TOPEX Voxi Family

VoiBridge / VoxiPlus

O.A.M. Manual

(Operating, Administration and Maintenance)

2008

About TOPEX!

TOPEX is a group of Romanian companies, founded in 1990, by 10 enthusiastic engineers experienced in telecommunications. Its activity is directed to the research, development and production of telecom equipment as well as service for them.

TOPEX becomes very quick the most important supplier of IT&C solutions for small to large companies as well as for telecommunications operators and providers in Romania. The company designs equipment for all existing mobile systems (GSM, CDMA), including 3G technology.

TOPEX is reperesented all over Romania by a wide network of local distributors through which the promotion, administration and product maintenance are running.

Due to our innovation power, authentic flexibility, real respect for our parteners and secure solutions that we provide TOPEX extended its business worldwide. Currently TOPEX delivers its products through its distributors to: Republic of Moldova, Bulgaria, Greece, Spain, France, Nigeria, Russia, UK, Turcia, Olanda, etc.

In order to achieve effective and flawless manufacturing for its products, TOPEX has carefully organized its Research and Development Department along with its production facility. This allows TOPEX to have maximum control of all the processes involved in the complex operations related to high-technology electronic manufacturing. At the present time, the Research and Development Department counts 30 specialists and the trend is ascending.

TOPEX's also considered the training and the service as part of the solutions it provides. Therefore, comprehensive trainings are organized at Topex Factory, complimentary for the company's clients. Service is also provided via internet, as all Topex solutions are designed especially to allow this, at the lowest cost.

TOPEX has implemented the quality management system according ISO9001 standard certified by SRAC since 1997, respectively by IQNET since 2002. TOPEX become a sector member of I.T.U. (International Telecommunication Union) since 2001.

For more details please visit www.topex.ro

The main lines of products of our company are:

- Fixed-Mobile Terminals (interfaces from GSM/GPRS/EDGE/UMTS to analog, ISDN BRI and PRI, or VoIP)
- Media Gateways for TDM, VoIP, GSM and CDMA (2G,3G),
- softswitches and other Next Generation Network solutions
- Signaling Converters and Protocol Translators (SS7, SIP, H323)
- Broadband Wi-Fi Mobile Routers
- VoIP gateways (miniature, GSM/CDMA, advanced)
- ATC Voice Communication Systems (Custom systems for special applications such as (air traffic control or railway dispatching center)

For further details please visit our website at: www.topex.ro

WEEE Directive Compliance

WEEE Directive

This symbol applied on your product or on its packaging means that this product fulfils the WEEE Directive. The product shall not be recycled as household waste; it will be disposed separately as sorted waste (electric and electronic equipments).
Regarding to WEEE Directive the recycling EE equipments must be accomplish separately in purpose of natural resources preserving and to avoid the occurring negative effects about human health and environment. The acquired product shall not be treated like household waste at the end of its life and will be returned to TOPEX Company at the address: ROMANIA, Bucharest, Feleacu Street no 10, code 014186 or sent to a specialized processing company.

! Please do not dispose your TOPEX product as unsorted waste (household waste), recycle it to protect the environment. Separate the packages according to waste disposal options and sort it for recycling.

For supplementary information contact us to:
Phone: +4021 408.39.00 or www.topex.ro

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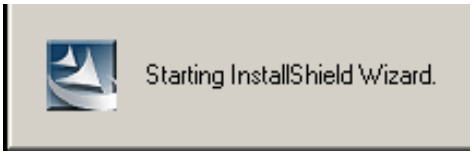
OAM manual is revision B / june 2008 / software version v5.1.12.

1 QUICK INSTALLATION USING O.A.M.

1. Install the OAM application comprises on CD deliveredc by TOPEX
2. Add VoiBridge equipment on position (Systems→ Add)
3. Set the own IP address
4. Set the network parameters
5. Verify if the connection is working (SSH, serial)
6. Download the Default configuration (“Actions” option, password “topex”)
7. Configure and install the cards by means OAM
   - assigned the psychical position of different card
   - assigned “Direction name”
7. Verify line status (installed lines / uninstalled lines)
8. Directions setting (parameters: DIR IP IN settings / DIR IP OUT settings)
9. Define the routing table for calls:
   PREFIX / PORT / DIR / SERV / HUNT / DIR IP / LCR
   Call Destination

2 OAM Starting

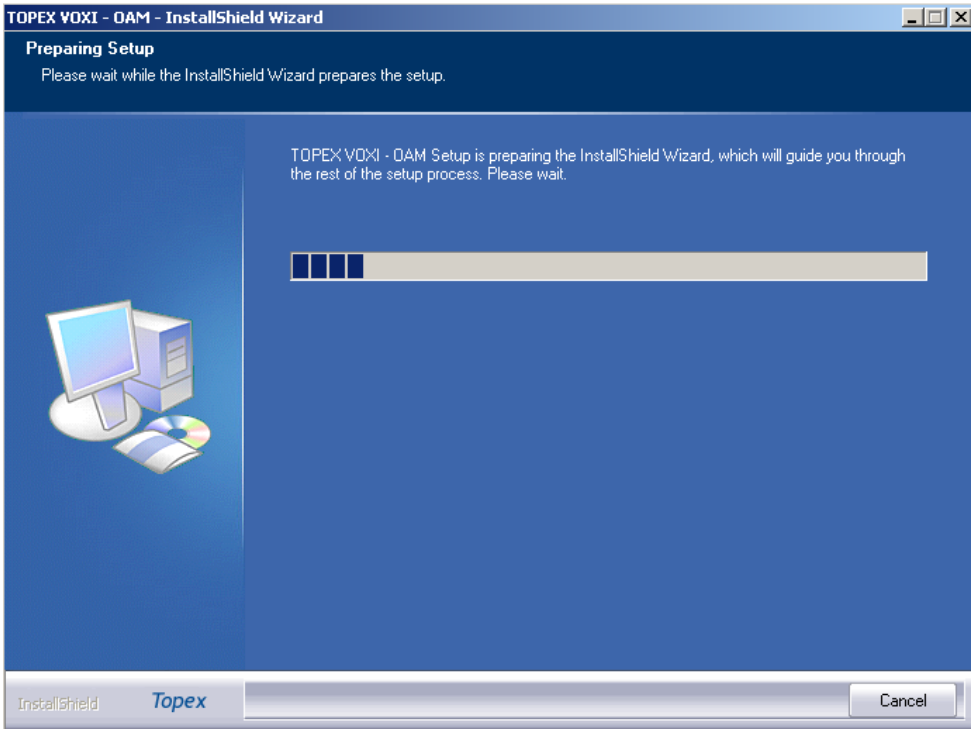
Insert the Topex CD-ROM into the drive. If the Autorun feature for the `CD-ROM enabled on your PC, the installation of OAM should begin by itself, since the program features auto-run:



The first installation screen shows: the Topex logo, the name of the program being installed and the equipment range for which the program is intended:

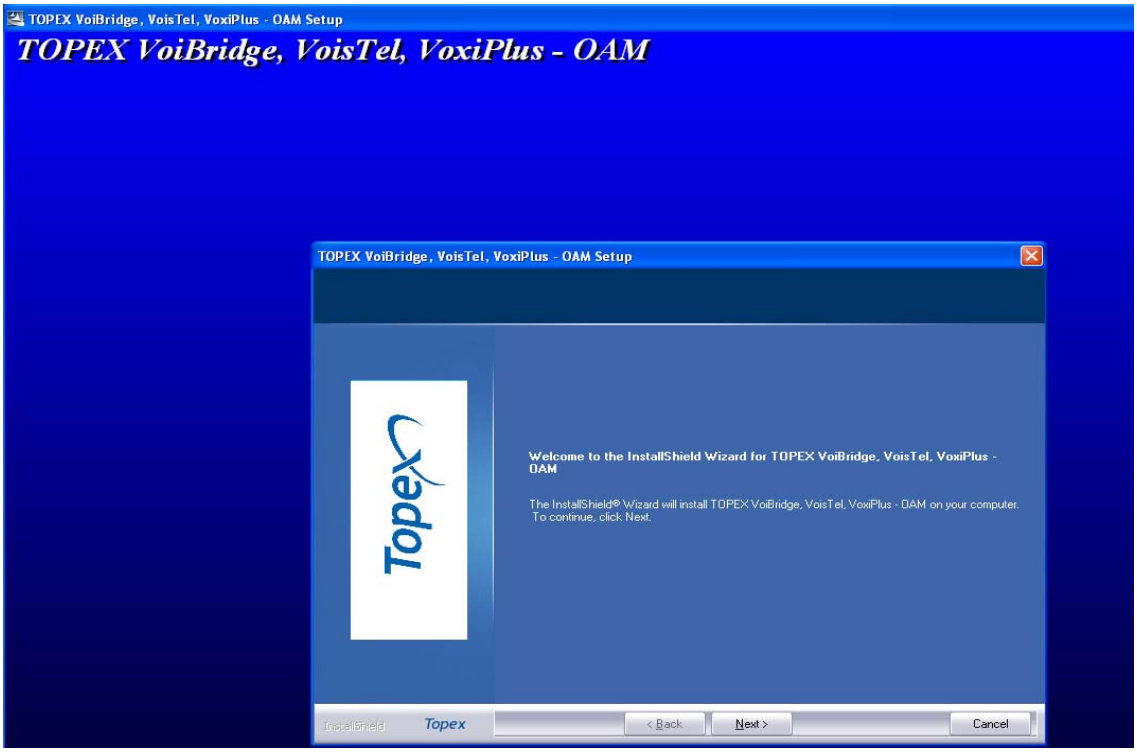
! *If the displayed equipment range is different means that the OAM CD is wrong and must ask TOPEX about proper OAM CD.*

Then a "Preparing Setup" screen is displayed, mentioning the name of the OAM program to be installed, such as "Topex VOXI –OAM" and showing a progress indicator for the Setup program. If you notice it is the wrong type of program, you can click the "Cancel" button at the bottom right corner to abort the installation.

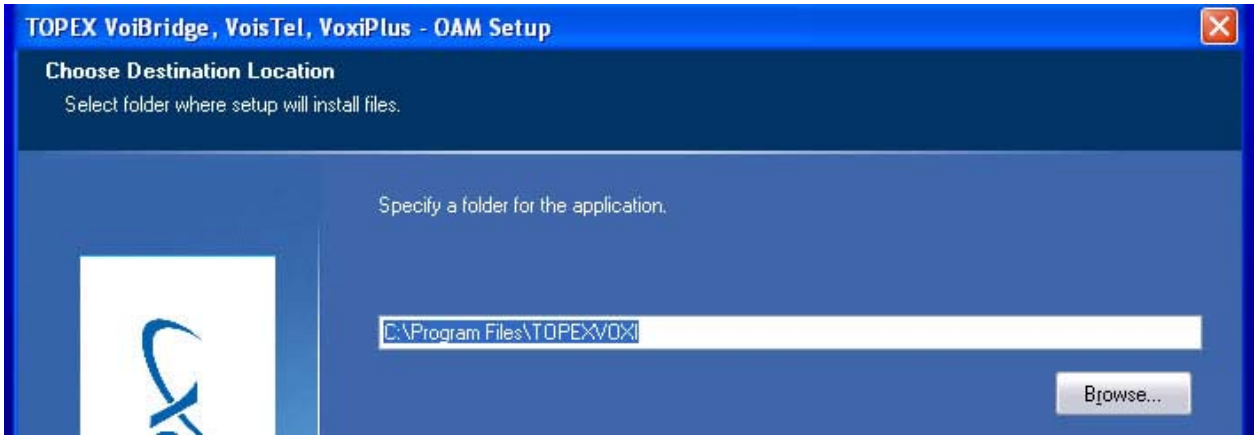


If the program is the right one, you just wait for the OAM program to be installed on your computer.

When the Welcome screen appears, mentioning the application name, just click Next to continue.



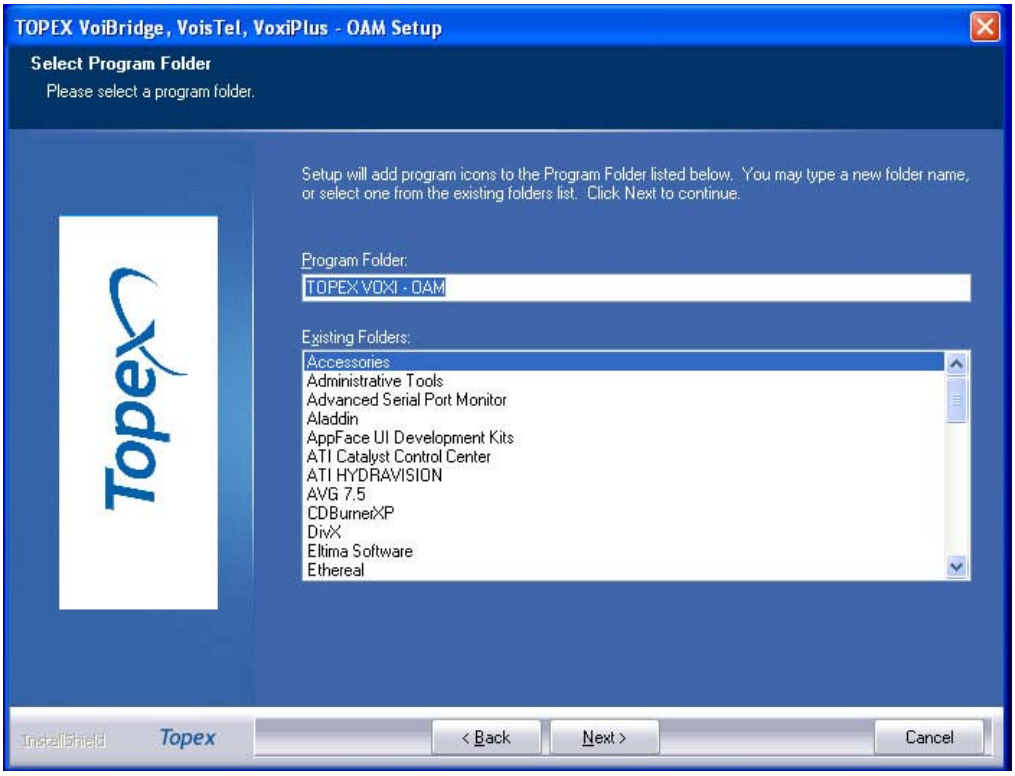
Now the window "Choose Destination Location" appears, asking you to specify the folder where the "gwconfig" application will be installed.



The default value is a folder named "TOPEXVOXI" in the "Program Files" system folder. Select "Next" to continue or "Browse" if you want to specify another folder for the installation. Usually, you do not need to change the default location on C: drive.

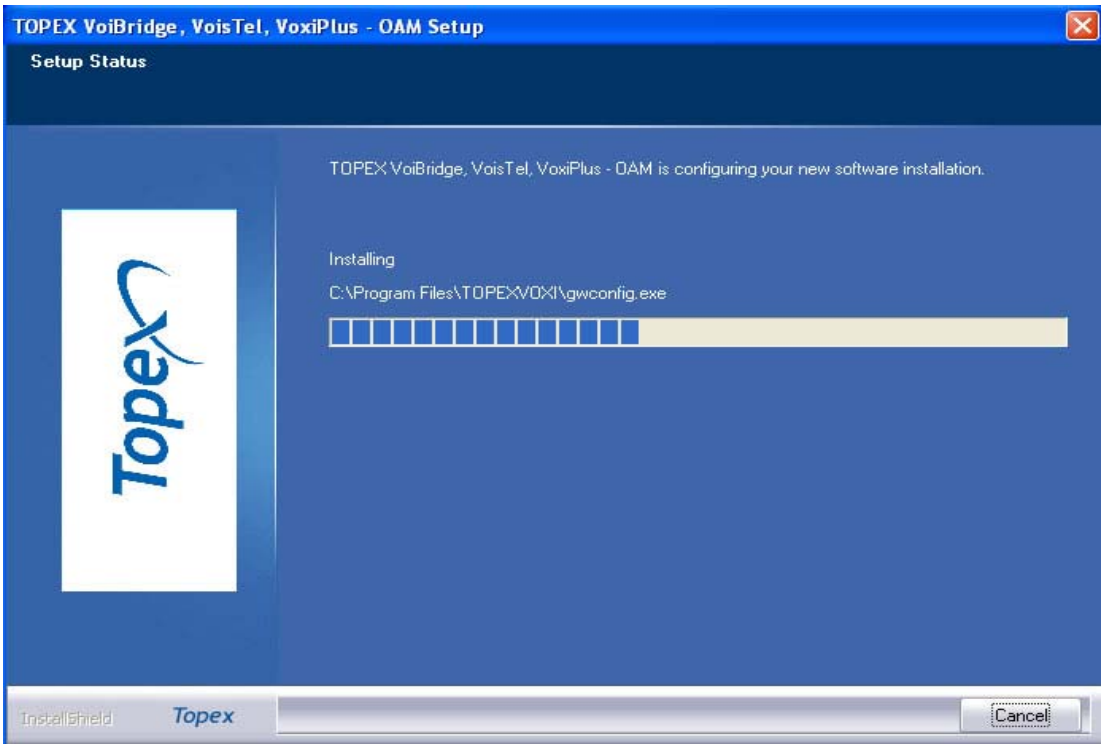
You may change the installation location to another drive and directory, according to your needs.

The "Select Program Folder" window comes next. Here you must specify the name of the Programs folder were the startup icon(s) for the OAM program will be installed. By default, the setup wizard will create in Programs a folder called "TOPEX VOXI-OAM". Again, you should not change this factory established value.

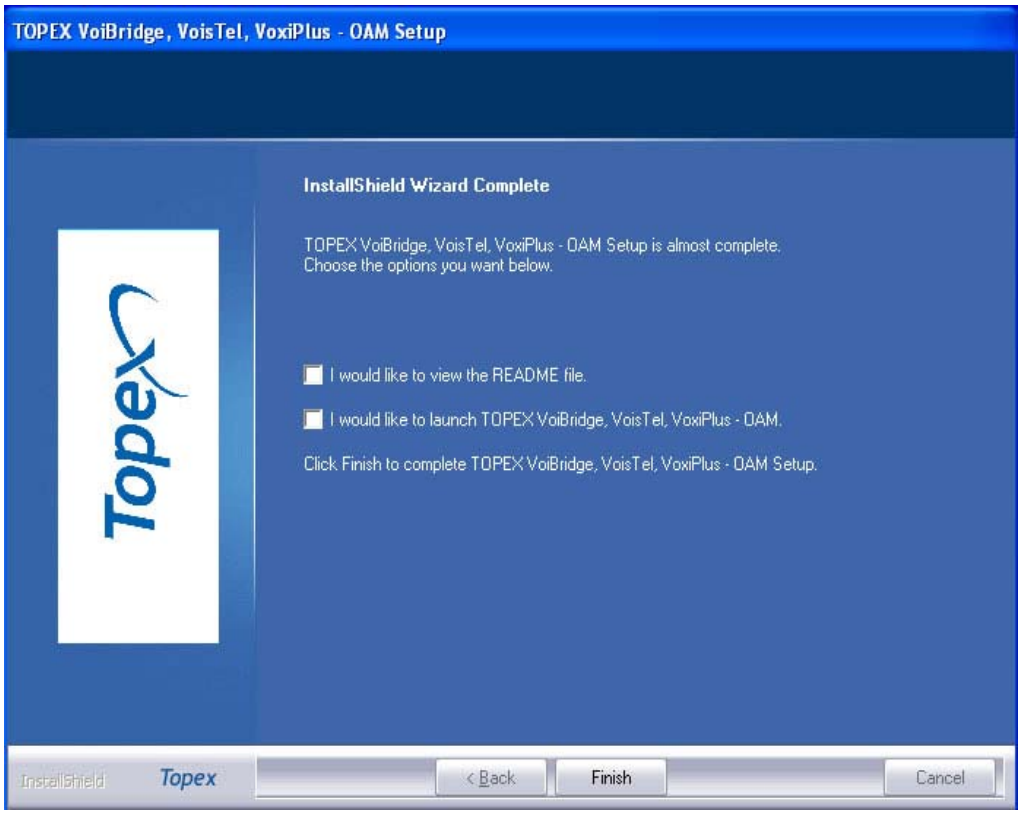


No matter if you have or not a dedicated folder for the OAM icons, you can later make shortcuts for them into your Desktop or in the Quick Start bar.

Click Next and the effective installation begins, you will see the message “Installing” and a progress bar that shows the progress of the installation:



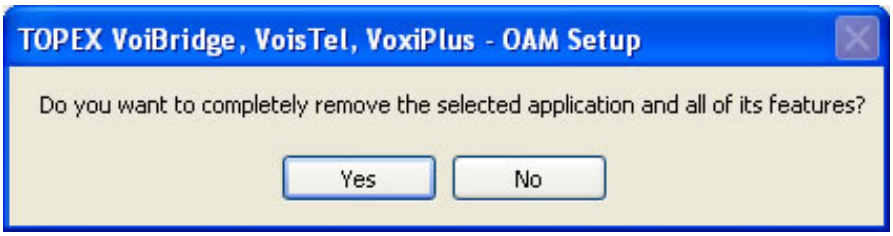
- When the setup is successfully ended, the window "Setup Complete" will show up. It asks you if you want to see the README file and if you want to launch the OAM program immediately.
- To end the installation, click the button "Finish".



- Now the OAM program is fully installed on your system.
- To launch it, go to the Start menu, select "Programs" and then click "TOPEX VOXI - OAM".

Note 1: If the **"autorun"** feature for CD's is not enabled on your computer, then you must launch the Setup program manually. Go to **"My Computer"** and click on your CD drive. You will see the contents of the Topex installation disk. Double click the "**Setup"** icon to launch the installation program.

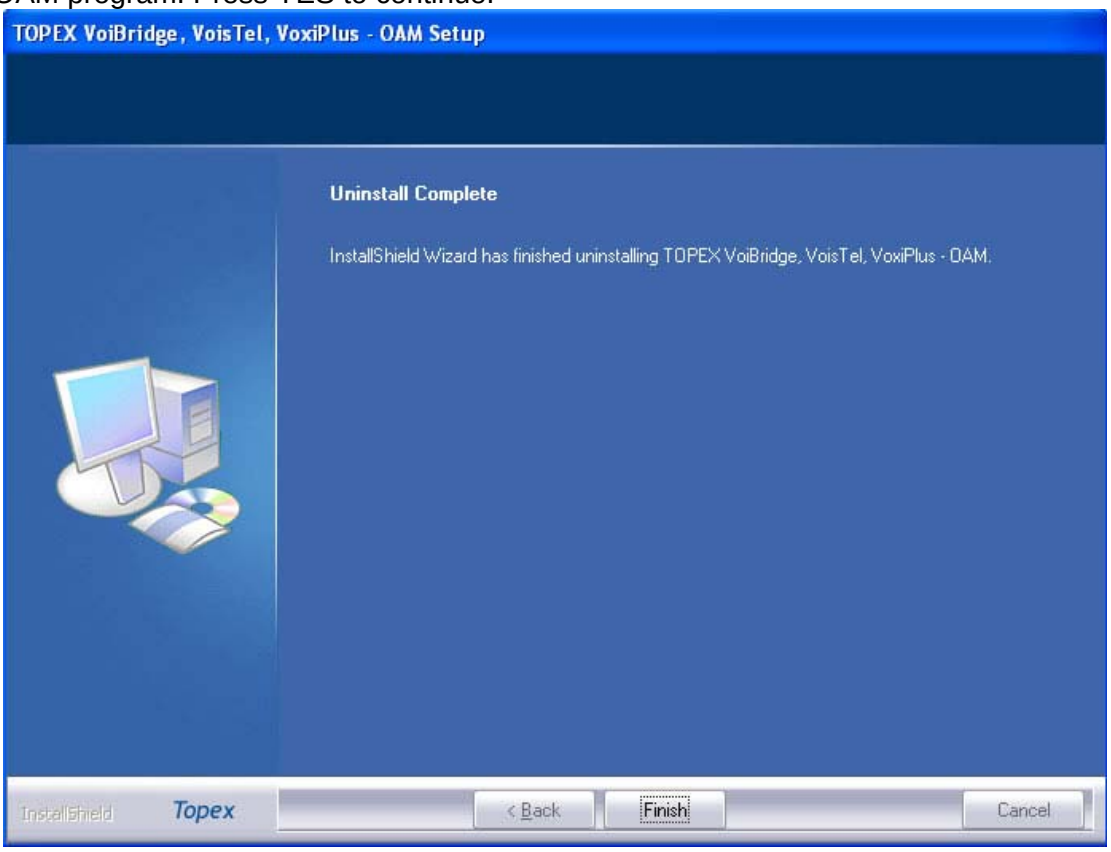
Note 2: You may want to remove the OAM program from your computer.
- From the Start menu, go to "Settings" and select "Control Panel".
- From Control Panel, double click the icon "Add or Remove Programs". Search in the list of currently installed program the application "Topex VOXI - OAM" and click the "Change/Remove" button if you want to uninstall the program.



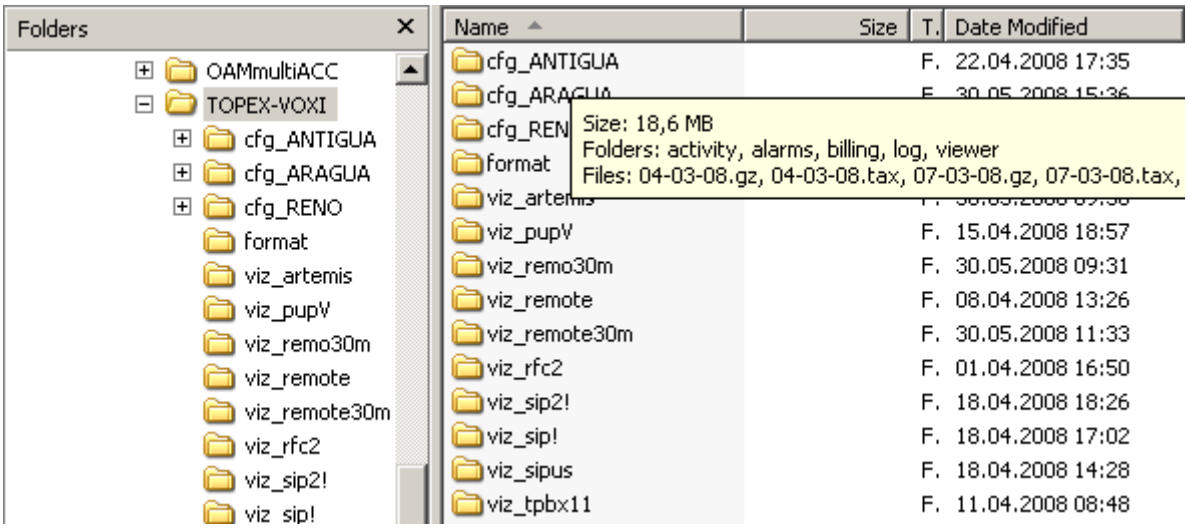
- A window called "Confirm File Deletion" shows up, asking you to confirm that you want to remove the OAM program. Press YES to continue.

Now the un-installation of the program begins.

You will see onscreen a progress indicator for the removing of the OAM program.



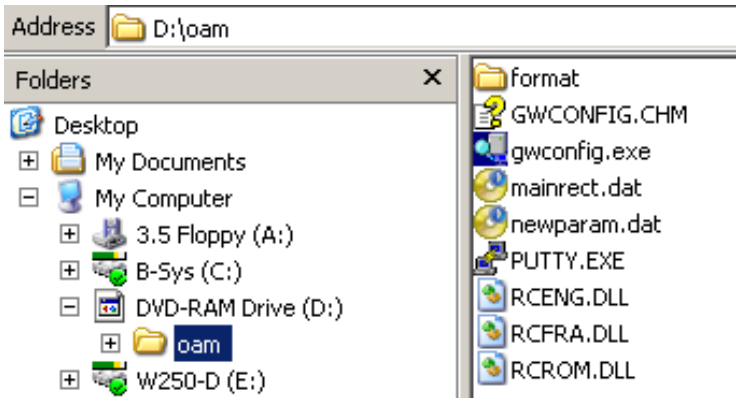
The OAM program is removed from your computer, but the TOPEX folder with all its sub-folders **won't be deleted**. Consequently, if you had different configuration, billing, alarms or monitoring files in this folder, the files will be kept.



Thus, when you re-install the OAM program over a previous installation, the configuration and log files won't be overwritten, so you may keep your settings, activity and billing records.

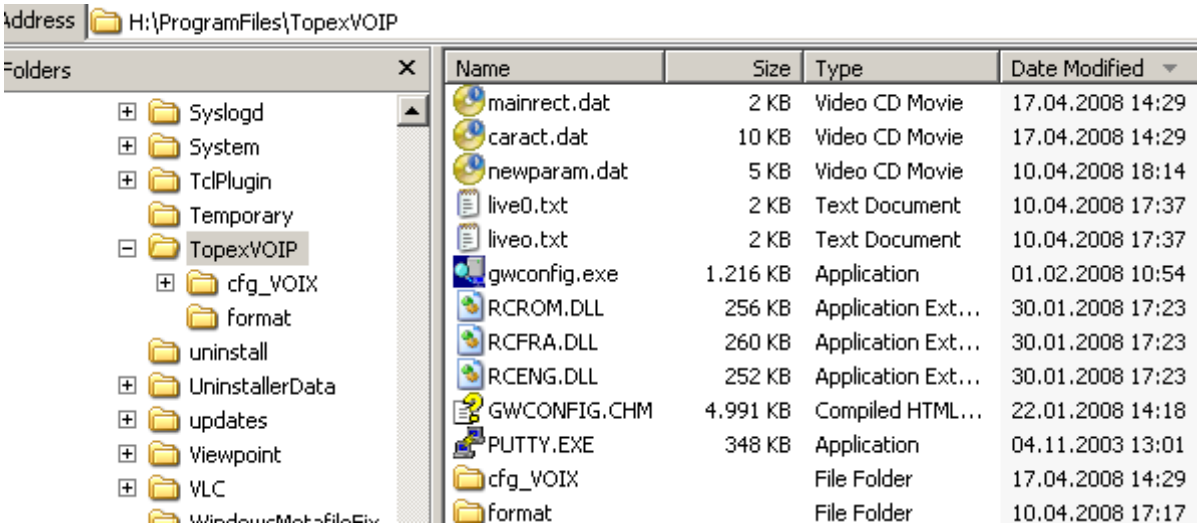
Note 3: on the Topex CD-ROM there is also a folder named "oam":

It contains all files which are necessary for the OAM application to run, in case that you need to copy them manually to a folder you want to employ, instead of using the installation wizard.



Note 4: if you have checked the box "I like to launch TOPEX VOXI - OAM" or when you go to "Start - Programs- TOPEX VOXI - OAM", the OAM program will start up.

From now on, in the selected directory (called TOPEX VoIP in this example) you will have several files including the “gwconfig.exe” OAM application:



The installation wizard adds here, besides the application for OAM, all the auxiliary files required, the online help, the PUTTY program that used for ssh, and the starting directories “cfg_VOIX” and “format”, where the configuration information will be stored.

To run the OAM , you should create a shortcut of the executable file on the desktop through the mouse right button.

Access the “Properties” option of this shortcut and click on the “**Shortcut**” tab.

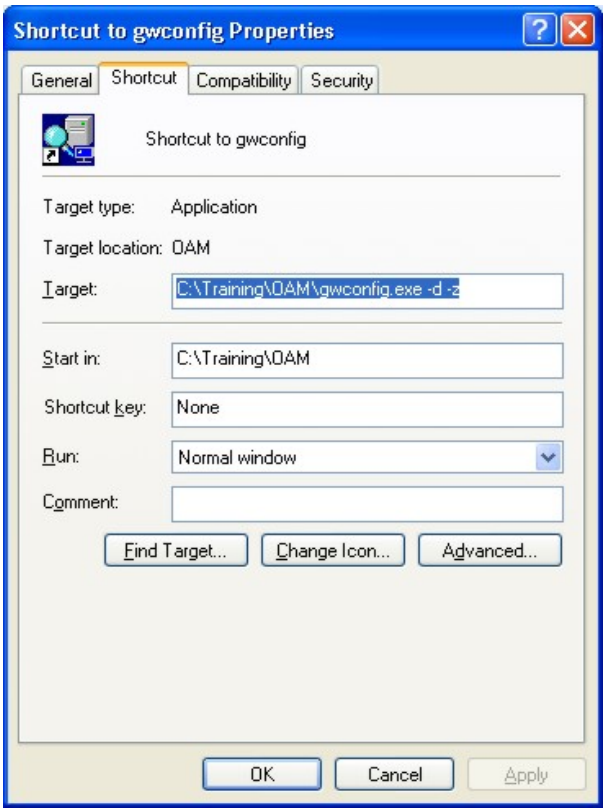
In the “**Target**” field is inserted the path where the shortcut is located. The path name can be for instance:

“C:\Training\OAM\gwconfig.exe”.

At the end of this name you may add the following parameters: “-d -z”.

In our example the path name will have the next form:

“**C:\Training\OAM\gwconfig.exe –d -z**”.



The parameters “-d” and “-z” have the following significance:

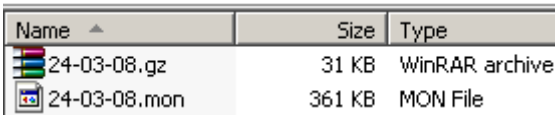
- “**-d**” parameter allows you to run the OAM application more than once in order to be connected to more than one VoiBridge equipment at the same time
- “**-z**” parameter allows the compression of the files which are downloaded and uploaded, for a faster access on the equipment.

Usage of starting parameters

- normally, the OAM application may be connected just to a single remote equipment at a time. So, if you are connected to Topex gateway no 1 and want to manage another gateway, you must first disconnect from the first one, then perform a connection to the other equipment. But the parameter “-d” allows multiple instances of the OAM program to be launched, so you can connect to several remote gateways at the same time.

- when the OAM is connected to a remote gateway, it may transfer back and forth configuration files, billing files, monitoring information, files about alarms and ASR, etc. The parameter ”-z” performs compression of these files, which get the extension “gz”. Compressed files require less space on the disk drive and they are also faster to upload or download.

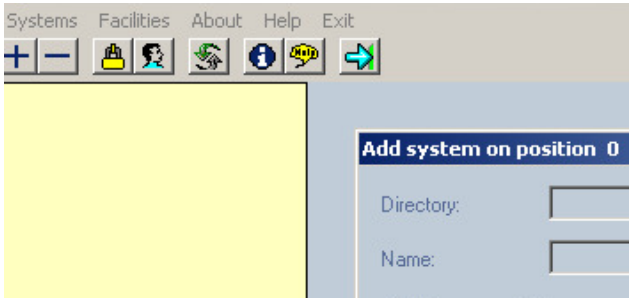
Compressed configuration files are generally ten (10) times smaller than the uncompressed file:



2.1 O.A.M. Connection

2.1.1 Achieve a connection

Run the “gwconfig.exe” application and access the main configuration window of the OAM application.



To add the first VoiBridge equipment to the OAM application, press the “**+**” button from the button bar or from the “**Systems => Add**”. You will see on the screen the “**Add system on position n**” window. Since the program is freshly installed and this is your first gateway, the value of “n” will be zero – you add the **first** VoIP gateway into the system!

The window “Add system on position n” shows up:

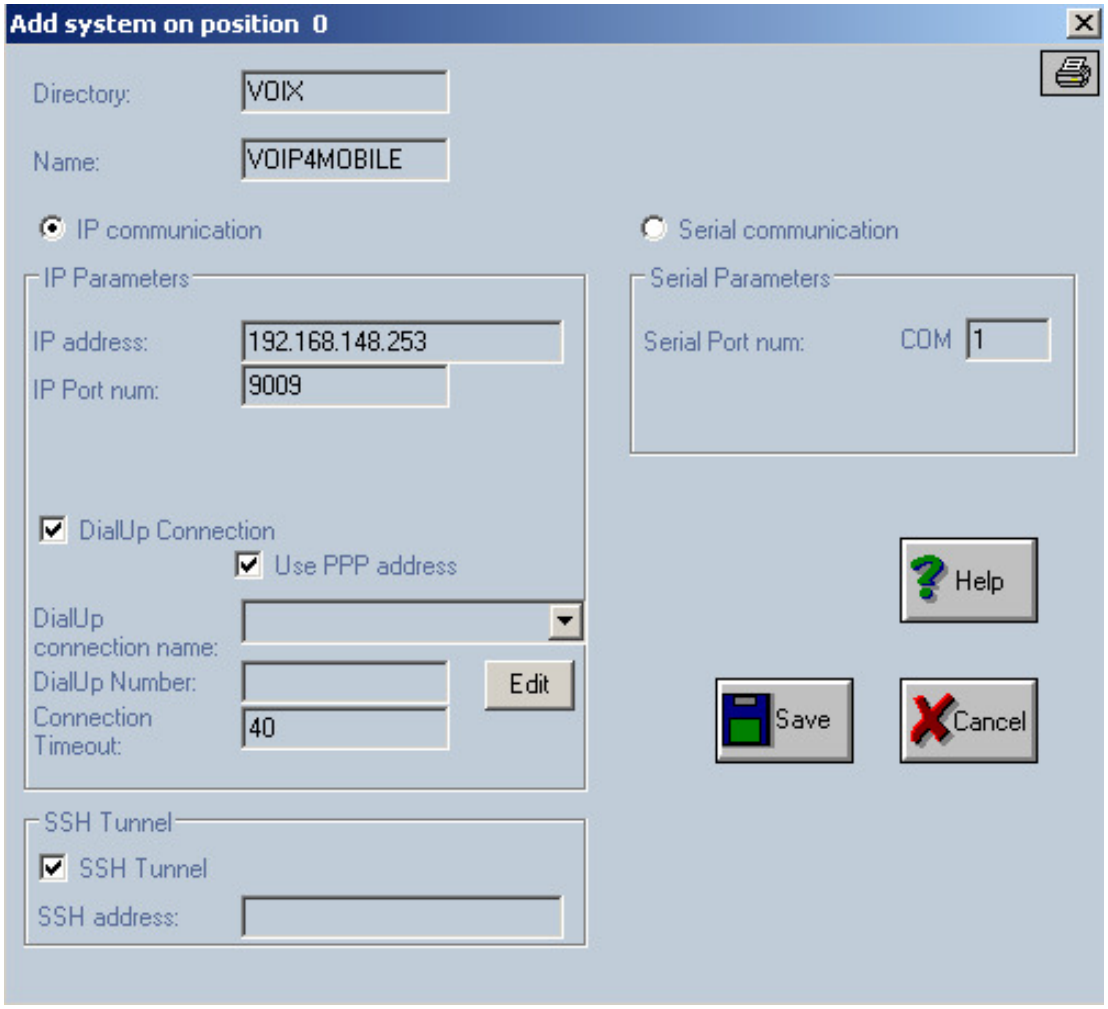


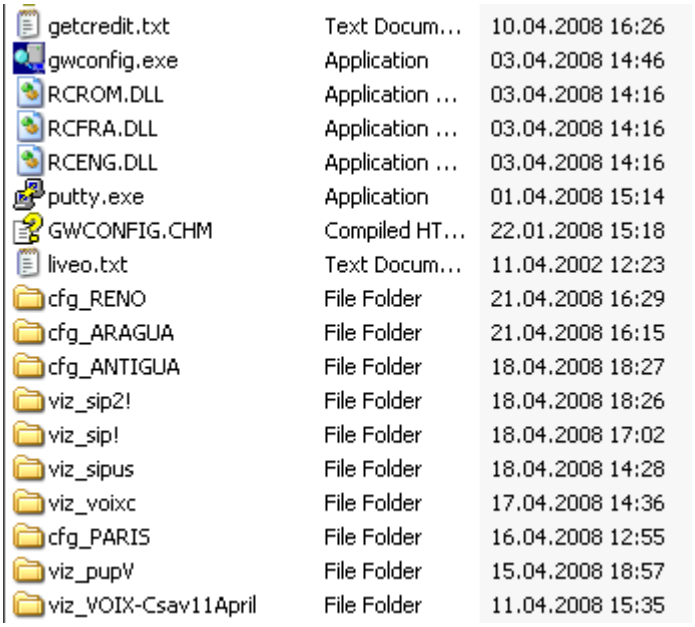
Figure 1 – Adding system to configuration

Fill in the following parameters:

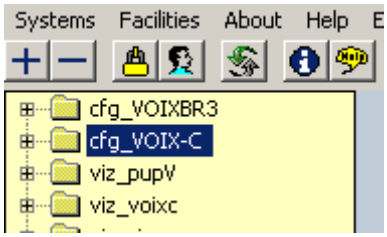
- **Directory** – the name of the folder from the hard disk of the PC where the equipment configuration will be downloaded;
- **Name** – the name of the remote equipment.

Warning: Do not use for the directory name characters such as ": / -" or blank spaces because the system will not allow the creation of the respective directory.

On the HDD of your computer the OAM program will create a directory with the following name: "cfg_Directory" where directory is the name typed in the "Directory" field. The folder will be created as a sub-folder in the directory where the executable 'gwconfig.exe' is located.



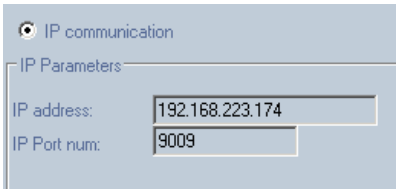
Likewise, the Name of the remote gateway is concatenated with the prefix "cfg_" and this it will be the text used in tree structure for identifying the system. The same name will be concatenated with the prefix "viz_" to indicate a gateway configuration that is stored locally.



- **Serial communication / IP communication** – here you must specify the type of link between the OAM computer and the remote equipment. There are two exclusive options for the communication with a TOPEX gateway, either IP or serial. Note that in the current implementation serial communication is not available for OAM, the serial link may be used only to access the Linux console of the equipment, not for the OMA program!

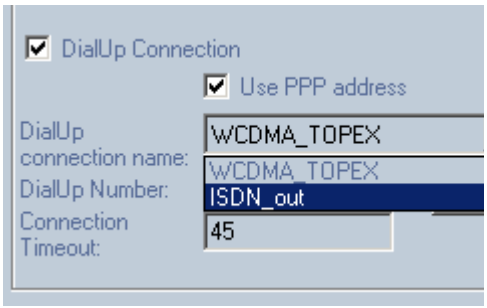
IP Parameters:

IP address - enter the IP address of the system. It can be a numeric IP address or a text address (in that case a DNS request will be made by software).
IP Port num - enter the number of the port through which the communication with the system is achieved.

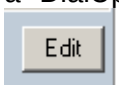


The default value is 9009. This value is established also into the gateway system and should not be modified without a good reason. If you must change it, remember to change accordingly the value on the remote gateway, otherwise the communication will be impossible .

DialUp Connection - in case of IP communication this setting allows if is checked, a dialup connection to be established. The Dialup Connection must be create from Windows (from "Dial-Up Networking") and the connection name must not contain spaces inside the text.
The checkbox "**Use PPP address**" is used to indicate to the OAM software to connect to the dialup server address after a successfully connection.



If the PPP address option is not validated, then the OAM program software will try instead to connect to the address specified in the "IP address" field. When this DialUp Connection" is checked, you may choose a dialup connection from "DialUp connection name". You must enter a "DialUp Number" which override the number used at the creation of the Dialup Connection.

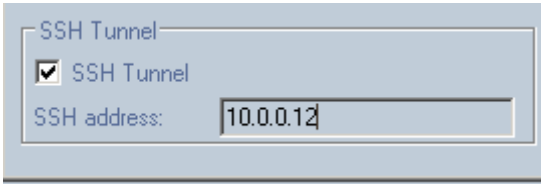


By pressing "Edit" button you can override the "User" and "Password" used in process of authentication.
The fields "User" and "Password" show up, allowing you to enter the parameters for correct log-in by means of the dial-up connection.

If you perform changes to "User" or "Password", those changes will also be made in the Windows registry.
The field "Connection Timeout" is the time in seconds to wait from the moment of starting the DialUp connection until achieving the connected status.

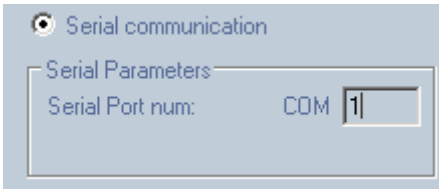
SSH Tunnel

If you check this box, you may connect to the VOXI gateway via secure a SSH tunnel.



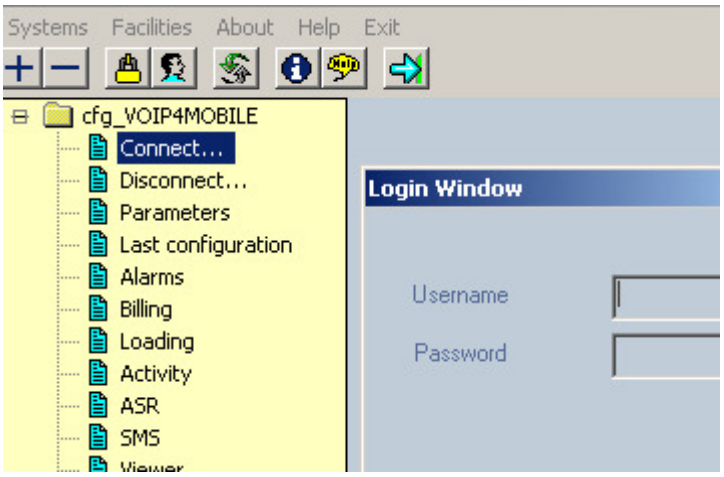
Serial

Port number for Serial communication. This is not implemented in the VOXI series, they use the COM port for HyperTerminal only, not for the OAM program.



When you have finished with the settings for "Ad System on position ", remember to press the Save button in order to keep the changes or Cancel if you don't want to save them!

Then on the arborescent structure of the newly added equipment press the “Connect…” button to log on the TOPEX VoiBridge equipment. In this example the name of the gateway is “VoIP4Mobile”

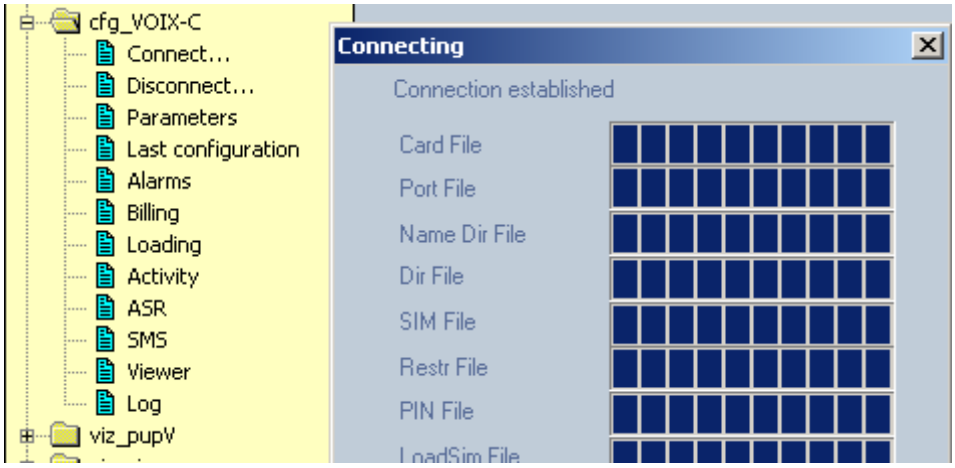


On the screen will appear a window that asks you for the user name and password for the connection to the VoIP gateway.
The default user name and password used for the connection to the VOXI equipment are:

- Username: **gsmgw**
- Password: **5tgb4rf**

2.1.2 Connection to OAM

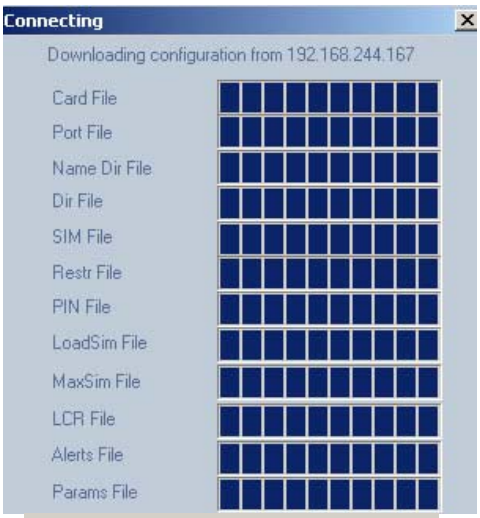
If you have entered the correct password, you are now connected to the respective gateway, so the OAM program shows “Connection established” and start downloading the configuration files from the remote equipment.



Once the connection is successfully established the OAM program tries to download the configuration information, you will see a progress bar for each transferred file:
· File with information about the cards that are installed on the equipment("card.cfg");
· File with information about the settings of port ("port.cfg");
. File with information about names of the directions ("dirname.cfg");
· File with information about established directions ("dir.cfg");
· File with information about the SIM modification table ("simindex.cfg");
· File with information about restrictions defined ("restr.bin");
· File with information about PIN codes on GSM/UMTS interfaces ("pin.cfg");
· File with information about loading on GSM/UMTS interfaces ("loadsim.bin");
· File with information about maximum usage time on GSM interfaces ("tmaxsim.bin");

· File with information about the LCR table ("lcr.cfg" - LCR is an abbreviation to Least Cost Routing);
· File with information about the CALLBACK table ("callback.cfg");
· Files with information for defining the parameters for alerting and programming of the test calls ("trafic.cfg");
· File with information concerning the parameters used for communication between a Topex device and the "gwconfig" OAM application, and the parameters used for debug purposes ("exec.cfg");
· Files with information about the signaling point allocation ("mtpcfg") and CIC (circuit identification codes) allocation to the ports of the Voxi equipment ("isup.cfg").These files are used when SS7 cards are presence (for VoxiPlus or VoisTel devices) ;
- File used for the configration of VOIP "voip.cfg" and "group.cfg".

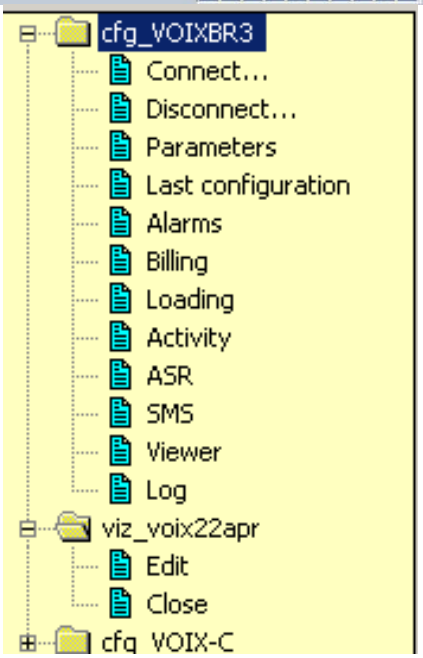
There are progress indicators for each of these files.



Together, all these files create a Voxi “configuration”

The OAM application "gwconfig" allows you to save such a Voxi configuration into a separate location (a directory on the hard disk).
Then you can edit (modify) this configuration even when you are no longer connected to the remote equipment, and later upload the saved and changed configuration it to one or more remote TOPEX boxes.

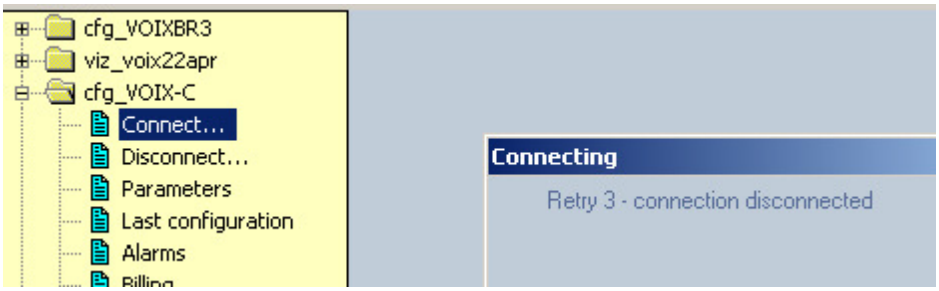
This way, if you have to configure several equipments with identical or close resembling settings, you will have less work to perform.



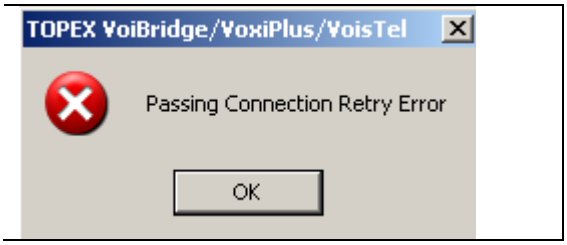
2.1.3 Connection problems

There are a few problems that can occur while attempting to connect to a remote TOPEX box:

1.The OAM program tries up to four times to connect to a system.



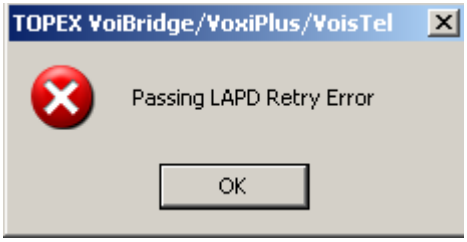
If it can't connect at the fourth retry it displays the following error message: **"Passing Connection Retry Error"**.



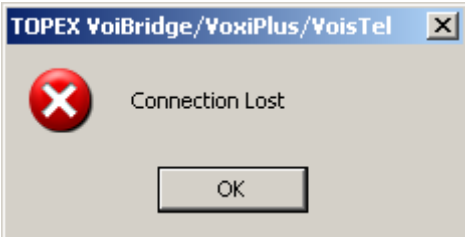
2. In the case of more than four attempts for login the following error message will show up: **"Passing Authentication Retry Error"**. If this happens, your will not be able to view or modify the system configuration.
3. If the program does not receive from the remote system one or more of the configuration files, it displays the following error message:
"**Downloading Configuration Failed**". If this happens, your will not be able to view or modify the system configuration.

4. If the dialog process between system and OAM computer falls (up to a number of consecutive errors) a **"Passing LAPD Retry Error"** message occurs.

