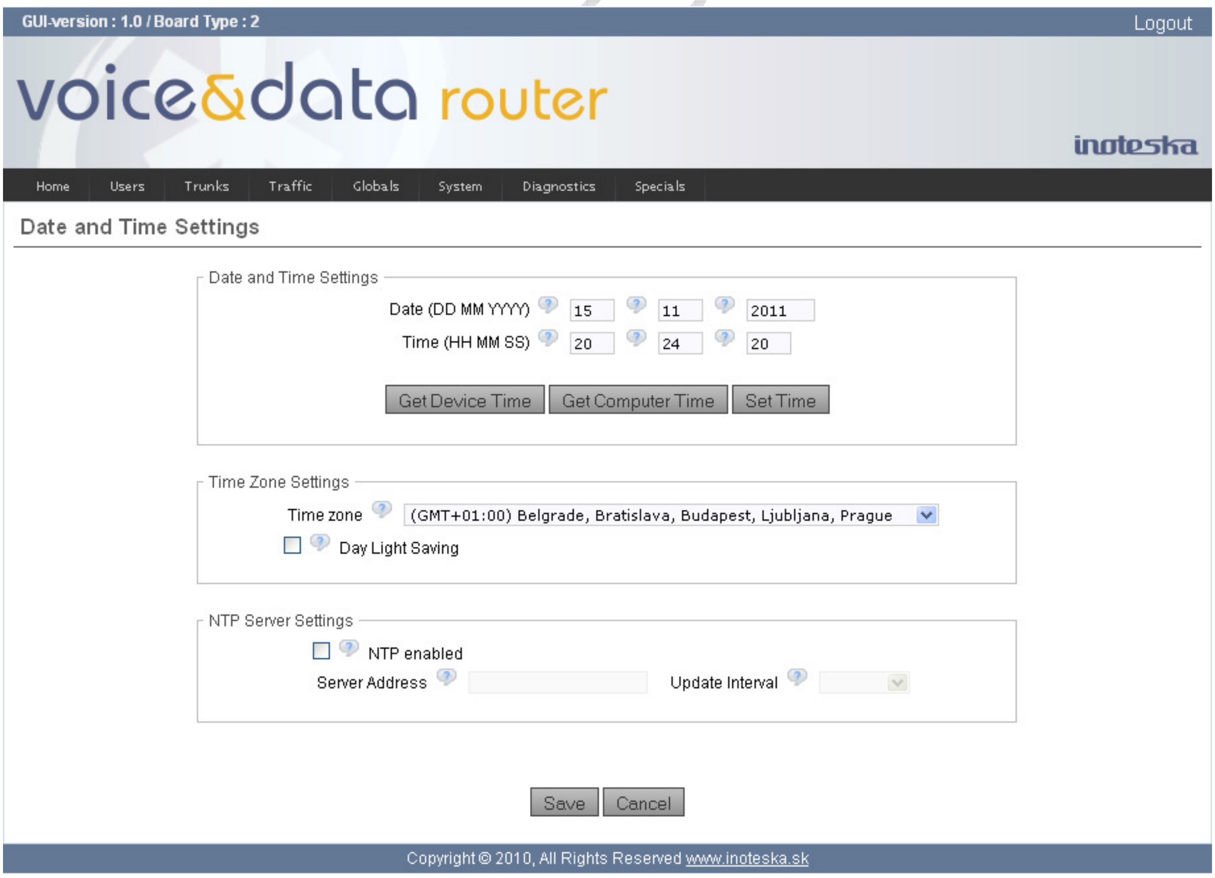



## Firmware Update



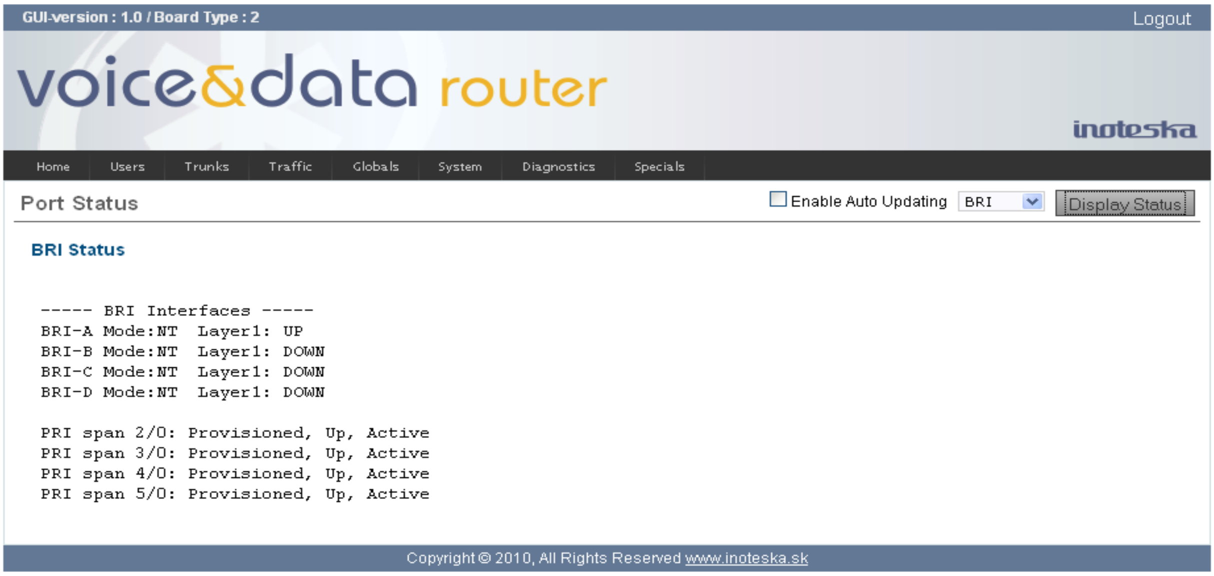

## Date&Time

**– Date&Time Settings, Time Zone settings, NTPServer Settings**

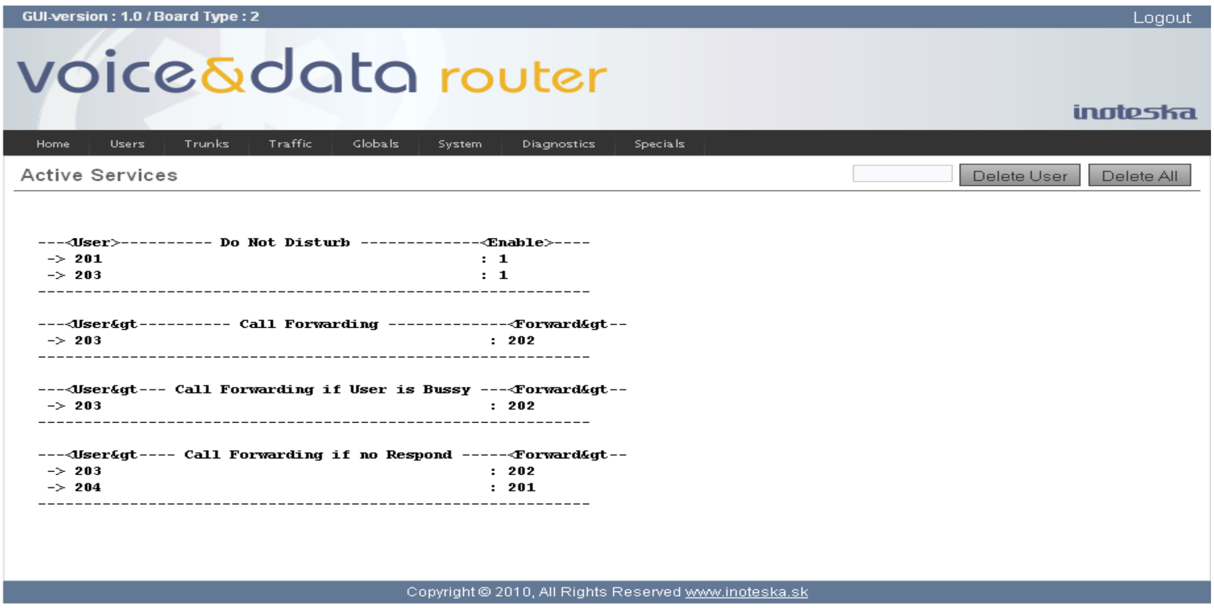


### 3.1.6 Diagnostics

**Port Status**

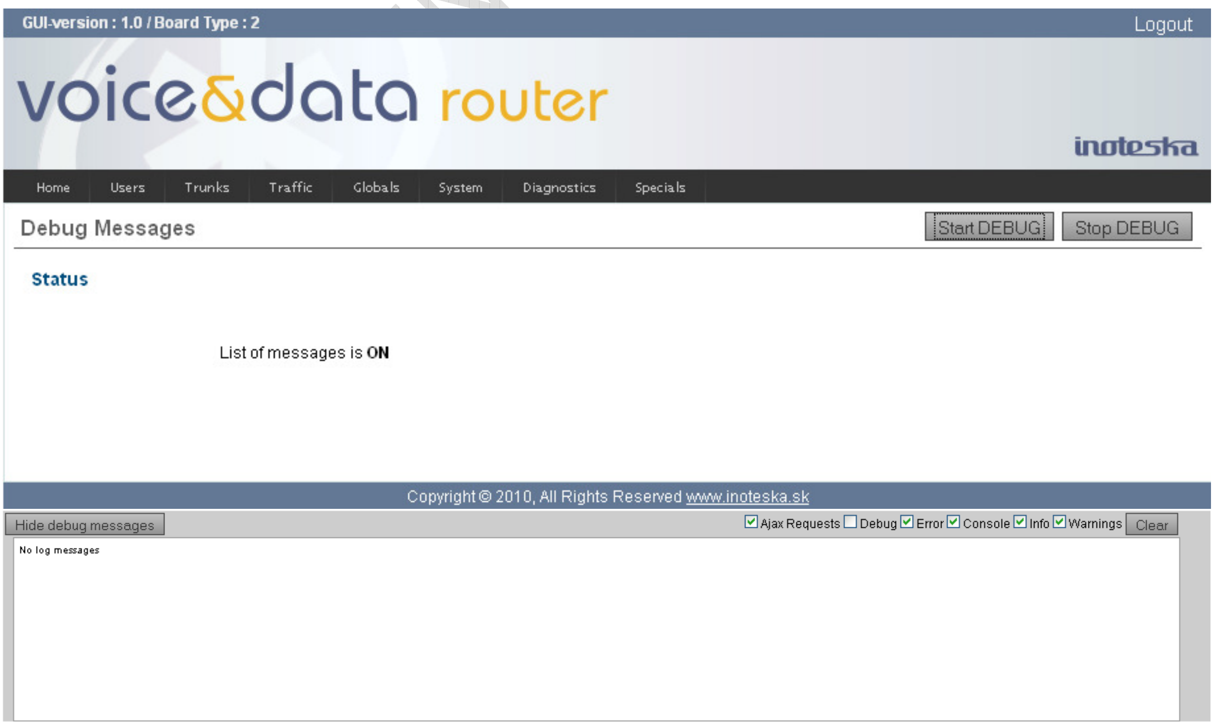




**Active Services**

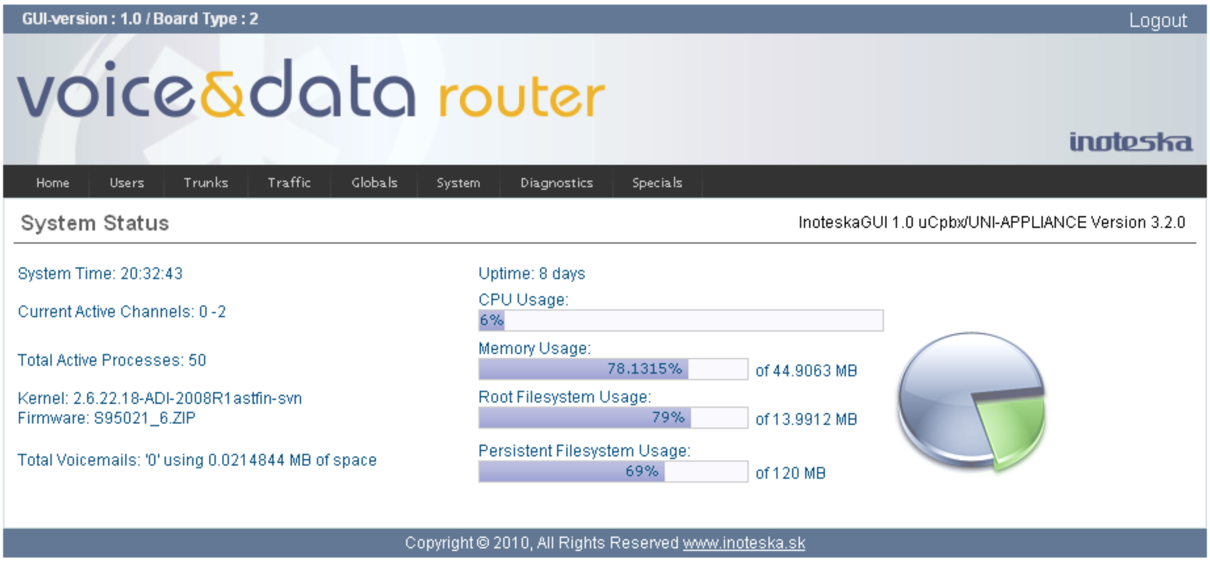


