

VOICE&DATA ROUTER BA

PRODUCT DOCUMENTATION



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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:

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

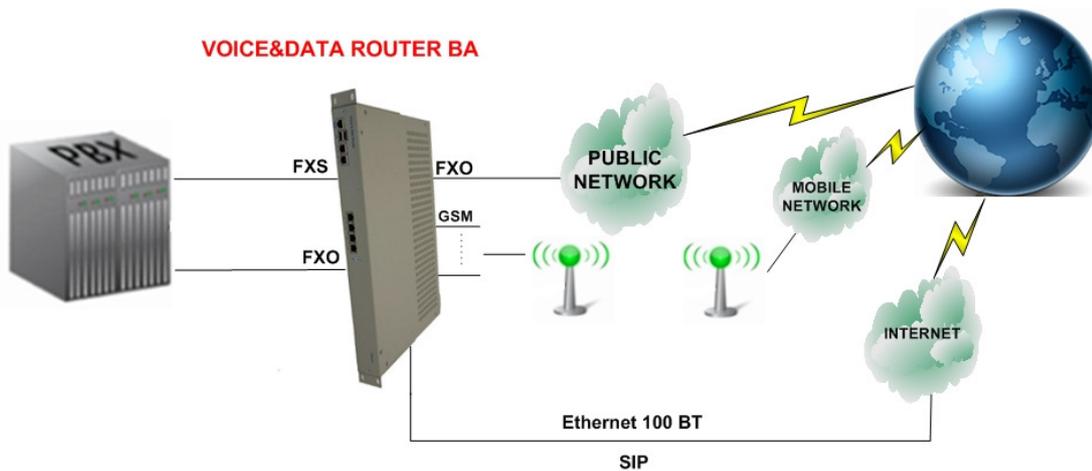
19" rack version 6U:

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

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

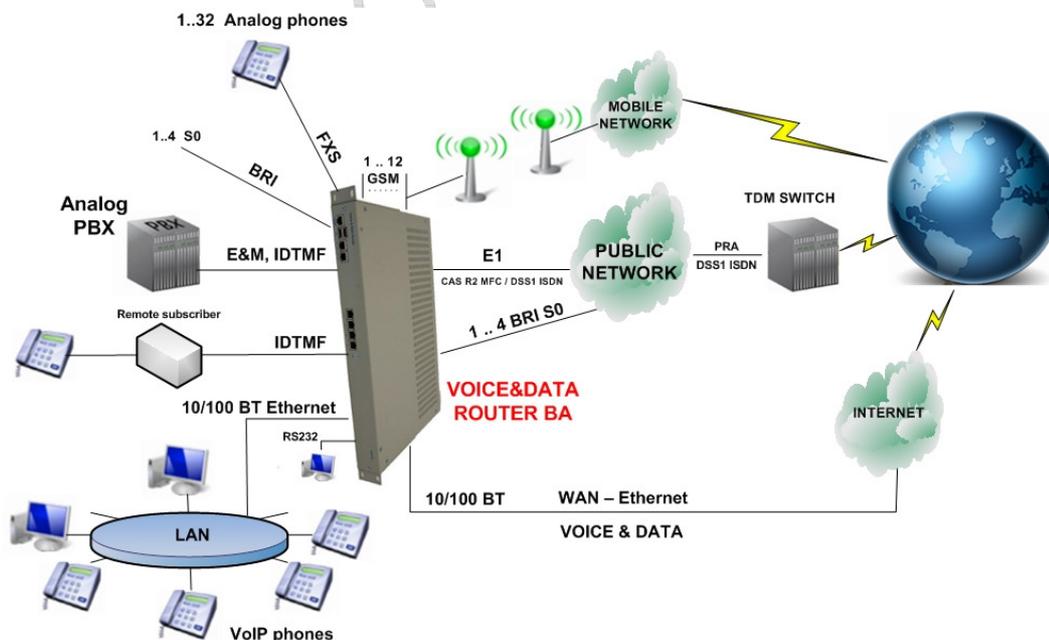
APPLICATIONS

- Voice & Data Router



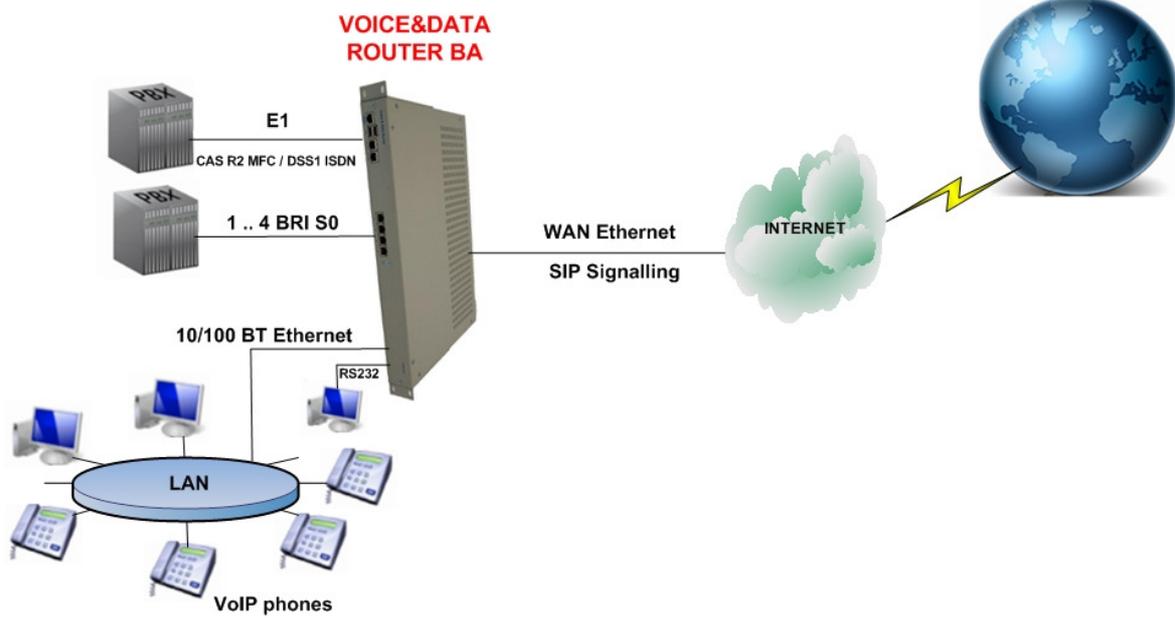
- VoIP, TDM PBX (SW Asterisk)