In this case you should try again to connect;

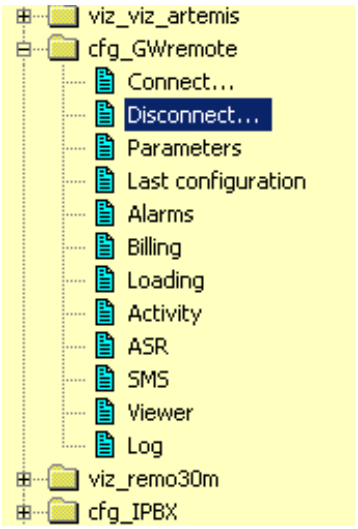


5. If the connection breaks down (because of the IP link) during the configuration downloading process or during the connected state, a **"Connection Lost"** message appears:

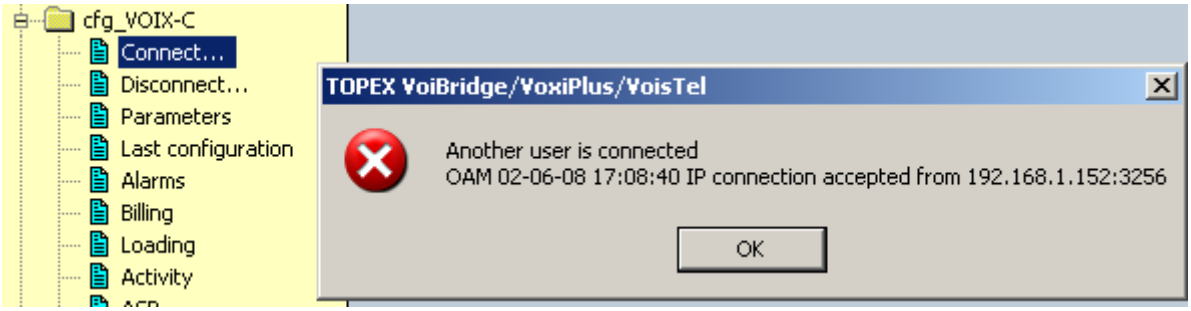


7. You can only connect to **one** TOPEX system at a time.

Thus, if you want to connect to another system, you must first **disconnec**t from the current one and only after this you can connect to the other remote equipment.



8. Also, only a single user of the OAM program may be connected to a specified TOPEX box at a time. So, if another user is already connected to the system that you want to manage at this moment by means of the "gwconfig" software from another PC, when you try to access the Topex equipment you get the following error message:



or:

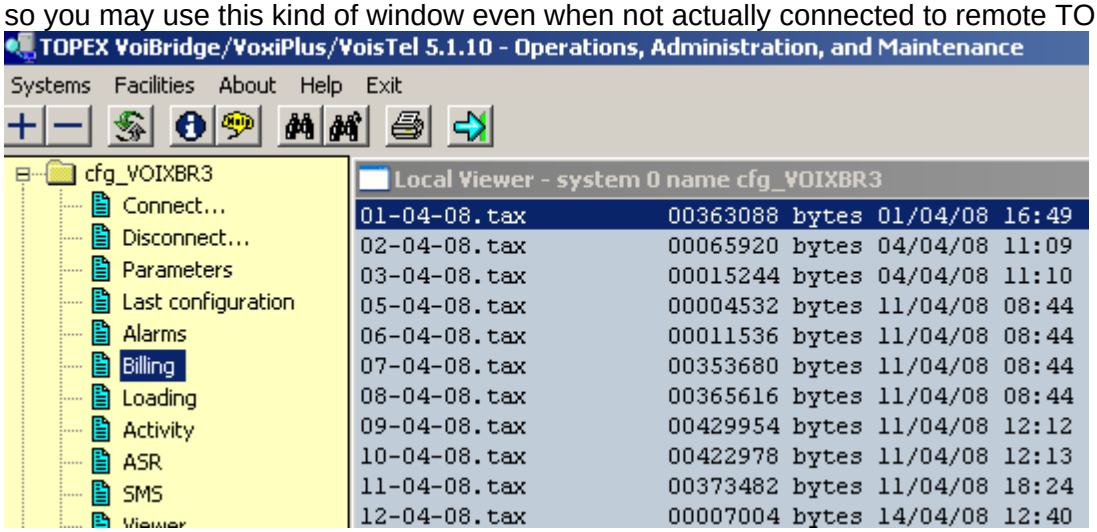
As you see, the message shows the time, date and **the IP address** from which the connection to that box has already been accepted.
If the message above is displayed, you must wait until the other user releases the equipment (performs Disconnect).

2.1.4 Types of windows

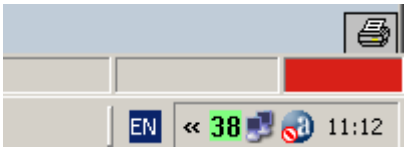
Please note that there are two types of windows used by the OAM program.

- There is one window displaying the **content** of the files. The title is either “Local Viewer” or “View Configuration”: 

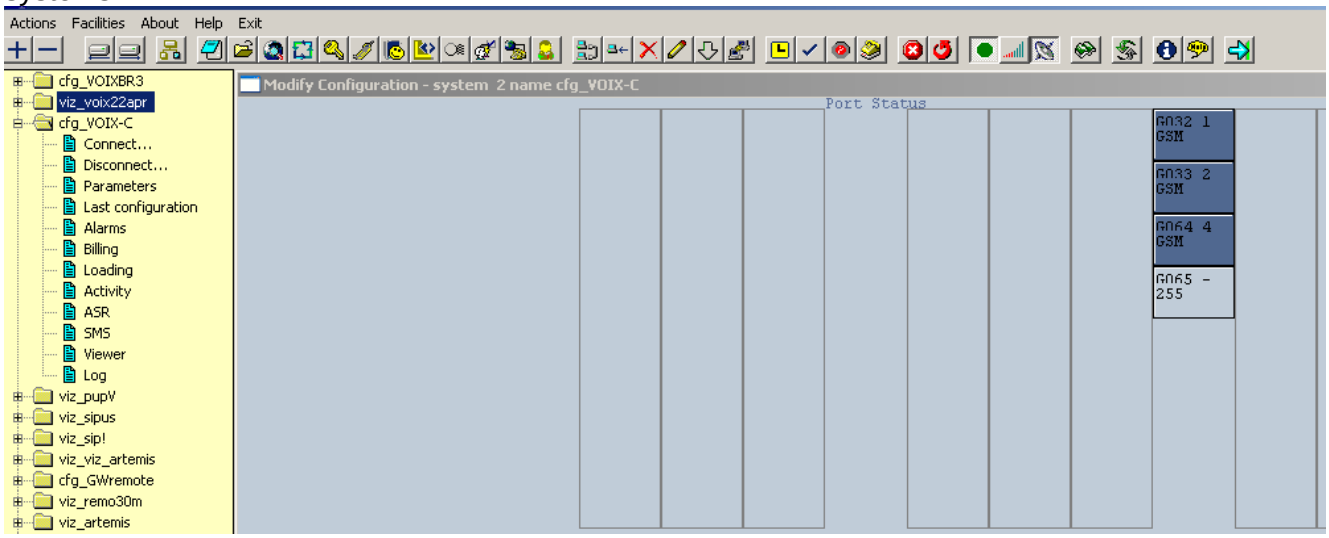
.
It deals with the configuration files stored locally in the specified directory of your hard disk drive, so you may use this kind of window even when not actually connected to remote TOPEX box:



The bottom rights corner (the sixth and rightmost column of the status bar) is colored in red, which indicates that there is no connection to the TOPEX box;

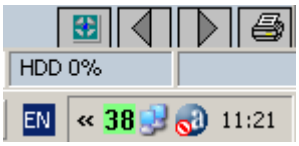


- the other kind of is window for viewing and changing the configuration (settings) of the Topex systems.



For this, you must be actually connected to the TOPEX equipment

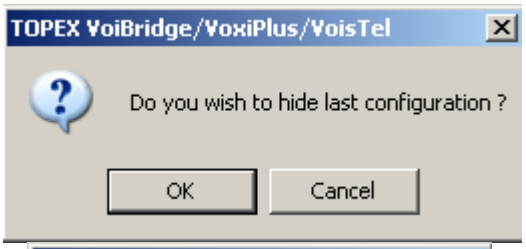
Notice that the sixth column of status bar (the bottom right corner) is colored in light blue instead of red, indicating a connected state:



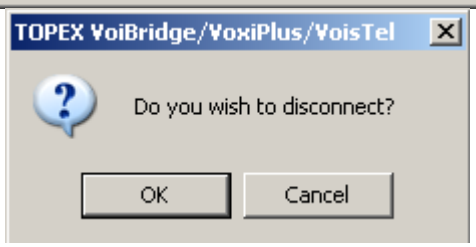
Also, the title of the window is now "Modify Configuration".

When you work with files content you will be able to operate upon the commands from the tree-like structure for the equipment you want and with the right mouse button you will be able to hide the window.

If you work (either change or just view) with a certain configuration, this can be hidden using the right mouse button.



If the OAM program is in the state of connection to one of the remote systems, by clicking with the right mouse button over the status bar, you will make pop up a window with the option of disconnecting from the current system.



2.1.5 Parameters for starting the OAM program

As mentioned previously, in order to allow additional facilities, the application may also be launched together with the following parameters in the command line:

1) The OAM software includes a protection against starting more then one instance. If it is necessary to start the software more then one time, you should use the following command to start the software: "**gwconfig.exe -d**" or "**gwconfig.exe -D**".
The "D" parameter allows simultaneous administration of several Topex boxes;

2) Parameter "-c" or "-C" allows automated connection of the administration program to a TOPEX box. The "c" parameter must be followed by a space delimiter and the three following fields: identification name for the remote system, user name and password (to allow automated log-in).
For example: "**gwconfig.exe -c TEST,<username>,<password>**"
In case of redundancy (equipments featuring dual processors) when the password is not explicitly requested - this parameter ("-c") must be used to provide the username/password to connect to each processor card:

3) Parameter "-s" or "-S" allows saving of several types of date. The information saved concerns the status of activation of monitoring, live monitoring and interrogation about mobile network information (cell IDs and signal levels)
For instance "**gwconfig.exe -s**" or "**gwconfig.exe -S**"

4) There are situations in which the configuration contains large files and thus the process of downloading is taking too much time. In such a situation the "z" parameter must be used in order to compress and transfer from and to processor card the compressed files.

The files are compressed by means of “GZip” and they get the extension .gz. These compressed files may be ten times or even more smaller than the original, uncompressed files

Name	Size	Type	Date Modified
02-06-08.gz	56 KB	WinRAR archive	02.06.2008 17:07
02-06-08.mon	668 KB	MON File	02.06.2008 17:07
29-05-08.gz	25 KB	WinRAR archive	29.05.2008 18:15
29-05-08.mon	267 KB	MON File	29.05.2008 18:15
30-05-08.gz	35 KB	WinRAR archive	30.05.2008 14:46
30-05-08.mon	406 KB	MON File	30.05.2008 14:46

All these parameters can be combined and added to the command line.

2.2 OAM DESCRIPTION

When the program starts, a screen called “TOPEX VoiBridge/Voxi version x.y.z – Operations, Administration and Maintenance” shows up:

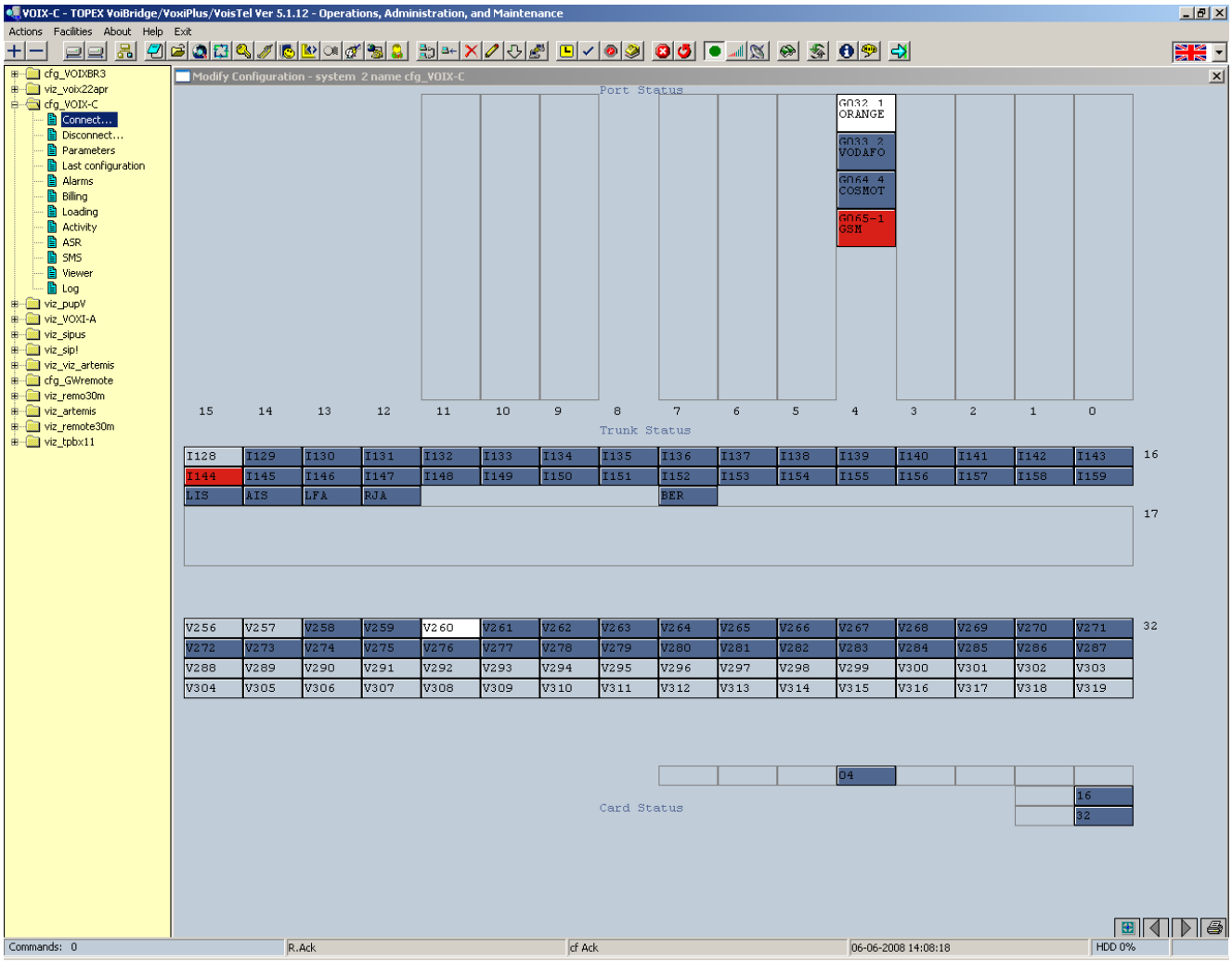


Figure 2 – Main OAM window

In the figure above there are displayed several types of cards and ports installed in the system. On position 4 there is installed the GSM card which contains four ports (G-032, G-033, G-064, G-065) corresponding to the GSM modules.

The main window of the OAM application contains the following items:

- First, a menu bar, with names such as Systems, Facilities, About, Help, Exit;
- Underneath it, a toolbar with different buttons marked with icons;
- To the left, the tree-like structure of connected gateways;
- The right panel, blue, which is the main window for Configuration or Viewing, showing the cards and ports of the connected Topex VOIX system;
- At the bottom, a status bar and some icons for additional functions.

In the upper right corner you may see the flags for selecting the language – the options are English, French and Romanian.

2.2.1 Configuration modifier window

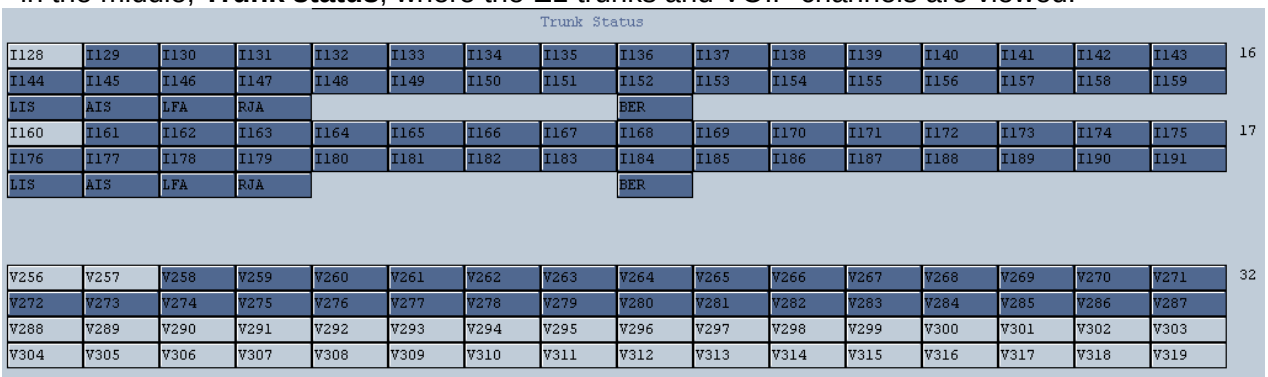
In this window you can view and change the current configuration (settings) of the equipment. The blue configuration window to the right of the screen is divided into three distinct areas:

- On top, the **Port status**, where the GSM ports or ISDN-BRI cards are viewed:

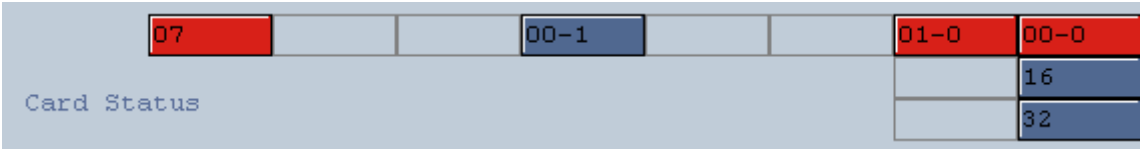
Depending on Voxi box type the "Modify Configuration" widow can display information concerning the GSM cards, the 2E1 (PRI) card or ISDN-BRI card (this only in case of the VoxiPlus or VoisTel boxes), as shown in the table below:

Indicating letters	Type and characteristics	Where it can be found
G	GSM/UMTS modules board with four ports	every equipment from the Voxi series
NT/TE	ISDN-BRI board with 4 ports	VoxiPlus and VoisTel
D	E1R2 board (32 channels)	VoxiPlus and VoisTel
I	E1ISDN-PRI board (32 channels)	VoxiPlus and VoisTel
T	E1SS7 board (32 channels)	VoxiPlus and VoisTel
R	E1R1 board (32 channels)	VoxiPlus and VoisTel
V	VOIP card - using H323 or SIP protocol (8, 16, 32 or 64 channels)	every equipment from the Voxi series

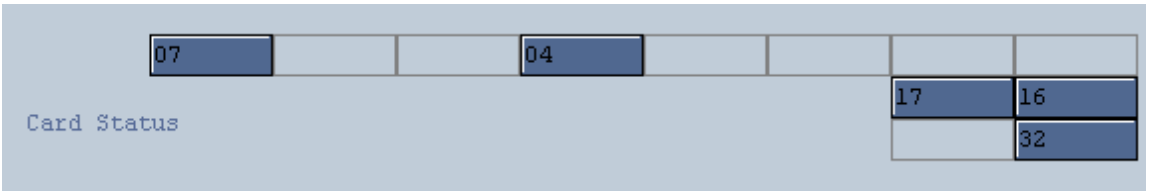
- in the middle, **Trunk status**, where the E1 trunks and VOIP channels are viewed:



- at the bottom, **Card status**, where the boards inserted in the equipment rack are viewed:

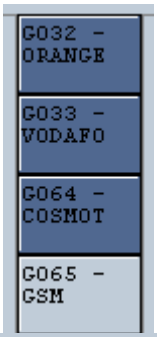


or:



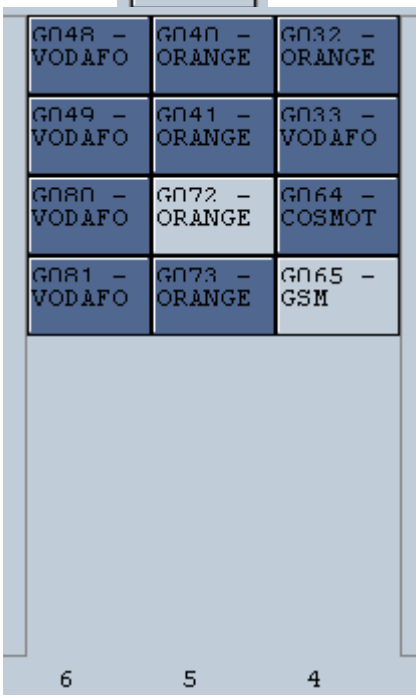
2.2.2 Port Status – GSM

The GSM card is represented as four rectangles displaying each two lines of text:
- the first line contains the 'G' character (to indicate a GSM card) followed by the physical position (in these example of this image the port positions 032, 033, 064 and 065)



- the second line shows the direction to which GSM module (port) belongs to (the first six letters are shown), the default value is the generic “GSM”.

The default GSM card will be always displayed on the card slot with index 4.
In case of the presence of the additional GSM cards (extended capacity equipments) - the next cards will be displayed to the **left** side of the first GSM card, as shown in this example on slots 5 and 6.

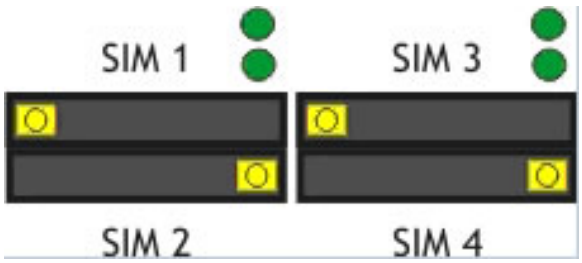


The possible positions for GSM cards are 4, 5, 6 and 7.

The basic VoiBridge features a single GSM card, the card positions 5, 6 and 7 are available only in case of hardware extensions which may use up to four cards, totaling 16 mobile modules.

The card positions to the right of the GSM card, that is from 0 to 3, are used for VoxiPlus and VoisTel equipped with ISDN-BRI/GSM card.

The positions of the SIM slots (holders for SIM cards) and corresponding indicator LEDs on the front panel of the GSM card is shown in the next image:

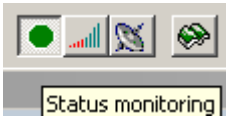


The relationship between the number of the SIM and the OAM representation is as follows:
SIM 1 - first port from the GSM card representation, SIM2 - the second port, SIM3 - the third port and SIM4 - the forth port.

For the case of the default GSM card position the relation between SIM number and GSM module number is:
SIM1-032;
SIM2-033;
SIM3-064;
SIM4-065

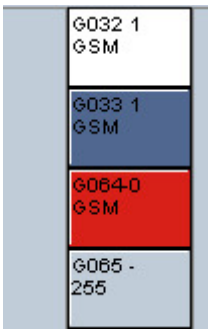
GSM Status

Too see the actual state of each port, selects the icon "**Status monitoring**" in a state of connection between the OAM program and the Topex box.

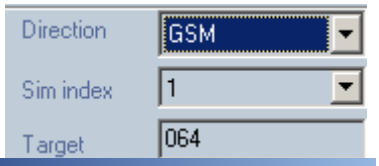


The port representation is changing according to its current state.
- port in use (in a call)d - **white color**
- port in alarm - **red color**
- invalid position -**light blue** (invalid position means either card error or the "Installed" field not set for the respective port)
- port available - **dark blue.**

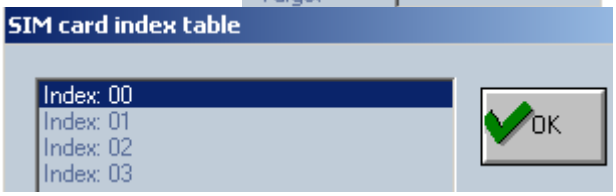
A functional GSM module with a properly registered SIM card is displayed in dark blue color.



In the above image module 032 is in a call, module 033 has a registered SIM (taking into account the relation between the SIM slots from the physical card and the GSM module representation, the SIM2 is registered and ready for calls), module 064 is in alarm and module 065 is not installed.
In the representation of a registered SIM it is also displayed a number - in above last image that number is "1" or “0”.
This number is a legacy from the previous version of TOPEX GSM where each GSM port featured a SIM holder with **four** positions for SIM card. Number "1" in that case was an indicator of the first SIM of that holder.
A sim card from the SIM holder was chosen based on SIM index value or based on SIM balancing algorithm.



The SIM index was created as a rule to specify which SIM card from the SIM holder (1,2,3 or 4) is to be used on specified time period and day of the week.



The SIM balancing algorithm was created to allow an equal mode of using each SIM from the SIM holder. In case of Voxi series - where each GSM module has a single SIM card - each time a SIM card is chosen from the SIM index or from SIM balancing algorithm then the SAME PHYSICAL SIM WILL BE USED. But by default the SIM index rule must be used – the SIM index which contains SIM card 1 should be used.

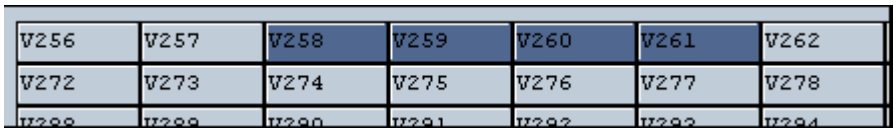
GSM module color

If the GSM module is displayed in red color (therefore no SIM registered) then after the port position and a line character it is presented a value from 0 to 5 which describes the cause of not having a registered SIM on that position.
The GSM module assigned to the direction ORANGE is displayed in red color with the cause of error "2" meaning "no physical SIM".

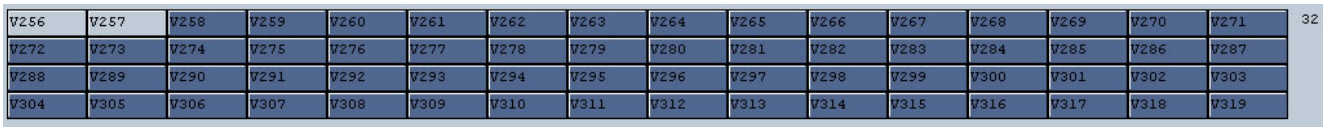
The GSM module 065 is in red color with the cause of error "3" meaning "no logical SIM" (a physical SIM is present, but it is not properly registered because of SIM index or Load Sim problems, calls cannot be made).
Here the cause of error is "0" meaning initialization phase, either for connection to the mobile network or programming (after it has received AT commands). This is just a temporary condition error, later the module should be available.

2.2.3 VOIP Card Status

The **VOIP card representation** features four rows, each with 16 channels. The total number of channels depends on how the box is equipped.
For example for a VoiBridge box there are just four VoIP channels, since only a maximum of four GSM modules are available for connection:



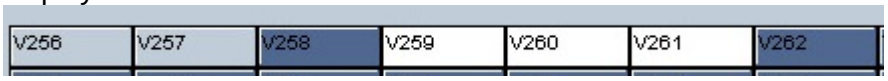
For each channel, the type of the VOIP card is shown as a small rectangle marked with the 'V' letter, followed by a three-figure number, which is the number of the channel. The numbers start form 256 and increase up to 319:



The protocols supported by the Topex VoIP card are H232 and SIP. The codecs used are G711, G723, G728, G729.

Please note that the first two channels on the VoIP card (respectively "V256" and "V257") **cannot be used as valid voice channels**. Thus the first two cases of the VoIP card will always be colored in light blue, indicating channels not available for voice.

Here also the uninstalled ports are colored in light blue, while the ports that are in a call are displayed with the color white:

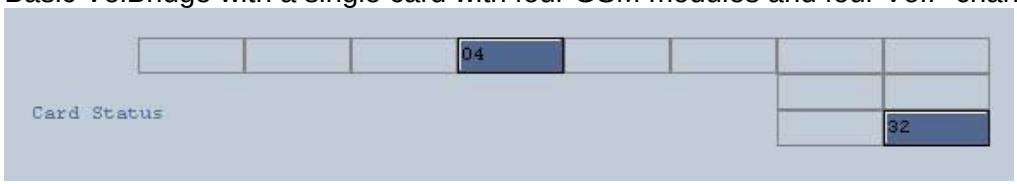


In the area "**Card status**" of the window there are three rows:

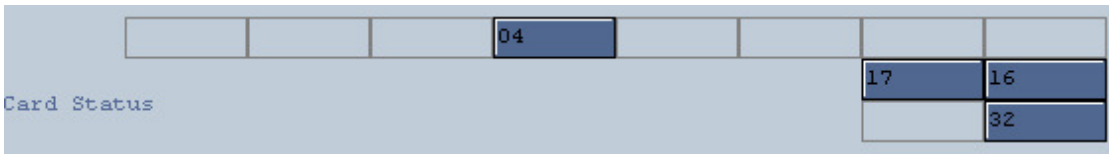
- the first row is a representation of the status of the boards installed in the cabinet of the equipment. Each board is shown as a small rectangle marked with the number of the card (from right to left). In the middle (on position 4 in case of VoiBridge) there is the first board with GSM nodules. The current status of the board is shown by its color.
- the second row show the one or two E1 trunks (positions 16 and 17)
- the third row shows the status of the VoIP trunk (position 32).

See below a few examples of “Card Status” in different VOIX configurations:

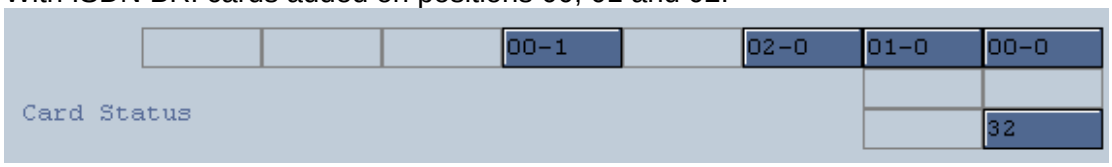
Basic VoiBridge with a single card with four GSM modules and four VoIP channels:



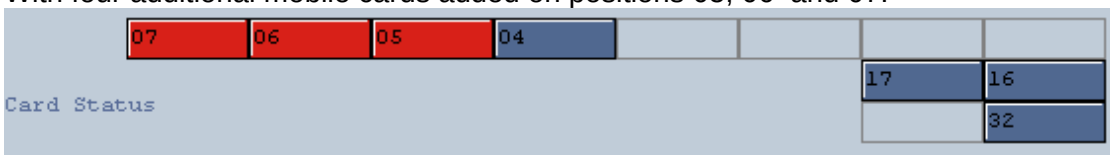
VoixiPlus with two E1 cards installed:



With ISDN BRI cards added on positions 00, 01 and 02:

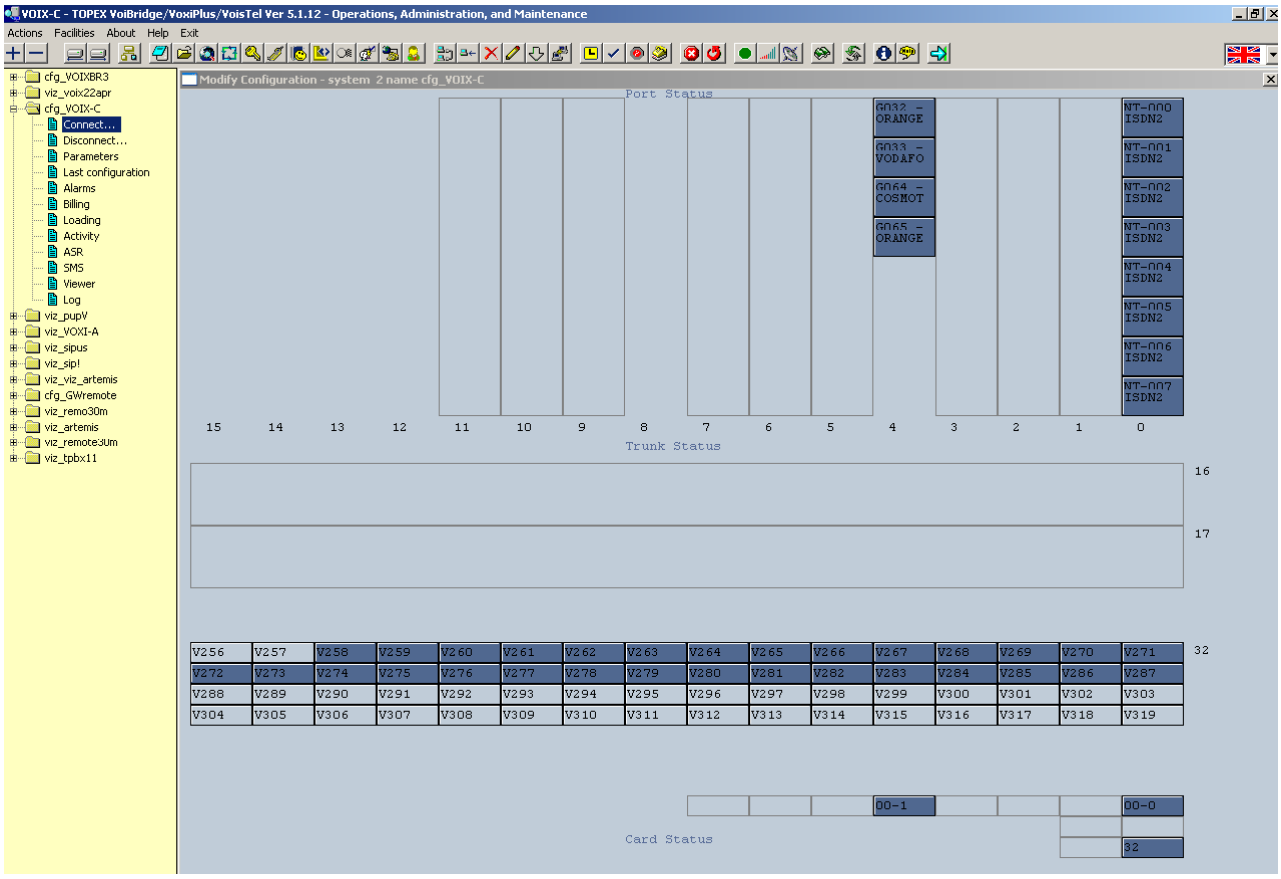


With four additional mobile cards added on positions 05, 06 and 07:

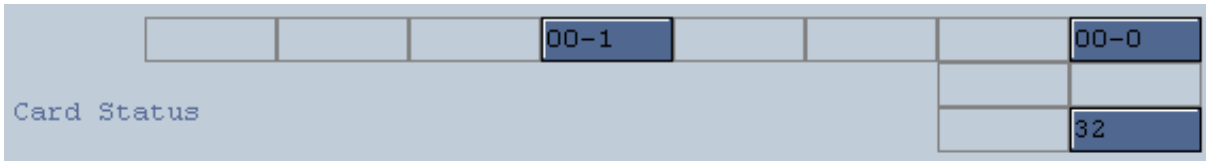


2.2.4 ISND Card Status

For a **VoxiPlus or VoisTel system** the OAM screen will display in addition to the GSM and VoIP cards explained previously, also another card, which may be either an E1 card or an ISDN-BRI card, depending of the customer configuration.
See below a configuration with an ISDN-BRI card:



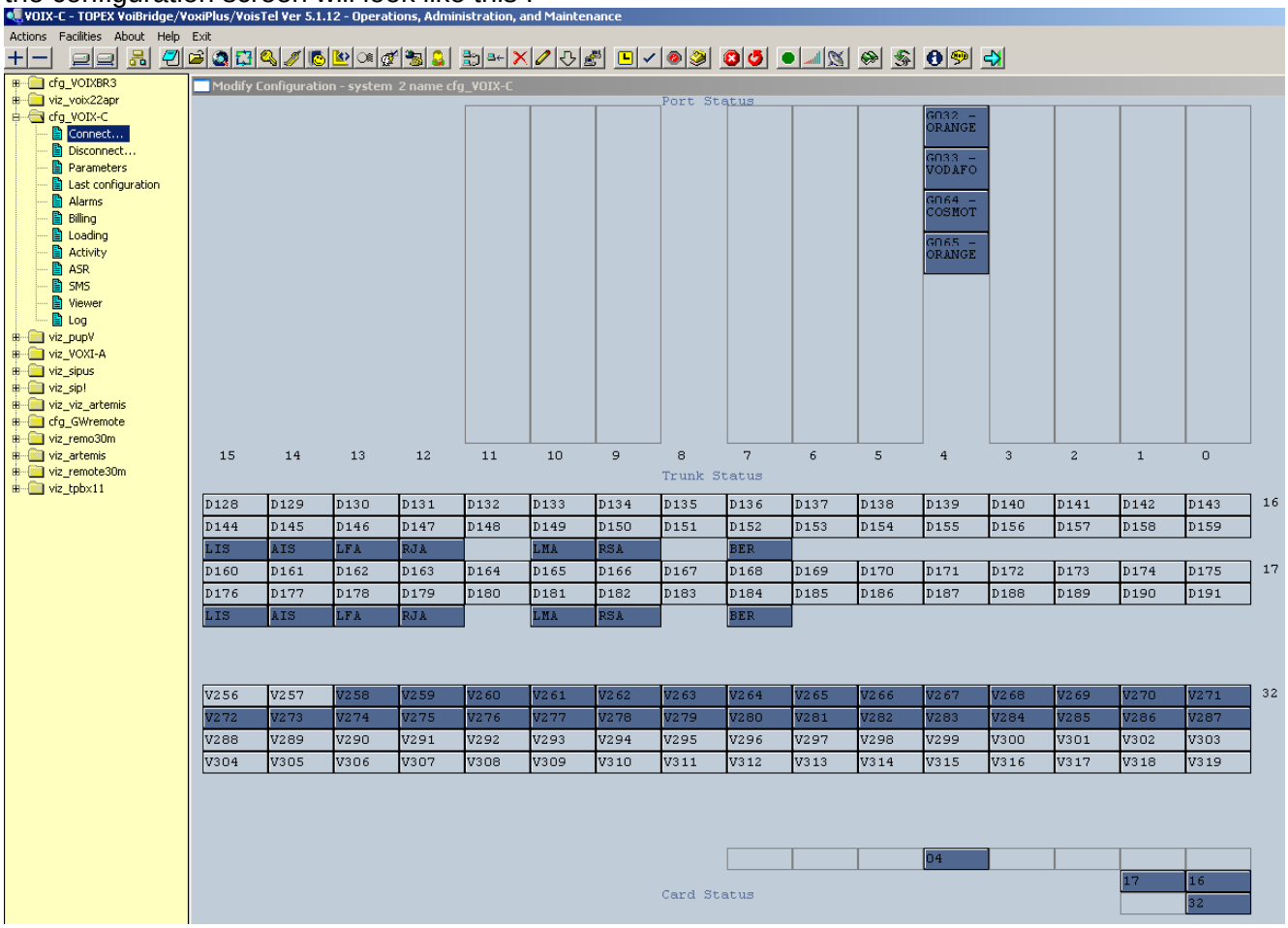
In case of the presence of the ISDN-BRI card, in order to highlight that the same physical ISDN-BRI card is equipped also with four GSM interfaces, in the Card Status zone this card appears with the index "00" followed by "-0" for its ISDN side and respectively "-1" for its GSM side:



So, remember that in case of ISDN cards, the OAM screen will show them in two places, once for the GSM part and once for the ISDN-BRI (4 interfaces, 8 channels) part.

2.2.5 E1 Card Status

In case of a VOIX box with E1 card (the letter D indicates that the E1 trunks use R2 signaling) the configuration screen will look like this :



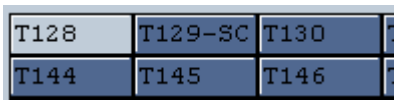
The **representation** of an E1 trunk features on the first two rows the channels.
For each channel, the type of the E1 interface is shown as a small rectangle marked with a letter, followed by a two-figure number - the number of the channel. In the above examples, channels are from D120 up to D191.
The third row includes representation of the alarms.
Each alarm button is marked with an abbreviation for the type of alarm it represents, as described in the table below:

LIS	Loss of Incoming Signal
AIS	Alarm of Indication Signal
LFA	Loss of Frame Alignment
RJA	Remote of Junction Alarm
LMA	Loss of Multiframe Alignment (not for an IDSN or SS7 interface)
RSA	Remote of Signaling Alarm (not for an IDSN or SS7 interface)
BER	Bit error rate

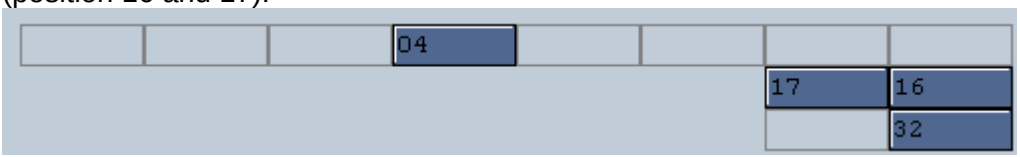
- Code representation used on drawing the E1 channels:

D	E1R2 trunk channel (32 positions on a E1 trunk)
I	E1ISDN trunk channel ISDN (32 positions)
T	E1SS7 (32 positions)
R	E1R1 (32 positions)

- Each channel from each E1 interface can be individually configured by clicking with left mouse button on the corresponding channel. For each channel the "Category" and the "Direction" parameters can be settled.
- Up to two E1 trunks can be installed in a VOXI system.
- The indication "SC" is used to highlight the signaling channel:

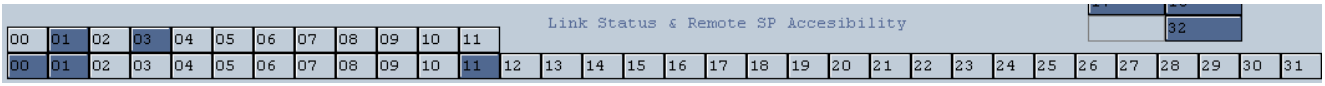


In the area "Card status" of the window, the second row shows the status of the E1 trunk (position 16 and 17).



2.2.6 E1-SS7 Card Status

In case of VoxiPlus or VoisTel equipped with E1 - SS7 interface, additional information will be shown in the Card status area. These include LinkStatus & REMOTE SP ACCESSIBILITY .



or:

This zone is displayed only if an E1 interface with SS7 signaling is present in the configuration. This zone is a representation of the accessibility of the remote (adjacent or non-adjacent) signaling points. The TOPEX box can handle connections to maximum 32 signaling points (SP) which are represented by boxes with values from 00 to 31. Values from 0 to 11 are used for adjacent signaling points while values from 4 to 31 are used for non-adjacent signaling points.

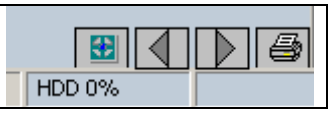
The colors used in displaying the "Remote SP accessibility" bar are:
- **light blue** for connection to a SP that is not installed (default state)

- **dark blue** for a connection to a SP which is installed (active). The information regarding the installed SP connections is read from the "MTP Configuration" file where you enabled the routes (option "Route x" where x is from 0 to 31):	

- **red** for a connection error (alarm) of an installed SP. This information is read again every 60 seconds.

-

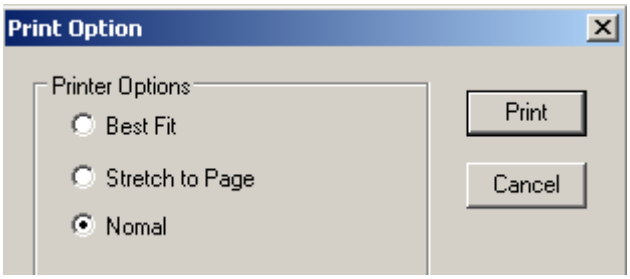
2.2.7 VOIP Card Configuration

At the bottom of the screen (bottom left corner) there are several icons: a green cross, left/right arrows, and printer.	

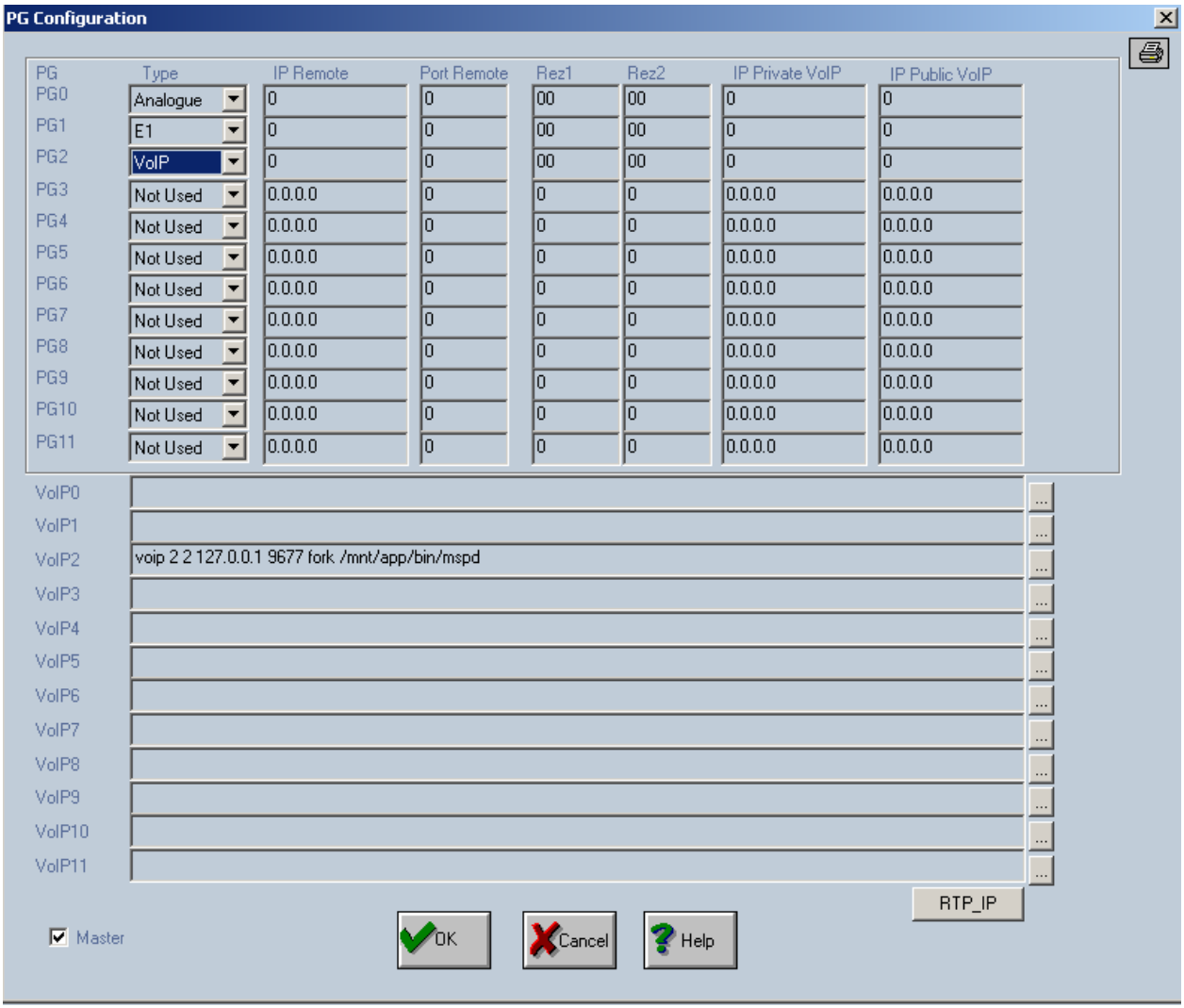
The most important is the first one, “  ” used to make the VoIP configuration.
Note: ***if this box is displayed on the OAM screen - then the "VoIP Configuration" icon () from the main toolbar is ignored by the main gateway application****. This rule is applied to the fields "VoIP Card IP address" and "Public IP address" which are in this case taken from the "PG Configuration" window (explained below). The rule is also applied to "VoIP Card MAC" and "IP GATEWAY MAC". The “VOIP configuration” icon on the toolbar on top of the screen is kept for compatibility with older equipments, which were configured this way.*

The arrows buttons are not used for Voxi series application. They are active in other equipments, where you need to navigate through several successive configuration pages of the OAM program.

The “Printer” icon sends directly to the default printer of your PC the current OAM screen. When you click it, the Print Option window pops up, asking you how to print the image: as a best fit to the printer paper, stretched or “normal”, as it is on the screen:



If you click the "  " icon, then the “PG Configuration” window is displayed:



Here you can see and modify the allocation of the slots that describe the configuration of the VOIX box. Currently only the first three (PG0 to PG2) are used, the others are reserved for extended equipments.

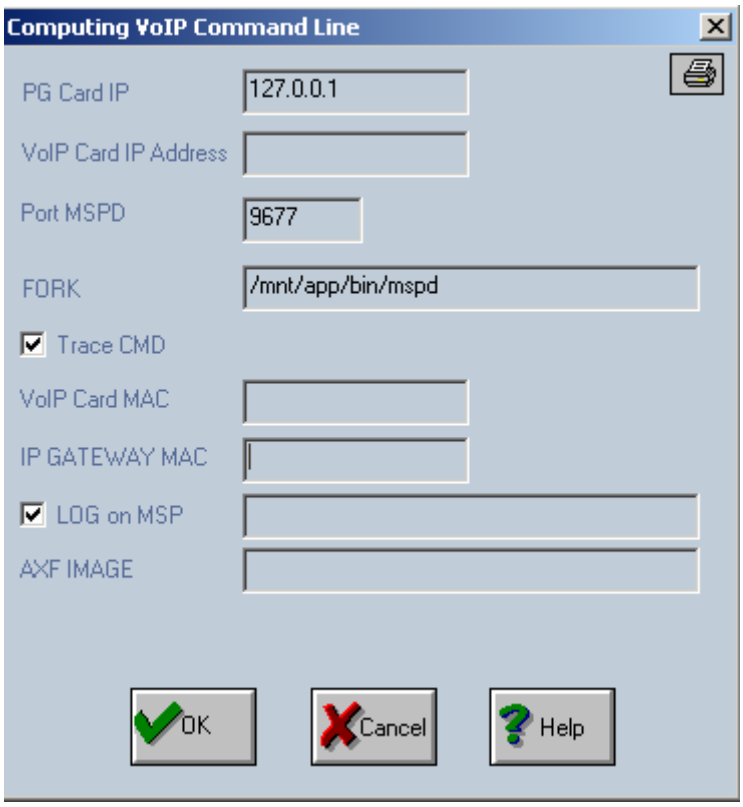
The "Type" field should be "Analogue" for PG0 (this slot contains the GSM and ISDN-BRI cards), "E1" for the second slot (representing the E1 trunks) and "VoIP" for the third slot (correspond to the VoIP card). **THESE FIELDS ARE COMPLETED BY DEFAULT WITH THE RIGHTS VALUES, YOU SHOULD NOT CHANGE THIS DEFAULT ALLOCATION OF THE TYPE OF THE SLOTS**.

Only for the VoIP slots a new line must be filled – in case of the VOIX equipments only the "VoIP2" line corresponding to slot 2 is of interest. This line includes the address of the processor card - 127.0.0.1, the MSPD port – 9677, and the forking action of the mspd application ("fork /mnt/app/bin/mspd).

The "IP Remote" and "Port Remote" are not used in Voxi configuration. The "Rez1" and "Rez2" fields are intended for further developments. Of interest to you are the fields "IP Private VoIP" and "IP Public VoIP", which represent the VoIP private IP - the VoIP IP card and VoIP public IP. Those fields are the same with "VoIP Card IP address" and "Public IP address" from the "**VOIP Configuration**" window. HERE YOU MUST ENTER YOUR VALUES, THOSE FIELDS ARE NOT FILLED BY DEFAULT. The address used in RTP packets is taken from "RTP_IP Parameters" window.

If the Voxi box is behind a gateway you may need to enter the MAC of the gateway and after it the MAC of the VoIP card.

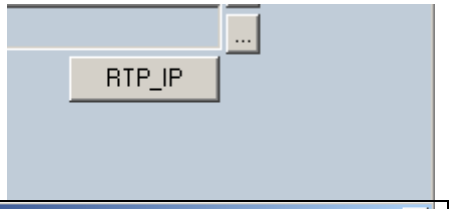
To edit a VoIP line, click the "..." button located in the right side of the respective line and the next window "Computing VoIP Command Line" shows up:



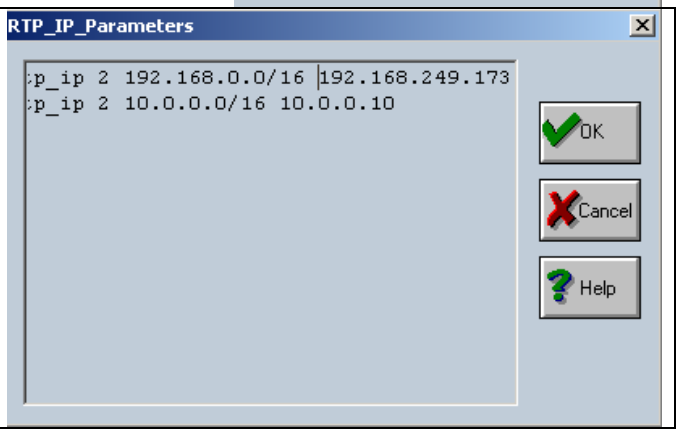
You should not change the other fields, they are present to increase the flexibility of the equipment.

*If the VoIP configuration is performed from the "PG Configuration" window then in the "**VOIP Configuration**" window the fields "PG Card IP Address", "VoIP Card IP Address", "Public IP Address", "VoIP Card MAC" and "IP GATEWAY MAC" will be empty.*

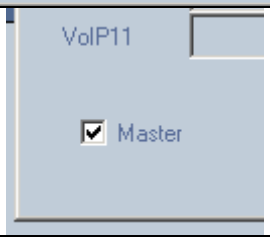
The "RTP_IP" button at the bottom right of the PG Configuration window is used to fill the IP used in RTP packets.



