ASTEROID-1U

The channel bank for Asterisk/Freeswitch

User’s manual

Version 3.0

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Attention! It is prohibited to use this product on communication lines without lightning protectors and extending outside one building.

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1. Introduction

1.1. Designation of Asteroid

Asteroid-1U is designed to be used for connection of telephone lines and GSM channels to Asterisk/Freeswitch. Server connection can be made through Ethernet port using TDMoE protocol. The product is delivered as a chassis for 4 slots and a set of plug-in modules.
The product line includes:

Designation	Number of serviced ports	Description
Asteroid-1UT	32 FXO/FXS, 8 GSM, 4 slots for modules	Modular chassis with controller, power supply, Ethernet.
Asteroid-1UT-EC	32 FXO/FXS, 8 GSM, 4 slots for modules	Modular chassis with controller, power supply, Ethernet and echo canceller module.
MFXO-8	8	Module for connection of external lines
MFXS-8	8	Module for connection of internal lines (telephone sets)
MGSM-2	2	GSM module

Chassis delivered as a set do not have modules, the modules shall be ordered additionally.

1.2. Product flow diagram

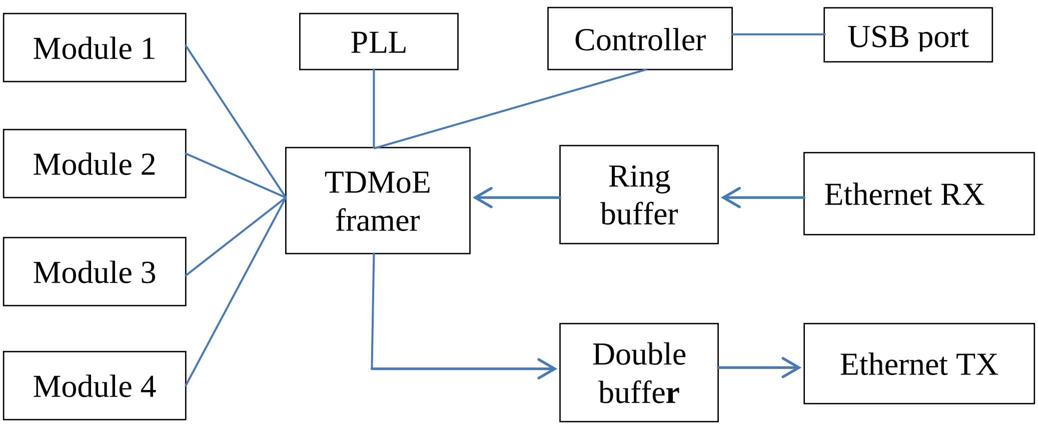


Figure 1. Asteroid-1U chassis flow diagram

Ethernet port is the main data transmission channel. Transferred packages format, TDMoE, is supported with the software DAHDI/Asterisk. The packages received through Ethernet port are transferred to the Ring buffer being half full. The buffer is designed to compensate for packets delay that may occur in the Ethernet network. Further, the packets are sent to TDMoE framer where sound and signal data are extracted. These data are transferred uniformly to the channel modules (Module1..Module4), for instance, FXO or FXS. PLL block is responsible for synchronization of the whole unit. The unit parameters are set through USB port and controller block.

1.3. Requirements to server and LAN environment

The TDMoE protocol is realized directly on Ethernet MAC level. For this reason, routers are not allowed between Asteroid and PC in Ethernet tract. Commutation switches (switches) can only be used.

Special attention shall be paid to the quality of Ethernet packets transfer (QoS) in the network where Asteroid provided. The device may operate in the mode of standard Ring buffer and extended Ring buffer.

The Ring Buffer in a standard mode compensate for packets delay up to 3-4 ms. In practice it means that the device may be connected through 2 – 3 switches. Intense external traffic on the server's port may result in the loss of the packets. Also, the loss of the packets may be caused by Asterisk launch on a virtual machine. At the same time, this mode has an advantage. The introduced delays are small, and echo canceller program may be used on the server not installing the module for hardware-controlled echo canceller.

The extended mode of operation of the ring buffer has been optimized for operation with virtual appliances. The allowed packets delay is 20-30 ms that practically does not limit the number of switches to the device. If this mode is planned to be used, the echo canceller hardware unit shall be installed.

The number of Asteroid devices that may be connected to the server is not limited logically and is defined by the PC operation speed and Asterisk configuration.

2. Technical characteristics

2.1. General parameters

Parameter	Value
Dimensions	430x250x45 mm
Weight	3 kg
Power consumption	50 W
Working temperatures	+5°C to +45°C
Storage and transportation temperatures	-40°C to +70°C
Relative humidity	Up to 80%
Power voltage (AC socket)	220V +- 20%

2.2. Ethernet interface parameters

Parameter	Value
Connector type	RJ45, 8 pins
Cable type	Symmetrical twisted pair (UTP)
Data transmission speed, mbit/s	100
Corresponding standard	IEEE 802.3
Work mode	Duplex, Half duplex, Auto negotiation

2.3. FXS/FXO ports parameters

Coding	PCM A or μ-law
Nominal signal level	0 dBm +- 0.5 dB
Nominal line impedance	600 Ohm
Echoed signal (300..3400 Hz), FXS	Less than -20 dB
Echoed signal (300..3400 Hz), FXO	Less than -12 dB
Gain-frequency variation (respectively to 1 kHz) in range 300..3400 Hz	+- 1 dB
Noise level	Less than -47 dBm
Loop current (FXS)	20 mA
Ring voltage (FXS)	100 V (peak-to-peak), 25 Hz

2.4. GSM ports parameters

Parameter	Value
Frequency range	850/900/1800/1900 MHz
External antenna connector	SMA-Female
Mode annunciation	4 LEDs/channel
Caller ID support	yes
Signaling type on the side of Asterisk	E & M

2.5. Console parameters

Parameter	Value
Standard	USB
Work mode	Serial port emulation
Data speed, kbit/s	38400
Flow control	none

3. Device connection

3.1. Location of operation controls, indicators and plugs

The back panel of Asteroid-1U chassis has:

- 4 slots for modules
- ETHERNET - 8-pin socket for connection to the local network
- ALRM – indicator of no flow TDMoE from the server, not lit under the normal operation
- LINK – indicator of Ethernet line integrity, lit under the normal operation
- CONSOLE - USB socket for console connection, socket type – B
- AC 220V – socket for power supply
- ON/OFF – power on/off button

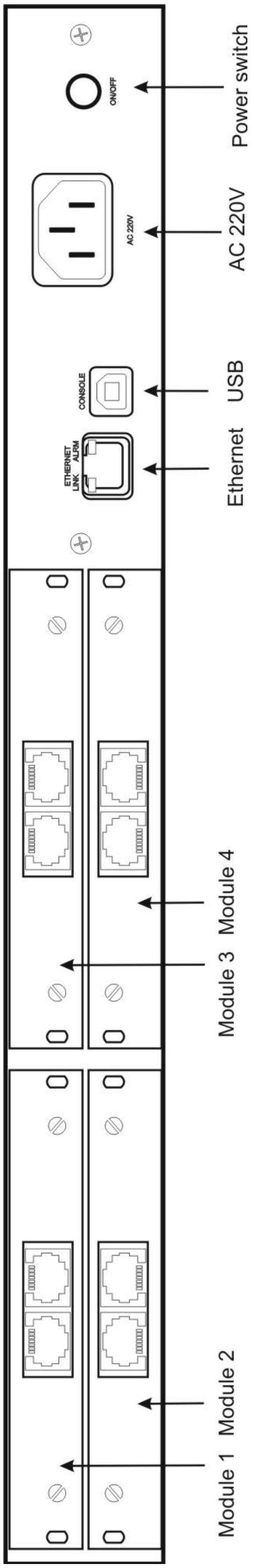


Figure 2. Chassis, back elevation

3.2. Chassis sockets description

Pin	Circuit	Direction
1	TX+	Out
2	TX-	Out
3	RX+	In
4,5		
6	RX-	In
7,8		

Table 1. ETHERNET socket signals

Pin	Signal
1	
2	Data-
3	Data+
4	Common

Table 2. USB console socket signals

3.3. Installation of equipment in 19" rack

Installation of enclosure in the 19" rack (cabinet) shall be carried out according to the Figure. The device chassis shall be supported on the both sides with horizontal rails attached to the vertical support of the cabinet. The device must not be fastened only with mounting angles on the device front panel as it will result to great mechanical stress and will deform the enclosure. The mounting angles on the front panel shall be used only to secure the enclosure in a horizontal plane.