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

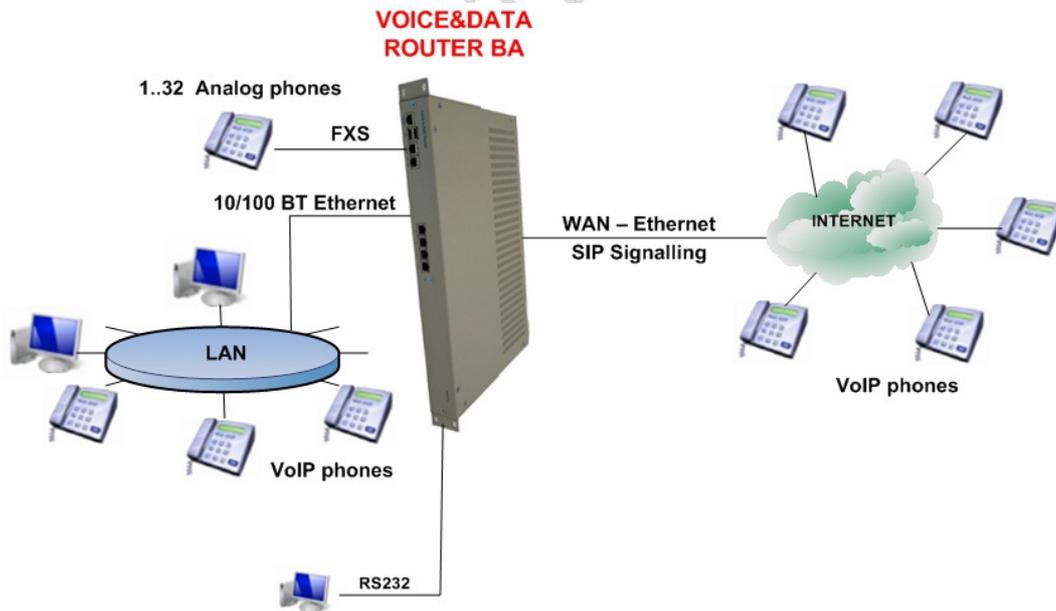


VOICE&DATA ROUTER BA

- TDM – VoIP gateway

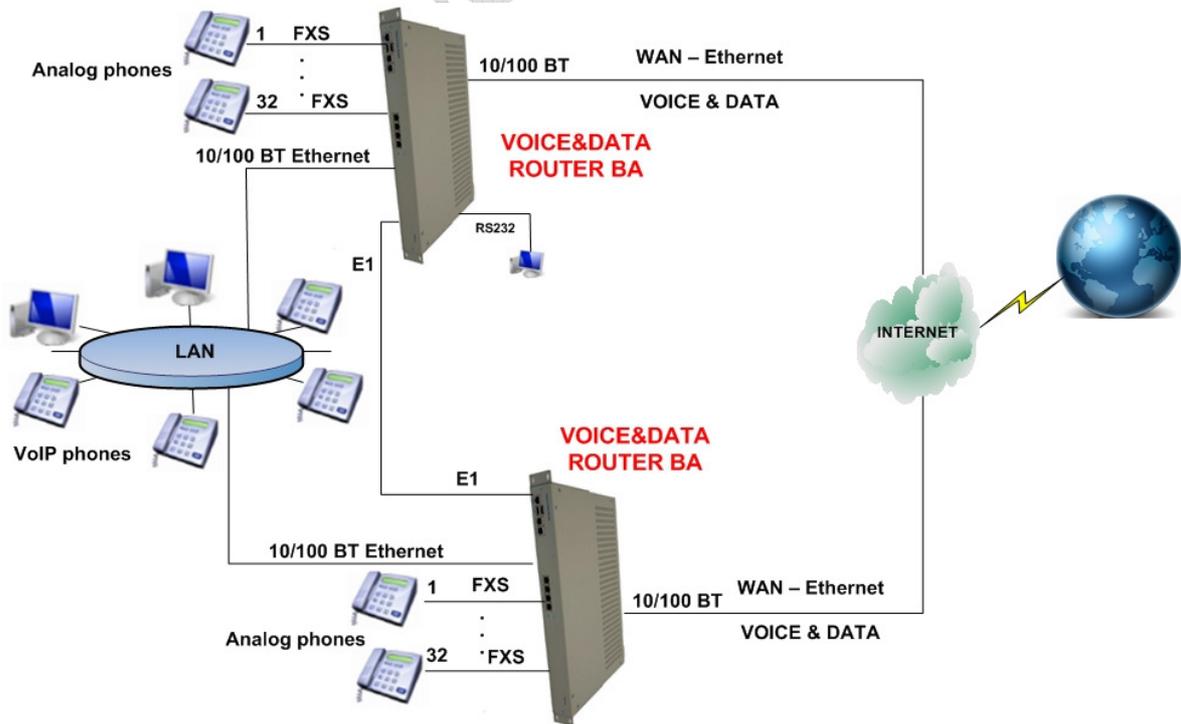
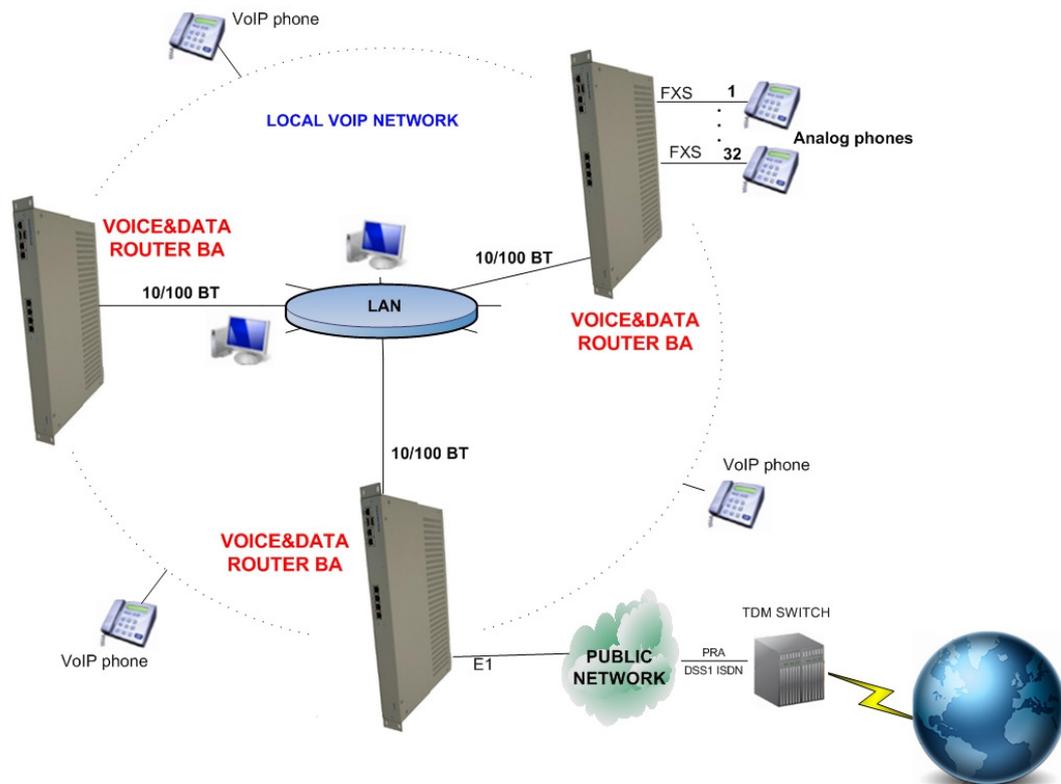