This value can be different if the TOPEX equipment has several Ethernet interfaces (but the VOIX box has just one interface for LAN and another for WAN).
In the "RTP_IP_Parameters" the first line must be edited when the IP address of the VOXI box is changed.



Finally, verify that the "Master" checkbox (located at the bottom right corner of the PG Configuration window) has to be enabled.



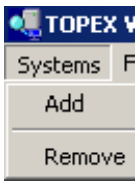
2.3 Menu description

The Menu bar features items Systems, Facilities, About, Help and Exit, which are described in this chapter:

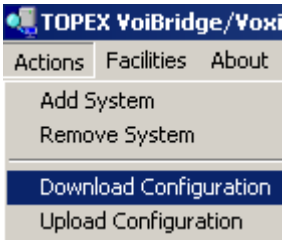
Systems		
	Add	Adds a TOPEX system to the tree like structure
	Remove	Delete a system from the existing tree-like structure
Facilities		
	Font	Chooses the font type for the text in the “Modify configuration” and "View configuration" windows
	Color significance	Shows the significance of colors used in "Modify Configuration" and "View Configuration" windows in the representation of cards and ports. The user is allowed to modify the colors at his choice.
	View print files	Opens a window in which you may see the text files into which other text files have been printed using the icon button “Print” (or the icon “Print”)
	Commands	Opens a window through which expert users may enter direct commands to the system
	License	Opens a window where is inserted the license activation code
About		Shows the software version.
Help		Opens up the window with the searchable help files for the program
Exit		Exit from the OAM program.

2.3.1 Systems

Through the OAM program you can configure and manage maximum 50 different TOPEX VoiBridge equipments.
At first, when OAM is not connected to a remote equipment, Systems allows you to add or remove Topex systems from the tree-like structure of configurations.

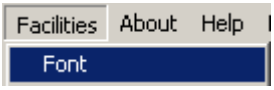


Then, When the OAM is connected to a Voxi box or when OAM is displaying the last configuration for a Voxi box - the first menu option named "System" is renamed as "Actions" and
contains as menu options all actions allowed through the OAM icons



2.3.2 FACILITIES => FONTS

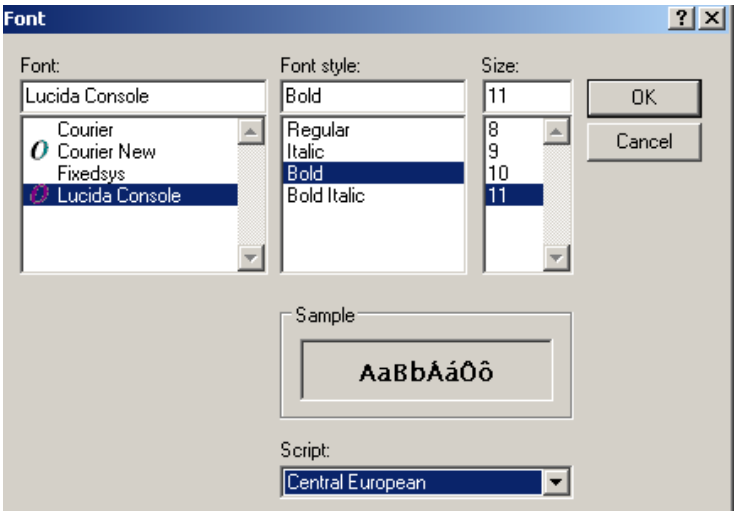
If you select "Facilities > Font" the font selection window shows up.



Here you can select according to your preferences

- the font (Courrier, Courrier New, Lucinda Console, Fixed Sys),
- the style (Regular, Bold, Italic)
- the size of characters (8, 9, 10 or 11 points)
- the script (Western, Central European, etc).

Remember that the changes of fonts will only affect the OAM windows called: "View Configuration" and respectively "Modify Configuration":

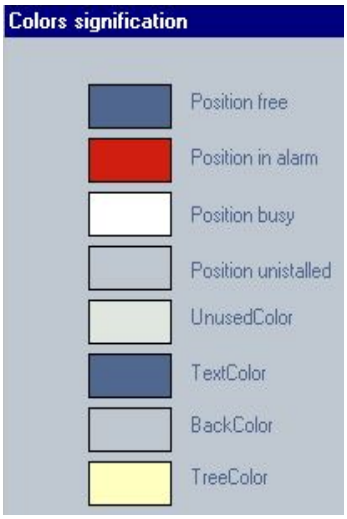


2.3.3 FACILITIES => COLORS SIGNIFICANCE

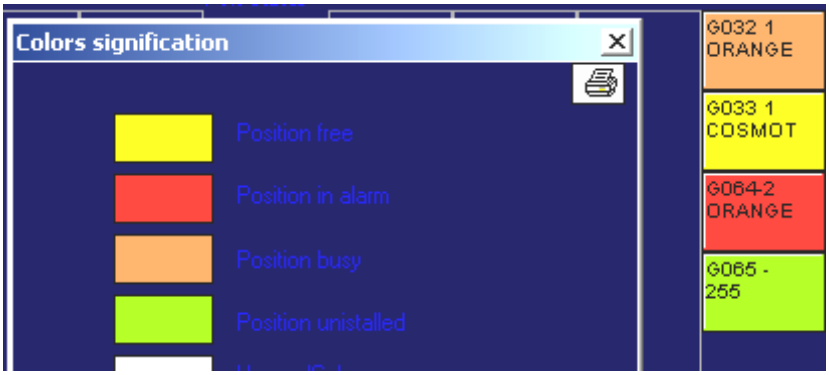
Here you may see or modify the color scheme used by the OAM program. This facility shows (and lets you change) the significance of the colors used in the windows "View Configuration" and "Modify Configuration", as explained next:

Dark blue for an available (operational) position;
Red for a position in alarm;
White for a position that is in use (busy);
Light blue for a position not installed (free slot);
You may change the colors for the text; for the background (the same colors for text and background will be also used for the dialog windows and for the tree-like structure.

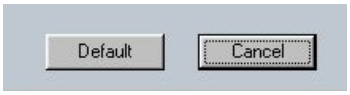
Each of these colors can be modified according to user's preferences by clicking on it.



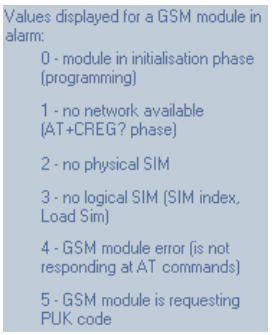
Note that the colors change immediately, there is no need for you to press a “Save” button to save the new color scheme:



If you want to restore the color settings to the original color set, you should use the "**Default**" button at the bottom

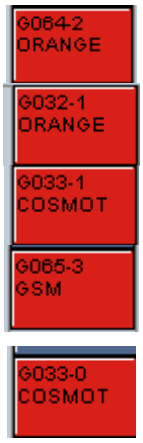


GSM Info – The text column at the right of "Colors signification" window explains the meaning of values shown for a mobile port in alarm. When a user selects the icon "**Status monitoring**" in a connected state, additional information will be provided for GSM ports. If the GSM module is displayed in red color (therefore no SIM registered) then after the port position and a line character a value is shown, describing the cause of the error. See below a few examples of Mobile ports in alarm, displayed in red color.



Here the type of error is "2" - "no physical SIM", meaning that no valid SIM is present in the slot of the module.

Error type “1” – no network yet available. The module may be in the process of registering with the mobile network.



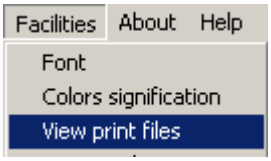
Error type “3” – no logical SIM. You may have a physical SIM card present, but the settings for SIM index and or loading prevent it from registering.
Error type “0” – programming. The module is in the programming (initialization) phase, try again later to see if it operate correctly.

Possible causes for the errors in the mobile modules are:

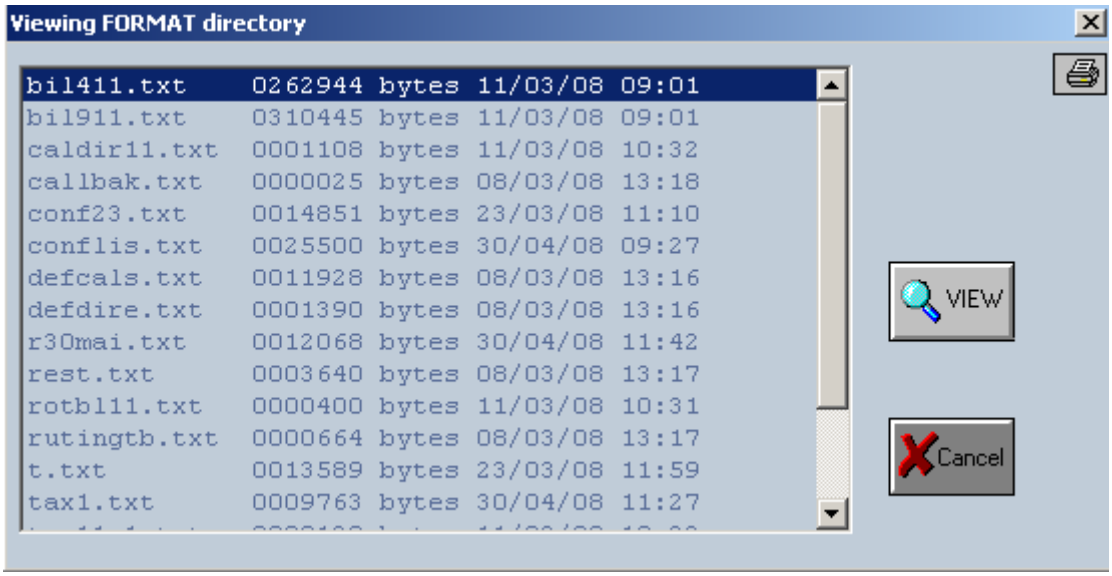
0 - GSM module in initialization phase (programming)
1 - no GSM network available ("at+creg?" phase)
2 - no physical SIM present
3 - no logical SIM. This value may be obtained in the situations such as: SIM with credit expired, when the "**SIM index**" rule is used for changing the SIM cards for the GSM module but for the current time moment no SIM value is specified in the "SIM index", or if "**LOAD BALANCING**" algorithm is selected to change the SIM on the GSM module and the "Load Sim" value (the actual usage time of the SIM) is greater than the SIM maximum time to use ("Max").
4 - GSM module error - the module is not responding at "AT" commands
5 - the GSM module is requesting PUK code.

2.3.4 FACILITIES => View printed files

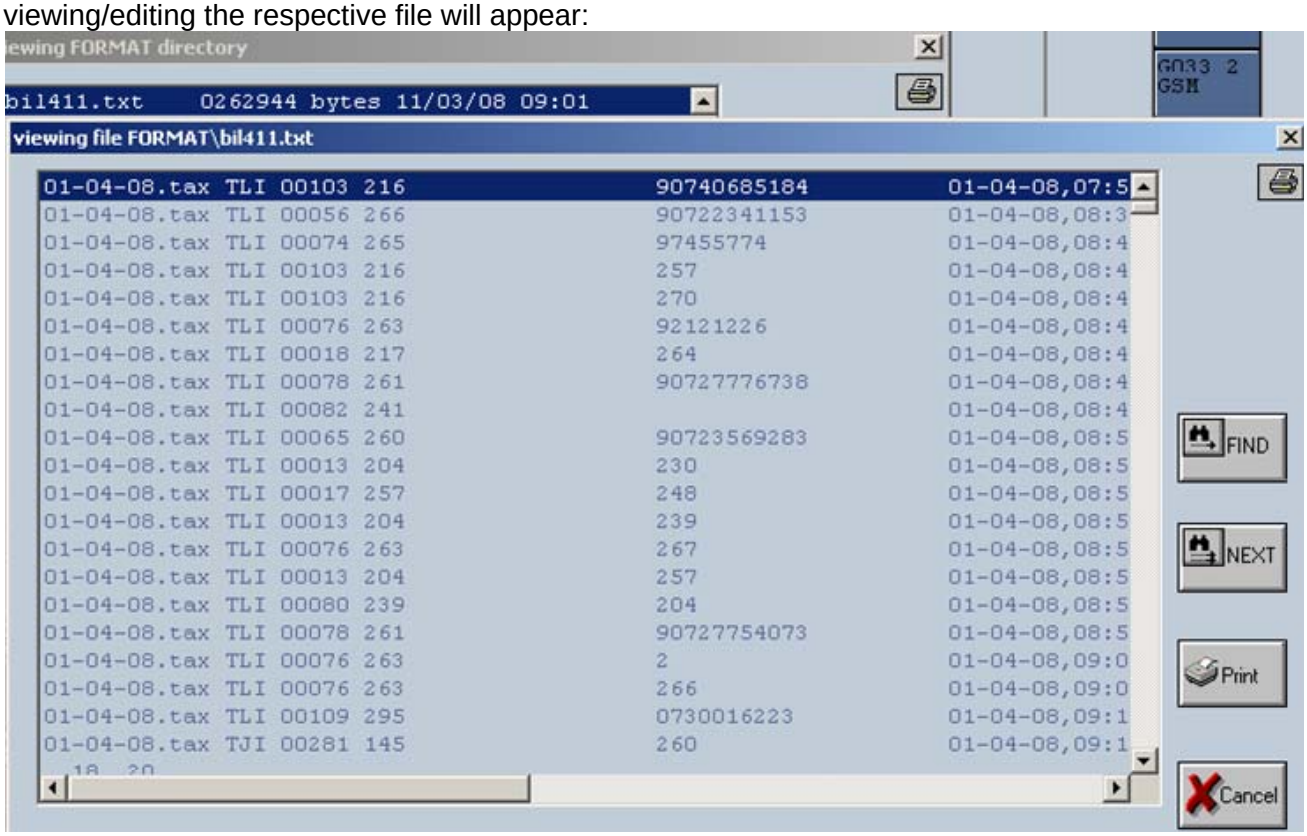
This command is a facility for viewing text files located in "FORMAT" folder. These files are generated by “printing” data into files instead of to a printer, from different locations of the OAM program.



All files in the FORMAT folder are listed:

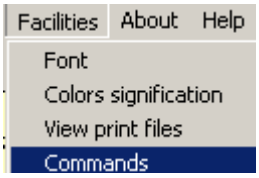


To see a file, select its name from the list and click the 'View' button. A window for viewing/editing the respective file will appear:



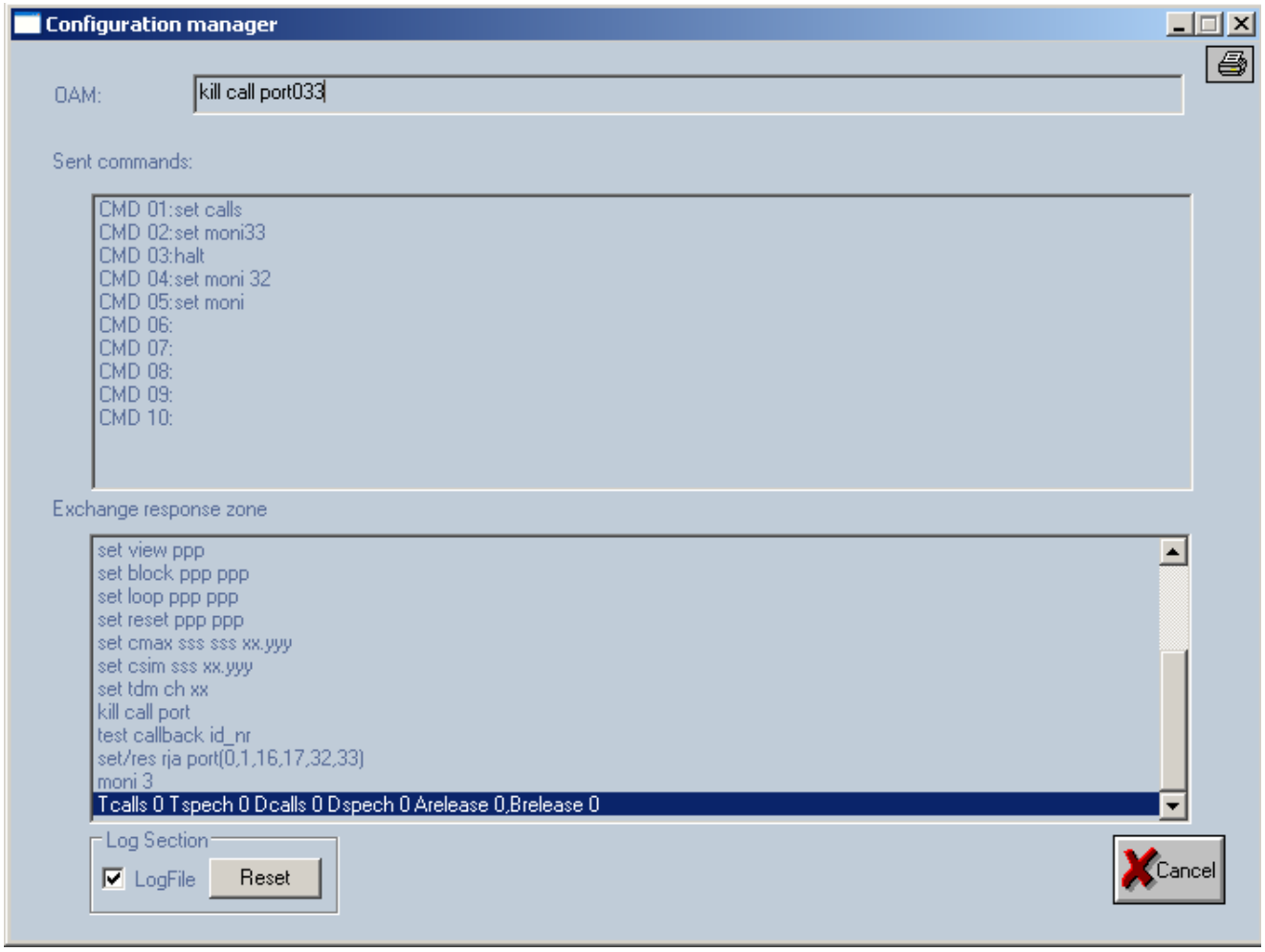
Now you can also perform searches by using the buttons marked 'FIND' and 'NEXT' or you can send to the printer a listing of the file.

2.3.5 FACILITIES => COMMANDS



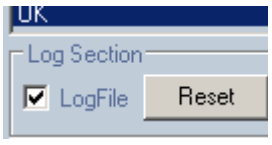
Use this option when you want to entering direct commands for the Topex gateway.

The window called “Configuration manager“ shows up:



On top, the field “OAM” is where you enter your commands.

The upper pane, “Sent commands”, shows the commands that you have issued, indexed form top downwards: CMD 01, CMD 02, etc, while the lower pane is called “Exchange response zone”, where you can see the answers (response) of the gateway to your commands.

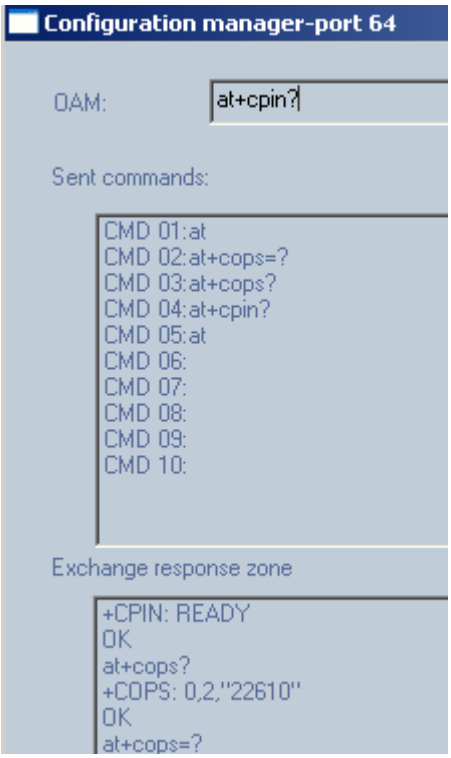


At the bottom left corner of the Commands window you have a **Log** box.

Log Section for “COMMAND” option - if you check LogFile box, a log file will be generated. The log file is always named "**command.txt**".

The option "Reset" is used to reinitialize the file used for saving the log.

The command window is useful for several purposes:



- Direct conversation with a mobile module – you can send them AT commands;
- Set PIN codes, display the number of calls, to configure the algorithms for SIM selection, etc.
- Start or stop the debug process;
- Display the statistics over the number of calls;
- Kill a current call on a port;

AT Commands for "COMMAND" option - if you want the mobile module to accept AT commands you must first place it in the monitoring mode.

Thus the first command in a series of AT commands will always be "**set moni xxx**", where "xxx" is the number of the respective mobile port (as it appears on the "gwconfig" panel).

The TOPEX equipment must send back the confirmation message "**moni xxx**", where 'xxx' is the position of the respective mobile port.

When the mobile port couldn't be placed into a monitoring state the following warning message is received: "**res moni**". If you receive this message (in the section "Exchange response zone") then you must type first "res moni' and after that "set moni xxx". After you receive the confirmation from the gateway, you can send AT commands to the mobile modules of the gateway. For instance, at+cops=? will show all the mobile networks that are active in your area.

Some of these AT commands are described in the table in **Appendix 2**.

There are also additional commands that are interpreted by the application running on the TOPEX VOXI gateway

To see the list of available commands, you must send the command "help" to the gateway.

You will get a response as shown:

The list of available additional commands is explained in **Appendix 3**.

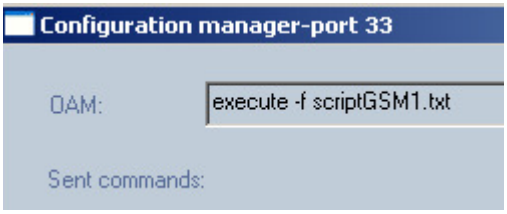
2.3.6 COMMANDS APPLICATIONS

In this paragraph there are shown some applications of the “Commands” feature:

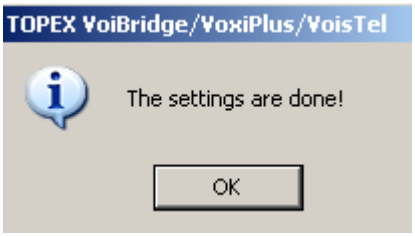
Execute AT commands

In "Configuration manager" window the user is also able to execute AT commands on all GSM modules. The execution of these commands is based on a script file which is given as command parameter. This file must be placed in the same folder with "gwconfig" OAM application. In this file you can enter a batch of AT commands to be executed over the GSM modules. The file must include the AT commands you want to be to be executed over the GSM modules, the expected response, the time delay to wait the appropriate response and the option to save the responses into a file.

The command to start this process is "*execute -f xxxxx.txt*" for all GSM modules (where "xxxxx.txt" is the script file), or "*execute -f xxxxx.txt -d GSM*" to start the process just for the modules belonging to the "GSM" direction. AT commands are read from the "xxxxx.txt" file



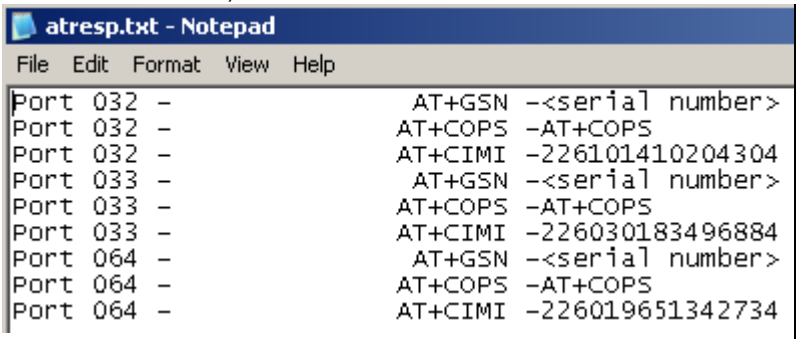
A window called "**AT Commands Confirmation**" is displayed for monitoring the process and the message “The settings are done!” is shown when the execution of the script has ended.



In order to clarify this facility, below is shown a commented example of a script file:

at+gsn
OK
10
1
at+cimi
OK
10
1

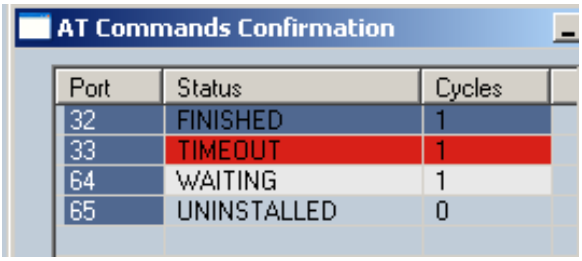
Each command is grouped as four lines:
1. The AT command to be sent to GSM modules; (in the example above these commands are "at+gsn" for IMEI code and "at+cimi" for IMSI code).
2. response wanted for the AT command; (in both cases the expected response is OK; the expected value is included in responses before the "OK" text).
3. time to wait for the expected response; the next field is a value (0,1 or 2) that has two meanings: the first significance (for a value of 0 or 1) is the time delay to wait for the response at an AT command. If the response comes in the specified interval then the process continues immediately. The second significance, if the value read from the file is 2, is this: the application waits all the specified delay and the last message received from the GSM module is saved into a file.
4. an option about the saving of the answer into a file; a 0 value means no savings are done, while a value 1 means answers will be saved in a file.



The name of this file is "atresp.txt"; If the value for the option is 2 then the last message received during the waiting time will be saved in "atresp.txt".

The option value 2 can be used to verify the credit for prepaid SIM cards.

Notes:
- before sending any AT commands the application starts the monitoring process (the same as selecting "Status monitoring"). This is done because the application needs to know which cards are in alarm and which ports are in alarm or in a call.
- this interrogation is performed just for the SIM cards currently registered. Because AT commands can't be sent to GSM modules which are in alarm, those modules will be skipped in the process. For GSM modules where a conversation is running, the commands are sent after the call is dropped. When the current interrogation for a module has reached the last port, then the process will be restarted for the skipped modules. The total number of restart cycles is 2.
- after a new interrogation process is started (when an "execute" command is performed) the file "atresp.txt" will be overwritten with the new response data.



The first column representing the port position. The second column displays the current "Status" of executing AT commands over that port.

The third (rightmost) column shows the number of interrogation cycles. The maximum number of restart cycles is 2.

Possible values for “Status” are:
- WAITING - AT commands are not yet sent to the port; (light blue color background)
- FINISHED - AT commands have been successfully sent to the port; (dark blue background)
- CONNECTED – the port was found in a call at the time when it was supposed to be placed under the process of executing AT commands; (red background). Call is dropped. AT commands are sent.
- PORT ERROR- the port was found in alarm at the time when it was supposed to be placed under the process of executing AT commands; (red background). The port will be interrogated in the next cycle.
- TIMEOUT- there was a timeout for receiving responses for an AT command; (red background). The port will be interrogated in the next cycle
- CARD ERROR- the card to which the port belongs was in alarm when the application was tried to execute AT commands for that port; (red background). This port will NOT be interrogated in the next cycle.
- UNINSTALLED- the port is not placed on any direction or the INST category is not given; (grey background). The uninstalled port will NOT be interrogated in the next cycle.

Voice Tests

Voice Quality Tests - in the same "Configuration manager" window the user can to perform tests over the quality of voice on two GSM modules. In order to make those tests, the user must first define the ports on which he wants to make the tests.

This selection is done in the "Callback Table":
<identity (number) to test quality of voice> 1 <code 1> <code 2>
The first field of the line is an arbitrary number used to start the test of the quality of voice. The second field must be "1" (callback functionality). The last two fields are containing the prefix used in routing table, the GSM port number and the number to call.
<code1>=<prefix>+032+0744000001 (GSM port position - 032, number to call -*0744000001*)
<code2>=<prefix>+033+0744000002 (GSM port position - 033, number to call -*0744000002*)

The Topex VOIX box will call the GSM module on first port, and following its answer, it will connect the two, allowing you to check the quality of voice for the respective connection.

For example:

9999 1 720320720000000 720330740000000 - the identity used for test is "9999". The prefix used in routing table for the test service is "72". The first GSM port is 32 and the second is 33, the first number to be called is "0720000000" and the second one is "0740000000".

In routing table must you must have a record to route "72" (the test prefix) to SERV ("Action" = SERV) and destination "11".

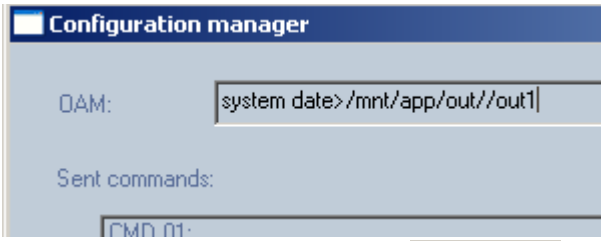
The command to start the voice test is performed from "Configuration manager":
test callbqack 9999

When the last command is executed, the gateway calls the first number 0720000000 from port 32. When the calls is answered at destination, the second part of the test is done. The phone number 0740000000 is dialed and the two are connected.
This way you can connect two by two all of the GSM ports of the Topex box and test the quality of the voice on the respective connections.

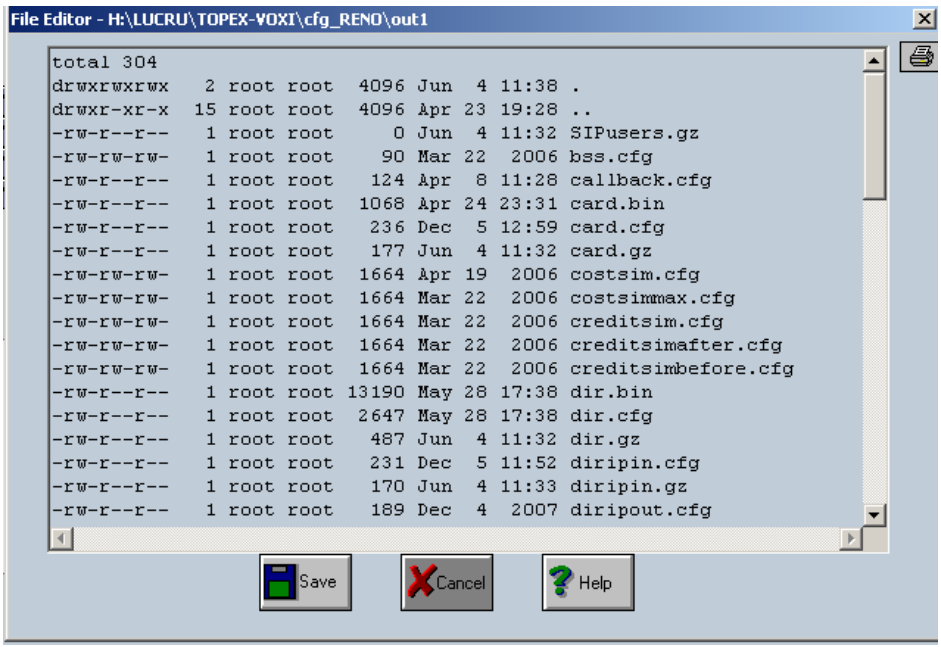
Linux commands

You can type in the "Configuration manager" LINUX-like commands that will be sent to the connected VOIX box. This way you can execute Linux commands from the OAM program, there is no need to use SSH or the serial connection.

The syntax must always begin with "system" characters. After a separation character (space character) a LINUX command may follow. By pressing ">" character after a command the software automatically adds the destination file for the command output. This destination is by default "/mnt/app/out/out".

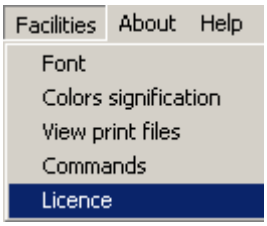


After such a command the user may analyze the output file by selecting the "Output File" icon command ("") from the toolbar on top of the OAM screen.



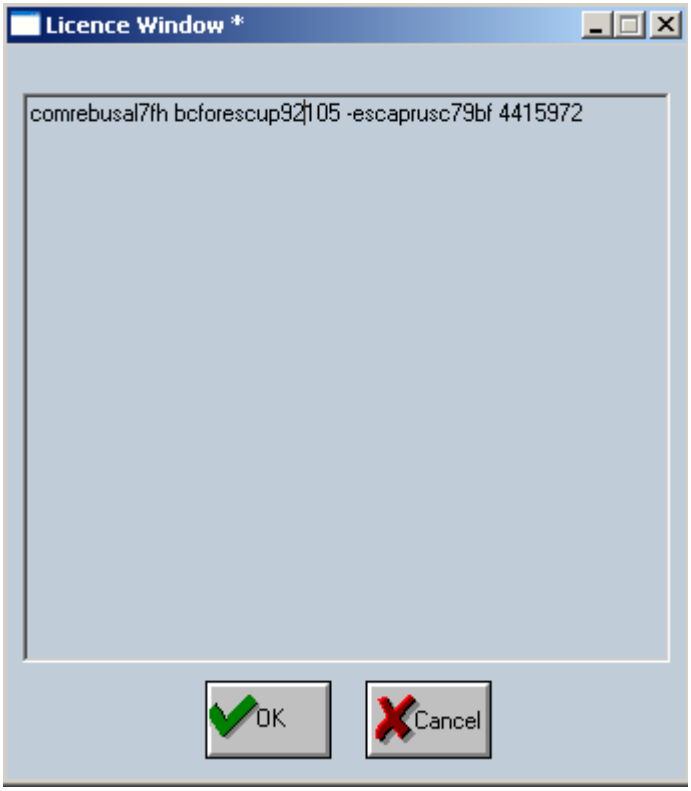
2.3.7 FACILITIES => License

This last option of “Facilities” menu is used to send a license message to the Topex gateway. The purpose of the "license" feature is to implement control over the time-limited application running on the gateway. This limited application will serve incoming calls only for a specific number of days. After the expiration of this period, the application will no longer process the calls.

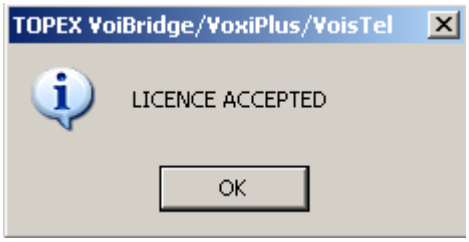


A license message is sent by Topex company, message that is unique for each gateway (it can't be used with another equipment).

The license-enabling message consists of several lines. This message can be received by the customer as an e-mail message. The user copies the message into the "License" window and it send to the gateway.



The gateway should answer with the confirmation message: "**LICENCE ACCEPTED**".



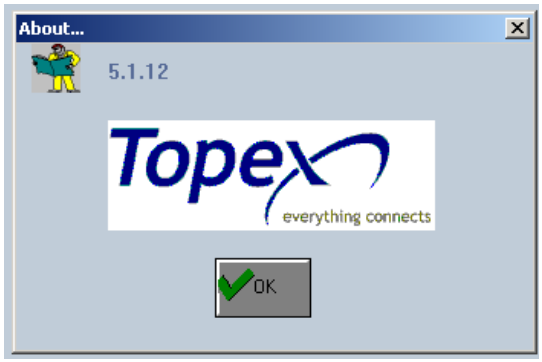
If the license message is not valid, the answer will be "**LICENCE REJECTED**".

If the license is accepted then the number of working day is displayed in the status bar.

2.3.8 About

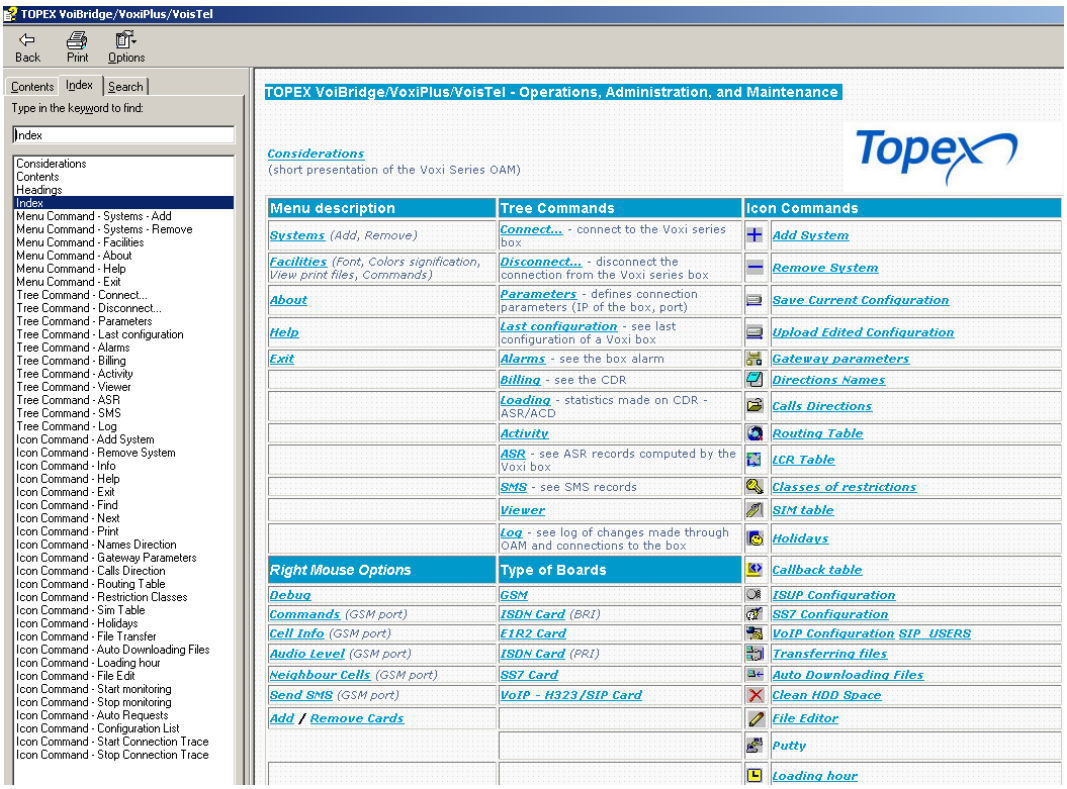
If you select "About" the window shown below will appear in order to tell you what version of OAM application software you have installed on your PC.

Select OK to close this windows



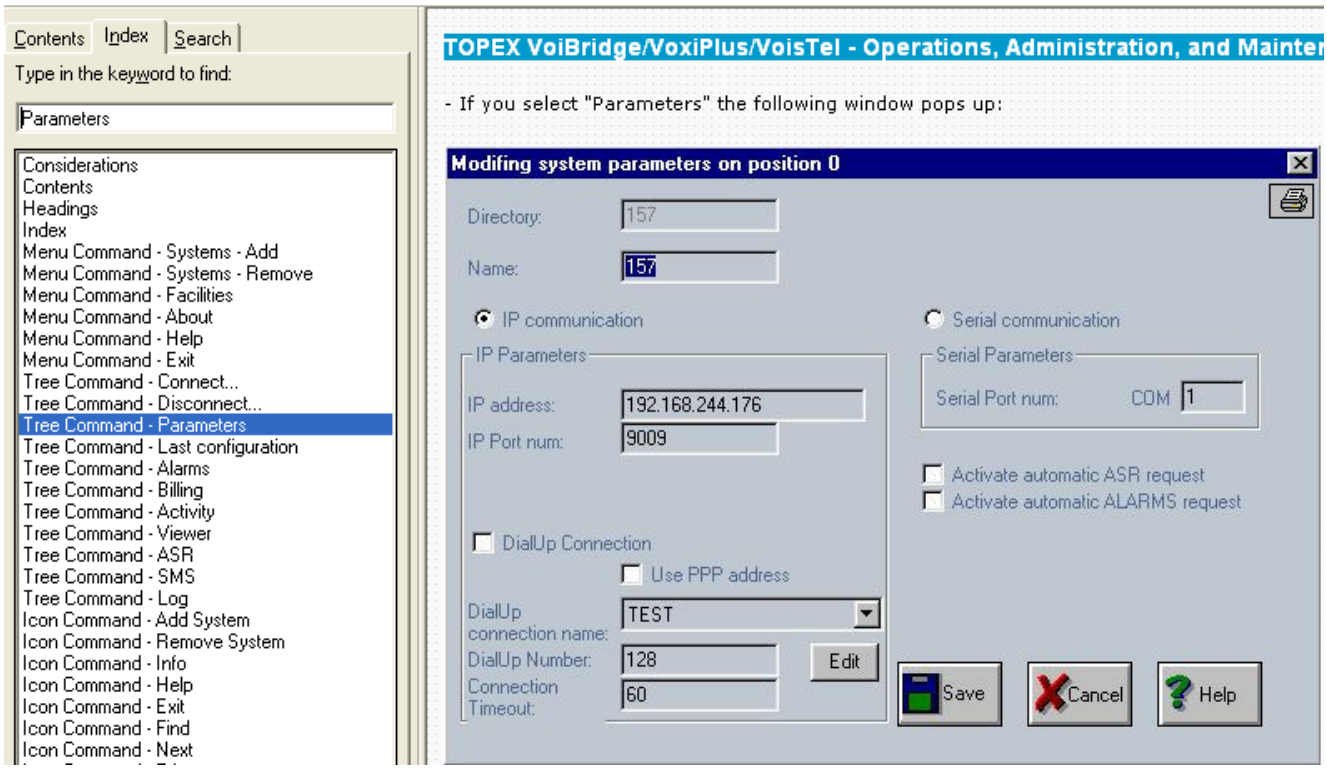
2.3.9 Help

This is the command for showing this manual in electronic format:



The left pane of the help window shows an index of pre-defined keywords where you can search for the item (general considerations, Menu command, Tree command Icon command) you want to learn about.

The right pane offers detailed pages structured as "Menu description" (Systems, Facilities and so on) and respectively "Tree Commands" (Connect, Disconnect, Alarms, Billing etc) and "Icon Commands" ("Gateway Parameters", "Directions Names", "Call Directions", "Routing Table", "LCR Table" and so on). The items colored in blue have active links to the corresponding section.



The left pane of the help screen also include the search facilities. You click on the "Search" tab and enter the word or the expression you want to look for, for example <<billing files>>, then click "List Topics" to start the search: The help program will display all the occurrences it has found, together with the name of the help page where they are located and the rank (1,2, 3 etc).

Please note that, according to the general rules for text search, if you enter several words in the "Search field", the help program will look for occurrences of EITHER word. This is like telling the help program <<search for 'billing" OR for 'files">>.

If you want to be more specific, you should select the boolean operator AND instead of OR. For instance, if you want to know about the billing files, you must search for "billing files", using quotes. In this case, the help program will display only the occurrences where ALL the words inside the quotes are found together.

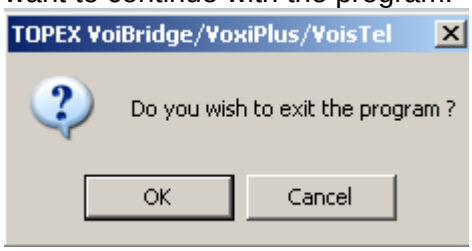
You select one of the occurrences and click "Display" to show the respective help page in the right pane. The words or expressions you look for will be highlighted (marked with reverse video). If you want to search further inside the help page, press Ctrl F and a "Find" window will show up. You type the expression to search in the field "Find what", You may select additional options: "Match whole word only" and "Match case" if you want to narrow the search. You must specify the direction of the search inside the page, Up or Down. "Find Next" shows the following occurrence of the word you want to search.

The Search function of the help program remembers the words you entered, even if you close down the help. When you run "Help" next time the same word or expression will be displayed.

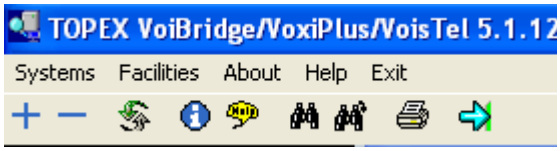
The window of each command now has a "Help" button of its own. This means that you no longer need to issue the "Help" command and then search or the item of interest. When you run a command, for instance "Add System", you just click its "Help" button and the respective help window will be displayed, showing you information about that particular command.

2.3.10 Exit

If you select "Exit" the following window will be displayed, asking you if you want or not to leave the program. Select "OK" if you really want to exit the OAM program or select "Cancel" if you want to continue with the program.



2.4 *Buttons (icons) description*



The toolbar with button icons is located under the menu bar.

Some buttons (Add/Remove System, About, Help, etc) are equivalent to options from the Menu or the System screens. This button toolbar is a dynamic bar, which changes aspect according to the state of the OAM program or with certain actions performed by the user.

The OAM software may be in one of these three states:

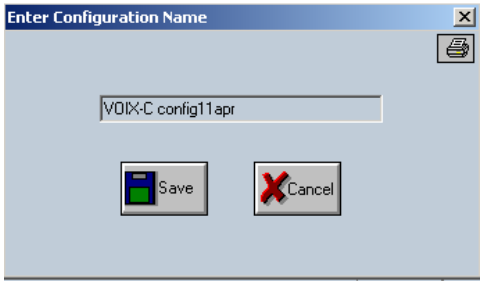
1. One TOPEX VOXI gateway connected - the last column of the status bar is colored in light blue, indicating a connected system  . In this case all option icons are available. Since the toolbar is very long, below it is shown in two pieces

2. You use “Local viewer” to see the files for a system, gateway connected - the last column of the status bar is colored in light red, indicating no connection:  . Only the following icons are available:

3. None of the above situations (no system connected):

The following table explains the icons from the button bar:

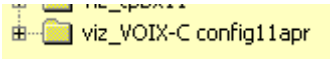
2.4.1  - Save current configuration



The "Save Current Configuration" (icon "  ") may be use only in a state of connection between the OAM application "gwconfig" and a Topex gateway.

When you press it, the "Enter Configuration Name" window is displayed, allowing you to type a name for the configuration you want to save.

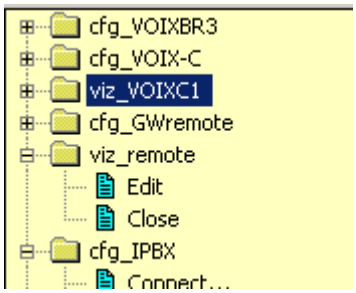
Click the "**Save**" button in order to validate the name.



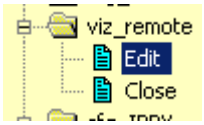
Now a new item will appears in the tree-like window located to the left side of the application panel.

The name typed in the "Enter Configuration Name" will be shown, starting with the characters "viz". This means the configuration can be saved and edited on the local computer, even without connection to a remote TOPEX equipment.

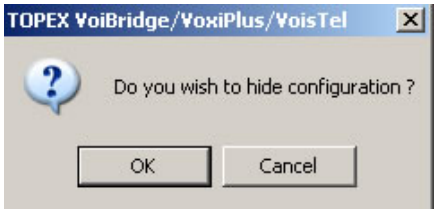
Types of Configuration - "viz" and "cfg"



The OAM program makes use of two types of configurations.
The prefix "**viz**" indicates a configuration saved as explained above, which is stored on your local computer.
The prefix "**cfg**" before the identification name of gateway refers to configurations of gateways to which OAM is able to make connections. These configurations are stored in directories corresponding to the remote gateways to which you connect.



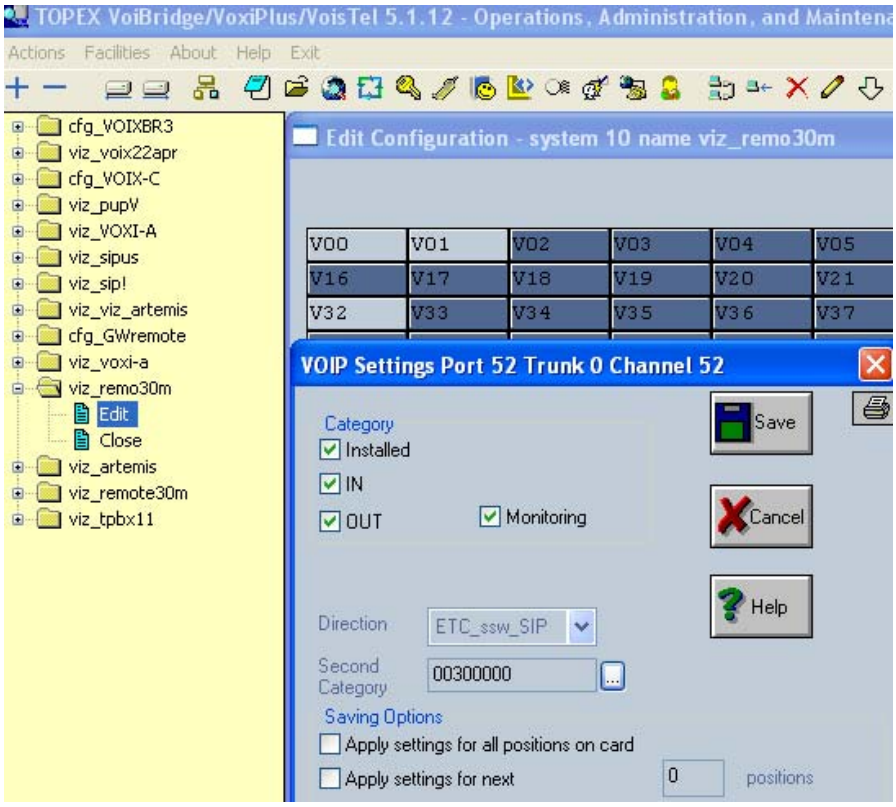
For a saved configuration (viz) there are just two possible options:
- "**Edit**" - this option will display the saved configuration in the same manner as for a connected gateway.
- "**Close**" - this is the option to close the "Edit" configuration mode.
When you pres close, the program ask if you want to hide the configuration.



Click "OK" to hide the configuration window.

You can make changes over this stored configuration.
All settings are preserved when the you choose the "Close" option.

Editing a saved configuration (viz_remo30m in this example):



2.4.2  - Upload edited configuration

You can use this option in a state of connection between the OAM application and a TOPEX gateway, to upload a saved (and edited) configuration to the remote system.

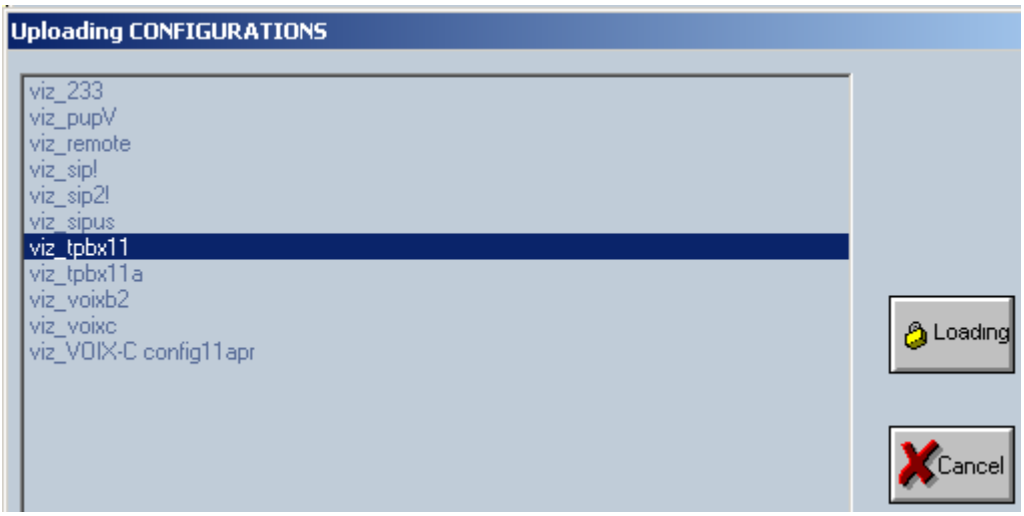
- Press the icon "  " and launch the uploading process;

A box called "Access password" shows up.

Uploading a new configuration can stop the operation of the gateway, so the program asks you for confirmation, before performing this.

You must enter the correct password and press OK in order to continue the uploading process. If you want to abort, press Cancel. The default password for this option is "topex".

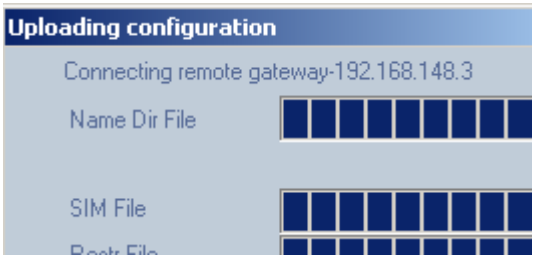
- Select the configuration you want to be uploaded. All available configurations are displayed in a list.



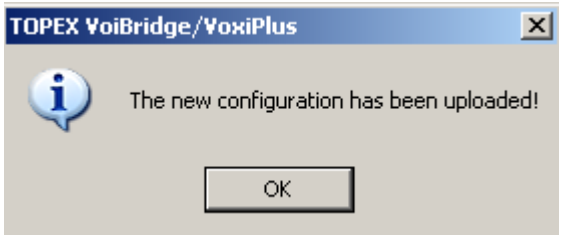
- Select a configuration from the list, then press the button "Loading" to upload it, or “Cancel” if you want to abort.

A window named “uploading configuration” shows up, telling you that it connects to the remote Topex equipment. The IP of the remote equipment is also shown.

The uploading of each configuration file is indicated by a progress bar.



- When the uploading process ends successfully, a confirmation message is displayed:



The OAM software will display on the screen the new configuration. This configuration is used also by the main application running on the TOPEX equipment.