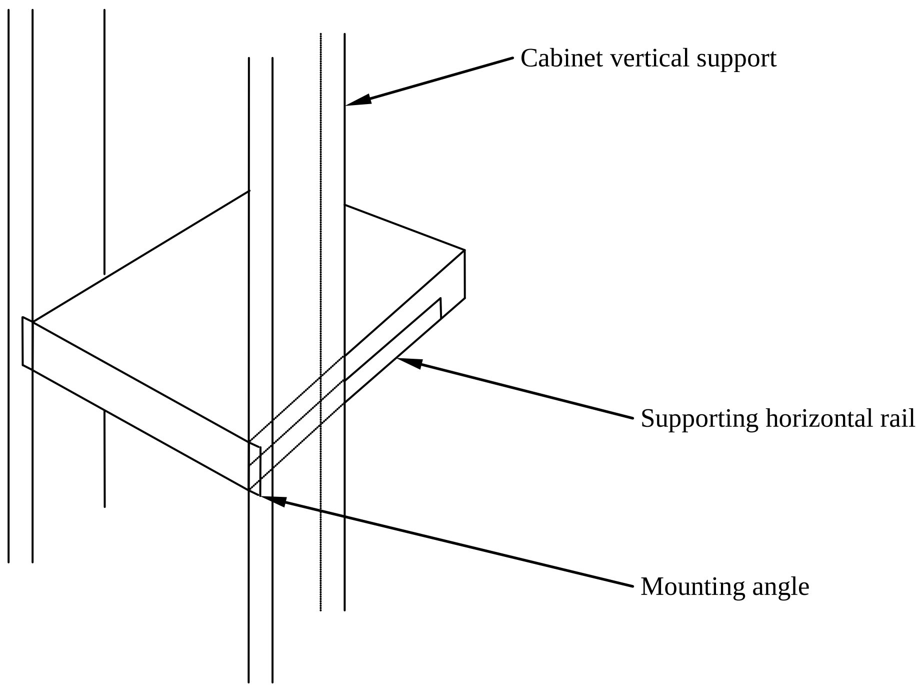


Figure 3. Device assembly in a rack

Supporting rails are the rack accessories and not included in the delivery set.
It is recommended to disconnect interface cables before enclosure installation.

3.4. Power supply and grounding

The device shall be powered from 220V AC mains with a standard three-wire cord with one ground wire. The power cable with the so-called Europlug is included in the set of delivery. **Grounding is obligatory**! Before connecting the device it is necessary to make sure that the socket has grounding pins connected to the facility grounding bus. Otherwise, the device enclosure and interface sockets may be exposed to the potential generated in the power line filter of the power supply. This potential is not dangerous for people but it may cause malfunction of the ports, computer and other units connected to the device.

Besides, failure to ensure grounding causes additional inducing from the device to the power line, and does not provide protective functions in case of a breakdown of the power supply unit.

Prior to connecting the power cable it is necessary to make sure that the power switch on the back panel of the device is “off”.

3.5. Lightning protection

The channel bank ports have galvanic insulation for the voltage up to 1,500V and are protected from static electricity. However, connection of terrestrial (air) lines, extending outside the building borders, to the ports shall be allowed only provided special overvoltage protective devices (PD). PD shall be included in the split between the cable passage and the channel bank port.

Special attention shall be paid to PD grounding and channel bank. Both devices shall be grounded in one point with minimum potential length of grounding buses. If possible, uninterruptable power supply shall be used for channel bank.

3.6. Procedures for extraction and installation of modules

Asteroid is a modular device. The user may change the module configuration only provided the following rules are complied with:

- **Switch off the chassis** using normal switch. 220V cable shall not be disconnected as it is used for device grounding.
- Wait for 30-40 seconds.
- Disconnect all cables connected to the ports of the module to be extracted. Disconnect the console from the chassis USB port.
- Unscrew the retaining bolts and extract the modules.
- When adding the modules, only good ones shall be used that have been received from the manufacturer.
- Relocation of modules from failed/defective chassis **is not allowed** as the module itself may be defective.
- The modules **must not** be installed or extracted if the lines are connected as there may be external voltage on the lines.

Failure to comply with these procedures may cause the device malfunction and cancellation of warranty obligations!

4. Device configuration via USB console

4.1. Connecting console

The console port is connected to the computer USB port with the help of the cable adapter included in the set of delivery. The console operates in the mode of COM port emulation, thus there shall appear an additional COM port in the system after the cable is connected. The device manager may be used for the purpose of control. A device, for example "USB Serial Port (COM3)" will appear in the folder "COM and LPT ports". If there is no COM port, it is necessary to install the driver of the USB device that is located in the folder \\asteroid\USB_console_drv\ on the disk included in the set. The driver is also available for downloading at http://www.ftdichip.com.

Once the COM port appeared, use the terminal program Teraterm (or hyperterm) with parameters 38400, 8b, 1s, np, flow control=off to connect. Remember to re-start the terminal program after switching off/on the channel bank.

4.2. The main configuration menu

After the power is on (or reset) Asteroid prints the main menu to the console and waits for a command. To configure the channel bank parameters , user should walk through the hierarchical menu system and choose the necessary parameters. After parameters modification, settings should be saved in non-volatile memory. There is the dedicated menu point for this.

The upper part of the screen displays the firmware version and the major settings and lines status. The lower part of the screen displays the current menu (see Figure).

```
Asteroid monitor, v1.7 Firmware: Asteroid 32Ch v1.1
Module:
 1    2    3    4
FX0  FXS  FXS  FXS
Link up, TDMoE down, SkipEr=0, SlipEr=7683, RxNuEr= 0, BuffPos = 3, Clk Master
MAC src: 005555555514, dst: 000000000000, Temperature(C): 0, EC Off
                    1 3 5 7   9 1 3 5   7 9 1 3   5 7 9 1
Line state: 1-32:   ........  ........  ........  ........

1. Configuration >>
2. Status >>
3. Test >>
9. Reset
```

Figure 4. The main menu

To choose menu points, use numbers 0-9. Other keys are ignored.

For exit to the upper menu level, press 0.

4.3. General settings

Configuration/Clock source – setting master/slave mode

Clock source	Synchronization mode
Master	Asteroid operates according to its own clock and is the master for Asterisk server.
Slave	Asteroid is the slave and adjusts to the TDMoE flow from the Asterisk server.

Table 3. Master/slave mode

Master mode shall be used if no other equipment is connected to the Asterisk server, except for the channel bank. In this case Asterisk shall synchronize from the bank.
Slave mode shall be chosen if the Asterisk external server already has its reliable source of signaling, for example, adapter E1.

Configuration/MAC – setting MAC address for the device Ethernet port. The user can arbitrarily choose MAC address different from the address set by the manufacturer. However, while choosing the address it is necessary to remember that all TDMoE devices in one network must have unique address.