- SIP PROXY Server

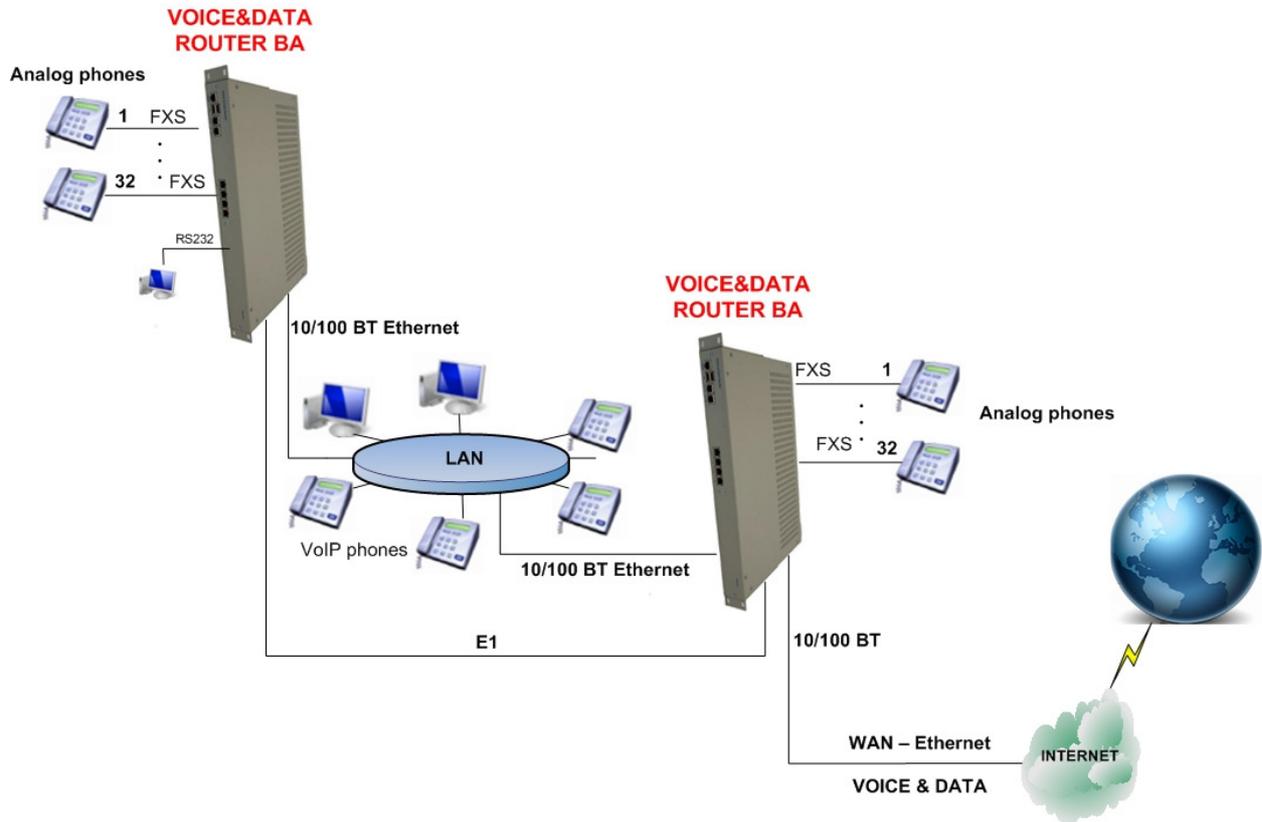


VOICE&DATA ROUTER BA

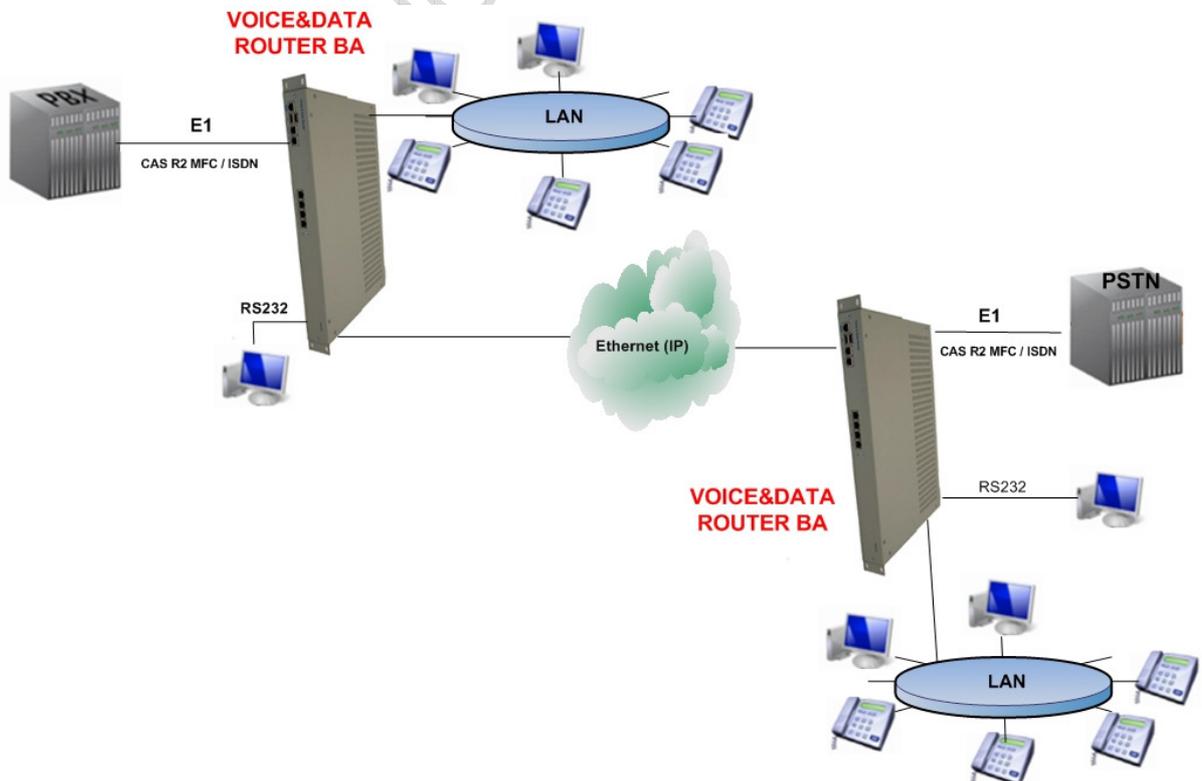
- Local network, extending of local network



VOICE&DATA ROUTER BA



- 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 Canceller - 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

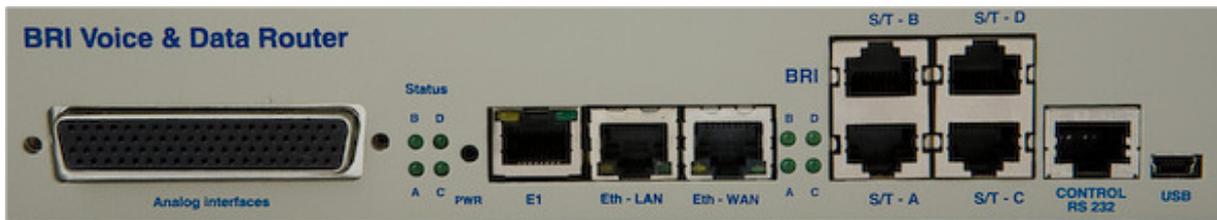
Install the device in a place where:

- 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

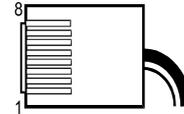
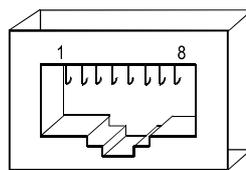
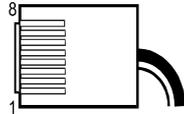
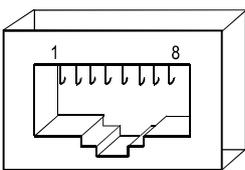
- 1 – input wire to device
- 2 – input wire to device
- 3 –
- 4 – output wire from device
- 5 – output wire from device
- 6 –
- 7 –
- 8 –

