Prodys IP Family
User Manual

Prodys
PROCESAMIENTO DIGITAL Y SISTEMAS, S.L

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INTRODUCTION

Prodys IP Family codec stands for a large range of compatible audio codecs from Prodys. Those codecs allow applications as dual and multi audio codec by supporting MPEG 2/4 AAC LC, MPEG 4 LD & HE and apt-X^{TM} amongst other standard algorithms, as well as bidirectional uncompressed audio transmission over IP.

A Prodys IP Family codec is also a Multi-network audio codec that can be used for audio streaming over a suite of communication networks:

- ISDN 1-BRI (U or S/T interfaces)[1].

- X21 synchronous data transmission links (Digital Leased Line: DLL)[2].

- LAN connector, 10/100Base-T Ethernet interface for audio transmission and control (audio streaming over private IP networks or Internet).

- PSTN connector, to connect to Public Switched Telephone Networks[3].

I.1 The set of Manuals

The User Manual is applicable to most of the common features provided by the Prodys IP Family codec.

For some specific features or restrictions, the user is referred to the proper Hardware and Reference Manual applicable to the codec in use. Installation

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[1] For some Prodys IP Family Codecs ISDN network capability is an option (Terminal Adapter module). Please refer to the proper Hardware and Reference Manual for this point.

[2] For some Prodys IP Family Codecs X21 network capability is an option (Terminal Adapter module). Please refer to the proper Hardware and Reference Manual for this point

[3] Only available on NomadaIP.

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requirements, physical and electrical parameters are also included in this document.

Hardware and Reference Manuals are available for:
- ProntoNet
- ProntoNet LC
- ProntoNet IP Decoder
- Nereus
- Nereus One
- PortaNet
- Nomada IP

If several Prodys IP Family codecs are managed by means of the ProdysControl please refer to the ProdysControl Manual too.

Prodys IP codecs provide the user with a control protocol which allows the user to develop customized management software. The control interface for this protocol is either the RS232 serial port or the Ethernet port. For detailed description please refer to the Prodys IP Family codec SDK User's Manual.

I.2 Application Notes

For specific subjects, Applications Notes and release update (What is new, Changelog…) the user is kindly referred to check our download area at www.prodys.net or contact support@prodys.net.

I.3 The User Manual

The information of the User Manual is arranged in the following sections:

- **Chapter I – Introduction**
  This chapter lists the content of most of the documentation available at Prodys.

- **Chapter II – The Web Browser Control**
  Prodys IP Family codec can be controlled from a Web Browser. This chapter describes how to enable the User Interface to the codec.

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- **Chapter III – Quick Start**
  This chapter is a quick guide aimed to let new user arrange his first audio communication.

- **Chapter IV – Configuration**
  This chapter is a comprehensive manual for all configuration parameters.

- **Chapter V – How does the Prodys IP Family codec work?**
  This chapter is a practical guide to help in understanding just how the Prodys IP Family codec works under different configurations. Singular application examples are included.

- **Appendix A – Technical specifications**
  This appendix describes the technical specifications in common for all devices of the Prodys IP Family.

- **Appendix B – Disconnection Codes**
  This appendix describes the meaning of the disconnecting codes showed on the display.

- **Appendix C – IP Ports descriptions**
  Detailed information of the IP ports is given for advanced users or private network managers.

- **Appendix D – Updating the firmware**
  This appendix describes how to update the Prodys IP Family codec firmware.

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WEB BROWSER CONTROL

Prodys IP Family codec can be controlled remotely by using an Internet Explorer web browser. The computer can be locally connected directly via a crossover CAT-5 cable, sharing the same Ethernet switch with the codec's LAN interface or remotely via the Internet.

To access the Prodys IP Family codec from the Internet Explorer, enter the IP address of the unit in the address bar. Keep in mind that the Prodys IP Family codec factory IP address is 192.168.100.100 and it could be necessary to modify the network configuration of the computer on which the web browser is running the first time.[4]

Installation Requirements

1.- Pentium 166 or higher.

2.- 64MB RAM minimum.

3.- Operating Systems:

Microsoft Windows XP, Microsoft Windows 2000,

Microsoft   Windows NT 4.0 Service Pack 6 or higher,

Microsoft Windows Millennium Edition (ME), Microsoft Windows 98.

Microsoft Windows Vista.

4.- Microsoft Internet Explorer 5.0 or higher.

The screen resolution must be 1024x768 minimum.

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[4] Some Prodys IP Family codecs are provided with a front panel display and keyboard. For these devices it is possible to set directly the proper IP configuration before gaining access by means of the web browser. Please refer to the "HW and Reference Manual" of your codec for details.

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> **The first time that the computer accesses the Prodys IP Family codec it is necessary to install the software. The computer will show the following window:**
>
>
>
> **The unit is supplied with the following IP address: 192.168.100.100**
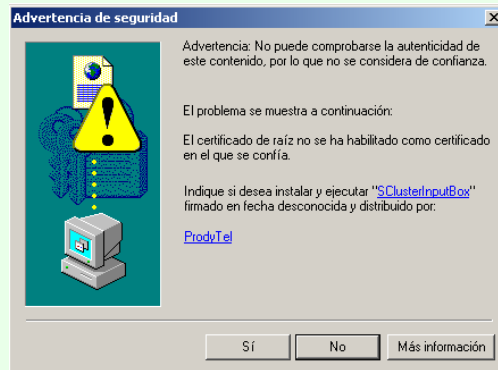
> Please refer to the Hardware and Reference Manual if the IP address is unknown.
> A procedure to recover the factory default configuration is described.

The first time the user accesses the Prodys IP Family codec web page, an OCX file has to be downloaded and installed on the computer. *Microsoft Internet Explorer can be configured to block OCX objects installation and/or execution.* So, depending on the configuration of the web browser, the following message can appear when first accessing the Prodys IP Family codec web page:

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Go to Internet Options in IExplorer, click on 'Security' tab, and set 'prompt' when downloading ActiveX signed and unsigned controls at Local and Internet zones.

Alternatively a manual OCX installation procedure is described at
http://prodys.net/downloads/OCXManualInstall.zip

This procedure is applicable if the **Internet Options for IExplorer** can not be modified by the user.
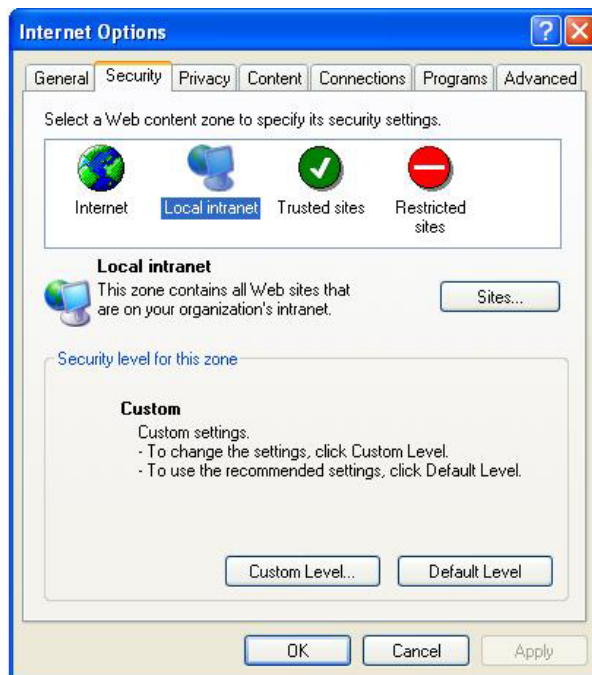
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Windows Vista: Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, *please disable UAC (User Access Control) on Windows Vista.* Once the OCX file has been installed in the computer, UAC can be enabled again.

Each firmware version might have a different OCX file, so the new OCX should be installed as it is done for the first access to the web page of the unit. If the unit was upgraded and, depending on the 'cache' configuration of the Internet explorer, there might be problems when accessing the web page, given that the old web page might be offered by the browser instead of the real one, which should be installed to replace the old one. In this case, a message indicating 'Incorrect Versions' will appear as soon as the user click on 'Control' or 'Monitor' on the Prodys login page. Click on F5 to skip the cache entries, and access to the 'real' web page. Even after pressing F5 and, depending on the IExplrorer configuration and/or version, this situation might continue. *In that case, go to Internet Options in IExplorer, click on 'General' tab, and delete temporary files.*

To access the Prodys IP Family codec from the Internet Explorer enter the IP address of the unit in the address bar as shown here:

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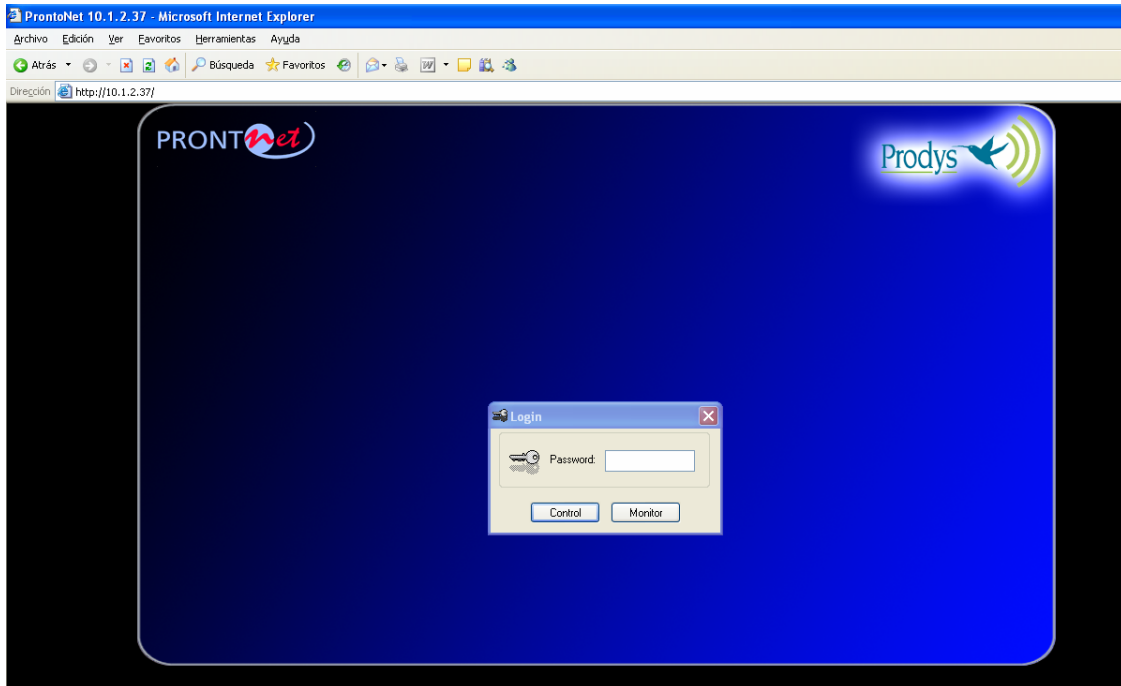
User can choose whether to gain the Web Browser access to the codec with monitor or control privilege. Bear in mind that only one web browser at a time can control the unit. However, it is possible to monitor the unit from several web browsers simultaneously.

If a unit is already being controlled by a web page and we try to get the control from another web browser, a message will appear. This message will indicate that the unit is already being controlled from another PC and the IP address of this computer.
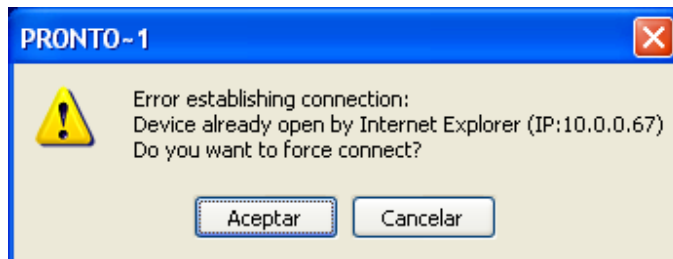
It is possible to get the control by pressing OK. Then, the connection of the old owner will be closed and the unit will be blocked for new controller.

This system is fully integrated with the ProdysControl. In that way, the control from/to ProdysControl can be revoked. It is possible to have a unit being controlled by ProdysControl and at the same time, web browsers monitoring the same unit, etc…

When entering the web page, the user can set the time-out period for the connection between the PC running the web browser and the unit. By default, this period is set to 5 seconds. To modify this parameter, click on the advance features button of the login window,

then define the new value from the following window:

The user should take into account that the longer the time-out, the less likely it is that the connection between the PC and the unit will be lost.

The user control Timeout definition does not affect the audio streaming protocol Timeout.

Base Port: When Prodys IP Family codec ports have been modified since the factory default configuration, it will be necessary to indicate the new port at the Prodys IP Family codec web browser address line. An example is given at the end of the green remarked box.

MORE ABOUT THE PRODYS IP FAMILY CODEC PORTS

Changing Prodys IP Family codec Ports: The Prodys IP Family codec configuration menu allows the user to configure which ports the unit will use for its TCP/UDP/IP communications from the web page.
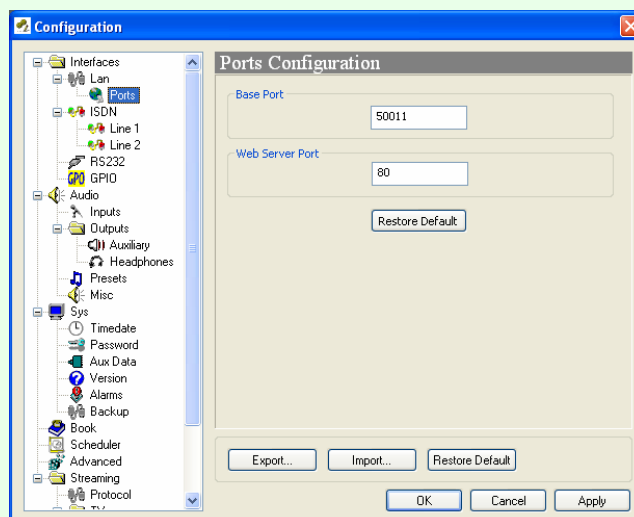
There are two different groups:

- **Web Server Port**: By default, it is TCP port 80. This is the internal web server port.

- **Base Port**: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit for management and for the audio communications over IP with Prodys V1 and V2 protocols (Prodys V3 & V4 use different ports other than these ones). From this base port on, up to 32 ports should opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041 (plus 51000 and 51001), both for UDP and TCP. Therefore any modification of the default values should be taken into account (e.g. opening /forwarding in the corresponding router/firewall if required).

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IMPORTANT

The following should be taken into consideration when changing the base port:

- To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:

http://<IP>:<Port> Example: 192.168.0.10:8080

Once the page is entered correctly, the web browser will display the "Home Page":

The Prodys IP Family codec Web page is arranged in three main areas:

- General Configuration area. ❶
- Control area. ❷
- Monitor area. ❸
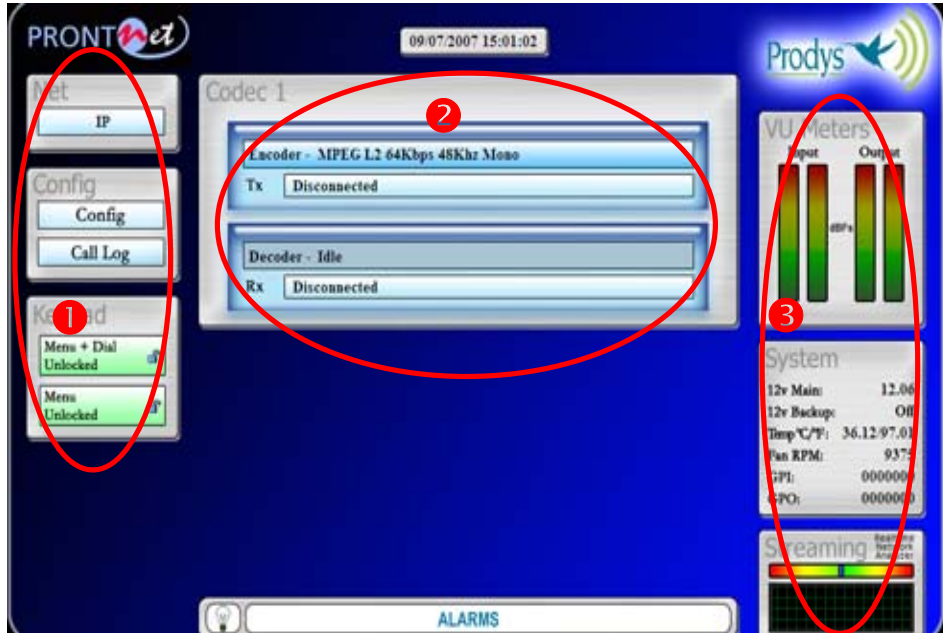
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By clicking on any button on the menu bar or any highlighted zone, individual configuration pages are displayed.

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Chapter III

QUICK START

This chapter guides new user thru the basic steps to start an audio connection over IP between two Prodys IP Family codecs.

Furthermore this example is also a good starting point for users focused on audio connections over ISDN or X21. (Please refer to section V.7 and V.8 for a detailed guide for those networks).

Calling to 192.168.100.102

Prodys IP codec
192.168.100.100

Audio streaming

Prodys IP codec
192.168.100.102
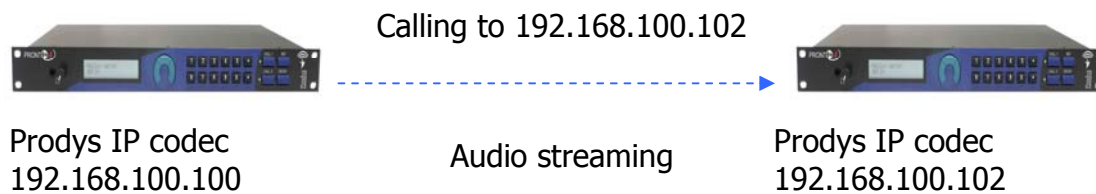
Given is:

- Both codecs are connected on the same Local Area Network.

- The user gains control access to the codec as detailed in chapter II.

- The codecs remains with the factory default settings (but the IP address). To recover the factory default configuration, please refer to heading of the configuration chapter (IV).

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III.1 Selection of the network interface and protocol

The user has to select the IP Network Interface and match the IP protocol on both codecs. For this example we propose Proprietary Protocol V3.[5]

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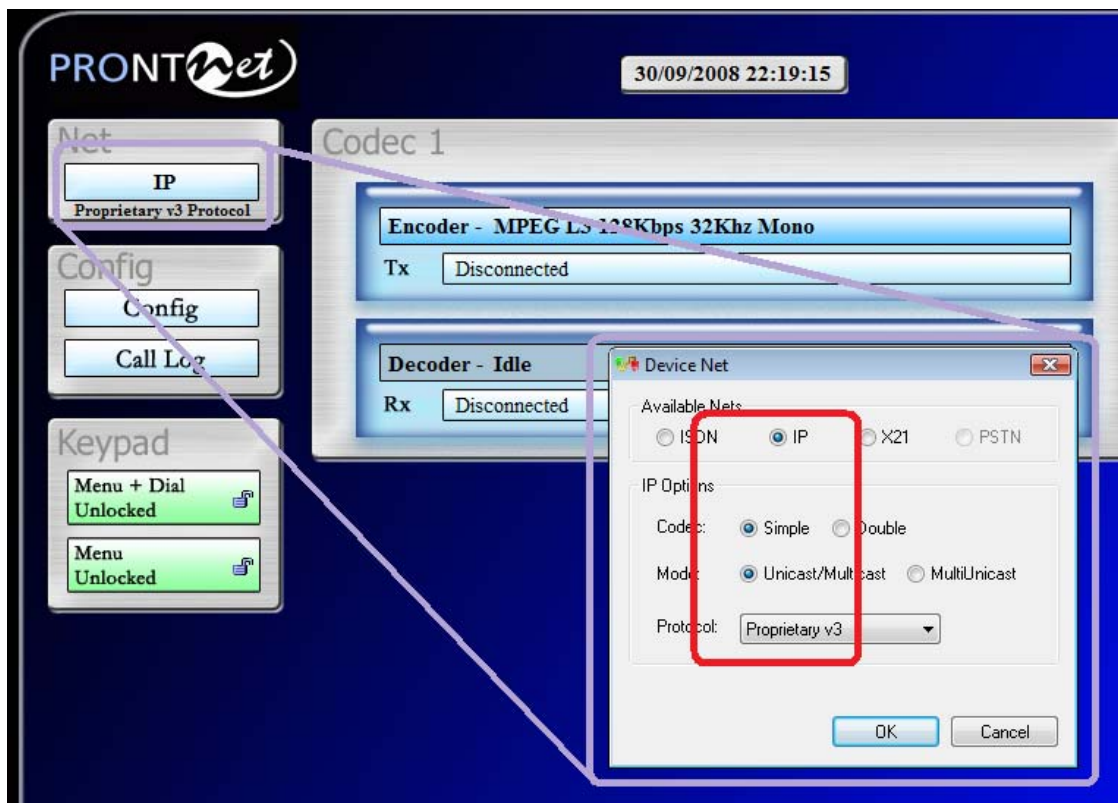
[5] Further details on section V.1

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III.2 Selection of the network profile

The user might select the network profile of his choice with just a handy shortcut.
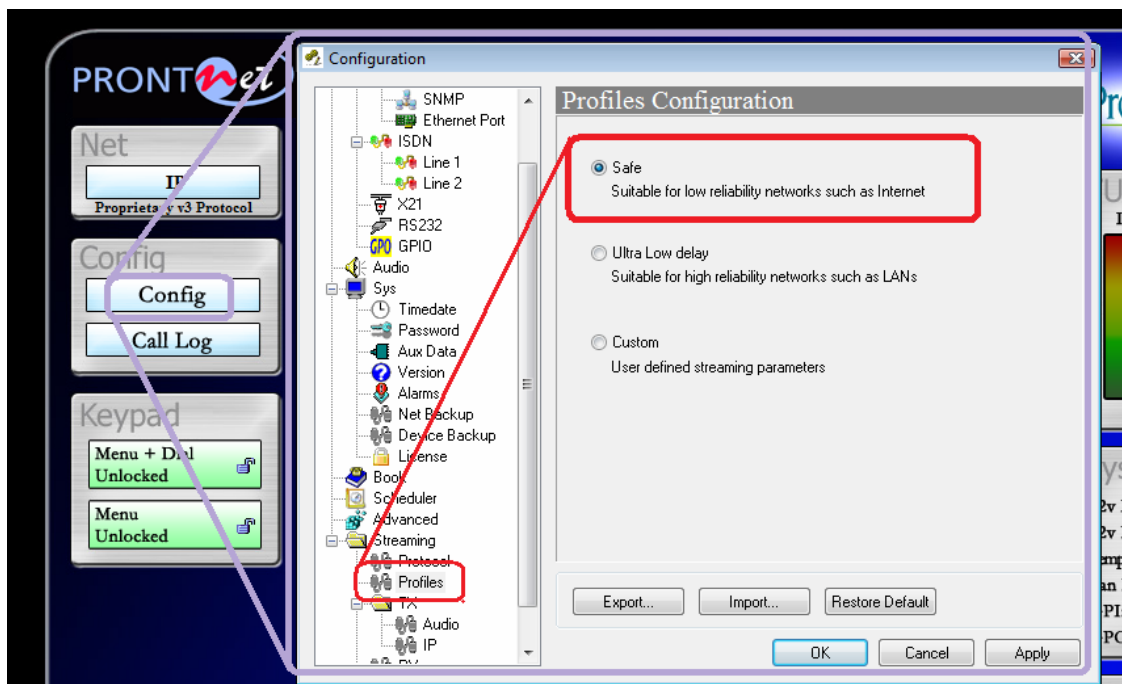
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III.3 Selection of the Encoder algorithm

The user is required to select the Encoder compression algorithm of his fond for both codecs in place.
It is remarked that the user is not required to define the Decoder compression algorithm, since Prodys decoding engine is fully automated.[6]

---
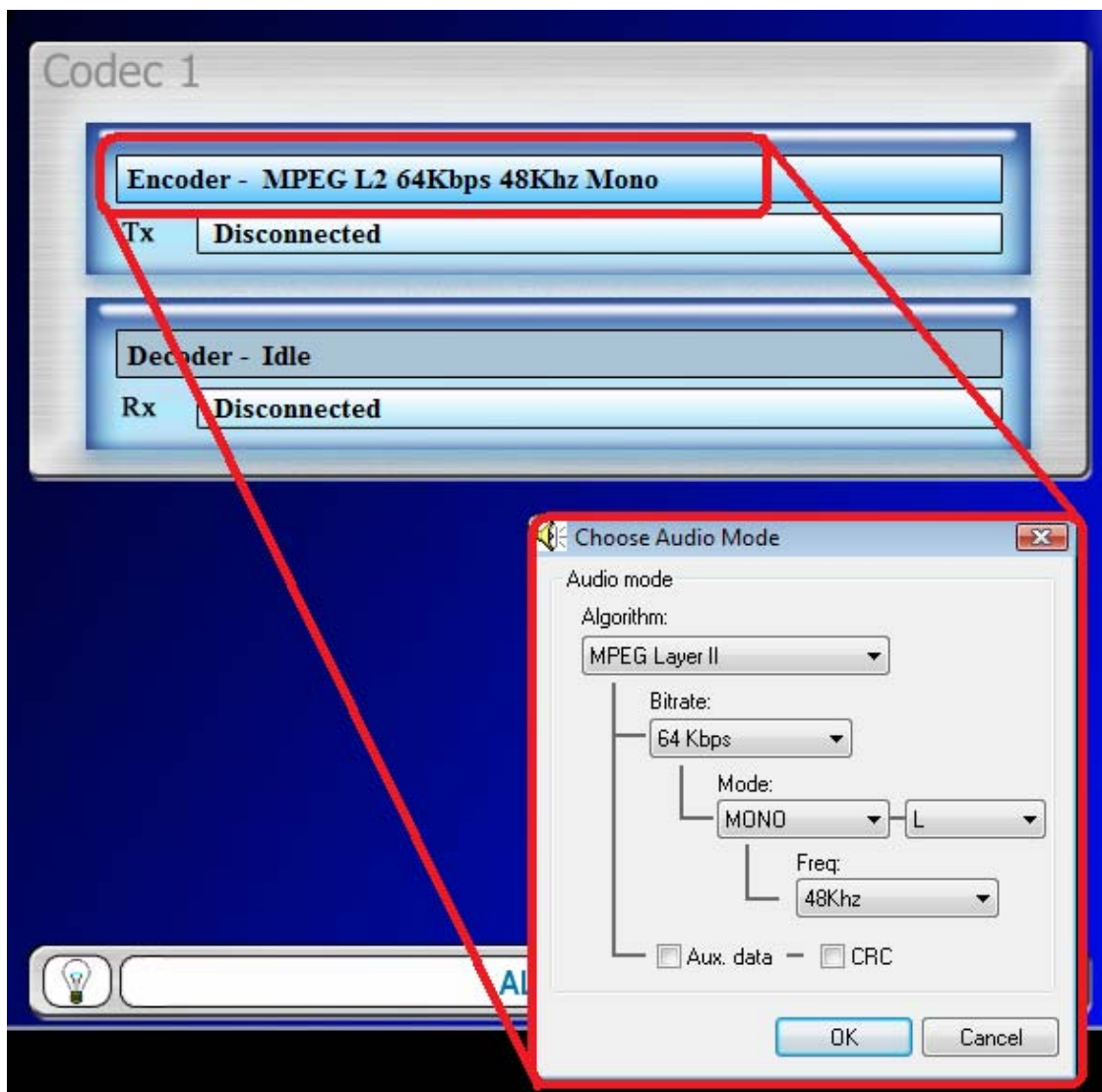
[6] Further details on section V.5

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III.4 Dialling the destination codec

The last step is to dial the destination codec according to the figure. Parameters to match are the IP Address of the remote codec and the codec number[7] (by default codec 1).
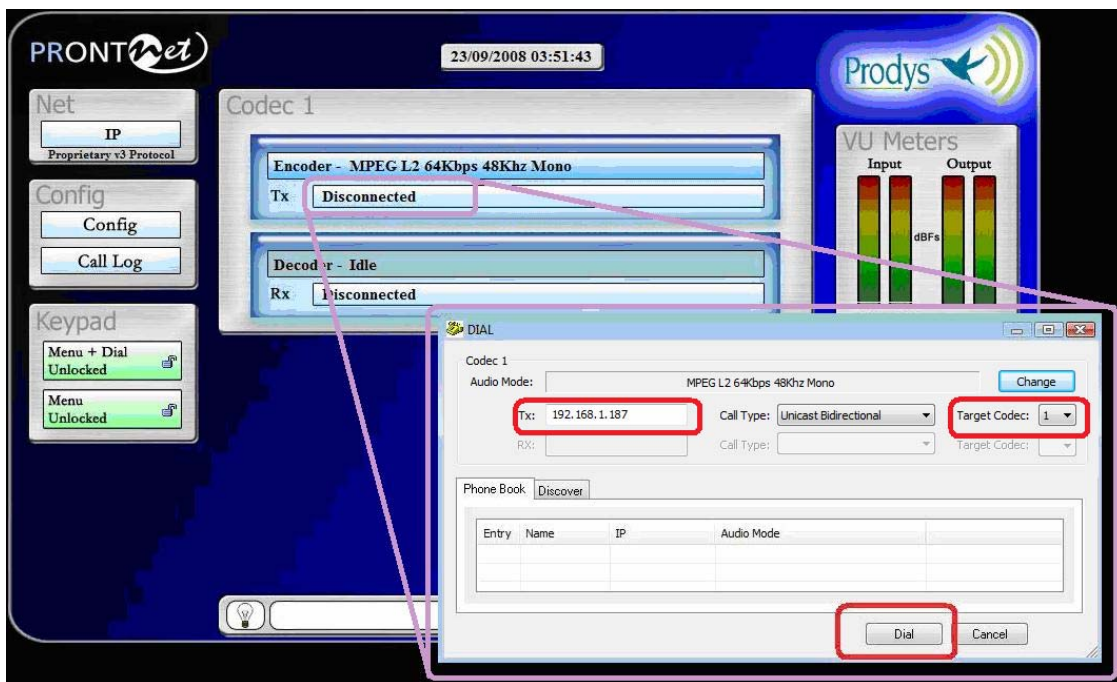
Immediately the audio streaming over IP takes place between both codecs. The user gets feedback of the dialling success with a green connection indication.

---
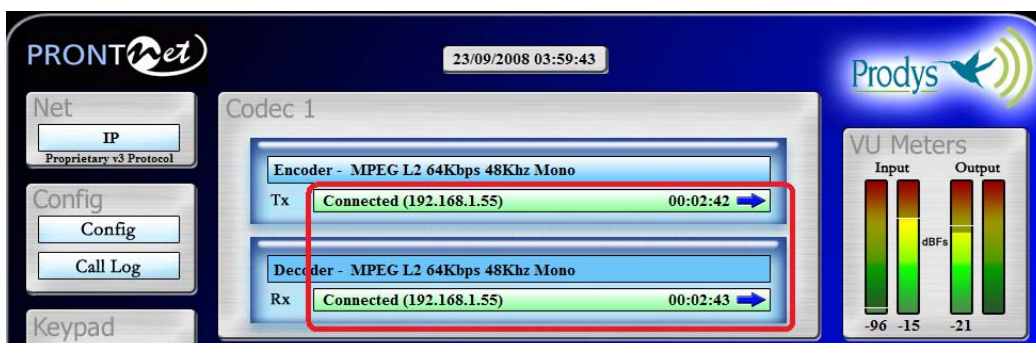
[7] Please refer to section V.4 for details on Target Codec.

III.5 Stopping a communication

The user might stop an audio stream in progress by double clicking the upper green connection indication at any time.