Configuration/DST MAC – destination address. If this field equals 0, the address is detected automatically. In this case the channel bank responds to the packets of the first server connected via Ethernet. If this field is not empty, the packets are transmitted and received only from or to the defined address. DST MAC field should be defined for point to point connections.

Configuration/Coding law=A – voice data are coded under A law
Configuration/Coding law=m – voice data are coded under m law

Configuration/Echo canceller – enable or disable the module of hardware echo canceller. This requires re-start of the bank channel, the configuration shall be saved and the device shall be turned off/on.

Configuration/FIFO – use the standard or extended mode of operation of the ring buffer. In the standard mode, the delay of packets from the server is allowed within 2 – 3 ms. In the extended mode, the delay may reach up to 20-30 ms. The extended mode is designed for server operation in the virtual environment.

4.4. Saving/Restoring parameters

Configuration/Factory – restoring factory settings (the current settings will be lost)
Configuration/Restore – loading settings from non-volatile memory
Configuration/Save – saving current settings to non-volatile memory

4.5. Testing and diagnostic tools

Test/Enter port to test – choosing the number of FXS/FXO module port to test. The port number may be within the range of 01..32.
Test/Line ring – Turn on or off the ring (for FXS modules).
Test/Line hook – Hook on or off (for FXO modules).

4.6. Device monitoring

The status information on malfunctions in the TDMoE protocol is displayed in heap of screen menu, in the TDMoE line.

Field	Transcript	Comment
SkipEr[1]	Skipped errors	Increases if the device buffer is overloaded, i.e. the packet was received but skipped as the device buffer contains 8 packets. In normal mode the number shall not increase.
SlipEr	Slipped errors	Increases if the device buffer is underrun and if the next packet from the Asterisk server is delayed. In case there is no flow of TDMoE packets, the number increases continuously. In normal mode the number shall not increase.
RxNuEr	Received Numeration Errors	Increases if the number of the received packet does not match the expected number, i.e. the packets are interchanged. In normal mode the number shall not increase.
BuffPos	Buffer fullness indicator	Indicates the current number of packets in the device buffer. Maximum number is 8, minimum number is 1. In normal mode the packets indicator shall be approximately in the middle of the buffer, i.e. it shall indicate approximately 4 packets.
Clk Adj	Clock adjusting indicator	If Asteroid is the master in terms of synchronization, the Clk Adj field indicates the current direction of clock adjustment - < (downwards) or > (upwards).

Table 4. TDMoE protocol status information

Field	Transcript	Comment
B	Busy	Hook off on the FXS port
R	Ringing	Ring from FXO port

Table 5. FXO/FXS line status

Notes:

1. Status is refreshed after pressing the key on the keyboard.
2. Error counters are cleared after choosing menu /Status/Clear.

[1]

5. MFXS/MFXO modules

Attention! FXO/FXS modules do not support hot swapping. Before adding or removing the module, the system power supply shall be switched off, and the port cables shall be disconnected.

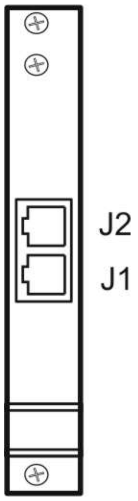


Figure 5. MFXS/MFXO module front panel

The FXO/FXS module front panel houses two connectors J1 and J2 of RJ45 type. Each connector is used to connect 4 telephone pairs (TIP/RING-pair). Signal routing for J1 and J2 is equal and is given in table:

Contact	Signal
1	TIP3
2	RING3
3	TIP1
4	RING0
5	TIP0
6	RING1
7	TIP2
8	RING2

Table 6. FXO/FXS port signals

To connect the telephone pairs, a standard UTP or STP patch cord may be used (see Annex B).

In case of DAHDI configuration, it is necessary to take into account that module 1 matches the telephones pairs connected to minor channels. Module 4 corresponds to senior channels. Inside the module, the minor channels are matched with the pairs connected to J1 connector. Hence, the distribution of DAHDI channels on module connectors may be as follows:

	Module 1	Module 2	Module 3	Module 4
Port J1	1-4	9-12	17-20	25-28
Port J2	5-8	13-16	21-24	29-32

Table 7. Matching of DAHDI channels to module FXS/FXO ports

6. MGSM module

MGSM-2 module is used for installation in the Asteroid channel bank. The module allows connection of two voice GSM channels to the server controlled by the PBX Asterisk software.

Potential areas of module application:

- Gateway VOIP - GSM
- Distributed PBX
- Interactive voice response

Major features:

- Number of GSM channels - 2
- Frequency range 850/900/1800/1900 MHz
- Connector for external antenna – SMA-female
- Indication of work modes – 4 LEDs/channel
- Installation of SIM cards – two slots on the module board
- Support of Caller ID
- Signaling type on the side of Asterisk – E & M

6.1. Module front panel

The Figure below displays the front module of GSM module with antenna connector and work mode indicators. The module uses external antenna that is common for the both channels. Antenna connector type – SMA, female.

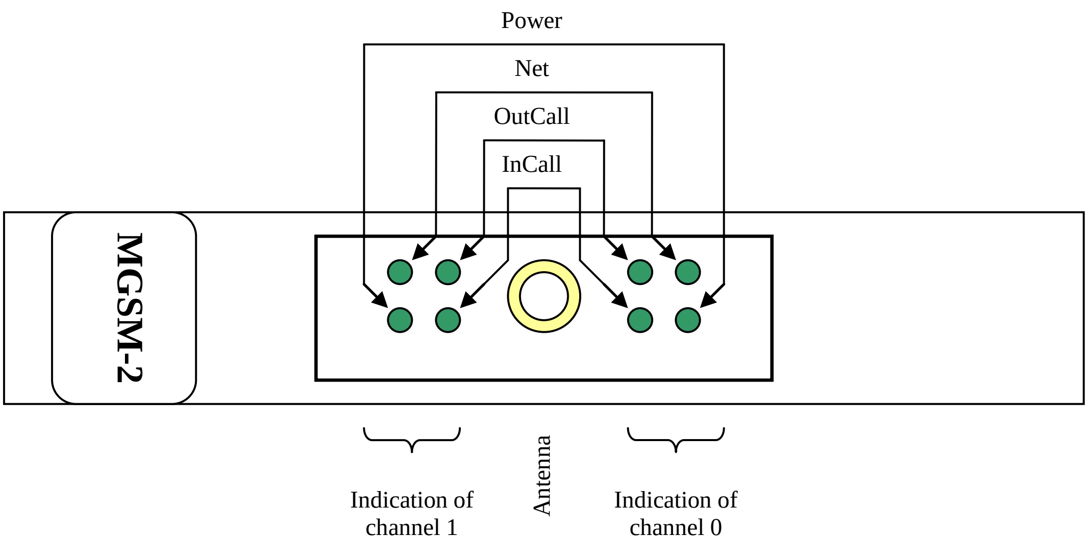


Figure 6. Front panel of GSM module

6.2. Indication

The both channels have the same work mode indicators consisting of four LEDs:

Power – indication of power supply to GSM channel.
Net – indication of GSM network.
InCall – indication of incoming call.
OutCall – indication of outgoing call.

Combination of LEDs may indicate different statuses of GSM channel.