- Rx -
- Rx+
- Tx -
- Tx+

Ethernet 10/100Base-T interface

RJ 45 connector

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



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

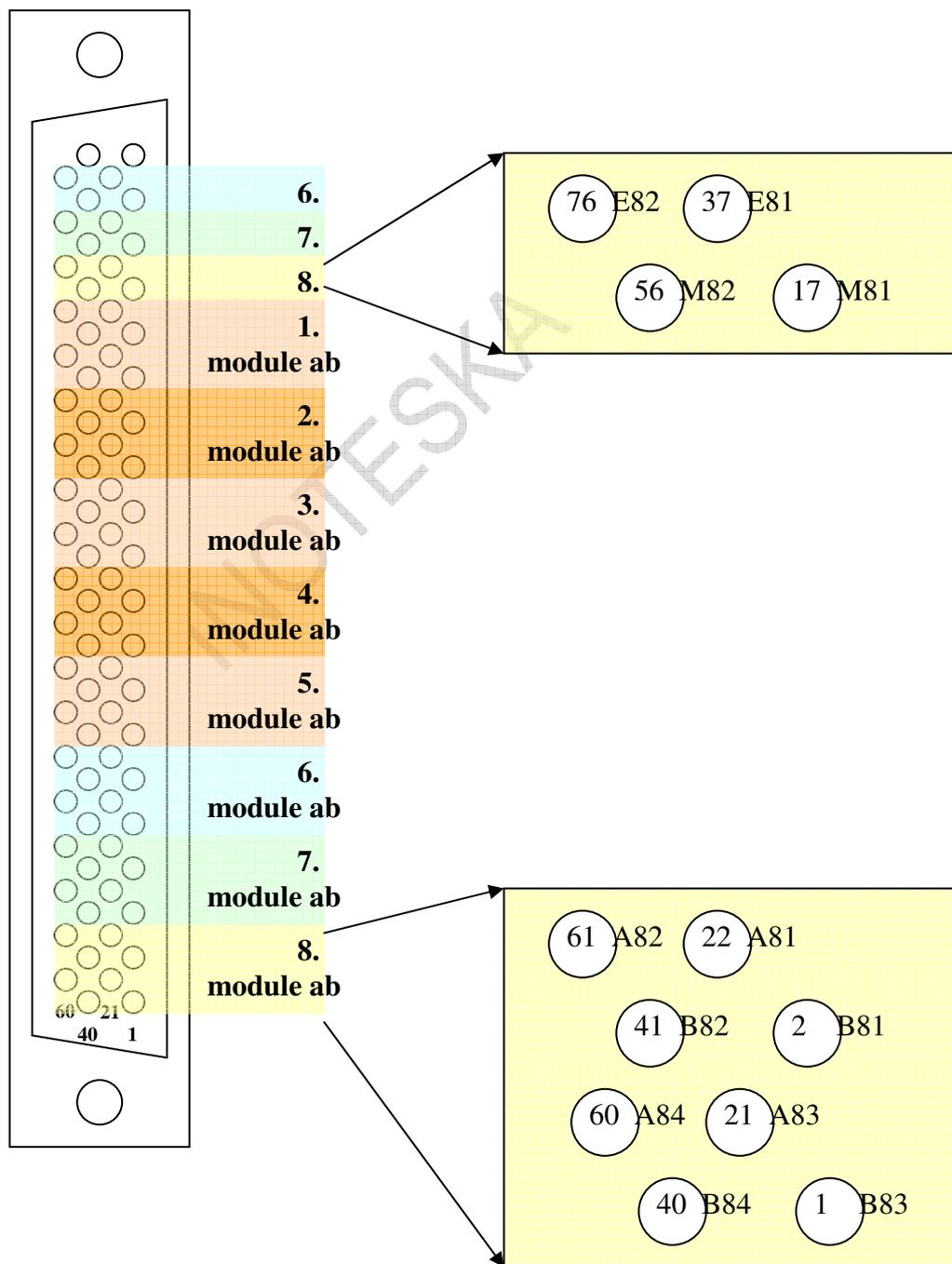
FXO module – dual

LB module – dual

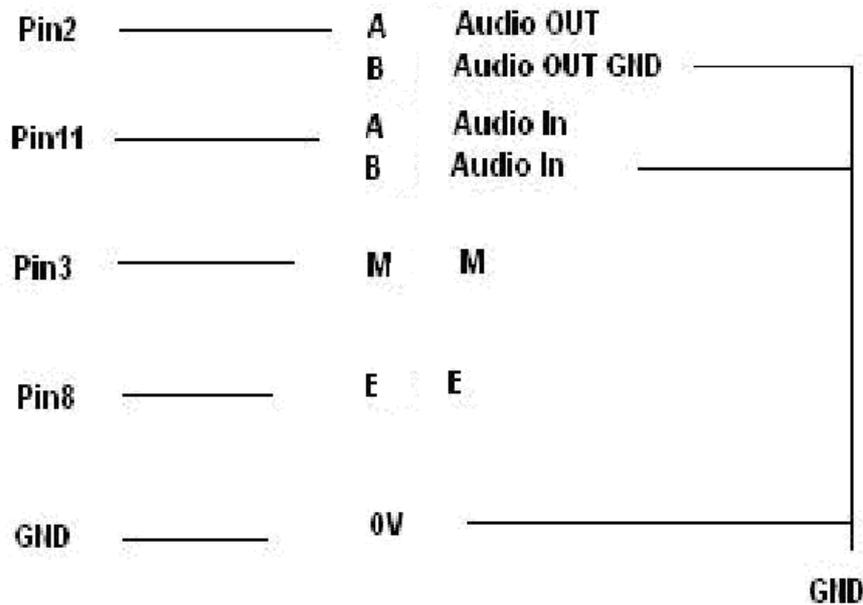
E&M module – dual

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

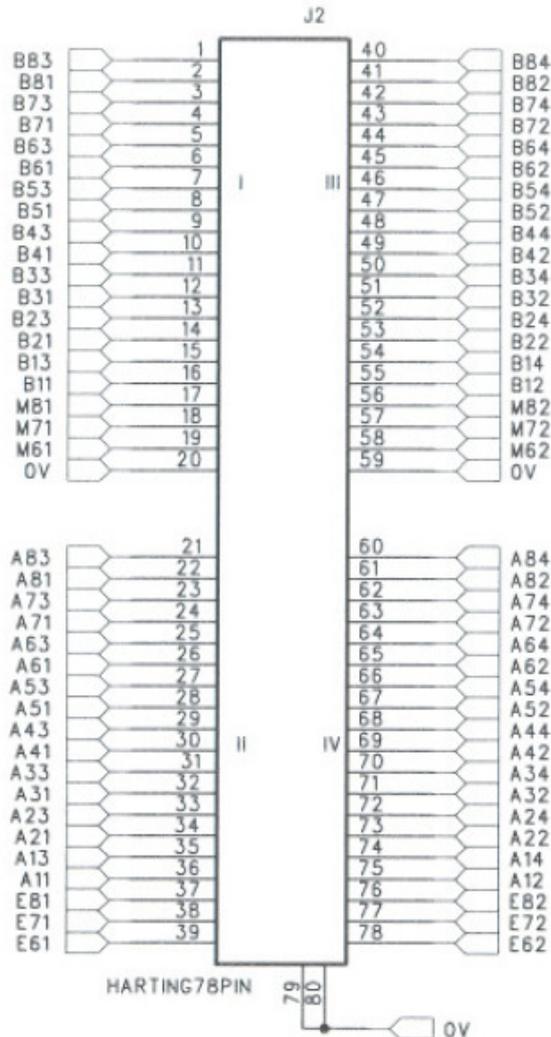
Signal	PIN	Meaning
A71	24	Audio Out 3
B71	4	Audio Out GND 3
A72	63	Audio In 3
B72	43	Audio In GND 3
E71	38	E 3
M71	18	M 3
0V	20	GND

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

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 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

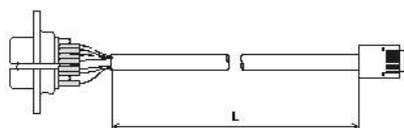
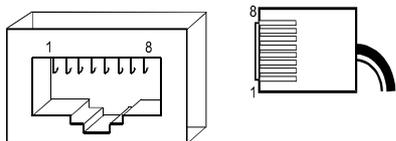
- modified product variant for max. 16 interfaces



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 0th 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 supervision. Web manager was tested with following browsers: **Opera, Google Chrome, Mozilla Firefox.**

Initial screen:

GUI-version: 1.0

voice&data router

inoteska

INOTESKA Configuration Engine

Username: admin

Password:

Login

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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.

GUI-version : 1.0 / Board Type : 2 Logout

voice&data router

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Home Users Trunks Traffic Globals System Diagnostics Specials

System Home Uptime : 09:40:26 up 6 days, 22:39, load average: 0.08, 0.01, 0.00

Device Configuration

Available Interfaces	Count
E1/T1	1
BRI	4
GSM	4
Analog Trunks	6
Analog Users	20
SIP Trunks	1
SIP Users	2

Identification

Device Information
TYPE=ITX4024201
SN=402420100014
Firmware=S95021_6.ZIP

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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
 – Analog Users

Trunks – SIP Trunks
 – E1/T1 trunks
 – BRI Trunks
 – Analog Trunks
 – 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

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

SIP Users Settings Create New User Modify Selected Users Delete Selected Users

List of User Extensions

<input type="checkbox"/>	Extension	Full Name	Host IP Address	DialPlan	
<input type="checkbox"/>	250	User 250	dynamic	users	Permit/Deny Edit Delete
<input type="checkbox"/>	251	User 251	dynamic	users	Permit/Deny Edit Delete

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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 addresses – for extensions with **dynamic** IP address – more information after click on Permit/Deny), **Edit** , **Delete**.