III.6 Encoder and Decoder status indication

Upon the selected network interface and protocol, there are several status indications available for the encoder and decoder part of the codec:
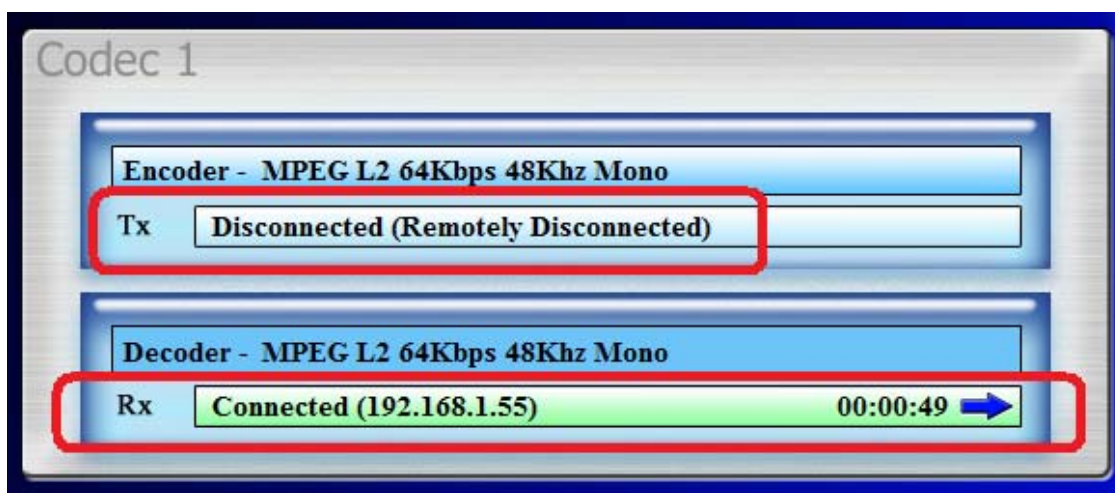
- **<No physical Line>:** The communication line is not physically detected. The user is kindly asked to check the network interface cable.

- **<Connected>:** The line is connected. An arrow will indicate if it is an incoming or outgoing call.
  The duration of the call also is indicated.

- **<Disconnected>:** The line is detected physically but no audio streaming is in progress. Beside the text appear the disconnection codes.

- **<Disconnected (Connection Rejected)>:** The far-end codec is addressed, but this codec can not take the call because the codec is busy or the Target codec is not available[8].

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[8] Please refer to section V.4 for details on Target Codec.

- **<Calling>:** A dialling is in progress or the codec is waiting for the far-end to complete the invitation (eg. manually answer at the far-end)

- **<Ring>:** Receiving an in-dialling call on the line.

III.7 Decoder Status

Further information is applicable only to the Decoder status:

- **<Searching>:** The decoder is not synchronized yet with the incoming audio stream at the receiver. The decoder engine is searching to get in sync with the proper compression algorithm.

- **<Framed>:** The decoder is synchronized.

- **<Stopped>:** Some restriction applies for the Decoder and therefore signal will not be available at the audio outputs. (eg. the codec has not got a valid Prodys' license for the incoming compression algorithm, or the Decoder and Encoder compression algorithms are incompatible on the same codec[9].)

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[9] The encoder compression selection must match the incoming audio stream for MPEG4 AAC HE and any APT-X compression.

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CONFIGURATION

Clicking on the <Config> button from the top web browser control page brings up the configuration page:
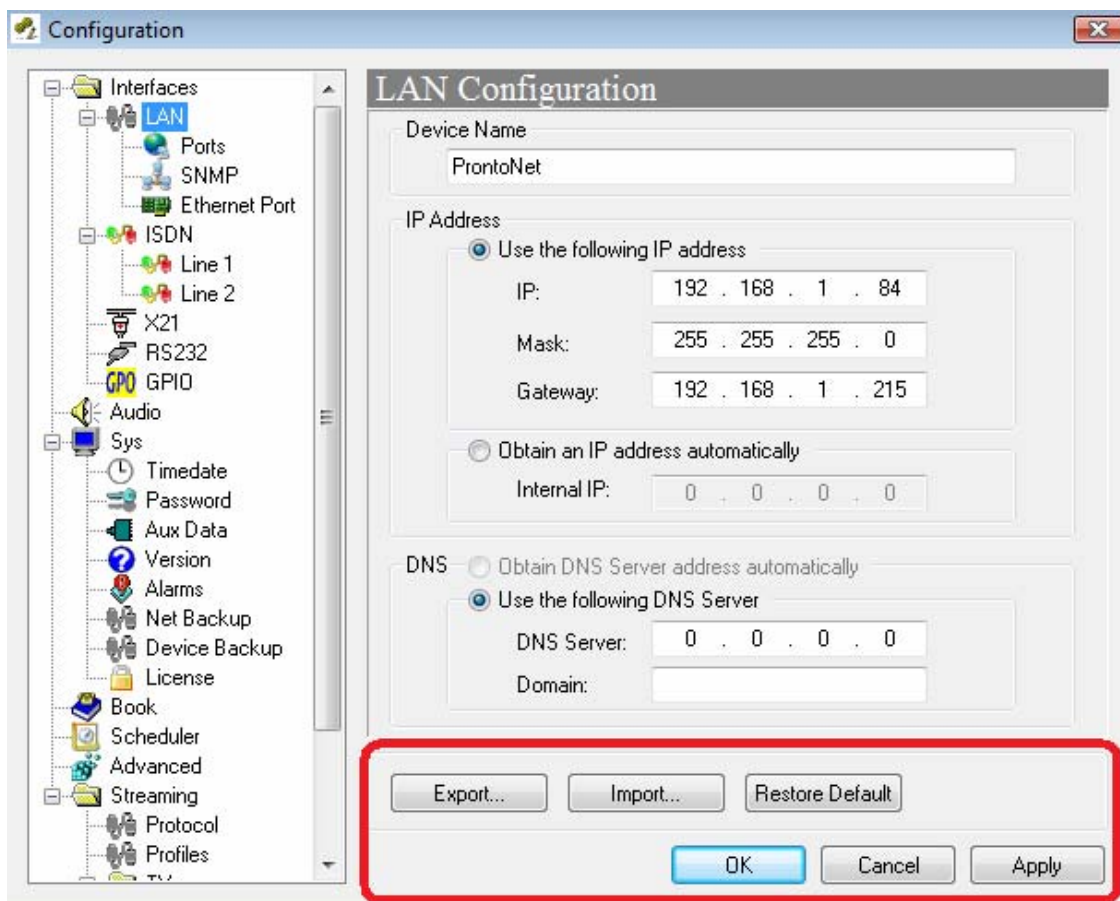
The factory default configuration can be restored by pressing <Restore Default> & <OK>. This option allows the user to restore the factory default configuration.

Please pay attention that even the current IP configuration will be erased after pressing <Restore Default>[10].

If the current IP configuration should kept, the user must re-enter manually this information before pressing <OK> or <Apply>

The Factory Default IP configuration sets to any DHCP values if this service is available on the local area network. In other case the default address 192.168.100.100 /24 is provided.

To apply the changes, the user can press either the <OK> button or the <Apply> button. By pressing the Apply button the changes will be applied but the configuration window will remain.

<Export>: This option copies the complete codec configuration to the computer disk. The user is requested to indicate the computer directory.

<Import>: It copies the complete codec configuration from the computer directory. This imported configuration is placed in force as soon the <OK> or <Apply> button are pressed.

IV.1 Interfaces

By clicking on the <Interfaces> menu, the port configuration dialog appears. The ports can be selected on the left side (LAN, ISDN, X21, PSTN, RS232 or GPIO). The right window shows the window dialog to configure the selected port. For example, if we select the LAN interface:

IV.1.1 LAN port

The first definition on this menu is the <Device Name>, which sets the IExplorer tab label with a meaningful indication. This name will be also available on the front panel display, and it will be used to send alarm email alerts when enabled.

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[10] From release 6.0.5 on.

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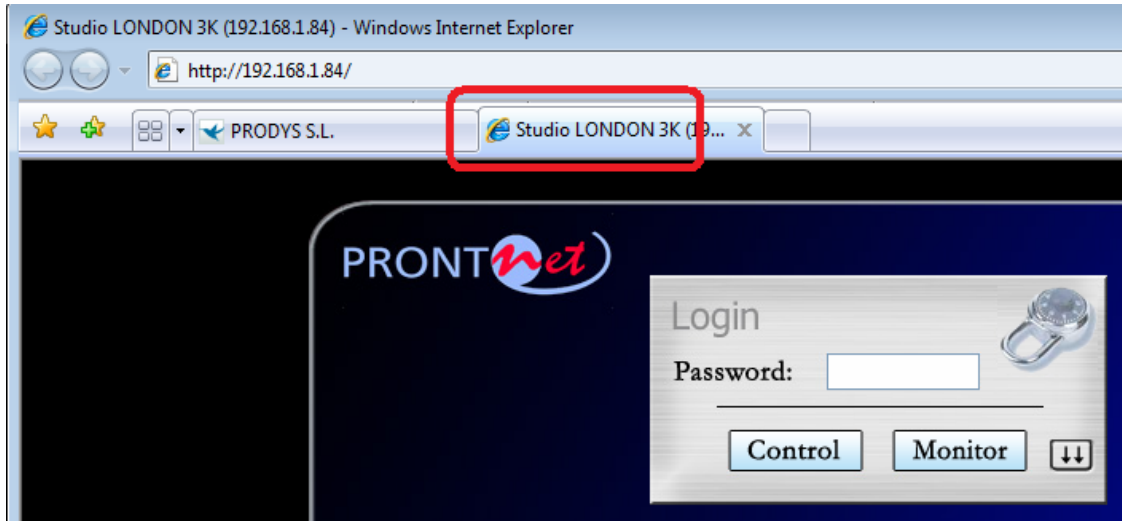
Prodys IP Family codec allows the assigning of the IP parameters either manually or automatically (by DHCP).

Manually → User must enter the following IP parameters manually:
- <IP address>
- <Mask>
- <Gateway IP address>

Automatically → Check "Obtain an IP address automatically" option to get the configuration automatically from a DHCP Server.

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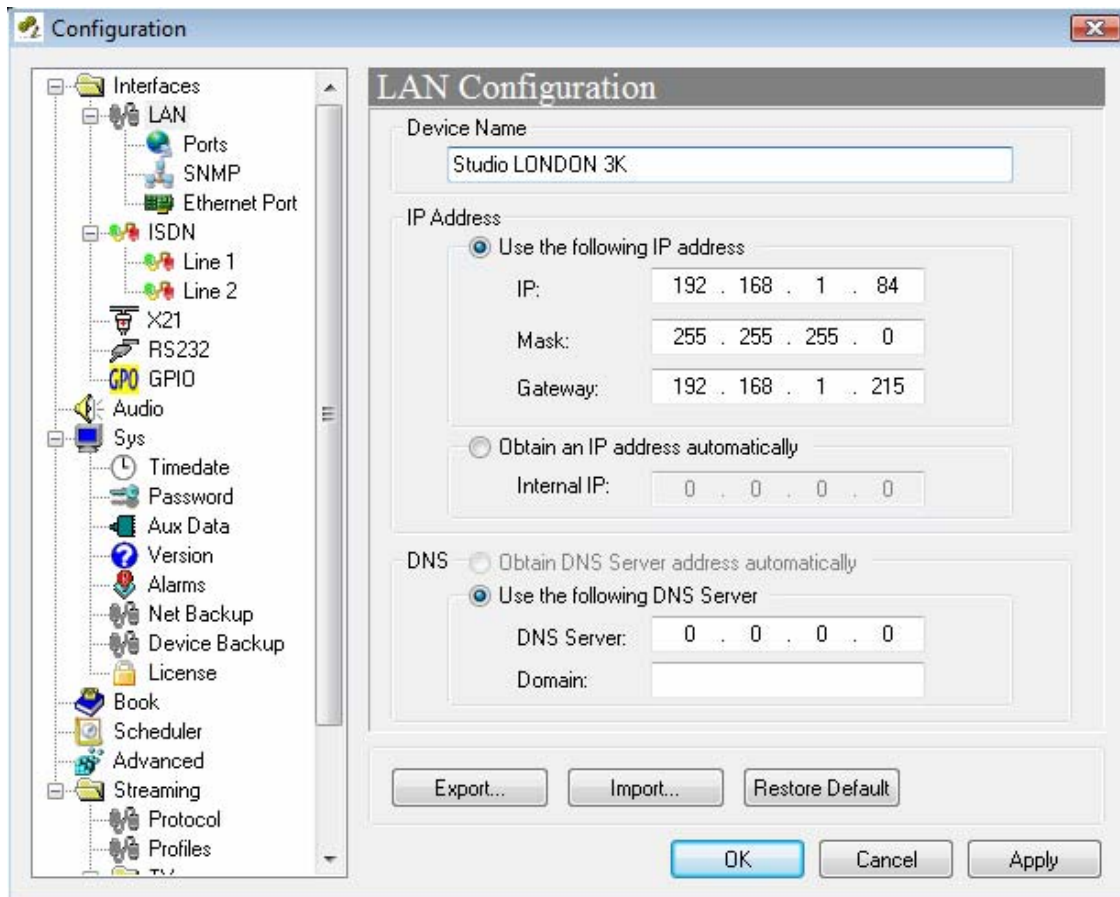
By setting DHCP, the codec will get its IP parameters when starting up. This IP setting might be different from time to time, that is why Prodys IP Family codec supports RIP2 protocol. This protocol allows the user to set an 'internal' IP address, in order that the unit can be identified regardless of the IP settings provided by the DHCP server.

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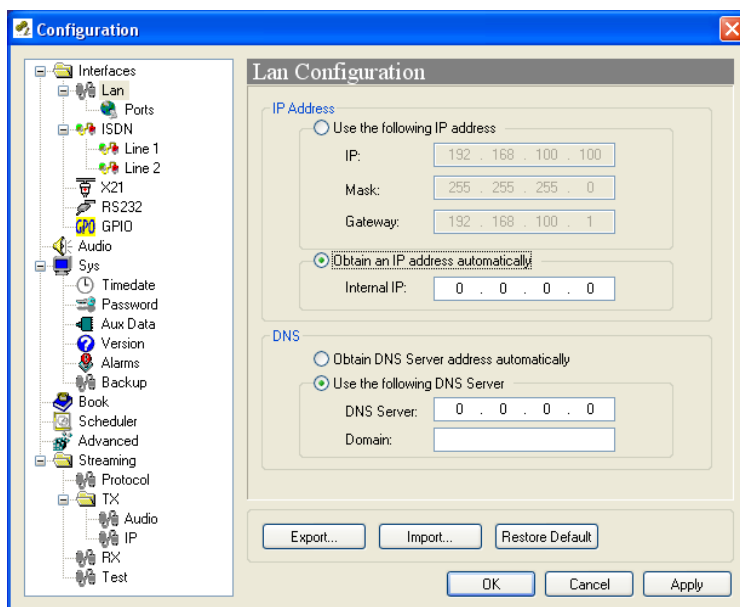
IV.1.1.1. DNS support

By specifying the IP address of a DNS Server, the user can make calls to 'names' instead of IP addresses. This new feature is available in any place where an IP address has to be entered. For example: the dial window, the phone book, SNTP IP server address...This is the DNS configuration windows:
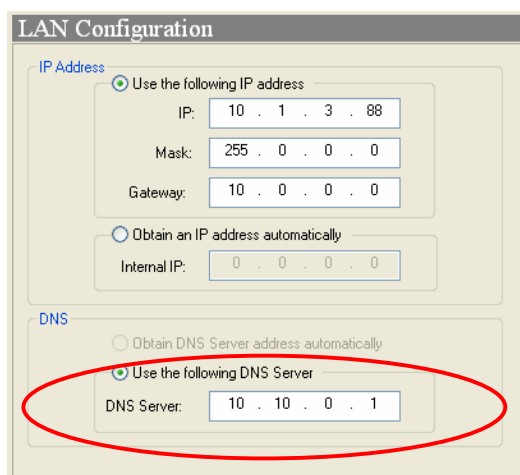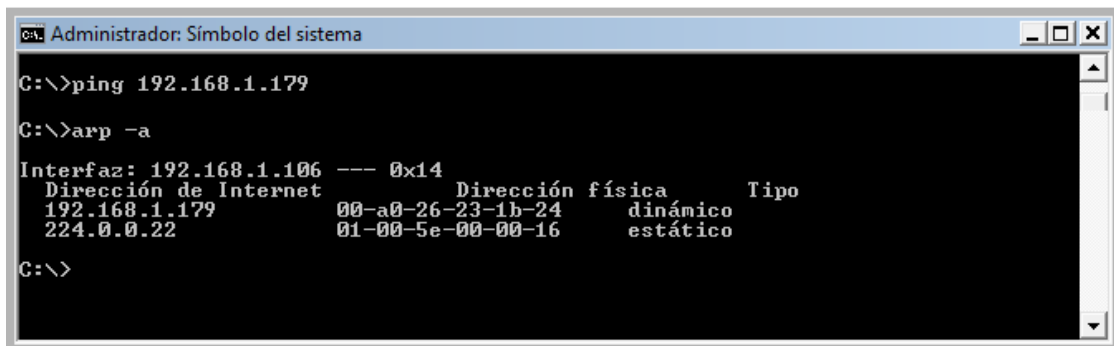
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IV.1.1.2. Codec MAC address

The user can retrieve the MAC address from the codec using any computer sharing the same subnetwork.

The procedure is:
1. "ping" to the known IP address of the codec. ("CMD tool" if Microsoft OS"
2. issue an "arp –a" command. The request MAC address is displayed inside the list.

IV.1.2 Ports

It is possible to configure which ports the unit will use when selecting Prodys proprietary protocols for IP communications.
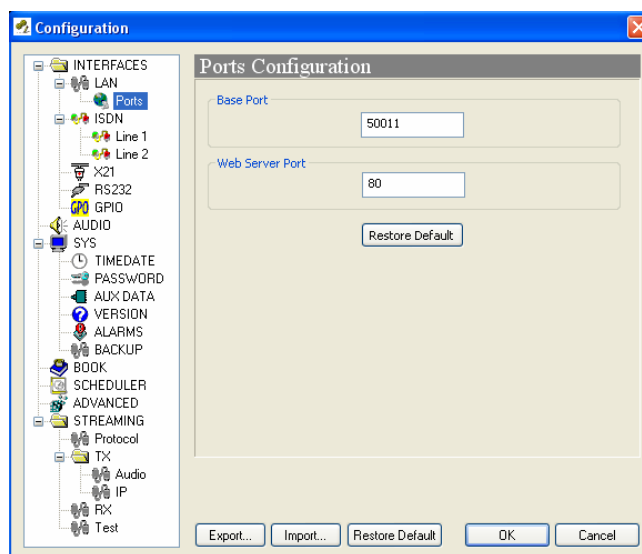
There are two different groups:

- **<Web Server Port>:** By default, it is TCP port 80. This is the internal web server port.

- **<Base Port>:** By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports **used by Prodys Proprietary Protocols V1 & V2,** along with the management ports. From this base port, up to 30 ports should be opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required), in order for the audio and management traffic is routed appropriately to the ProntoNet. This is the simplest way to deal with the port configuration of routers/firewalls. In case only the management ports have to be opened a more detailed list of ports and different configuration examples are provided in chapter V.6.4 More about Prodys Proprietary Protocols. The configuration and ports used by Prodys Proprietary Protocols V3 and V4 are describe on section V.6.6.

The following things should be taken into consideration when changing the base port:

1. To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:

   http://<IP>:<Port> Example: 192.168.0.10:8080

   To log into the unit, click on the advance features button of the login window:

and specify the base port which we want to connect to:

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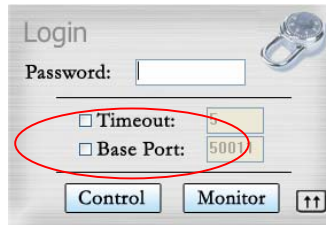
When establishing a call, the new base port must be entered after the IP address, separated by a colon. Example: When calling to a unit which IP address is 192.168.1.2, and base port 40011, the following call destination should be entered into the 'Dial' window: **192.168.1.2:40011**.

When establishing a bidirectional call with Prodys IP codecs using Prodys **Proprietary Protocols V1 or V2**, two connections are made automatically, one for each direction. For the receiver to call properly to the caller (*only in case the base port in the caller is not the default one*), it should have an entry on its phone book with the IP address of the caller and the base port of the caller. Example: if the caller has the IP address 10.1.5.0 and its base port is 40011, in the receiver, the following entry should be configured in its phone book:
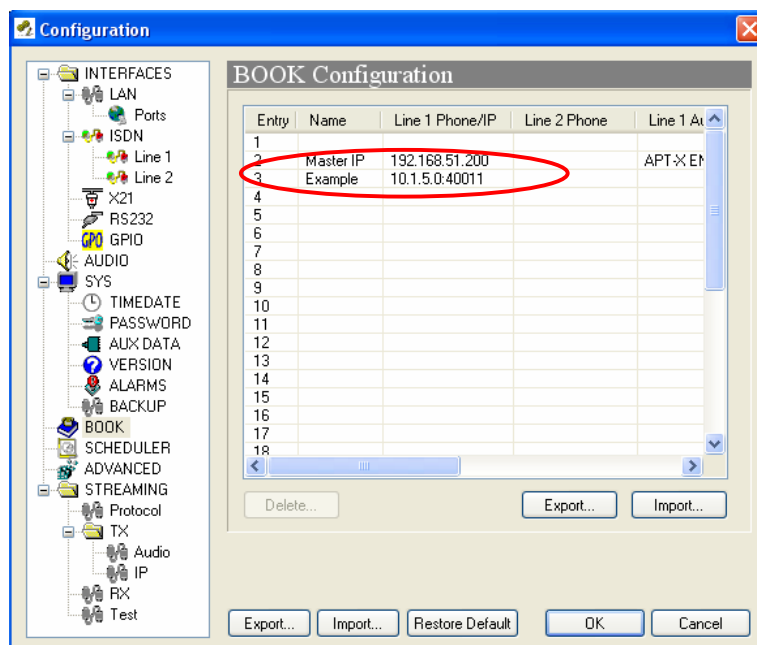
NOTE: Take into account that this is not necessary when working with Prodys Proprietary Protocol V3/V4 protocol.

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IV.1.3 SNMP configuration

It is possible to set the SNMP community.
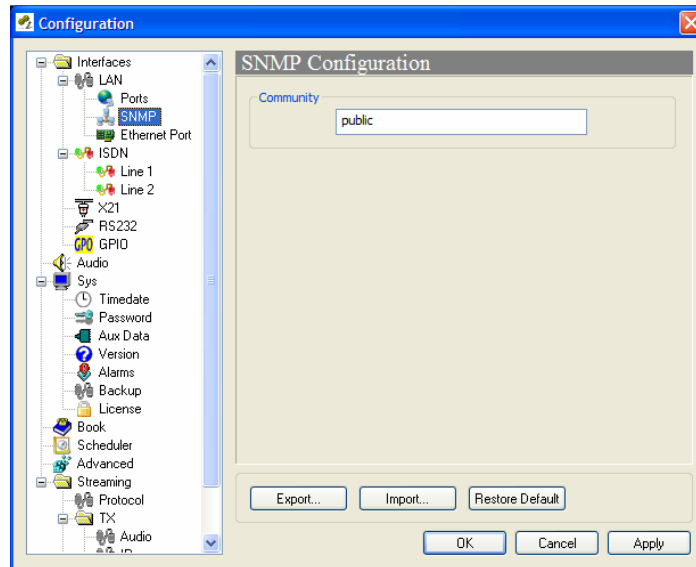
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IV.1.4 Ethernet Port Configuration

It is possible to set speed and duplex configuration to the following values:
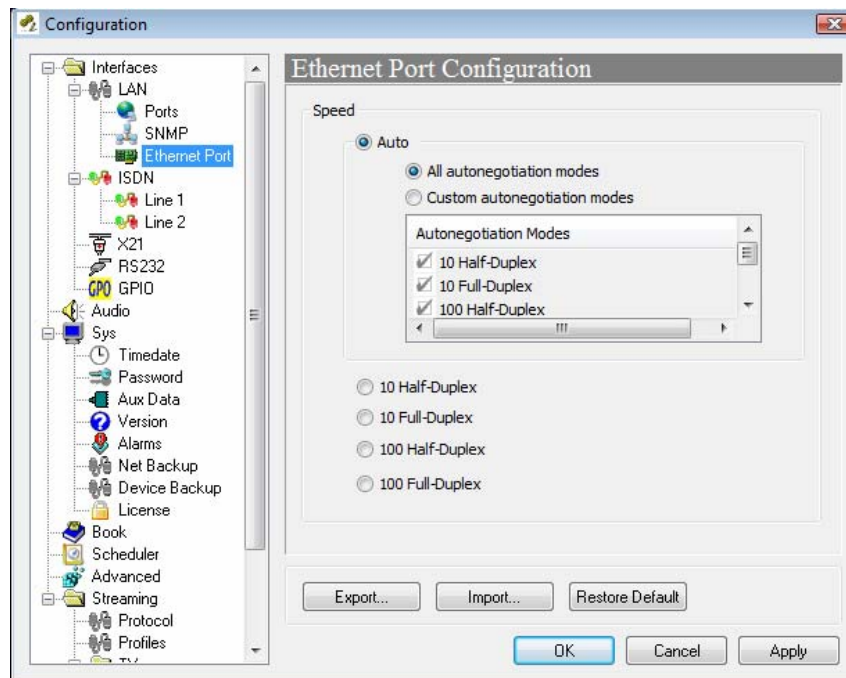
NOTE: When the unit is connected to a device which does not support auto-negotiation, and it is set to work in a fixed mode, the Ethernet interface must be configured to match the configuration of that device, so 'Auto' should not be selected in this case.

IV.1.5 ISDN Terminal Adaptor configuration

The first dialog allows the selection of the ISDN protocol according to the national network in place. There are four options:
Euro-ISDN, 5ESS, DMS 100 and NAT1:

The options Line 1 and Line 2 allows the user to configure in each line (B channel) the Local Number, Calling number filters and the answering mode:
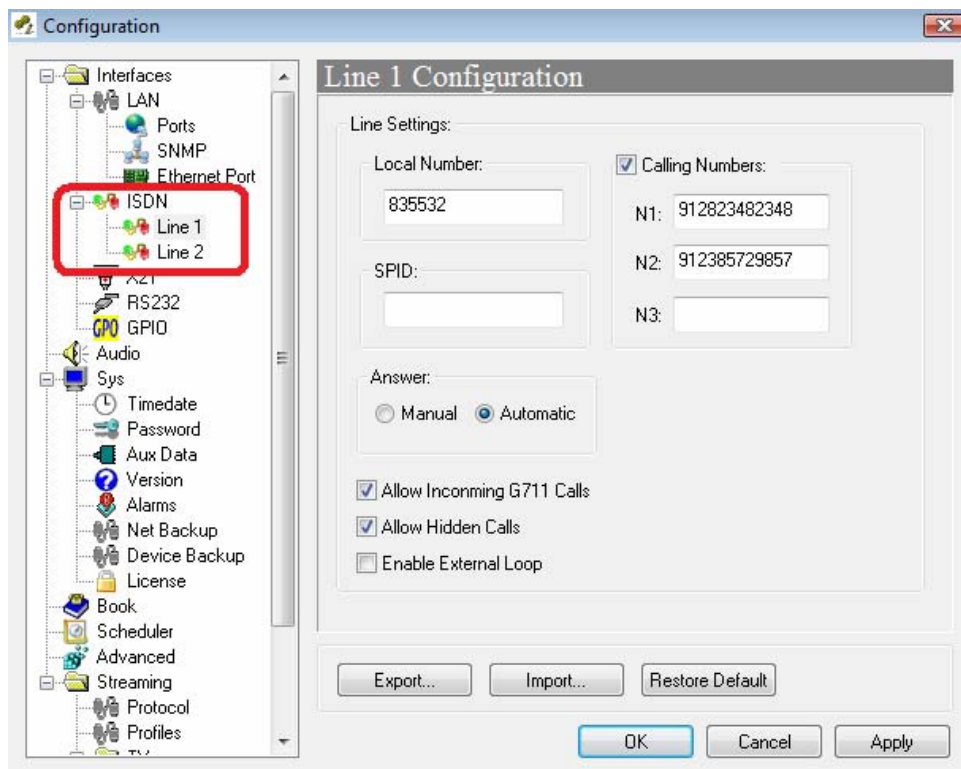
- **<Automatic>** or **<manual> answer:** Incoming calls can be answered automatically or manually upon this option. If the mode is set to manual,

the red-blinking <Connection> bar must be pressed to accept the call and connect to it.

- **<Calling Number>:** It is possible to define up to three numbers for each line which performs as an in-dialling filter. This is, the line will only connect to calls with these caller IDs.

- **<Local Number>:** It is also possible to assign a single number to each line in such a way that the line will only respond to calls made to this local number. This can be used if you need to map an ISDN directory number to a specific audio port.

- **<Allow incoming G711 calls>:** the ISDN indicates calls on G711 as voice mode. The Prodys IP Family codec recognises this type of incoming call and proceeds automatically to reconfigure the Encoder and Decoder to this mode. As such, the user need not configure anything to allow the automatic reception of incoming calls using G711.

IV.1.6 X21 Port:

From this option the speed of the X21 port is selected.
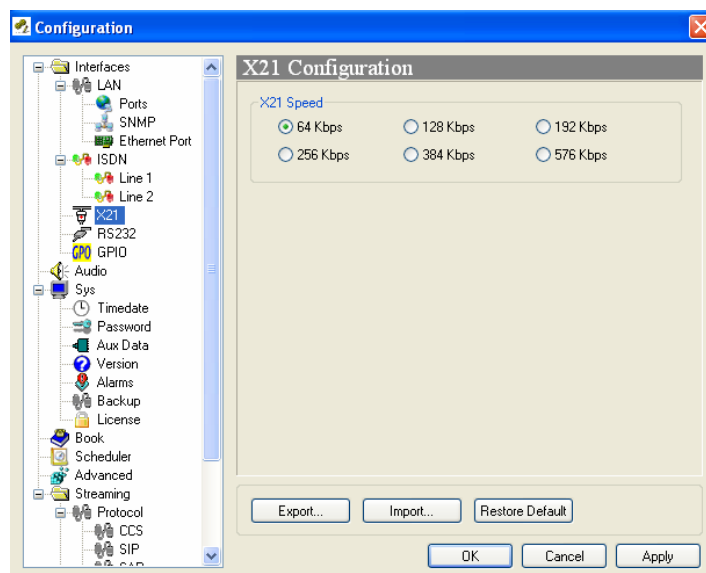
IV.1.7 RS232 Ports

There might be up two RS232 ports for use as auxiliary data ports, depending of the model. These ports allow the transmission and reception of data along with encoded audio. The port configuration will be always set to 8 DATA bits, NO parity, 1 START bit and 1 STOP bit. The bit rate can be adjusted to between 300 and 9600 bps via software.
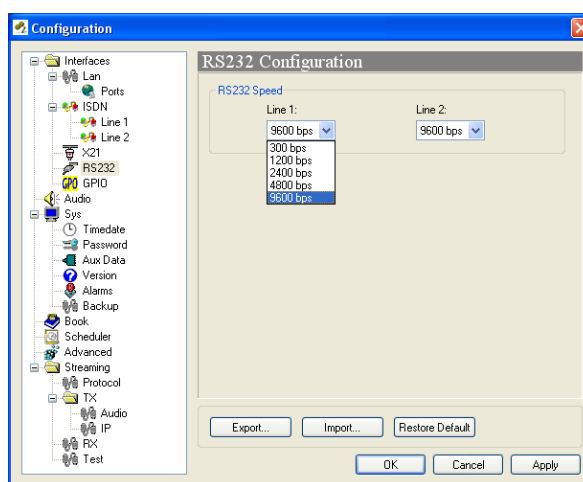
IV.1.8 GPIO Port

From this menu the GPIO port is configured. The amount of GPIO available might be different for the specific Prodys IP Family codec, although the GPIO definition for operation remains in any case.

INPUTS	
Transparent	Under this configuration, the state of the input will be present in its homologous output in the Prodys IP Family codec connected on the other end.
Connect Line1/2	When this input is activated, the Prodys IP Family codec will proceed automatically to connect the line 1/2. The user can select which entry of the book will be used for this purpose.
Hang-Up Line 1	When this input is activated, the Prodys IP Family codec will proceed automatically to disconnect the line 1/2.
Mute Left	When this input is activated, the left audio output will be muted.
Mute Right	When this input is activated, the right audio output will be muted.
Acknowledge alarms	When this input is activated, the alarms will be acknowledged.
Answer Line1	Line 1 will be picked up when receiving a call.
Answer Line2	Line 2 will be picked up when receiving a call.
Call/Hang-Up Line 1/2	With the same input the user can establish and finish the connection..