Channel status	**Dislplayed as**	**Comment**
Displaying the GSM controller firmware number	**InCall** & **OutCall** blink simultaneously in 0.5 sec flashes	The number is displayed when module power supply is switched on. The number of flashes represent the firmware number. The option is available starting from number 6.
Channel is blocked because of SIM card	**InCall**, **OutCall** blink alternately in 0.5 sec flashes with a total pause 5 sec	No SIM card, SIM card cannot be read or there is a PIN code.
The channel is searching the base station and attempts to register	Two 0.5 sec flashes **Net** and a 5 sec pause	If the channel stays in this state for long, it means the low signal level or that the SIM card is blocked by the operator
Low level of signal	One 0.5 sec flash **Net** and a 5 sec pause	Connection is possible but the signal level is very low. Change the antenna orientation or its location.
Readiness	**Power**, **Net** lit constantly	The channel has been registered in the operator network, the signal level is good, the channel is ready to receive calls.
Incoming call	**Power, Net, InCall** lit constantly	Call received GSM -> Asterisk
Outgoing call	**Power, Net, OutCall** lit constantly	Call received Asterisk -> GSM

6.3. Module switching on and initialization

After the module is switched, the channel controllers carry out the GSM chip initialization. If initialization is successful, **Power** LED is lit. Then, the network signal level is checked. If the level is less than -93dBm, the controller goes into the waiting cycle and leaves it only when the signal of sufficient level is received, for example, after the external antenna has been removed to the area of better reception. The next stage includes verification of registration in the operator’s network stated in the SIM card firmware. If registration is successful, **Net** LED is lit. Otherwise the registration requests will be repeated periodically. Thus, in the normal mode two lit LEDs – **Net** and **Power** – indicate that the channel is ready for calls.

Please note that if there is no SIM card, the channel controller removes power supply from the GSM chip, and **Power** LED fades. The devices does not support “hot swapping” of the SIM card. Restarting is needed after installation or replacement of the card.

Once the channel is initialized, it may be in one of the three various statuses – call waiting, incoming call processing (GSM -> Asterisk) and outgoing call processing (Asterisk -> GSM).

When the module is switched on, **InCall** and **OutCall** indicators also display the version number of the channel controller software. The number of simultaneous flashes of these indicators corresponds to the number of the firmware version. If there is no flashes – version 0, one flash – version 1 etc. The interval between the flashes is approximately 0,5 sec.

6.4. Signaling and transfer of address information

Address transfer between Asterisk and GSM module is carried out through the E & M protocol. Within this protocol, the module and Asterisk may transfer their states to each other as signals Off-hook (Call start/channel active) and On-hook (Ring off/channel inactive). Address transfer between the parties is ensured with DTMF tones via the sound channel. Inside the module, the channel controller is responsible for analyzing and re-coding the signalization between the GSM chip and Asterisk. For instance, in case of incoming call the Caller ID received from the GSM network is read by the controller from the GSM chip and transformed into DTMF tones that are transferred to Asterisk. 0-9, as well as *, # symbols will be transferred. Other symbols, for instance the international call prefix “+”, will be ignored.

Vice versa, in case of outgoing call, DTMF tones will be accepted by the channel controller from Asterisk and transferred to GSM chip as a dialing command. The number containing symbols 0-9, *, and more than 3 symbols may be dialed.

Besides, the channel controller shall have several reaction algorithms to the state of the called GSM subscriber. Further, the algorithms of incoming and outgoing calls will be discussed in detail.

6.4.1. Incoming call (GSM -> Asterisk)

In the reference mode Asterisk and the channel are in On-hook state. When the call is received from the GSM network, the channel reads the number of the calling subscriber (Caller ID), and then it changes its state to Off-hook on the internal line E&M. When Off-hook state is sent to Asterisk, it is ready to receive the number dialing. The channel starts sending Caller ID after a 300 ms pause as DTMF tones with the times: sending/pause 200 ms. At this time the GSM subscriber is in the waiting mode until Asterisk on the E&M line changes its state to Off-hook indicating the subscriber's response. The signal of the subscriber's response is immediately transferred to the GSM network, and through connection of the sound channel is ensured.

The call may be ended upon the initiative of either party. Upon receiving the disconnection signal from the GSM network, the channel generates On-hook in the E&M internal line resulting in the termination of the call in Asterisk. And vice versa, when receiving On-hook signal from Asterisk, the channel disconnects the GSM subscriber.

If disconnection is required on the part of Asterisk without the subscriber's response, it shall look like a short-term Off-hook (more than 300 ms) with subsequent On-hook.

The Figure illustrates the process of connection establishment.

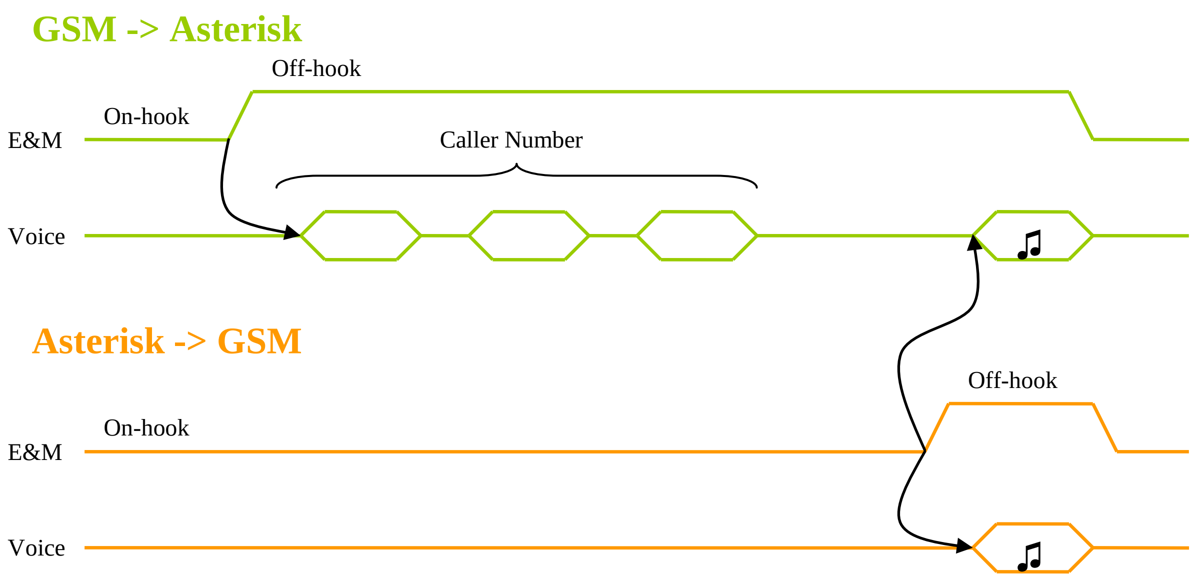


Figure 7. GSM incoming call diagram

6.4.2. Outgoing call (Asterisk -> GSM)

The outgoing call may be realized under three scenarios, each of which takes into account its peculiarities of GSM network operation.

1. Group call
2. Priority call
3. Single call

Single call scenario is set by default.

To make a call according to scenario 1 or 2, prefixes shall be used before the number of the called subscriber.

Single call

Dialing is initiated by the Asterisk server be means of switching the E&M internal line in the Off-hook state. The number digits are sent to the GSM channel as DTMF tones through the sound channel. A 4-second timeout or # symbol are the symptoms of the number end. After the full number has been received, the channel initiates the call through GSM network. Then immediately the GSM channel in E&M line changes the state to Off-hook, that means the subscriber’s response for Asterisk. Thus, the sound channel is enabled, and the Asterisk subscriber hears the call sending signal, audio response of GSM station and called subscriber.

The call may be ended upon the initiative of either party. Upon receiving the disconnection signal from the GSM network, the channel generates On-hook in the E&M internal line resulting in the termination of the call in Asterisk. And vice versa, when receiving On-hook signal from Asterisk, the channel disconnects the GSM subscriber.