The screenshot shows the 'Edit User Extension - 250' configuration page. The interface includes a navigation bar with 'Home', 'Users', 'Trunks', 'Traffic', 'Globals', 'System', 'Diagnostics', and 'Specials'. The main content area is divided into several sections:

- General:** Extension (250), Name (User 250), DialPlan (users), Password, and CallerID.
- Host Options:** Host Type (dynamic), Host IP Address (dynamic), and Host Port (5060).
- Enable Voicemail for this User:** A checkbox that is currently unchecked.
- Access PIN code:** Access PIN code, Mailbox (10), and Email Address.
- Codecs:** First (a-law), Second (G.729), Third (None), Fourth (None), and Fifth (None).
- VoIP Settings:** Quality (2), NAT (checked), Can Reinvite (unchecked), DTMF Mode (RFC2833), and insecure (no).
- Other Options:** Pickup Group (1).
- User's Services:** Call Forward (checked) and Do Not Disturb (checked).
- User's Permission:** User's Permission Group (Users) and User's Call Limit (1).

At the bottom of the form are 'Update' and 'Cancel' buttons. The footer contains the copyright notice: 'Copyright © 2010, All Rights Reserved www.inoteska.sk'.

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 from 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

Quality: Quality 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 streams 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 Forward: Call Forward 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

GUI-version : 1.0 / Board Type : 2 Logout

voice&data router

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Home Users Trunks Traffic Globals System Diagnostics Specials

Users > SIP Users

Edit Permit/Deny Option

Add Permit/Deny Option

IP

Network Mask



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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**.

GUI-version : 1.0 / Board Type : 2 Logout

voice&data router

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Home Users Trunks Traffic Globals System Diagnostics Specials

Analog Users Settings

List of Analog Users

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

GUI-version : 1.0 / Board Type : 2 Logout

voice&data router inoteska

Home Users Trunks Traffic Globals System Diagnostics Specials

Users > Analog Users

Edit Details

General

Position 1 Channel 44

Extension Full Name

DialPlan

User's Services

User's Permission Group

Pickup Group

3-Way Calling Call Transfer Call Forwarding Do Not Disturb

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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

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

SIP Trunks Settings

Provider Name	DialPlan	Hostname/IP	Username
Inoteska	trunk	192.168.1.1	Inoteska

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

The screenshot shows the 'Edit SIP trunk Inoteska' configuration page. The fields and their values are as follows:

- Provider Name: Inoteska
- DialPlan: trunk
- Hostname: static, 192.168.1.1 : 5060
- Username: Inoteska
- Authuser: (empty)
- Fromuser: (empty)
- Fromdomain: (empty)
- Password: Inoteska
- Contact: (empty)
- Qualify: 2
- Insecure Type: very
- Codecs: First u-law, Second a-law, Third GSM, Fourth None, Fifth None
- CallerID: (empty)
- VoIP Settings: Register , NAT , Can Reinvite , DTMF Mode RFC2833
- Call Limit: 0
- Fax & Modem Support: Enable T.38
- Fax Passthrough Codec alaw
- Modem Passthrough Codec alaw

Buttons: Save, Cancel

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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 from 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 authentication, 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 authentication

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 streams 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.

GUI-version : 1.0 / Board Type : 2 Logout

voice&data router inoteska

Home Users Trunks Traffic Globals System Diagnostics Specials

E1/T1 Trunks Settings

E1/T1 Interfaces

Interface	Name	Line Mode	Signaling	DialPlan	Status
A	E1A	E1	isdn	TestE1	ON

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Home Users Trunks Traffic Globals System Diagnostics Specials

Interface A

General Settings

Enabled

Name Line Mode Signaling

DialPlan

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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**.

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Trunks > E1/T1 Trunks

Details: Interface A

Line Parameters

CRC Clock Mode Slave 1 Longhaul RX Longhaul TX

ISDN Details

Switch Type EuroISDN Mode TE Restart L3

Channels

Channels 1-15,17-31 Signaling Channel 16

Channel Selection Ascending Cyclic

Parameters

Overlap Dial

Echo Cancellation default

AOC

Generate AOC Charging Table1 Add Incoming Charging

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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 coming 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.

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

BRI Trunks Settings

BRI Interfaces

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

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

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Interface 2

General Settings

Enabled

Name DialPlan

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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**.

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Interface 2

Line Parameters

Clock Mode

ISDN Details

Mode

Channels

Channels

Channel Selection Cyclic

Parameters

Overlap Dial

Echo Cancellation

AOC

Generate AOC Charging Add Incoming Charging

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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.

Position	Name	Trunk Type	DialPlan	CID Number	Channel		
17	EM_1	E&M	trunk		60	Unused	Edit
18	EM_2	E&M	trunk		61	Unused	Edit
25	FXO_A	FXO	FromFXO		68	Unused	Edit
26	FXO_B	FXO	FromFXO		69	Unused	Edit
29	trunk_em_29	E&M	trunk		72	Unused	Edit
30	trunk_em_30	E&M	trunk		73	Unused	Edit

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Each interface can be **Edited** or set as **Unused**.

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Home Users Trunks Traffic Globals System Diagnostics Specials

Edit Trunk Settings

General Settings

Trunk Type

Name

DialPlan

CID Number

Busy Detection

Enable

Count

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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.

GUI-version : 1.0 / Board Type : 1 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

GSM Trunks Settings

GSM Interfaces

Interface	Name	DialPlan	Channel	
1	GSM1	from_gsm	128	<input type="button" value="Edit"/>
2	GSM2	from_gsm	129	<input type="button" value="Edit"/>
3	GSM3	from_gsm	130	<input type="button" value="Edit"/>
4	GSM4	from_gsm	131	<input type="button" value="Edit"/>

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Edit parameters of each GSM interface.

GUI-version : 1.0 / Board Type : 1 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Edit Trunk Settings

General Settings

Name

DialPlan

PIN Number

Echo Cancellation

Audio Level

From GSM

To GSM

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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.

Calling Rules Settings

Calling Rules Groups

DialPlan is a collection of Call Rules. DialPlans are assigned to Users and Trunks.

Users to User

Calling Rules: Users to User	New Calling Rules Group Name	<input type="text"/>	Create
LocalUsers			Edit Delete

Users to Trunk

Calling Rules: Users to Trunk	New Calling Rules Group Name	<input type="text"/>	Create
ToAnalog			Edit Delete
ToGSM			Edit Delete
ToSip			Edit Delete
ToTDM			Edit Delete

Trunks to User

Calling Rules: Trunks to User	New Calling Rules Group Name	<input type="text"/>	Create
FromTrunk			Edit Delete

Trunks to Trunk

Calling Rules: Trunks to Trunk	New Calling Rules Group Name	<input type="text"/>	Create
FromFxo			Edit Delete
TDM_Analog			Edit Delete
To_SIP			Edit Delete

Users to User – Edit

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Traffic > Calling Rules

Edit Calling Rule: LocalUsers

Pattern

Permission Group

Send to Local Destination

Local Destination

Save Cancel

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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, preceded 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

GUI-version : 1.0 / Board Type : 2 Edit CallingRule ! Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Traffic > Calling Rules

Edit Calling Rule: ToGSM

Pattern

Permission Group

Send this call through trunk

Use Trunk Two Step Dial

Strip digits from front Strip digits from ID front

and Prepend these digits before dialing and Prepend these digits to ID

Use FailOver Trunk 1

fail over Trunk Two Step Dial

Strip digits from front Strip digits from ID front

and Prepend these digits before dialing and Prepend these digits to ID

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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

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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.

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 FailOver 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.

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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

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For example, the extension _NXXXXXX would match normal 7 digit dialings, while _1NXXNXXXXX would represent a three digit area code plus phone number, preceded 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 this call through trunk

Use Trunk - Defines the Trunk that calls, matching the specified pattern, will be placed through.

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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 FailOver 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**.

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DialPlans Settings

DialPlan is a collection of Call Rules. DialPlans are assigned to Users and Trunks.