2.4.3  - Gateway parameters

In a connection state, the "Gateway Parameters" icon can be used for editing several parameters of the remote TOPEX equipment. (Not to be confused with the “Parameters” command of the tree like structure, which changes the parameters of the OAM application and may be used even when the remote VOIX box is not connected to your PC):

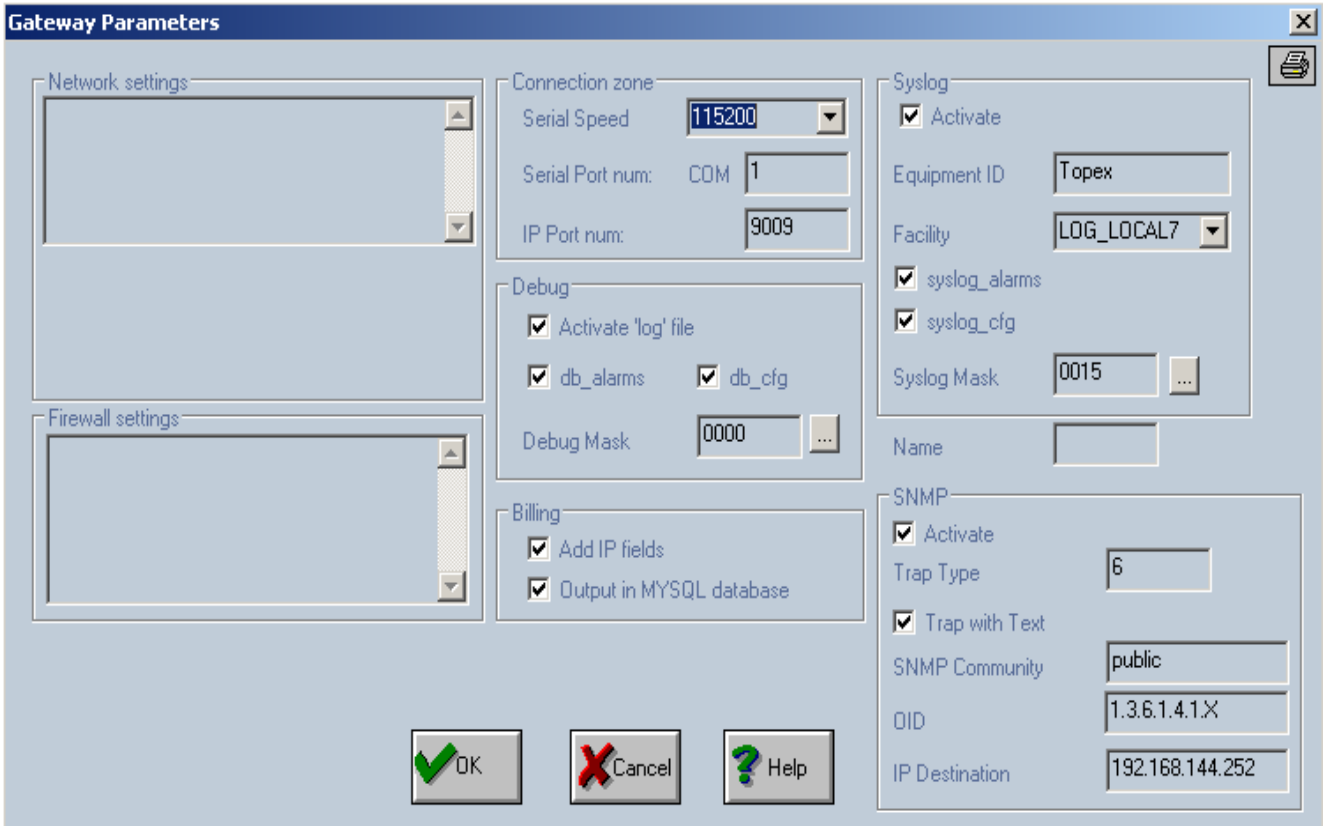


Figure 3 – Gateway parameters window

The gateway parameters are grouped into six areas:

- Network and Firewall settings (not used in the current implementation);
- Connection zone;
- Debug;
- Billing;
- Syslog;
- SNMP.

A. Network settings – this area allows you to change network settings for the TOPEX VoiBridge equipment:

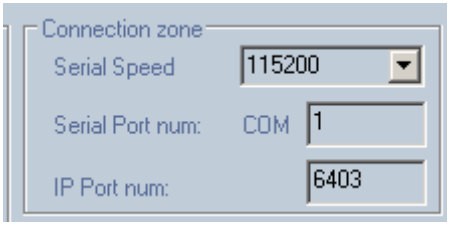
IP ADDR, NETMASK and GATEWAY don’t appear in the “Network settings” window. The “Network settings” window is non-editable.

- **IP address** of the gateway (the most important parameter) – in the image the record started by “IP ADDR”;
- **Netmask** (line that starts with "NETMASK");
- **Gateway address** (line started by "GATEWAY").

After you change even one parameter from the "Network settings" the gateway must be restarted (press the "" button) in order to use the new value of the parameter.

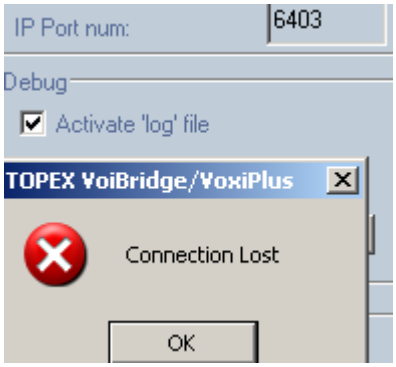
B. Connection zone - these settings are related to the serial connectivity and IP connectivity of the TOPEX equipment with the "gwconfig" software.

Here you can set parameters for the TOPEX VoiBridge system.
Of course, the parameters set for the gateway must correspond to those established for the OAM program!



The available settings are:

- **Serial Speed** - the value used for serial communication between the program running on the PC and the Topex equipment. Allowed values are 0 (it means unused port and must be set when you want to is intended to install a dialup server on the gateway), 4800, 9600, 19200, 38400, 57600 and 115200 (default value);
- **Serial Port num** – specifies the number of the serial port on the gateway to be used for serial connection with OAM computer, such as COM1 in the example; *Note that in case of VOXI equipments the serial connection **cannot** be used for OAM, it is reserved for direct link by means of the Linux console.*
- **IP Port num** – the port used for IP connectivity; the default value is 9009.

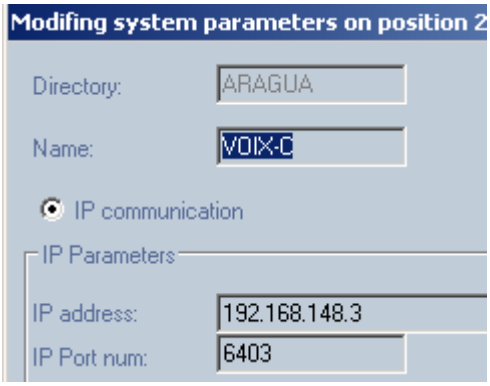


If this value is changed from OAM software, then ***the connection will be immediately lost***, as you see in this example, where the number of the port for IP connection has been changed to the value 6403.

The loss of IP connection is normal, since now the remote gateway and the OAM program are set to different IP port numbers!

To restore connection, you must go to the “Modify System parameters” window of the OAM program and change the “IP Port num” accordingly.

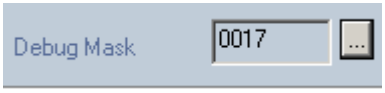
Then you may connect again form the OAM program to the remote Topex equipment, both using the new value for the IP port number!



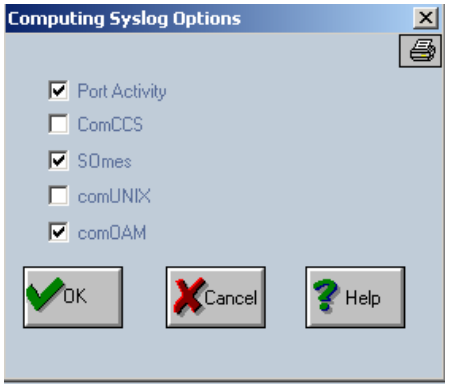
C. Debug - activates several options for generating special log files, useful for debug (diagnosing and repair) operations. These debug files have extension "log" and the name corresponding to the current day (format day-month-year: dd-mm-yy).

- **Activate 'log' file** - when the option is checked the debug messages will be generated in the debug log of the TOPEX equipment. These debug files have extension "log" and the name corresponding to the current day (format day-month-year: dd-mm-yy);
- **db_alarms** - when the option is checked the alarm messages will be shown in the debug log of the TOPEX equipment. A log file is created, if this was enabled from the "Alerts" icon "" of the toolbar. Those debug file have the extension "log" and the name corresponding to the current day (format day-month-year: dd-mm-yy);
- **db_cfg** - when the option is checked any changes on the configuration of the Topex gateway will be shown in the gateway debug log;

Debug Mask - The option sets the debug mask (four bytes) that establishes for which items the debug logs will be generated.



The items that may be recorded in the logs are: "Port Activity", "ComCCS", "SOmes", "comUNIX" and "comOAM". To edit the "Debug Mask" field, the button "..." must be pressed. Then the "Computing Debug Options" window will be displayed.



Here you just check the boxes for the options you need to generate debug information.

"**Port Activity**" - generate debug records for all ports of the TOPEX VoiBridge;
"**ComCCS**" – option currently unused

"**SOmes**" - debug the main application running on the gateway;
"**comUNIX**" - debug the commands that are received from the Web server (using an Unix socket);
"**comOAM**" is used to debug the communication with OAM program.

See below samples of messages from a debug log of a VOXI box:

If option "db_alarms" was set then messages preceded by "RUN" will appear in the debug file:
RUN Com Err cd f cat 43 (this is an alarm message on card 'f' which is card number 15 from the Topex box)
RUN Com Err cd 10 cat 4003 (this is an alarm message on card '10' which is card number 16 from the VOIX box)
RUN NLIS 4 (LIS error on card 32)
RUN NLFA 4 (LFA error on card 32)
RUN NAIS 4 (AIS error on card 32)
RUN NRJA 4 (RJA error on card 32)

If option "db_cfg" was set then messages preceded by "CFG" will appear in the debug file:
CFG Open for read ../cfg/trafic.cfg
CFG Alarm run 0 ASRalarm 80 timeasr 12(min) nr min calls 1 alarmmask 0 timetest 0
CFG Save ASR at 30 min
CFG Debug 1 Pstart 0 Pstop 320 SaveFile 1
CFG Trafic run 0
CFG Trafic idle time 5
CFG Trafic seize time 30
CFG GEN TRAFFIC CALL 0 1234567891108 120 port 257 mod 0. These messages contains operations performed over the configuration files.

If option "Port Activity" is validated in the creation of debug file then messages starting with "OAM" will appear in the debug file:
00257 7235150 pid 1 E1CCSidle ACTIV channel 1
00257 7235150 Digits 1234567891108 Ident 120
00257 7235200 Route N 123456 ... NT 123456 ... RxT 9000
00257 7235200 Dialdig M_CALL len 30
00258 7235200 pid 2 E1CCSidle CALL mod 1 mycat 540f b_pid 1 prtb 101 param 0
00258 7235200 Digits Ident 120
00257 7235250 Dialdig ACK_CALL->Dialdest
00257 7250250 Dialdest TimeOut
00258 7250250 E1CCSackout TimeOut|CANCEL ->BUSY. In this case messages were generated for ports 257 and 258.

If option "SOmes" was validated in the creation of debug mask then messages starting with "GEN" will appear in the debug file:
GEN COADA 6 TRI 145 TRA 5 CPS 508 CPL 256 DTMF 0 R2 0 CH 0 CMD 0
GEN Local Time Wday 03 Mday 07 Date 07-04-04 Hour 11:19:10 SysDay 2
GEN Run60sec 6872131 us Tcalls 15 Tspech 0 Dcalls 1 Dspech 0 Arelease 1,Brelease 0
GEN Local Time Wday 03 Mday 07 Date 07-04-04 Hour 11:20:11 SysDay 2
GEN Run60sec 6465996 us Tcalls 15 Tspech 0 Dcalls 1 Dspech 0 Arelease 1,Brelease 0
GEN COADA 5 TRI 145 TRA 5 CPS 508 CPL 256 DTMF 0 R2 0 CH 0 CMD 0

If option "comOAM" is validated in the creation of debug mask then messages starting with "OAM" will appear in the debug file:
OAM ADD CARD 16 CAT 4003 (this is a message which indicates that the card 16 was added to the gateway. Value "4003" is a category which describes the type of card.)

D. Billing – This section contains two fields related to billing, which are activated by checking the boxes near the fields:

Add IP fields – if it is checked, the application will automatically add the necessary IP fields (for instance the signaling IP field);

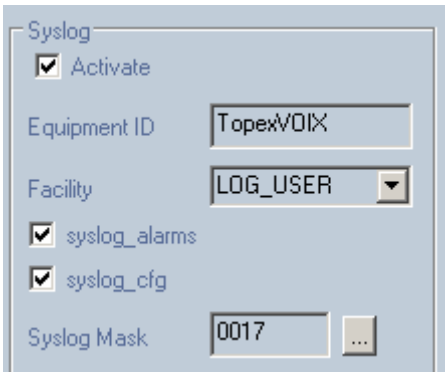
- **Output in MYSQL database** – if it is checked, thus option determines the output of equipment billing files in a MySQL database installed on a different server.

E. Syslog – if you activate this option (by checking the “Activate” box), the debug messages are sent to a selected IP address.

This Syslog is an UNIX-like feature that allows logging of debug messages and sending them to a remote machine;

The destination IP address is written on target in the file "/etc/syslog.conf". All logged messages will contain the identification of the TOPEX equipment.

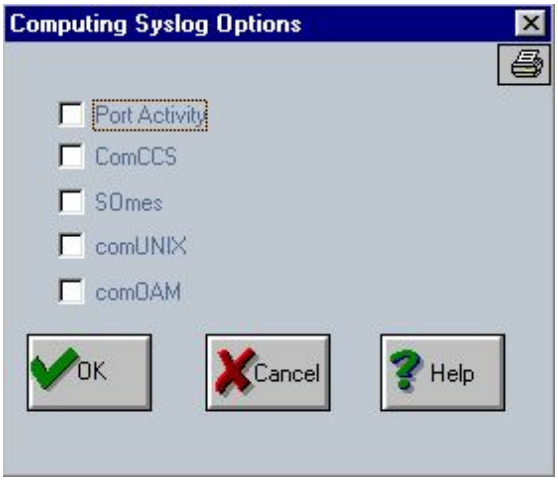
You must enter this identification in the field "Equipment ID".



- "**Facility**" list allows you to select the type of syslog messages. You may choose: "LOG_USER", or one of "LOG_LOCAL0" to "LOG_LOCAL7";
- **syslog_alarms** - when the option is checked the alarm messages are sent to the syslog destination;
- **syslog_cfg** - when the option is checked then any changes made upon the configuration of the gateway will be also sent to the syslog destination;
- **Syslog mask** – similar to Debug mask, this option establishes about what items ("Port Activity", "ComCCS", "SOmes", "comUNIX" and "comOAM") will be generated syslog message.

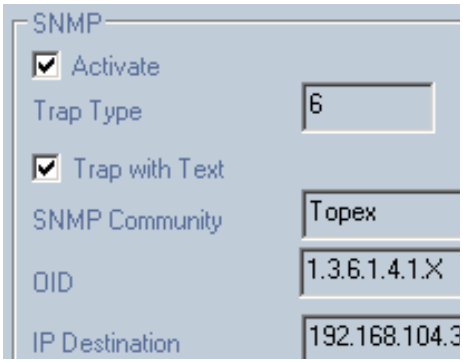
It works similar to debug mask.

Click on "..." button and the next window shows up, allowing you to select the items which will be included in the syslog.



F. SNMP - Simple Network Management Protocol is used to monitor the devices inside a network by means of "trap" messages that are issued each time a significant event has occurred.

When you activate this option (by checking the box "Activate") alarms can be sent as traps to a selected IP address ("IP destination"). The user can choose the type of the trap (default value is 6 - "Enterprise Specific"), the community name (default value is "public"). When an alarm appears the value sent is 0, and when it disappears the value sent is 1.
You can check the box "trap with Text" – then together with the trap containing the 0 / 1 value, a second trap is also sent, with the text specifying the alarm which occurred.



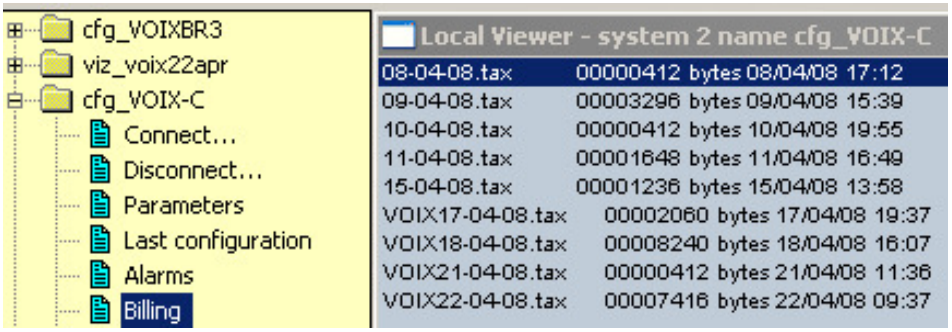
The **OID** of traps are derived from the specified OID ("1.3.6.1.4.1.x").

Name - this field is used as a prefix for the names of different files (alarm, billing, activity, log and ASR) created by the application running on the Topex gateway.

If you leave this field empty, then the name of the files is given by the current date and an extension, for example "12-04-08.tax" is the billing file created on the 12th April 2008).

If the "Name" field is filled, then the names of these files will be given by the name you have typed, concatenated with the current date (in format DD-MM-YY) and with the corresponding extension. For example if you type "Topex", the billing file created on 12-04-08 will be named "Topex12-04-08.tax".

If instead you type VOIX in the field “Name”, as shown above, the billing files created form this moment on (starting with the 17th April in the following example) will have this name as prefix, followed by the date, and the same extension “tax”, as you can see here:



2.4.4 - Directions names

Routing of calls is based upon dialing rules and DIRECTIONS (groups of trunks).

Such a “direction” includes one or several trunks or subscribers with similar routing characteristics.

Press icon for selecting the "**Direction names**” option and performing changes into the list of names of directions.

On the screen appears the window „**Define direction names**”, a list with the directions already defined on the TOPEX box.

Such pre-defined names for directions are: LOCAL, GSM, E1R2, ISND, MYVOIP, H323, SIP, MYVOIP, SENDCALL, SENDSMS, etc.

Depending of the gateway main application, there is a maximum of 20 directions (for simple VoIP to GSM gateways) or 250 directions (for VOIX equipments that act also as an IP-PBX).

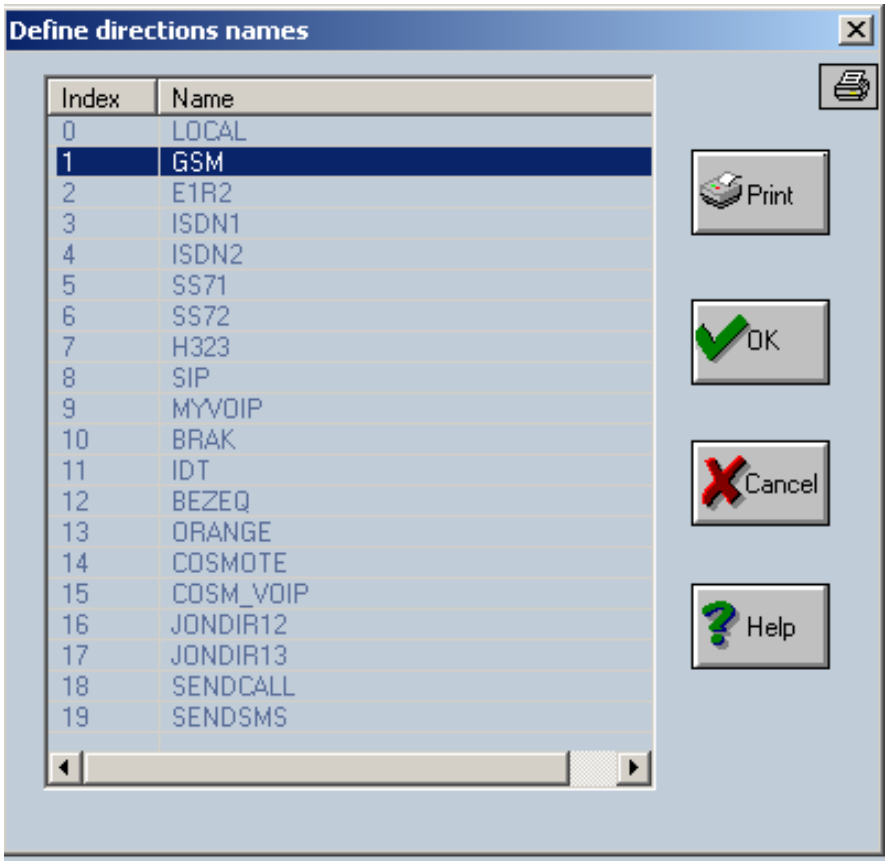


Figure 4 – Define / modify the names for the directions

These “Directions” are groups of inbound or outbound trunks that have common routing characteristics The maximum length for the directions names is 19 alphanumeric characters, which must not contain blanks (instead, underscore character is used for separation).

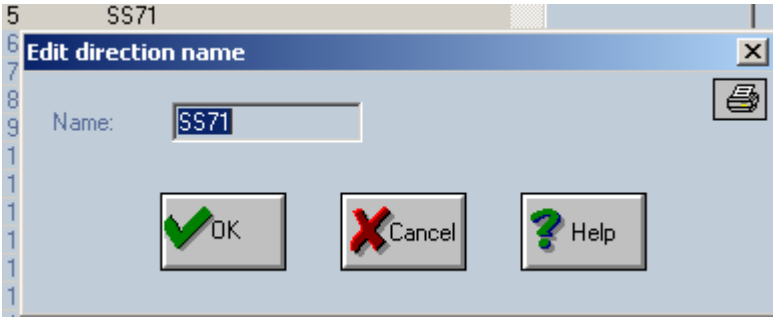
First, you define (or edit) the names of these “directions” and then you use the icon command “Define call directions” to modify the table with directions for calls, according to your requirements.

To ensure adequate routing of calls, you must assign one or more trunks or ports to each direction.

Edit a name of direction

Select from the list the name of the direction you want and double click the name.

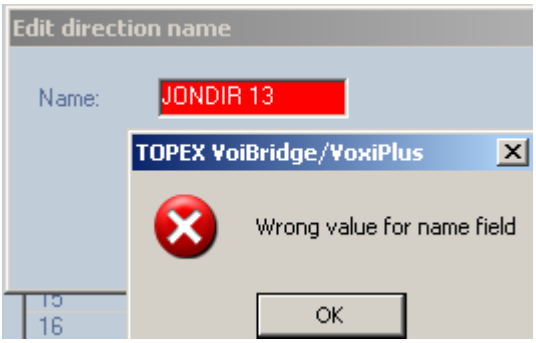
A window called “Edit direction name” appears, and the field Name holds the current name of the selected direction:



The name of each direction must be a unique identifier - an alphanumeric text (digits and letters) maximum 19 characters long.

If you type more characters, they will be ignored. The first character of the name must not be a digit.

In addition, the text that you enter for a direction name must NOT contain spaces (blanks). Otherwise, an error message will be displayed, like in this example, where the name included a blanc (space) character.



Warning: The generic direction used for the group of the VoIP channels – “**MYVOIP**” – must not be changed!

2.4.5  - Calls directions

- Click on the icon button "Calls direction" ("  ") to open up the window for defining and editing the table with directions for the calls. For VoiBridge there are a maximum of 20 directions (trunk groups), but for equipments with the IP-PBX feature the max number of trunks is 250.

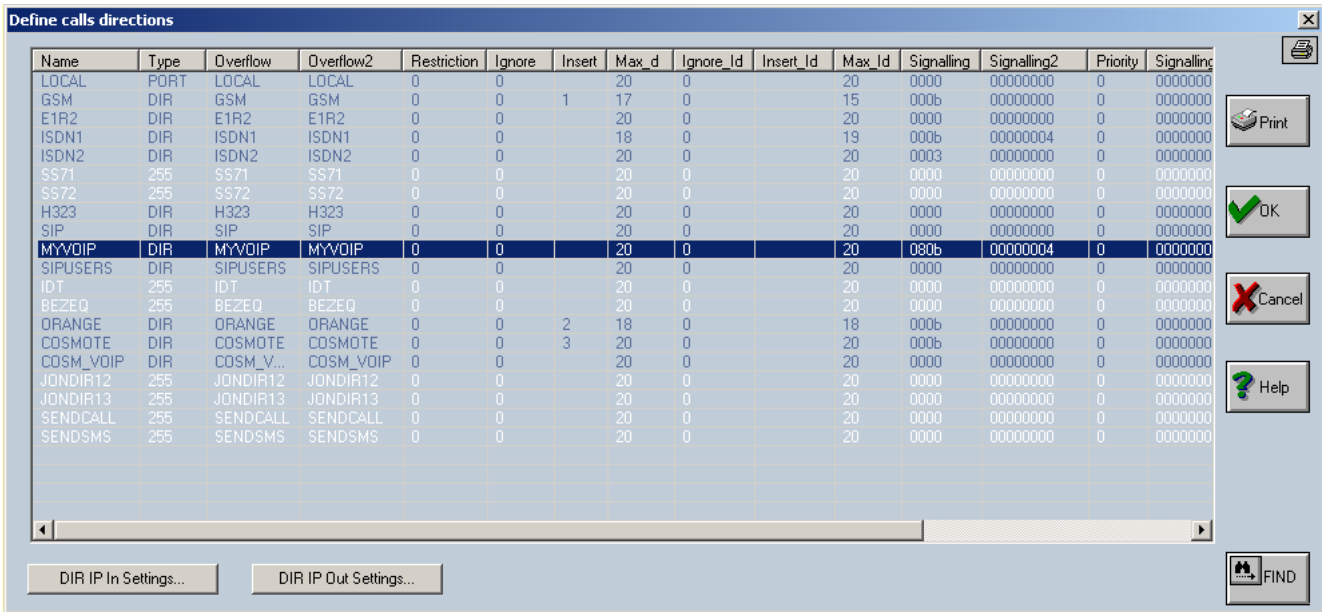
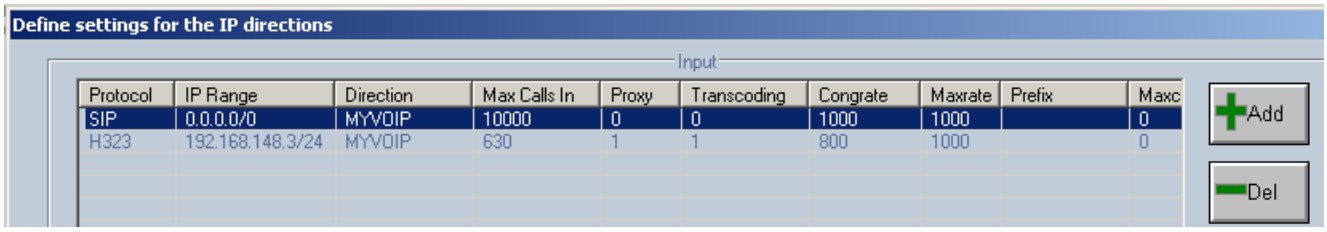


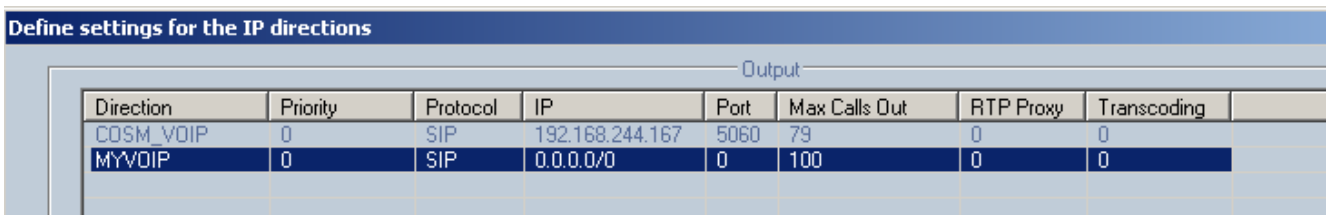
Figure 5 – Define calls directions name

At the bottom of this window there are two buttons named "**DIR IP In Settings...**" and "**DIR IP Out Settings**".

The first one is used to control the incoming access into the TOPEX equipment for VoIP calls:



The second one is used to establish the outgoing destinations for VoIP calls.



- To edit the parameters of a direction, double click on its name and the window "Edit call directions parameters" will show up:

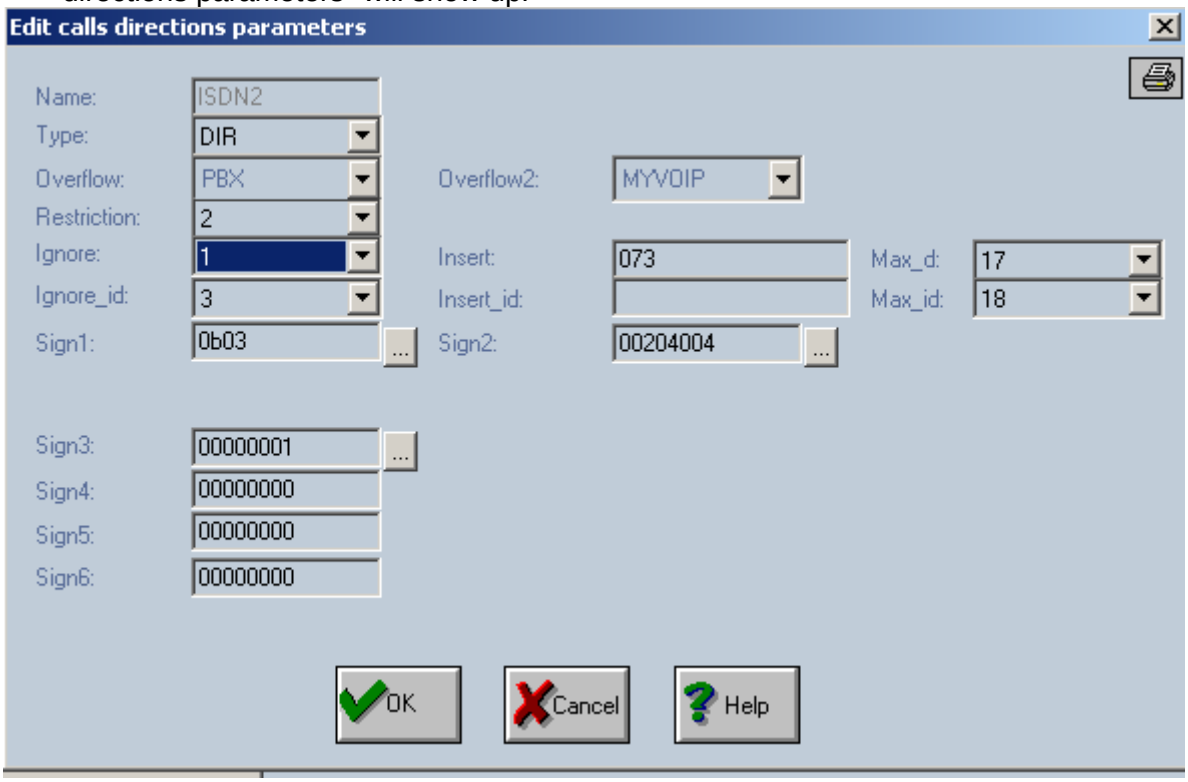


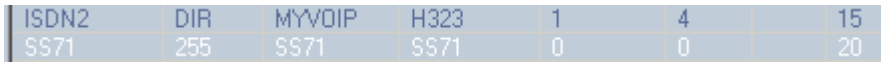
Figure 20 – Edit calls directions parameters

The parameters of a direction are:

- **Name** - the name of the direction. The field shows up in gray color, because it **cannot** be edited here, the name was already defined or modified in “Define direction names“;
- **Type** – it can be either PORT or DIR and specifies how that direction is addressed.

The associated list contains two strings "PORT"= local extension and "DIR"= trunk or group. The default value of "255" indicates that the respective direction is **ignored**;

When the "255" value is assigned to a direction, then the entire row of that direction will be colored in white instead of the normal blue color:



The name of the direction which has "255" assigned to the field "Type" will be displayed in **red color** in the list used to assign a direction for each kind of port.

It will also shown in red color in all places where the destination will be a direction name.

This is to remind you that you must properly assign (define) the respective direction.

- **Overflow** – The user can set up to three priorities for each routing rule, if the Topex gateway cannot route the call to the primary destination, it will try to route to the second or third destination, which are named “overflow directions” for the current direction. The calls will be re-routed to this overflow direction (SIPREG in the image above) when the current direction is unavailable (fully busy or out of service);

For this Overflow direction you may select a name of a defined direction from the corresponding drop list:

- **Overflow 2** - second overflow direction for the current direction. The calls will be re-routed to this second overflow direction when the first overflow direction becomes unavailable; you choose a name of a defined direction from the corresponding list;

For Overflow2 you can also select a name of a defined direction from the corresponding drop list:

- **Restriction** - the class of restriction applied to that direction; from the list a number from 0 to 19 can be selected;
- **Ignore** - specifies how many digits are “ignored” (dropped) from the numbers received on that direction; the first x digits of the received number will be ignored. The list contains a range of values from 0 (none ignored) up to 20 digits (all the digits are dropped). For example, if you set here 3 and the called number is 021334455, the Topex gateway will drop 021 and send 334455 as outgoing number;
- **Insert** - specifies what number of digits will be “inserted” (added) in the number received to that direction. The maximum allowed is 16 digits. If you don't want any digit to be inserted you must enter “---” for this field. For example, if you set here 02 and the called number is 998877, the Topex equipment will add 02 and send 02998877 as outgoing number

Note: If you set both Ignore and Insert, first the dropping operation is performed and then the adding of digits is performed.

- **Max_d** - maximum length of digits that may be dialed on that direction. When the maximum number is reached the system will automatically send out the call to routing analysis, without waiting to see if the caller part still sends digits. This option is especially useful when you define directions for which the number of digits to be dialed is precisely known (for example the phone numbers for certain GSM networks have exactly ten digits). Valid values are form 0 to 20 digits;
- **Ignore_id** – Same as Ignore and Insert, but applied to Identity instead of phone number. It drops from the identity of the caller (Caller ID) the number of digits you have specified; The maximum allowed is 20 figures. The Ignore command is performed before the Insert command. Valid values are form 0 to 20 digits;
- **Insert_id** - Adds to the Caller ID the specified figures; The maximum allowed is 16 digits;
- **Max_id** - The maximum number of digits from the Caller ID to be sent to the subscriber who has been called. Valid values are from zero up to 20 digits.

- **Signaling** - contains 4 digits in the 'xyzw' format with the following significance:

Digits	Purpose	
x (digit 1)	**Bit (from right to left)**	**Interpretation**
	bit 0 (mask 0x1000)	**DIRCHECKCALLBACK** - when this bit is set then for an incoming call on this direction the callback table will be analyzed with the received identity (caller identity)
	bit 3 (mask 0x8000)	**DIRMODULECDMA** - when this bit is set all GSM ports placed on the selected direction will be treated as CDMA modules
y (digit 2)	**bit 0 (mask 0x0100)**	**DIRTESTNET** (used in case of a direction that contains GSM modules) - when this bit is 1 (one) then the outgoing GSM module will be tested if it is registered
	bit 1 (mask 0x0200)	**DIRGOODASR** - when this bit is set then a RELEASE message is sent following a delay of 5 seconds when a congestion situation is encountered on GSM part. The call will wait on the specified time a free GSM resource.
	bit 2 (mask 0x0400)	**DIRCATCALL** - when this bit is set then all calls will be cut (stopped) on the GSM modules for which a reprogramming is necessary (for example when a SIM must be changed because of the algorithm used)
	bit 3 (mask 0x0800)	**Verify CLIR** - when this bit is set then the CLIR setting is verified each time after the CLIR setting is sent to a GSM module
z (digit 3)	Audio level (in range 0-7 dBm for the directions with GSM interfaces) where '0' is the biggest level value and '7' the lowest value	
w (digit 4)	**bit 0 (mask 0x0001)**	**Receive identity** - when is set identity is received on selected direction
	bit 1 (mask 0x0002)	**Send Identity** - when is set identity is sent on selected direction
	bit 2 (mask 0x0004)	**Load Balancing Algorithm** - establish the algorithm for changing SIM on GSM interfaces (when not set, the algorithm for minimal cost is used (SIM selection by time periods). When the bit is set, the "load balancing algorithm" is used (equal usage time for each SIM)
	bit 3 (mask 0x0008)	**Coupling the ring-back tone** - by setting it, you allow coupling of ring-back tone while dialing on the next link, before the called party answers

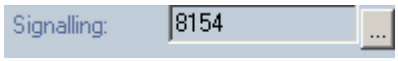
You can either input the individual values for the digits of “Signaling” or use a simpler means, to press the button "..." located left of the “Signaling” field.

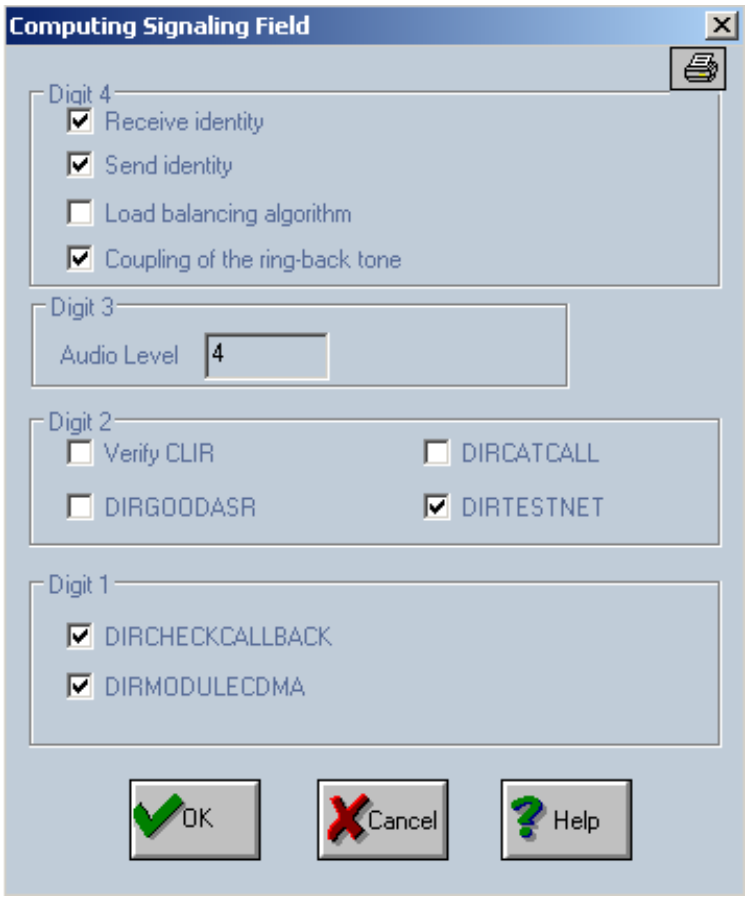
In this case a window called “Computing Signaling Field” shows up, assuring you an easy way of setting up the "Signaling" bits for you.

You just check the items you want and the program computes and inserts for you the individual bits. Thus you will have:

or

or



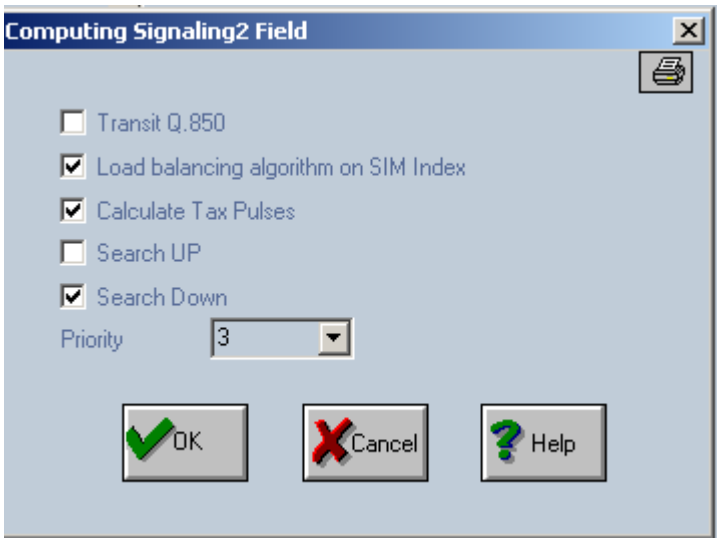


- **Signaling 2** - contains eight (8) digits that will be fully used in further developments.

"**Transit Q.850**" – not currently used

"**Load balancing algorithm on SIM index**" – in case of multiple SIM per module, it is used to enable the load-balancing algorithm ("equal load") on the SIM or RUIM cards that are already selected by SIM index algorithm

"**Calculate Tax Pulses**" – Compute the taxation (billing) pulses.

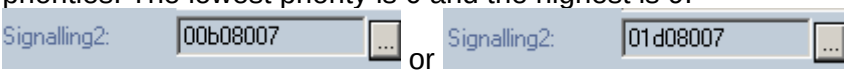


For each direction (group of trunks) of incoming calls, you can establish the computing (and sending out) of the taxation (billing) pulses. These pulses for billing will be generated according to rules you specify and will be stored in the billing files.

Note: You can establish the prefixes that will be charged with billing pulses in “Routing Table”.

The "Search UP" and "Search Down" fields are used for establish a rule in which channels are selected.

The "Priority" field is useful in order to implement routes with the same prefix to have different priorities. The lowest priority is 0 and the highest is 9.

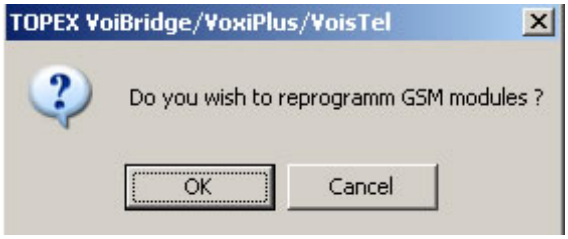


Verifications are performed for the fields that are involving insertion operation: "Insert" and "Insert_id". The allowed characters are digits from 0 to 9, "*", "#", "f" and "F". The characters "\","/","-" and "." are ignored in the saving process. In case of an error An error message is displayed.
By pressing "OK" you validate the settings for the "Signaling" field.

Note: If the field "Signaling" for a direction (for a record with type "DIR") is changed by modifying "Receive identity", "Send identity" or "Audio Level", then all the GSM modules belonging to that direction will be re-programmed (reset) in order to properly activate the sending and the receiving of the identity.

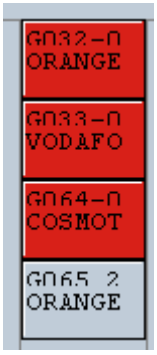
Also the audio level will be re-programmed.

The program asks for your confirmation by displayed as follows message will be:



When you click OK, all installed GSM modules will be resetted. The calls will be interrupted, and the modules will be displayed in red color on the OAM screen.

After restart, the modules will operate with the new settings and the color will be again dark blue indicating normal functioning.

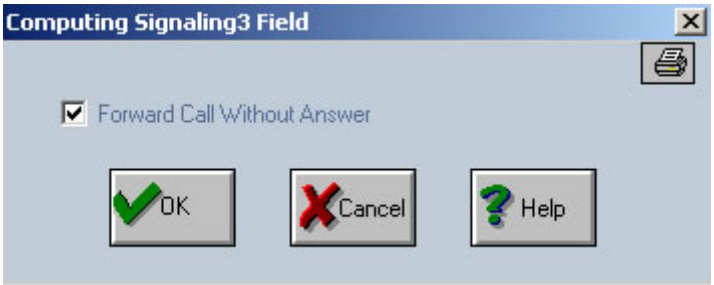


Sign3 – Right now, here is available a single option, to be used in case when incoming calls from GSM network have to be routed to a destination.

In order to accept an incoming call on a GSM module, the IN and DISA option must be activated in the settings of the GSM port.

The call can be forwarded directly to a destination if digits are inserted in the definition of the direction to which the GSM port belongs.

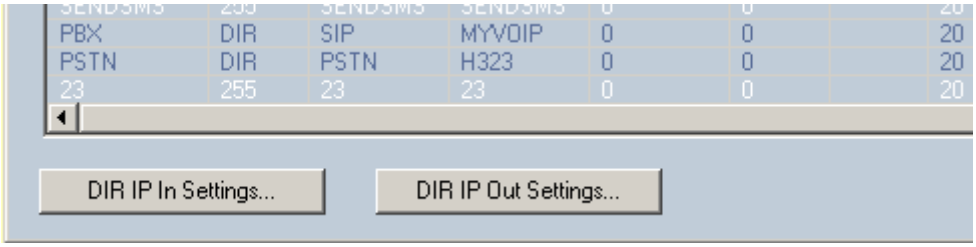
If the "**Forward Call Without Answer**" option is activated then the call will be forwarded to destination and the answer to the GSM side will be committed only when the destination (for example VoIP) will answer. Until then, it is put on hold on ring-back tone



If the "**Forward Call Without Answer"** option is not checked, the incoming GSM call will be answered immediately, before the answer from the destination. In this case the mobile caller must pay for the call, even before it is actually connected to destination.

2.4.6 DIR IP Settings

This is just a feature of the Calls Directions paragraph, but it is important enough to be detailed. At the bottom of the „Define calls directions" page are located two buttons „**DIR IP In Settings...**" and „**DIR IP Out Settings...**" which allows the definition of the input and output directions for the VOXI box:



a) DIR IP In Settings – this section is used to perform direction settings for VoIP incoming calls.

Press the button to define the characteristics of the input directions in the gateway. On the screen will appear the „**Define settings for the IP directions**" which contains the following parameters:

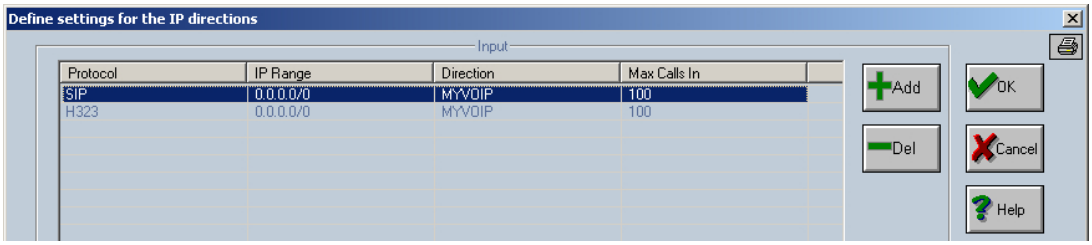
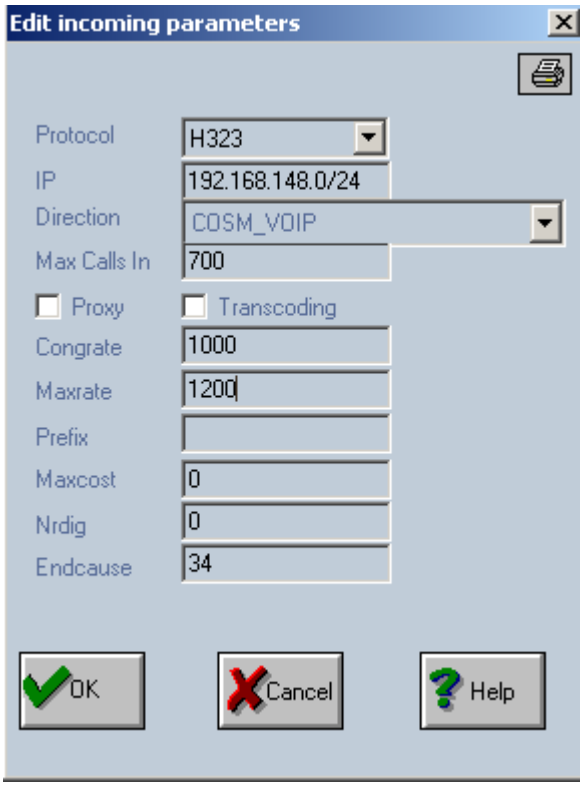


Figure 6 – Define settings for the input IP directions

Here the user must allocate to each pair <protocol, IP range> a direction and a maximum number of incoming calls. You can Add or remove (Del) records about incoming VoIP calls. The name of the direction can be one of the already defined directions, so it can be "MYVOIP" - the generic direction used for group the VoIP channels or another direction if you want to use ignore / insert features.

If you use the generic MYVOIP name, all VoIP calls will be treated in the same way, but if you use specific direction names, you can define different rules for different incoming IPs, for instance for those using the SIP, respectively H323 protocols. This allows you to perform operations such ignoring / inserting digits on the incoming number or identity (see "Calls direction") like for ordinary calls.

To edit the parameters of the input directions double-click the respective direction name and on the screen will appear the „Edit incoming parameters" page which contains the following fields:



- **Protocol** – the protocol used for the respective direction. You can select SIP or H.323.
- **IP** – the IP range for the incoming direction. Here you can complete either a single IP (such as 192.168.144.57) or a range of IP values (for example "192.168.1.0/24").
- If you use an IP without specifying the range, then calls are accepted just from that particular IP; It is the same as using "range /32"

Below are shown a few examples of using the IP range:

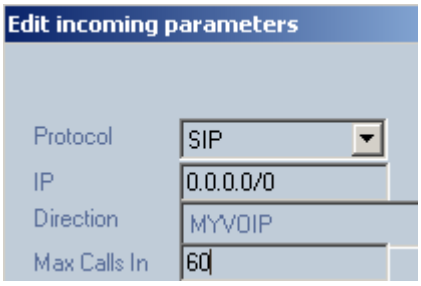
192.168.1.0/24 then the IP addresses allowed are 192.168.1.x;
192.168.0.0/16 then the IP addresses allowed are 192.168.x.x;
192.0.0.0/8 then the IP addresses allowed are 192.x.x.x;
0.0.0.0/0 then the IP addresses allowed are x.x.x.x (calls are allowed from **all** IP address!);

- **Direction** – the direction name. It is chosen from the list with defined directions;
- **Max Calls In** – the maximum number of calls which can be receipted on a single direction. In the previous example the maximum number of calls is 700. If you enter 0 (zero) in the field "Max Calls In", this means no restrictions are placed upon the number of incoming calls.

To add a new input direction press the „Add" button with a green plus sign and to delete an input direction, press the „Del" button with a green minus sign.

Example:

SIP 0.0.0.0/0 MYVOIP 60 - for SIP protocol calls are allowed from all IP address, and maximum 60 calls are allowed;



Note: The maximum number of incoming calls controls the number of simultaneously calls accepted from the specified IP. If you enter 0 (zero) in the field "Max Calls", this means no restrictions are placed upon the number of incoming calls. VOIX The equipment will accept any number of calls.

The default value is 1000.

Proxy - must be enabled if the source IP is behind a NAT.

Transcoding - must be enabled when the source and the destination have different codecs. This feature works only on equipments with VoIP card included.

Congrate - this is the congestion rate. It should not be zero. The default value is 1000.

Maxrate - this parameter represents the total number of setup calls on a second. It should not be zero. The default value is 1000.

Prefix - this parameter was added in case when calls are coming with different prefixes from the same IP source. In case of two such prefixes - the user can assign two different directions for the same IP source.

Maxcost - this parameter will be used in further developments.

Nrdig - with this parameter - you can control the number of expected digits for each call coming from the specified IP.

Endcause - this field has is the release code used when the received number has a different number of digits then the expected one - "NrDig" value. The default value is 34.

b) DIR IP Out Settings... – this section is used to perform settings for outgoing VoIP calls.

Press this button and on the screen you will see window called „Define settings for the IP directions" related to the output direction.

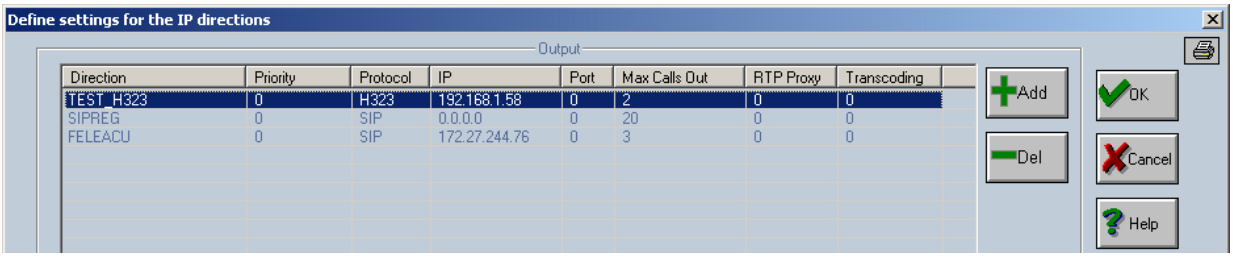


Figure 7 – Define directions for the output IP directions

Here the software application features a possibility to allow overflow over an IP direction. The user defines for each direction name a protocol (SIP or H323), a destination IP, a port used for signaling for example 1720 for H323 and 5060 for SIP) and the maximum number of outgoing calls.

In order to route a call on VoIP to a destination IP address - there are two possibilities of setting in "Routing Table".

- If you use "DIRIP" in "action" field then you can provide a VoIP protocol (SIP/H323) but just one IP;
- If you use "DIR" in "action" field then you can provide a direction name (you don't have any VoIP channel assigned to that direction - because all VoIP channels are already assigned to the "MYVOIP" direction). Here in "DIR IP OUT" section you'll make the relation to an IP address and a VoIP protocol (SIP/H323).

To edit the output directions features press double click on the direction name and set the parameters from the „Edit outgoing parameters".

Direction – select the direction from the list with defined directions names;

Priority - this parameter represents the priority of the selected direction. This parameter can be also set in "Signaling2" field in the direction definition ("Define Calls Direction");

Protocol – the VOIP protocol used for the respective direction;

Figure 8 – Edit outgoing IP parameters

IP – the IP range for the incoming direction. Here you can complete either a single IP (such as 192.168.1.58) or a range of IP values (for example "192.168.1.0/24"); The maximum number of incoming calls is set to "2" (as an example, the default is "1000"). If you use an IP without specifying the range, then calls are accepted just from that single IP. It is the same as using range /32;

Port – the port number related to the selected direction;

Max Calls Out – maximum numbers of calls on the output direction;

RTPProxy - must be enabled if the destination IP is behind a NAT;

Transcoding - must be enabled when the source and the destination has different codecs. The codec may be changed if, for instance if the destination IP does "know" just one codec, so we must perform a transcoding.

Click OK to save the settings performed over IP Out directions.

2.4.7  - Routing table

Defining the table with rules for routing of the calls.