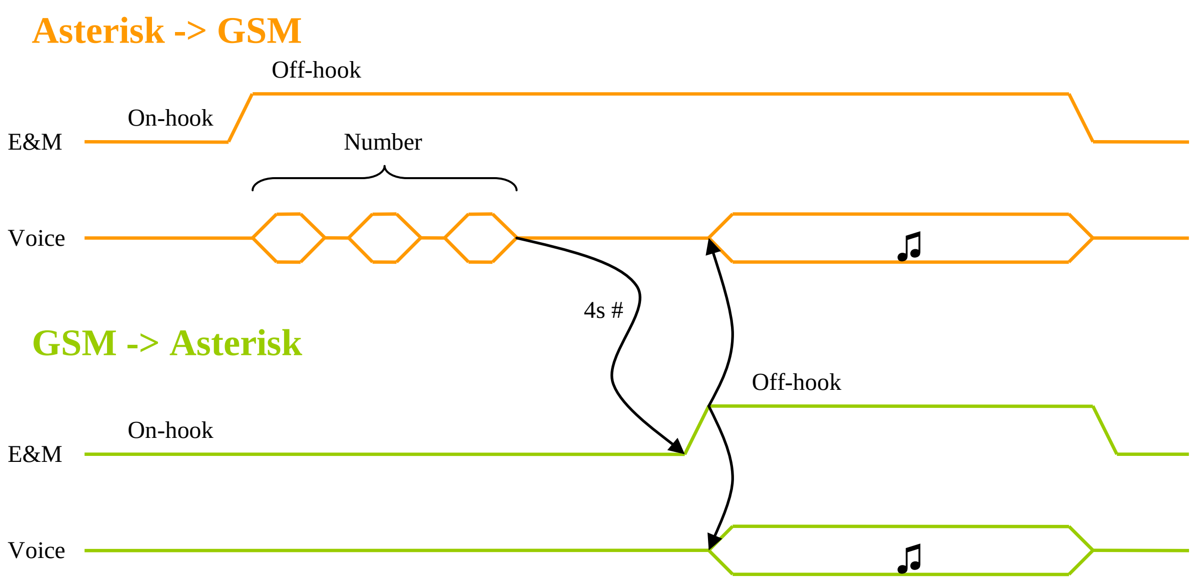


Figure 8. GSM outgoing single call diagram

Singe Call scenario may be used for address call of a single GSM subscriber or during the accompanying transfer of the call.

Group call

To make a call under the Group call scenario, it is necessary to add prefix *1* before the number of the called subscriber. For instance, to dial 1234567, the sequence *1*1234567 with optional # in the end shall be sent to the GSM channel.

Dialing is initiated by the Asterisk server be means of switching the E&M internal line in the Off-hook state. The number digits are sent to the GSM channel as DTMF tones through the sound channel. A 4-second timeout or # symbol are the symptoms of the number end. After the full number has been received, the channel initiates the call through GSM network. From the moment of number transfer to the GSM subscriber's answer the call may be terminated only upon the initiative of Asterisk. Sound notifications from the GSM network "subscriber is not available", "balance is insufficient for making the call", as well as "busy" and "no answer" notification tones are unavailable for the calling subscriber, and subsequent disconnection of the GSM call does not result in the interruption of the Asterisk call, the notification is ignored. The GSM channel changes the state to Off-hook only provided an answer of the GSM subscriber. After the subscriber's answer, the sound channel is enabled, and the conversation is possible.

In the conversation mode the call may be terminated upon the initiative of either party. Upon receiving the disconnection signal from the GSM network, the channel generates On-hook in the E&M internal line resulting in the termination of the call in Asterisk. And vice versa, when receiving On-hook signal from Asterisk, the channel disconnects the GSM subscriber.

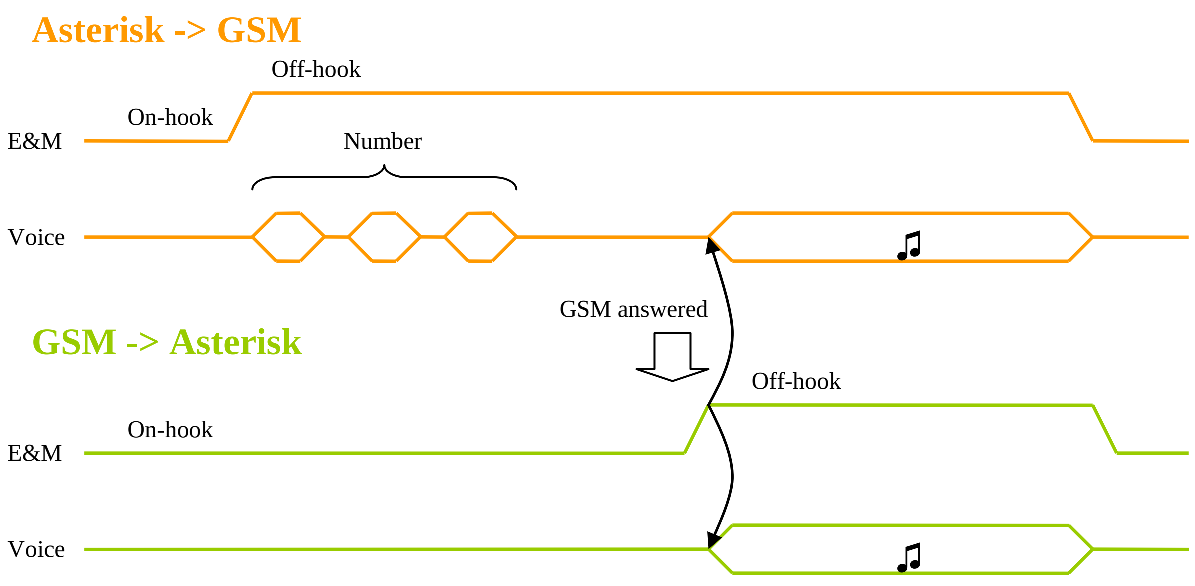


Figure 9. GSM outgoing group call diagram

Group call scenario may be used for calling a group of subscribers when several GSM channels call each member of the group simultaneously. At that, it is necessary that the call last until at least one called subscriber answers.

Priority call

To make a call under the Priority call scenario, it is necessary to add prefix *2* before the number of the called subscriber. For instance, to dial 1234567, the sequence *2*1234567 with optional # in the end shall be sent to the GSM channel.

Dialing is initiated by the Asterisk server be means of switching the E&M internal line in the Off-hook state. The number digits are sent to the GSM channel as DTMF tones through the sound channel. A 4-second timeout or # symbol are the symptoms of the number end. After the full number has been received, the channel initiates the call through GSM network. From that moment the GSM channels waits for the GSM subscriber's response. In case of notifications from the GSM network "busy", "subscriber is not available", "no answer", "balance is insufficient" the short-term Off-hook with subsequent On-hook is sent to Asterisk. The GSM channel changes the state to Off-hook only provided an answer of the GSM subscriber. After the subscriber's answer, the sound channel is enabled, and the conversation is possible.

The call may be terminated upon the initiative of either party. Upon receiving the disconnection signal from the GSM network, the channel generates On-hook in the E&M internal line resulting in the termination of the call in Asterisk. And vice versa, when receiving On-hook signal from Asterisk, the channel disconnects the GSM subscriber.

The connection time chart displays the situation when the GSM subscriber is busy.