Disable FAN	This control enables or disables the internal FAN

> The inputs are activated by grounding.
>
> Transparent configuration only works with audio algorithms that support ancillary data and have been previously activated.

For electrical in/out ratings and examples of application please refer to the "Hardware and Reference Manual" of the codec.
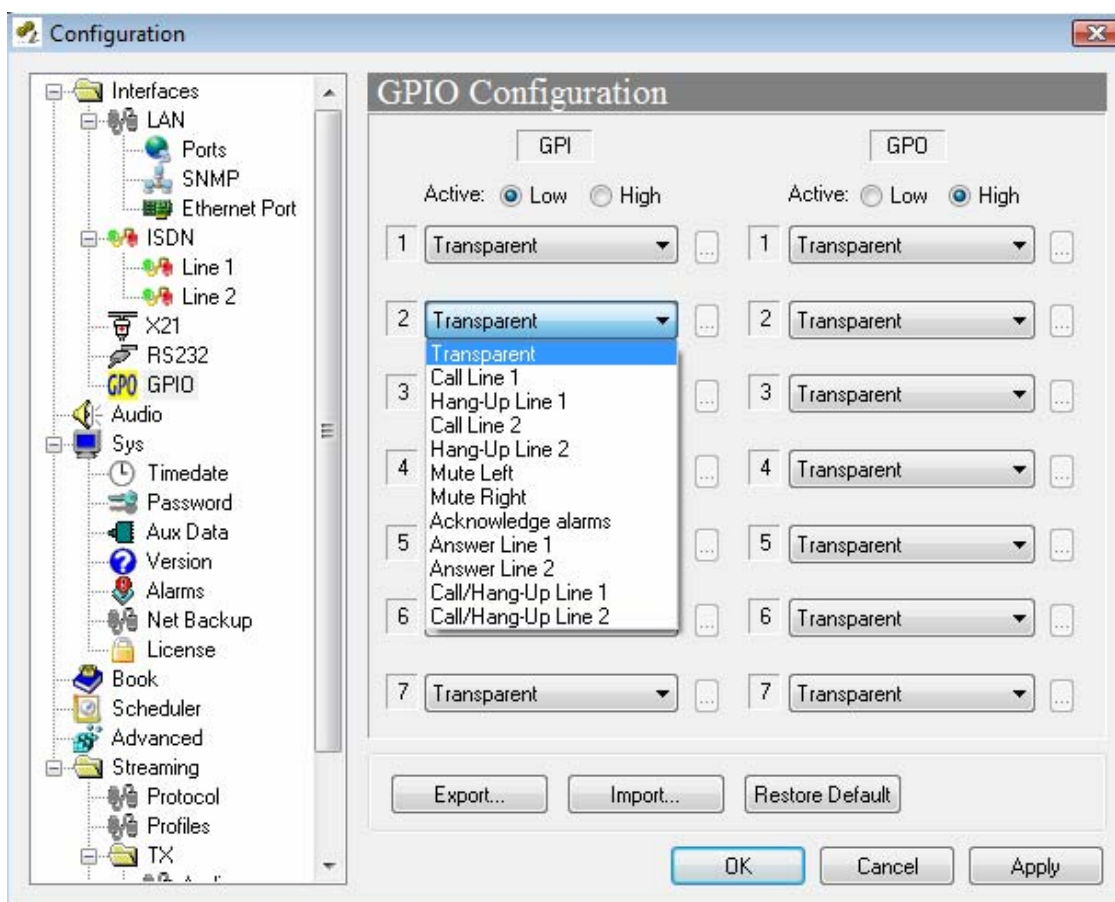
OUTPUTS	
Transparent	Under this configuration, the state of the output will be the same as the corresponding input of the Prodys IP Family codec connected on the other end.
Line Connected	The output will be activated when one line is connected. The user can define which lines are to be monitored. (The picture below shows the application for multi-unicast connections)
Line Disconnected	The output will be activated when one line is disconnected. The user can define which lines are to be monitored. (The picture below shows the application for multi-unicast connections)
Decoder 1 Framed	The output will be activated when the Decoder 1 is Framed.
Decoder 1 NOT framed	The output will be activated when the Decoder 1 is NOT Framed.
Decoder 2 Framed	The output will be activated when the Decoder 2 is Framed.
Decoder 2 NOT framed	The output will be activated when the Decoder 2 is NOT Framed. It is only useful when the Prodys IP Family codec is working in ISDN mode as a DUAL CODEC.
Backup Active	The output will be activated when the ISDN Backup is working.
Backup NOT Active	The output will be activated when the ISDN Backup is NOT working.
Alarm Active	The output will be activated when one Alarm is activated.
Alarm NOT Active	The output will be activated when there are NOT Alarms activated.
DTMF detected	DTMF is detected by the Decoder.
Line 1/2 Status	Line Disconnected: GPO set to '0'. Line Connected: GPO set to '1'. Receiving a call: GPO blinking (1 second period).

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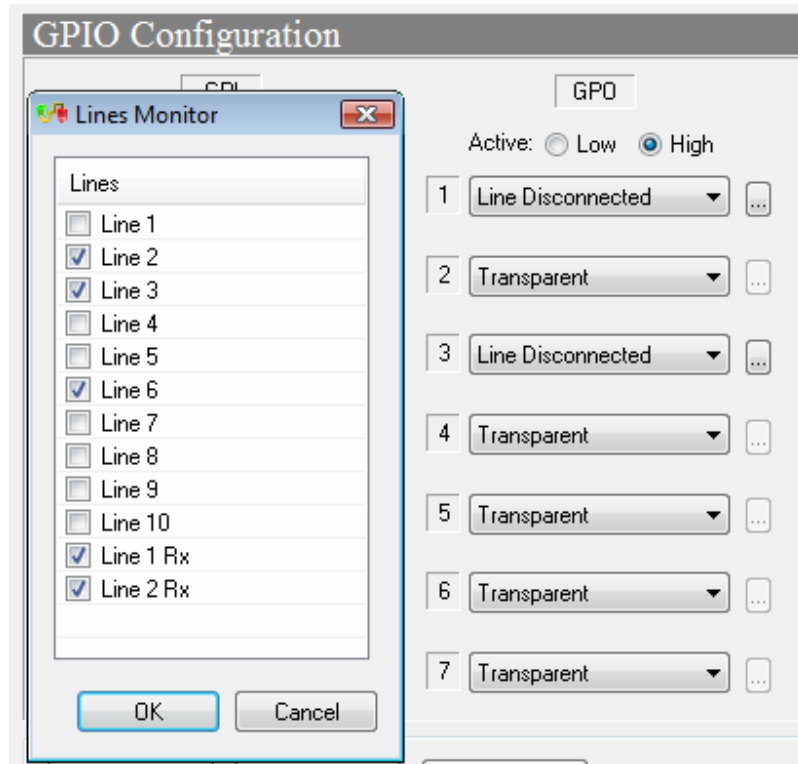
When a GPO is configured to monitor alarms <Alarm active>, it is possible to select which alarms will enable this output.
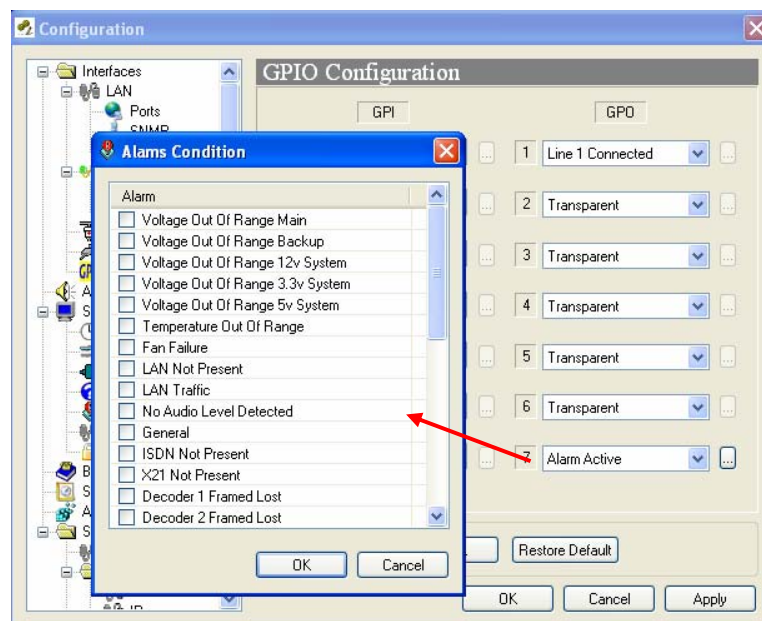
Temperature out of range alarm will arise when the temperature goes over 50ºC.

Lan traffic alarm will come up when more than 90% of the LAN bandwidth is in use.

This GPO will be activated when one or more of the selected alarms arise.

In addition, it is possible to program a GPO to be enabled when a DTMF is detected in either Decoder. The decoder number and the DTMF which will enable the GPO must be defined by the user.
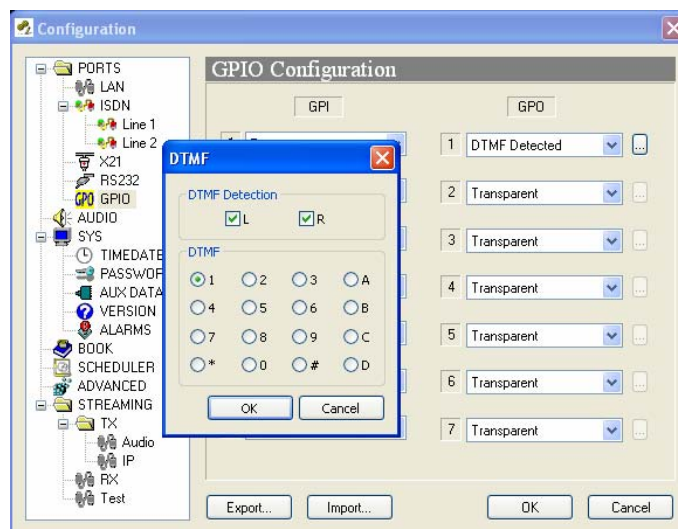
IV.1.9 Audio Configuration

<Analog> or <AES/EBU> (digital audio) input is selected at this menu. The digital input mode provides several choices:

The <AES/EBU> which just takes and supplies the audio information from /to the AES/EBU digital frame.

The <AES/EBU Transparent> is a special input/output mode which sets also the network transmission mode to <PCM 24 bits> avoiding any audio payload processing at all by the codec. This mode allows the exchange of non linear audio at the input (or data over the AES3 interface) between codecs (e.g. transmission of Dolby E streams).

Audio gain adjustment: Audio inputs and outputs can be adjusted digitally. Each audio channel can be adjusted independently.
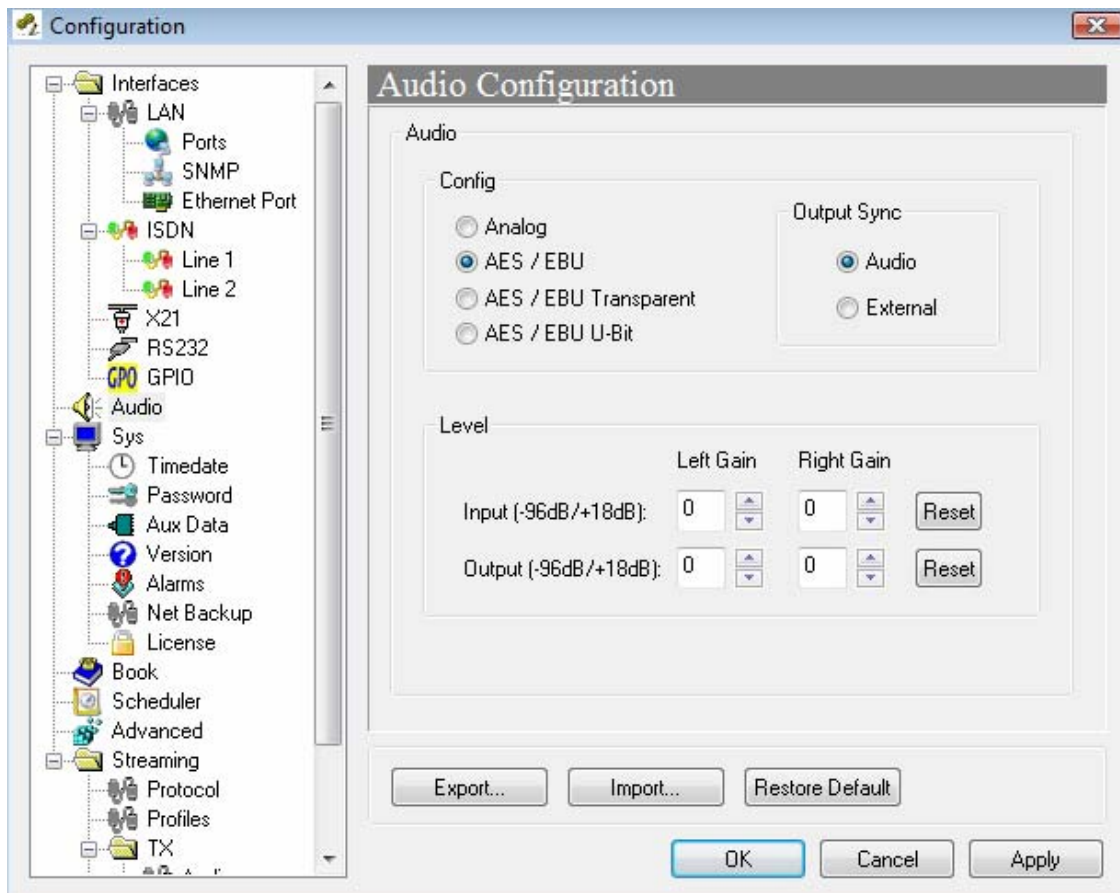
The <AES/EBU U bit> option allows the user to route also the User Bit data from the AES/EBU frame along with the audio stream over IP. This can be useful to control remotely some third party devices sensitive to the User Bit of the AES/EBU frame.

Since the decoded audio will be always available on both output interfaces (analog and digital), several syncs could be applicable for the digital stream over the AES3 interface. Syncs could be:

- **<External>**, which means the AES3 derives its sync from [sync input] of the digital audio interface. As opposed, if <External> is not selected with option <Analog>, the internal AES3 clock is enabled.
- **<Audio>**, which means the AES3 derives its sync from the [AES/EBU input] of the digital audio interface.

IV.2 System Configuration

Grouped here are functions that affect the general operation of the unit.

- **<Loop>:** This sets up an Encoder-Decoder loop. Its purpose is to help the user to check audio connections before streaming. The loop takes place in accordance with the encoder configuration in place at that moment.

- **<PLL>:** This is a special option that allows you to select the PLL reference clock when NET = IP.

- **<Redial>:** It is possible to configure a number of redials for attempting the first call of this communication. The Prodys IP Family codec will try the connection in the following cases:
    - 1.- The unit is trying the connection for the first time.
    - 2.- The unit was connected and lost the connection because the line dropped or the line was disconnected from the other end.

- **<Keypad>:** This option allows the user to lock automatically the frontal panel keypad.

- **<Monitor>:** This option allows the user to hide or display additional information on the main window:

    - VU meters: Indicates audio levels.
    - System: Monitor voltages, temperatures, and fan operation.
    - Streaming: Shows clock-sync algorithm operation and Buffer Occupation Graph (Real Time Network Analyzer). This algorithm comes into scene when receiving audio over IP. This display is active when the audio data is "framed" - Under normal circumstances the indicator should be in the middle Green area.
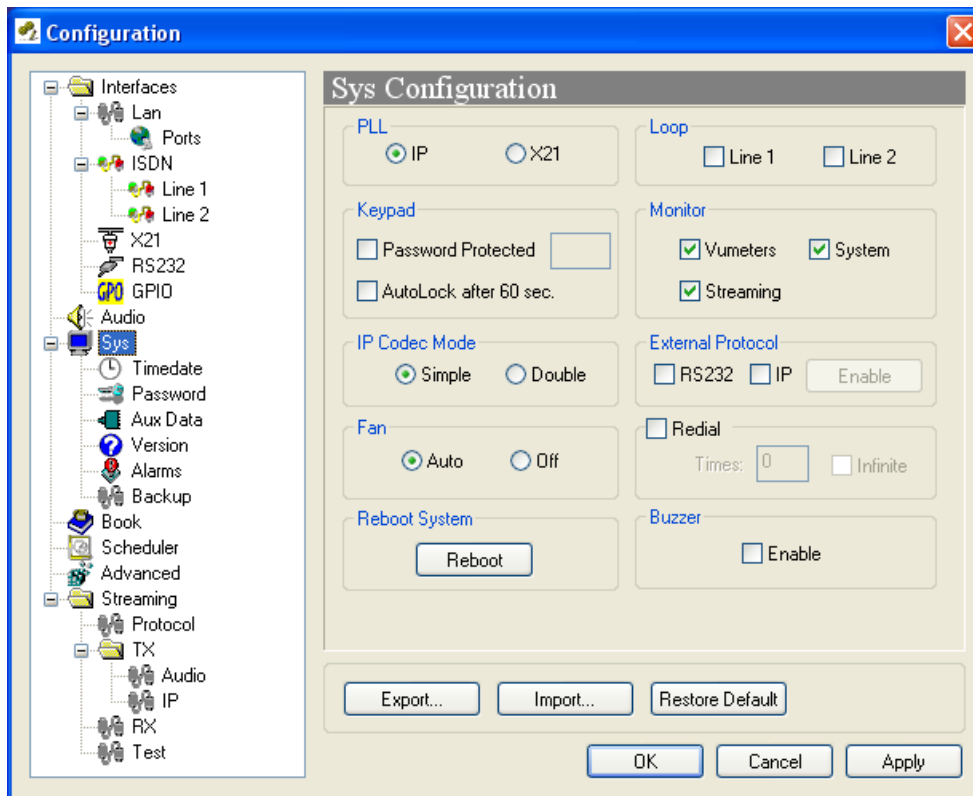
- **<Reboot System>:** Restarts completely the codec with last configuration. This option could play as a hardware reset for codecs placed remotely.

- **<External Protocol>:** The Prodys IP Family codec Software Development Kit provides a tool to manage the Prodys IP Family codec units from external applications or devices different than ProdysControl. In this way, it is possible to customize the control of the codecs and integrate them in a global management system.

  Please refer to the Prodys IP Family codec SDK User's Manual.

- **<IP Codec Mode>:** There are two different operation modes: SIMPLE and DOUBLE mode. In SIMPLE mode, the unit will work as usual, that means, it cannot establish two independent communications via IP. In DOUBLE mode, the unit can establish two independent communications via IP.

  In DOUBLE mode over IP, the following restrictions will be applied:

a) Only mono signals can be encoded and decoded.
b) PCM and AAC HE streaming is not supported.
c) Apt-X and enhanced Apt-X are supported only at 16kHz sampling frequency

- **<Buzzer>:** It is possible to enable/disable the buzzer from this option. The buzzer warns the user about incoming calls when the unit is configured as an ISDN, IP and PSTN codec and the answer mode is set to manual.

IV.2.1 TimeDate

This option allows to synchronize time and date by using SNTP protocol (Simple Network Time Protocol). SNTP operates always in the client-server model and for this reason Prodys IP Family codec can work as SNTP server or SNTP client.
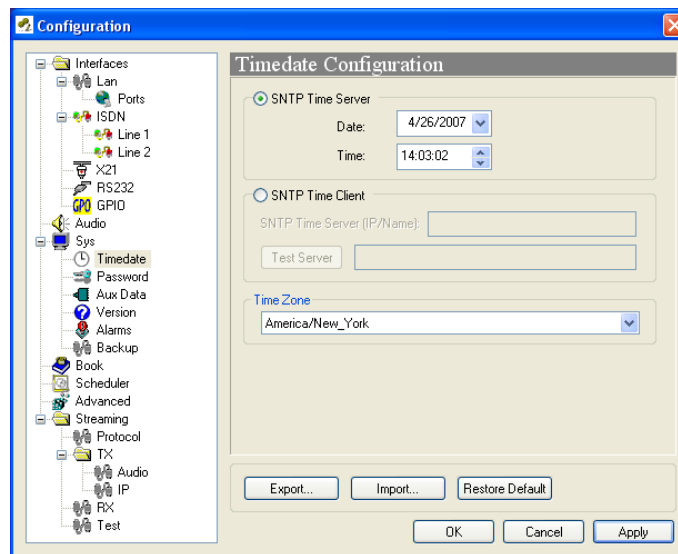
IV.2.2 Password and User Profile definition

Whenever the web browser is started, it is necessary to enter a security password. The user profiling is available at the <Password> menu. Different password for the administrator and other users can be provided.

Enabling or disabling some codec functions, the administrator defines the required profile for any user. The users identify themselves at the login display by their password.

Furthermore the defined restrictions are applied also to the keypad access by means of the codec's front panel. Codec menus are enabled according to the password given into the keypad.
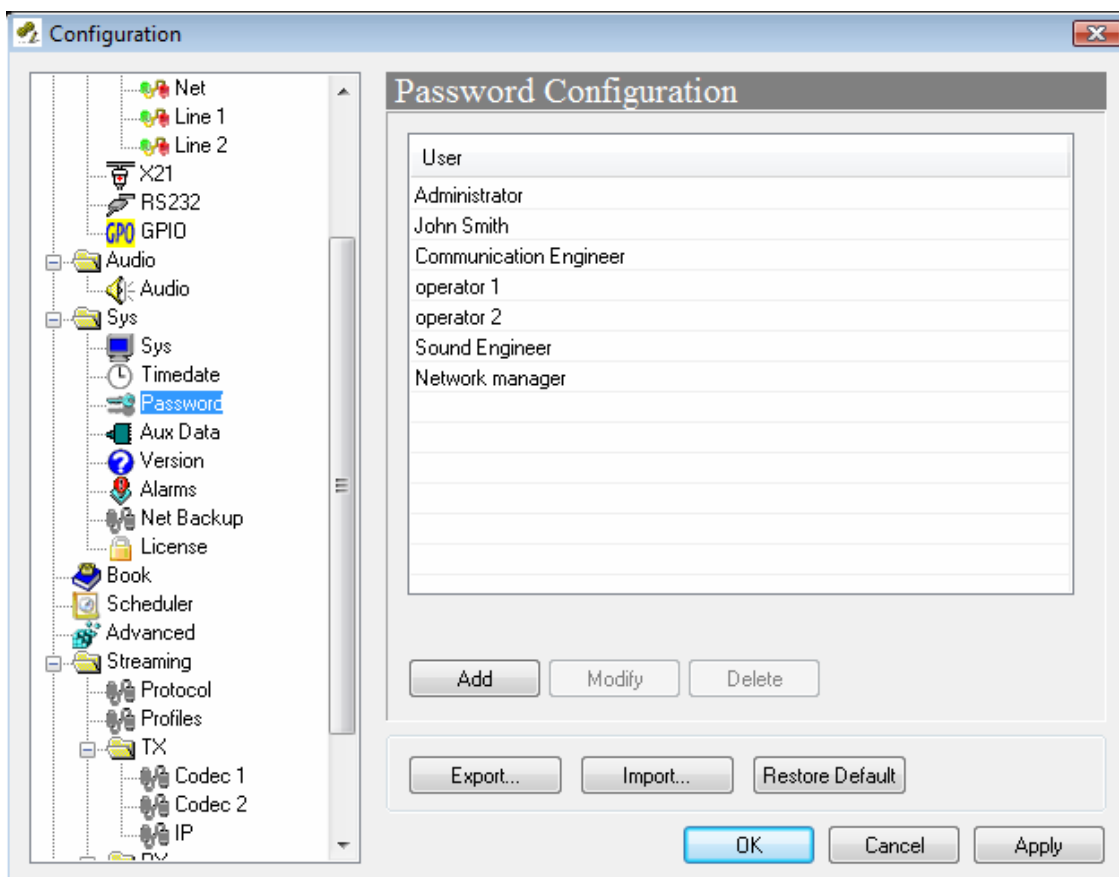
The codec administrator can prevent basic configuration conflicts and still enable non-technical staff making use of the codec.

The user profiles do not define access priority. This means any user might attempt to log into the codec at any time. If the codec is in use by another operator, he got a warning message which must be confirmed before get a successful login. This procedure and a smart user profile definition allow a very flexible and concurrent management of the codec.
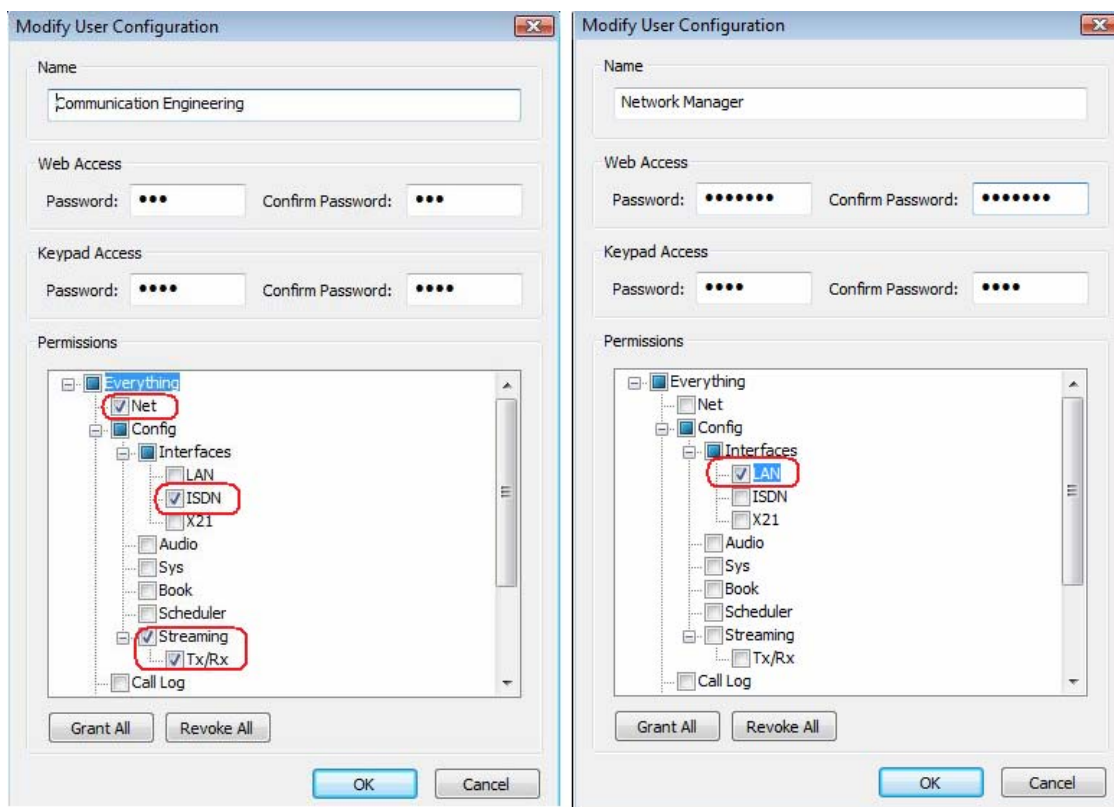
On the previous picture an example is given for two independent management profiles.

The "Network Manager" profile is allowed only to set the IP address of the codec according to his private network plan. The "Communication Engineer" is prevented to modify the IP address, but he is allowed to switch the codec to IP streaming or ISDN streaming. Furthermore this expert is enabled to modify any IP streaming relevant parameter (streaming protocol in use, time between packets, jitter buffer).
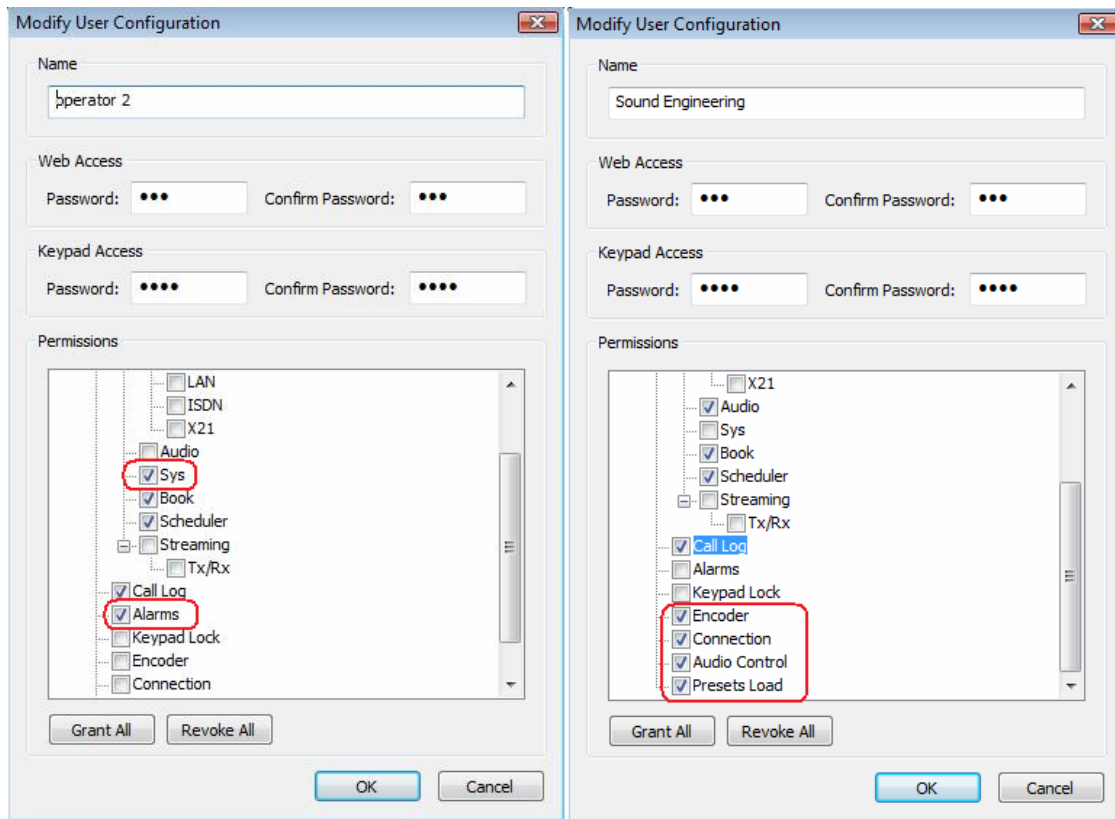
On the previous picture an example is given for two other independent profiles, but with some common access rights.

The "Operator" profile requires definition of the alarm type to monitor, license updates, calendar and clock setting among others, Book and Scheduler updates. On the other hand the "Sound Engineer" is enabled to start and stop audio streams, to choose the best compression algorithm for this communication and other settings.

It is remarked that the user profiles prevents only modification of the some configuration parameters; the read access of all parameters is always granted. Back to the last example: the "Sound Engineer" can read the complete record of alarms, but he can not disable or delete alarms from the log.

User profile can be modified or deleted. The user profile definitions are part of the codec configuration; therefore exporting and importing the codec configuration include the complete user profile definition.

IV.2.3 Aux Data

- RS232: There can be up to two RS232 ports for use as auxiliary data ports. These ports allow the transmission and reception of data along with the encoded audio. Port A is always ready. Port B is only available in some codecs of the Prodys IP Family when the codec is configured as a DUAL Codec, therefore able to operate two totally independent communication channels. When one option is checked, the corresponding decoder will allow the reception of ancillary data.

  Over ISDN, it is possible to choose between Prodys IP Family codec format and Pronto2/3 format (in which the GPIOs information cannot be sent). To be backward compatible with ISDN Pronto2/3 Prodys devices, select Pronto2/3 format.

  When selecting IP as communication interface, the auxiliary data can be sent via a different path, different from the audio one (Aux. data embedded in audio). This method has 3 big advantages: less delay independent from the audio codification delay; the possibility to send/receive auxiliary data regardless of the compression mode used for the audio communication; and the possibility to send/receive the User bit of the AES/EBU frame. The drawback is that the audio and data delay won't be the same.