DialPlan - Users

DialPlan - Users	Calling Rules	Create New DialPlan	
users	LocalUsers, ToSip, ToTDM, ToGSM, ToAnalog, services	<input type="text"/>	New DialPlan Edit Delete

DialPlan - Trunks

DialPlan - Trunks	Calling Rules	Create New DialPlan	
trunk	FromTrunk, TDM_Analog, To_SIP	<input type="text"/>	New DialPlan Edit Delete
TestE1	To_SIP	<input type="text"/>	Edit Delete
FromFXO		<input type="text"/>	Edit Delete

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DialPlan – Users - Edit

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Home Users Trunks Traffic Globals System Diagnostics Specials

Traffic > Dial Plans

Edit DialPlan users

	Calling Rules	Add Calling Rule	
↓	LocalUsers	LocalUsers	Add Delete
↑ ↓	ToTDM		Delete
↑ ↓	ToSip		Delete
↑ ↓	ToAnalog		Delete
↑	ToGSM		Delete

User's Extensions

Use Services Codes

Ignore Pattern

Ignorepat

Back

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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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Traffic > Dial Plans

Edit DialPlan trunk

Calling Rules		Add Calling Rule	FromTrunk	Add
↓	FromTrunk			Delete
↑ ↓	TDM_Analog			Delete
↑	To_SIP			Delete

Back

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

AOC Tables Settings

AOC Tables

Create/Delete/Edit AOC Tables.

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

voice&data router inoteska

Home Users Trunks Traffic Globals System Diagnostics Specials

AOC Tables Settings

AOC Tables Time Tables Holidays

AOC Tables

Name	Create New Table	Create
Table1		Edit Details Delete Table

[Download AOC File](#)
Right Click on the above link and download using the 'Save Link As..' option

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AOC Tables – Edit

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Prefix Table: Table1

Prefixes Time Table table2

Prefix Table

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Time Tables

Create/Delete/Edit Time Tables.

GUI-version : 1.0 / Board Type : 2 **Voce&Data Router Reloaded !!** Logout

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AOC Tables Settings

Time Tables

Name	Create New Table <input type="text"/>	<input type="button" value="Create"/>
table2		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table3		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table4		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table5		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table6		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table7		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table8		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>
table9		<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>

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Time Tables – Edit

GUI-version : 1.0 / Board Type : 2 Logout

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Time Tables: table3

Create New Item

Work Day Time Interval									
1	2	3	4	5	6	7	8	9	
00:00	60	0	1100	0	0	0	30	55	<input type="button" value="Set"/> <input type="button" value="Back"/> <input type="button" value="Edit Details"/> <input type="button" value="Remove"/>

Create New Item

Weekend Time Interval									
1	2	3	4	5	6	7	8	9	
00:00	60		1100				30	55	<input type="button" value="Edit Details"/> <input type="button" value="Remove"/>

Create New Item

Holiday Time Interval									
1	2	3	4	5	6	7	8	9	
00:00									<input type="button" value="Edit Details"/> <input type="button" value="Remove"/>

Holiday Table

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Holidays

Create/Delete/Edit Holidays Tables.

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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AOC Tables Settings

Holidays

Create New Table

Name	
HolTab1	<input type="button" value="Edit Details"/> <input type="button" value="Delete Table"/>

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Holidays – Edit

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Holidays: HolTab1

Date

Holidays

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3.1.4 Globals

SIP settings

General Preferences

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

SIP (Session Intitiation Protocol) Configuration Settings

General Preferences TOS Debug Notify NAT Misc Jitter Buffer Codecs

Context

Realm for digest authentication

UDP Port to bind to

IP address to bind to

Domain

Allow guest calls

Overlap dialing support

Allow Transfers

Enable DNS SRV lookups (on outbound calls)

Pedantic

Always auth reject

SIP Domain Support

From Domain

Auto Domain

Allow External Domains

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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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Intitation Protocol) Configuration Settings

General Preferences **TOS** Debug Notify NAT Misc Jitter Buffer Codecs

TOS for Signalling packets ?

TOS for RTP audio packets ?

TOS for RTP video packets ?

Music On Hold Interpret ?

Music On Hold Suggest ?

Language ?

Enable Relaxed DTMF ?

RTP TimeOut ?

RTP HoldTimeOut ?

Trust Remote Party ID ?

Send Remote Party ID ?

Generate In-Band Ringing ?

Server UserAgent ?

Allow Nonlocal Redirect ?

Add 'user=phone' to URI ?

DTMF Mode ?

Send Compact SIP Headers ?

Max Registration/Subscription Time ?

Min Registration/Subscription Time ?

Default Incoming/Outgoing Registration Time ?

Min RoundtripTime (T1 Time) ?

Time between MWI Checks ?

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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 promiscdir when redirects are made to the local system will cause loops since Asterisk is incapable of performing a 'hairpin' call.

Add 'user=phone' to Uri - 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/outgoing 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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Initiation Protocol) Configuration Settings

General Preferences TOS **Debug Notify** NAT Misc Jitter Buffer Codecs

Sip Debugging ?

Enable SIP debugging ?

Record SIP History ?

Dump SIP History ?

Status Notifications (Subscriptions) ?

Subscribe Context ?

Allow Subscribe ?

Notify on Ringing ?

Save Cancel

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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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Initiation Protocol) Configuration Settings

General Preferences TOS Debug Notify **NAT** Misc Jitter Buffer Codecs

Extern ip

Extern Host

Extern Refresh

Local Network Address

NAT mode

Allow RTP Reinvite

Save Cancel

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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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Initiation Protocol) Configuration Settings

General Preferences TOS Debug Notify NAT **Misc** Jitter Buffer Codecs

FAX Passthrough

T38 fax (UDPTL) Passthrough

Outbound SIP Registrations

Register

Register TimeOut

Register Attempts

Video

Max Bitrate (kb/s)

Support for SIP Video

Generate Manager Events

NonStandard G.726 Support

Save Cancel

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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

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Initiation Protocol) Configuration Settings

General Preferences TOS Debug Notify NAT Misc **Jitter Buffer** Codecs

Enable Jitter Buffer ?
 Force Jitter Buffer ?
 Log Frames ?
 Max Jitter Buffer ?
 Resync Threshold ?
 Implementation ?

Save Cancel

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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, usually sent from exotic devices and programs. Defaults to 1000.

Implementation - Jitterbuffer implementation, used on the receiving side of a SIP channel. Two implementations are currently available - 'fixed' (with size always equals to jbmaxsize) and 'adaptive' (with variable size, actually the new jb of IAX2).

Codecs

GUI-version : 1.0 / Board Type : 2 Apply Changes Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Globals > SIP settings

SIP (Session Initiation Protocol) Configuration Settings

General Preferences TOS Debug Notify NAT Misc Jitter Buffer **Codecs**

Allowed Codecs u-law a-law GSM G.729 H.261 H.263 H.264

Save Cancel

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Select allowed codecs from the list.

RTP/UDPTL Settings

GUI-version : 1.0 / Board Type : 2 Logout

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RTP/UDPTL Settings

RTP/UDPTL Port Range

Start port

End port

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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.

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Home Users Trunks Traffic Globals System Diagnostics Specials

CallGroups Settings

Call Groups List

Call Group Name	Interfaces		
BRI	BRI:BRIA, BRI:BRIB, BRI:BRIC, BRI:BRID	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
GSM	GSM:GSMA, GSM:GSMB, GSM:GSMC, GSM:GSMD	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>
ANALOG	FXO:FXO_A, E+M:EM_1, E+M:trunk_em_29	<input type="button" value="Edit"/>	<input type="button" value="Delete"/>

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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.

RingGroup List	Members		
Product	250, 201, 206, 222	Edit	Delete
test	202, 209	Edit	Delete
Products	250, 202, 210, 206	Edit	Delete

Ring Groups - Edit

GUI-version : 1.0 / Board Type : 2 Edit RingGroup ! Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Edit RingGroup - Product

RingGroup Name

Ring Group Members		Available Users
250 (SIP) User 250	<<<	251 (SIP) User 251
201 (TDM) User 201	<<	202 (TDM) User 202
206 (TDM) User 206	<	203 (TDM) User 203
222 (TDM) User 222	>	204 (TDM) User 204
	>>	205 (TDM) User 205
		207 (TDM) User 207
		208 (TDM) User 208
		209 (TDM) User 209
		210 (TDM) User 210
		211 (TDM) User 211
		212 (TDM) User 212

Ring Group Options

Strategy

Seconds to ring each member

If not answered Goto

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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.