- Press on the icon "**Routing table**" () to open the window with the routing table. You can define maximum 128 of routing rules (there are also situations in which the TOPEX machine can handle more routing records such 1024 or 4000 records). The user can consult the current settings of the TOPEX machine regarding the number of directions and routing records in the license window);

- Use the buttons of the right sight of window for performing the following operations: add a routing rule to the table, insert a new record, edit a routing rule, and delete a rule for routing.

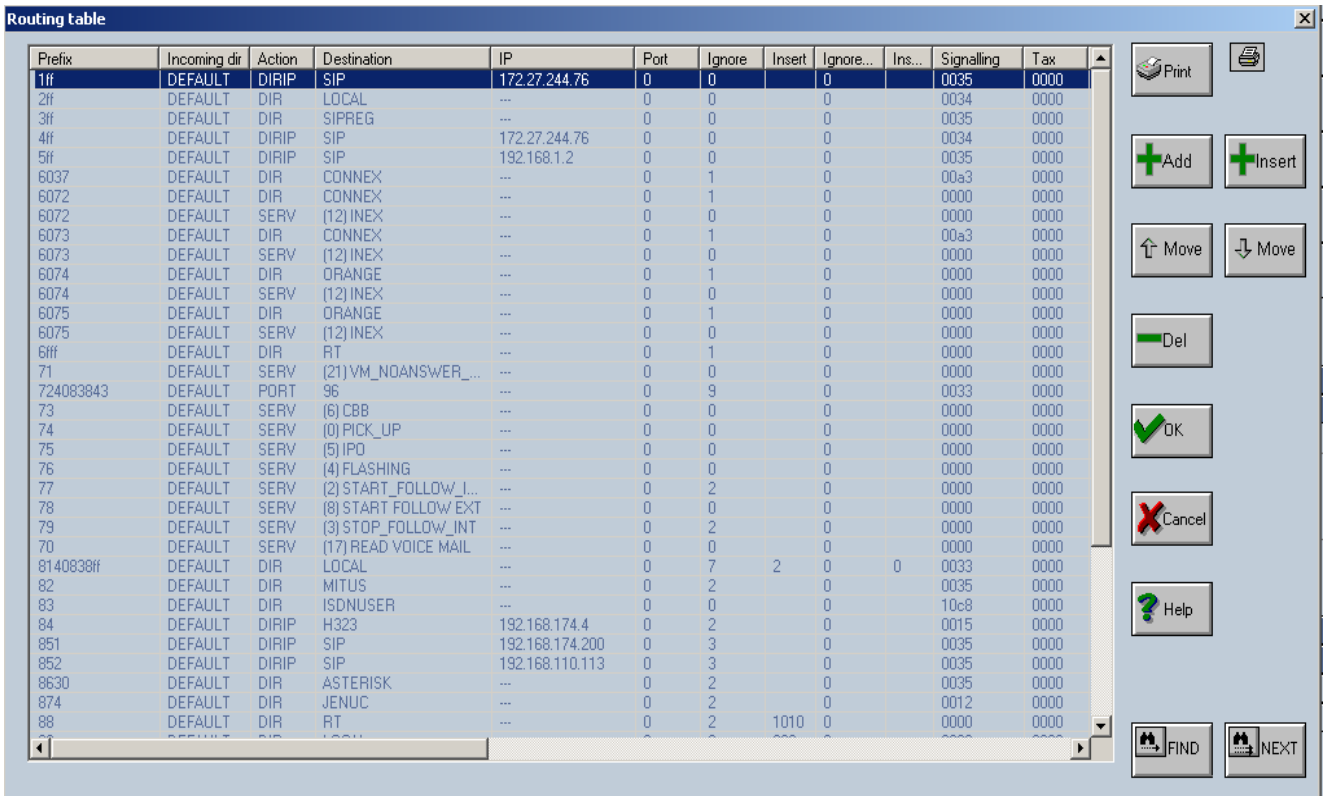
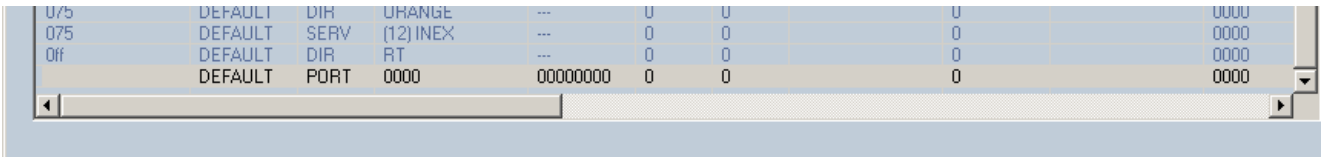
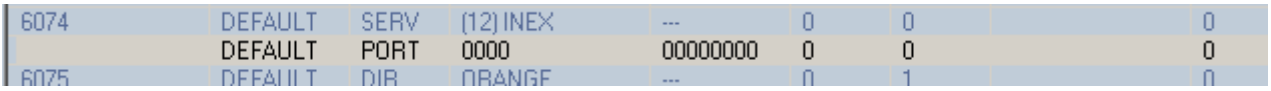


Figure 9 – Routing Table

Add – add a routing rule to the table. A new record will be added at the bottom of the list (the new line will have an empty "Prefix" field);



Insert – insert a new record over the selected record from the table (the new line will have an empty "Prefix" field);



Del - delete a rule for routing. First select it from the list and then click the "Del" button (the deleted line will have an empty "Prefix" field);

You can have the same prefix more then once - case in which you can use "Move" buttons to change the prefix order - for the same prefix - the upper line will have a greater priority than the lower one.

Note: For several fields the length is limited: for “Prefix”, “Insert” and “Insert_Id” maximum 16 digits are allowed, for “IP” the maximum is 15 characters and for fields “Signaling”, “Tax” and “Port” 4 digits is the maximum.

To edit a routing rule double click it and the editing window will show up:

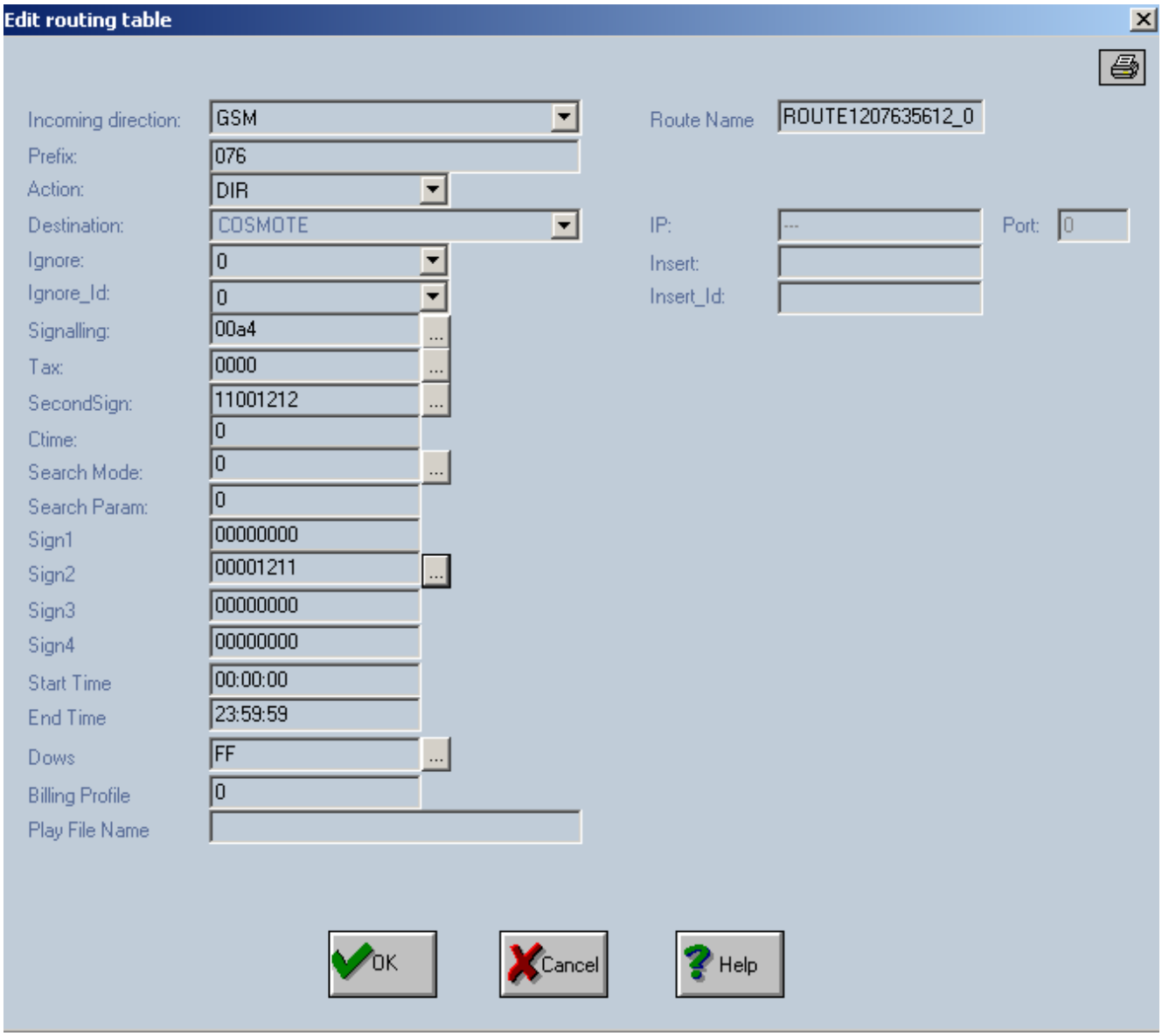
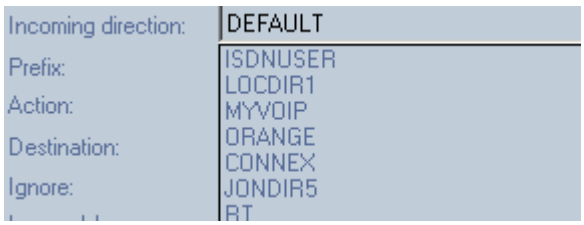


Figure 10 – Edit routing table

Incoming direction – this parameter is used in case of using several routing records with the same prefix in order to make the difference between incoming sources. It has to be interpreted here as incoming source direction. A "DEFAULT" value means that there is not rule to be applied.



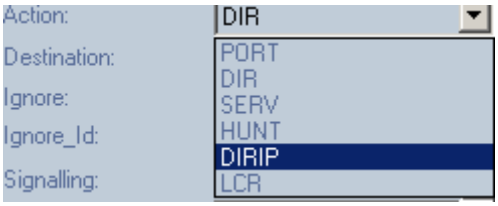
Prefix – the first digits of numbering, which are needed for routing the call (for example to direct all the calls to a mobile telephony network on a certain route enter the prefix 0722 or 0723 or 0788).

Prefix	Incoming dir	Action	Destination	IP	Port	Ignore	Insert	Ignore_Id	Insert_Id	Signalling	Tax
23	DEFAULT	DIR	MYVOIP	---	0	1		0		0033	0000
24	DEFAULT	DIRIP	SIP	192.168.2...	0	2		0		0035	0000
25	DEFAULT	DIRIP	H323	192.168.2...	0	2		0		0035	0000
33	DEFAULT	DIR	MYVOIP	---	0	1		0		0033	0000
4	DEFAULT	SERV	(4) FLASHING	---	0	0		0		0000	0000
074	DEFAULT	DIR	ORANGE	---	0	0		0		00a4	0000
076	DEFAULT	DIR	COSMOTE	---	0	0		0		00a4	0000

- If you leave empty this field the respective routing rule will NOT be taken into account at saving action;
- Inside a prefix string, the value “f” can be used to indicate any digits from zero to nine (0-9). For example, “1f2” means al prefixes from 102, 112, and so on until 192. This feature is very useful because it allows you to reduce the number of records in the routing table.

Action - specifies the action to be taken for the call.

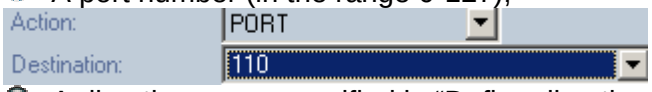
It is a drop list of six strings, you may choose either : “PORT”, “DIR”, “SERV”, “HUNT”, “DIRIP” or “LCR”.



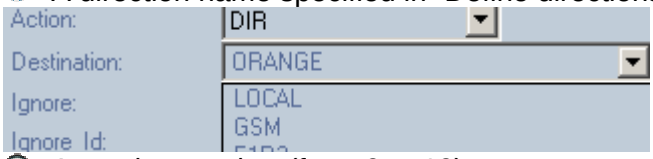
- **PORT** - the call will get out through the port specified in the "Destination" list;
- **DIR** - the call will get out by the direction you have specified in the "Destination" list;
- **SERV** - the call will get out through the service that was specified in the "Destination" list;
- **HUNT** - the call will get out through the hunting group that you have specified in the "Destination" list;
- **DIRIP** - in such a case the "Dest" field will be interpreted as VoIP protocol - SIP or H323. The user must provide destinations IP address and port (in case when the default values of 1720 for H323 and 5060 for SIP are not used);
- **LCR** - the call will get out by analyzing the LCR table for the index specified in the "Destination" field. This field can take a value from 0 to 6. Each value represents a rule to be applied in order to find a direction at the specified moment.

Destination – the destination, which may be EITHER :

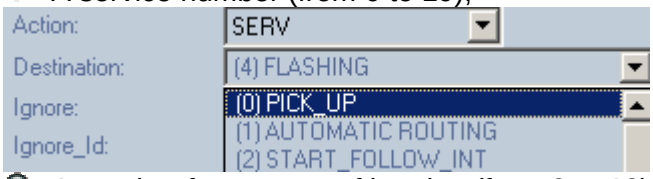
- A port number (in the range 0-127);



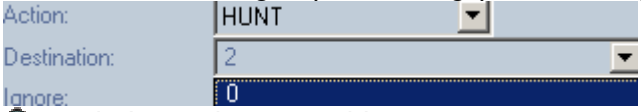
- A direction name specified in “Define directions names” (from the list);



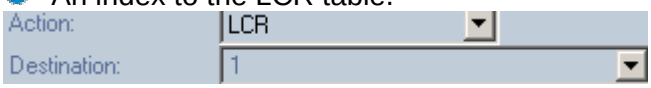
- A service number (from 0 to 19);



- A number for a group of hunting (from 0 to 19);



- An index to the LCR table.



Ignore – the number of digits that will be ignored from the numbering sent out through “Destination”. There is a list with values from 0 to 20;

Insert – option complementary to “Ignore”. Adds to the number sent out through “Destination” the digits specified; the maximum allowed is 16 digits.

Ignore_Id – the number of digits that will be ignored from the Caller ID sent out through "Destination". There is a list with values from 0 to 20;

Insert_id – option complementary to "Ignore_Id". Adds the specified digits to the Caller ID sent out through 'Dest', in this example 30; the maximum allowed is 16 digits.

IP – used when "DIRIP" is used to fill the "Action" field. It is the remote IP address used to establish voice over IP connection;

Port - the number of port used in voice over IP signaling. It is used when "DIRIP" is specified in the action field.

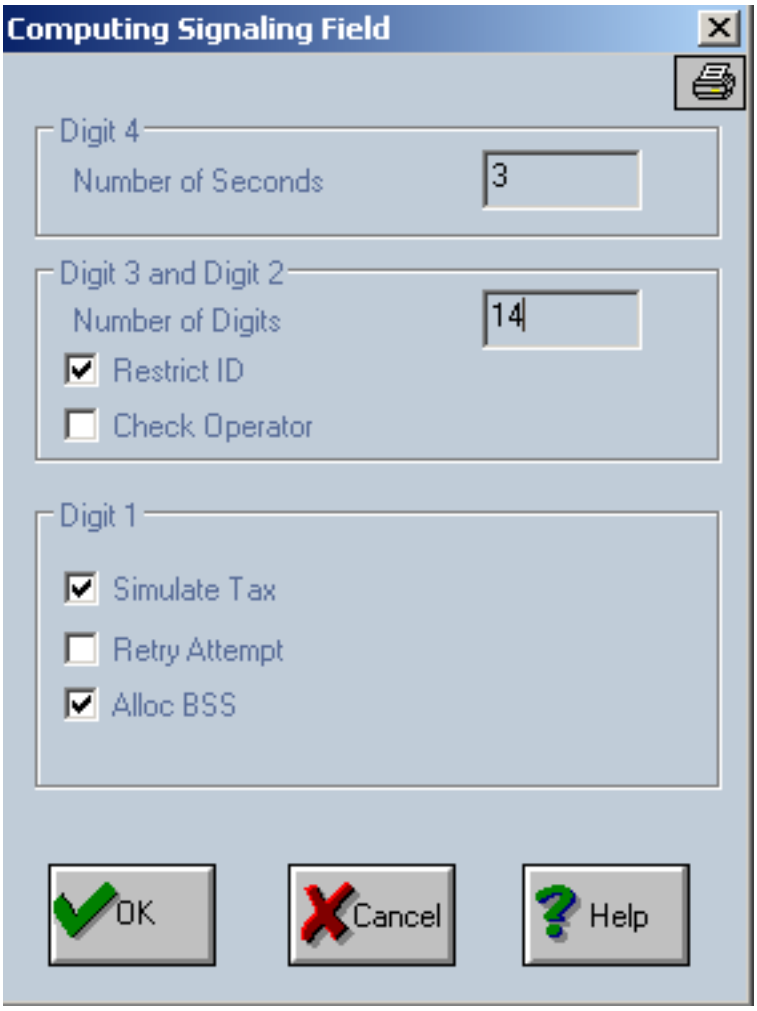
Note: The Insert and Ignore operations for the phone number and Caller ID are processed first in the "Directions IN" and then in the "Routing Table" steps. The "Directions OUT" does no longer perform ignoring or adding of digits to the phone numbers.

Signaling – four hexadecimal digits of the type "xyzw", explained in the following table:

Digits	**Interpretation**	
	Bit (from right to left)	**Meaning**
x (digit 1)	**bit 3 (mask 0x8000)**	**Alloc BSS** - this option is used in the situations when the ring-back tone must be identified in order to declare the call as answered. This option is useful in cases when the gateway application must make the difference between a call answered without ring-back tone and a call answered after a ring-back tone. Additional software must be installed on the gateway.
	bit 1 (mask 2000)	**Simulate Tax** - is used in case of FXO junction - in which the answer at destination can't be recognized. In such a situation this option has to be validated. The call is considered as answered as soon as the call is made on output link.
	bit 0 (mask 0x1000)	**Retry Attempt** - when this bit is 1 one retry attempt will be made in case of a first failure on this direction; when this bit is 0 no retry attempts will be made
y (digit 2)	**Check Operator (mask 0x0800)** - is used when portability facility is desired. For each call, a database interrogation is performed. The portability database can be located on the same gateway or on another PC. Additional software must be installed on the gateway. **Restrict ID (mask 0x0400)** - is used for SS7 direction in order to indicate that the identity is restricted. The identity can be hidden if in the routing record - the ignore identity field is put to maximum digit allowed - 20.	
z (digit 3)	Enable to set the number of digits which are waiting to take the action specified in the field "Action"	
w (digit 4)	Enable to set the number of seconds in which digits are waiting to take the action specified in the field "Action"	

Note: Value of 'z' can take values in range '0' to 'f' (where 'f' means 15 digits). When a bigger number of digits is required, the value of 'y' may be used. A group of 'yz' with values '10' means 16 digits.

You may fill the individual bits for Signaling or instead you can click the button "..." and a window for automated setting of "Signaling" field will show up like this:



The program performs validations upon some values - the field called "Number of Seconds" can be filled with values from 0 to 15 and the field "Number of Digits" with values from 0 to 20.

After pressing the button OK the result should be something like:

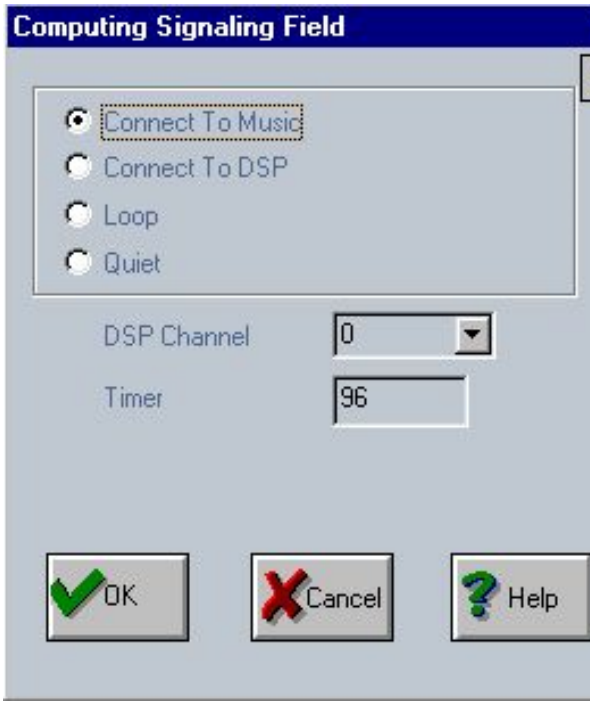
Note 1: If you have selected the action “SERV” and the service number is “4”, the incoming call will get a flashing tone.

This is used for testing purposes (verifying the connection of TOPEX VoiBridge to other equipment).

Note 2: There is a situation when the "Computing Signaling Field" window will look different.

In this case, the “Signaling” field will specify actions concerning the flashing tone provided to the incoming call.

When "Action" field is "SERV" and "Destination" field is "FLASHING", the window will look like shown in the next image:



- **Connect To Music** - the TOPEX equipment provides to the incoming call a flashing tone (the normal greeting music of the system);
- **Connect To DSP** - the welcoming tone will be obtained from a DSP (Digital Signal Processing) circuit (with possible values from 0 to 63);
- **Loop** - the "Tx" and "Rx" are looped together. This way, the caller can verify the continuity of the circuit, by sending out a certain tone on Tx and getting back the same tone on Rx;
- **Quiet** - no tonality will be provided to the incoming call; the user will not hear anything. The "Quiet" option in used for performing measurements;

The "**Timer**" field is used to specify the duration of the greeting tone, as follows: if a "0" value is used then the tone will be heard forever otherwise the "Timer" value will set the interval of time (in seconds) during which the selected tone will be played to the incoming call.

Setting Call Release Cause

One of the "Services" available is setting the cause for release of a call as you wish.

When "Action" field is "SERV" and "Dest" field is "PLAY_RELEASE",

the Computing Signaling Field window will look like this:

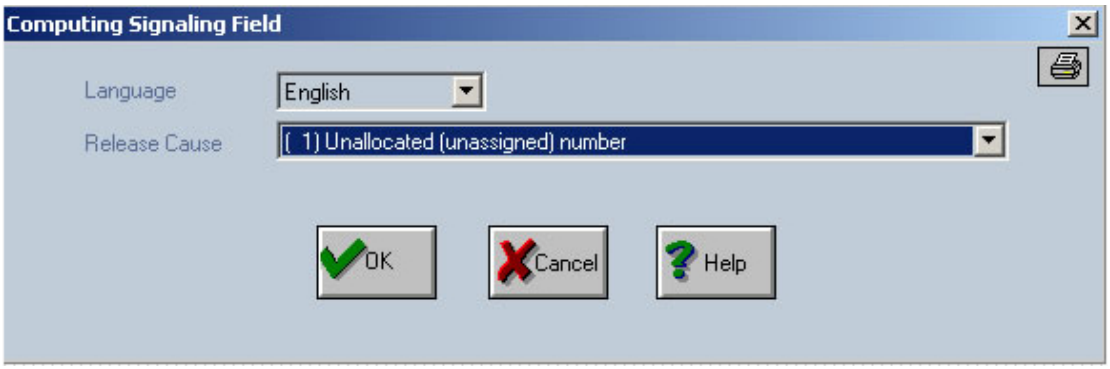
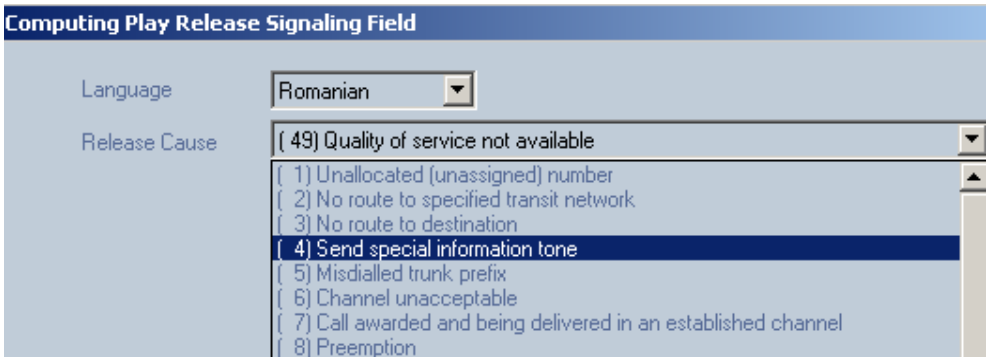


Figure 11 – Computing signaling field for Play Release

What it does:
For the prefix specified in routing table with action ="SERV" and destination = "PLAY_RELEASE", the call will be released with the cause specified in "Release Cause".

You may select every possibility out of the drop-down list, as shown in the next image.

Also, a pre- recorded message stored on the TOPEX equipment may be played on such a situation, and you may select the language used:



Play File

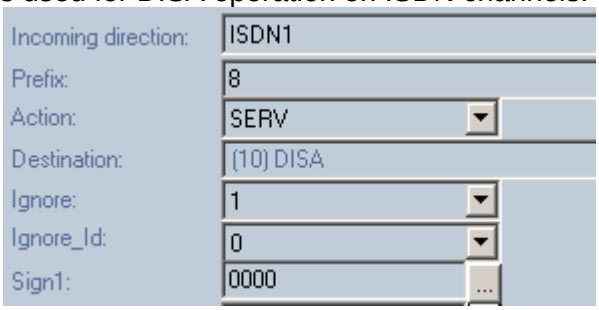
When "Action" field is "SERV" and "Dest" field is "PLAY_FILE" then incoming calls which are routed (based on this record) will be connected to the voice file specified in "Play File Name" field.

Voice Tests

The pair of settings "SERV" and "Dest=SELECT PORT" is used for testing the quality of voice through the two GSM modules from the gateway (as explained in "Facilities-Commands" chapter).

DISA on ISDN

The pair of settings "SERV" and "Dest=DISA" is used for DISA operation on ISDN channels.
For example a record could be:
"Prefix = 8"
"Action = SERV"
"Destination = DISA"
"Ignore =1"
"Sign1 = 0000"

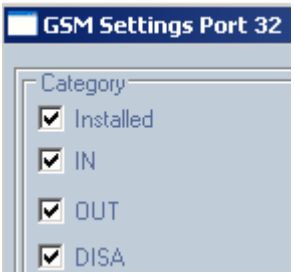


The meaning of this record is: calls are coming from ISDN with the prefix (first digit) '8'. This digit is deleted (because the value in the field "Ignore" is 1). DISA tone is provided. All those new digits, which are collected in DTMF tone, will be analyzed according to the routing table. If you want to be sure that all the digits from the original number are deleted you can use instead "Ignore = 20".

If you want to force the call to be outputted only on a GSM direction, you can have in the list starting with '8' the "Insert" field completed with the first digit of a mobile call.

For example "Insert = 0". Then, in the record for prefix "0" in the routing table you must set "Ignore =1" so one digit is deleted.

Note: For a GSM port, the DISA tone is provided for it if you select the appropriate Category checkbox in its GSM Settings configuration window:



Tax – type of billing. You can establish the prefixes that will be charged with billing pulses in "Routing Table". To can handle charging issues you must change the field "Tax".

Prefix	Action	Dest.	IP	Port	Ign	Ins	Ign_id	Ins_id	Signaling	Tax
0	DIR	GSM		0	0		0		00a4	110a

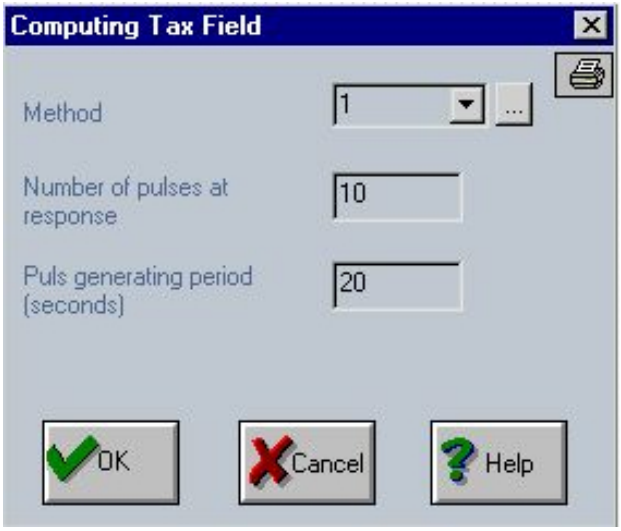
There are three methods (rules) for configuring calculation of tax pulse. The first digit (the leftmost of the four) of "Tax" field is used to differentiate between those methods. Allowed values for the method are: 0 – no method, 1 – method 1, 2 – method 2 and 3 – method 3.

Method 1

"**1xyy**" – upon answering the call is charged with "x" pulses.

During the state of conversation the call is charged with one pulse every "yy" seconds.

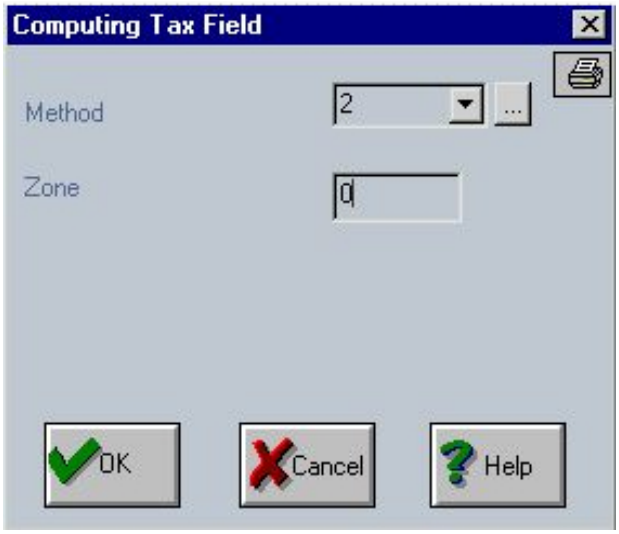
So if you select "1" you must specify the number of pulses at response (10 in the next image) and the time period for pulse generation (20 seconds in the adjacent example).



Method 2

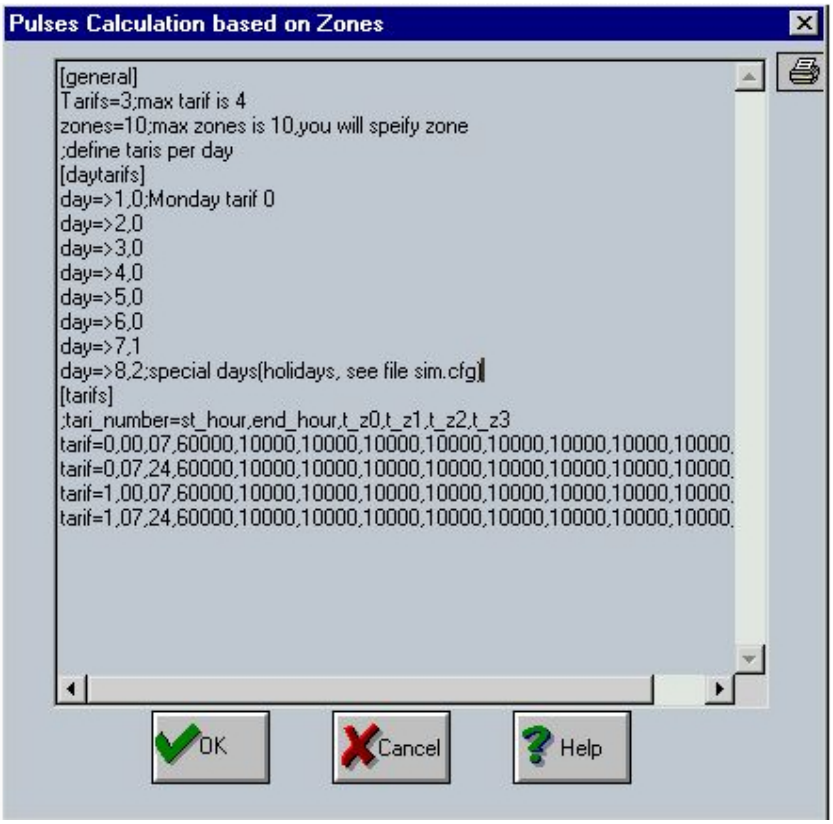
"**200x**" – the calls are charged according to several zones and tariffs. If you select Method 2 you may change only the "Tariff Index" value.

The list of all zones and tariffs is defined in the next window, "Pulse calculation based on Zones" that is displayed by pressing "…" button.



You can easily handle zones and tariffs. A maximum number of ten zones and four tariffs can be defined.

In the adjacent image you can notice the tariff allocation on each day of the week.
First, you have the day-tariff assignment. “day=>1,0” means that the tariff 0 is applied on each “Monday”. The days of the week are allocated beginning with Monday – 1 up to Sunday – 7. There is also an 8th day of the week, the holidays the first value is 8. These special days that begin with 8 are defined in “Define Holidays”
After day – tariff allocation the tariff – zone correspondence follows:



- Each line starts with a triplet “tariff=x,yy,zz”, where “x” is the number of the tariff and “yy-zz” is the time interval when the settings that follow are applied;
- After the characters “tariff=x,yy,zz” come the ten columns, the zones showing time period when a pulse is generated. The temporization values are in msec, so if a value is “60000” this means 60 seconds

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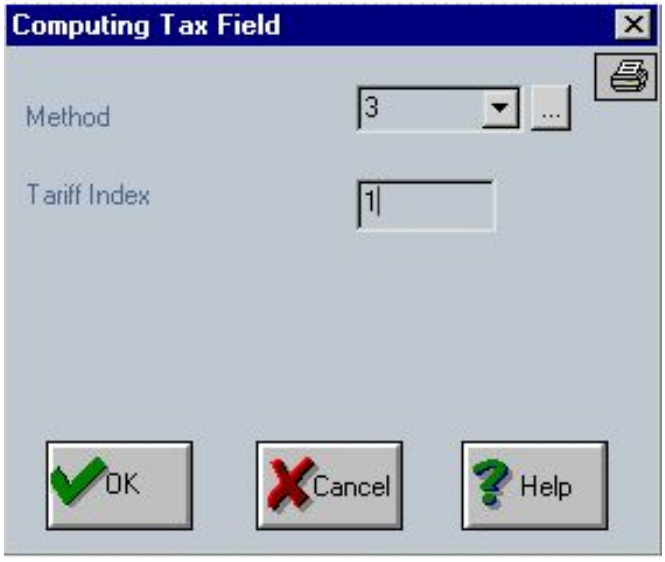
Method 3

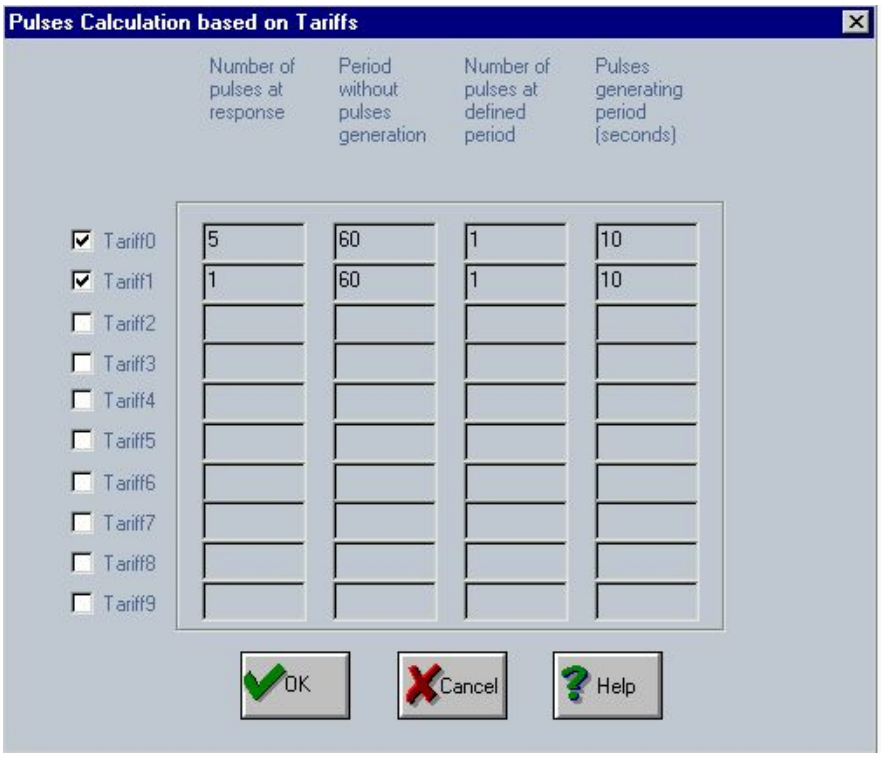
“ **3xxx**” – the calls are charged according with a tariff. This is an extension (refinement) of Method1. Besides the number of pulses upon answering and the period for generating pulses, now you can specify also a period without tax pulses and the number of pulses per taxing period. This kind of billing is used by several mobile telephony carriers. If you select method 3, you may change only the “Tariff Index” value.



There are maximum 10 tariffs, so values for “Tariff Index” can be value from 0 to 9.

The list of all tariffs is defined in the window “Pulse Calculation based on Tariffs”, which is displayed by pressing the button “…”.

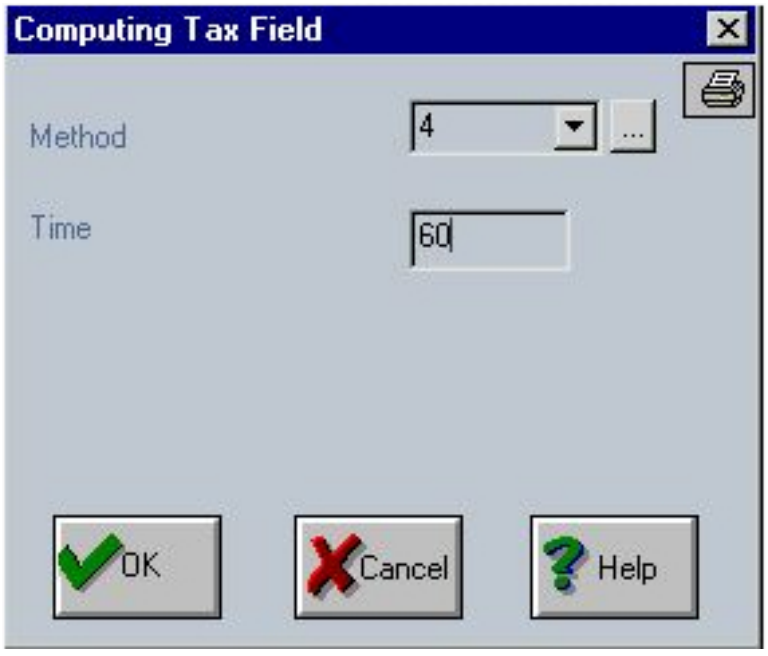
In the adjacent picture, “Tariff1” is defined as follows:
- one pulse is sent upon answering
- then follows a one minute pause, for 60 seconds no pulses are sent
- after 60 seconds one pulse is sent every 10 seconds.



In the “Tax” field, there is also a facility for limiting the maximum duration of a call limit.

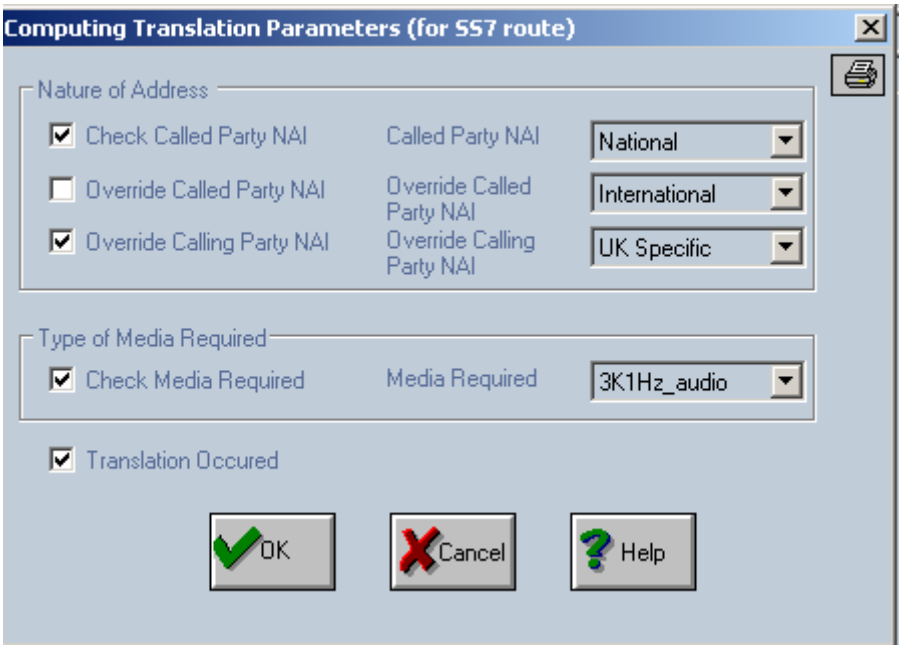
For this, you select “4” for the value of the field “Method”. This is NOT really a method for calculating the charge for a call!

With this, you may impose a time limit for the call. The value for “Time” is in minutes, so the example above means that no calls longer that one hour will be allowed.



SecondSign –parameter for SS7 signaling.

Here you can specify (in case of a SS7 route) some translations parameters as follows:



The first zone is related to "Nature of Address" information. You have the possibility to override both the "Called Party" nature of address and the "Calling Party" nature of address.

The "Called Party" nature of address is changed by enabling the first two options "Check Called Party NAI" and "Override Called Party NAI".

The original "Called Party NAI" - Subscriber, Unknown, National, International and UK Specific - can be changed to "Override Called Party NAI" which contains the same list as the first one.

Moreover the "Calling Party NAI" can be override by selecting a value from "Override Calling Party NAI" and enabling the "Override Calling Party NAI" option.

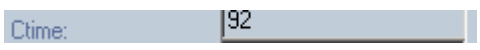
The second zone named "Type of Media Required" - the route will be available just for the specified type of communication media.

Possible values are "speech", "64k_unrestr" and "3K1Hz_audio".

The "Translation Occurred" is to indicate for SS7 that a translation of number has occurred.

Ctime – Call Time

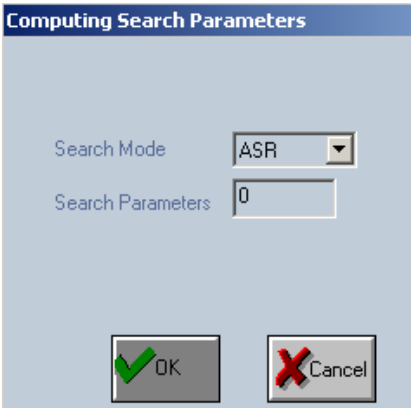
This parameter establishes the maximum call duration (in minutes). It is similar to "Tax" method 4. If this value is set here (not 0) the "Tax" method 4 will be ignored.



Search Mode - this parameter is used for routing in case of using routes with the same prefix.

In such a case a method for overflowing and dividing the traffic between several routes must be provided.

Each route from such a group must have "Retry Attempt" option in "Signaling field" (0x1000).



Overflow is performed if one of the following situations occurs:

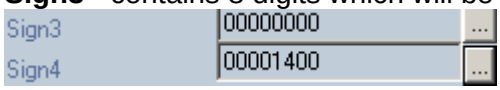
- When the call is routed on VoIP - the main application is checking if number of simultaneously calls is greater then the maximum number of output calls established in "**Diripout settings**" (see DIRIPOUT section);
- When the call is dropped from the remote side with a release cause which is set for rerouting in "trafic.cfg". For example to reroute in case of congestion message, you must add in the "trafic.cfg" file a line with "rerouteoncause 34 1".

We assume these situations in the following explanations.

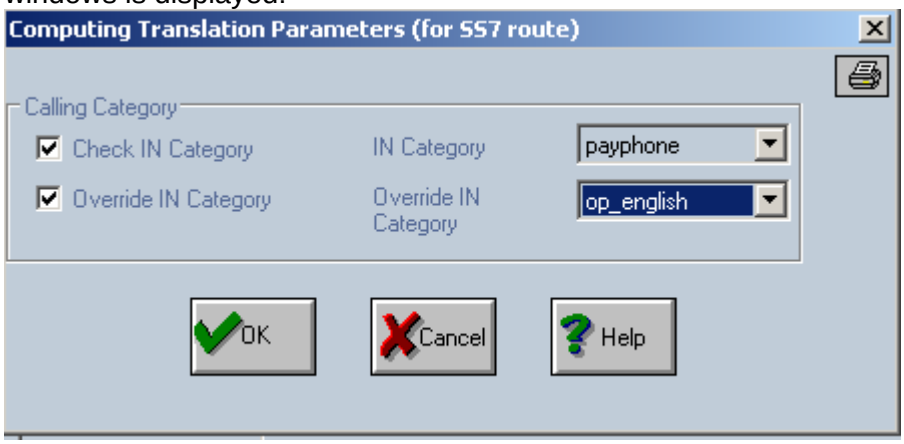
This parameter is used in conjunction with "Search Param" parameters:

Search Mode	Search Param	Significance
ASR	Not used	The route will be chosen based on ASR value
ACD	Not used	The route will be chosen based on ACD value
Priority	Not used	The call will be routed based on direction priority (direction specified in "dest" field). Calls from the routing group (with the same prefix) will go mostly on the direction with the highest priority. If the maximum number of calls is reached (for example for a direction specified in "DIR IP OUT" settings when "Max Calls Out" value is passed
Down	Not used	The route will be chosen from the first to the last one. Depending on the position in routing table the first route from the group will have the highest priority.
Up	Not used	The route will be chosen from the last to the first one. Depending on the position in routing table the last route from the group will have the highest priority.
Circular	Not used	The route will be chosen circularly.
Percent	Specifies the percent	Calls will be routed based on percentage. The "Search Parameters" represents in this case the percentage value. The application running on TOPEX machine knows the number of calls on each machine.

Sign3 - contains 8 digits which will be fully used in further development.

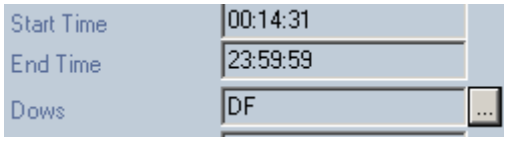


Sign4 - is used for a SS7 route for translating purposes: the "IN Category" can be overridden. Press the "…" button to the right and the "Computing Translation Parameters (for SS7 route)" windows is displayed:



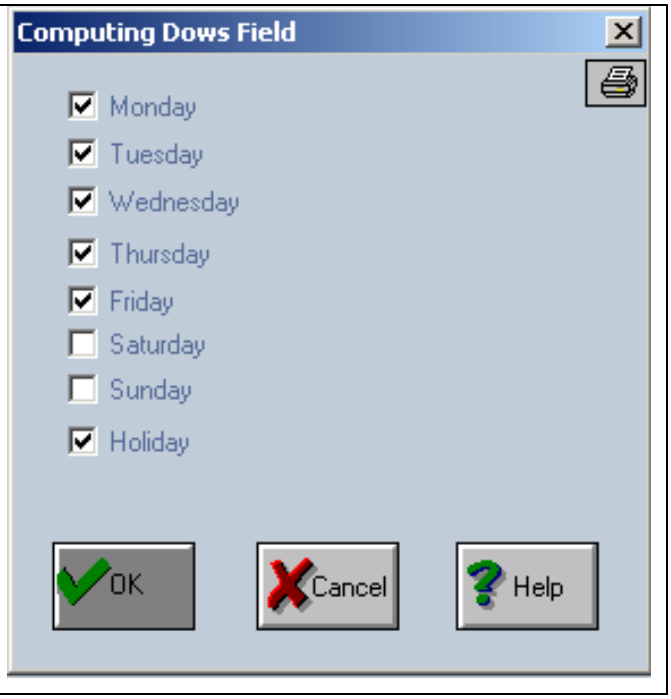
To override the incoming category - the boxes "Check IN Category" and "Override IN Category" must be selected. In "IN Category" list - you select the incoming category which will be checked and, if you want, replaced with the value in "Override IN Category". The possible values are: "unknown", "op_french", "op_english", "op_german", "op_russian", "op_spain", "op_rsrv1", "op_rsrv2", "op_rsrv3", "notused", "ord_subscr", "prio_subscr", "data_call", "test_call", "payphone", "uk_oper_call", "uk_admin_diverted".

Sign5 - contains 8 digits which will be fully used in further development.
Sign6 - contains 8 digits which will be fully used in further development.



The next parameters are used to establish when the routing records are valid: "StartTime", "EndTime" and "Dows".

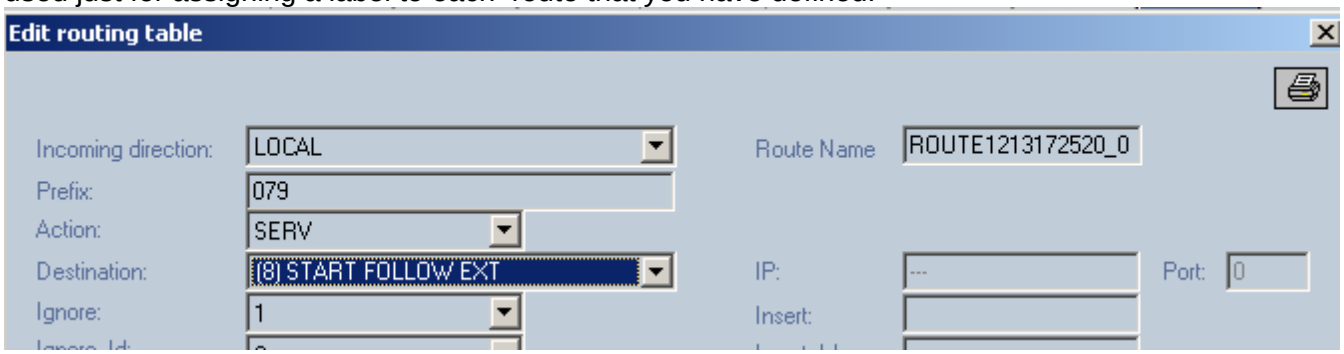
The first two parameters represent the time period when the record is valid (calls can be routed based on the selected record). The default setting is the whole day, that is beginning at 00.00.000 and ending at 23.59.59.

The "Dows" field is used to set the days of the week when the record is valid. Click on the "…" button to the right to display the window "Computing Dows Field" where you can mark the checkboxes for the days of the week. You may have , for instance, one routing record valid for the working days, and another valid only on Saturday and Sunday.	

The "**Billing Profile**" field will be used in further development.

The "**Play File Name**" field is used when "Action" field is "SERV" and "Destination" is "PLAY_FILE". It represents the name of the file to be played.

The "**Route Name**" field in the upper right corner of the windows is generated automatically. It is used just for assigning a label to each route that you have defined:

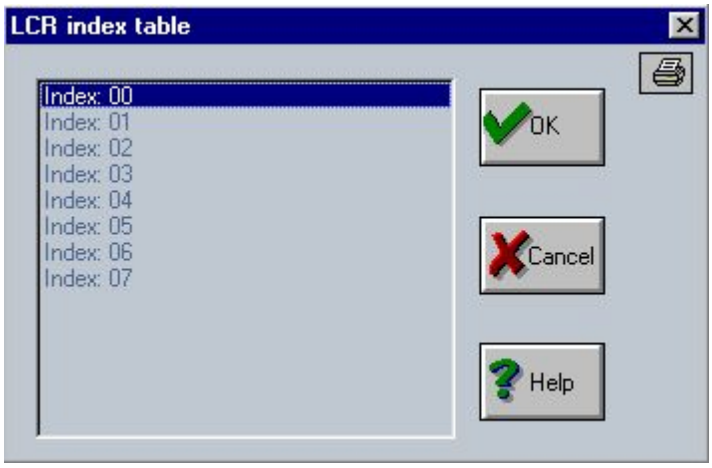


or:

2.4.8 - LCR Table

In this table there are displayed the 8 indexes of LCR rules.

Each index includes information about the selected direction, for specified time intervals on each day of the week and about the days that were declared national holidays (non-working days).



To edit a certain LCR index, double click on that index and the next window will be displayed on the screen:

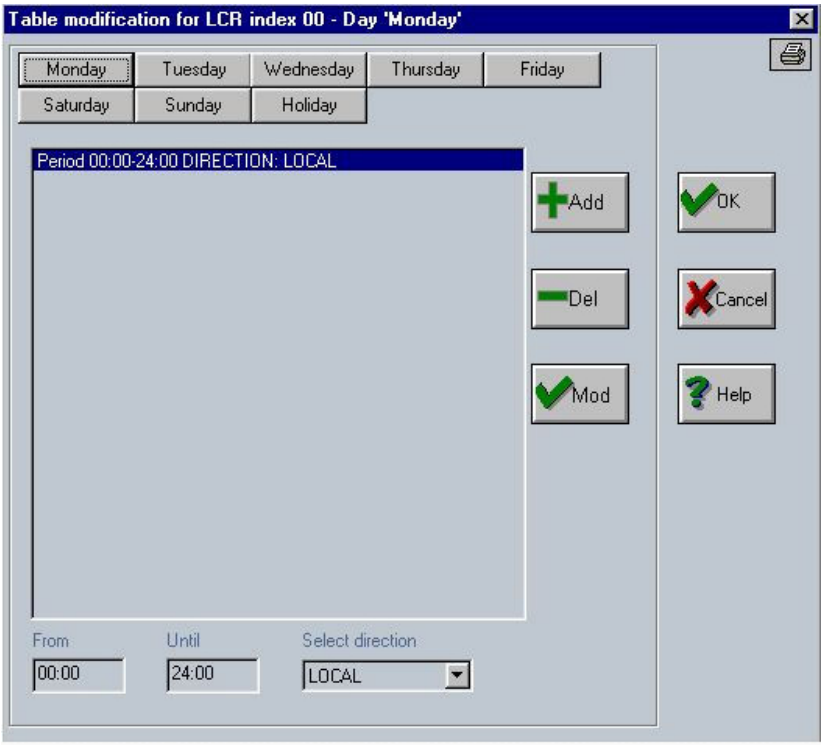


Figure 12 – LCR table

Note: when the dialog window for "Table modification" appears, the information is displayed by default for "Monday".