- GPO: Under 'transparent' configuration, the state of the input will be present in its homologous output in the Prodys IP Family codec connected at the other end. Since there is only one GPO connector, user must select whether to select GPO information coming from Decoder1 or from Decoder2, or none.
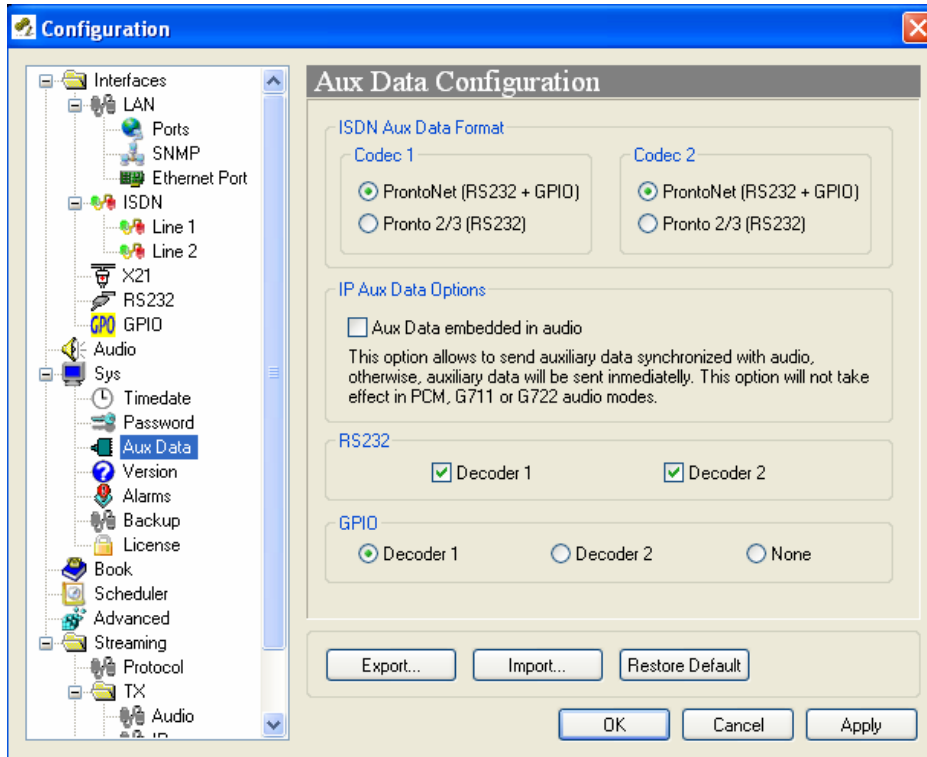
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IV.2.4 Software Versions
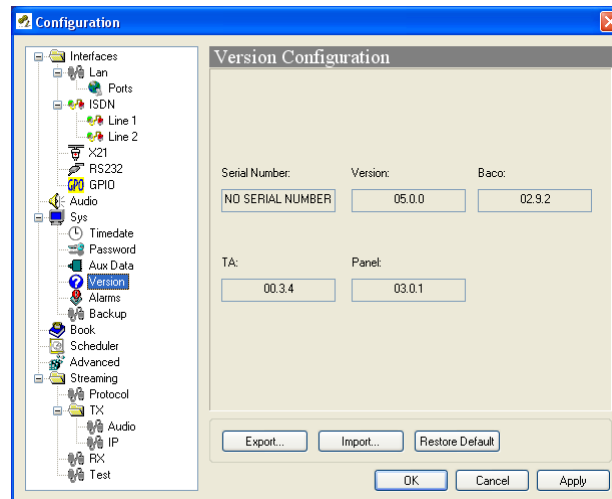
IV.2.5 Alarm paging

This menu is available for setting alarm paging over IP networks. Notifications are available by means of:

- SNMP traps
- SMTP email

Please refer to section IV.9 for details on the user Alarm configuration.

The user might enter multiple IP destinations for SNMP traps to be sent. Usually a Network Management Tool will display and log the alarm information accordingly. Integration of the Prodys IP Family of codecs into an overall customer network operation and management centre is feasible with this protocol.

These SNMP V1 traps will notify any event status selected, this is, an SNMP V1 trap will be sent when an alarm is activated or deactivated, including timestamp and further information.

A complementary or alternate alarm paging tool is to notify via email. Several email destinations might be selected for automated notification of events.

The alarm paging by email requires the configuration of some mail server parameters as:

- **<IP/Host Name>:** any valid IP address or Host Name for the SMTP Server.

- **<Port>:** default SMTP port is 25.

- **<User>:** User to access to the SMTP services.

- **<Password>:** Password to access to the SMTP services.

NOTE: Take into account that some email servers requires encrypted messages. Prodys IP codecs do not support it.

![ProdysIP Manual](header)

ProdysIP Manual

IV.2.6 Net Backup

Network Backup Configuration. The Prodys IP Family codec allows the user to use the ISDN line as a backup when the unit works in IP or X21 mode. Please referrer to section 0 for details.

IV.2.7 Device Backup

The Device Backup feature provides N+1 (including 1+1) protection switch algorithm for multi codec facilities. Please refer to section V.10 for details.

IV.2.8 License

Some codec features are licensed by Prodys and are not available for the factory default. This allows Prodys to meet different pricing policy according the real customer requirement for advanced features.

The user might check and update the codec licenses for new features at the <License> menu.

- <Yes>: The feature is licensed for this codec (i.e. for this serial number).

- <No>: The feature is not available for this codec.

Steps for upgrading the feature licenses are:

- Contact sales@prodys.net for the new feature with indication of the codec <Serial Number>. The serial number is displayed on the <License> menu.

- The user is provided with a License File by Prodys

- Press de <Upgrade> option on the <License> menu.

- Upload the License File from the computer directory to the codec.

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It is noted that the feature license is valid for any future Prodys IP Family versions. In other words the user is allowed to update the codec version any times, the feature license will not be revoked by the version update process.
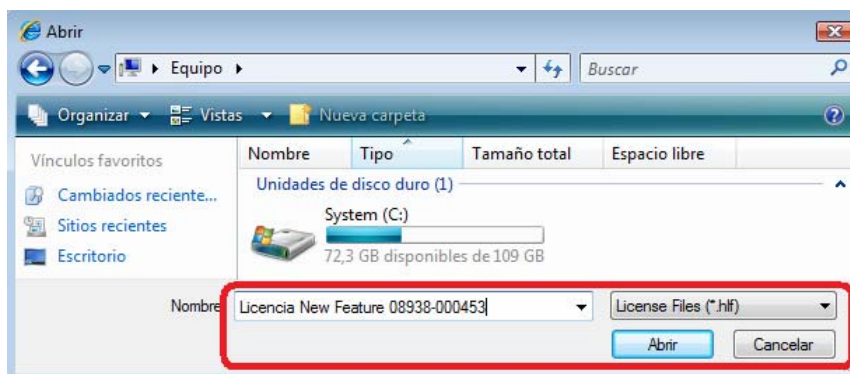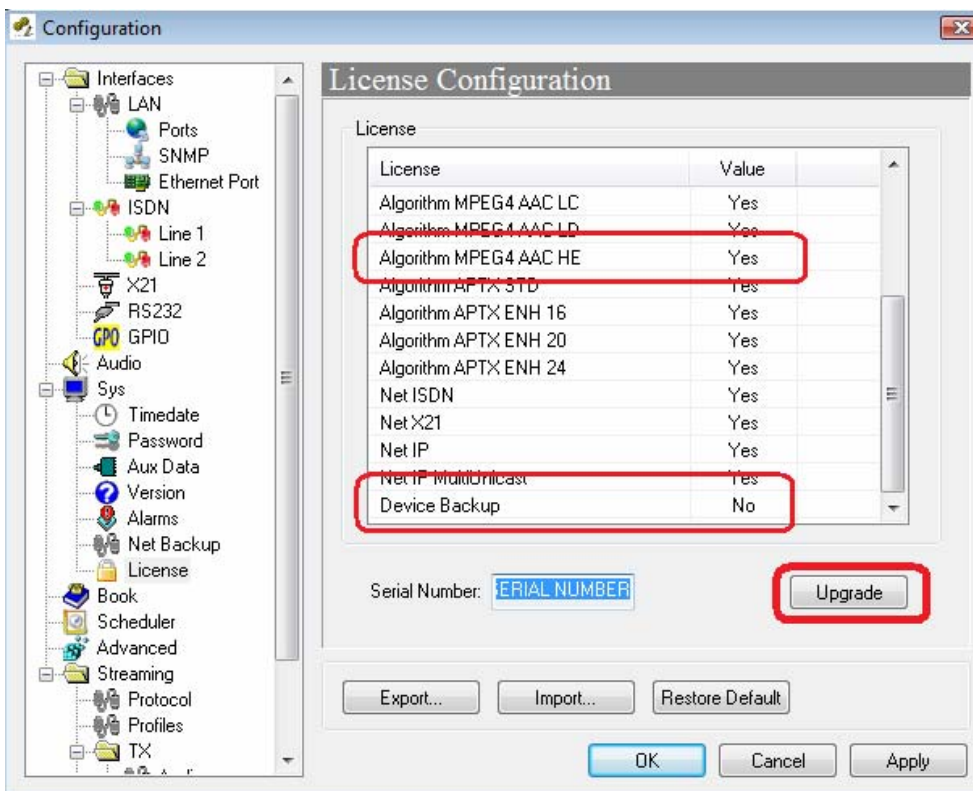
IV.3 Book index

When the BOOK menu is selected, the phone book window is displayed on the right side. The Book index records the user name, a telephone number (optionally two), an IP address. Up to 64 registers are available.

> The Book index keeps ISDN and IP destinations apart. Therefore for accessing the ISDN Book index the codec must be set to the NET=ISDN. Conversely for accessing the IP Book index the codec must be set to the NET=IP.

The proceeding to edit a phone book index is as follows:

1. To select one index by clicking on its area.
2. To select ISDN numbers or IP Address depending on which network will be used. Only it is possible to select one of them.
3. To enter the ISDN number/numbers or the IP address according to the before selection.
4. To configure the encoder is optional. In case of that it is not entered, the unit will proceed to call in the current encoder configuration.
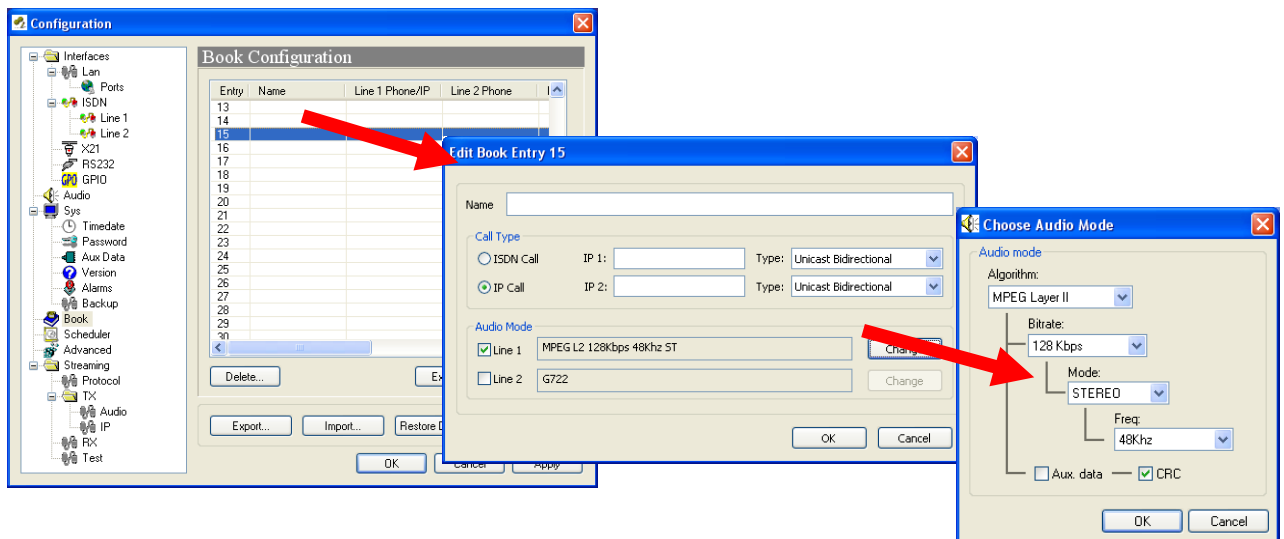
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The user is allowed exporting to or import from the Book in XML format on the computer disk. Additional options are preview end printing of the index content.
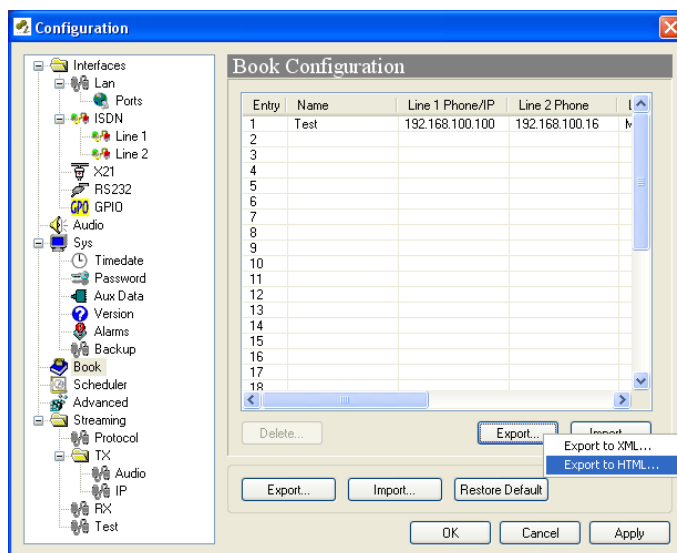
IV.4 Scheduler

The Scheduler allows the user to program calls to be dialled automatically. In addition, the duration and the period for each call can be defined.
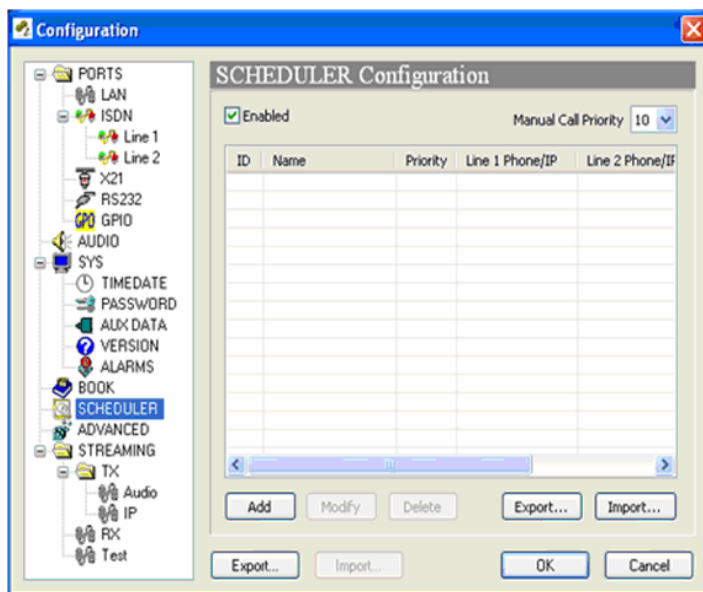
IV.4.1 Enabling and disabling the Scheduler

The Scheduler dialog has a checkbox which allows the activation/deactivation of the Scheduler. When the Scheduler system is disabled, none of the scheduled calls will progress. In addition, each scheduled call can be enabled/disabled one by one.

IV.4.2 Automatic and manual call

It is possible to define the precedence for manual or for automatic calls. Thus, the user can decide if one automatic call will take precedence over a manual one. The call priority can be set to a number between 1 and 10 (highest priority). The codec will dial any higher prioritized scheduled call against any ongoing lower prioritized call. The default priority for automatic calls is 1, and the default one for manual calls is 10. Manual calls mean those initiated by the user from the web browser control. It is possible to modify the default priority for the manual calls from the Scheduler dialog, by changing the value of the field 'Manual call priority'.

IV.4.3 Programming the scheduler

To add a new automatic dial, click on the 'Add' button in the Scheduler configuration window. The following window will be displayed:

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This dialog allows the user to configure all the parameters related to an automatic dial.

<Name>: This field allows the user to enter a brief description to identify this call. This name will appear in the call list of the Scheduler. By default, the name will be "Scheduled Call" followed by a number.

<Start>: The "Start Date" and "Start Time" fields permit the user to configure the date and time when the scheduled call or pattern will be enabled.

<Priority>: By clicking on "Priority" the user can change the priority value assigned to the current entry.

<End>: There are two methods to define how long the call will last: "Recurrence and duration" and "End date and Time".
- o **"Recurrence and duration"**

This option allows the programming of call repetitions based on time or date: Time recurrence and Date recurrence.

To configure the unit for "Date recurrence" mark the corresponding checkbox in the Scheduler configuration window and click on <Recurrence>. The following window will be displayed:

This window is arranged in two different sections:

<Recurrence pattern>: This section allows the user to configure the dates when the call will be made. It is possible to program daily, weekly, monthly and yearly patterns.

<Range of recurrence>: This section is divided in two different parts:

- o <Starts on>: This option sets the start date and time for the configured pattern.
- o <Ends on>: It is possible to define the end of the configured pattern in three different ways: Without an end date, after a certain number of repetitions, or for a specific date.

By clicking on <Preview and edit> tab, the user will obtain a calendar wherein those days when this pattern is active will be coloured on red.
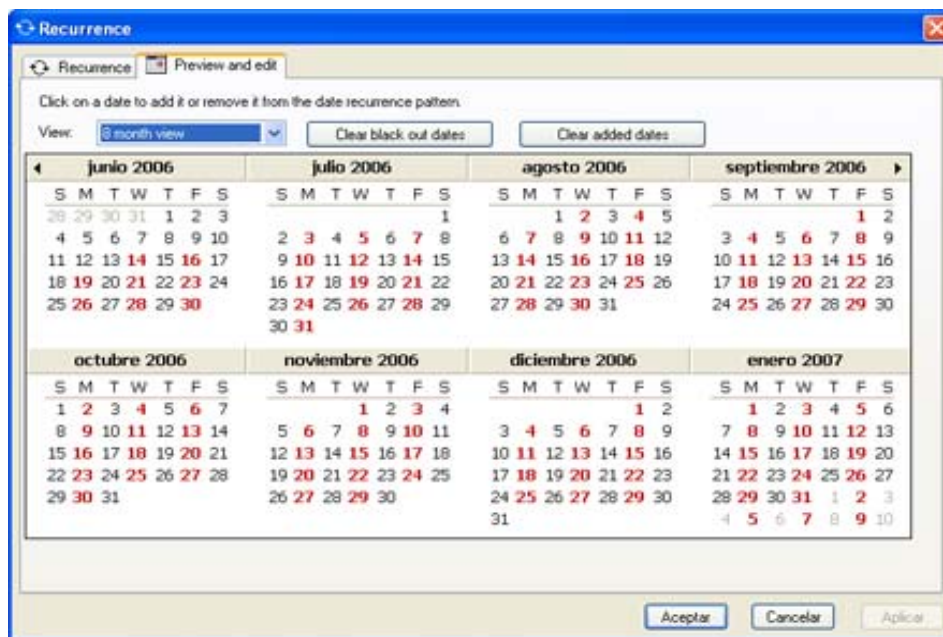
To configure the unit for <Time recurrence> mark the corresponding checkbox in the Scheduler configuration window and click on <Recurrence>. The following window will be displayed:
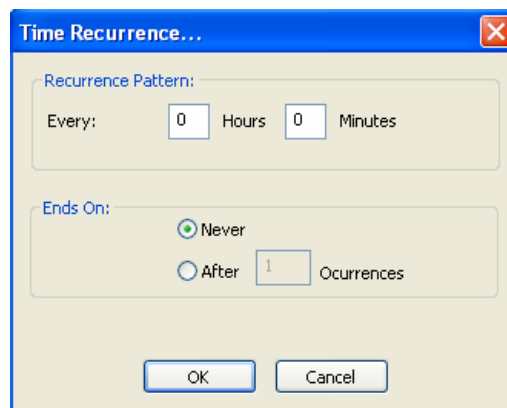
This window is arranged in two different sections:

-   "Recurrence pattern": This section permits the user to configure the period of time for this call to be repeated.
-   "Ends on": From here, it is possible to establish the end of this recurrence. It can be   never or   after an specific number of occurrences.

    o   **"End date and time"**

This option allows the user to establish the duration of a programmed call by defining the end time and date. The call will be established in the date and time defined by the **"Start Date"** and **"Start Time"** fields.

IV.4.4 Scheduled call configuration

It is necessary to configure the type of network, the numbers and the compression mode that will be used for call establishment. To do that, the user can configure these parameters manually, or automatically, by getting them from one of the entries of the phone book.

IV.4.5 Manual configuration

To configure these parameters manually, clic on "Edit…". The following dialog will be displayed:

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From this window the user can enter all the parameters involved in a call establishment.

IV.4.5.1. Automatic configuration from the phone book

To configure these parameters automatically, click on "Load From Book…" A window with all the entries of the phone book will appear. From this window the user can select any of the existing entries by clicking on any of them and on the OK button. The selected entry will be read and its configuration will be stored for the current pattern.
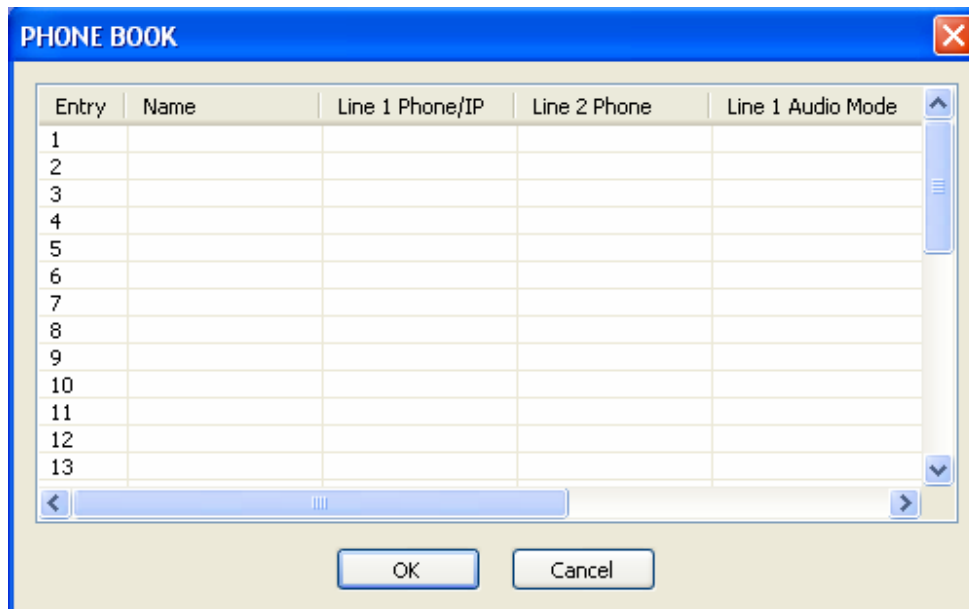
IV.4.6 Modifying existing scheduled calls

To modify a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Modify", the user will be able to edit all the parameters related to the select pattern.

IV.4.7 Deleting existing scheduled calls

To delete a pattern or scheduled call, the user has to select one scheduled call from the Scheduler list on the Scheduler Configuration Window. Then, by clicking on "Delete", the select pattern will be deleted.

IV.4.8 Copying and pasting scheduled calls

From the Scheduler configuration window, the user can copy and paste previously configured scheduled calls.

To copy a pattern, select one from the list and right-click on it and select the option <Copy Call(s)>.

To paste a pattern, select one from the list and right-click on it and select the option <Paste Call(s)>.

IV.4.9 Monitoring scheduled calls

When one call related to one of the patterns defined in the scheduler of the unit is in progress, the message "SCHEDULED CALL IN PROGRESS" will be displayed on the web page for the duration of the call.
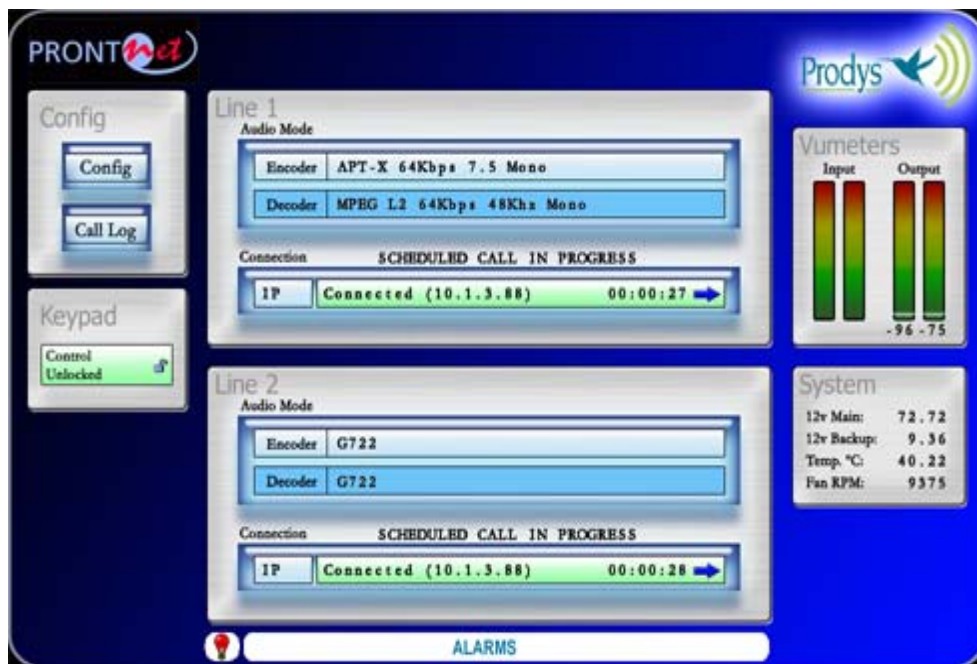
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IV.5 Advanced Configuration

Advanced configuration parameters might be set on this dialog box. For any modification of the current value double click the proper option.
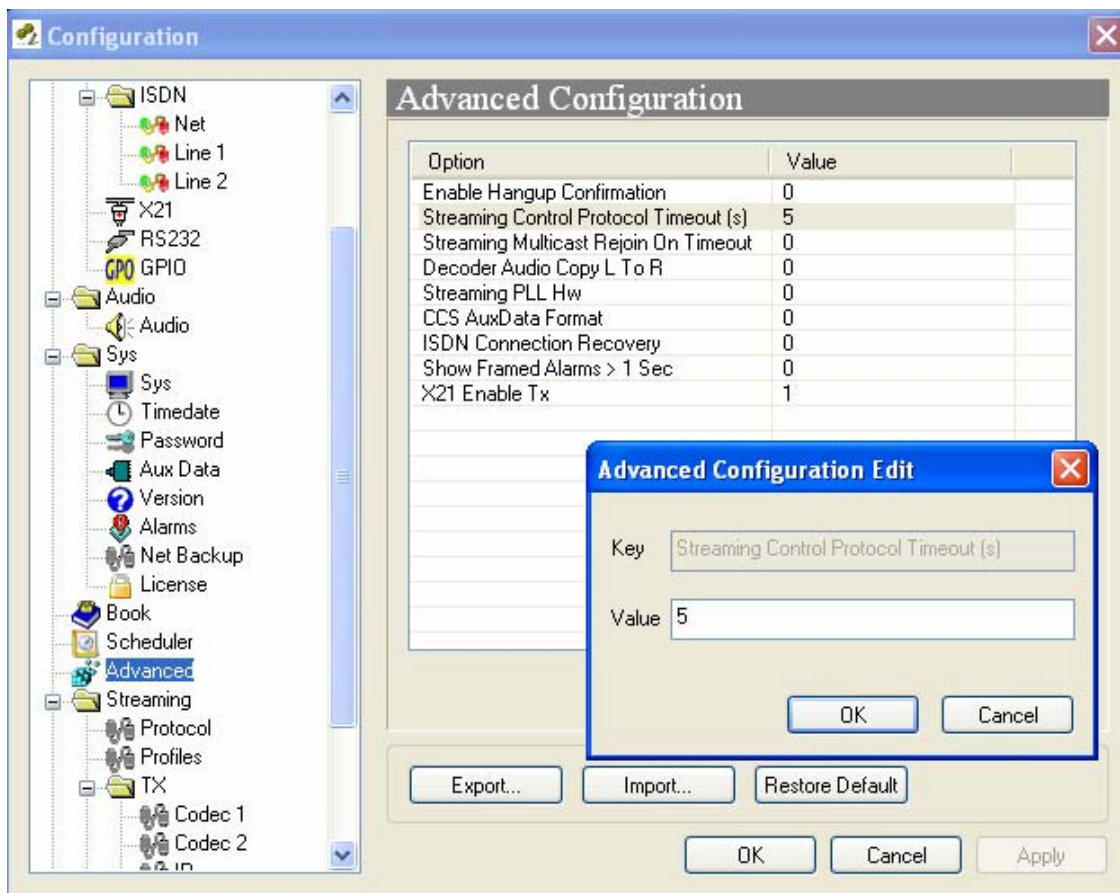
<Enable Hangup Confirmation>: If the value is <1>, the application will ask confirmation before disconeccting the line.

<Streaming Control Protocol Timeout>: When the link is down, or the mate codec is unreachable for the specified number of seconds, the audio connection drops automatically. After the protocol timeout a new communication might be started.

<Streaming Multicast Rejoin On Timeout>: IGMP protocol functionality.

<Decoder Audio Copy L to R>: This option is enabled with value <1>. This feature is mainly valuable for hearing mono decoded signals on the left and right output channel.

<Streaming PLL HW>: Please, contact support@prodys.net for further information about this. The default and advisable value for this parameter is <0>.

<ISDN Connection Recovery>: It is possible to force the unit to reconnect when the ISDN connection drops, even though the connection was hung up intentionally from the remote end.

<Show Framed Alarms >1s>: Short audio gaps, (shorter than 1 second), can be filtered from the alarm list with this advanced configuration option. This avoids alarm flood streaming over error prone networks.

<X21 Enable Tx>: It is possible to disable only the transmission on the X21 interface. To do that, this parameter should be set to '0' and the X21 port should be enable by clickgin on the 'CALL1' key or by clicking on the connection bar on the web page.

IV.6 Streaming

The Streaming folder displays several menus which provide the user access to the key IP network parameters:

- Selection of a communication protocol compatible with the mate codec.
- Adaptation to the actual network performance.
- Information about the audio streaming efficiency over the IP network, delay, etc.

IV.6.1 Protocols

Several communication protocols are available for starting, monitoring and ending an audio stream over the IP network. From now on those are referred to as just "protocols".

The list of protocols available for Audio over IP streaming is as follows:

- Proprietary

- Proprietary V2

- Proprietary V3

- Proprietary V4

- CCS NetStar /AudioStar

- RTP

- SIP

- SAP

Usually the newest proprietary protocols include features from the former proprietary protocols. Prodys still provides the full range of protocols for compatibility with legacy codec versions. In general it is recommended to use the latest protocols, which includes all the company experiences up to the date.
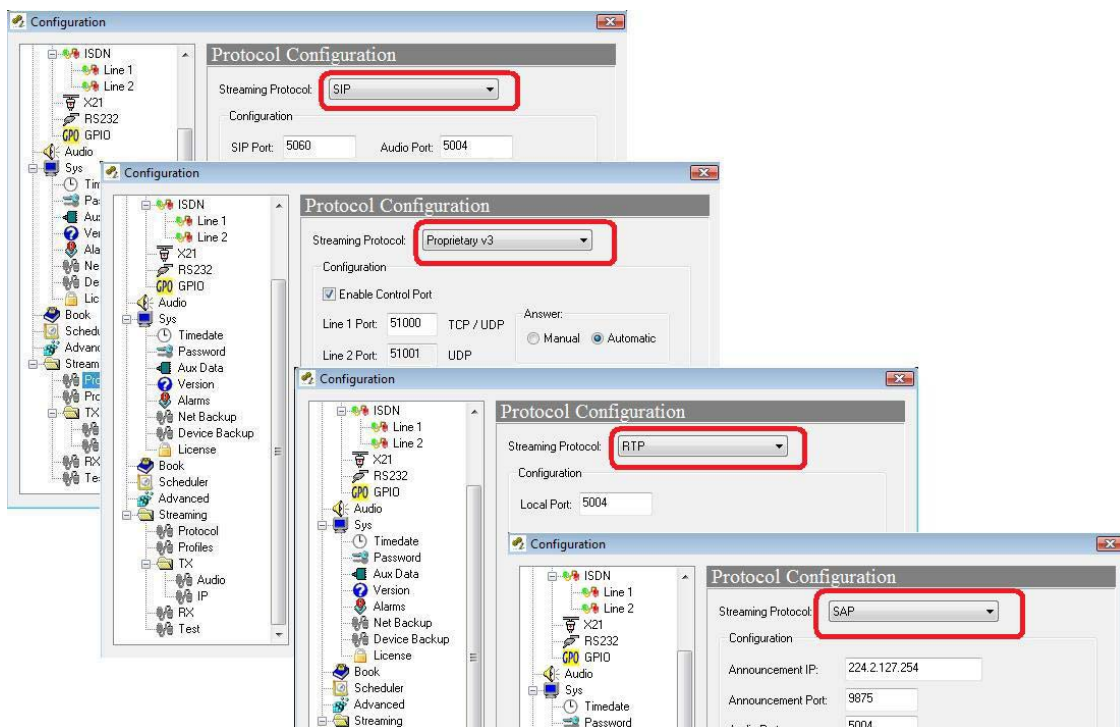
IV.6.2 Proprietary Protocol (V1)

This is Prodys` legacy IP protocol since 2004. It has been proven to be very robust and useful for a broad amount of customers.