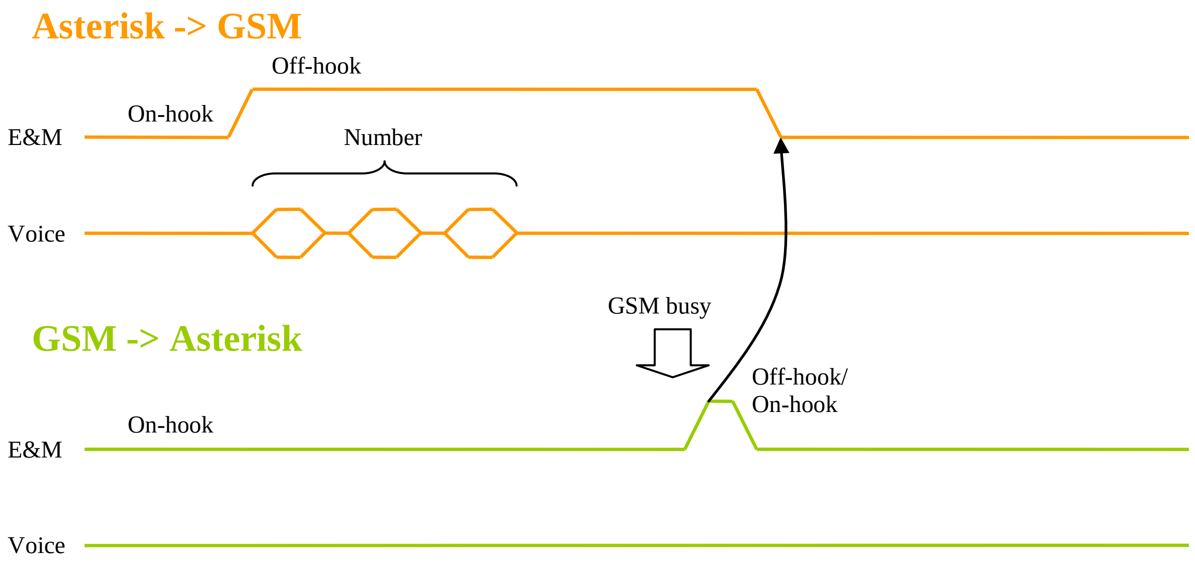


Figure 10. GSM outgoing priority call diagram

Priority call scenario may be used in the situations when several subscribers need to be called (1,2,3) in turns, in priority order. If subscriber 1 is busy, the call goes to subscriber 2, then to subscriber 3. Please, note that the GSM channel does not repeat calls, it shell be defined in the Asterisk connection scenarios.

Scenario / Characteristics	Single call	Group call	Priority call
Prefix before numbers	*3* or no prefix	*1*	*2*
Enabling sound channel	Immediately after dialing	After the GSM subscriber’s response	After the GSM subscriber’s response
Reaction to the following GSM notifications: “busy”, “subscriber is not available”, “no answer” “insufficient funds”	Audible for the Asterisk subscriber, call release follows	Ignored, not audible for Asterisk subscriber	Line release
Transfer of the subscriber’s response to Asterisk	Immediately after dialing	After the GSM subscriber’s response	After the GSM subscriber’s response
Reaction to the GSM subscriber’s response	Ignored, everything is included	The subscriber’s response is transferred to Asterisk, the sound channel is enabled	The subscriber’s response is transferred to Asterisk, the sound channel is enabled
Call termination before receiving GSM subscriber’s response	Possible for either party	Only upon initiative of Asterisk	Possible for either party
Call termination after receiving the answer of the GSM subscriber (conversation)	Possible for either party	Possible for either party	Possible for either party
Typical situations	Single call of the GSM subscriber, accompanying transfer	GSM subscriber’s group call, waiting for the first answer	Alternative call of several subscribers, waiting for the first answer. Direct transfer.

Table 8. Summary table of outgoing call modes

6.5. Configuration of DAHDI/Asterisk for GSM module

6.5.1. Channel numbers

8 channels are assigned to each module, including MGSM-2. As GSM module is two-channel, it takes only two channels in the slot – 4th and 8th. Thus, the following DAHDI channels correspond to the module in the slots:

	Module 1	Module 2	Module 3	Module 4
DAHDI channel	4 8	12 16	20 24	28 32

Table 9. Correspondence DAHDI channels with GSM module ports

6.5.2. DAHDI configuration

To use MGSM module, it is necessary to describe channels in the file system.conf. e&m signaling type is stated for the module. If we assume that the module is installed in the third position of the Asteroid module bank, then system.conf is as follows:

dynamic=eth,eth0/00:55:55:55:55:01/0,32,1
alaw=1-32
echocanceller=oslec,1-32
e&m=17-24

Please, note that in the examples above the signaling is given in total for the whole slot.

6.5.3. Asterisk configuration

The GSM module channels shall be described as necessary in the file chan_dahdi.conf. By the example of the Asteroid module bank, configuration will be as follows:

```
;==== GSM lines ====
signalling=em
context=from-gsm
echocancel = no
echocancelwhenbridged = no
relaxdtmf=no
callerid=
txgain=-3.0
channel=20
channel=24
```

Besides signaling, it is also possible to adjust the signal levels (rxgain, txgain); enable echo canceller (echocancel); enable the parameter relaxdtmf for better identification of DTMF tones.

The scenario for processing the call from the number, that no scenario has been established for, shall be written in the file extensions.conf:

```
exten => _.,1,Answer(10)
exten => _.,n,NoOp("CID unknown: ${EXTEN}")
exten => _.,n,Hangup()
```

7. Hardware echo canceller

The echo canceller module (EC) eliminates reflected signal produced by FXS/FXO ports (see Figure).

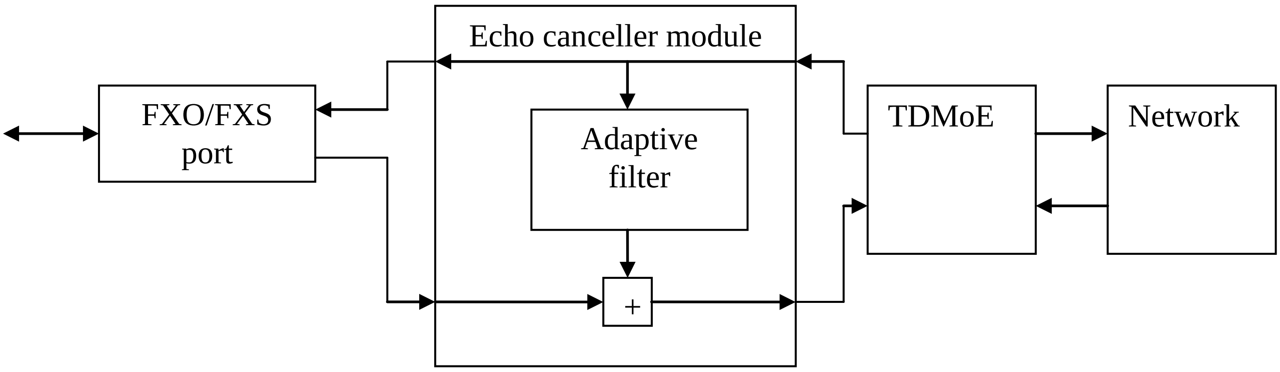


Figure 11. Echo canceller module structure

The echo effect is produced by the reason of not ideal analogue telephone line and significant signal delay from one subscriber to another (more than 30 ms). EC eliminates reflected signal from the signal "line -> network" and passes the signal "network -> line" unchanged. So, remote (network) subscriber does not hear its own reflected signal.

EC module characteristics:

- Linear echo components suppressed at the level of -30..-40 db
- Echo tail is 256 ms
- Automatic EC disconnection in case of modem and fax detection

EC module shall be enabled through the device console in compliance with cl. 4.3.

The use of Asteroid modules with embedded hardware echo canceller notably reduces Asterisk server computation expense. The software echo cancellers in Asterisk shall be stopped by changing the following parameters in the file **/etc/asterisk/chan_dahdi.conf**:

```
echocancel=no
echocancelwhenbridged=no
echotraining=no
```

8. Connection to Asterisk

The Asterisk server setting manual is applicable to all devices operating under the TDMoE protocol. By this reason, the manual is drawn up as a separate document “User manual for DAHDI/Asterisk setting” available for downloading at
http://parabel.ru/d/manuals/dahdi/tdmox-ru.pdf

Summary table of correspondence between DAHDI driver channels and Asteroid-1U ports is given below.