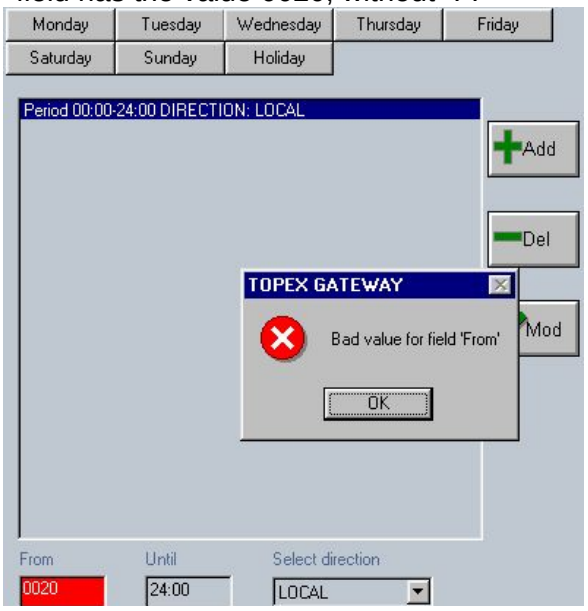
For each day of the week, a list shows up with the time periods (intervals) that specify the direction to be used. In the previous image, for 00:00 until 24:00 the direction "LOCAL" will be used. The periods are structured as hours and minutes separated by character ":".	Period 00:00-11:00 DIRECTION: LOCAL Period 11:01-11:30 DIRECTION: GSM Period 11:31-17:30 DIRECTION: LOCAL Period 17:31-21:30 DIRECTION: MYVOIP Period 21:31-23:59 DIRECTION: SENDSMS

When the user modifies the selection in the list, the fields which are located at the bottom of the window are filled with the line information: period "From" to "Until" and the current direction.

These values can be modified and validated by the option "Mod". If you select the “Add" option, a new line will be added in the list. The "Del" option is used to delete a record from the list.

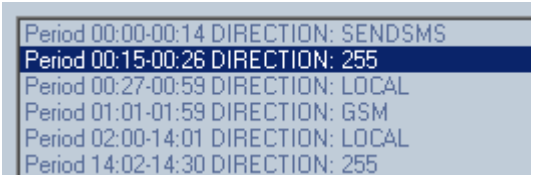
The periods are checked before adding or modifying operation.
An error message is generated if the character ":" is not preserved or the values used are wrong (hours or minutes).

In this example the “From” field has the value 0020, without “:”.



The field in error is shown with red, and the error message is “Bad value for field From”.

Note: The lines that containing in the field “Select direction” the default value of "255" instead of a declared name direction are not saved in the "LCR" file.



This way the format of the "LCR" file is compacted in order to get a smaller size.

An example of LCR file content is as follows:

```
#---LCR INDEX FILE---
#Day type Mon=0x01 Tue=0x02 Wen=0x04 Thu=0x08 Fri=0x10 Sat=0x20 Sun=0x40
Holyday=0x80
#s index_lcr day_type_bitmap(%x) start_hour:min end_hour:min dir(%x)
l 00 1 00:00 24:00 0
l 00 1e 00:00 24:00 1
l 00 20 00:00 00:01 f
l 00 20 00:01 12:01 11
l 00 20 12:01 14:01 13
l 00 20 14:01 23:01 a
l 00 20 23:01 24:01 9
l 07 13 00:00 24:00 1
l 07 4 00:00 11:45 1
l 07 4 11:45 12:23 8
l 07 4 12:23 12:31 1
l 07 4 12:31 18:50 8
```

Each line contains a letter 'l', a LCR index number (for example 00 and 07), a mask (each day of the week has an identifier constant (Mon=0x01 Tue=0x02 Wen=0x04 Thu=0x08 Fri=0x10 Sat=0x20 Sun=0x40 Holyday=0x80), periods used to define an interval and finally an index to specify the direction used. This is the index in the "Directions Names" table (a value from 0 to 19).

Note: If the same periods and the same direction are used on different days (but for the same LCR index) then instead of two records, in the file will be written only one record with the field "mask" completed by an OR operation between those days. For example "1e" is used to identify Tuesday, Wednesday, Thursday and Friday.

2.4.9 - Classes of Restrictions

Press the “Restriction Classes” icon (") for defining certain classes of restrictions for the calls. On the screen will appear the “Restriction classes” window which contains the restrictions for every defined direction.

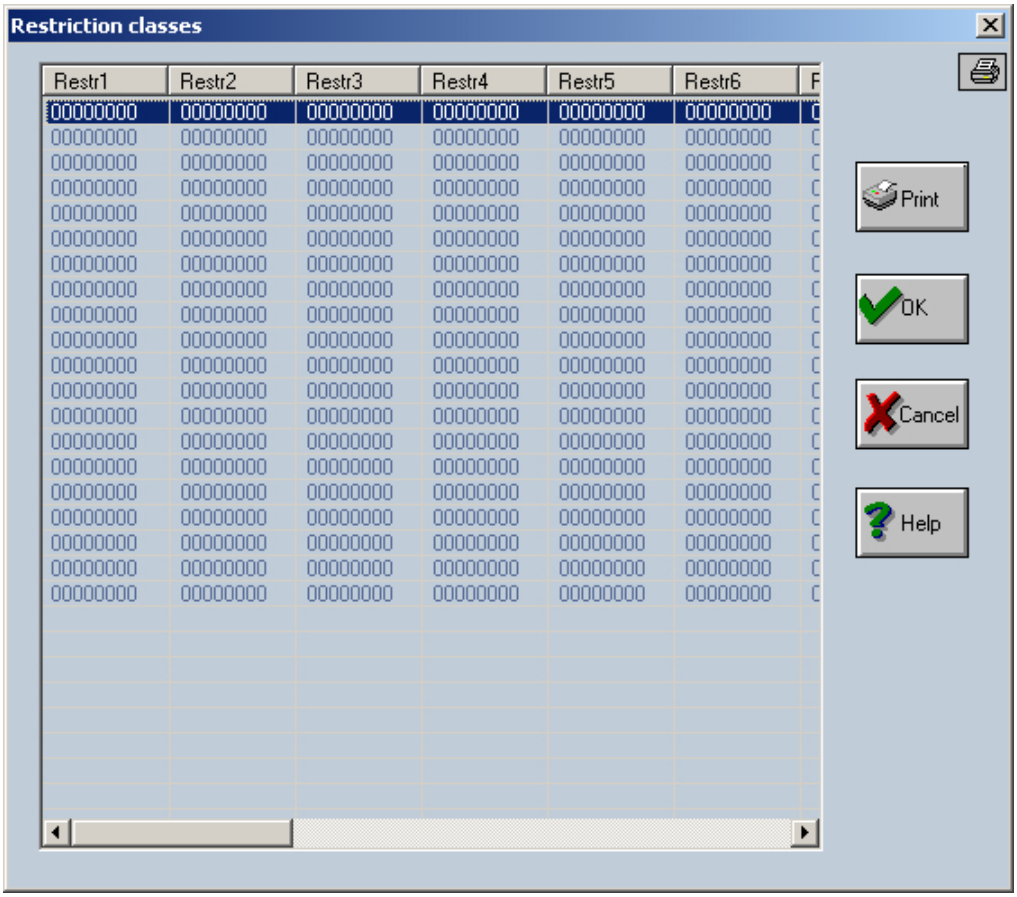


Figure 13 – Restriction classes

Double click on the class line for editing or modifying the restriction of a class. Use the “Edit” window that appears for inserting the restriction. There are maximum 20 restrictions for each class.

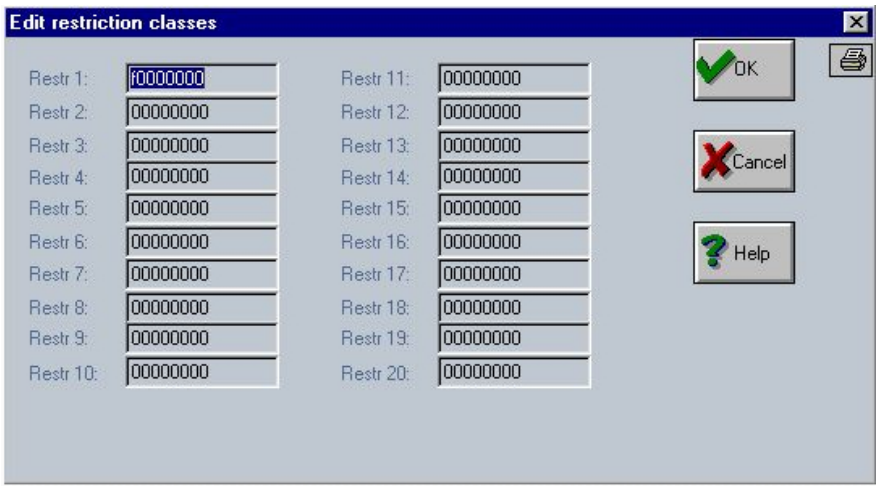


Figure 14 – Edit restriction classes

Rules for establishing restrictions:

Each restriction has eight characters. The characters can be any digit (from zero to nine) or the letters 'a' or 'f':

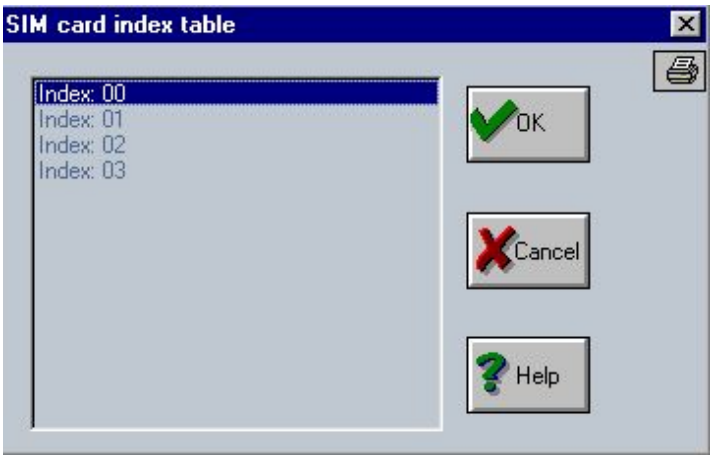
- '**a**' represents the digit 0 (zero). For instance a subscriber with the restriction aa000000 won't be able to make calls started with digits "00" (international phone calls).
- '**f**' represents "any digit". For example, if you define the restriction f0000000 for a certain port it won't be able to call any number - it will be able only to receive phone calls.
- '**0**' signifies the end of the restriction rule. If in a restriction field there are eight "0" figures this means that no restriction has been defined for that field.

A class can inherit one or more restriction classes. For instance, if in a class we have the restriction 000000xx, where xx is the hexadecimal value corresponding to another class of restrictions, then our class will inherit all the restrictions that have been defined in the xx class, to which there will be added the restrictions defined in the current class.

2.4.10  - SIM Table

This is Table with indexes for selection of active SIM cards.

Press the SIM icon ("  ") and the window “SIM card index table” will appear. This is a table where the four indexes of SIM cards are displayed (00, 01, 02, 03).

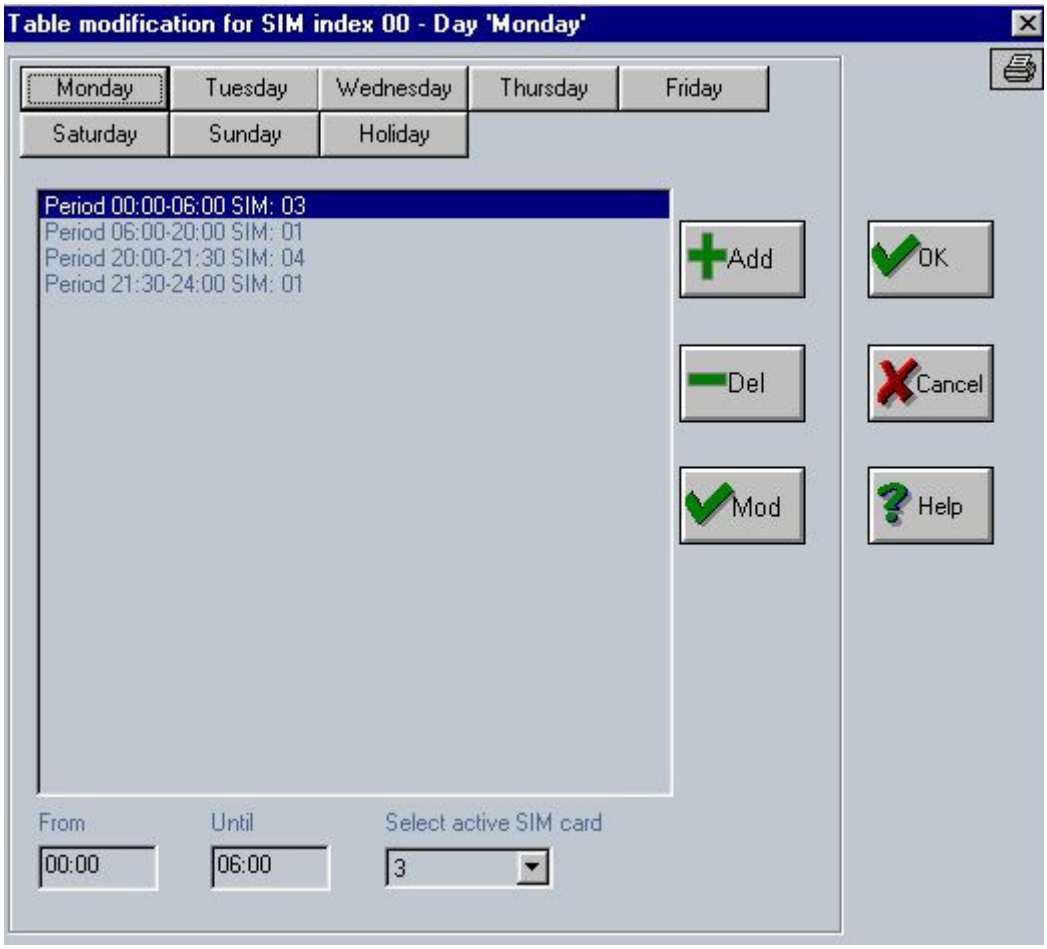


Each index includes information about the SIM card selected for each hour interval of each day of the week and even the days that were declared national holidays (non-working days).

To edit a SIM index, double click on the respective index and the “Table modification for SIM xx” window will be displayed on the screen.

In the list are displayed the periods (intervals) that specify the SIM/RUIM card (1, 2, 3 or 4) to be used. The time periods are structured in hours and minutes separated by the colon character ":".

When the user changes the selection in the list, the fields which are located at the bottom of the window are automatically filled with the line information: period "From" to "Until" and the active SIM card. These values can be modified and validated by option "Mod". If user selects "Add" option a new line will be added in the list. The "Del" option is used to delete a record from the list.



Notes:

- when the dialog window for "Table modification" appears, by default the information is displayed for "Monday".
- The SIM index table was developed for equipments with a **four-SIM holder** for every GSM module. But the Voxi series equipment have just one SIM inserted into each GSM module - so the possibility is only to change between the SIM card 1 and no SIM (255). By default on SIM index 0 the SIM card 1 is set to be used.

The periods are checked before adding or modifying operation. An error message is generated if the character ":" is not preserved or the values used are wrong (hours or minutes).

The lines that contain "255" instead of an active SIM card are not saved in the "simindex" file. The format of the "simindex" file is compacted in order to have a smaller size. An example of simindex file content is as follows:

#---SIM INDEX FILE---
#Day type Mon=0x01 Tue=0x02 Wen=0x04 Thu=0x08 Fri=0x10 Sat=0x20 Sun=0x40 Holyday=0x80
#s index_sim day_type_bitmap(%x) start_hour:min end_hour:min sim(%x)

```
#---HOLYDAYS---
#h holydayday holydaymon
s 00 1f 00:00 06:00 2
s 00 17 06:00 20:00 0
s 00 1f 20:00 21:30 3
s 00 1f 21:30 24:00 0
s 00 8 06:00 20:00 1

s 00 60 00:00 24:00 0
s 00 80 00:00 12:00 0
s 00 80 12:00 24:00 1
s 01 ff 00:00 06:00 0
s 01 ff 06:00 12:00 1
s 01 ff 12:00 18:00 2
s 01 ff 18:00 24:00 3
s 02 ff 00:00 24:00 1
s 03 ff 00:00 24:00 0
h 01 01
h 01 01
h 25 12
h 08 01
```

Each line that specifies a SIM index begins with the letter 's', then follows a SIM index number (00, 01, 02 and 03), a mask (each week day has an identifier constant (Mon=0x01 Tue=0x02 Wen=0x04 Thu=0x08 Fri=0x10 Sat=0x20 Sun=0x40 Holyday=0x80), periods used to define an interval and a SIM card value.
In the file the active SIM card is written as a number from 0 to 3. In all locations in the "gwconfig" OAM software values from 1 to 4 are used for SIM cards (on GSM holders the values 1,2,3 and 4 are shown).

Note: if the same time periods and the same active SIM card is used on different days (for the same SIM index) then instead of two records in the file will be written only one record with the field "mask" completed by an OR operation between those days. For example "1f" is used to identify Monday, Tuesday, Wednesday, Thursday and Friday.

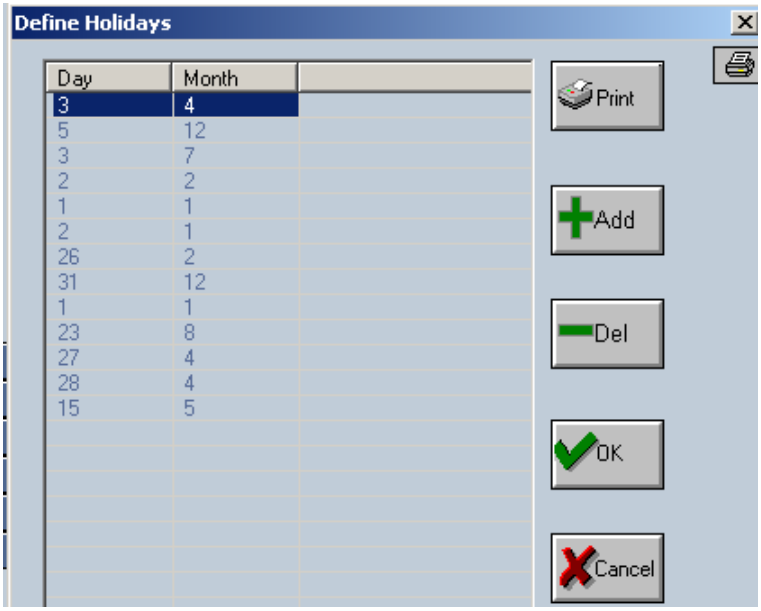
Note: the records starting with 'h' character are used for definition of holidays (non-working days) that nay have a different tariff.
See next paragraph for “Defining Holydays”!

2.4.11 - Define Holidays

In order to define the “holidays” (non-working days with special GSM tariffs), click the smiley button. This is needed if your mobile network operator has special (lower) tariffs on Sunday or national holydays.

Press the icon button ("") and the “Define Holydays” window will appear on the screen, as shown:

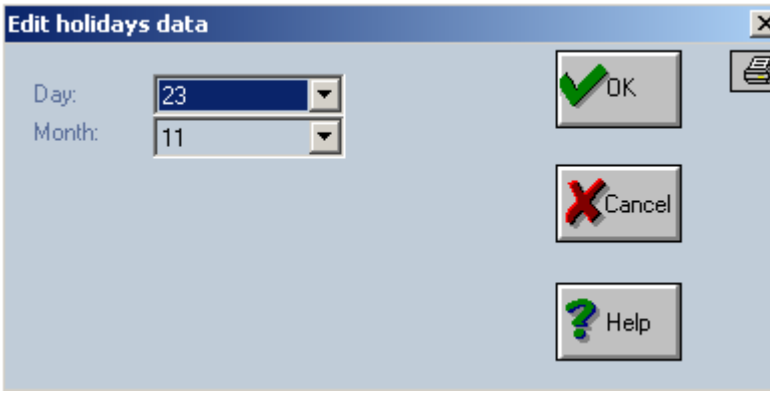
This window shows a default list with the national holidays and you can edit the list, add or remove non-working dates with special tariff.



To add a holiday in the list click the "Add" button. Once a record has been added you have to double click on it in order to be available for editing, as shown in the following window.

To delete a holiday from the list, select the date from the list and click the "Del" button.

To edit or modify these data double click them and on the screen will appear the „Edit holidays data” window which contains two fields, Day and Month, where you must insert the new holiday data.



2.4.12 - Callback Table

Press the icon ("") for defining the table with callback subscribers and their access rights and the "Callback table" window will appear:

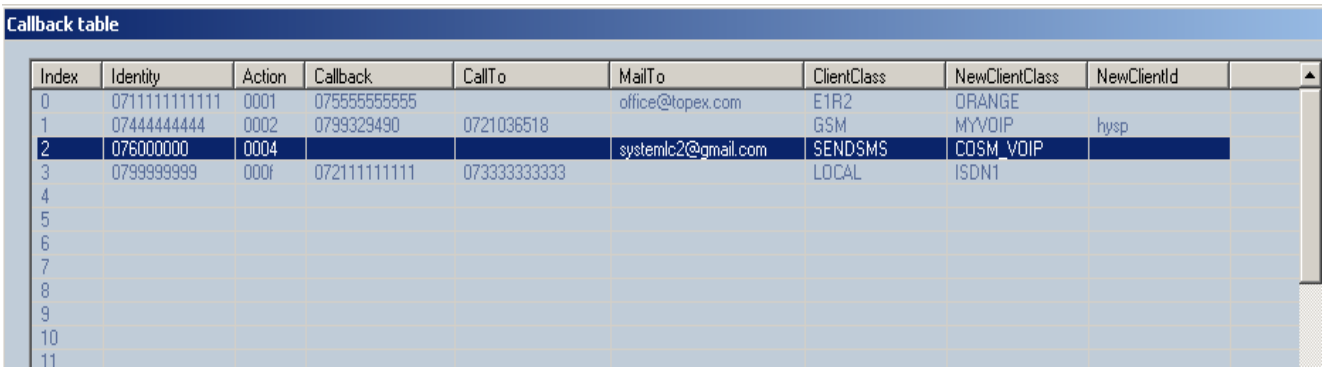


Figure 15 – Call back table

Call-back services assure lower costs for the mobile subscribers: from a mobile phone you just call the VOXI equipment and hang up. The software checks identity on incoming calls and if the telephone numbers are in its database calls them back. You don't have to pay anything for the connection!

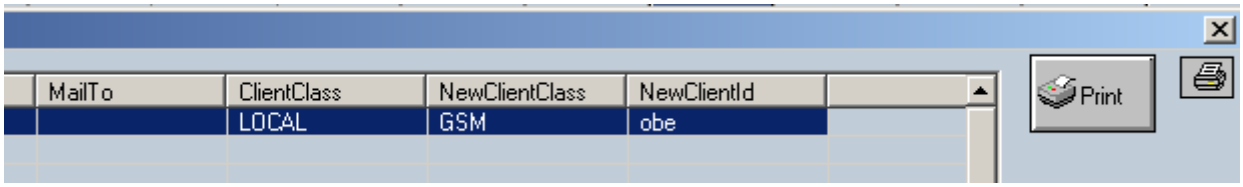
This is very useful for mobile users in the field that have prepaid cards, which feature an expensive tariff. They can ring the TOPEX equipment located at headquarters, and the equipment will call them back, put them into connection with other mobile subscriber or allow them to send e-mail messages via SMS.

Callback table contains the following fields: "Index", "Identity", "Action", "Callback", "CallTo", "MailTo", "Client Class", "NewClientClass" and "NewClientID".

Callback has several implementations (depending on the "Action" field which can take the values "CALLBACK", "CALLBACKONSMS", "SMSTOMAIL" and "DISA"):

- If the caller identity of an incoming call is found in the "Callback table" and the field action contains the mask "CALLBACK" then the caller party will be called back. The back call is sent to the phone number specified in "Callback" field (it can be the same number with the number specified in the "Identity" field or a different number). If "CallTo" field is filled with a phone number, then the initial caller will be called and put into connection with this phone number;
- if the caller identity of an incoming SMS is found in the "Callback table" and the field action contains the mask "CALLBACKONSMS" then the caller party will be called back. The reverse call is sent to the phone number specified in "Callback" field (can be the same number with the number specified in the "Identity" field or a different number-). If "CallTo" field is filled with a phone number then the caller will be called and put into connection with this phone number;
- If the caller identity of an incoming SMS is found in the "Callback table" and the field "Action" contains the mask "SMSTOMAIL" then the text content of the SMS will be sent to the IP address which is specified in the field "MailTo";

- If the caller identity of an incoming call is found in the "Callback table" and the field “Action” contains the mask "DISA" then a "DISA" tone will be provided to the caller part. The subscriber who ringed to the TOPEX equipment is called back and receives DISA tone (if the field "CallTo" does not contain another phone number). "Direct Inward Services Access' allows the mobile subscriber to get access to private telephonic facilities (he can dial numbering).
- the next three fields named "**ClientClass**","**NewClientClass**" and "**NewClientID**” were added in the latest firmware versions:



They allows to change the incoming direction from the current one to a new different direction; based on such an action, the user can operate changes on the received number and identity in the "Define Calls Direction" window.
For example if the call is coming on a GSM port belonging to the "GSM" direction then the "ClientClass" will be set to "GSM".

The "**NewClientClass**" field will be used in further developments.

To edit a routing rule double click it and the editing window will show up:

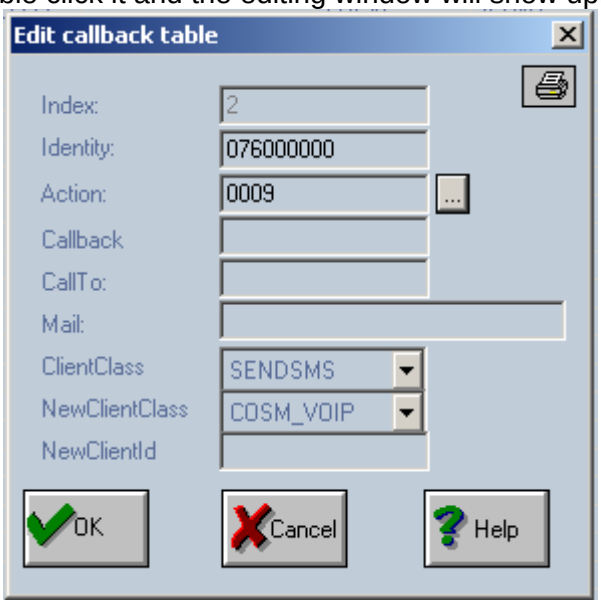
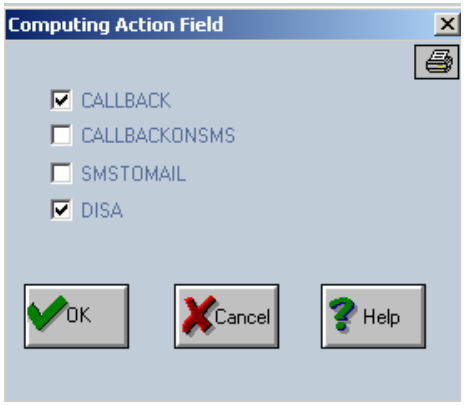


Figure 16 – Edit Call back table

Notice that the "Action" field has a button "..." to its right. If you want to edit "Action", click the button located to the right.

The window "Computing Action Field" shows up as indicated below:
Currently the following options are available "CALLBACK", "CALLBACKONSMS", "SMSTOMAIL" and "DISA".



CALLBACK – The subscriber with the number form Identity is called back. If the field Callback includes a different mobile number, that number is called and the two subscribers are connected.
CALLBACKONSMS – this option allows calling back upon reception of a SMS message. The subscriber who has sent SMS can also be connected to another subscriber.
SMSTOMAIL – you can send e-mail messages via SMS. Previously, the address of the recipient must be filled in the field Mail. The text of the message to be transmitted is sent by SMS.
DISA – the subscriber who ringed to the TOPEX equipment is called back and receives DISA tone. This happens if the field "CallTo" does not contain another phone number. "Direct Inward Services Access' allows the mobile subscriber to get access to private telephonic facilities (he can dial numbering).

Note: Callback calls are recorded in the SMS files.

2.4.13 - ISUP Table

In case of SS7 signaling you must configure the control part of the connection, ISDN User Part. It defines procedures used to setup, manage, and release trunk circuits that carry voice and data calls over the public switched telephone network.

Click the appropriate icon ("") and the "ISUP Configuration" window will appear:

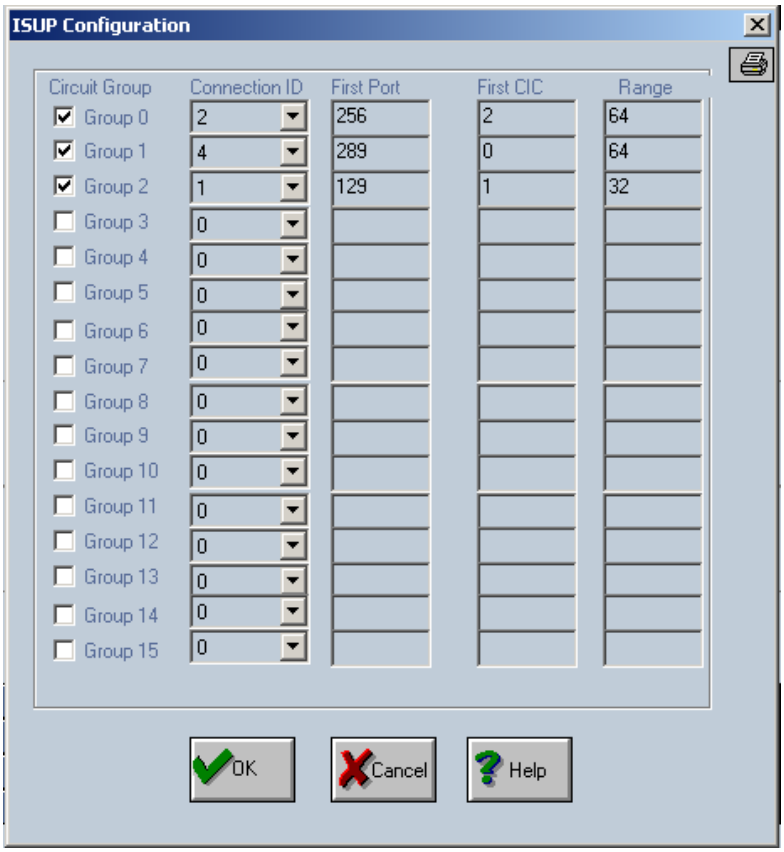


Figure 17 – ISUP Table

The "ISUP Configuration" window allows the allocation of the circuit groups (in range from 0 to 15).
- each circuit group is enabled by the option "Group x", where "x" is from 0 to 15.

- "Connection ID" can take values from 0 to 31. The " Connection ID" fields are completed in "MTP Configuration" window displayed after the user selects respective icon.

- "First Port" contains the first port from a SS7 card.

- "First CIC" (Circuit Identification Codes (CIC)) and "Range" are used to define all CIC values.

In a VOXI equipment, the SS7 trunks can be placed on card 16 (trunk 2), 17 (trunk 3), 32 (trunk 6) and 33 (trunk 7). Thus the ports for card 16 are in range 128-159, for card 17 in range 160-191, for card 32 in range 256-287 and for card 33 in range 288-319. For example the field "First Port" must be filled with the value 256 for an SS7 card installed on card 32. The range establishes the total number of channels from the field "First Port". For a single SS7 card the range field will be 32. If two consecutive SS7 cards are used (for example on positions 32 and 33) the range can take the value "64".

2.4.14  - MTP Configuration

After configuring the Control part for the SS7 link you must set up the Message Transfer Part. Click the appropriate icon ("  ") and the "MTP Configuration" window will appear as shown in the image below:

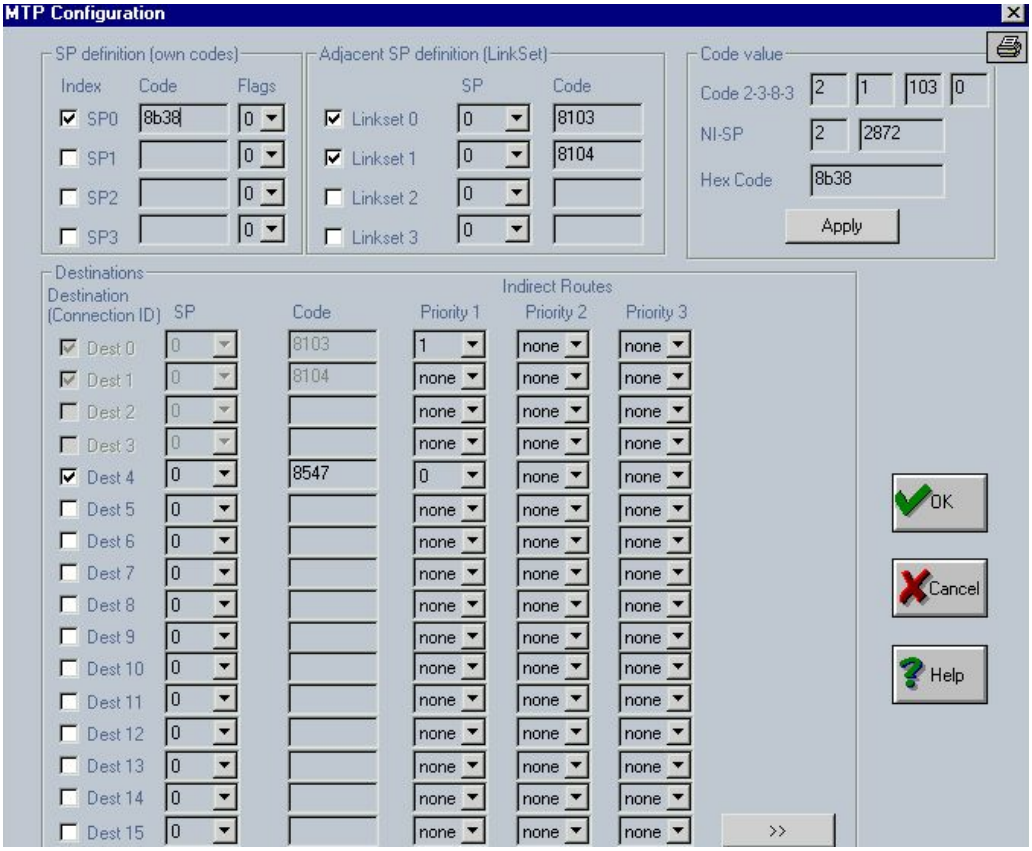


Figure 18 – MTP Configuration

In this window you must performed the definitions for signaling points (SP):

- "SP definition (own codes)" - this is the zone where the own (gateway) signaling points are defined. Each "SP" is enabled by marking it in the appropriate checkbox. For each SP the "Code" and "Flag" must be set. For "Flag" a value of "1" is indicating that this node is an STP (signaling transfer point).

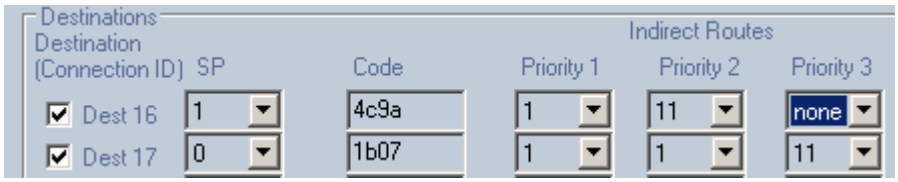
- "Adjacent SP definition (LinkSet)"- this is the zone where the adjacent signaling points are defined. Each "Linkset" is enabled by marking it in the appropriate checkbox. For each linkset the associated "SP" and the "Code" must be set. The SP list is filled with the SP values enabled in "SP definition (own codes)" zone.

- "Destinations" - this is the zone where the destinations are defined. Each "Destination (Connection ID)" is defined by enabling the appropriate checkbox "Dest" field. The first four destinations are filled based on "Linkset" definitions - "SP" and "Code" fields are completed directly if the correspondence "Linkset" is defined. This is why they are colored in gray.

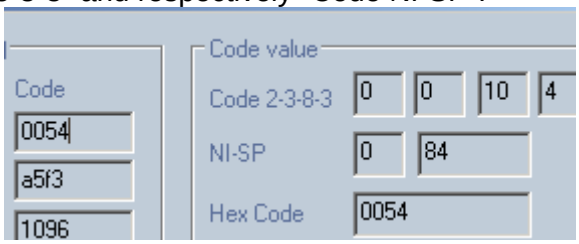
The user can define indirect routes using the "Priority" fields. For each priority, the linkset is set as follows: there are three priorities - represented as three columns: "Priority 1", "Priority 2" and "Priority 3". Priority 0 is reserved for the direct route between the own SP and the destination. For each priority and for each destination, user can select one of the enabled link sets (the linkset list is filled with the linkset values enabled in "Adjacent SP definition (LinkSet)" zone). For each destination (with a number greater than 3) the associate "SP","Code", "Priority" and

"Linkset" can be set. The SP list is filled with the SP values enabled in "SP definition (own codes)" zone.

Note1: the own SP code, the adjacent SP code and the nonadjacent SP code must all have the same "NI" field.
Note2: the ">>" button is used for destinations from 16 to 31. Seldom are needed more than 8 Destination SPs, but if you do need more than 16, click the button and a “second page” for defining Destination SPs will show up, listing SPs with numbers from 16 up to 31:



Note3: the zone "Code value" is used to automatically convert a hex code used to define a signaling point to the other known formats: "Code 2-3-8-3" and respectively "Code NI-SP".
Point with the mouse inside a "Code" edit field.
And The fields from the "Code value" zone will be filled with values . The option "Apply" is used to fill back the original "Code" edit field (an <ENTER" action on one of the fields from the "Code value" zone will cause a recalculation to be performed).



Examples:
1) to declare an own SP0 code "8b38", non-STP - check "SP0" in the "SP definition (own codes)" zone and then click the mouse on the edit field located to the right side of the "SP0" checkbox - you can directly complete the point code or you can edit the code value in the "Code value" computing zone. Here you can input the code in different formats. When you finish you press <ENTER> and "Apply" button. You select "0" in the "Flags" field.

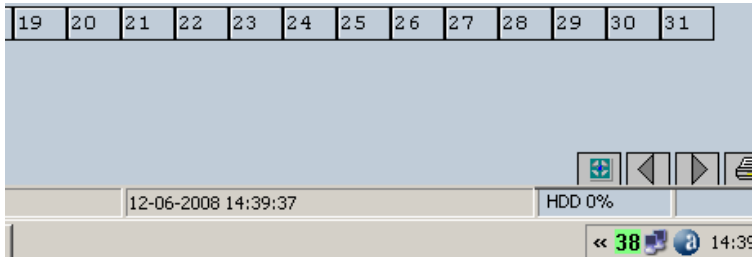
2) to declare an adjacent destination code (code "8103") - to the SP0 using linkset 0- check "Linkset 0", select '0' in "SP" list and fill the point code value with "8103".

3) to declare a non-adjacent destination code (code "8547") - to the SP0 using linkset 0 on Connection ID 4. Select "Dest 4" checkbox, select "0" from "SP" list, complete code field and select linkset on the highest priority available box.

4) to declare an indirect route (linkset 1) to an adjacent destination code "8103" (linkset 0) - select linkset 1 in the highest priority available - on the row starting with "Dest 0".

2.4.15 - VoIP Configuration

Note: *The configuration of the Voice over IP boards of the Topex equipment is done now by means of the icon “* *“ located near the bottom of the OAM screen*:



The icon for the "VOIP Configuration" in the menu bar on top of the OAM screen is kept for compatibility with older versions.

Also, here you can set up the codecs to be used. But you should remember that Because of the new mode of configurating the VoIP card, the **"VOIP Configuration"** screen shown here contains several fields **THAT WILL BE IGNORED** if the "PG Configuration" icon is displayed in the bottom-right zone of the OAM screen.

Click the button to bring up the VOIP Configration window:

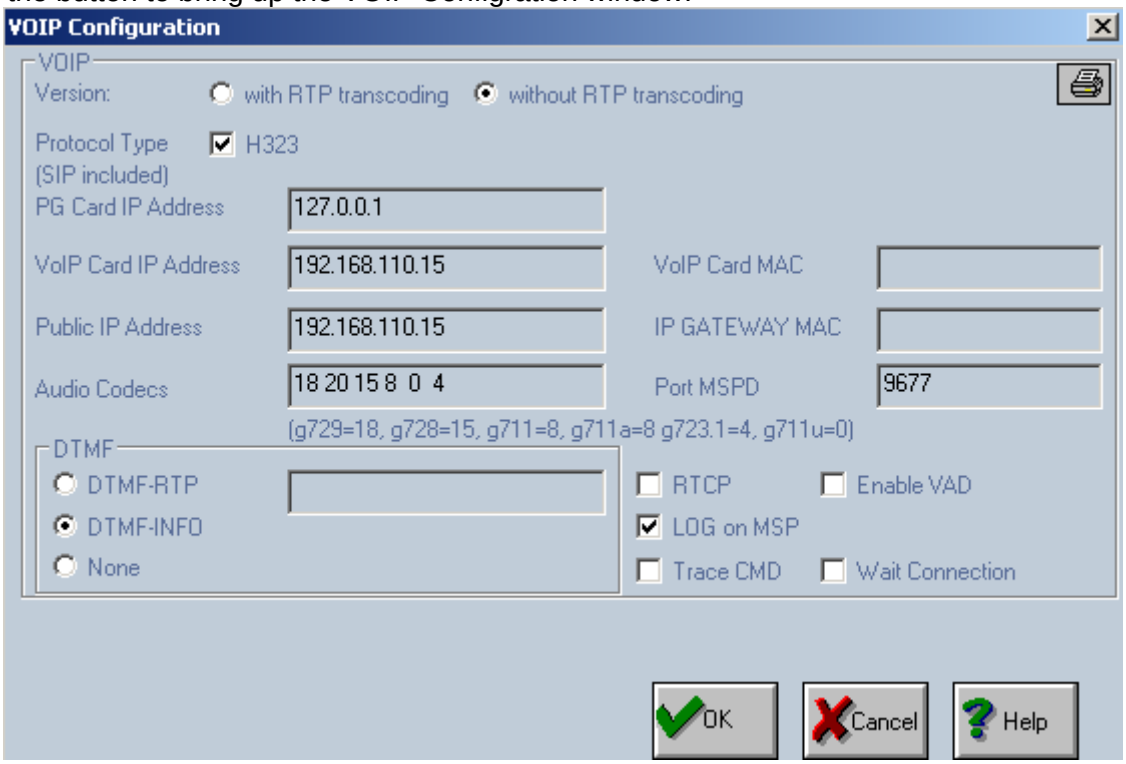


Figure 19 – VoIP Configuration

First you must decide which version of VoIP configuration you will use: **with** or **without** RTP transcoding. Transcoding means that our equipment performs conversion of codecs, that is the codecs used on the output are different from the ones used on the incoming call. For an incoming call from VoIP that goes out also through VoIP, the RTP channel is processed by the Voxi equipment and transcoding may be performed. No transcoding means that the equipment performs VoIP signaling, but does not process the RTP channel, it goes directly from source IP to destination without going through the Topex equipment. In this case transcoding cannot be performed.
The conversion of codecs may be required if the caller uses a certain type of codec, for instance g711, but the destination supports only g728, so the Topex equipment must perform transcodig. But if the called party also supports g711, no transcodig is required, and the RTP channel from the caller may be connected directly to destination.
Check the option "**without RTP transcoding**" if you are sure that no translation of codecs will be required.
Then select the VoIP **protocol** to be used. The new series of equipments feature included **SIP** protocol, so mark the checkbox in the "Protocol Type" field if you want to use **H323**. When you click the "OK" button the OAM application will load the settings from VoIP zone and depending upon "Protocol Type" selection also the settings from H323 or SIP zone will be saved.

- "**PG Card IP Address**" - the IP address of the processor card (PG card). A legacy form equipments with several plug-in cards, this field is supposed to be filled with the same value as the ones you have specified in the "Network settings - IPADDR" (icon "Gateway Parameters"). For Voxi series box this field is set by default to "127.0.0.1".

- "**VoIP Card IP Address**" - the IP address of the VoIP card. THIS FIELD HAS MEANING ONLY WHEN "PG Configuration" icon is not displayed in the bottom-right zone of the OAM screen.

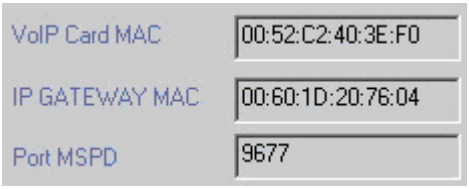
There are two situations concerning the PG Card IP address and respectively the VoIP card IP address.

1) when the Voxi box is installed in a LAN, then both addresses must be in the IP range addresses of that LAN. In this case the following ports must be opened if a firewall is used: the ports declared for RTP when the VoIP card was installed (for example 3000-3063; to see the range of ports assigned for RTP, go to window "H323 Settings" - window which is described on the bottom of the window in the area called "IP Card Settings"). RTP ports are UDP and must be forwarded to the IP address of the VoIP card. The ports 1718, 1719 and 1720 are TCP and they must be forwarded to the IP address of the processor card (PG).

2) when the PG card is using a public IP address, then the VoIP card must also use a public IP address.

- "**VoIP Card MAC**" - this value represent the MAC address assigned to the VoIP card.

The physical address is preset or is given by TOPEX. THIS FIELD HAS MEANING ONLY WHEN "PG Configuration" icon is not displayed in the bottom-right zone of the OAM screen



- "**Public IP Address**" - when the TOPEX gateway is used inside a LAN and when the H323 signaling is performed with a machine from the internet, then in this field you enter the external IP address of the LAN gateway. When H323 signaling is performed with a machine located inside the same LAN, then in this field you enter the same value as in the "VoIP Card IP Address" field. THIS FIELD HAS MEANING WHEN "PG Configuration" icon is not displayed in the bottom-right zone of the OAM screen.

- "**IP GATEWAY MAC**" - this field is filled in case of making RTP connections to a machine which is NOT in the same address range. THIS FIELD HAS MEANING WHEN ONLY WHEN "PG Configuration" icon is not displayed in the bottom-right zone of the OAM screen.
1) when the VoIP gateway is installed in the same LAN, then a program (MSPD) running on the Topex equipment automatically detects and fills the MAC of the target gateway. You will enter in the field “MSPD Parameters” the command “--no-gw”. This means no outside access is required.
2) when the remote VoIP gateway is on the public IP address, then you must first use an utility (MSPD executable) to search and find out the MAC of the target gateway, then you must type this MAC address in the field “IP Gateway MAC”.

- “**Audio Codecs”**:
A list of pairs <codec no><packetization time>, which are NOT delimited by commas, only by spaces (blanks).

This is the list of codecs to be used by the RTP protocol. VoIP uses several different codecs, having different parameters (bit rate and complexity).The codecs used by VOXI are coded with one or two digit numbers, such as 0 for g711u, 4 for g723.1, 8 for g711 or g711a, 15 for g728 and 18 for g729.

The value for packetization time (in milliseconds) is usually 20, but it may be as long as 30 msec or as short as 10 msec. For a specified codec, like g729, the number of channels supported depends of the packetization time. Increasing the packetization time reduces the packetization overhead, so when going form 20 msec to 10 msec the number of voice channels goes down form 60 to only 40.

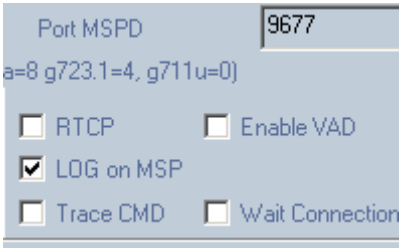
The order of the codecs does matter, for incoming calls the VOXI equipment will look in the list form left to right, so if codec “18” is at the beginning of the list, G729 will be the first used.

Examples: 

or 

- "**Enable VAD**" - this checkbox option allows the "Voice Activity Detection" facility to be used. Typical voice conversations can contain up to 35 to 50 percent silence.

On VoIP networks, both conversation and silence is packetized. VAD sends out RTP packets only when voice is detected, thus decreasing bandwidth by 30-50 %. This way the Topex will send voice packets only when it has voice input. When it detects silence on RTP, it will send to the other side a “silence packet” that use less bandwidth and allows to be interpreted for generating comfort noise.



- “**LOG on MSPD**” – enables logging on MSPD. The name of the log file will be “day-month-year_mspd.log”, such as 08-06-17mspd.log
THIS FIELD HAS MEANING ONLY WHEN "PG Configuration" icon is not displayed in the bottom-right zone of the OAM screen.

- "**Port MSPD**" - MSPD is a program that controls the VoIP, in this field you can enter the number of the port to be used by MSPD, such as 9677 that is the default. You should NOT change this value!
- “**RTCP**” – RTP Control Protocol - This checkbox establishes if the equipment uses or not the control protocol for RTP, RTCP. This s used for control and diagnostic on RTP sessions. Like RTP, RTCP typically runs on top of UDP and is defined in the IETF RFC1889. RTCP is a companion protocol to RTP that is used to maintain Quality of Service . RTP nodes analyze network conditions and periodically send each other RTCP packets that report on network congestion.

DTMF
This configuration area establishes the means of sending out DTMF in the RTP packets. There are two methods, transmission of DTMF out of band in RTP packets, or sending out INFO messages (as digital telephone-events).
In SIP protocol, the INFO method would be used for the carrying of mid-call signaling information along the session signaling path (for example to send the DTMF digits generated during a SIP session). if you check this “DTMF-RTP”: options, you must also complete the Parameters field to the right

“None” : the sending of dual-tone multi-frequency signals is not treated by the Topex equipment.

Example for DTMF-RTP option: *dtmfRTP 101 100 1*

The syntax is :
- payload type for DTMF in RTP (according to RFC 2833)
- payload type for DTMF in RTP redundancy
- redundancy scheme: 0 or 1, where 0=IETF and 1=AAL2

2.4.16  - SIP Users

The "SIP USERS" button is used to declare the SIP USERS - this option is used in case when SIP users are registered to the TOPEX Voxi equipment (the case of VoisTel).

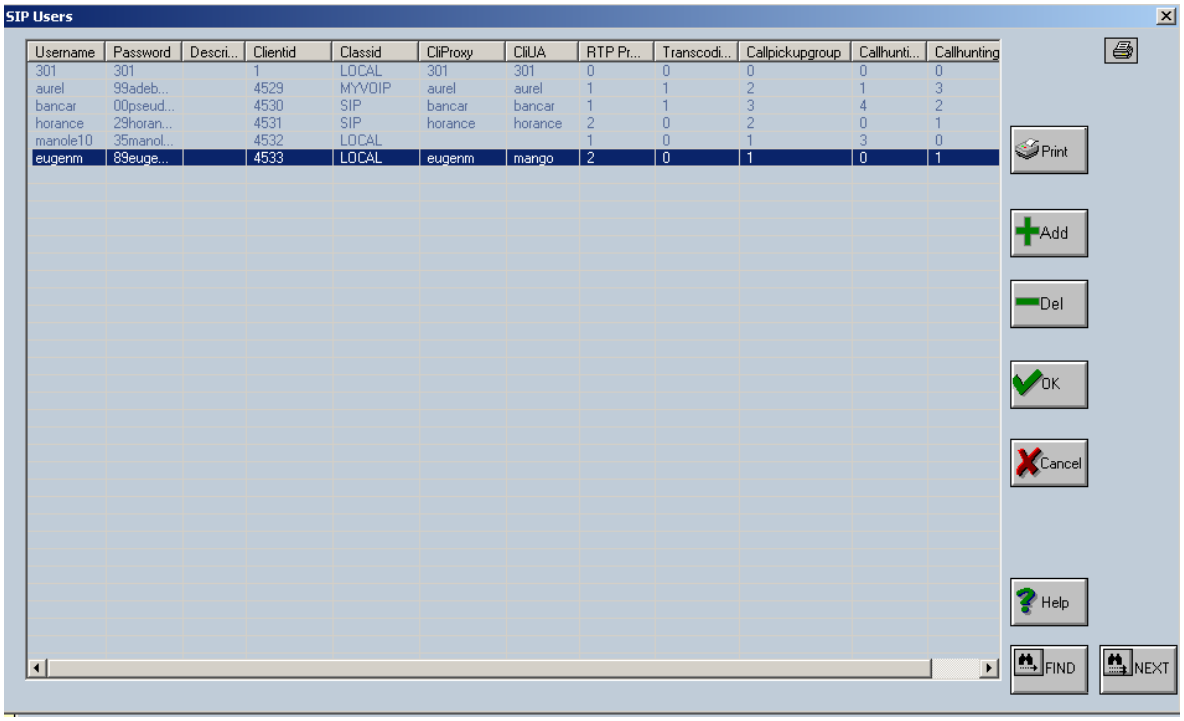


Figure 20 – SIP Users definition

This window - named "SIP Users" shows for each SIP user:
- SIP username
- the registering status is indicated in the "username" background - blue - for a user hat is registered and red - for unregistered user
- password - used together with SIP username for SIP user registration
- description - a relevant description for the SIP user;
- clientid - unique value allocated to each SIP user
- Class ID – the class to which SIP user belongs; it represents a direction name as it is defined in "Directions Names";
- CLIProxy - this is the ANI used for calls between SIP users.
- CLIUA - identity of the SIP user - for SIP outgoing calls (not the case between SIP users registered to the connected Voxi box), PSTN calls.
- RTP_proxy – is used when the SIP user is behind a NAT; the available options are: “Not Used”, “Used except same NAT”, “Always Used”; by default the selected option is “Not Used”;
Transcoding – enables/disables the transcoding option;
Call PickUp Group –it can be selected a value from 1 to 50 representing the PickUp group to which the user is allocated;
Call Hunting Group – when the call is directed to the user, and this one is not answered for several reasons - the call will be directed to another user; if this one also does not answer, then the call will be directed to a third user. The routing call algorithm is based on „Call Hunting Priority” parameter.
Note: in order to accomplish this feature - the users must be in the same hunting group!
Call Hunting Priority – represents the priority of the user inside the hunting group; the „0” value has the highest priority, and „10” the lowest priority.