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Home Users Trunks Traffic Globals System Diagnostics Specials

Carriers Settings

List of Carriers

Name	Carrier Prefix	Ringing Tone	
Carrier1	0990	Generate	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

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For each operator you can define:

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Globals > Carriers

Carrier's Options

Details

Name

Carrier Prefix

Security Code

Ringing Tone

Save Cancel

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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**.

Permission Settings Save Changes Cancel Changes

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

Permission Groups

1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	Permission Group
<input checked="" type="checkbox"/>	Administrator															
<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	Users
<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	Guest
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																
<input type="checkbox"/>																

Permission Flags

Permission Flag	Flag Name
1	Local calls
2	National calls
3	International calls
4	flag3
5	flag4
6	flag5
7	flag6
8	flag7

Services Codes

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Services Codes Settings

Set Call Forwarding

Set Call Forwarding if user is busy

Set Call Forwarding if no respond No respond time:

Clear Call Forwarding

Clear Call Forwarding if user is busy

Clear Call Forwarding if no respond

Set Do Not Disturb

Clear Do Not Disturb

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CDR Settings

CDR Details

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CDR Settings

CDR Details

CDR Note

CDR User to User

CDR Trunk to User

CDR User to Trunk

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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.

GUI-version : 1.0 / Board Type : 2 Logo

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Home Users Trunks Traffic Globals System Diagnostics Specials

CDR Settings CDR Details CDR Records

CDR Records Refresh View Delete File

```

rec0,201,2011-11-03 13:37:40,3,s,
rec0,201,2011-11-03 13:37:54,3,s,
rec0,201,2011-11-03 13:38:34,6,s,
rec0,201,2011-11-03 13:38:59,4,s,
rec0,201,2011-11-03 13:52:45,4,s,
rec0,201,2011-11-03 13:52:56,2,s,
rec0,202,2011-11-03 13:53:11,3,s,
rec0,202,2011-11-03 13:53:20,3,s,
rec0,202,2011-11-03 13:58:04,4,s,
rec0,202,2011-11-03 13:58:31,3,s,
rec0,202,2011-11-03 14:01:46,0,s,
rec0,201,2011-11-03 14:02:43,0,s,
rec0,202,2011-11-03 14:06:07,3,s,
rec0,202,2011-11-03 14:15:10,3,s,
rec0,205,2011-11-03 14:30:53,4,s,
rec0,205,2011-11-03 14:31:20,2,s,
rec0,205,2011-11-03 14:31:30,3,s,
rec0,205,2011-11-03 14:33:07,7,s,
rec0,205,2011-11-03 14:34:09,5,s,
rec0,2011-11-03 14:34:24,6,s,
rec0,2011-11-03 14:34:40,3,s,
rec0,2011-11-03 14:36:24,4,s,
rec0,9049,2011-11-03 15:04:44,2,250,IN
rec0,9049,2011-11-03 15:34:33,1,205,IN
rec0,89,2011-11-03 15:34:43,0,205,IN
rec0,205,2011-11-04 10:09:59,3,s,
rec0,205,2011-11-04 10:13:08,3,s,
rec0,205,2011-11-04 10:13:30,2,s,
rec0,89,2011-11-04 13:40:41,13,205,IN
    
```

Announcement Settings

Management of announcement files.

Settings

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Announcement Settings Settings Upload File Record File Play Sound

Announcement Files

#	Announcement File Name	Download File	Delete File
1	welcome_message.wav	Download File	Delete File
2	test2.wav	Download File	Delete File
3	dlhyoznam.wav	Download File	Delete File
4	test2_is.wav	Download File	Delete File
5	test4.wav	Download File	Delete File
6	oznam1.wav	Download File	Delete File
7	oznam00.wav	Download File	Delete File
8	oznam0.wav	Download File	Delete File
9	testujme.wav	Download File	Delete File
10	abc.wav	Download File	Delete File
11	test9.wav	Download File	Delete File
12	tst.wav	Download File	Delete File
13	janko.wav	Download File	Delete File

Upload File

The screenshot shows the 'voice&data router' GUI. At the top, it says 'GUI-version : 1.0 / Board Type : 2' and 'Logout'. The main header features the 'voice&data router' logo and the 'inoteska' brand name. A navigation menu includes 'Home', 'Users', 'Trunks', 'Traffic', 'Globals', 'System', 'Diagnostics', and 'Specials'. The current page is 'Announcement Settings', with buttons for 'Settings', 'Upload File', 'Record File', and 'Play Sound'. The 'Upload File' button is highlighted. Below the navigation, the section is titled 'Upload Announcement File'. It contains a large empty rectangular box for file upload, with a small globe icon on the left and a 'Prehľadávať...' button on the right. The footer of the interface reads 'Copyright © 2010, All Rights Reserved www.inoteska.sk'.

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.

The screenshot shows the 'voice&data router' GUI. At the top, it says 'GUI-version : 1.0 / Board Type : 2' and 'Logout'. The main header features the 'voice&data router' logo and the 'inoteska' brand name. A navigation menu includes 'Home', 'Users', 'Trunks', 'Traffic', 'Globals', 'System', 'Diagnostics', and 'Specials'. The current page is 'Announcement Settings', with buttons for 'Settings', 'Upload File', 'Record File', and 'Play Sound'. The 'Record File' button is highlighted. Below the navigation, the section is titled 'Record New Announcement File'. It contains a large text box with the following instructions: '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.' Below this text box, there is a 'Record File' section with a dropdown menu for 'Extension' (currently showing 'SIP User -- 250'), a text input field for 'Announcement File Name', and a 'Start' button. The footer of the interface reads 'Copyright © 2010, All Rights Reserved www.inoteska.sk'.

Play sound

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Announcement Settings Settings Upload File Record File Play Sound

Play Announcement File

Select a user and the name of sound file.
After pressing "Start" button selected user is ringing.
Pick up the phone and the sound file is played.

Play Sound

Extension: SIP User -- 250 Announcement File: welcome_message.wav Start

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3.1.5 System

IP settings

– settings for LAN,WAN, DNS, HOSTNAME

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

IP Settings

LAN eth1

Mode: static

IP Address: 192.168.2.136

Mask: 255.255.255.0

Gateway:

MAC Address Default

WAN eth0

Mode: static

IP Address: 192.168.1.136

Mask: 255.255.255.0

Gateway: 192.168.1.123

MAC Address Default

DNS

Server Address: 192.168.1.210

Server Address:

HOSTNAME Inoteska

Save Cancel

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Routing Tables

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

GUI-version : 1.0 / Board Type : 2 Logout

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Routing Tables Settings

Create New Modify Selected Delete Selected

List of Routes

No Data Found!!!

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Add or Edit Route

IP address:

Mask:

Gateway:

Save Cancel

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Firewall

General Settings

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Firewall Settings Edit Input Firewall Edit Forward Firewall Edit DNAT

General Settings

Enable Input Firewall Global Policy

Enable Forward Firewall Global Policy

Enable Source NAT

Save & Apply Cancel

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Enable Input Firewall - Enables/Disables Firewall functions.

Global Policy - Select Accept for enable or Drop for disable as a default policy.

Enable Forward Firewall - Enables/Disables Route functions.

Global Policy - Select Accept for enable or Drop for disable as a default policy.

Enable Source NAT - Enables/Disables NAT functions for LAN to WAN traffic.

Input Firewall Details

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Firewall Settings

Input Firewall Details

Order	Protocol	Policy	Interface	Src Address	Src Port	Dst Port	
↓	tcp	ACCEPT	wan	10.1.1.0/24		22	Add New Delete All Edit Delete
↑	tcp	ACCEPT	wan	195.91.0.0/17		80	Edit Delete

Back

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

Input Firewall Rule

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Firewall Settings

Input Firewall Rule

Protocol

Policy

Interface

Src Address /

Src Port to

Dst Port to

Save Record Cancel

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Protocol - The rule is applied for selected protocol only.