Selection of sub-parameters are not applicable for this protocol, therefore this menu is blank. However the user should keep in mind that the factory default configuration for Proprietary Protocol V1 is affected by the menu <Port Configuration> (please refer to IV.1.2).

Information for advanced users is available at V.6.4.

IV.6.3 Proprietary Protocol (V2)

Prodys` second IP protocol enhances the former V1 protocol with additional network performance monitoring features like Lost or Disordered Packets. keep in mind that the factory default configuration for Proprietary Protocol V1 is affected by the menu <Port Configuration> (please refer to IV.1.2).

IV.6.4 Proprietary Protocol (V3)

Prodys` third IP protocol is build up with a different approach although any former features are still available. Prodys' V3 protocol is aimed to make the streaming over Internet easy for the user.

More specific, the V3 offers a NAT traversal ability. This means that the caller end of a new IP communication does not require asking for router or firewall adaptation before streaming. This protocol enables to stream from any place like convention centres, sport stadiums, hotel, etc. without meeting the network administrator.

At the destination, usually the studio, the gateway must forward the ports defined at the V3 protocol menu.

Another application for the V3 protocol is allowing one-way IP connections (e.g. satellite or radio links). These applications require disabling the Control Port.

The V3 protocol compared with V4 protocol allows the lowest delay possible for low delay compressions (PCM, G722 and APT-X), since the time between packets (TBP) can be set to the absolute minimum. Attention should be paid if very small TBP are in use due the performance stress applied on the overall network (i.e. routers, etc.).

IV.6.5 Proprietary Protocol (V4)

This protocol is similar to the V3 protocol, but emphasizes the streaming robustness over Internet based on the Forward Error Correction (section V.6.15) application.

The proprietary protocol V4 interlaces the redundant audio information with the main audio information on the same UDP Destination Port. V4 users can accommodate the FEC overhead to the available bandwidth or bit error rate of the network. This selection ranges from 4% to 100% in 24 steps.

In the example configuration <Audio packets per FEC packet =5> means that one packet loss over 5 audio packets could be healed. The overhead provided in this case is about 20% in regard with FEC disabled.

It should be noted also that FEC introduces additional audio delay, which could affect significanty low delay communications. The account for the additional delay can be made based the amount of audio packets sent between two consecutive FEC packets.

The general audio delay formula is:

  + network delay
  + jitter buffer
  + encoding /decoding delay
  + FEC number x Time Between Packets
  = Transmission delay

In general 100% FEC (FEC=1) is the best choice for low delay communications.

Following example is provided:

+ Given an average 16ms network delay
+ Given an 10ms jitter buffer provision (<User Line Delay>)
+ Given a 25ms encoding /decoding delay for a low delay G722 compression
+ Given a FEC=2 by TBP=5ms
= 61ms total transmission delay

Out-of-order arrival of audio packets at the destination codec could an issue for internet transmissions or complex IP infrastructures with load-balancing routers. Now our V4 protocol, like our RTP/SIP/SAP protocols do, manages this network hurdle transparently for the user.

Similar to V3, the V4 protocol allows one-way IP connections (e.g. satellite or radio links). These applications require disabling the Control Port.

Prodys Proprietary Protocols V4 and V3 support NAT traversal.

IV.6.6 CCS NetStar /AudioStar protocol

This protocol is applicable if compatibility with CCS Musicam IP codecs is required.

It is remarked that this protocol does not include automatic procedures for taking an incoming call or streaming supervision. Therefore the user is required to enable or disabling manually any incoming or outgoing stream.

IV.6.7 RTP protocol

This protocol performs plain streaming of audio embedded into RTP/UDP packets. The main application is also compatibility with a very wide choice of audio streaming devices.

It is remarked that this protocol does not include automatic procedures for taking an incoming call or streaming supervision. Therefore the user is required to enable or disabling manually any incoming or outgoing stream.

IV.6.8 SIP

The European Broadcast Union (EBU) is promoting the interoperability of audio codecs for any manufacturer. For this purpose the use and the application of a subset of the Internet Protocols has been proposed. This effort will allow to setup in a friendly way audio streaming communications between several vendor equipments.

The deployment of SIP Protocol Servers among the network will support calling remote parties just by invoking their network name regardless of the actual public o private IP addresses. Neither is required to manage the office routers and the firewall's TCP/IP ports for any new communication path.

The physical location, often related to fixed IP addresses or subnets, is not further meaningful. In this sense, portable audio codecs will profit with an easy call procedure. Neither is required to agree ahead on the audio compression type and data stream rate, because the SIP Protocol manages by itself to negotiate the convenient communication details with the remote party.

For Prodys' former customers performing a call using SIP is much the same as using Prodys' proprietary protocols given that the SIP configuration is set.

SIP supports up to date only unicast streaming. Any Prodys IP Family codec's audio compression mode is supported also if SIP protocol is in use.[11] For further details please refer to the "Technical Description" at this document.

▪ **<SIP Port>:** TCP/IP port dedicated to SIP signalling for establishing, updating or finishing a call. Port 5060 is mandatory by the standard if direct calls without gateways are expected.

▪ **<Audio Port>:** TCP/IP port dedicated to the RTP communication, this is the actual port for the audio streaming payload. By default the number 5004 is reserved.

▪ **<Public IP Address>:** This parameter is necessary when making connections through a router to the Internet.[12] The user must copy the public IP address of his router into this box.

---

[11] The apt-X compression is still undefined by the EBU Tech 3326 standard. For compatibility reason this mode is not supported if SIP protocol is selected.

- **<Automatic>** or **<manual>** answer of incoming call request can be defined.

If SIP server support is granted by your network:

- **<Server IP>:** IP address of the SIP Server. At this point is registered periodically the proper user information for full protocol support.

- **<Server user> & <Server password>:** Some SIP servers requires authentication before proceeding the register of SIP users.

- **<Username>:** Your alias on the Internet regardless of your current IP address. By this alias you are identified for other SIP participants.

- **<Timeout>:** Time in seconds before the user information registered at the previous server is flushed. Therefore the information about the SIP user are updated from time to time.

- **<FEC>:** Configuration for the Forward Error Correction applicable to the outgoing audio stream is defined at this section for SIP. For further details on FEC please refer to section V.6.15.

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[12] Routers translate local or private IP addresses into public WAN IP addresses (e.g. Internet). This translation of address is broadly known as NAT.

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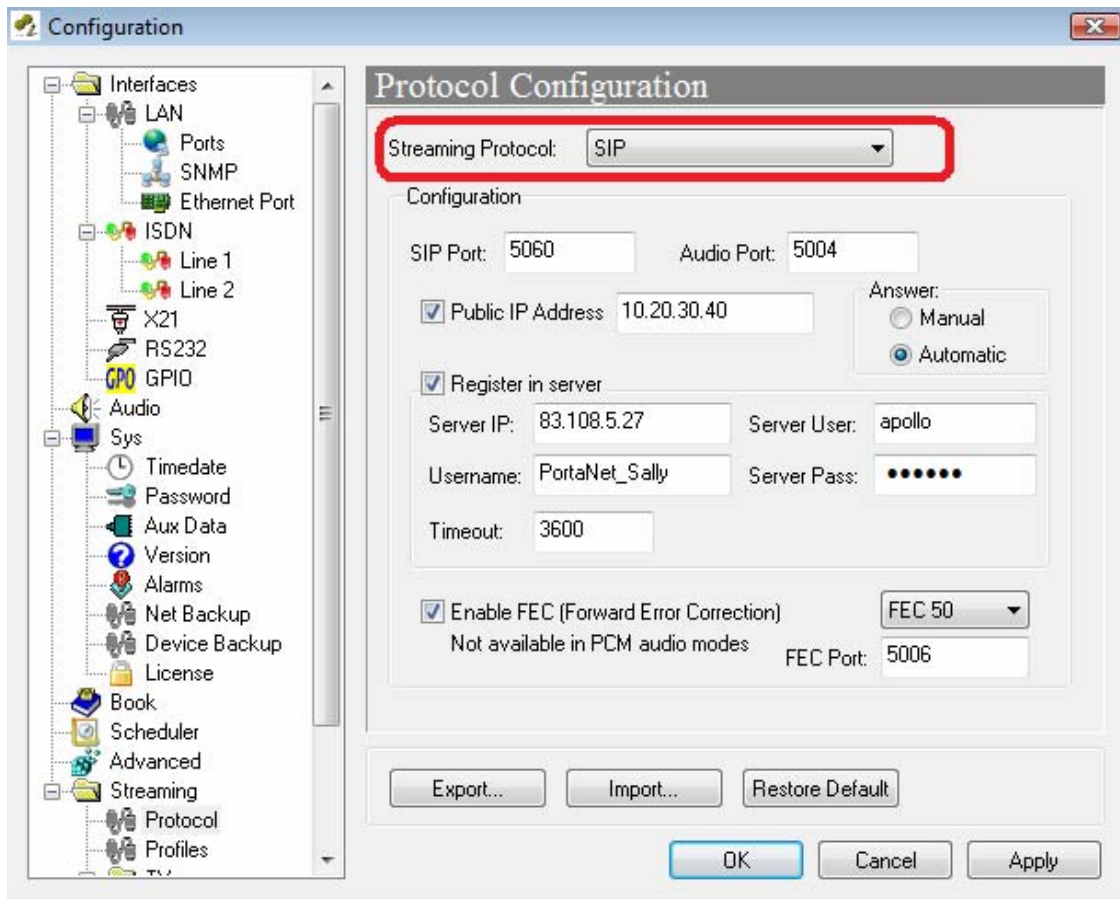
IV.6.9 SAP

When selecting SAP as communication protocol, only multicast calls will be available (point-to-multipoint). In addition, the user will have to indicate the announcement IP address and port on which the audio streaming is 'announced'.
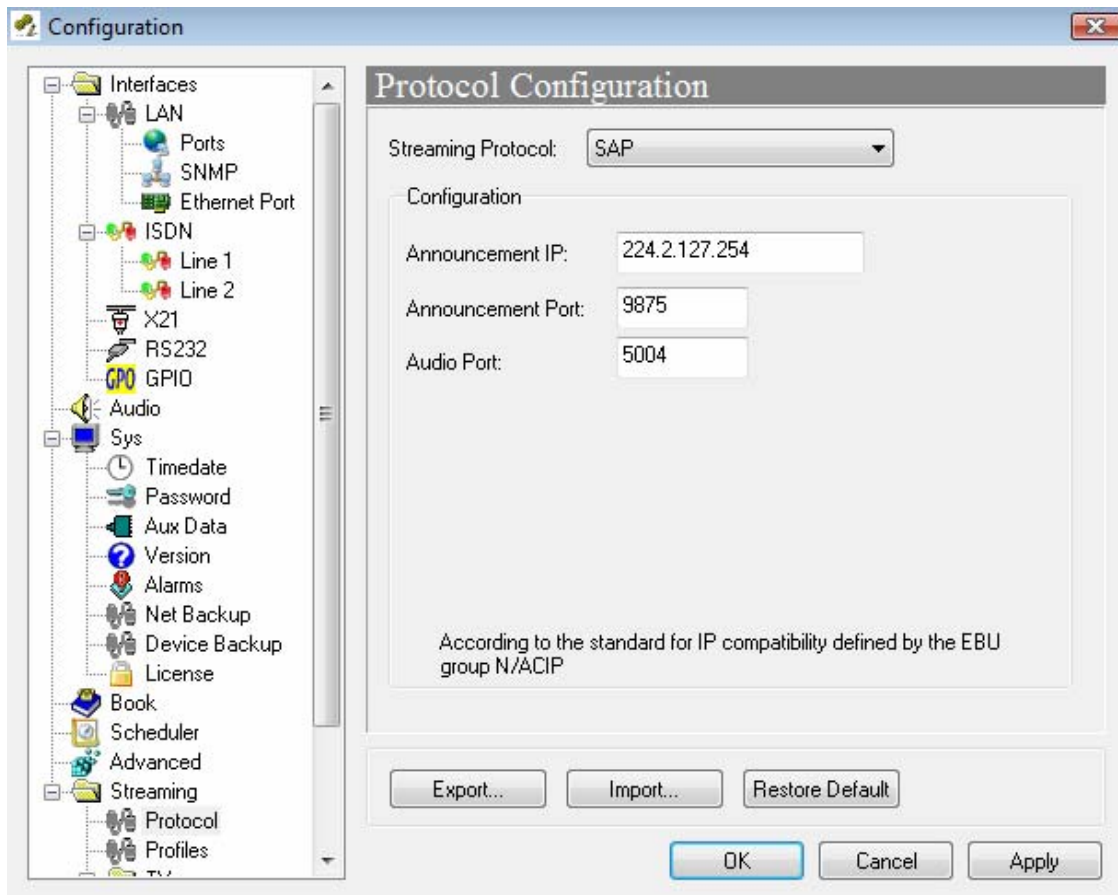
The default values for these parameters are the default ones defined by the SAP standard (RFC 2974). This address and port will be used by the transmitter to 'advert' its audio streaming broadcasting. In the case that these parameters are modified by the user, they should be configured to the same values at both ends, the sender and the receiver.

When making a call with SAP as communication protocol, only Multicast Tx and Multicast Rx call types will be available. To launch an audio streaming over IP with SAP the user shall select Multicast Tx as call type from the dialing window, and type in the multicast IP address to send the audio to.
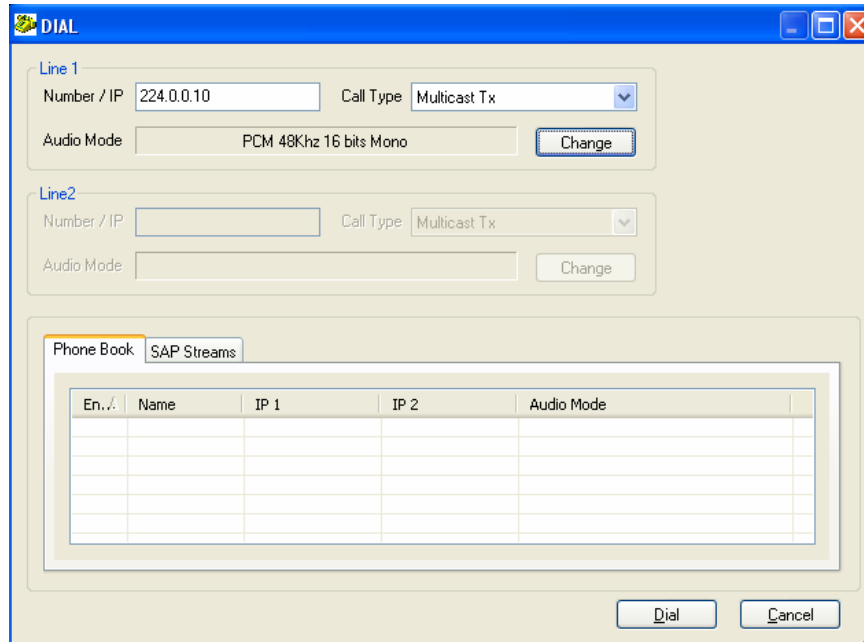
To receive this audio streaming, the user shall select the Multicast Rx call type, and will introduce the same Multicast IP address as the transmitter.
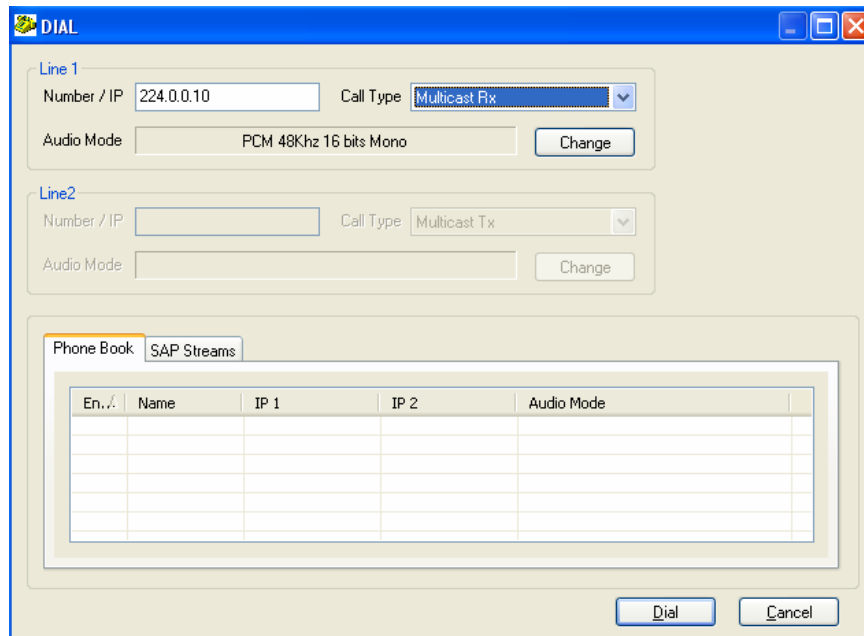
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As many receivers as required can join this multicast address and receive the audio, without any increment in the required bandwidth, given that this communication is based on IP multicast technology.

IV.6.10 Profiles for audio streaming over IP Networks

The <Profiles> menu provides a simple and quick shortcut to predefined codec configurations for audio streaming over IP Networks. Given is that the user might deal with different IP network performances.

Several predefined network profiles are available:

- <Safe>: Suitable for low reliable networks such as Internet.

- <Ultra Low delay>: Suitable for high reliable networks such as LAN.

- <Custom>: User defined streaming parameters. Please refer to sections IV.6.11 and IV.6.12 for details.

IV.6.11 Tx

This submenu defines some subtle parameters for fine tuning of the IP protocols.

<Audio>: This menu allows the user to configure the audio transmission parameter (TBP: Time between packets) and to know in advance, the required bandwidth and delay of the connection.

NOTE: Bear in mind that this is a transmission parameter, so it won't be available for those units like ProntoNet IP Decoder, which does not have encoding capabilities.

Access this by pressing the "SYS" button on the web interface select STREAMING – TX. You will be presented with a screen similar to below:
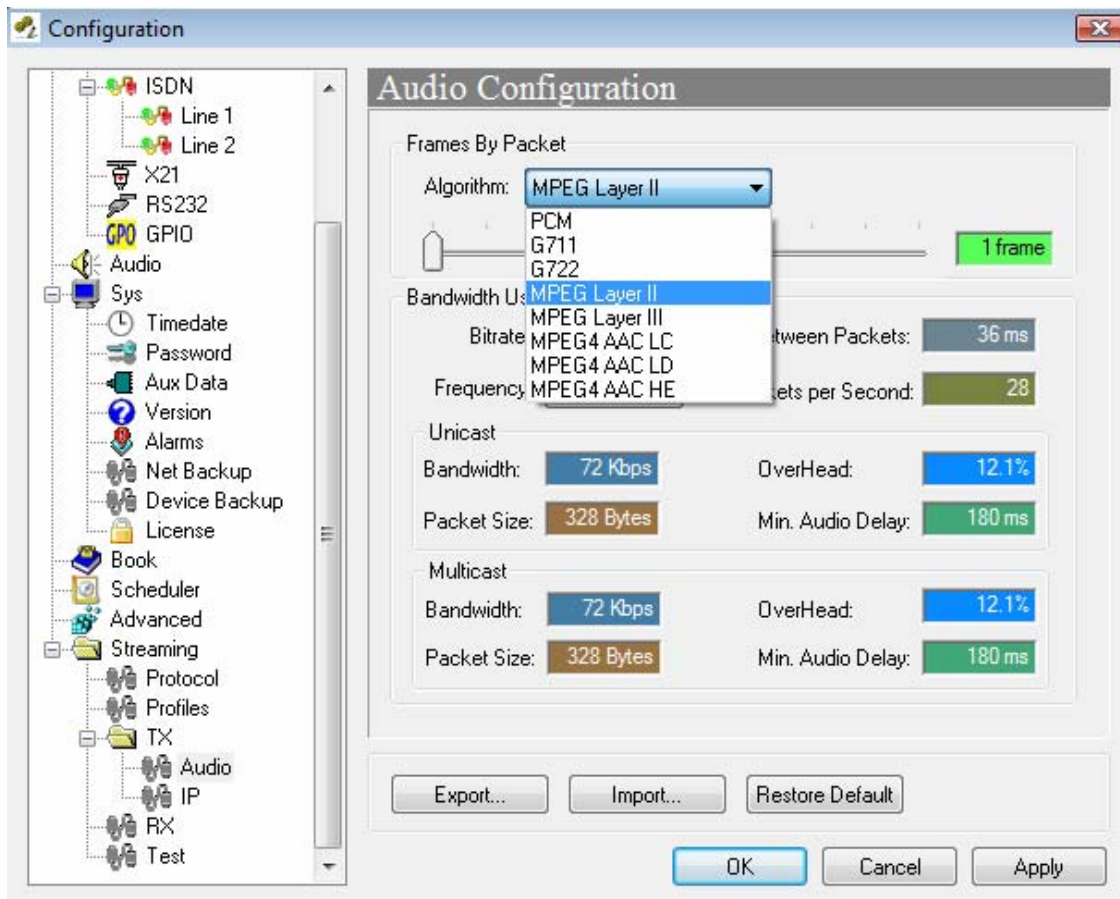
From this window, the user can know beforehand, the bandwidth and delay of the connection.

So, the 'Time Between Packets' parameter, is directly related with the packet size and therefore with the occupied bandwidth (overhead) and delay. Therefore, the appropriate value for this parameter is a trade off between delay and bandwidth: **The larger the block size (TBP), the higher the delay, but the smaller the required bandwidth and overhead (more efficient use of IP packets), and vice versa.**

If there is no delay requirements in the connection, it is always advisable to select the highest value, since bandwidth will be smaller. Also, the more frames over the network (with small TBP values, frames decrease in size and increase in number), the more likely it is that the jitter will grow.

NOTE: Keep in mind that there are compression algorithms that introduce delay itself. When that delay is higher than the TBP minimum value, the value is automatically adapted to the algorithm to the values shown on the left, and user is not allowed to change it, given that it does not make sense to increase it and it is not possible to be decreased due to the codification process.

> Remind that the <Tx> menu only defines the operational parameters for several protocols. For taking in force the actual audio streaming protocol, the user must apply the selection at the <NET> and the <Protocol> according section III.1.

<IP> menu: This menu allows the user to configure some IP header fields supporting QoS.

- Streaming Packets Quality of Service: This option allows the user to configure the value in the ToS field in the IP header.
- Streaming packets Time To Live: Configure the Time To Live parameter.
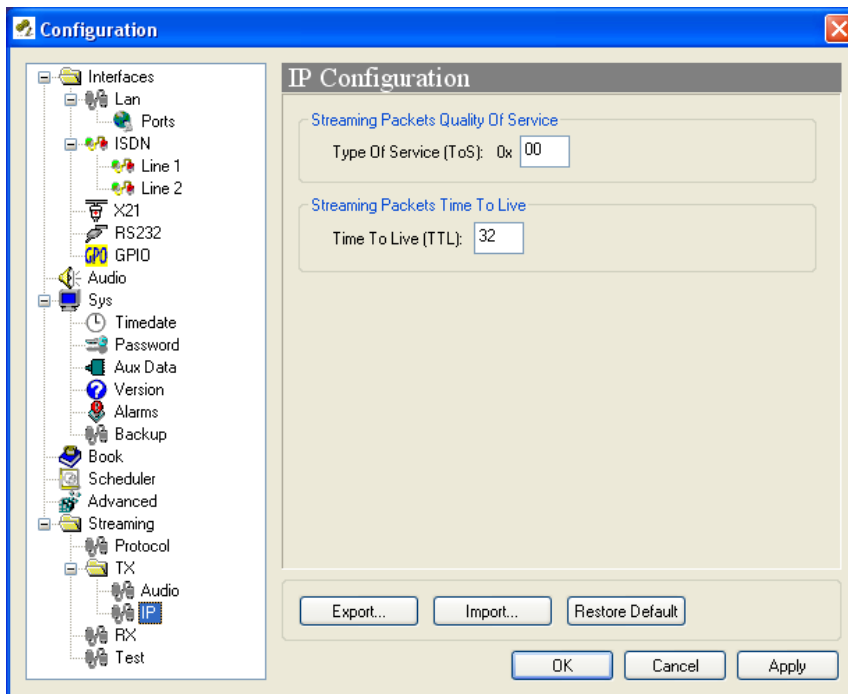
IV.6.12 Rx

A decisive factor in real time audio streaming is the 'jitter', or delay variation. To deal with the jitter in the connection, Prodys IP Family codec provides a tool which allows the user to modify the size of the reception buffer, and so, to compensate the jitter. The maximum value for this buffer is 10s.

Once the user knows the "jitter value" of the IP connection, the jitter buffer should be set:

■ "Jitter Value" is the difference between the maximum and minimum delay. If the jitter value is 0 it means that the delay is constant and it is not necessary to adjust the jitter correction buffer in the reception side. If the delay is not constant, it is necessary to adjust the buffer size in order to guaranty not audio drops, even when the delay reaches the maximum value.
■ The unit of this buffer is miliseconds, and the range is 0-10000. The delay grows as many as the number of miliseconds the buffer has been set to.
■ The "jitter value" can me measured by the streaming tool commented before.
■ **Rule: Max User Line Delay + TBP >= Jitter value.**
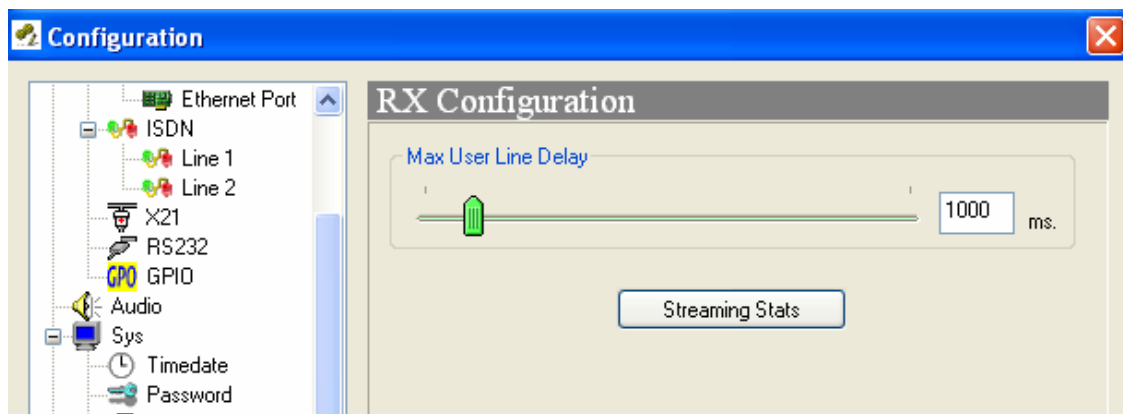
This value can be changed during the audio connection.

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IV.6.13 Test

This tool allows the user to check, in real time the upload/download bandwidth, delay, jitter and packet loss in an IP link between two Prodys IP units. This information will be used to adjust the streaming parameters in order to achieve the best quality in the audio streaming connection.

Before making an IP connection, be it unicast or multicast, the user should first check the quality of the IP connection. Prodys IP Codecs comes fitted with a 'Test Tool' embedded in the web page. No additional software is required. With this tool, the user can measure the most important parameters of an IP connection: Download and upload bandwidth, delay, jitter, lost packets and packet disorder:
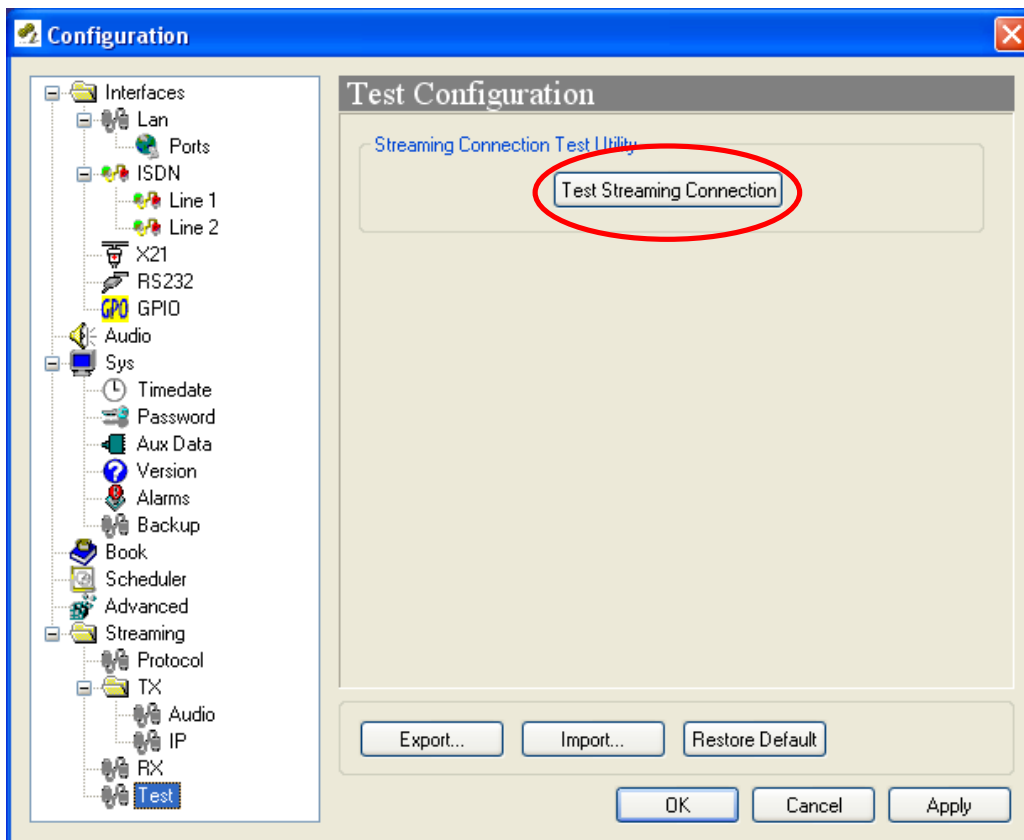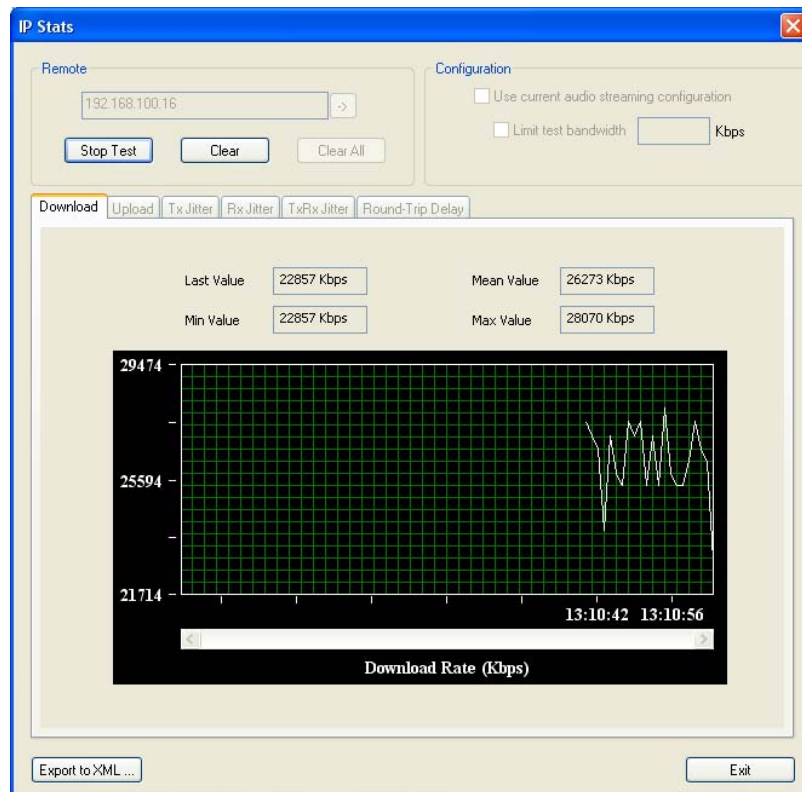
This tool is only available when lines are disconnected.