## Identification


## Debug messages



## System Status




## File Editor


**IP Tables Tests**

**SMS Manager Tests**

### 3.1.7 Specials

**AOC Web Update**

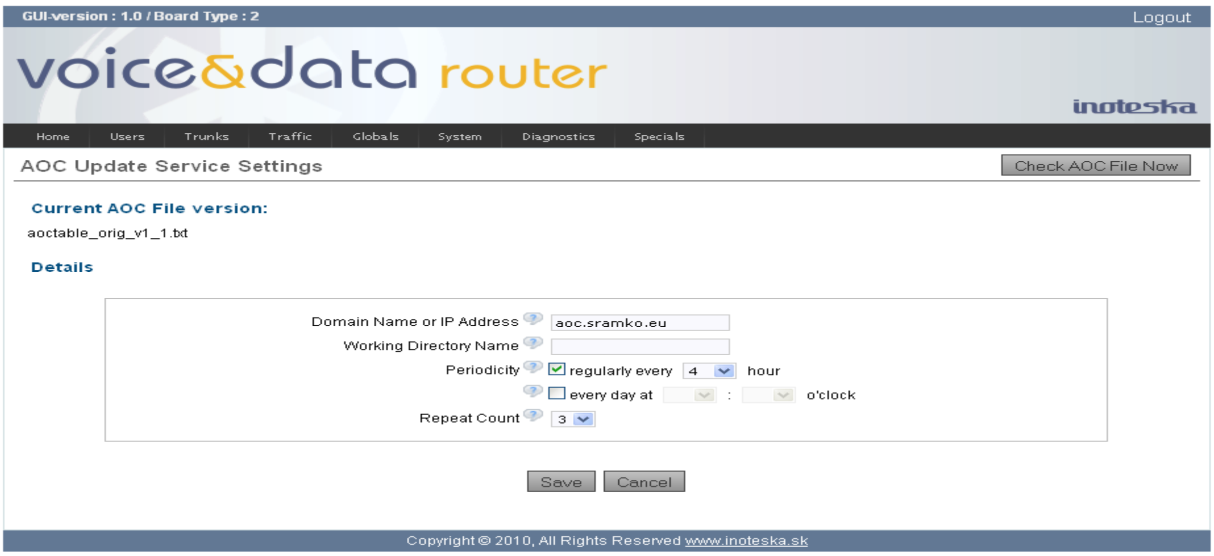

**Domain Name or IP Address :** name or address of web server where AOC files are saved

**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 – see pic below.



# 4. SALES CONDITIONS

**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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Podtureň-Roveň 221
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033 01
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**Tel.:** + 421 44 55 679 11
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**Hotline:** + 421 902 774 538

**Web:** www.inoteska.sk
**E-mail:** email@inoteska.sk



**VAT no.:** SK2020428300
**Bank information:** Všeobecná úverová banka a.s.
**Account no.:** 616243342/0200
**SWIFT code:** SUBASKBX
**IBAN:** SK3402000000000616243342

***Sales department:***

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| Ing. Ľubica Brtáňová | 0903 826 079 |
| Ing. Pavel Wolf | 0903 800 133 |

***Technical department:***

| | |
|---|---|
| Ing. Pavol Perdek | 0903 519 908 |
| Ing. Pavel Wolf | 0903 800 133 |