Module type	Module number (lot)	Module port	DAHDI channels	DAHDI protocol	Asterisk protocol
MFXS-8	1	J1	1-4	fxols	fxo_ls
		J2	5-8		
	2	J1	9-12		
		J2	13-16		
	3	J1	17-20		
		J2	21-24		
	4	J1	25-28		
		J2	29-32		
MFXO-8	1	J1	1-4	fxsls	fxs_ls
		J2	5-8		
	2	J1	9-12		
		J2	13-16		
	3	J1	17-20		
		J2	21-24		
	4	J1	25-28		
		J2	29-32		
MGSM-2	1	Sim0	4	e&m	em
		Sim1	8		
	2	Sim0	12		
		Sim1	16		
	3	Sim0	20		
		Sim1	24		
	4	Sim0	28		
		Sim1	32		

Table 10. Correspondence between DAHDI channels and Asteroid ports (summary)

Explanations to the table.
“Module number” means the position of the module in the chassis, see Back Panel Figure
“Module port” means the number of channel (port) inside the module
“DAHDI channels” mean the channel number in DAHDI driver corresponding to this hardware port
“DAHDI protocol” means the protocol to be stated in the DAHDI configuration for this module
“Asterisk Protocol” shall mean the protocol to be stated in the file chan_dahdi.conf for this module (or in web interface)

9. Point-to-point connection

Asteroid may be used independently without connecting to the Asterisk external server. The couple of devices connected through the Ethernet port form the system of subscriber multiplexing.

For such connections, it is necessary to use channel banks with complimentary configuration. FXS slot on one side shall correspond to FXO slot on the other side. In this case Asteroid will ensure transparency of signaling. Thus, off-hook on the side of FXS will result to off-hook on the side of FXO. In other direction the call on the FXO side will result to the call on FXS.

Synchronization in the channel banks shall be set complementary according to the scheme “master-slave”. MAC address of one Asteroid shall correspond to DST MAC – address of another Asteroid. Herewith, the MAC and DST MAC fields shall not be equal to zero.

The network infrastructure shall ensure packet delay not more than 2-3 ms. Thus, the point-to-point connection via Ethernet may be used within the local network or by optical communication lines. The following parameters shall be set.

Parameter	Asteroid #1	Asteroid #2
Configuration/Clock Source	Asteroid master	Asteroid slave
Configuration/MAC	00:55:55:55:55:01	00:55:55:55:55:02
Configuration/DST MAC	00:55:55:55:55:02	00:55:55:55:55:01

10. Warranty provisions

The warranty covers all defects of manufacture or accessories within the warranty period. The warranty requirements shall be sent to the manufacturer enclosed with a written description of the defect, circumstances of arising, as well as the copy of the device passport.

The manufacture undertakes to repair or replace the device free of charge within the warranty period, except for the following cases:

a) if the device has the signs of extraneous interference or there was an attempt of unauthorized repair (soldering, unoriginal components, firmware of micro programs provided by third parties);
b) mechanical damages – splits, cracks, dents, path destruction in the result of impacts and falls;
c) damages caused by ingression of foreign objects, liquids, insects, animals inside the device;
d) operation of the device with direct connection to air (surface) communication lines without lighting protection;
e) violation of the procedure for extraction and installation of the modules enlisted in the user manual;

The warranty period is 2 years from the selling date stated in the device passport.

11. Delivery set

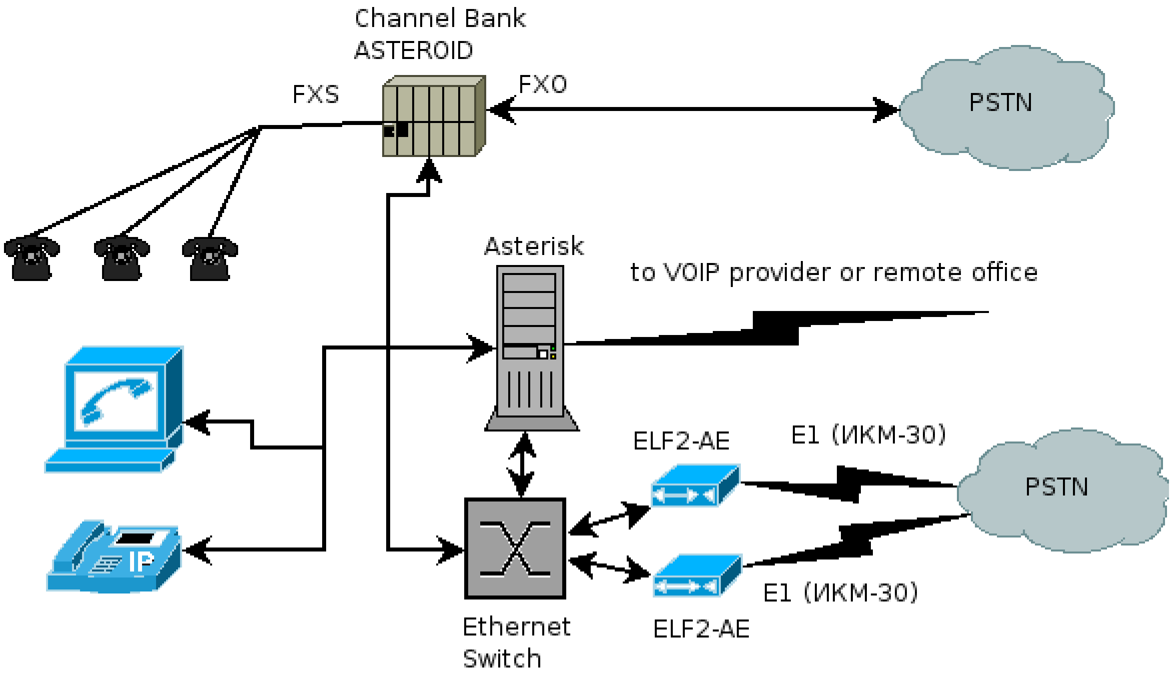
The device is shipped with the following accesories:

- ASTEROID – 1 pc.
- CD (DVD) with documentation – 1 pc.
- Power cable (Europlug) – 1 pc.
- USB console cable – 1 pc.

Cables for connecting FXO/FXS, GSM antenna are not included in the delivery set and are delivered seprately.

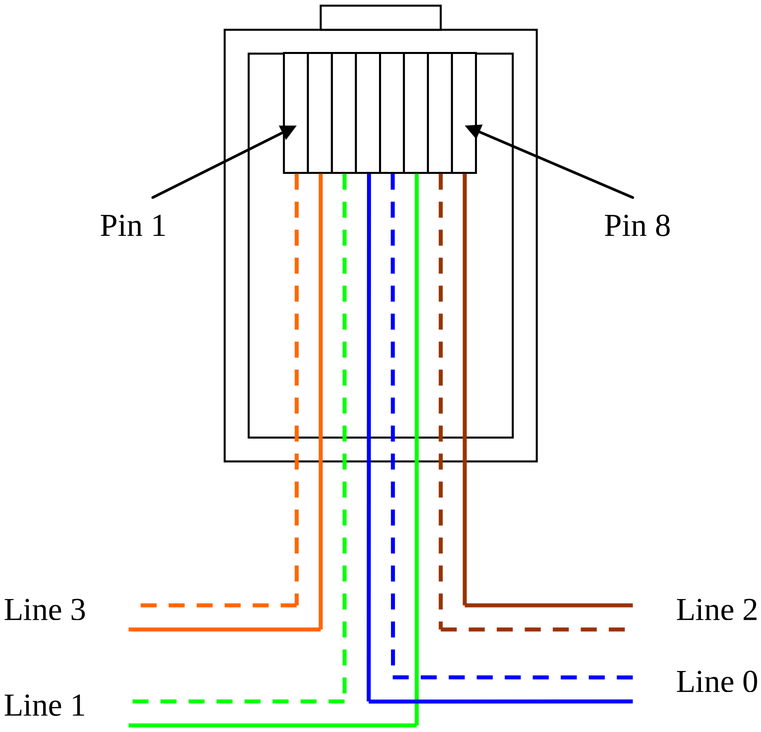
Appendix A. Application schemes

Corporate PBX with VOIP connection



In this scheme the office can be connected to the IP telephony provider or corporate VOIP network. It allows to cut the expenses on telephony.
The following dial plan can be supposed. Local consumers make international calls. **Asterisk** software receives such call, analyzes it and determines that the call is international, and then forwards it through the IP telephony provider network. If the VoIP network is unavailable, it sends the call through the public telephone network.