Policy - Select Accept for enable or Drop for disable this rule.

Interface - The rule is applied 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

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Firewall Settings

Forward Firewall Details

Order	Protocol	Policy	Src Interface	Src Address	Src Port	Dst Interface	Dst Address	Dst Port	
	udp	ACCEPT	lan			wan	8.8.8.8		<input type="button" value="Add New"/> <input type="button" value="Delete All"/> <input type="button" value="Edit"/> <input type="button" value="Delete"/>

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Forward Firewall Details - Edit

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Firewall Settings

Forward Firewall Rule

Protocol

Policy

Src Interface

Dst Interface

Src Address /

Dst Address /

Src Port to

Dst Port to

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Protocol
Policy
Src Interface
Src Address
Src Port

Dst Interface
Dst Address
Dst Port

DNAT Details

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Firewall Settings

DNAT Details

Order	Protocol	Src Address	WAN Port	LAN Address	LAN Port		
	tcp		8080	192.168.1.1	8080	<input type="button" value="Add New"/>	<input type="button" value="Delete All"/>
						<input type="button" value="Edit"/>	<input type="button" value="Delete"/>

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DNAT Details - Edit

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Firewall Settings

DNAT Rule

Protocol

Src Address /

WAN Port to

LAN Address /

LAN Port to

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Options

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

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Change Password

Enter New Password

Retype New Password

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Reset to Factory Defaults

Warning: By resetting your PBX Appliance/System to factory defaults, you will lose all your configuration !
You can take a backup of your current configuration from the [Backup page](#).

Keep Network Settings

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Reboot Appliance

Warning: Rebooting the appliance will terminate all active calls.

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Backup&Upload

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

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Backup / Restore Configurations Create New Backup Upload Backup File

List of Previous Configuration Backups

#	Name	Date			
1	backup2_2011mar30_090512	Mar 30, 2011	Download from Unit	Restore	Delete
2	backup2_2011aug11_082814	Aug 11, 2011	Download from Unit	Restore	Delete
3	backup2_2011jun08_160550	Jun 08, 2011	Download from Unit	Restore	Delete
4	backup2_2011jun08_082525	Jun 08, 2011	Download from Unit	Restore	Delete
5	backup2_2011sep09_140813	Sep 09, 2011	Download from Unit	Restore	Delete
6	backup2_2011jun08_123843	Jun 08, 2011	Download from Unit	Restore	Delete
7	backup2_2011oct24_175944	Oct 24, 2011	Download from Unit	Restore	Delete
8	backup2_2011nov15_144033	Nov 15, 2011	Download from Unit	Restore	Delete

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Upload a previous backup File List of Backups Upload Backup File

File Uploading

 Upload

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Firmware Update

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Firmware Update

Firmware Source

 Use TFTP Url Address

Use HTTP Url Address

Use File

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Date&Time

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

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Date and Time Settings

Date and Time Settings

Date (DD MM YYYY)

Time (HH MM SS)

Time Zone Settings

Time zone

Day Light Saving

NTP Server Settings

NTP enabled

Server Address Update Interval

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3.1.6 Diagnostics

Port Status

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Port Status Enable Auto Updating

E1/T1
 BRI
 GSM
 Analog
 SIP
 ETH

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Port Status Enable Auto Updating

BRI

BRI Status

```

----- BRI Interfaces -----
BRI-A Mode:NT Layer1: UP
BRI-B Mode:NT Layer1: DOWN
BRI-C Mode:NT Layer1: DOWN
BRI-D Mode:NT Layer1: DOWN

PRI span 2/0: Provisioned, Up, Active
PRI span 3/0: Provisioned, Up, Active
PRI span 4/0: Provisioned, Up, Active
PRI span 5/0: Provisioned, Up, Active
    
```

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Active Services

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Active Services

```

-----<User>----- Do Not Disturb -----<Enable>-----
-> 201 : 1
-> 203 : 1
-----

-----<User>&gt;----- Call Forwarding -----<Forward>&gt;-----
-> 203 : 202
-----

-----<User>&gt;----- Call Forwarding if User is Busy -----<Forward>&gt;-----
-> 203 : 202
-----

-----<User>&gt;----- Call Forwarding if no Respond -----<Forward>&gt;-----
-> 203 : 202
-> 204 : 201
-----
    
```

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Identification

Identification

Details

```
[BASIC]
TYPE=ITX4024201
SN=402420100014
ID=6086
DATE=20.12.2010
MAN=103,105
DPS=ITB 256 02
PROD=INOTESKA
ADR=PODTUREM-ROVEN221,LIPTOVSKY HRADOK,03301
TEL=+421445567911,+421903360360
MAIL=MAIL@INOTESKA.SK
WWW=WWW.INOTESKA.SK
HW=2000
RD=D
SDRAM=64M
FLASH=8M
NAND=128M
CARD=SD
FPGA=ICDP1C6Q240C8
CPU=BF527
CON=1
E1=1
ETH=2
ANA=32
BRI=4
GSM=4
SERVP=F3DE1AF32994DA01EA9299293C6157B0
[CONFIG]
TYPE=ITX4024201
SN=402420100014
ID=6086
USERS=108
USERB=108
E1=0
ANA=0,1,2,3,4,5,6,7,8,9,10,11,12,13,14,15,16,17,18,19,20,21,22,23,24,25,26,27,28,29,30,31
ETH=0,1
BRI=0,1,2,3
[FUNCTIONS]
```

Debug messages

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

Debug Messages Start DEBUG Stop DEBUG

Status

List of messages is **ON**

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Hide debug messages Ajax Requests Debug Error Console Info Warnings

No log messages

System Status

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System Status InoteskaGUI 1.0 uCpbx/UNI-APPLIANCE Version 3.2.0

System Time: 20:32:43	Uptime: 8 days
Current Active Channels: 0 -2	CPU Usage: <div style="width: 6%;"><div style="width: 6%;"></div></div> 6%
Total Active Processes: 50	Memory Usage: <div style="width: 78.1315%;"><div style="width: 78.1315%;"></div></div> 78.1315% of 44.9063 MB
Kernel: 2.6.22.18-ADI-2008R1astfin-svn Firmware: S95021_6_ZIP	Root Filesystem Usage: <div style="width: 79%;"><div style="width: 79%;"></div></div> 79% of 13.9912 MB
Total Voicemails: '0' using 0.0214844 MB of space	Persistent Filesystem Usage: <div style="width: 69%;"><div style="width: 69%;"></div></div> 69% of 120 MB



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File Editor

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File Editor

Config Files

- logger.conf
- aoc.conf
- users.conf
- sip.conf
- modules.conf
- musiconhold.conf
- rtp.conf
- indications.conf
- codecs.conf
- features.conf
- cdr_custom.conf
- http.conf
- cdr.conf
- asterisk.conf
- zapata.conf
- manager.conf
- manager/traffic.conf
- manager/ext_globals.conf
- manager/tdm.conf

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File Editor

Create New Config File

New FileName

(Ex: newfile.conf)

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[IP Tables Tests](#)

[SMS Manager Tests](#)

3.1.7 Specials

[AOC Web Update](#)

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AOC Update Service Settings Check AOC File Now

Details

Domain Name or IP Address

Working Directory Name

Periodicity regularly every hour

every day at : o'clock

Repeat Count

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

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

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

Repeat Count : number of repeated files control (used when control is executed with error)

Check AOC File Now: runs immediate control of AOC file. After successful completion, name of currently used AOC file is displayed – see pic below.

GUI-version : 1.0 / Board Type : 2 Logout

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Home Users Trunks Traffic Globals System Diagnostics Specials

AOC Update Service Settings Check AOC File Now

Current AOC File version:
aocstable_orig_v1_1.bt

Details

Domain Name or IP Address

Working Directory Name

Periodicity regularly every hour

every day at : o'clock

Repeat Count

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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 majeure.

Delivery:

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

Contact:

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Account no.: 616243342/0200

SWIFT code: SUBASKBX

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Ing. Pavol Perdek	0903 519 908
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