The Buttons "Add" and "Delete" are used to add SIP users or Remove SIP Users.

To register a new user, click the "Add" button then the next window is displayed:

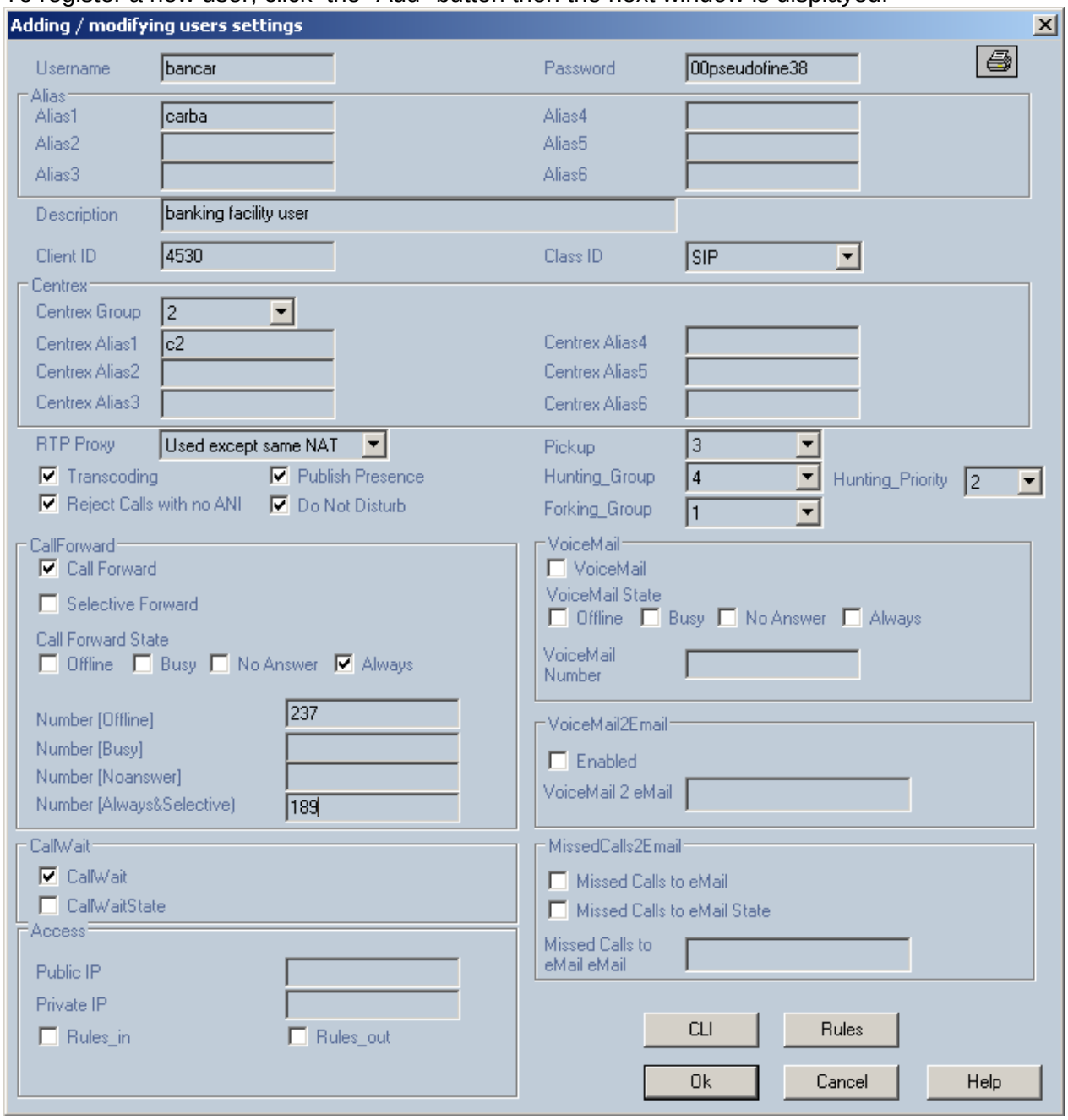


Figure 21 – Add/Modify settings for SIP Users

In the previous image picture you can notice all settings which are available for a SIP user. Most of them are available in the "Adding/modifying user settings" window.

In order to set SIP user identity settings click the "CLI" button located at the bottom right of the screen. For establishing rules for DNIS (number) or ANI (identity) – use the "Rules" button located near CLI.

For each SIP user the possible settings are:

- **SIP username**

- the registering status is indicated in the "username" background - blue - for user registered and red - for unregistered user

- password - used together with SIP username for SIP user registration

Alias zone:

- alias - includes the alias through which a SIP user client can be called; there are allowed a maximum number of 6 alias fields.
- description - a relevant description for the SIP user;
- clientid - unique value allocated to each SIP user
- Class ID – the class to which SIP user belongs; it represents the number of a direction as it is defined in "Directions Names";

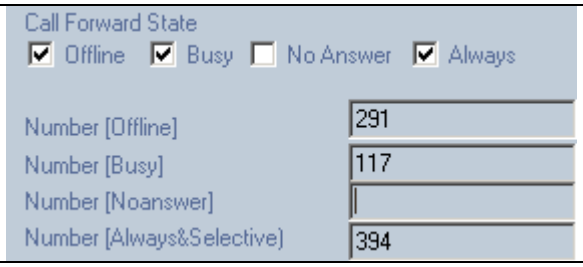
Centrex zone*:*
- Centrex Group – represents the "Centrex" group to which the SIP user belongs;
- Centrex Alias - from 1 to 6 - the same SIP user can have more alias which are all used for calling by the SIP users belonging to the same center group.
- RTP_proxy – is used when the SIP user is behind a NAT; the available options are: “Not Used”, “Used except same NAT”, “Always Used”; by default the selected option is “Not Used”;
- Transcoding – enables/disables the transcoding option;
- Reject Calls with no ANI – if is set then the calls to this SIP user without ANI are rejected;
- Publish Presence – allows to report the SIP user status to a presence server.

PickUP/Hunting Groups
Call PickUp Group – a value from 1 to 50 can be selected, representing the PickUp group to which the user is allocated;
Hunting Group – when the call is directed to the user, and this one does not answer for several reasons - the call will be directed to another user; if this one also does not answer then the call will be directed to a third user. The routing call algorithm is based on „Call Hunting Priority” parameter.
Note: in order to accomplish this feature, the selected users must be in the same hunting group!
Call Hunting Priority – represents the priority of the user inside the hunting group; the „0” value has the highest priority while „10” has the lowest priority.

Forking Group
Option for Sequential Call Forking, it works in two modes:
- when the same pair "Username/Password" are used for registration from different phones, then an incoming call from that user will be sent to each phone. Such a feature is useful when a certain user has more then one SIP phones in different locations, such at office or at home;
- several SIP users can be grouped in the same „Forking Group”; when a SIP user from such a group is called, then **all** phones from that group will be called.
Note: in each case or “forking groups”, the first user which answers will take the call.
Note: the forking group values are from 1 to 50.

Call Forward area*:*
Call Forward – allows to redirect calls when the SIP user belongs to one of the states for call forward;
Selective Forwarding – enables the selective redirection;
Call Forward State – this is a characteristic which contain one or more states which may activate the redirection
- „Offline” – the user is not registered;
- „Busy” – the user is in another call;
- „No answer” – the user does not respond;

- „Always” – all calls will be redirected.

The next four fields are the redirect numbers are filled for each possible situation („Offline”, „Busy”, „No answer” and„Always”).:	

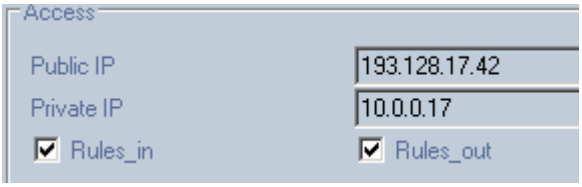
Call Wait area:
- "Call Wait" – from the administrator side it enables to send calls to a SIP user even when this phone is not in „idle” state;
- "Call wait state" – at user level. If you check it, then the „Call wait” option is activated on the phone side;

Voice Mail area:
Voice Mail – if is enabled then the voicemail box message is activated;
Voice Mail State – this is a characteristic which contain one or more states which activate the voicemail box:
- „Offline” – the user is not registered;
- „Busy” – the user is busy;
- „No Answer” – the user does not respond;
- „Always” – all calls are redirected to voice mail box;

Voice Mail Number – the voice mail number to which calls will be redirected;

Voice Mail2Email area:
Voice Mail to Email State – enables to send the voicemail messages to the specified email account;
Voice Mail 2 Email – the email account where the message are sent;

Access area:



Public – this field is set with the public IP address from which the SIP user is accepted;
Private – this field is set with the private IP address from which the SIP user is accepted;
Rules_in - if is enabled - the incoming rules set in "Rules" section are taken into account;
Rules_out - if is enabled - the outgoing rules set in "Rules" section are taken into account

In order to set the SIP identity the OAM user will press the "CLI" button:
The "SIP users - CLI settings" window is displayed.

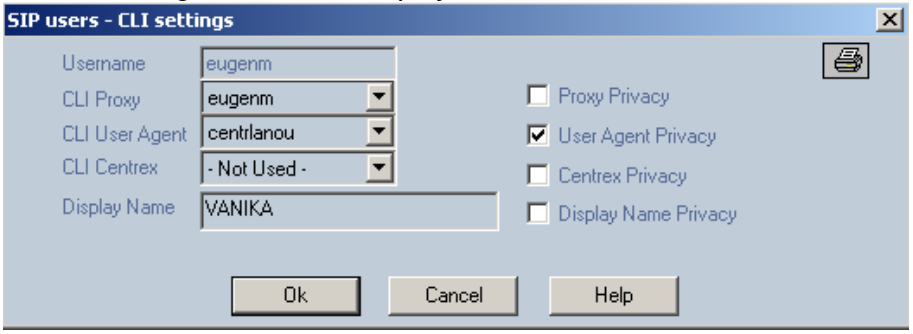


Figure 22 – CLI Settings for SIP Users

The available fields are:
- CLI Proxy - this is a characteristic used for calls between SIP users. This field contains a list of all defined aliases together with the SIP username ; the default value is the SIP username
- Proxy Privacy - by default this field is not enabled - case in which the identity is sent. If this field is enabled then - the identity is not sent.

- CLI User Agent - identity of the SIP user - for SIP outgoing calls (not the case between SIP users registered to the connected Voxi box), PSTN calls. This field contains a list of all defined aliases together with the SIP username ; The default value is the default username
- User Agent Privacy- by default this field is not enabled - case in which the identity is sent. If this field is enabled then the identity will not be sent.

- CLI Centrex - this is a characteristic (ANI) used between SIP users from the same "Centrex Group". This field contains a list of all defined Centrex groups
- Centrex Privacy- by default this field is not enabled,- case in which the identity is sent out. If this field is enabled then the identity will not be sent.

- Display name – the name attached to the user which will be sent to destination; this field is sent for SIP calls (as display name) and on E1-SS7 trunk (as presentation number).
- Display Name Privacy - by default this field is not enabled, case in which the display name is sent. If this field is enabled then the display name is hidden

In order to set the RULES for incoming / outgoing calls the "RULES" button is used:

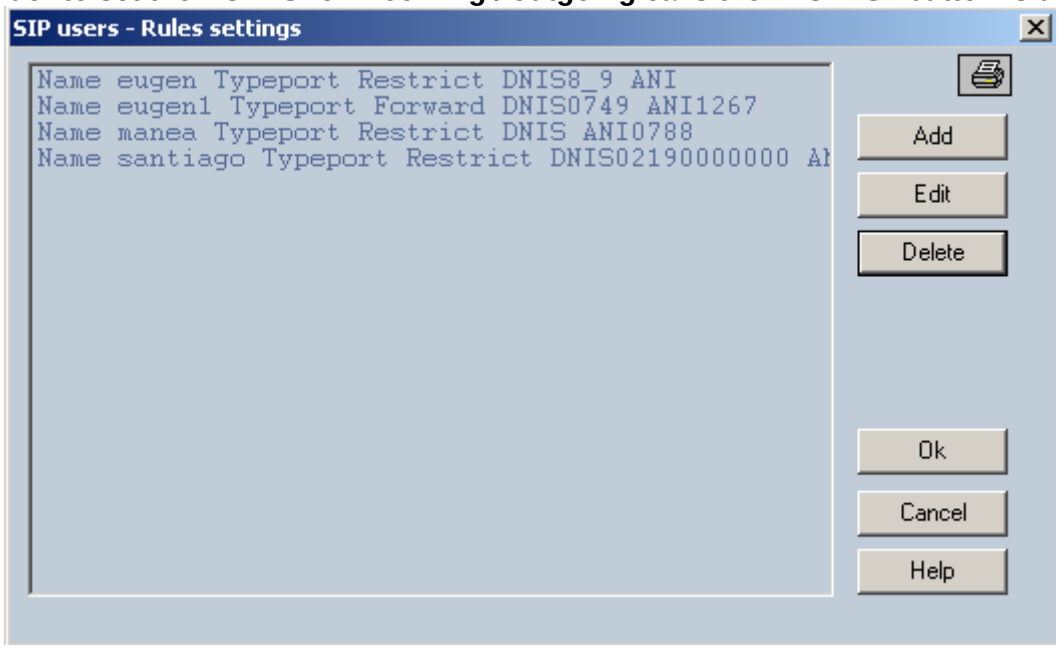


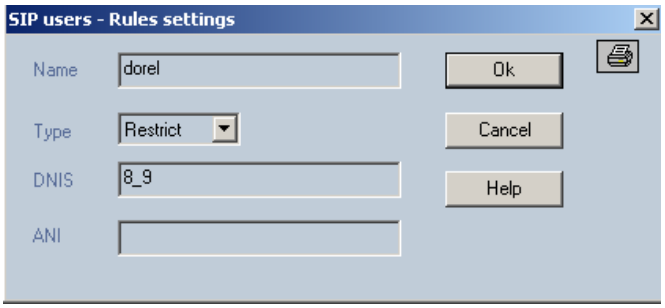
Figure 23 – File Transfer Manager

The "SIP users - Rules Settings" window contains a list of defined rules for the selected SIP user. Each line contains a single rule. The format of the rule is:
Name xxxx Typeport DNISyyy ANIzzz

There are there three options: “Allow”, “Restrict” and “Forward”, with those you can specify allowed, restricted and forwarded numbers for the SIP user.

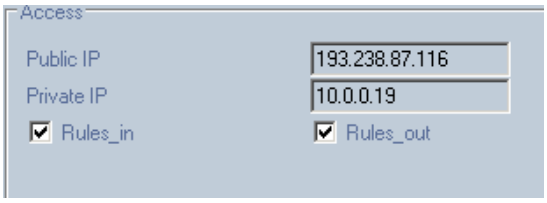
A rule can be deleted with the "Delete" button. "Add" button is used to add a new rule and "Edit" button is used to edit an existing rule.

The next window named "SIP users - Rules settings" is displayed when "Add" or "Edit" button is selected.



The rule interpretation depend on how rules settings are established in "Adding/modifying user settings" window:
- if "rules_out" is enabled then the DNIS - number - dialed by the SIP user is checked among the set rules
- if "rules_in" is enabled then the ANI – (identity) received bye the SIP user is checked among the set rules
For example if DNIS is filled with "0219000000" and "rules_out" is enabled and "Type" is "Restrict" then the specified number will not be allowed to be dialed by the SIP user.
If you wish to restrict access to number such "89" then a rule with DNIS="89%" will be created - the "%" character will be interpreted as ANY DIGITS after 89.
If you want to specify a combination like "8x9" where 'x' means ANY DIGIT, then you'll use "8_9" (the underscore "_" MEANS ANY DIGIT).

For incoming rule - you can restrict access from specified ANI sources.



2.4.17 - File transfer

You can transfer files between the VOXI equipment and your PC. By clicking on the associated icon ("), the window for file exchange labeled “Traffic files manager” will be displayed:

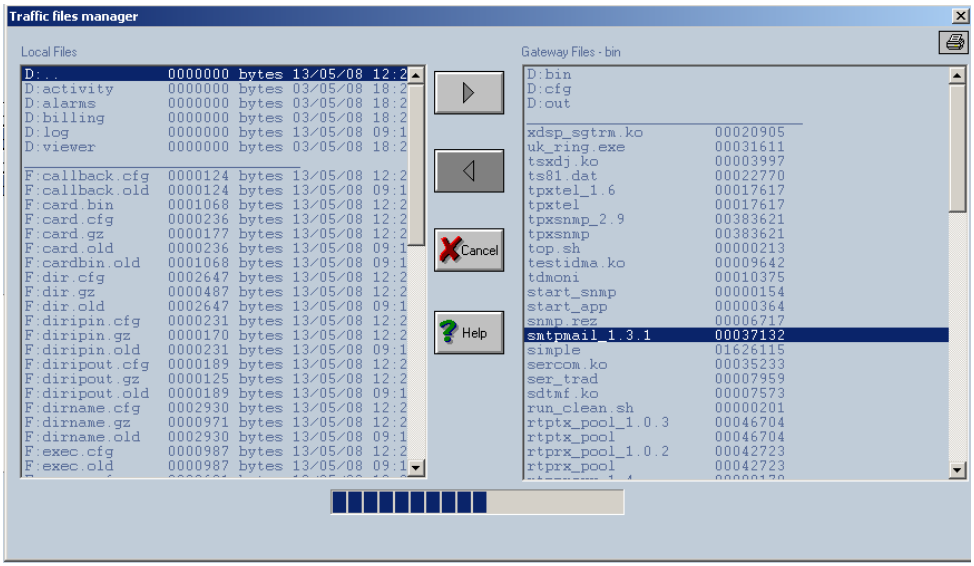


Figure 24 – File Transfer Manager

"**Gateway Files - current**" – the left panel of the window shows up the structure of files and folders from the OAM program (the directory of your system associated with the equipment).

You will see directories named "Alarms", "Billing", "Activity", "Viewer" and “Log”. You can change between those directories and the root directory of the connected system.

“**Gateway Files – cfg**” - the right panel of the window shows the files and folders structure located on the TOPEX VoiBridge equipment that is connected to the PC.

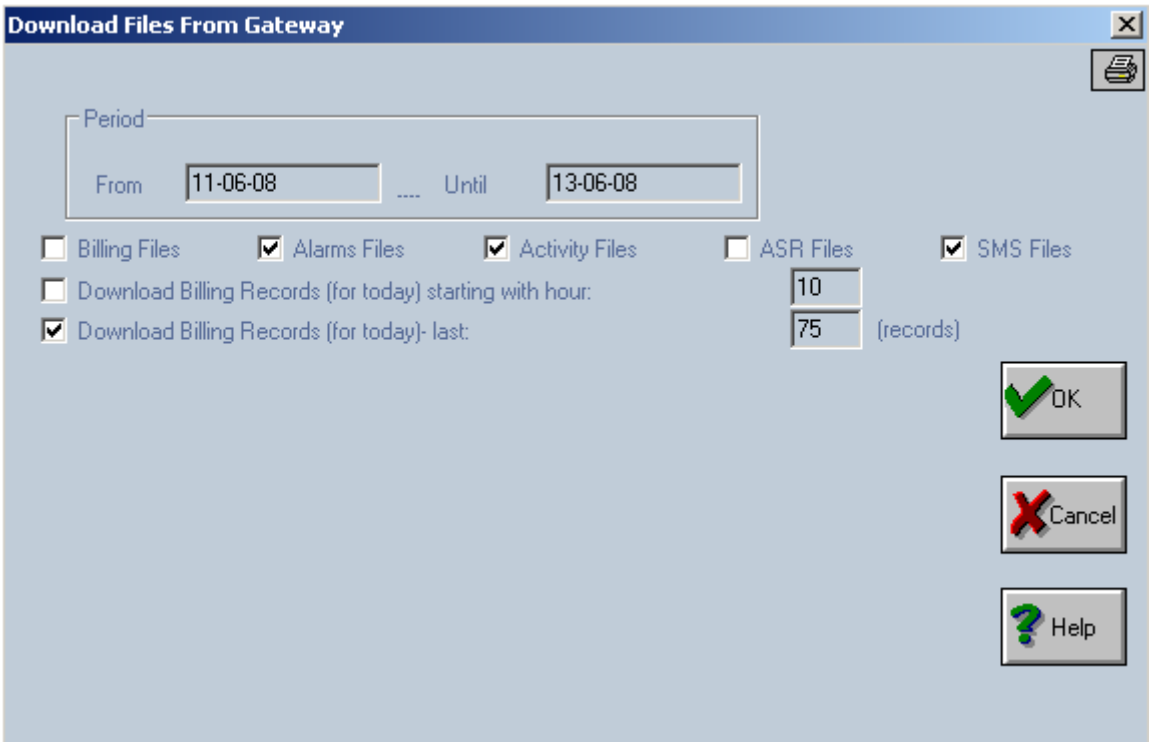
On top of this panel you should see on the remote gateway three directories: “bin” (where is located the main telephony application), “cfg” (where the configuration files are stored) and “out” where the output files are written by the system. These files contain recordings for alarms, billing, ASR, monitoring (activity) etc.

Note1: The file transfer is performed in “COPY” mode, not in the “MOVE” mode. This means that if you download a file from the VOXI equipment to the OAM computer, the original file is left intact on the remote TOPEX system.

Note2: It is mandatory to download files to appropriate destinations: alarms (*.alr format) and asr (*.asr format) in "Alarms" directory, billing (*.tax format) in "Billing" and activity (*.mon format) in "Activity".

2.4.18  - Auto downloading files

Automatic transfer of file is allowed in the connected state. Click the icon to open the window “Download Files form Gateway”



First you define the period of interest (for instance from 11 to 13 June) and then you can select the file types you want to download: “Billing”, “Alarm”, “Activity”, “ASR” and respectively “SMS”.

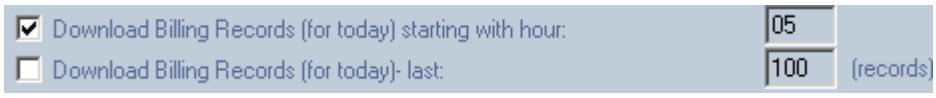
Auto options for billing files:

"**Download Billing Records (for today) starting with hour**:" - used to download out of the billing file for the current day only the records starting with the time (hour) you want. The specified hour may be changed.

"**Download Billing Records (for today) last: 100 (records)**" - used to download the last xx0 records also from the billing file of the current day. The number of last records to be downloaded may be changed from the default value of 100.
For any of those two options the resulted billing records will be automatically displayed.

Note: The last two options for downloading billing records are mutually exclusive; they cannot be set at the same time.

or



2.4.19 - Clean HDD Space

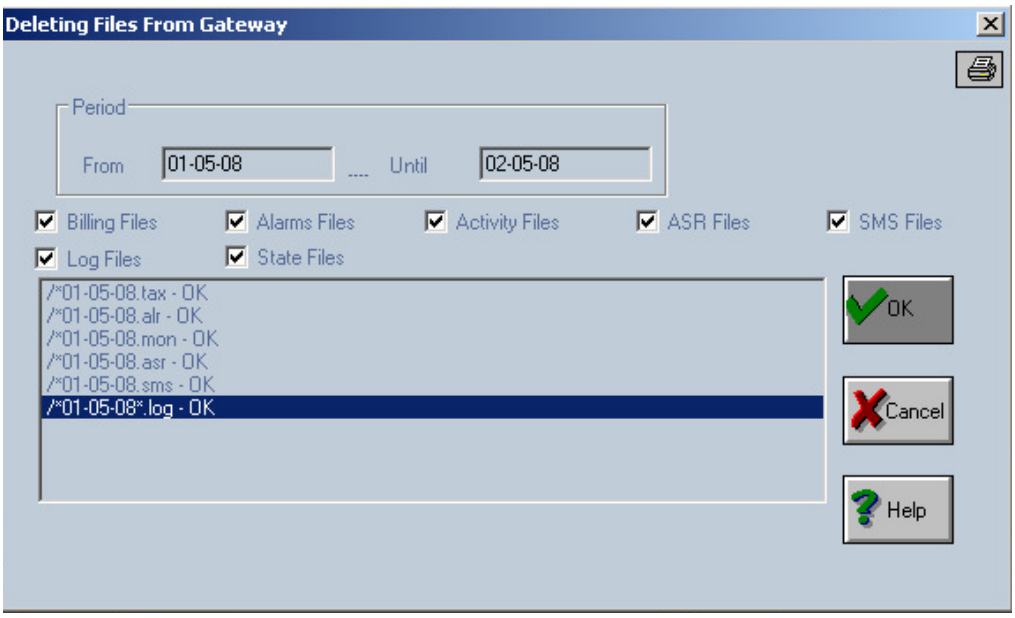
This option could be needed only if the equipment has been in use for a long time without connection to the OAM program.

In this case, it has stored many large files on the hard disk and you may need to clear up space.

You must choose the associated icon ("Clean HDD Space").

This operation is protected by a password, which is “topex”

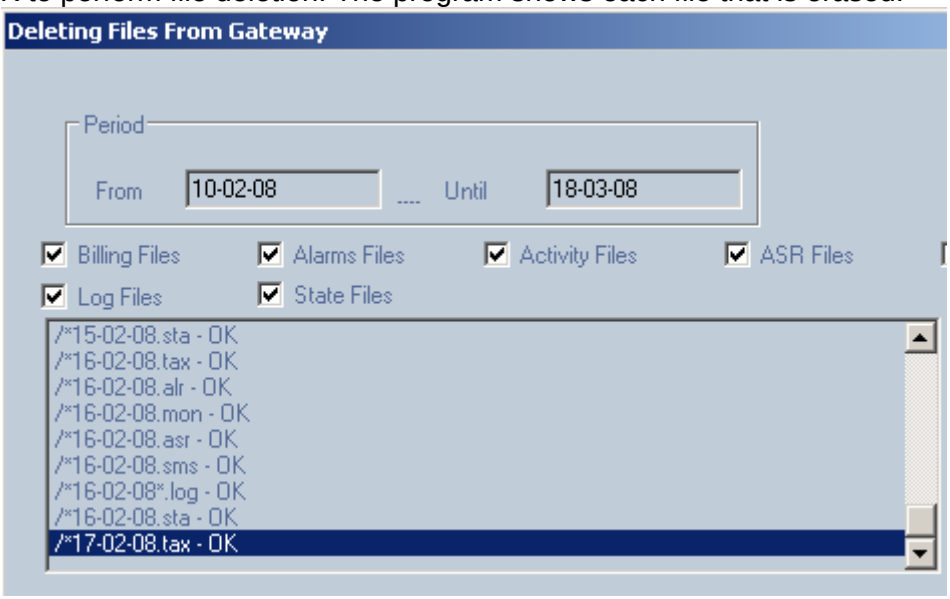
The window "Deleting Files From Gateway" allows you to choose a time interval and the type of files to be deleted:



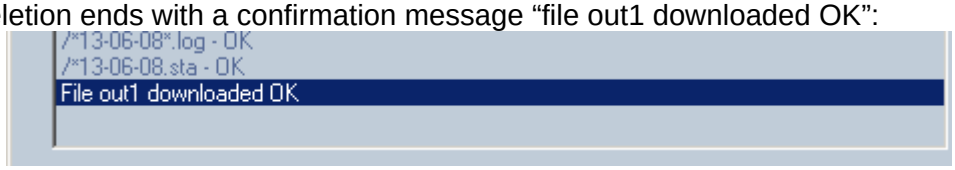
The type of files can be:

- Billing files (extension "*.tax");
- Alarms files (extension "*.alr");
- Activity files (extension "*.mon");
- ASR files (extension "*.asr");
- SMS files (extension "*.sms");
- Log files (extension "*.log"); these files are created by the gateway application when "Validate Debug" and "Validate Saving to File" are checked in "Parameters for ALERTING" window;
- Status files (extension "*.sta"); these files are used to store the state for all ports (by state of a port we mean if it uninstalled, free, busy or alarm). These files are used by the web application.

Click OK to perform file deletion. The program shows each file that is erased:



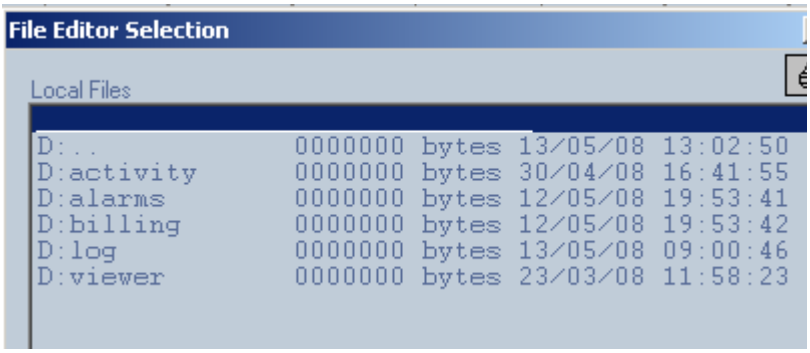
The deletion ends with a confirmation message “file out1 downloaded OK”:



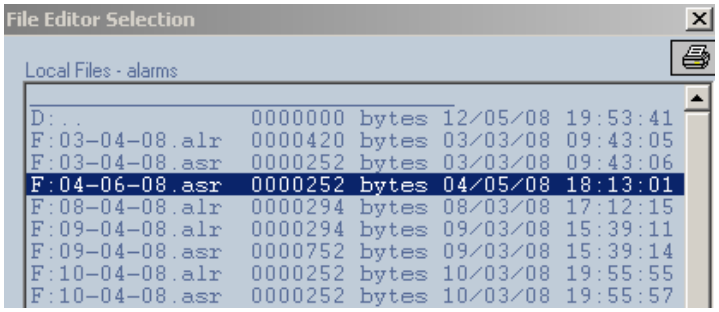
2.4.20 - File Editor

This option ("") allows you to edit the configuration (.txt) files that can be loaded later into the TOPEX system.

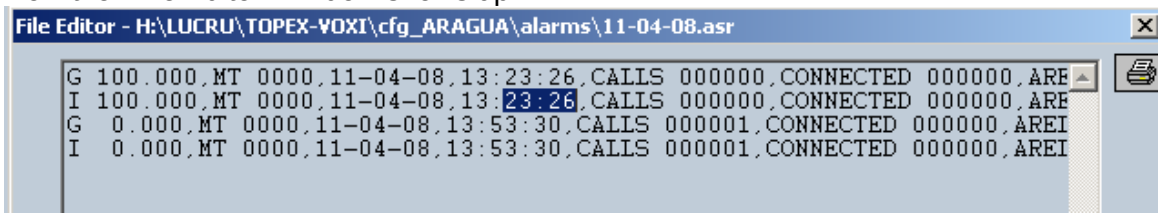
The first window “File Editor Selection” allows you to navigate through the subdirectories structure in the system root directory in order to select the file you want to edit.



Go to the directory of interest, for example with ASR and alarms files. Double click the file you want to see or edit::

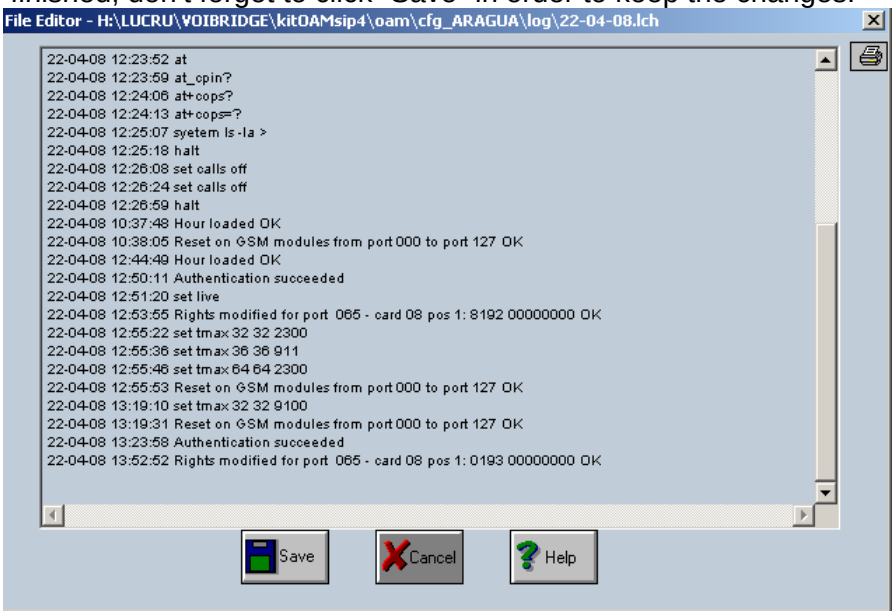


Now the “File Editor” window shows up:



There is a limit for the size of files to be edited; they should not be over 65,535 bytes in length.

When you are finished, don’t forget to click “Save” in order to keep the changes:



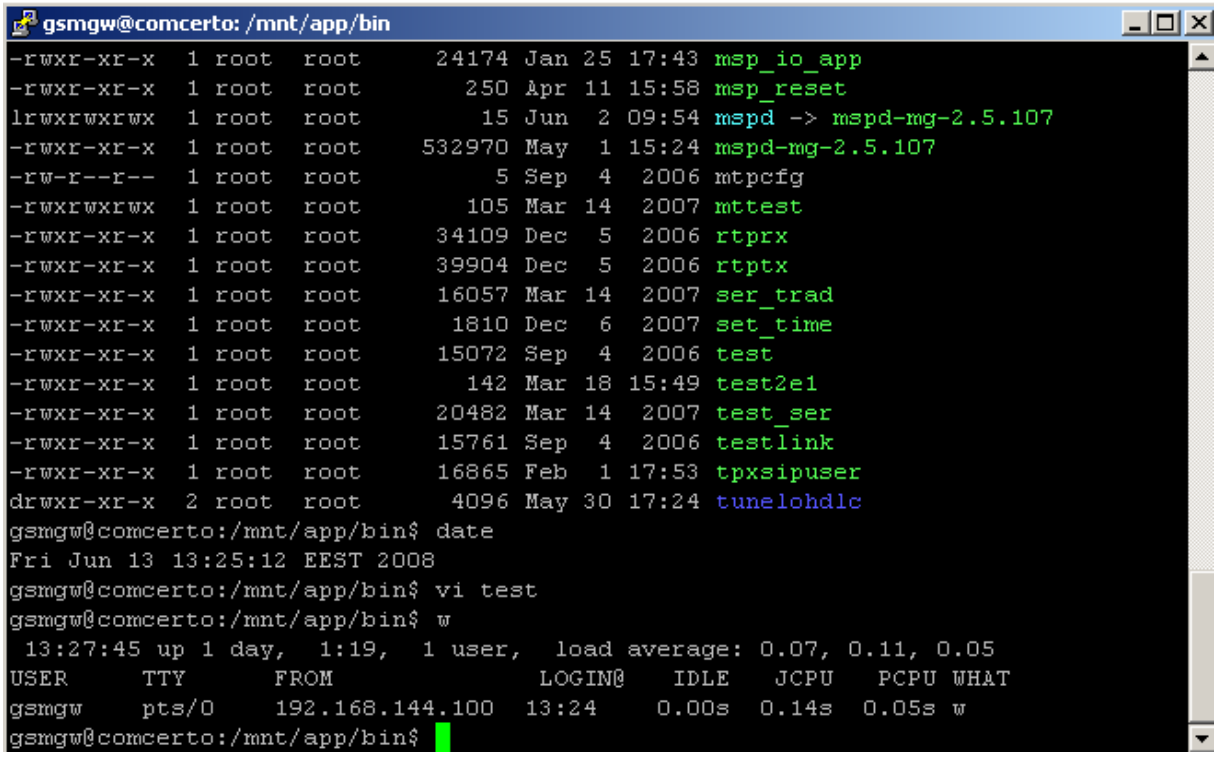
2.4.21  - PUTTY

Click on the Putty icon "  " to launch directly the "putty" application.

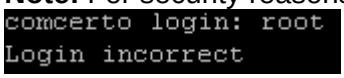
In a state of connection between "gwconfig" program and a remote TOPEX equipment, Putty will connect on port 2222 to the same IP address where "gwconfig" software is connected. The username will be sent automatically (it is taken from the name used for logging into the VOXI equipment).

With Putty you may administrate and configure directly the remote VoiBridge gateway. You have real-time access to the log file, you can see what processes are running, the level of HDD usage, and all sort of low-level items that cannot be seen from the OAM program.

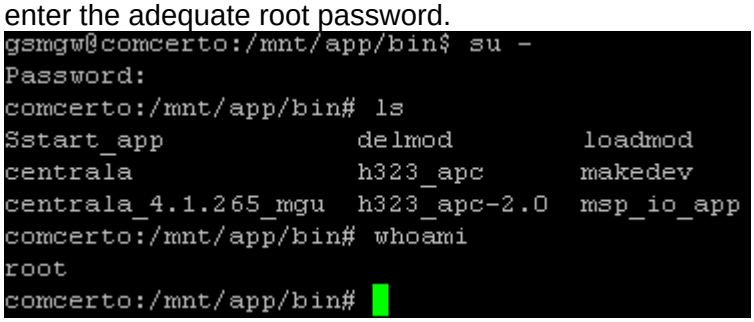
For debugging purposes, you can see if the telephony application is running correctly, etc.



Note: For security reasons, the direct logging as „root" cannot be performed.



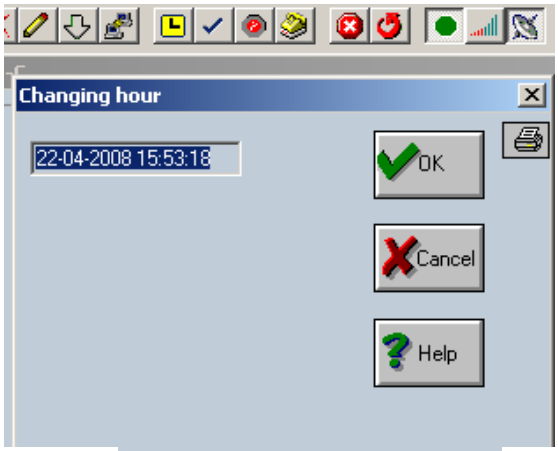
However, after login as user you may perform switching to root with the command su-, if you enter the adequate root password.



2.4.22 - Loading hour

You can upload the time into the TOPEX system. Loading hour is very important because you modify the gateway clock: time and date. So this action is protected by password. This password is always "topex".

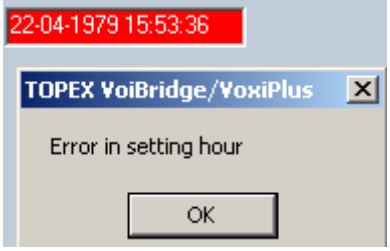
If you enter the correct administrator password, the window “Changing hour” shows up.



The default value shown in its field is the date and time that are set on the computer where the OAM program is running.

You may leave this value or enter a different one of your choice.

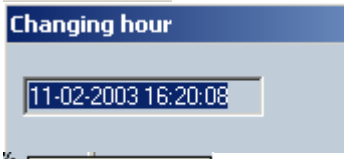
Before sending the modified time value to TOPEX system, the program performs a validity check.
If you enter for the time and date values that are NOT VALID you will receive the error message shown to the right and you must start the “Loading hour” procedure al over again!



Note1: this function performs a Synchronization, that is it sets the same time on the remote VOXI box and on your PC, as shown in the following example:

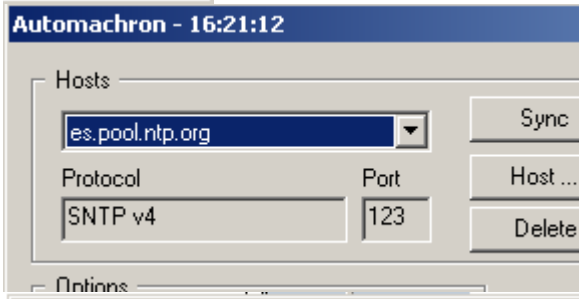
Local time before Sync:

Use **Loading Hour** to change the date/time:

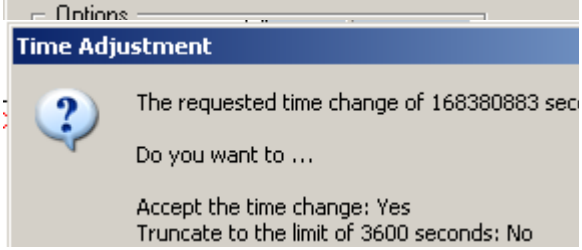


After the change is performed:

Now use a NTP client to get the current UTC time for your computer:



Since the time difference is very big, the NTP request asks for confirmation



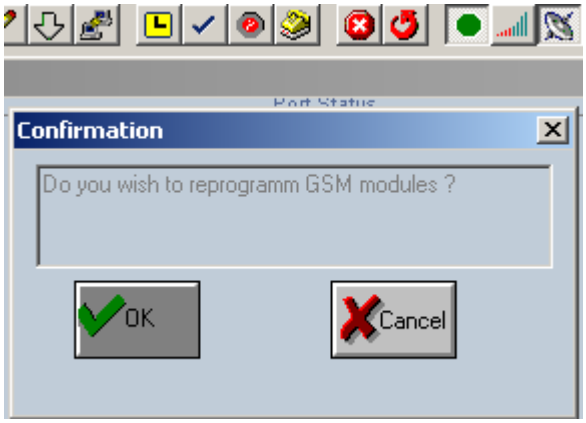
The local time of the PC has been restored:

But on the remote VOXI gateway the time is still set to the old value:
Use again Loading Hour to synchronize!

Note2: The time you have uploaded will hold true only until the first reset of the TOPEX system. After reset, the time and date will revert to the previous value (that was in the BIOS of the TOPEX equipment.

2.4.23 GSM - Reprogramming

This reset command ("") is useful in situations when there is need to refresh the settings for all the GSM modules.



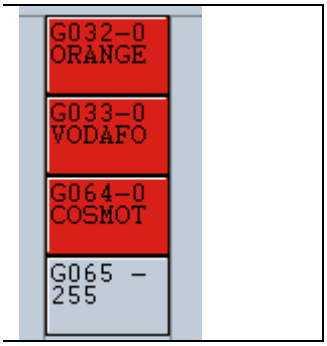
This GSM refresh must be issued to allow the proper setting of the receiving or sending of the identity through GSM modules and also the setting of the audio level.

The program shows you a confirmation message:

Press OK to proceed

All installed GSM become colored red for a few seconds, showing they are not available

Then they will start operating again, with the new settings



2.4.24  - Alerts

This option ("  ") establishes sending out the alerts, recording of ASR values and enables several debug and testing functions. Alerts are used in case of alarms or if ASR values fall below a selectable level.

In case of occurrence of different pre-programmed events, the TOPEX system can issue warnings (alerts) and also send out a description of the events. The alarms may be sent to the administrator of the equipment in three ways:

- **By e-mail (a warning message that has attached the alarm files);**
- **By SMS;**
- **As a call on the mobile phone of the operator (several beeps).**

The "Alerts" command shows the window for setting up the parameters for the alerts. This large window has several configuration areas:

Alert zone" - is used to activate (enable) the transmission of the alerts. It includes four smaller areas: Parameters, Dial Alert, SMS Alert and MAIL Alert.

The "**Parameters**" area contains the following fields:

- **Alert activated** - must be checked in order to activate the alerts;
- **ASR alert limit** - is used to set a level for activating ASR alert transmission when ASR value falls below the specified value. The ASR value is read at time interval specified in "Time period for checking";
- **Alarms for alerting** - validate the type of alarms for which alerts are transmitted;
- **Timer for testing alerts** – field used for testing alerts. For a "0" value in this field the alerts will be sent only in real situations.

"**DIAL alert**" - is used if you want the alerts to be sent via phone calls. You must enter the phone number to be called upon alert. You may also set the number of beeps (1, 2, 3 etc) to be sent out in case of alert.

"**SMS alert**" – used in case of alerts by sending a SMS message to a mobile phone. You must enter the phone number to be alerted by SMS. You can also change the content text of the SMS alert message.

"**MAIL alert**" is used in case of sending e-mail alert messages to an email address. In case of activation of this kind of alert the following e-mail related parameters must be set: "Gateway address", "Mail From", "Mail To", "Subject", "Server" and "Port".
You also have the facility to send the alert message with the current alarm file for the TOPEX equipment as an attachment.

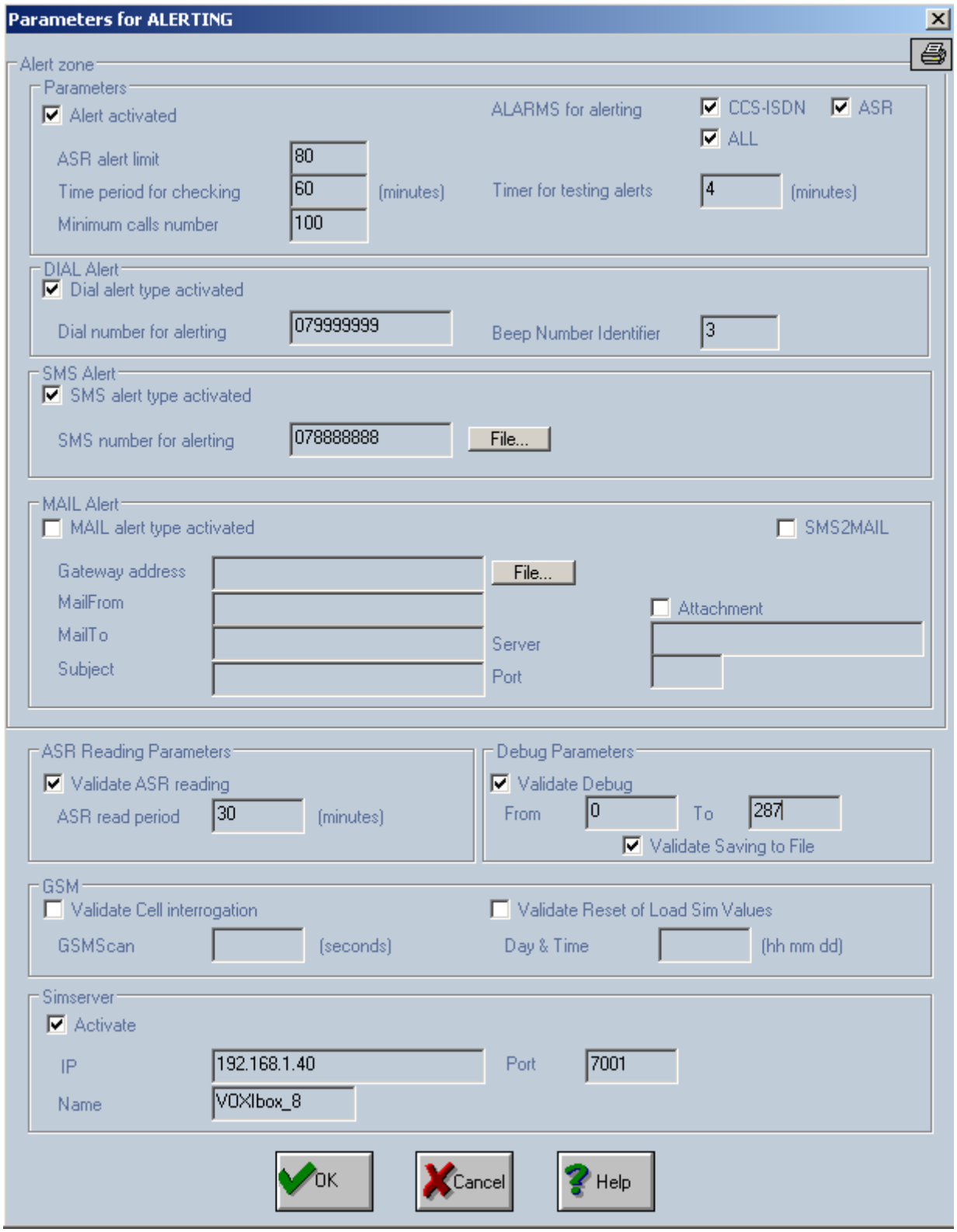


Figure 25 – Alerts window

Note: *The following fields must not contain spaces inside the value: "**Gateway address**", "**MailFrom**", "**MailTo**", "**Subject**" and "**Server**"*

"SMS2MAIL" - this option is used to forward all incoming SMS to an email address. This option is placed here because it works together with the settings established in the "MAIL alert" zone. The content of the SMS will be sent to the address specified in the field "MailTo".

"**ASR Reading Parameters**" – establishes the ASR reading procedure. First, to activate the ASR reading process the option "Validate ASR reading" must be checked. Then the interval (in minutes) for reading and saving ASR values is set in "ASR read period" field.

"**Debug Parameters**" - used for the debugging process. This facility is intended to help debugging the TOPEX system: log files for the equipment are created and respectively displayed. If the "Validate Debug" box is checked then all activities on ports specified in range "From" and "To" is displayed. If "Validate Saving" is also checked, this information will be saved in log files on target.

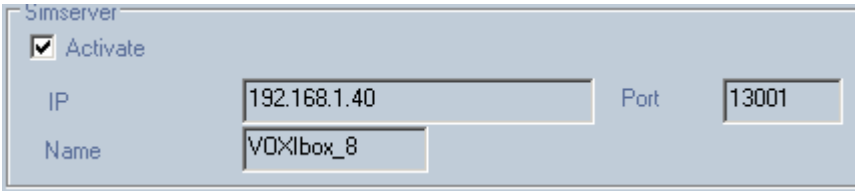
"**GSM**" - is used for:

1) "**CELL, LEVEL and Channel**" interrogation. The GSM scanning is launched by checking the option "Validate Cell Interrogation". The "GSMScan" value is the value in seconds at which each GSM module is searched upon cell, level and channel.

2) The reset of the "**Load Sim**" values - the user can enable the option "Validate Reset of Load Sim values" and specify the "Day and Time" at which those times of SIM using values are resetting. User will fill the time moment in "hour minute day" format. These values are separated by space character.

There are the following text limits at the fields: "ASR alert limit", "ASR read period", first port and last port - 3, "Time period for checking" - 4, "Minimum calls number" - 4, "Timer for testing alerts" - 3,"Beep Number Identifier" - 2, "Gateway address", "Mail From", "Mail To", "Subject" and "Server" – 40 characters.

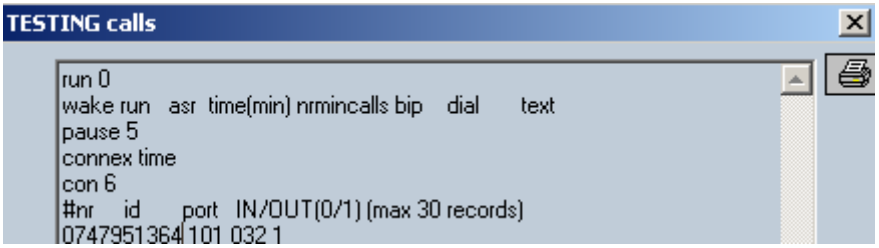
"**Simserver**" - is used in case of VoiBridge equipment which is working with GSM cards Simserver.



The "simserver" zone contains an activation checkbox "Activate", the IP of the simserver and the port to connect to (default value is 13001). Finally the "Name" is the gateway identification name, name which is also set in simserver to recognize the gateway client.

2.4.25  - Testing calls

This command allows you to perform phone calls for testing the equipment. If you click this button, the window will appear where you can enable and define phone test calls that will be performed by the TOPEX equipment.



First, you must enable this feature, with the option "run 1" on the first line.
The default is set to "run 0", which means the test calls are disabled. For each test call two timing parameters: **"paused xx"** and **"con yy"**will be used. These represent the time interval (in seconds) between two successive attempts (xx) and respectively the time to keep the call connected (yy).

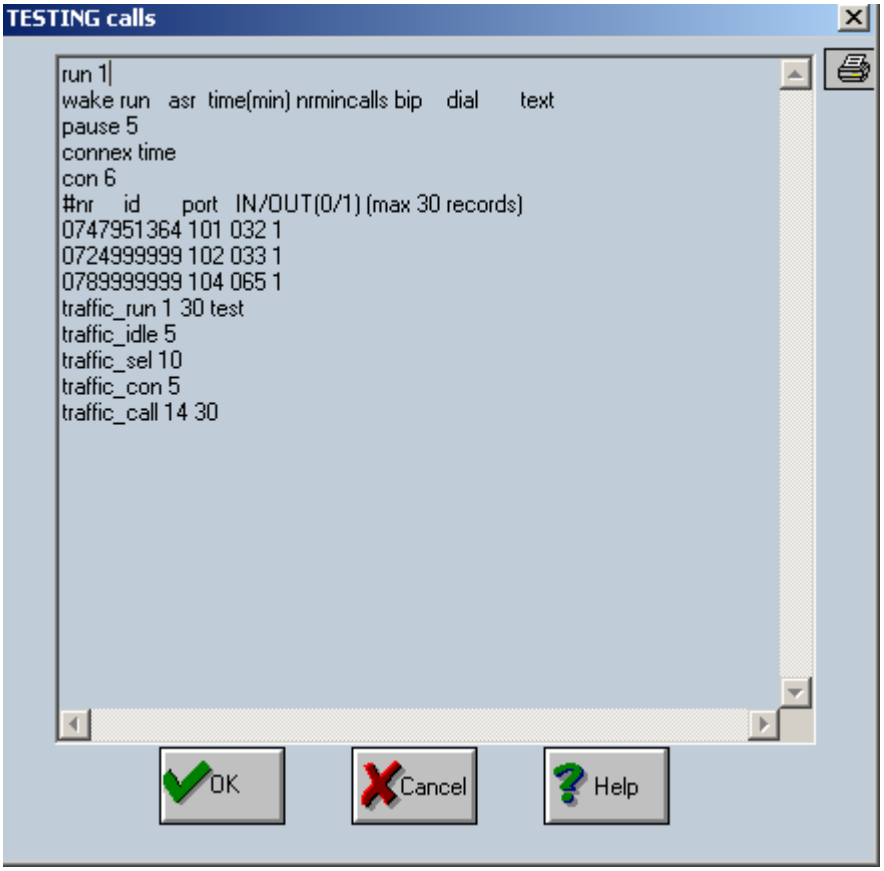


Figure 26 – TESTING calls enabled

Then the lines that begins with "#nr" contain the definitions of the test calls (maximum 30 records). Each record line will contain the number to be dialed, the identity of the call, the caller port and a value that may be 0 or 1.