Appendix B. Example of cable termination for connection of Asteroid telephone ports.



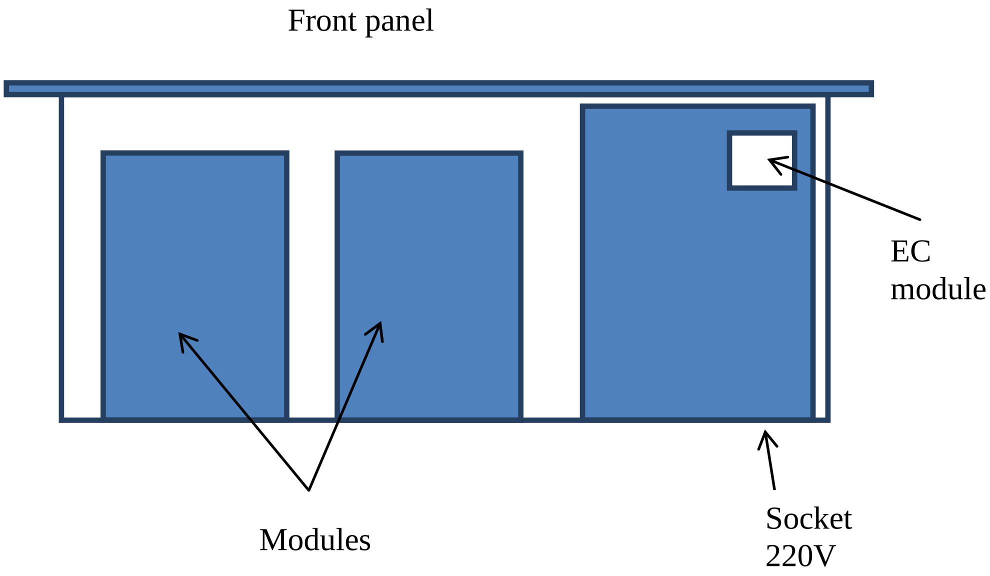
Note

1. Standard patch cord for local networks may be used for connection.
2. View on the connector RJ45 from the opposite side of the latch.

Appendix C. Echo canceller module installation

To install the echo canceller module:

1. Switch off the chassis
2. Remove the upper cover undoing 7 screws
3. Install the module in the slot on the main printed circuit board (see Figure)
4. Examine visually and make sure that the module is installed in the slot without any track and support displacement
5. Install the upper cover again
6. Turn on power, and switch on the module through the console menu
7. Store configuration
8. Turn off and on power again (re-start)



Appendix D. Frequently asked questions and answers

Q. I cannot connect the USB console
A. USB interface is realized on the chip of FTDI - FT232 company. First of all, make sure that the computer "sees" it in the list of USB devices.

In linux command lsusb:
#lsusb
Bus 004 Device 002: ID 0403:6001 Future Technology Devices International, Ltd FT232 USB-Serial (UART) IC

In Windows in the device manager, in the section COM and LPT Ports *USB Serial Port (Com N)* shall appear

To make sure that it is exactly Asteroid, disconnect the USB cable – the device shall disappear from the list.

If there is no such device in the list, try to change the cable or use another USB port.

Further, the terminal program may be set. This program shall display on the screen the data transfer between the computer and the device through the USB bus.
In Linux minicom is usually used, in Windows - Putty, Teraterm, Hyperterm etc.
The device that appears when the cable is connected shall be set as a port for the terminal program. In Windows it is Com N (N – the port number dynamically assigned by the system). In linux the port hand shall appear in the special device files:

*#ls /dev/ttyUSB**
/dev/ttyUSB0

This file shall be set for minicom as a Serial Device.
Other port parameters shall be set incompliance with the user manual.

Finally, if the terminal program screen is still empty after the settings have been made, press the space bar. The thing is that the device waits for the reaction from the user and displays menu only after the key is pressed.

Q. The settings made through the USB port are regularly reset.
A. There may be two reasons.
The first reason. All parameters on the screen menu are changed before the first re-start of the device. When the device is-restarted, they are newly loaded from non-volatile memory. To save the parameters, it is necessary to save the configuration in the respective menu.
The second reason. If the parameters were saved, and the settings are nonetheless reset, it is possible that the program minicom "is guilty". It is designed to be used with modems, thus it has the so called modem initialization string in the settings – Init string. It is a set of symbols that is sent through the port to the modem when the program is started. As in our case we connect not the modem but the device with screen menu, the initialization line is a random set of commands for it, and in the result the device chaotically "runs" across the menu and resets the settings. It is necessary to reset to zero the parameters of minicom program – Init string and Reset string in the menu Modem and dialing.

Q. The telephone is connected to FXS port, there is no buzzer

A. The reason is that the device is not the VOIP gateway, it is the channel bank. The channel bank ensures only hardware frame for the port. The main scope of functions on audio processing and signaling is performed by IP PBX – Asterisk in our case. Thus, the buzzer will appear only after DAHDI setting (driver level) and Asterisk itself.

Q. Asterisk and DAHDI are set, there is a connection with Asteroid, but no calls go through, and no buzzer

A. Perhaps, the type FXO/FXS is confused in the configuration. In the DAHDI/Asterisk system the signaling type of the connected party (partner) is specified. Thus, for example, for FXO port fxs_ls signaling is defined.

Q. There is a lot of ground noise on the telephones, the conversation on the next line is surveilled

A. Most probably, the pair at the FXS slot is connected improperly. For instance, one wire in the pair is taken from one port, and the other wire – from another port. In this case the line won’t function, as all the ports are powered from one source, and there is electric connection between the ports. However, the sound quality will be diminished.

Q. Clicks are heard during the conversation, fax sessions are interrupted.

A. It is the sign of incorrect synchronization of Asterisk – Asteroid system.
Objectively, it may be determined by the increasing numbers on the counts SlipErr, SkipErr in the device console.
If the system is set up correctly, the numbers shall not increase at all.
Variants of incorrect synchronization setup are given below.

Wrong	Right
Two masters in the system. Asterisk is set to operate according to internal clock, Asteroid – master.	Asteroid – master, Asterisk synchronizes according to it
Asterisk is set to operate according to internal clock, Asteroid – slave.	Computer clock is not accurate, so it is better to synchronize from the specialized device – E1 card or Asteroid. Correct variant: Asteroid – master, Asterisk synchronizes according to it.
E1 card is set in master mode. Asterisk synchronizes according to Asteroid. Asteroid – master.	As a rule, E1 cards may be synchronized only from E1 line and may not operate under the clock imposed by Asterisk. Thus, in any case it is necessary to choose E1 card as the source of synchronization for Asterisk, and to synchronize Asteroid based on Asterisk. Asteroid-slave.

TODO.

1. Extended FIFO – number of packets