How to run the tests**:**

1.- First of all the user must enter the IP address of the remote Prodys IP Family codec. This dialog stores the last used IP addresses.
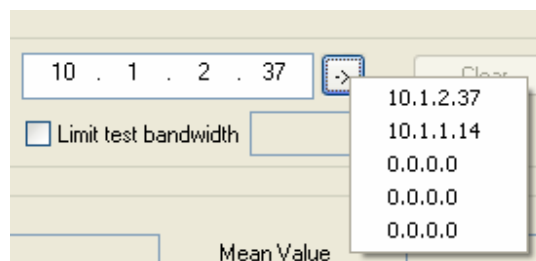
2.- Select the test you wish to perform.
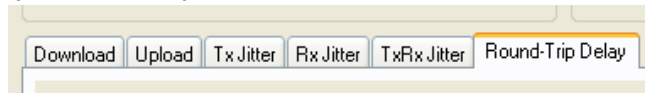
3.- To run the test just press the test button:
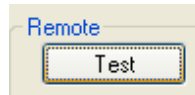
4.- The values will be shown in a graph, as well as the max, min, last, and mean(avg) values.

4.- Press "STOP" to stop the test or "CLEAR" to reset the readings.

5.- The user can limit the bandwidth used by the test tool:

By default this test will attempt to send as much IP traffic as possible. If you are using this test on a "live" network then it is desirable to limit the test bandwidth to that which will actually be used. It is also possible to simulate the same packet rate and bandwidth as that configured for the audio connection[13].
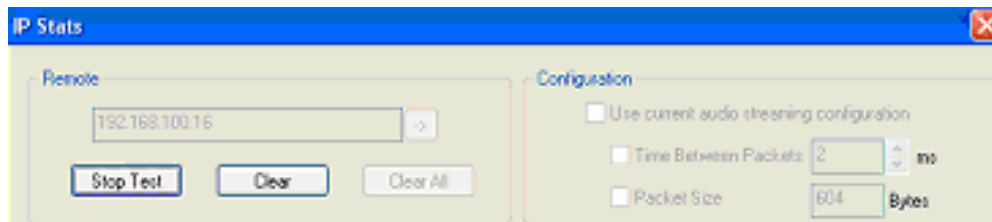
An accurate figure of total bandwidth used (audio data + overhead) can be obtained from the "Audio Configuration" screen.

Once the test is finished, it is possible to export the results to an XML file.

_____
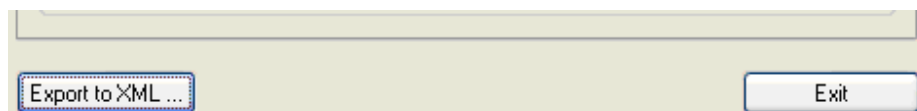
[13] Available from version 5.0.0 onwards.

IV.6.14 Real Time Analyzer

It is possible to monitor the streaming operation in real time by ticking a checkbox.
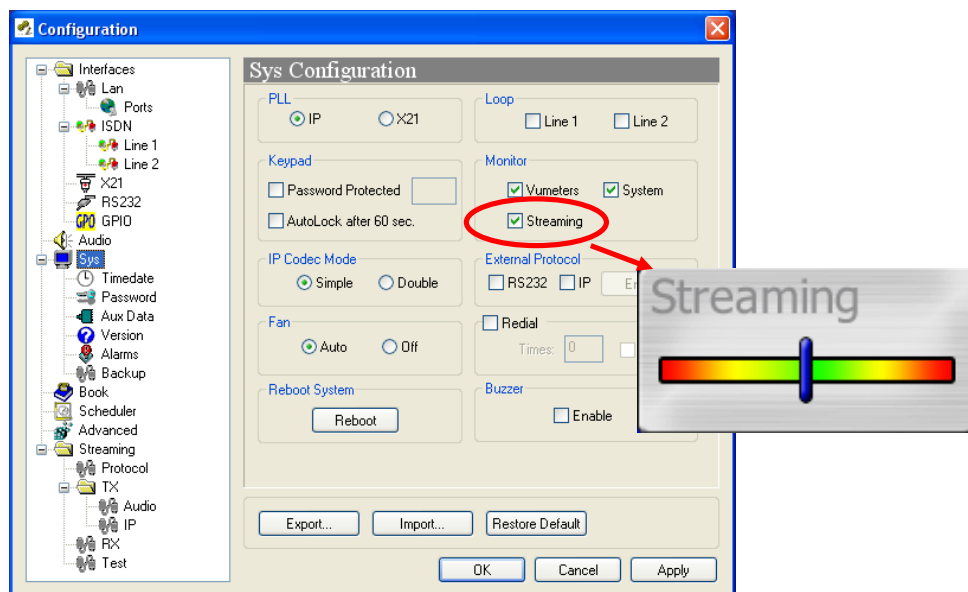
That indication represents the instant decoder buffer occupation.

When indication is green or even yellow means the streaming is working fine. Depending on the algorithm, the variation will be quicker or slower but always should be going up and down from yellow to the center (green).

If the bar moves towards the red area, it means that it is getting closer to buffer overflow (red right) or buffer underflow (red left). Of course, if you have buffer overflow or underflow you will have a cut in the audio. If the indication goes to red for an instant, usually doesn't affect the streaming, but of course, if you have a constant red indication you are having problems with the streaming.

Buffer underflow might be caused by lost packets or because the clock of the AD converter at the transmitter side which runs slower than the DA converter at the receiver side.

Buffer overflow might be caused by the opposite scheme: The clock at the transmitter side is running quicker than the clock at the receiver side.

Prodys PPL adjustment mechanism is aimed at dealing with this clock jitter and thus to compensate for this variation and guarantee proper audio over IP.

IV.7 Preset Configurations

Beside our familiar option of storing and loading any amount of codec configuration on the PCs, it is also possible to store and load any type of codec configuration inside the codec memory.

This function is very useful for non expert codec users, portable application, quick reconfiguration or if the notebook is not available at the event. The <Presets> can be loaded from the front panel and from the web control interface.
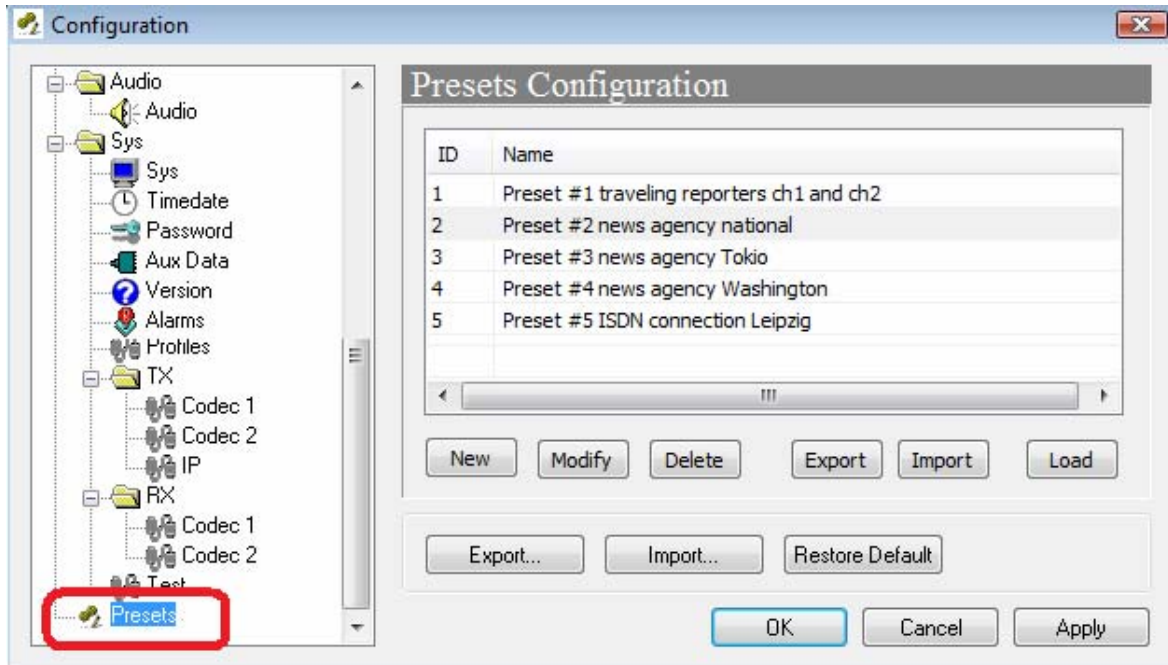
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The configuration option is enabled by the left hand matrix according to the picture below. For any parameter not explicitly defined at this menu it is applicable the current value in use[14]. Essential parameters for the application should be explicitly defined at this menu.

---
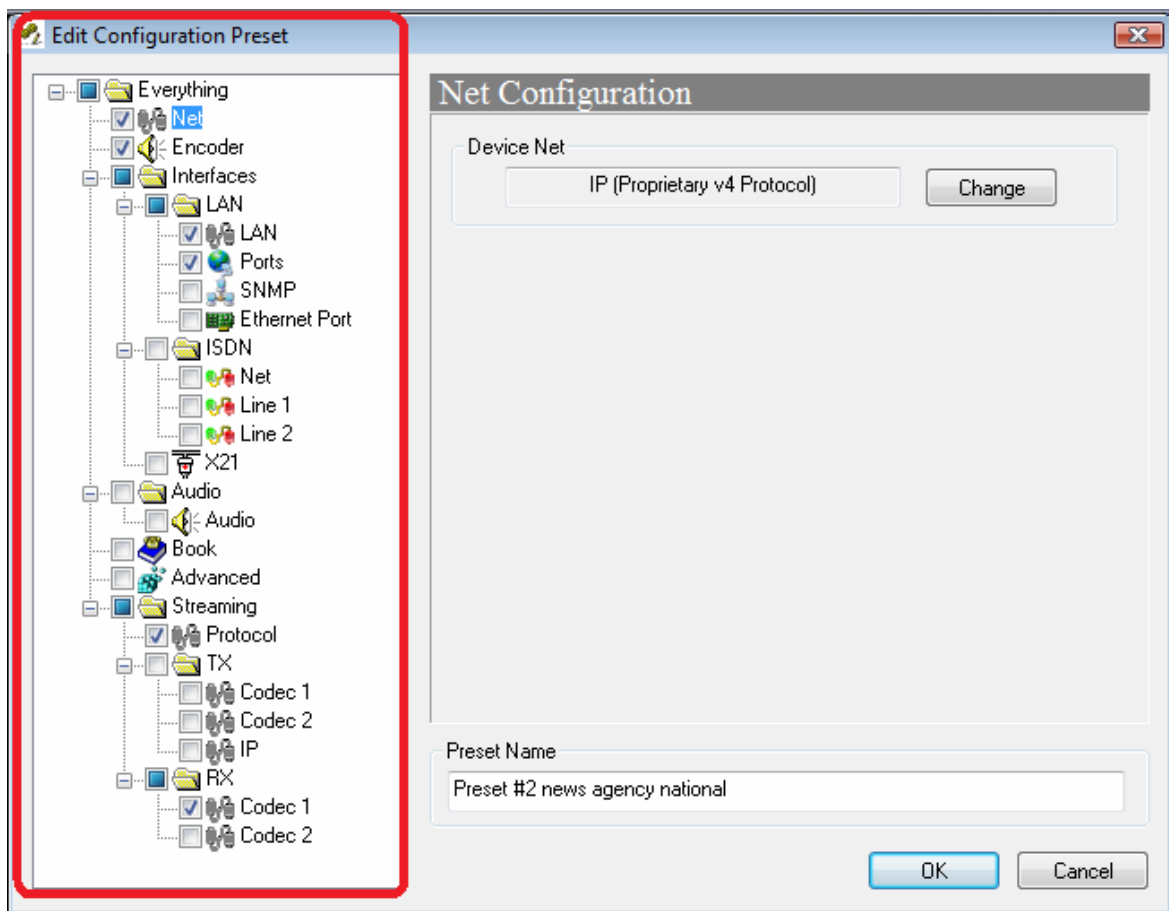
[14] In other words the definition of a new Preset is actually the "delta" definition of the current configuration in place.
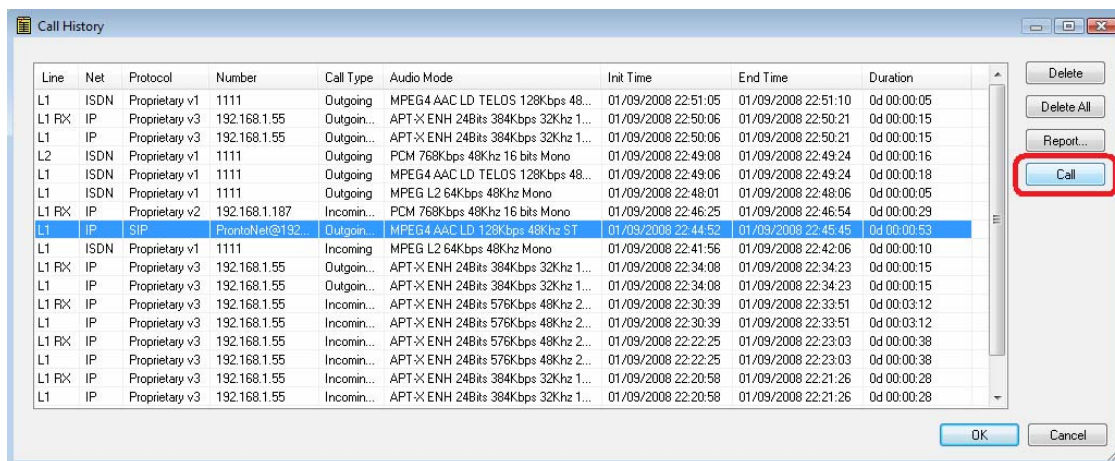
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IV.8 Call Log

An history report has been included to record the input and output calls according to the following information: telephone/IP, audio modes, date and start/end time, length of each call, etc…

A report of the calls in html can be created allowing preview and print.

A dial short cut is possible from the Call Log window. The user might restart the communication with the same NET parameter, destination ID and audio compression mode as any of the former communications logged.

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IV.9 Alarms

The alarm window allows the user to select the alarms that the unit will check. The unit will notify the occurrence of each of the selected alarms. It is possible to configure the unit to send SNMP traps or emails to notify alarms information. See chapter IV.2.5 Alarm Paging.

IV.9.1 Selecting Alarms

The unit allows many different alarms. The selection menu is opened by clicking over the Alarm area.

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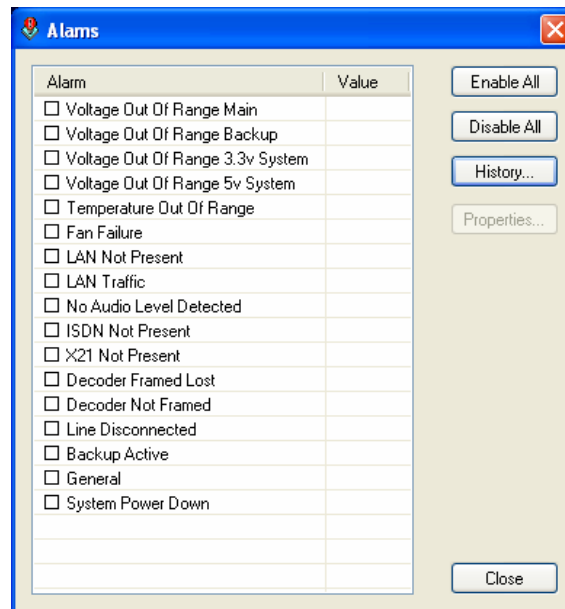
- **Voltage Out of Range 12 v Main:** The voltage from the main power supply (AC/DC) is out of range.
- **Voltage Out of Range 12 v Backup:** The voltage from the backup power supply is out of range.
- **Voltage Out of Range 3.3 v System:** The voltage from the 3.3 DC/DC converter is out of range.
- **Voltage Out of Range 5 v System:** The voltage from the 5 DC/DC converter is out of range.
- **Temperature Out of Range:** The temperature is out of range[15].
- **Fan Failure:** The fun is activated if the internal temperature is over 34º. The Fan Failure alarm will be activated if the fun doesn't work when the temperature is over this threshold.
- **LAN NOT Present:** Not physical level detected in the LAN port.
- **LAN Traffic:** The LAN traffic is higher than the 90% of the capacity of the network.
- **No Audio level detected**: No audio is present on the input/output (as configured). A threshold and a time period can be defined to activate the alarm.
- **ISDN NOT Present**: Not physical level detected in the ISDN port.
- **X21 Line Not Present**: Not physical level detected in the X21 port.

---

[15] The temperature is above 50ºC.

- **Decoder 1/2 Framed Lost**: The Line is connected but the decoder is not framed anymore. The alarm will be activated also if the unit is turned off. In this case, the alarm will indicate the date and time when the decoder lost the synchronization, that is, when the unit was turned off. The alarm will be activated until the unit is framed once the line is connected again. If the connection is not possible, the alarm will be marked as finished.
- **Decoder 1/2 Not Framed**: The Line is connected but the decoder is not framed. This new alarm doesn't require that the decoder was framed previously. For this reason, it is necessary to specify a window time to decide when the alarm will be activated once that the NOT FRAMED condition has been detected.
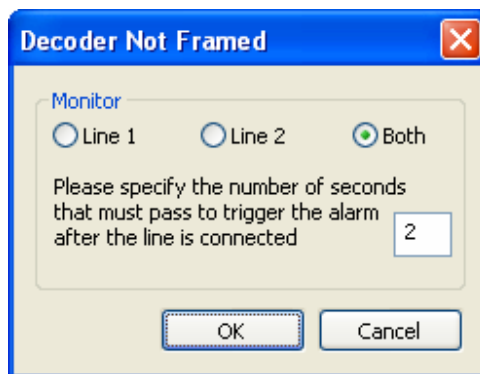
Otherwise, the alarm would come up as soon as the connection is established, given that the Decoder needs some time to detect the incoming compression mode.

- **Line Disconnected**: Line 1/2 has been disconnected unexpectedly, manually disconnected or both.
- **Backup Active**: The ISDN backup line is working.
- **General**: In internal hardware error was detected by the software.
- **System Power Down**: This new alarm allows the user to know if the unit was turn off and the time that it remained in that condition.

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IV.9.2 Monitoring Alarms

When one of the selected alarms is detected, the alarms window is highlighted.
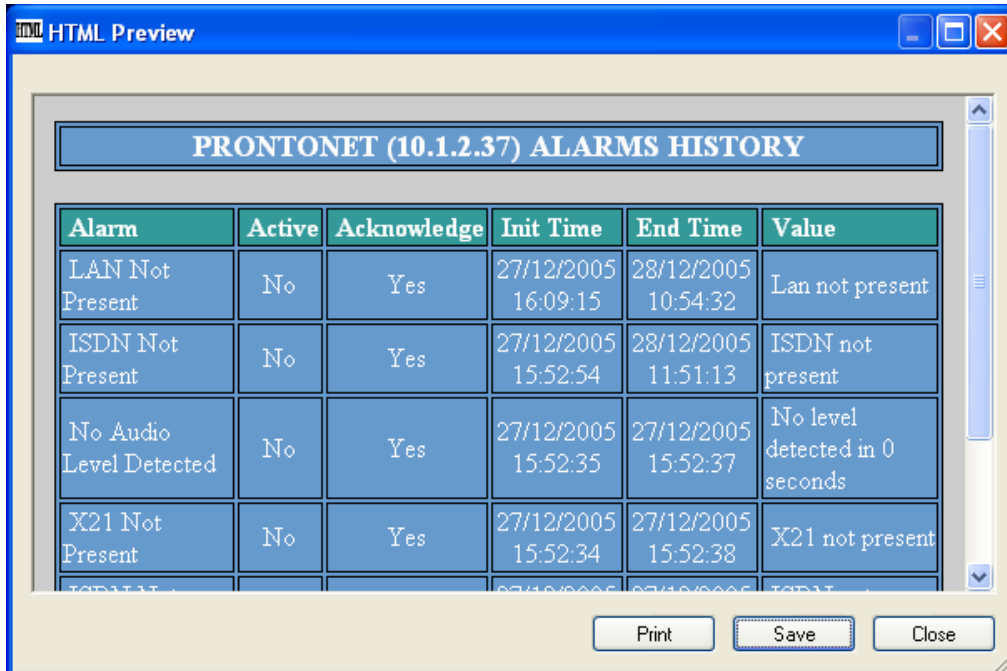
IV.9.2.1. Alarms History

Alarms are saved on the non volatile memory of the Prodys IP Family codec. Therefore, it is possible to know what happened in the past even if the unit was disconnected of the network. The alarms history can be showed by clicking over the lamp or over "History" option in the alarm selection dialogue.
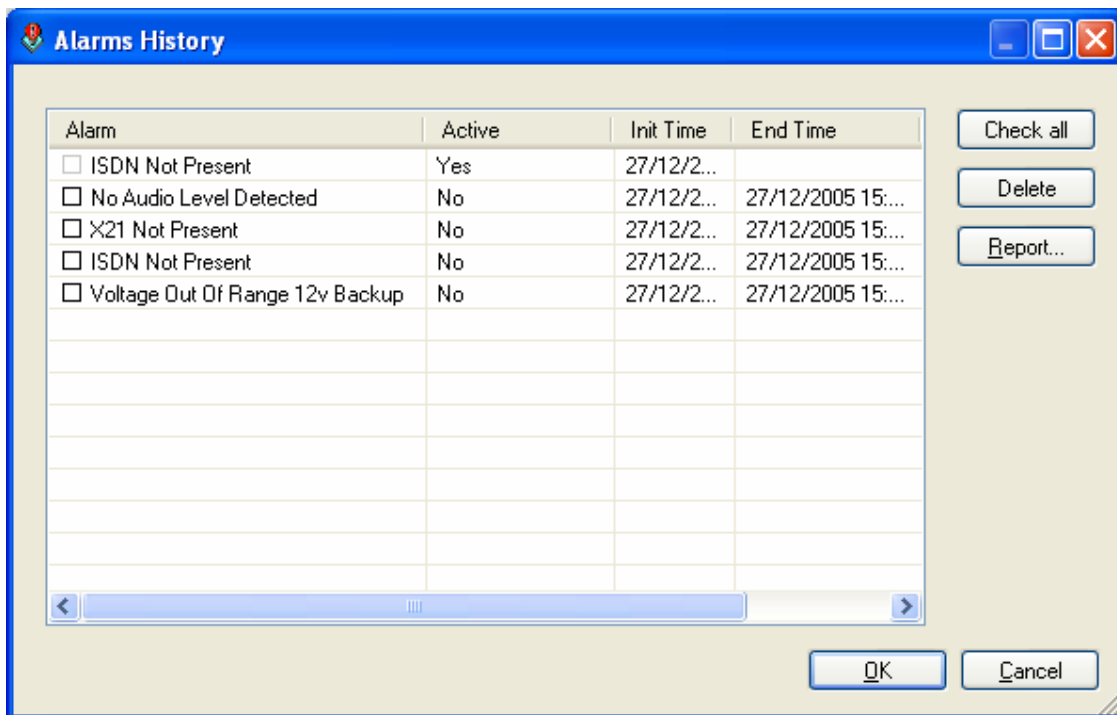
An historical report of alarms in html allowing preview and print is available.

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HOW DOES PRODYS IP FAMILY OF CODECS WORK?

Given the different nature of the three communication standards that the Prodys IP Family codec supports, the device requires network specific settings for matching the transmission media. What is more, in situations, Prodys IP Family codec is capable of dealing with two independent communications at the same time. This chapter is a practical guide to help in understanding just how the Prodys IP Family codec unit works under different configurations, in particular the non conventional ones.

V.1 Selecting the network interface

Firstly, when we refer to the 'network interface' we are always referring to the communications port that is used for the transmission of 'audio data'. The NET option from the top menu is applicable for this selection. There are three settings available for the Prodys IP Family codec:

- Prodys IP Family codec operating as "IP CODEC": The codec will be configured to establish communications via the Ethernet port using the IP protocol.

- Prodys IP Family codec operating as "ISDN CODEC": The codec will be configured to establish communications via a basic access ISDN connection. The Prodys IP Family codec has a multi-protocol ISDN terminal socket ready for connection to a basic rate 2B+D interface.

- Prodys IP Family codec operating as "X21 CODEC": The codec can be configured to establish communications via dedicated digital lines at 64, 128, 192, 256, 384 and 576 Kbps.

- Prodys IP Family codec operating as "PSTN CODEC"[16]: The codec will be configured to establish communications via any PSTN network.

> **The Ethernet interface is always available for remote control of the unit, even if it is configured to operate as an ISDN, PSTN or X21 codec.**

You should first select the network type - i.e. the network interface - to be used for audio data transmission and reception. The menu will only show the configuration options relevant to the network type selected. Of course, there are a number of general parameters that will be the same across all network types and so will not be affected when the network type is changed.

V.2 Configuration parameters that are dependant on the network type selected

There are two fundamental reasons for which the user must change the Prodys IP Family codec configuration parameters when a new network interface is selected:

- The available bandwidth over the selected network (IP, ISDN, PSTN or X21). This places limitations on the bit-rate that can be selected and the amount of compression. As an example: over ISDN the bit-rates for any algorithm must fit into 64 or 128Kbps.

- The number of available channels: Number of bidirectional communications which can be achieved. Note that over ISDN and IP, up to two independent mono communications can be achieved.

The following table shows what the differences between the various communications networks regarding Prodys IP Family codec operation:

---

[16] PSTN is only available on NomadaIP.

NET	ALGORITHM	BIT RATES (Kbps)	Nº CHANNELS
IP	PCM	16 bits @ 48 KHz.	2 Mono[17] or 1 Stereo
	G711	64	
	G722	64	
	MPEG1,2 LII	64,128,192,256,384	
	MPEG1,2 LIII	64,128,192,256	
	MPEG AAC LC.LD	64,128,192,256,384	
	MPEG 4 AAC HE	24,32,48,56,64,128	
	aptX™	64,128,192,256,384,576	
ISDN	G711	64	2 Mono or 1 Stereo
	G722	64	
	MPEG1,2 LII	64,128	
	MPEG1,2 LIII	64,128	
	MPEG 2,4 AAC LC	64,128	
	MPEG 4 AAC LD,HE	64,128	
	aptX™	64	
X21	G722	64	1 Mono or 1 Stereo
	MPEG1,2 LII	64,128,192,256,384	
	MPEG1,2 LIII	64,128,192,256	
	MPEG AAC LC,LD	64,128,192,256,384	
	MPEG 4 AAC HE	24,32,48,56,64,128	
	aptX™	64,128,192,256,384,576	

V.3 Prodys IP Family codec working as a "DUAL CODEC" over ISDN

When we select ISDN as the network interface, it is possible to establish two totally independent MONO connections for each 64 Kbps B Channel. This means that for each channel or line of communication (Line 1 or Line 2 as seen by the Prodys IP Family codec) it is possible to send an audio signal encoded with any of the available codec algorithms. The menu offers the option to configure Encoder 1 and Encoder 2 separately. However, the user must remember that in this mode it is only possible to work with MONO signals on each channel, given that Prodys IP Family codec has only one stereo audio input and one stereo

[17] Over IP, the second MONO communication is fixed to G722.

audio output. To avoid incorrect configuration there are therefore some restrictions while working with the Prodys IP Family codec as a "DUAL CODEC":

- If Encoder 1 is configured in any mode other than DUAL or STEREO/JSTEREO the Encoder 2 option is disabled and the Prodys IP Family codec will not work as a "DUAL CODEC".

The Encoder 2 is not available also when NET = ISDN and bit rate = 128 Kbps.

- Encoder 2 can only be configured to work in MONO.

Further section shows how the Prodys IP Family codec operates as an ISDN CODEC in detail. Restrictions are listed for the Encoder 1 in DUAL or STEREO mode, as well as for receiving calls from other units that are working in any of these modes.

V.4 Prodys IP Family codec working as a "Double CODEC" over IP

A Prodys IP Family codec is a stereo signal capable codec. However the codec provides a mean to split one stereo codec into two independent mono codecs.

This option is enabled at the NET menu.

- <Simple Codec>: This is the setting by default. Stereo audio signals and any compression algorithm might be enabled.

- <Double Codec>: This setting enables two independent mono codecs. Following restrictions must be remarked:

    d) Only mono signals can be encoded and decoded.
    e) PCM and AAC HE streaming is not supported.
    f) Apt-X and enhanced Apt-X are supported only at 16kHz sampling frequency
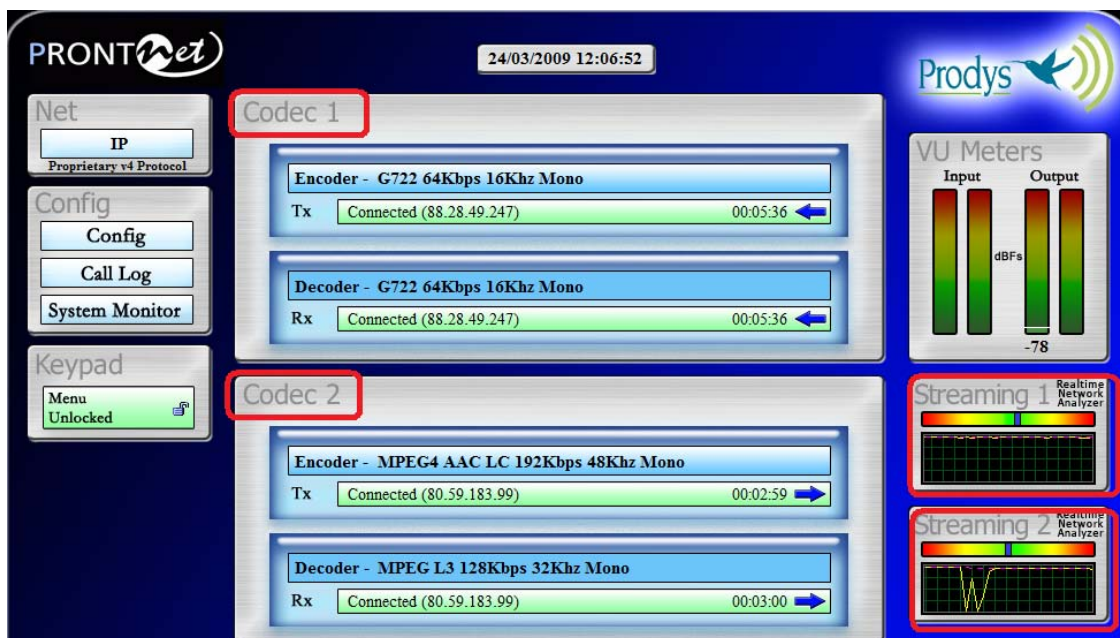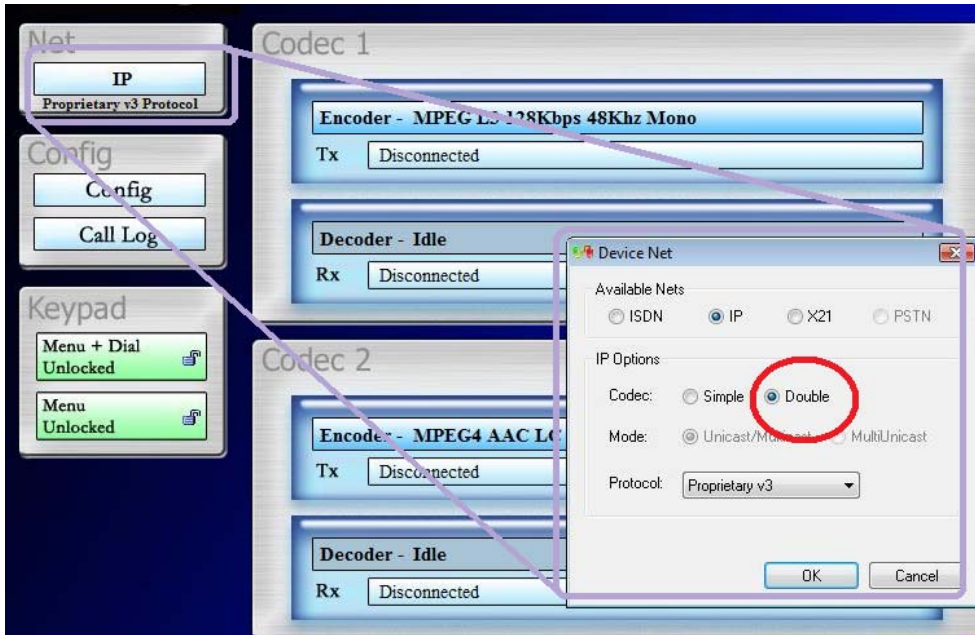
Codec 1 and the Codec 2 are not required to match their <TX configuration> (Time Between Packets definition) or <RX configuration) (RXbuffer definition).