If the value in a record is '0' the phone number will be first analyzed according to the trunk definition (the direction that includes the specified port) and then the routing rule will be applied (the routing table). If the value in the record is '1' the phone number will be first analyzed only according to the routing table (it will be sent out directly from the port!).

Example: A test record can be "0765999999 103 064 1". It means an outgoing call with number '0765999999" and identity "101' is sent on GSM port number 064.

Note1) a good method for testing calls and to analyze the routing process is to add a subscriber card and to add a record in the "TESTING calls" window with an installed port from the subscriber card (that port must have also the option of making calls). For example if the card 15 is a subscriber card then a record for port "120" will be: *"0700000000 120 120 0"*
Note2) for making test calls on E1 trunks or GSM cards an example was giving before:
For sending the number digit by digit a value "0000" must be used to fill the "Signaling" field for the direction that contains the test port.
Note3) test calls are very useful in finding the available number of GSM channels. Test calls are simultaneously generated towards the vocal messaging. After the test calls sequence is started, the number of connected calls can be observed by using "Live monitoring" facility.
For instance, below you can see the results of performing three simultaneous test calls on GSM modules with SIM card for different operators:

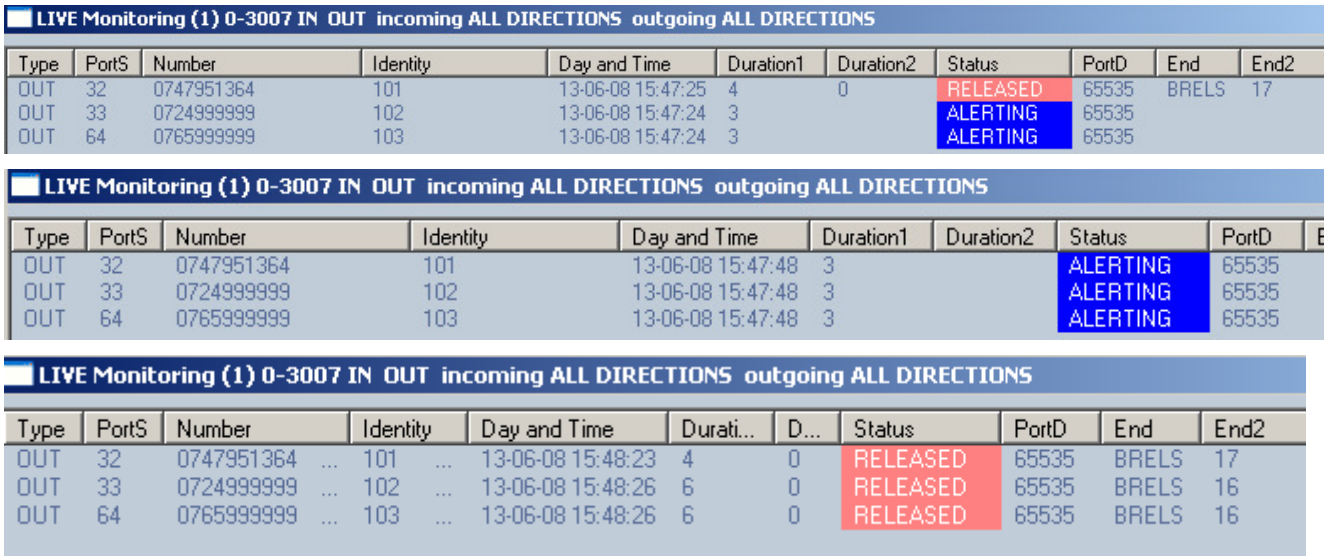


Figure 27 – Live monitoring of 3 mobile test calls

Delayed test Calls
Moreover a new method can be used to generate **non-simultaneously** calls. Previously, the time delay between attempts and the time for keeping the calls connected was the **same** for all test calls from the list. This new method allows the user to create a list of test calls that have **variable** time delays between them. Also, not all calls from the list are launched at the same time.

This new method includes the following parameters: (the lines beginning with "#" are comments, used just to show you the structure of commands for test calls):
#traffic_run 1/0 nr_simultaneous_calls(max 4 calls) call_type
traffic_run 0 1 test
#traffic_idle t_idle(seconds of idle time betwen calls) rand_add(random added seconds)
traffic_idle 10 5
#traffic_sel t_sel(seconds of selection time for each call) rand_add(random added seconds)
traffic_sel 5 3
#traffic_con t_con(seconds of connection time for each call) rand_add(random added seconds)
traffic_con 1 2
#traffic_call nr id
#traffic_call 5113 100
traffic_call 5112 100
traffic_call 5777 100

You must notice that each line contains the prefix "**traffic**_" concatenated with "**run**", "**idle**", "**sel**" ,"**con”** and "**call**".

In order for this new method to work, the user must create a direction with the name "SENDCALL" (icon "Directions Names"). The "Type" field must be completed with "DIR" in "Calls directions".

All those test calls are generated from this fictive direction "SENDCALL".

- "**traffic_run 0 1 test**" – to disable this second method for launching calls you type "0" after "traffic_run". If you want to enable the method you must type "1".

The second number specified in this line is the number of test calls that can be generated at the same time (in the example above it is "1"). The list with the test calls is described by the lines beginning with "trafic_call".

The "test" word is used to specify the tonality heard by the called party (in this case a flashing tone).

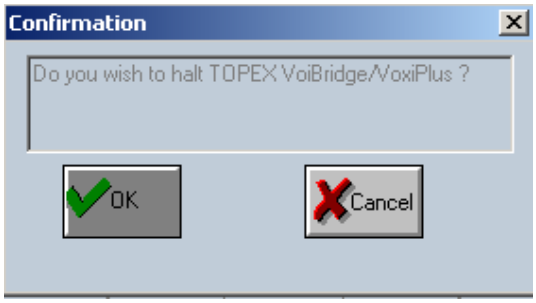
- "**traffic_idle 10 5**" – specifies the time to wait before to launch a new test call. The first digit (in this case 10) means the number of seconds to wait. To this value you may add the next value (it can be present or not), which means a range for a random value. The random number can be between zero and that value;
- "**traffic_sel 5 3**" – specifies the value used to simulate the selection time for test calls. Here also the first digit (in this case 5) means the starting number of seconds to simulate the selection time. To this start value may be added the next digit (which can be or not present) for randomizing. In this example, “three” means a random value between zero and three. Thus the number used to simulate the selection time for test calls will have get random values between 5 (5+0) and 8 (5+3);
- "**traffic_con 1 2**" – specifies the value used to simulate the conversation time for test calls. The first digit (in this case 1) means the starting number of seconds to simulate the conversation time. To this start value may be added the next digit (which can be or not present) for randomizing. In this example, two means a random value between zero and two. Hence, the value used to simulate the duration of the test calls will take random values between 1 (1+0) and 3 (1+2);
- "**traffic_call nr id**" – specifies the number ("nr") to be dialed and the identify ("id").

2.4.26  - Halt and Reboot commands

Those two commands are similar to the LINUX commands.

 - The first command, **HALT**, is equivalent the “halt” command in Linux and stops the machine. This assures normal shutdown of the VOXI box.

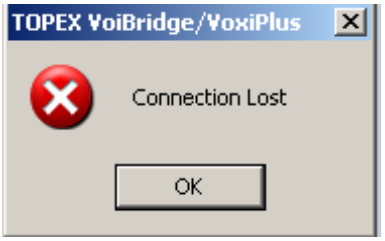
 - The second, **REBOOT**, restarts the machine.



The program asks you for confirmation.

Click OK if you really want to halt or reboot the remote VOXI gateway

In both cases, you will see the message “Connection Lost”, because the VOXI box is no longer online.



After reboot, you should perform again “Connect..” in order to re-establish the link between the OAM program on your PC and the remote VOXI gateway.

You should always use HALT before powering down the VOIX equipment.

After HALT command, the TOPEX system may be safely disconnect from the power supply. Incorrect shut down of the equipment may cause damage of the file system and other problems such as an increased delay in startup due to the file system check.

2.4.27 - Status monitoring

For viewing online the state of the TOPEX VoiBridge (while in the connected state): you can select the corresponding icon (") to start monitoring the system.

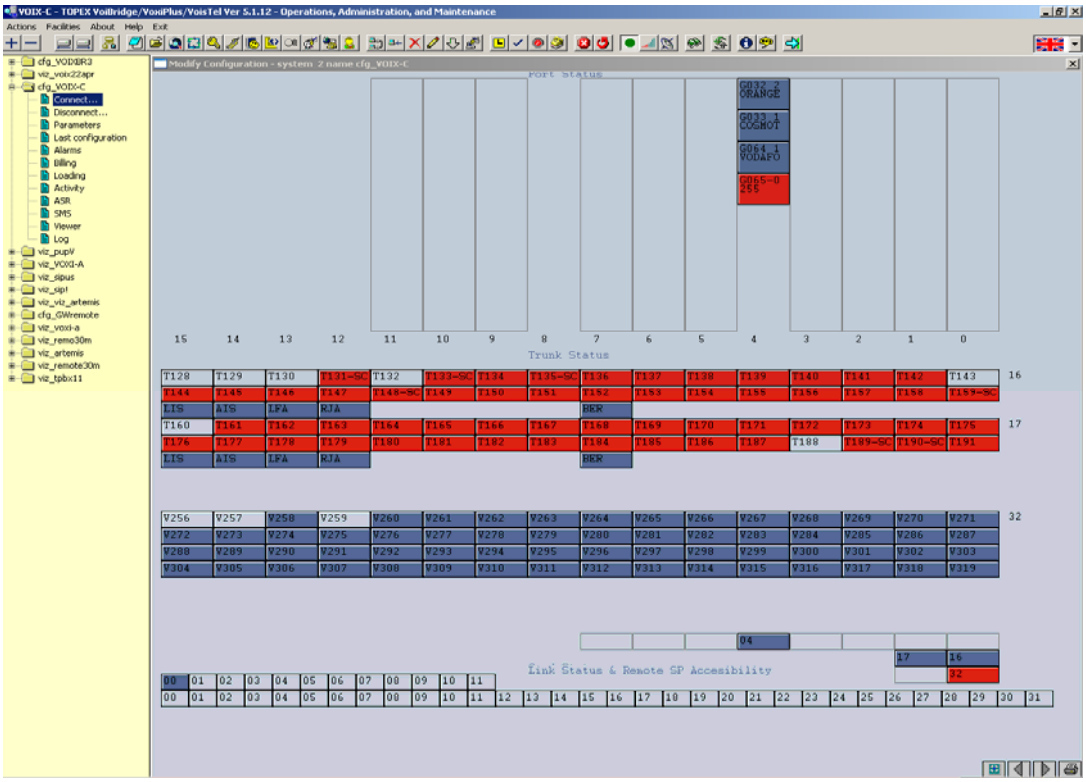


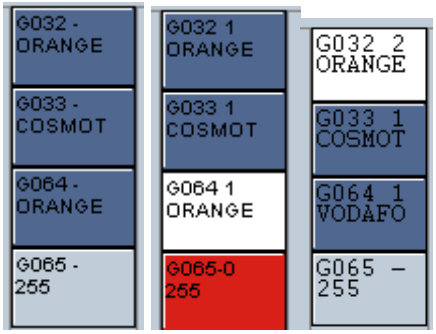
Figure 28 – Status monitoring started

After the monitoring process is started then the icon will be displayed inside a gray rectangle "  ". If you want to get out of the state of monitoring the TOPEX equipment, you stop the interrogation by clicking again the appropriate icon from the toolbar.

While in monitoring state, a command to view the state will be sent to the TOPEX system at regular intervals (5 seconds).

The significance of colors of ports is detailed in chapter 5.1 – Menu Description.

Colors for ports (port status):
white color – shows which port is opened (used)
red color - shows which port is in alarm . There are two situations for GSM positions: first situation when no SIM number is shown (only a line - can be viewed) and the second situation when the SIM number is displayed. For the first situation there are a few possibilities: there is no SIM card physical installed on that GSM port or PIN code is incorrect or the GSM module is defective.



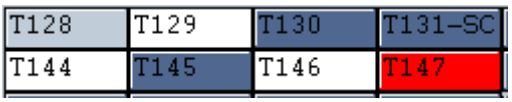
For the second situation the cause is that SIM module is not registered into a GSM network: antennas are not connected or there is congestion in GSM network or SIM card is inactivated in GSM network or there is a problem in radio frequency section.

dark blue color – indicates an available port.
light blue color - shows an invalid position. Invalid position means either that the "Installed" field is not set or there is a card error. Card errors will show up in red color in the Card Status area at the bottom of the screen.

Card error will appear in the absence of an real card in the equipment (but which is present in OAM configuration) or when the card is present, but it is damaged and therefore cannot be interrogated by the processor card.

For E1 trunks (in case Voxi boxes equipped with E1 interfaces) you can see alarmed channels and the situations of possible alarms on trunk: LIS, LFA, LMA, AIS, RSA, RJA, BER.

For a trunk type E1-ISDN, the indication "SC" is used to display the status of the signaling channel: when link is up the dark blue color is used; the red color is used to indicate that the link is down (LAYER2 alarm); the white color is used to indicate that the system is performing a restoration of the link.



At the bottom of the screen, for Card Status area the following colors are shown: dark blue for installed board, light blue for not installed cards and red for alarm on that card.

2.4.28  - Cell, Level and Channel info

When the program is in connected state with TOPEX VoiBridge equipment you may select this option from the toolbar by clicking the icon "CELL, LEVEL and CHANNEL Info". This option will display info about the mobile interface: Cell ID, signal level and GSM channel (frequency).

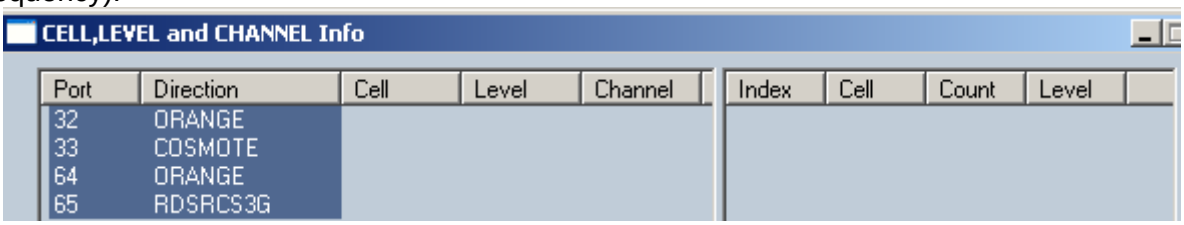
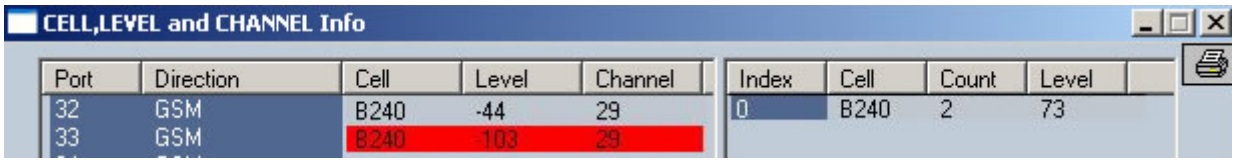


Figure 29 – Cell, level and channel info

When the cell, level and channel monitoring process is started then the icon will be displayed in a gray rectangle.

The window "CELL, LEVEL and CHANNEL Info" will show a list of all mobile modules (modules that have the "Installed" option set in the "Category" field and for which there is no alarm indication on that port).

Depending on the type of mobile module and the features enabled, the ports with active SIM will show values in the fields "Cell", "Level" and "Channel": ID of the mobile cell, such as B240, signal level in dBm (-44) and GSM channel (29).



The right side of the “Cell, Level and Channel Info” window shows statistics for each cell to which mobile modules are registered. For each cell the total number of modules connected is shown in "Count" and the average signal level is shown in the field "Level". The column “Index” is simply an index of the recordings present in this window: 1, 2, 3 and so on.

2.4.29  - Live Monitoring

If you want to view online the calls routed by the TOPEX VoiBridge equipment, just click on the ("  ") icon to start live monitoring of calls through the equipment. If the calls monitoring process is started then the icon will be displayed inside a gray rectangle.

Following the validation of the icon, several types of monitoring windows can be displayed:

a)- LIVE Monitoring (n), where n may be from 1 to 4
b)- LIVE Monitoring – Report on DIRECTIONS
c)- LIVE Monitoring – IP signaling

a) The first type of window, “LIVE Monitoring (n)”, is used to see on line the calls going through all the range of ports of the Voxi gateway.

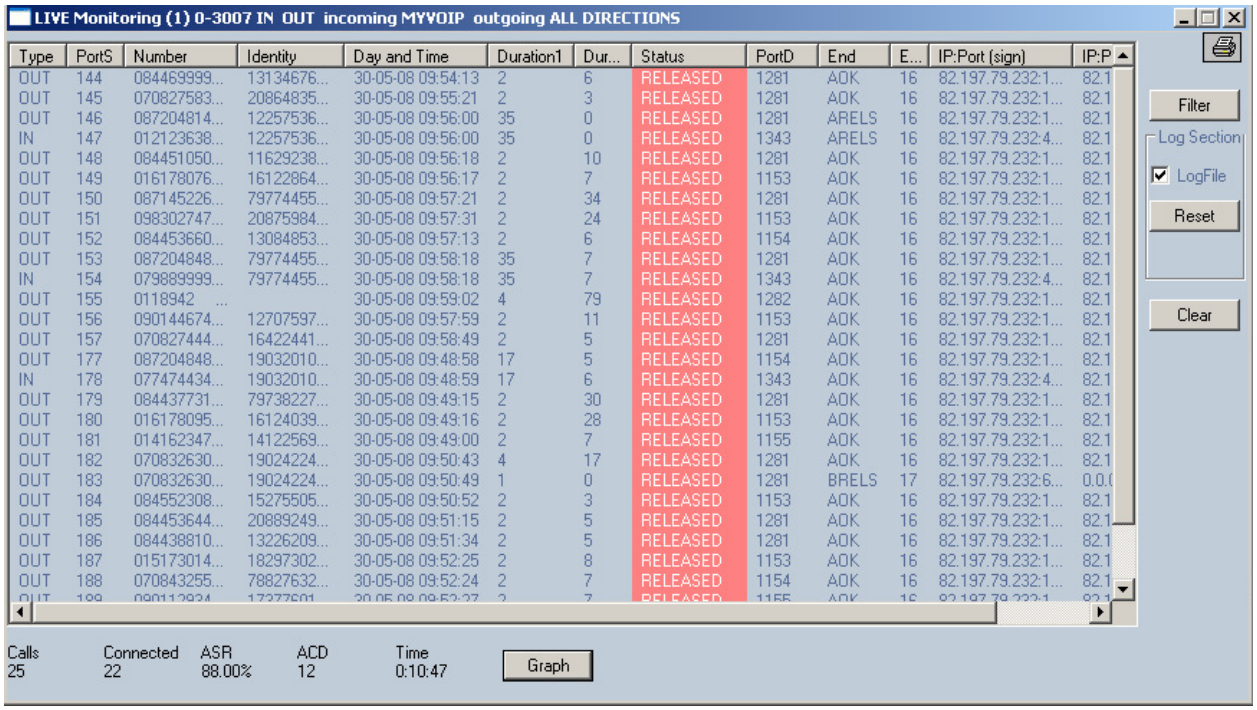
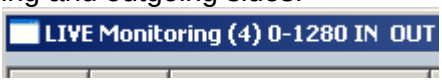


Figure 30 – Live Monitoring (n) window

The range is defined from 0 to 3007, for all directions on incoming and outgoing sides.
You have the possibility to open up to 4 (four) separate windows of that type.



For VOXI equipments a single Live Monitor window is enough, but the OAM program has provisions for extended capacity boxes, where several users are connected at the same time.

Notice that the tooltip displayed for the live monitoring icon is “Calls monitoring – new window”.
When all the four windows are opened the icon tooltip will show “Calls monitoring – no window available”.

Through those four windows you have the possibility to filter calls from different directions on both incoming or outgoing ways.

The digit “n” shown between the round brackets (and) represents the index of the window - it can be a value from one to four. In this example you see the window (2):

In the following you will see the window (1):

The monitoring of calls can be stopped at any time by pressing one more time the same icon. When the calls monitoring is started, the following window will be displayed when first information about a call in progress is received:

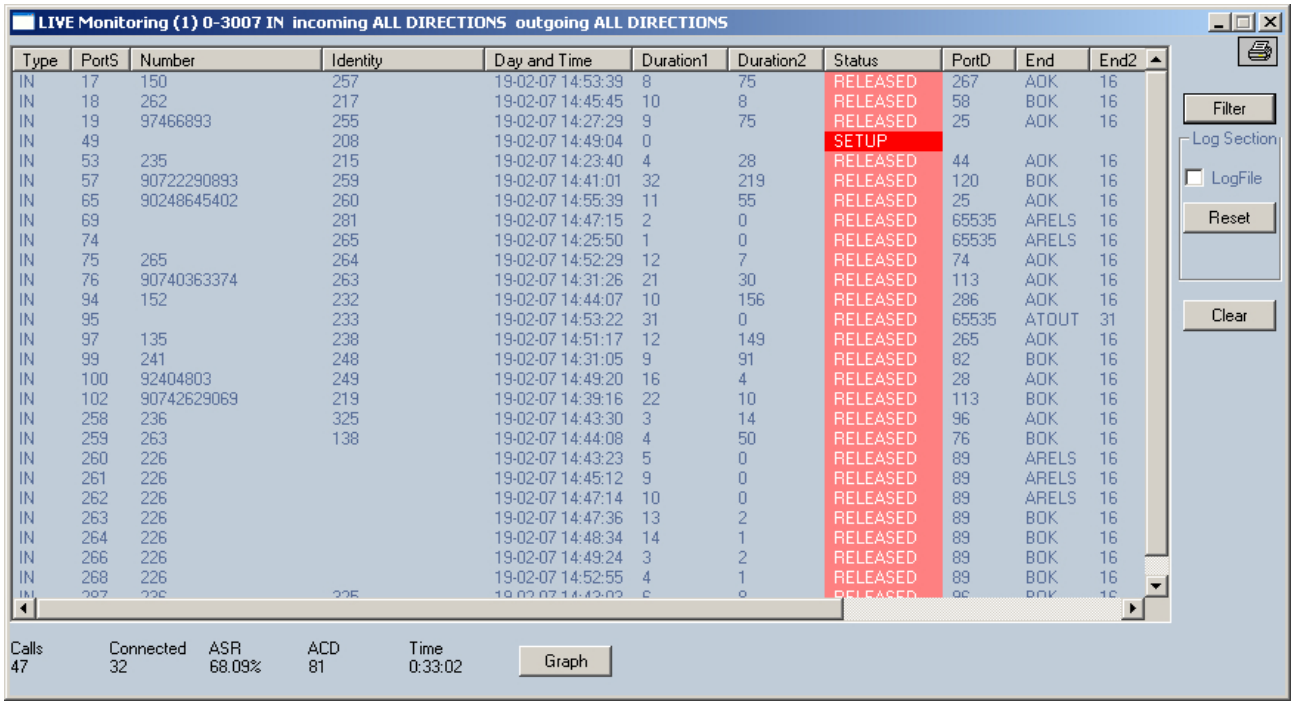


Figure 31 – Live Monitoring

In the „Live Monitoring" window each call is displayed, containing the parameters described below:

- **Type** – call type. It can be an input or an output call;
- **PortS** – source port number;
- **Number** – the phone number of the destination (the dialed number);
- **Identity** – the caller identity (the subscriber phone number);

Type	PortS	Number	Identity
OUT	144	0844699...	13134676...
OUT	177	0872048...	19032010...
IN	178	0774744...	19032010...
OUT	179	0844377...	79738227...
OUT	180	0161780...	16124039...

- **Day and Time** – the call date and time initiation in the format dd-mm-yy hh-mm-ss;
- **Duration 1** – the selection time (in seconds);,
- **Duration 2** – the call duration (in seconds);

Duration 2 is the selection time, that is the time elapsed between the moment when the call enters the VOXI gateway and the called party answer (in seconds);

	Duration1	Dur...
2	35	0
0	2	8
4	2	17
7	2	10
4	2	6
8	2	5

- **Status** – shows the state of the calls which can be: "SETUP", "PROCEED", "ALERTING", "CONNECTED" or "RELEASED"; each state has an associated color.

0	RELEASED	1153
	PROCEED	1281
	SETUP	

- **PortD** – the destination port;

it can take a value of "65535" until the incoming call is routed through an destination port;

Status	PortD
RELEASED	1281
RELEASED	1281

or

Status	PortD
RELEASED	65535
ALERTING	65535
ALERTING	65535

- **End** – mode of call finalization, computed by the equipment.

It indicates the mode of ending the call.
Possible values for this field are: AOK, BOK, ARELS, BRELS, AINEX, BINEX, ACONG, BCONG, ASERR, BSERR, ANERR, BNERR, ANANS, BNANS, ABUSY, BBUSY, ATOUT and BTOUT:

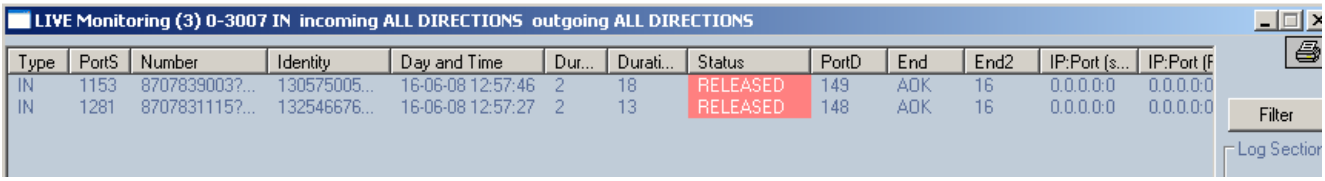
PortD	End	E...
1154	AOK	16
1343	AOK	16
1281	AOK	16
1153	AOK	16
1155	AOK	16
1281	AOK	16
1281	BRELS	17

  - The first character indicates who has released the call: A = caller party or B = called party;
  - The following characters are keywords detailing how the call was ended: OK - ANSWER (response in the destination part), RELS - RELEASE (release in other situation then ring-back tone or busy), INEX - (non-existent from equipment point of view - no defined route), CONG (congestion from equipment point of view - no available resources), SERR (Signaling error), NERR (Network error), NANS - NO ANSWER (release on ring-back tone), BUSY (release on busy situation) and TOUT - TIMEOUT (timer expiration).
- **End2** – the mode of call finalization as sent by the remote party, that is mainly ISDN calls - possible values are ' 31' (normal call) for non ISDN calls or values from the ISDN standard for release code.
- "IP:Port (sign)" – the IP signaling port – filled in case of incoming voip calls and viewing IN records or in case of outgoing calls and viewing OUT records (IN, OUT settings are performed through the "Filer" option).

Status	Por...	End	E...	IP:Port (sign)	IP:Port (RT
RELEASED	1281	AOK	16	82.197.79.232:17...	82.197.79.:
RELEASED	1282	AOK	16	82.197.79.232:17...	82.197.79.:
RELEASED	1153	AOK	16	82.197.79.232:17...	82.197.79.:
RELEASED	1154	AOK	16	82.197.79.232:17...	82.197.79.:
RELEASED	1281	AOK	16	82.197.79.232:17...	82.197.79.:

- "IP:Port (RTP)" – the IP signaling RTP– filled in case of incoming voip calls and viewing IN records or in case of outgoing calls and viewing OUT records (IN, OUT settings are performed through the "Filer" option).

Note1: The fields "Duration 2", "End" and "End2" are completed only when the status field is "RELEASED".



Note2: Initially the LIVE MONITORING window displays the IN records (the incoming side of each call), not the output side of the call. This is performed in order to avoid having the same call into two records. Normally, an incoming call should have, in case of routing success, another correspondent line in which the source port is the destination port from the first record.

Filters

In the “Live Monitoring” window press the „Filter” button located on the right part of the window in order to filter the calls from the TOPEX VoiBridge equipment.

On the screen will pop up the window „Live Monitoring Filter” which contains the parameters:

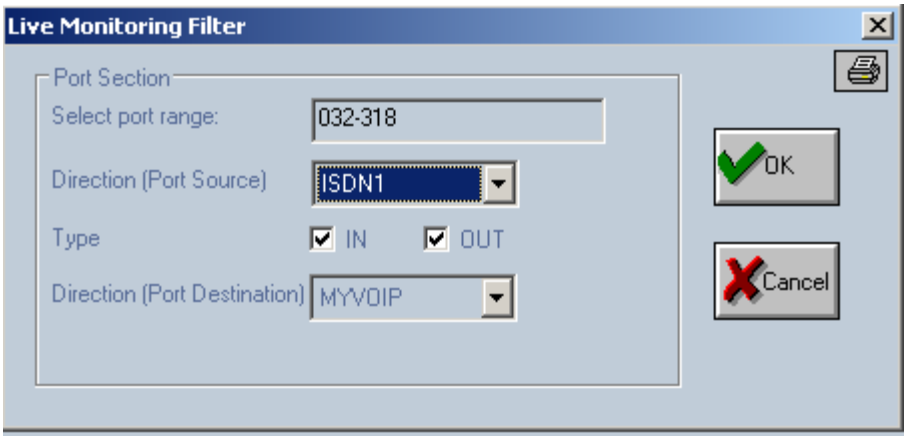
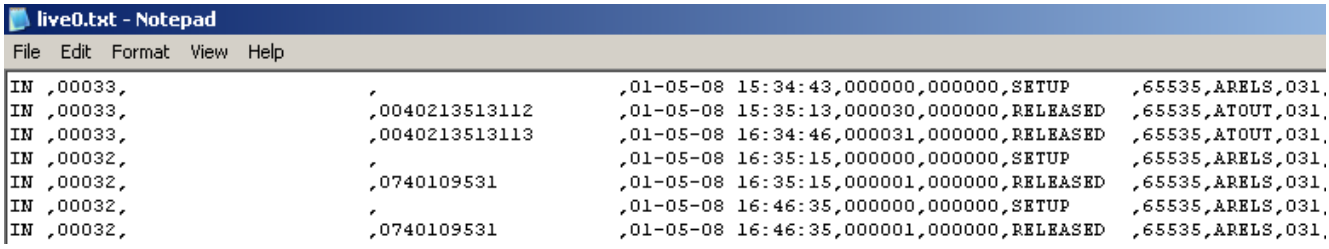


Figure 32 – Live Monitoring Filter

- **Select Port Range** – you can select a range of ports for which to display the "Live Monitoring". The calls monitoring will be displayed for the ports in range from "xxxx" to "yyyy". Those two values must be separated by the minus character "-".
- **Direction (Port Source)** – the input direction name;
- **Type** – You can select the option "IN" or "OUT". Those two options are about the type of the call (from the point of view of the TOPEX gateway): incoming or outgoing calls. The default option is “IN”, it is automatically selected at startup of live monitoring.
- **Direction (Port Destination)** – output direction name. Press the „OK” button to save the settings made in this window.

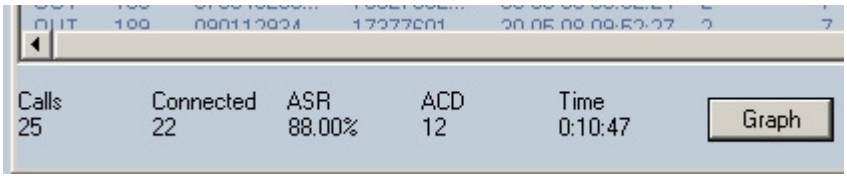
Note: You may also save all online messages regarding the progress of the calls into a log file. The name of this log file is always "liven.txt" (where ‘x’ is 1,2,3 or 4 depending of the “LIVE Monitoring” window rank).

The content of the file can be reset with the option "Reset". It can be viewed with any text editor:



Statistics

At the bottom of the „**Live monitoring**" window is displayed a calls statistic which displays the parameters:



- **Calls** – the number of calls from the gateway;
- **Connected** – the number of calls which are connected to the destination;
- **ASR – ASR (Answer Seizure Ratio)** statistics for calls;
- **ACD – ACD (Average Call Duration)** statistics for calls;
- **Time** – the total time of the calls achieved through the TOPEX Equipment.

To the right of these parameters is located the „Graph" button which determines the screen appearance of the „Statistic on last 60 minutes" window which contains two diagrams with the ASR and ACD parameters.

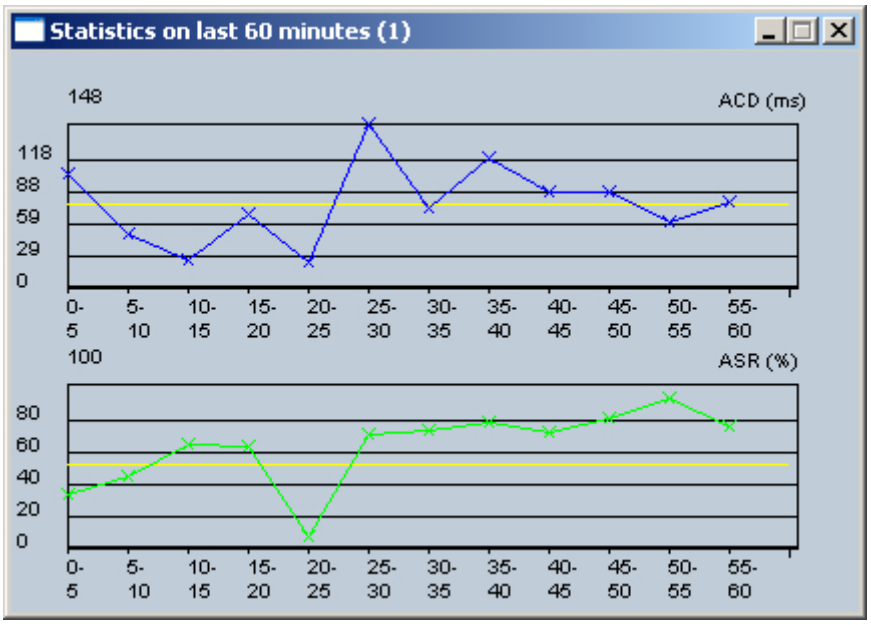


Figure 33 – Live Mon Statistics

In the upper part of the previous picture you can see a statistic on the ACD (Average call duration). The color which is used for ACD is blue, and for the average ACD is yellow. On the bottom half of the last picture there is a statistic on the ASR. The color which is used for ASR is green, and for the average ACD is yellow.

The calculated values for both ASR and ACD are highlighted in blue color. Tooltips with the calculated values are displayed if the user places the mouse over those points.

b) Reporting on directions

When you activate Live Monitoring, , it is also displayed a window „Live Monitoring – Report on directions" which contains the online monitoring report for every direction:

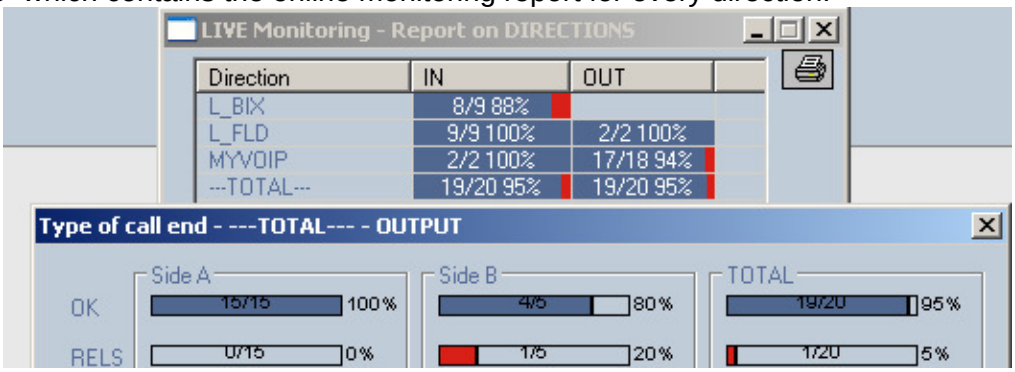
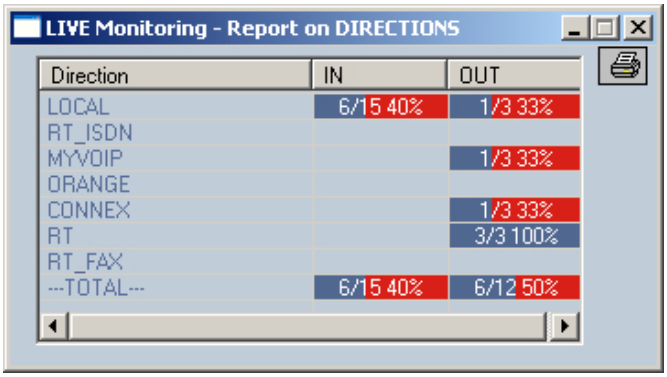


Figure 34 – Report on DIRECTIONS

Here are displayed only the directions for which at least a port is installed on the gateway will be displayed.
There is an indication on each direction about the number of connected calls (displayed with blue color), number of attempts calls (displayed with red color) and online ASR (in percents).



If you click on a column (IN or OUT) for a direction, a new window will be displayed (window named "Type of call end"). See the example below for INPUT calls:

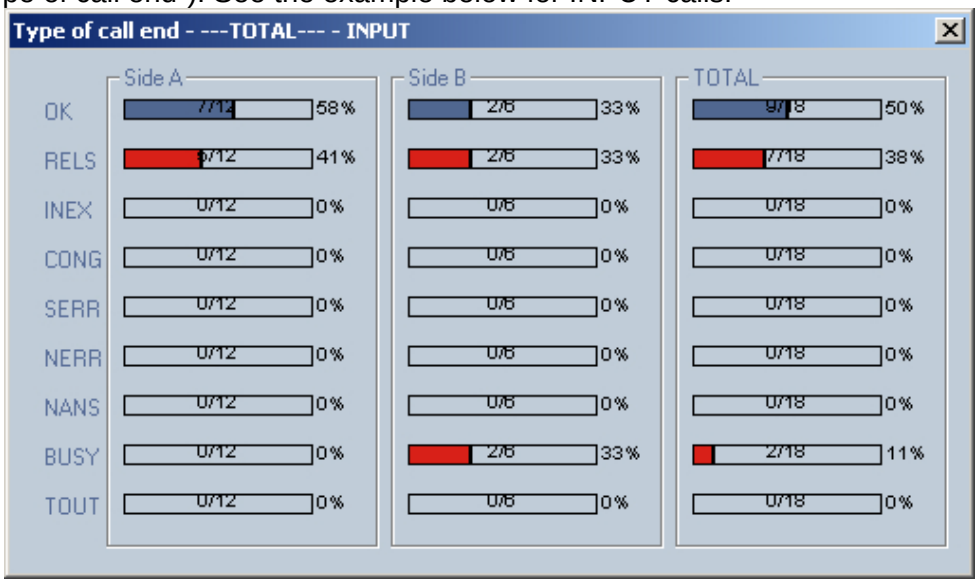


Figure 35 – Type of call ending

In this window the user can see detailed information about the ending of a call attempt. The information is structured in three columns:
- First column "Side A" contains details about all calls released from caller party;
- Second column "Side B" contains details about all calls released from called party
- The third and final column, "Total", contains all of the calls.

For each zone the causes for release of the calls are detailed.

c) **LIVE Monitoring – IP Signaling**
This is a single window that provide access to the VoIP calls which are passing through our Voxi equipment.

This happens when the Topex gateway is performing just the VoIP signaling and when the RTP is made without using the resources of the VoIP channels, or in case of incoming VoIP calls. The first type of calls do not use the VoIP ports of the Voxi box and therefore **cannot** be displayed in one of the normal windows of the LIVE MONITORING function. This is why you need the “IP signaling” window to be able to monitor the calls that are bypassing the Voxi equipment.

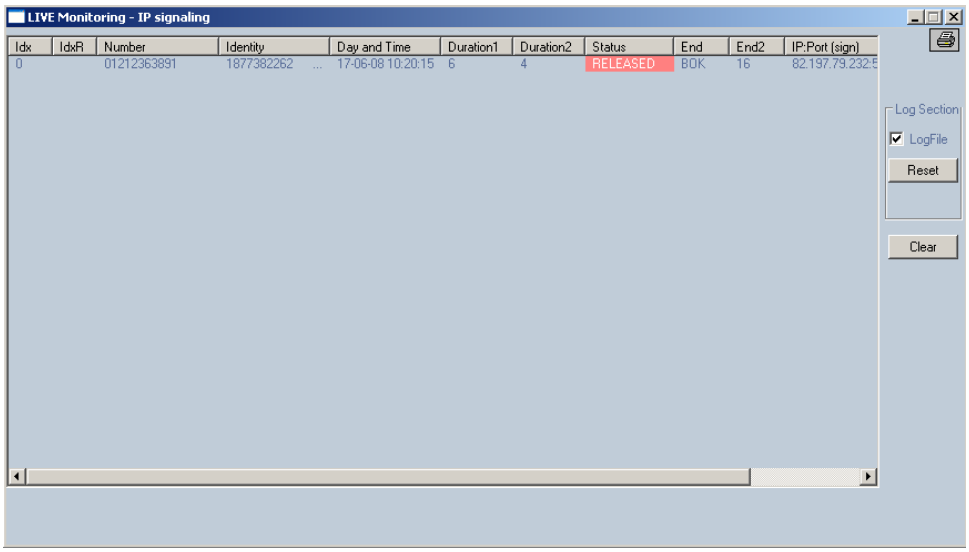


Figure 36 – Live Monitoring IP signaling

These records are just **temporary** records, this is why the image above is almost empty – you can see a singe IP call. For the first type of monitoring windows, the records are preserved after disconnection until a new records is coming on the same input or output port. But in case of IP signaling windows, the records are deleted after 20 seconds.
To close live monitoring windows – you should simply close down all windows of the first type. Then the Live monitoring windows for Directions and IP signaling will be also closed, automatically.

Note1: the fields "Duration2", "End" and "End2" are filled only after the end of the call, when the status field takes the value "RELEASED".
Note2: You can select a range of ports for which to display the "LIVE Monitoring".
The format to be used is "xx-yy". The calls monitoring will be displayed for the ports in range from "xx" to "yy". Those two values must be separated by the minus character "-".
The option "Set" must be used to validate the new port range. The default range which is used is ports from 0 to 319.
Note3: You can select the option "IN" or "OUT". Those two options are about the type of the call (from the point of view of the TOPEX box): incoming or outgoing calls. The default option is “IN”, it is automatically selected at startup of live monitoring.
Note4: You may also save all online messages regarding the progress of the calls into a log file The name of this log file is always "liven.txt". The content of the file can be reset with the option "Reset".

2.4.30  - Configuration list

The icon "  " displays a visualization window with the settings of all ports (rights, number, target, direction, SIM index regarding type of port).

The list is built in physical order of ports starting with 000, and there are shown ALL the ports, even if many are not allocated:

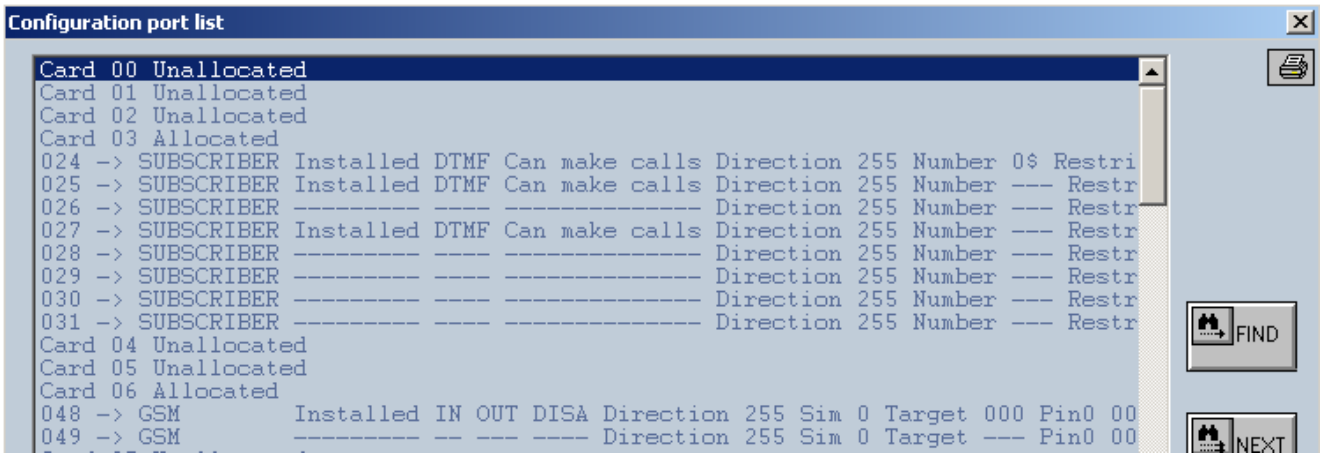


Figure 37 – Configuration port list

In case of a VoiBridge equipment, the first port positions (cards 1-3, ports 000 to 0031) are not used, so they are not shown in the picture below, where the first allocated card is Card 04 with port 032:

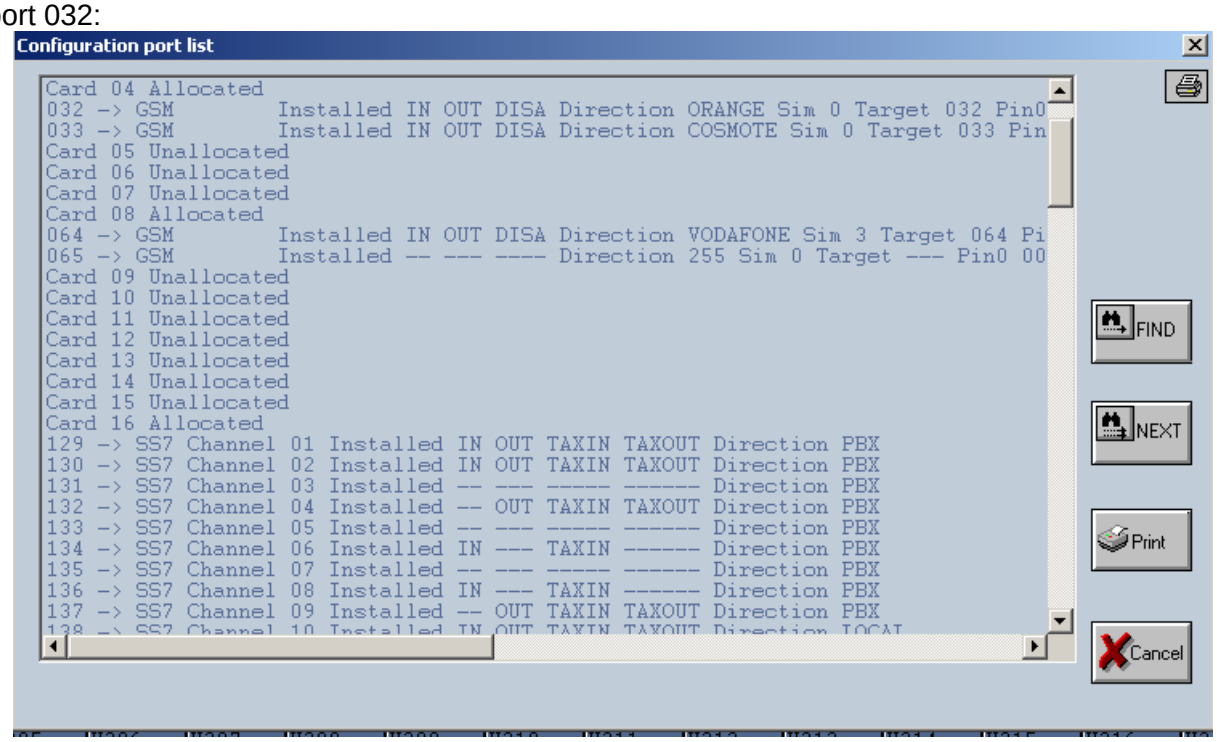


Figure 38 – Ports list for VoiBridge

In the picture above it can be seen the GSM positions for which are given the following settings: 'Installed', 'IN', 'OUT', direction, SIM index, target, Pin1, Pin2, Pin3 and Pin4 values and the SS7 channels with their respective settings.

2.4.31  - Automatic requests

Press the icon "  " for auto-download of alarms and ASR info from all of the TOPEX systems that are connected (using IP communication).

Click on it and the OAM program will automatically launch (at pre-established time intervals) interrogation requests for the installed TOPEX systems. The request may be about alarmed modules and about ASR value (both general and instantaneous). This is very useful when you have several TOPEX equipments, because you no longer need to connect manually to each one to ask it for the alarms and ASR information.

- **Period** – set the time interval (in seconds) for the interrogation of all TOPEX systems that are connected. These values must be greater or at least equal to 60 seconds;
- **ASR** - select a limit for the ASR value: when the general or instantaneous value is under the established level;
- **ALARMS Alert** - pane of the Automatic Requests window allows you to select different kind of alarms. When these alarms occur in the TOPEX system, they will be shown and stored.

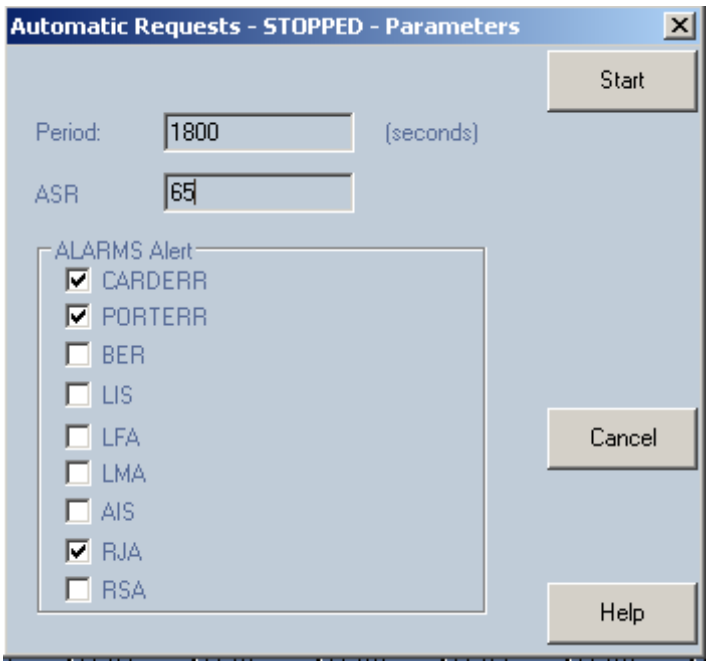
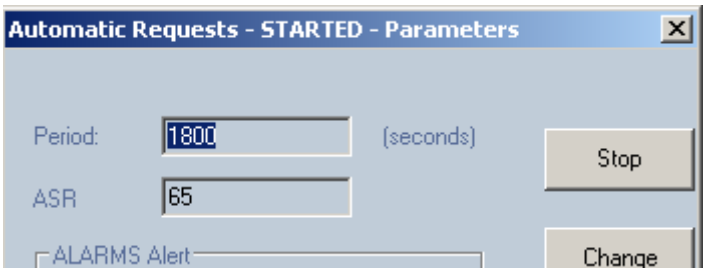


Figure 39 – Automatic Requests

When the interrogation process is not started the text "STOPPED" is shown in the dialog window title. You must select the types of alarms. The start of the automatic interrogation process is allowed by choosing "Start" option. The interrogation period and ASR alert value may be changed at any moment by selecting "Change" value.

When interrogation process is started the text "STARTED" will be shown in the dialog box title. "View" option is for displaying a statistics on last 24 values for instantaneous ASR (for the TOPEX systems that are in automatic interrogation process).



When the specified period expires the request for system interrogation will be launched. The text "AUTO REQUEST IN PROGRESS!!!" will be displayed in the first box of the status bar.

In the next box to right side all messages will enclose the TOPEX system name (connection name).

During the automatic interrogation procedure it is allowed to connect to a system only if the OAM software is in a pause between two interrogations. If it is in the middle of an interrogation, you may still stop the automatic procedure that is in the process (when status bar shows the text "AUTO REQUEST IN PROGRESS!!!") by selecting the "Disconnect" command from any of the systems.

The manufacturer reserves the right to modify the product and manual for the purpose of technical improvement without prior notice. The manufacturer guarantees the good functioning of the product provided that it has been correctly installed and the directives for storage and usage have been respected.

The warranty implies exclusively repairing or replacing the defective unit. The warranty does not include any indirect losses or loss of profit. The manufacturer is not liable for any damage, whether direct, indirect, special, incidental, or consequential, as a result of using the TOPEX VoiBridge equipment.

It is certified hereby that the TOPEX VoiBridge equipment is manufactured in concordance with the legal provisions concerning responsibility towards the quality of delivered products, fulfills the quality parameters specified in its “Users Manual” and is fit for the purpose for which it has been designed.

It also warrants that the equipment will perform substantially in accordance with the accompanying documentation.

Any comments, suggestions and proposals of yours concerning our products are welcome and we are gladly waiting for your feedback.

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