Therefore two independent mono communications can be provided. Accordingly the "Real Time Analyzer" supports also double codec operation.
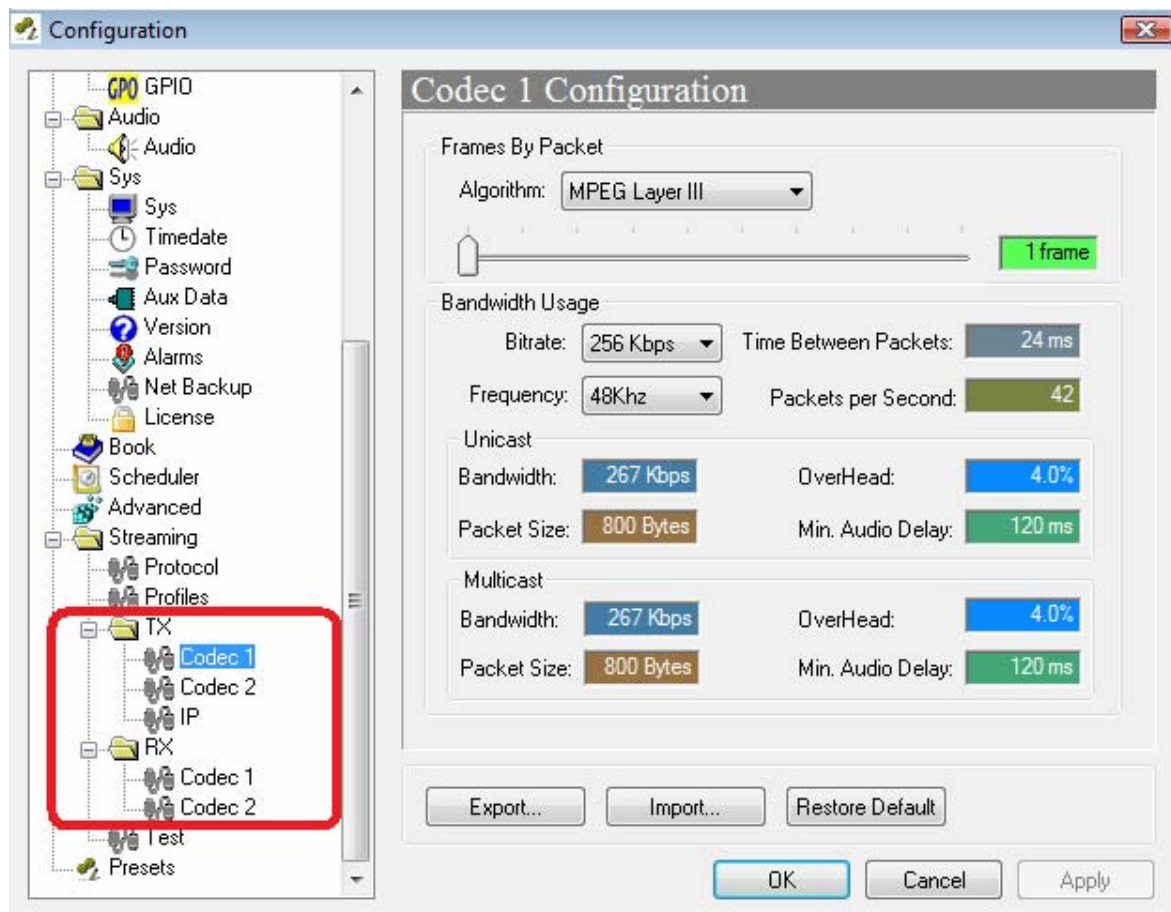
The <Dial> menu allows selecting the <Target codec>: 1 or 2. This means that if the far-end codec is also in double mode, the current communication might address the Codec 2 (the second mono codec) directly.
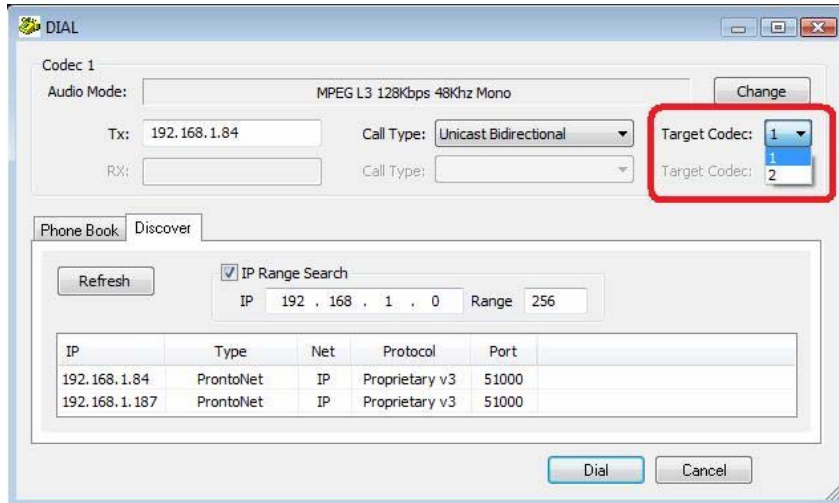
V.5 About how the Decoder works and automatic searching

The Prodys IP Family codec Decoder system does not require to be configured by the user; detection of the incoming audio stream parameters is automatic (given the NET and the Protocol settings matches). However, it is helpful to highlight some of the logic behind of the automatic operation. The Prodys IP

Family codec is able to synchronise automatically to the following algorithms under the given conditions:

Algorithm	IP Codec (NET = IP)	ISDN Codec (NET = ISDN)	X21 Codec (NET = X21)
G711	YES	YES	NO
		The system detects incoming calls such as voice calls and configures the Encoder and Decoder to G711.	Algorithm not available.
G722	YES	YES	YES
		Needs G722 encoded audio since it is synchronised by means of statistical framework. **Automatic detection is unsuccessful if the Encoder is configured to aptX , AAC HE or J52.**	Needs G722 encoded audio since it is synchronised by means of statistical framework. **Automatic detection is unsuccessful if the Encoder is configured to aptX, AAC HE or J52.**
MPEG1,2 LII MPEG1,2 LIII MPEG2,4 AAC LC & LD	YES	YES	YES
		Detects all modes excepts if the Encoder is configured to aptX. **Automatic detection is unsuccessful if the Encoder is configured to aptX, AAC HE or J52.**	Detects all modes excepts if the Encoder is configured to aptX. **Automatic detection is unsuccessful if the Encoder is configured to aptX, AAC HE or J52.**
MPEG AAC HE & aptX^{TM}	YES	NO	NO
	AptX not with SIP/SAP as IP protocols	It is necessary that the Encoder is also set for the proper algorithm	It is necessary that the Encoder is also set for the proper algorithm
PCM Linear Audio	YES	NO	NO
		Not Available.	Not Available.

Taking into account previous table the user must also bear in mind the following:

1. Over X21 or ISDN, it is only possible to set encoding and decoding algorithms independently for those algorithms that supports automatic synchronising. For example, it is possible to transmit on G722 and receive on MPEG Layer II.

> **Although G711 is detected automatically in ISDN mode, the Encoder is also configured automatically because it is not possible to use it in any other combination.**

2. If we want to receive an algorithm that does not support automatic synchronisation it is necessary to configure the Encoder to the same algorithm. In the case of apt-X or Mpeg4 AAC HE, the compression parameters on the Encoder should match those on the Decoder.

3. If we want the Encoder to switch to the algorithm detected by the Decoder, the Encoder should simply be set to AUTO mode.

> **AUTO does not work with those algorithms that do not support automatic synchronization.**

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V.6 The Prodys codec operating over IP (Prodys Proprietary Protocols)

The operation of the Prodys IP Family codec over IP offers three operational modes: UNICAST, MULTICAST and MULTIUNICAST[18].

V.6.1 UNICAST communications

The term UNICAST is used in the networking world to refer to the connection to a single destination. Unidirectional or bidirectional audio streaming is possible with this mode.

UNICAST is the default mode and is available for any IP protocol, but for SAP.

Given is an example for Prodys Proprietary Protocol flow-control:

Prodys IP Family codec ----- Calling to 192.168.0.2 -----▶ Prodys IP Family codec
Audio P-RTP
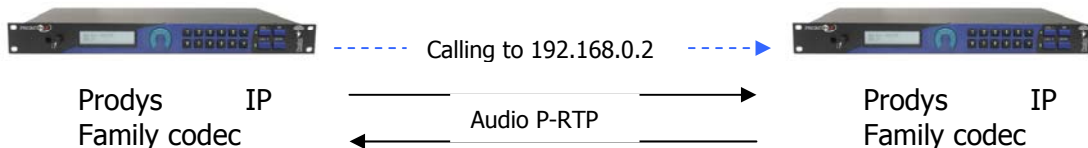
V.6.2 MULTICAST communication

The MULTICAST mode stands for point-to-multi-point communications. This is one transmitter streams audio towards one or several receivers
In order to start a MULTICAST communication, the connection must be requested from both ends (transmitter and receivers). Any participant, the transmitter of the audio stream and all the receivers of the stream must join to the multicast group. It does not matter which participants joins first. Of course the actual audio stream will be not available for decoding before the transmitters joins to the multicast group; up to the audio availability any receiver will just keep the audio outputs muted.

---

[18] MULTIUNICAST is available from version 5.2.1 onwards.

It is noted that a multicast group is defined by the IP address, which must be any address included in the 224.0.0.0 /4 range.
As soon the users dials a destination belonging to this address range, the codec changes automatically to the multicast mode.

The request to join to a multicast group is performed just entering the multicast IP address into the dial menu of Encoder or Decoder (according whether the audio stream must be streamed out from or decoded on this codec).

The multicast operation can be shown in the following diagram:
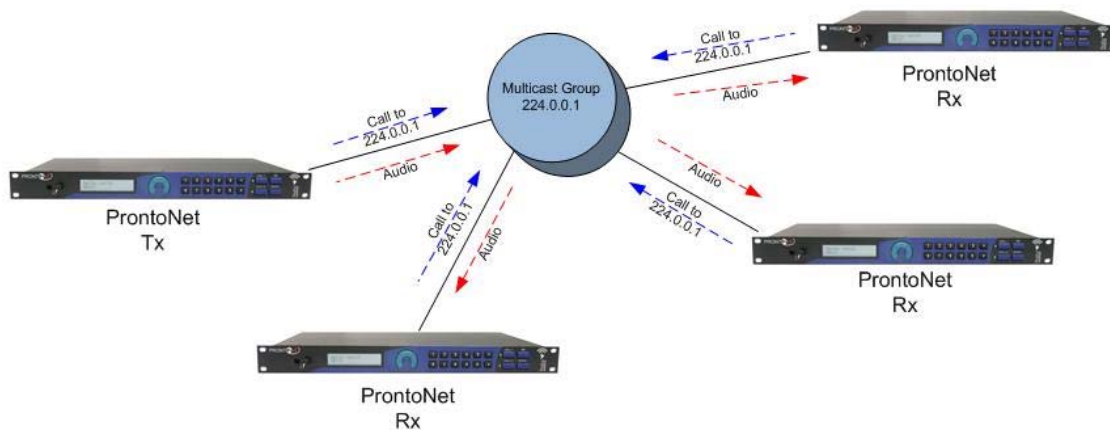
■ **Multicast considerations:**

▪ Internet Protocol (IP) multicast is a bandwidth-conserving technology that reduces traffic by simultaneously delivering a single stream of information to thousands of corporate recipients and homes.

▪ Multicast is based on the concept of a group. An arbitrary group of receivers expresses an interest in receiving a particular data stream. This group does not have any physical or geographical boundary—the hosts can be located anywhere. Hosts that are interested in receiving data flowing to a particular group must join the group using IGMP. All this is done automatically by Prodys IP Family codec when establishing a connection.

▪ Multicast traffic is rejected by the Internet infrastructure, since most IP servers on the Internet do not currently support the

multicasting part of the protocol, except when using VPNs, because VPN's encapsulates IP packets as unicast frames, so routers simply see an ordinary packet.

- All IP multicast group addresses will fall in the range of 224.0.0.0 to 239.255.255.255, but some of them are reserved, that's why the range of addresses from 224.0.1.0 through 238.255.255.255 are called globally scoped addresses.
- Several transmitters to the same multicast group must be avoided for preventing of decoding artifacts.
- For transmitting MULTICAST audio the Prodys proprietary protocol "Prodys eXtended Real Time Protocol (PX-RTP) will be used.
- The Prodys IP Family codec transmitter cannot be in AUTO mode for the encoder.
- To guarantee a constant delay all the units must synchronise their clocks. Each receiver will activate a clock-sync algorithm that adjusts its PLL (Phase-Lock Loop).

V.6.3 MULTI-UNICAST

It happens that the MULTICAST mode is not applicable for streaming over the Internet (section V.6.2). Then MULTI-UNICAST overcomes the lack of Multicast support on some networks.

This option allows the user to establish point to multi-point connections by means of many unicast connections. One Prodys IP Family codec can send the same encoded audio stream up to 10 different destinations.

It has to be remarked that this mode demands multiple streaming bandwidth over the network and the codec Ethernet interface, since every individual bandwidth accounts for the total.

This special feature is enabled thru the <NET> menu as the depicted in the screen shot. Any Prodys Proprietary Protocol is suitable for this purpose.
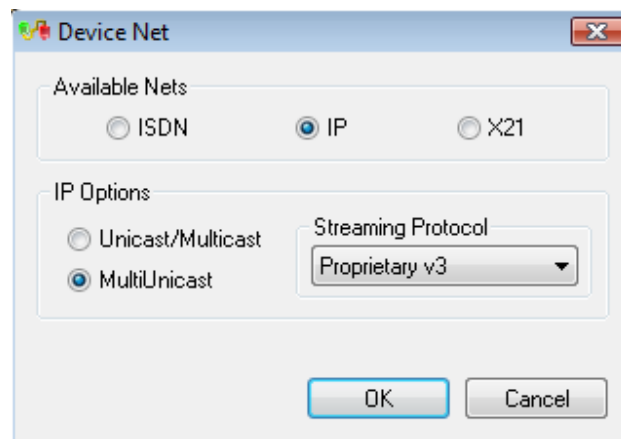
The main window will show all the available connections:

ProdysIP Manual
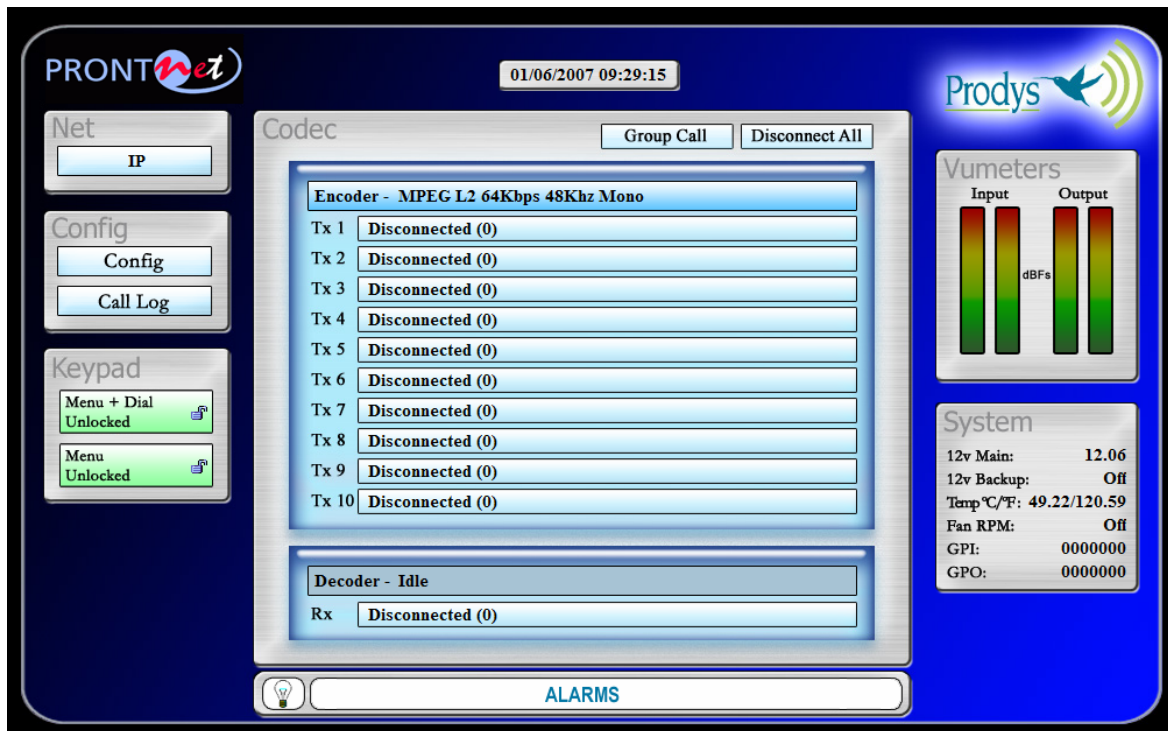
Depending on the compression mode, up to 10 outgoing connections can be available:

- MPEG L2, L3 and AAC: Up to 10 Tx + 1 Rx.
- PCM, G711, G722, APTX: Up to 3 Tx + 1 Rx.

The picture displays an independent control box for each connection. Each of the connection box will show the actual line status. The procedure to establish a call is the same as that for Unicast connections.

Some remote codecs might drop its communication to the local codec while other remote codecs keep the current link. In addition, it is possible to make or hang up several connections at the same time by using the 'Group Call' and 'Disconnect All' buttons respectively.

As for Unicast connections, the Tx and Rx parts of the connection are divided so that it is possible to receive from one end at the same time the unit is sending audio on one or more Tx connections.

> This operation mode is incompatible with either <Double IP codec> or <Net Backup> enabled.

V.6.4 More about Prodys Proprietary Protocols

Prodys has developed this proprietary set of protocols to carry out IP streaming connections, due to the lack of a standard in this regard:

- Prodys Real Time Control Protocol (P-RTCP): This is a protocol based on TCP that allows for the establishment and termination of a connection as well as for the negotiation of the codec mode (automatic audio synchronisation in all modes).
- Prodys Real Time Protocol (P-RTP): This is a protocol based on UDP used for the transmission of audio.
- Prodys eXtended Real Time Protocol (PX-RTP). This is a protocol based on UDP used for the transmission of multicast audio.
- Prodys Upgrading/Identifying Protocol. This protocol is based on UDP and used to identify/upgrade the units.
- Prodys External Protocol (P-XP). This protocol is based on TCP and can be used for controlling the units from an application other than the web page or ProdysControl.
- Prodys Auxiliary Data Protocol (P-AUXP). This protocol is based on UDP and used for transmitting/receiving auxiliary data.
- Prodys U-bit Protocol (P-UbP). This protocol is based on UDP and used for transmitting/receiving the User Bit from the AES/EBU frame.

This set of protocols formed the Prodys Proprietary Protocol V1.

V.6.5 More about Prodys Proprietary Protocol v2

This set of protocols is based on the Prodys v1, and it uses the same ports. This new set of protocols (version 2) is not compatible backward, and it introduces some new fields in some of the protocols previously listed so that it is possible to measure lost and disordered packets in real time and during the audio connection. Proprietary protocol version 1 will allow the user to get information about jitter and buffer usage, but will not allow the user to obtain information about lost and disordered packets.

This information will be saved separately for each connection in RAM memory. Up to 24 hours of data from different calls can be stored.

Once the connection is established, the user can access the 'real time network analyzer' by clicking on the 'buffer occupation graph', in order to get information related to:
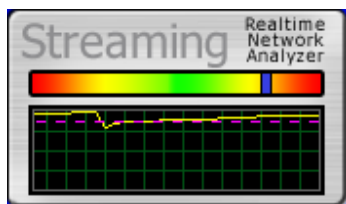
-Rx occupation: With average, maximum and minimum usage. Very low percentage of buffer occupation will cause audio interruptions.
-Jitter: High jitter values will match with low buffer occupation.
-Lost and disordered packets.
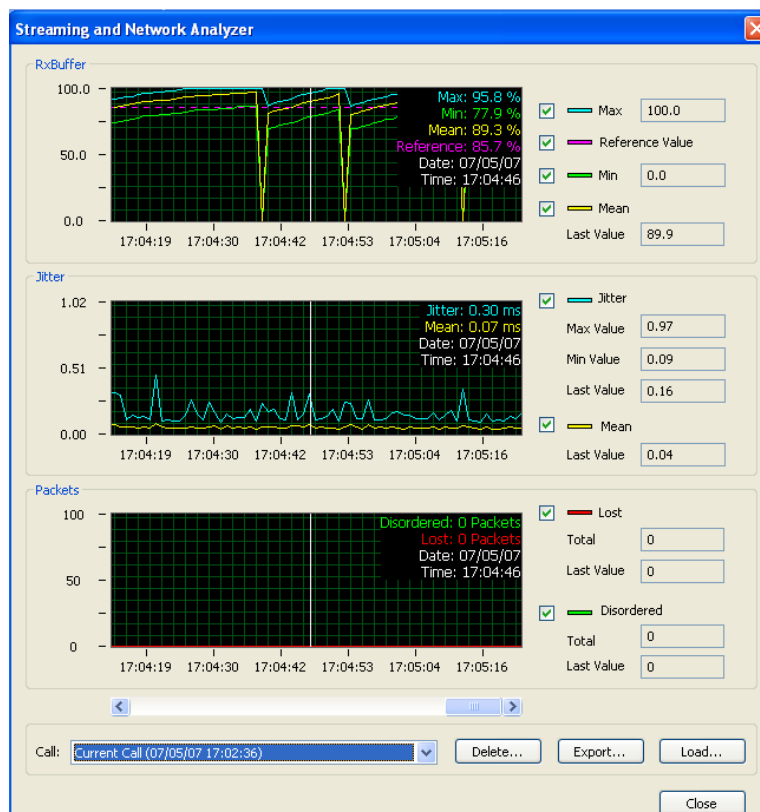
All the information is displayed in different graphs, synchronized to each other, so that the user can move through all the data very easily.

Data from each connection is stored independently so that it is possible to access data from connections other than the current one. In addition, it is possible to delete, export or import data from any previous call.
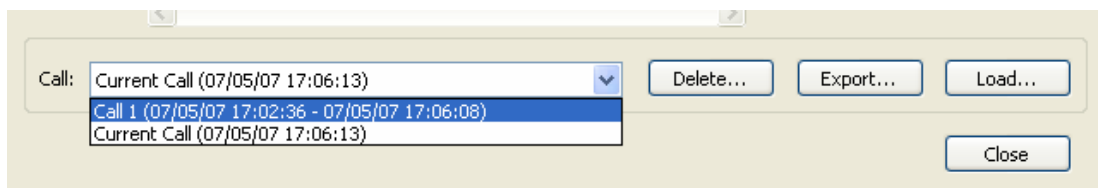
V.6.6 More about Prodys Proprietary Protocol V3 & V4

Another set of proprietary protocols (version 3) is available. This new set of protocols is not compatible backward. This new set of proprietary protocols (version 3 & 4) applies **NAT Traversal** techniques.

This new protocol multiplexes auxiliary data (RS232+GPI) into the audio stream, so that it does not need an additional TCP or UDP port for it, reducing the number of ports in the communications. In addition, this protocol has been modified so that no configuration at all is necessary in the router at the caller side: **NAT Traversal** Protocol. Only at the callee side, the router has to be configured to allow the incoming call to proceed.

This will make it possible to make calls from anywhere, regardless of whether there is a router which can be configured by the user or not. This is very useful for remote broadcasting from hotels, conference rooms or any other public location which offers a non configurable IP access to the internet, for example. More information on this protocol can be found on chapter

V.6.7 Common default IP ports for management and provision

The management ports are independent of the audio streaming protocol, but the Streaming and Audio Control ports depend on the protocol selected for the audio streaming.

According to the user application some or all ports should be considered at the NAT port definition of the default router assigned to the codec.
More about the application of every port is available at appendix "C".

HeraFlash &
Prodys Control
UDP:50013

Remote Web Control
TCP 80: HTTP
TCP 50011: Web Page
TCP 50017: ProdysControl

Test Streaming Tool
TCP 50033
UDP 50033

External Protocol (P-XP)
TCP 50031: Control Port
TCP 50035: Status Port

V.6.8 Default streaming IP ports for Prodys Proprietaty protocols

More about the application of every port is available at appendix "C".

V1 & V2 Audio Streaming
TCP 50019: P-RTCP (Control)
UDP 50021: P-RTP Unicast L1
UDP 50023: P-RTP Unicast L2
UDP 50025: PX-RTP Multicast L1
UDP 50027: PX-RTP Multicast L2
UDP 50037: P-AUXP Datos auxiliares
UDP 50039: P-UbP U-BITs

V3 Audio Streaming
TCP 51000: P-RTCP (control)
UDP 51000: P-RTP Unicast /Multicast L1 & P-AUXP Aux. Data (Multiplexed)
UDP 51001: P-RTP Unicast /Multicast L2 & P-AUXP Aux. Data (Multiplexed)

V4 Audio Streaming
TCP 52000: P-RTCP (control)
UDP 52000: P-RTP Unicast /Multicast L1 & P-AUXP Aux. Data (Multiplexed)
UDP 52001: P-RTP Unicast /Multicast L2 & P-AUXP Aux. Data (Multiplexed)

V.6.9 Table of default ports per protocol

Ports	DEFAULT PORTS[19]				
Management	**HeraFlash & Prodys Control** UDP:50013 TCP 50017	**Web Page** TCP 80 TCP 50011		**Test Streaming Tool** TCP 50033 UDP 50033	**External Protocol (P-XP)** TCP 50031 TCP 50035
Streaming and Audio Control	TCP 50019 UDP 50021 UDP 50023 UDP 50025 UDP 50027 UDP 50037 UDP 50039	TCP 51000 UDP 51000 UDP 51001	TCP 52000 UDP 52000 UDP 52001	UDP 5060 UDP 5004	UDP 5004
Protocol	Prop. V1,V2	V3[20]	V4[21]	SIP	RTP

---

[19] The ports listed here are the default ones defined either by the corresponding standards or by Prodys Proprietary Protocols. All these ports are configurable by the user.

[20] The number of ports for a simple audio connection over IP has been reduced on Prodys Proprietary Protocol V3. This new protocol multiplexes auxiliary data (RS232+GPI) into the audio stream, so that it does not need an additional TCP or UDP port for it, reducing the number of ports in the communications.

[21] The number of ports for a simple audio connection over IP has been reduced on Prodys Proprietary Protocol V3. This new protocol multiplexes auxiliary data (RS232+GPI) into the audio stream, so that it does not need an additional TCP or UDP port for it, reducing the number of ports in the communications.

V.6.10 SIP (EBU TECH 3326 standard for audio over IP)

The European Broadcast Union (EBU) is promoting the interoperability of audio codecs for any manufacturer. For this purpose the use and the application of a subset of the Internet Protocols has been proposed. SIP/SAP/RTP and SDP are the main protocols, although SIP is commonly used to refer to the whole standard. This effort will allow to setup in a friendly way audio streaming communications between several vendors of equipment.

The fist version of this standard was released in 2007 as EBU Tech 3326 standard.

The deployment of SIP Protocol Servers among the network will support calling remote parties just by invoking their network name regardless of the actual public or private IP addresses.

The physical location, often related to fixed IP addresses or subnets, is not further meaningful. In this sense, specially portable audio codecs will profit with an easy call procedure. Neither is required to agree ahead on the audio compression type and data stream rate, because the SIP Protocol manages by itself to negotiate the convenient communication details with the remote party.

For Prodys' former customers performing a call using SIP is much the same as using Prodys' proprietary protocols given that the SIP configuration is set.

V.6.10.1. Enabling SIP protocol for Prodys IP codecs

To enable SIP calls the codec administrator has to:

- Provide an IP network.
- Enable the IP network on the root NET menu.
- Enable the Unicast / Multicast on the root NET menu.
- Enable the SIP option at the Configuration >Streaming >Protocol menu.
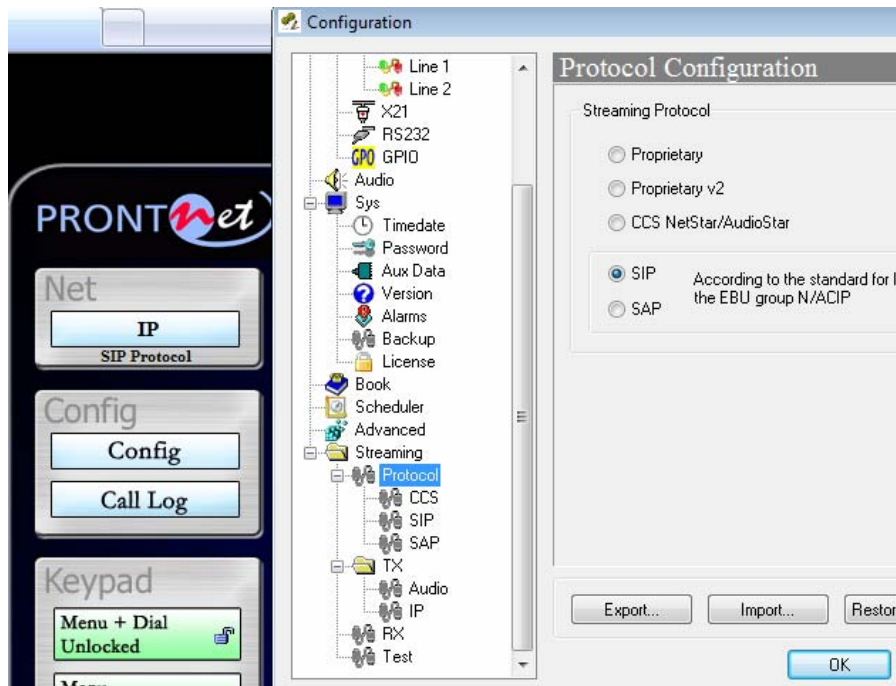
V.6.10.2. Configuring SIP

Any SIP call actor requires the setting of two IP ports; one for the signalling messages and another one for the actual audio data stream. By default the signalling port 5060 and audio port 5004 are provided complying to the corresponding standards for SIP and RTP (RFC's 3261 and 3550). The communication mate must agree with both port numbers to make a call possible.

ProdysIP Manual
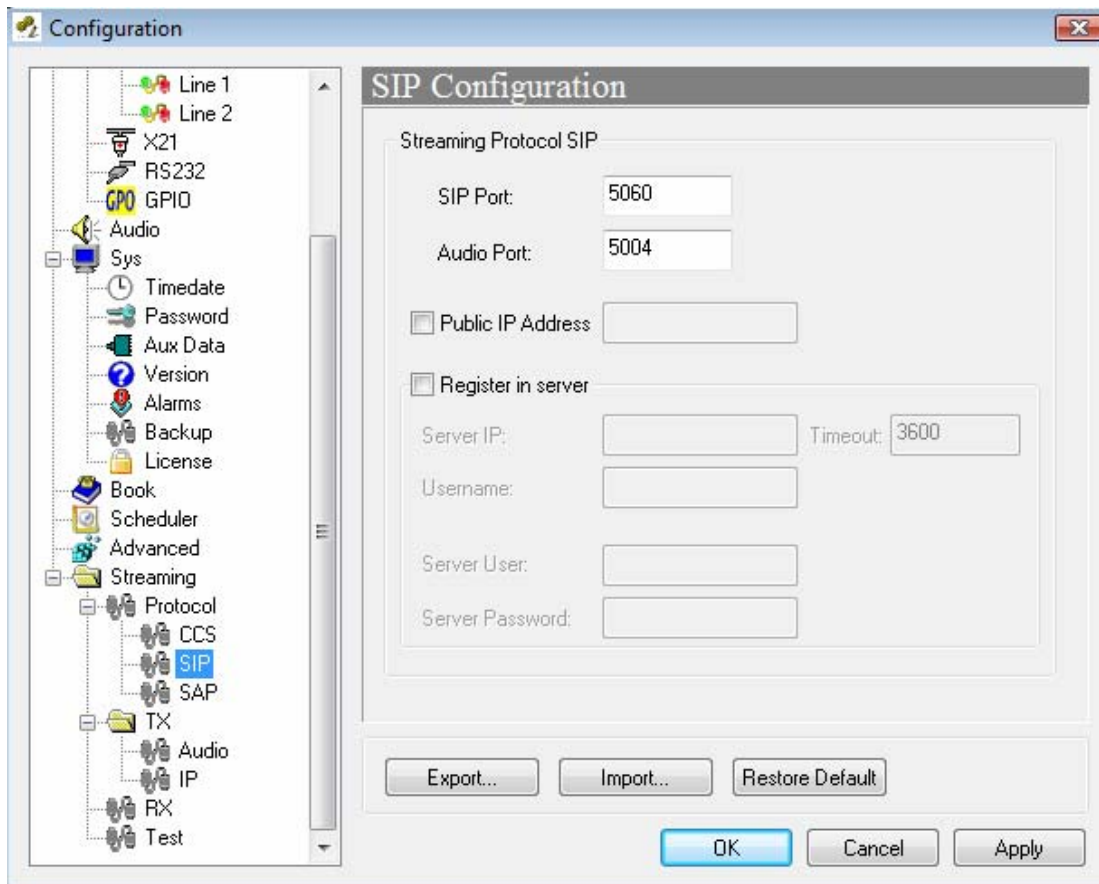
V.6.10.3. **Public IP Address**

If the codec has to manage SIP calls thru a router/firewall with NAT (Network Address Translation) other parameter must be set.

- The Public IP Address of the router's WAN interface must be configured.

- It is recommended to purchase static IP addresses access from Internet Service Provider.

If the codec is not going to connect through a router/firewall which translates from private to public IP address (NAT), it is not necessary to use this field, and should be left empty.
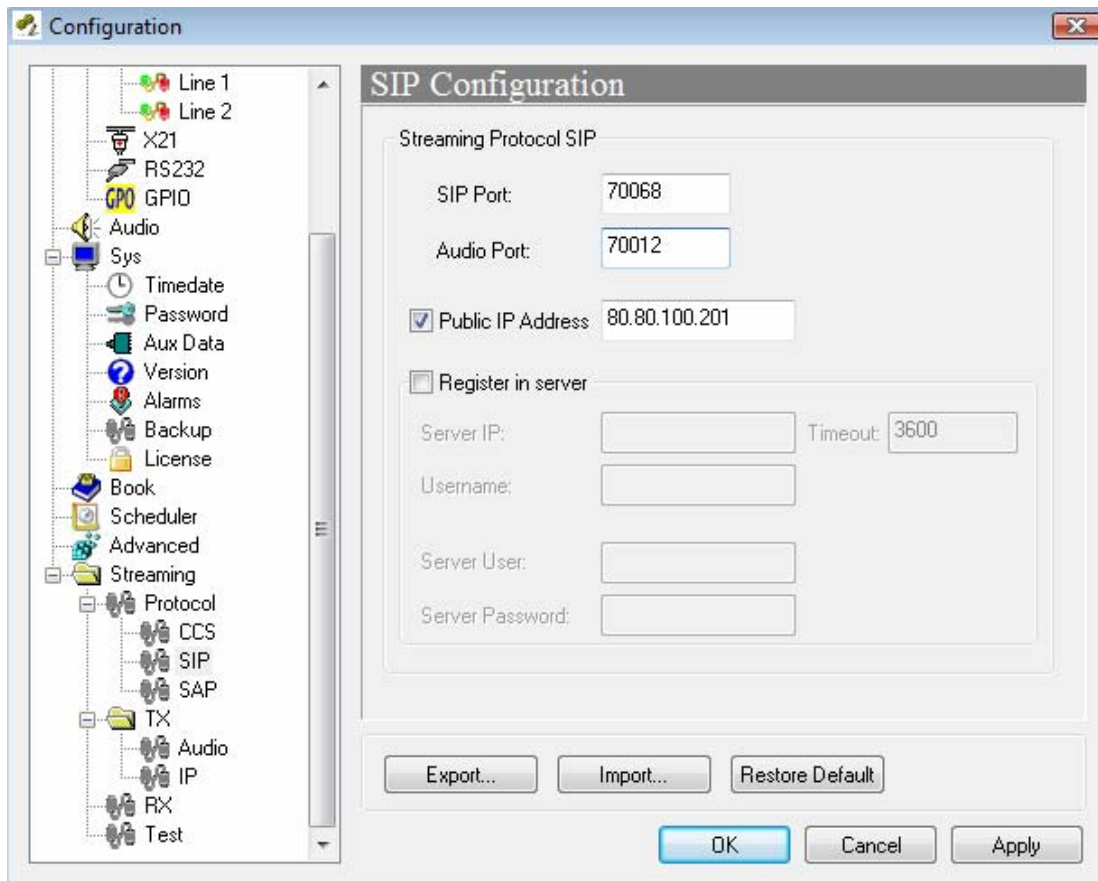
NOTE: There are different ways to discover automatically the Public IP Address from the codec which are expected to be included in the next firmware version.

V.6.10.4. SIP Server Registration

If a SIP Server is to mediate the signalling between two calling parties, each codec has to register on this server.

▪ **<Server IP>:** IP address of the SIP Server. At this place the codec registers periodically the proper user information for full protocol support.

▪ **<Timeout>:** Period (in seconds) for refreshing the user information registered.

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- **<Username>:** Your alias on the Internet regardless of your current IP address. By this alias you are identified for other SIP participants. This username will be sent along with user and password information to the SIP server.

- **<Server user> & <password>:** Most SIP servers requires authentication before proceeding registering of SIP users. This parameters should be assigned to you by the SIP Server administrator.
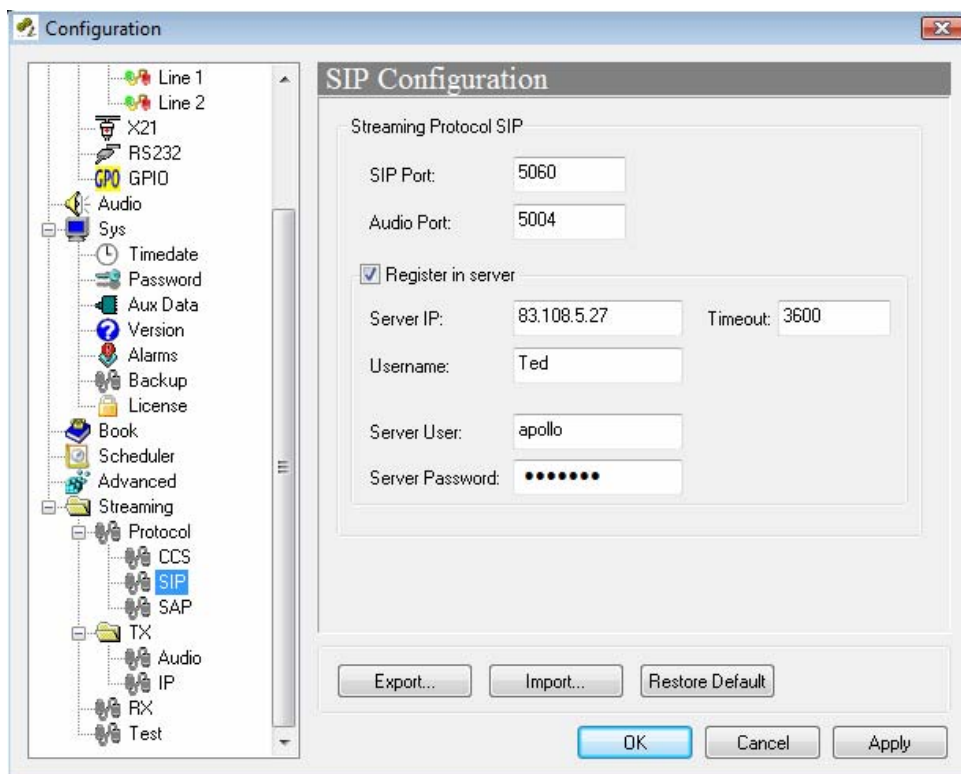
V.6.10.5. Calling with SIP

Once all parameters has been properly configured, it is possible to establish an IP connection with SIP in two different ways:

- Peer to peer.
- Through SIP Server.

Peer to peer SIP calls

When making a call peer to peer with SIP protocol, just proceed as usual, entering the corresponding IP address or domain name of the receiver peer in the dial window, select the type of call, unidirectional or bidirectional and click on ok. On the front panel, just press the CALL1 or CALL2 keys, select the type of call and enter the IP address of the remote end.

Calls through SIP Proxy Server

Once registered with the Sip Server the codec can be called by usgin only its username. Using a SIP server avoids to share continuously with communication mates IP details like IP address.

Please take into account that the address might (optional) contain the SIP server address preceded by the registered name of the called party. It is the SIP server which redirects the call to the destination if it is registered. Please, see the example below.
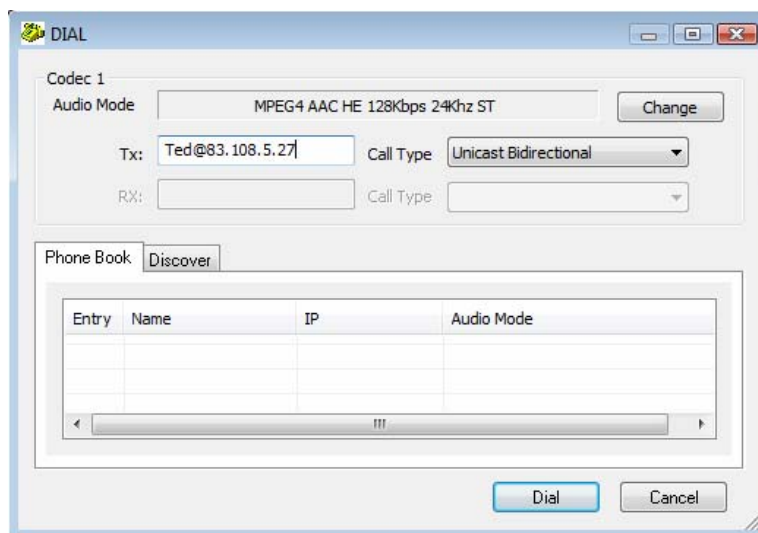
In this example, this call will be headed to the Sip server at IP 83.108.5.27, and to the user registered as 'Ted' on that server.

There are several Public SIP servers available. Please, bear in mind that some of these servers act as Proxy SIP Servers, in the sense that not only the signalling

but also the audio data goes thought them. This implies a longer audio delay in the communication, as well as a restriction on the supported compression modes for the audio connection.

V.6.11 Router configuration

■ As we mentioned before, both Prodys Protocols and SIP (EBU TECH 3326 standard protocols), use several ports in order to achieve IP connections between units.

■ If the user tries to connect two units through one or several routers, they should be configured in order to forward these ports packets to the recipient units (this is usually called Network Address Translation NAT, or NAPT).

■ A new NAT Traversal IP Protocol has been implemented (Prodys Proprietary Protocol V3). This protocol has been modified so that no configuration at all is necessary in the router at the caller side: NAT Traversal Protocol. Only at the callee side, the router has to be configured to allow the incoming call to proceed.

This will make it possible to make calls from anywhere, regardless of whether there is a router which can be configured by the user or not. This is very useful for remote broadcasting from hotels, conference rooms or any other public location which offers a non configurable IP access to the internet, for example.

■ In addition, routers/firewalls should let ICMP packets pass through from WAN interface to LAN interface, in order to let the unit ping packets to reach the recipient codec. This is only necessary till version 4.8.0, in which this protocol was removed.

■ Bear in mind that the Prodys IP Codec subnet should be the same as the router subnet, and its gateway should be the private (LAN) IP address of the router which the Codec is connected to.

■ The user should bear in mind that other protocols such us SNTP or SNMP use UDP ports for their communications, and that should be taken into

account when configuring the routers/firewalls when required. That is, SNMP 161 UDP port should be enabled/forwarded in the corresponding router/firewall when trying to control the units via SNMP from outside our network. SNTP 123 UDP port should be configured only when one the units acts as a time server, and it is going to receive time requests from outside our network.

■ On the other hand, when establishing a connection between Codecs, the IP addresses to call, will be the public (WAN) address of the corresponding routers.

■ Also, user has to keep in mind that some new Operating Systems like XP Home SP2 comes fitted with a firewall enabled by default. This **firewall** might be configured to disable incoming traffic, thus not allowing **connection to the web page**.

V.6.12 Examples of port setup for calling over IP

This chapter is aimed at providing a fundamental understanding of TCP/UDP port configuration for those scenarios where a router/firewall with NAT (Network Address Translation) requires the configuration (open/forwarding) of the incoming traffic based on the TCP/UDP port number in the router.

Prodys is ahead of the development of the current standard from the EBU group for audio distribution over IP, although we also support our own proprietary protocols for those cases where they better suit customer needs. Therefore, this chapter comprises both, configuration for Prodys Proprietary Protocols and for SIP/SAP/SDP/RTP protocols (complying to EBU Tech 3326 standard)[22]. From now on we will use only SIP to refer to all protocols included in this standard.

V.6.12.1. Different network topologies and port configuration

This chapter describes how to configure and operate with Prodys IP codecs when making IP connections for different network topologies.

---

[22] SIP is the commercial name commonly adopted to refer to the EBU standard for audio over IP. This standard also comprises other protocols such us SAP, SDP and RTP.

The following picture depicts a complex network where star topology and array configuration have been merged.

- If only one single codec behind a router/gateway is to connect, you might refer to the "One-to-One topology" chapter below.
- If multiple codecs are deployed sharing the same gateway (e.g. Nereus or several ProntoNets on one site), the configuration description in "Array of codecs in a back-to-back configuration" section below must be taken into account.

- The third scenario would be that where only one codec (A) behind a router is to communicate with other codecs (B,C) at the remote site sharing a common router. In this case, the configuration detailed on "One-to-One topology" chapter would be applied to A, whilst B and C would require the configuration defined in "Array of codecs in a back-to-back configuration".
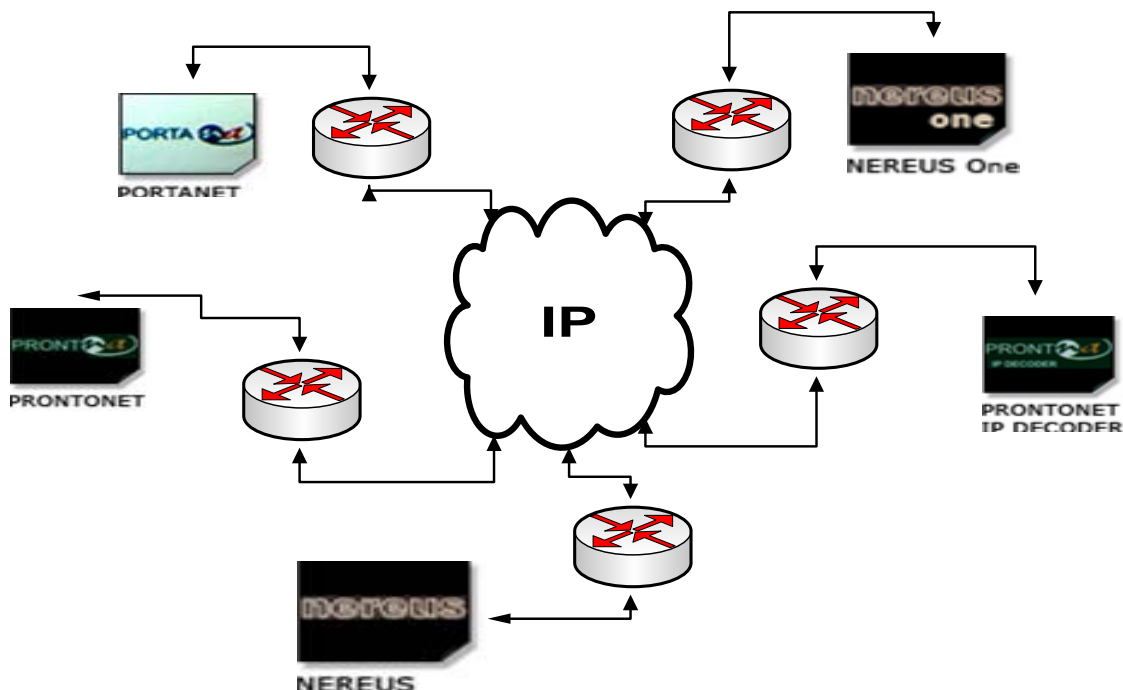
V.6.12.2. One-to-one topology

This section describes how to connect two codecs over an IP backbone or Internet in a one-to-one scenario where only one codec is placed behind the router/gateway.
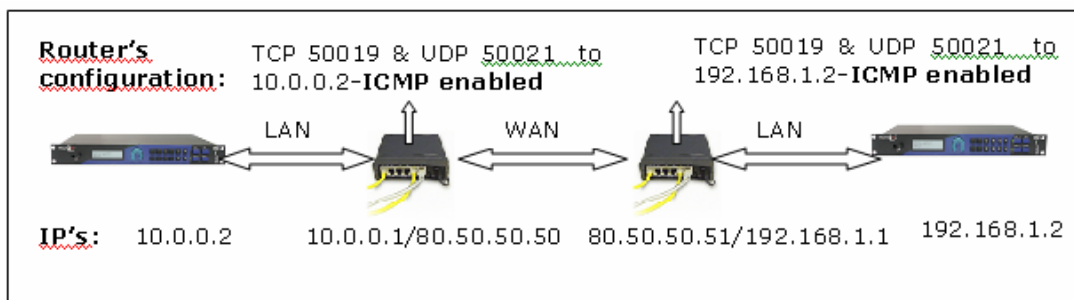
It is assumed one IP-gateway at each place (labelled as WEST and EAST). In this way, the WAN IP address of the router determines the external internet address of the codec behind this gateway.

- In order to call from WEST to EAST, the user has to call to the Internet (or public) IP address of the EAST router: 80.50.50.51. If we try to call to the local IP address (192.168.1.2), it will be consider a LAN address and the connection will not succeed.

- Router NAT configuration: Let all Prodys ports packets pass through and be forwarded to local IP address of ProntoNet (10.0.0.2 or 192.168.1.2). Please refer to the user manual of your router; usually you have to look for "NAT", "PAT" or "Port Forwarding" description of your network device. Here follows an example of NAT/NAPT configuration for the labelled as 'WEST'.

Entry for router	First port	Last port	Local address	Map port
Codec West	80	80	10.00.2	80
Codec West	5004	5004	10.00.2	5004
Codec West	5060	5060	10.00.2	5060
Codec West	50011	50039	10.00.2	0 [23]

---

[23] Map port "0" means for most of the router configuration that the external port number matches the port number for the local area network. Given an example: an incoming request from the

Please make sure that for both codecs the default port assignment is provided. Take into account that this router configuration only applies when the default ports for IP communications are used. In case you want to make sure that the default ports will be used for the connections, the user can restore the default port configuration from the port configuration window on the control web as shown in the picture below, by clicking on the 'Restore Default' button under the Web Server Port entry box[24].

NOTE: Take into account that with the new Prodys Proprietary Protocol V3, the router configuration is only necessary at the callee side, as this protocol applies NAT Traversal techniques.
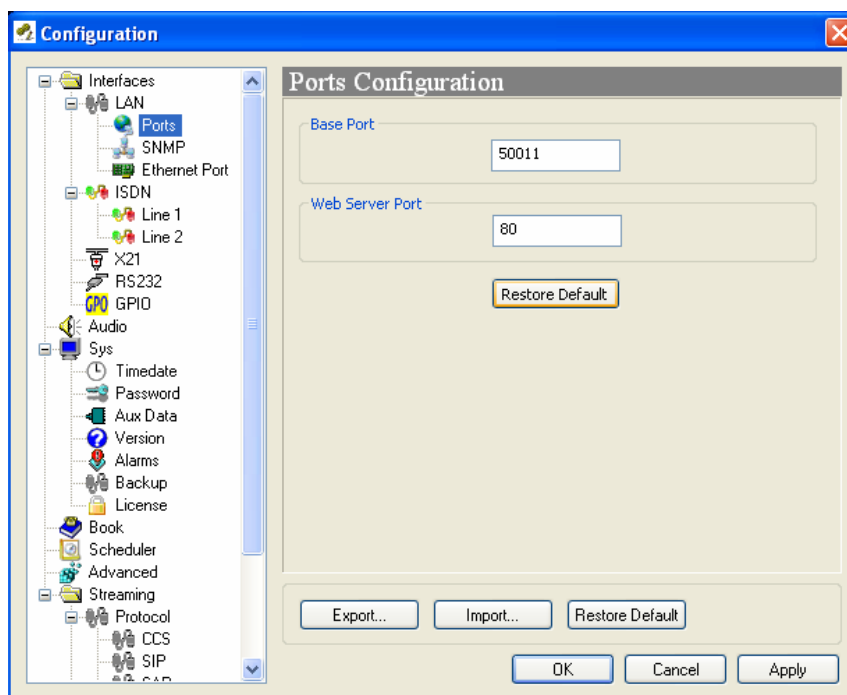
WAN interface to the port number 50021 will be forwarded to the codec on the same port number 50021.

[24] The other button labelled as Restore Default at the bottom of the screen will restore the factory default configuration for all settings, not only ports.

V.6.12.3. Array of codecs in a back-to-back configuration

This section describes an example of how to match an array of four Prodys Family codecs with a mate array of four Prodys codecs at an opposite place. The following arrangement allows to share up to four high quality audio stereo signals between both sites. The same procedure can be extended for any number of codecs inside the array.
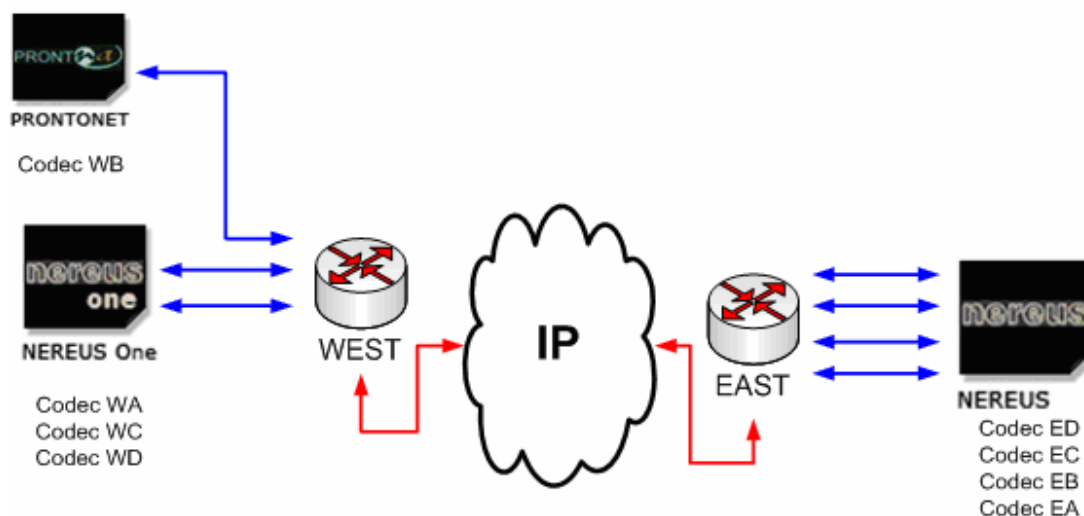
It is assumed one IP-gateway at each place (labeled as WEST and EAST). In this way, the WAN IP address of the router determines the external internet address of all four codecs behind this gateway. Let us write a place holder for this example as:

- YY.XX.ZZ.WW (any valid external IP address of your IP backbone or IP provider). (IP address gateway at site "West").
- YY.XX.ZZ.EE (any valid external IP address of your IP backbone or IP provider). (IP address gateway at site "East").

Behind the gateway "West" four codecs are available on the LAN. Let us write a place holder (any valid local IP address) for this example as:

- LL.YY.WW.AA (codec A)
- LL.YY.WW.BB (codec B)
- LL.YY.WW.CC (codec C)
- LL.YY.WW.DD (codec D)

The next step is to avoid confusion when four Prodys Family codecs has to be addressed by means of one single gateway thus, sharing one common Public or External IP address. This is achieved in two steps:

- By putting in place a "Network Address Port Translation" mechanism inside the gateway. Please refer to the user manual of your router; usually you have to look for "NAT", "PAT" or "Port Forwarding" description of your network device. Please, find below an example of router configuration for the router labelled as 'WEST':

Entry for West	First port	Last port	Local address	Map port	
Codec WA	50001	50060	LL.YY.XX.AA	0	25
Codec WB	50061	50120	LL.YY.XX.BB	0	26
Codec WC	50121	50180	LL.YY.XX.CC	0	
Codec WD	50181	50240	LL.YY.XX.DD	0	
Codec WA	5004	5004	LL.YY.XX.AA	5004	27
Codec WA	5060	5060	LL.YY.XX.AA	5060	
Codec WB	5104	5104	LL.YY.XX.BB	5104	
Codec WB	5160	5160	LL.YY.XX.BB	5160	
Codec WC	5204	5204	LL.YY.XX.CC	5204	
Codec WC	5260	5260	LL.YY.XX.CC	5260	
Codec WD	5304	5304	LL.YY.XX.DD	5304	
Codec WD	5360	5360	LL.YY.XX.DD	5360	

figure 6

---

[25] The First Port number can be any available port number for the gateway. We recommend to agree with your network administrator which the port ranges are fully available for the codecs for not to overlap with other IP services of your network.

[26] Map port "0" means for most of the router configuration that the external port number matches the port number for the local area network. Given an example: an incoming request from the WAN interface to the port number 50073 will be forwarded to the codec B on the same port number 50073. In other case one port by one must be defined according the Prodys Family User Manual.

[27] Ports 5X60 and 5X04 in this example are only used in case of using SIP protocol.

NOTE: When calling with Prodys Proprietary Protocol V3 is not necessary to configure NAPT (Port Forwarding) in the router placed at the caller side, only at the callee side.

- The following step is to define a range of ports to be used by the unit for its IP communications which matches the port forwarding range defined for NAT/NAPT on the router/gateway. This can be done by the user from the 'Ports Configuration' window from the web page or via the menu screen on the front panel.
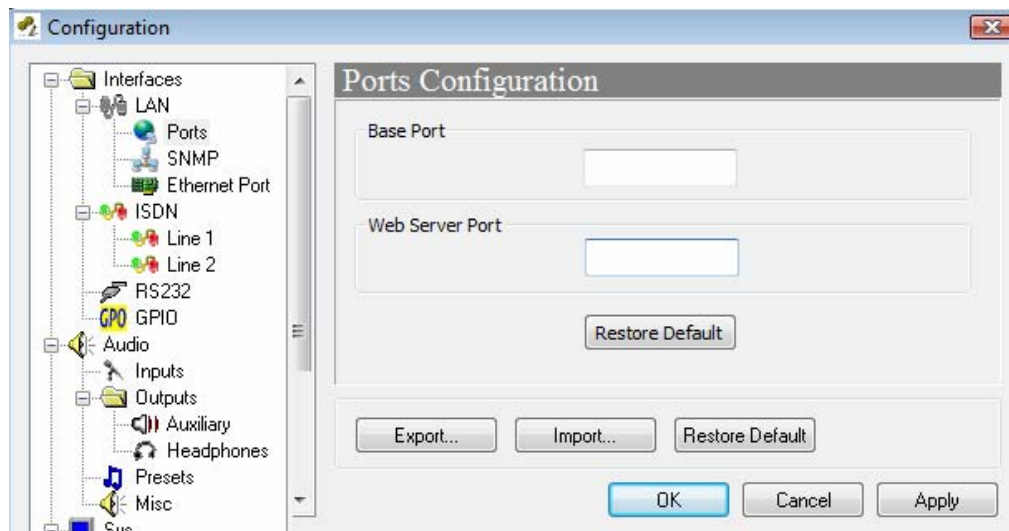
In case of using SIP for establishing IP audio connections, the configuration window would be the following:

ProdysIP Manual
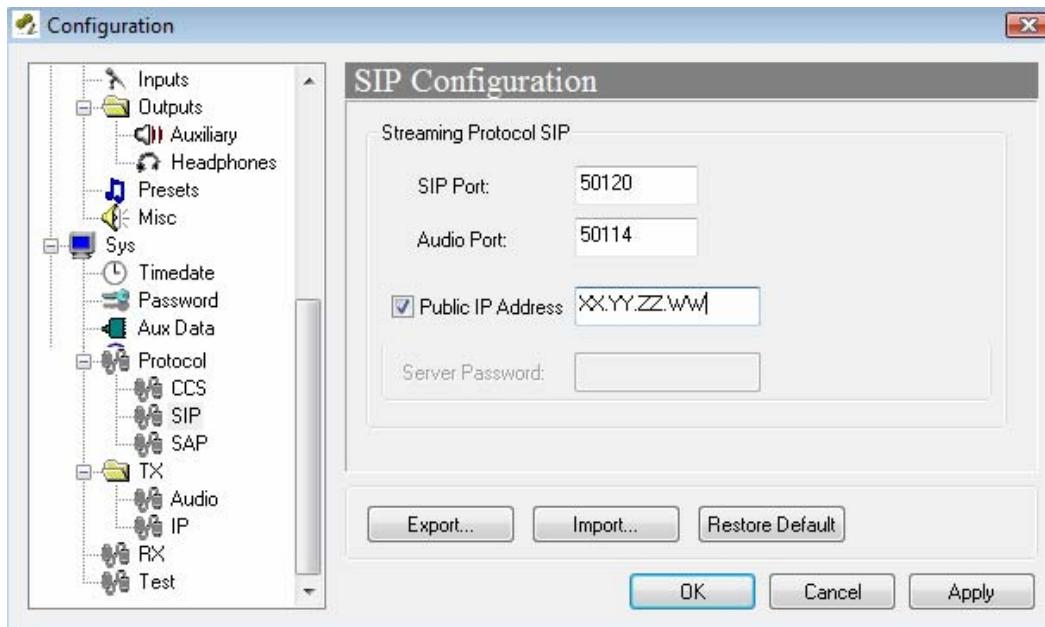
figure 8

Any SIP port and Audio port inside the gateway assigned range is valid. In this way each codec is defined with different SIP and Audio ports along the codec array.

NOTE: As per the current firmware version (5.5.0), it is mandatory to enable the Public IP Address option in the SIP configuration when making calls through routers/gateways with NAT[28], and to provide the public IP address of the router's WAN interface manually.

Here follows a table summarizing the different port configurations depending on the units for both Prodys Proprietary Protocols and SIP/RTP standard protocols.

Entry for West	Web Server Port	Base Port	SIP Port	RTP Port
Codec WA	50001	50010	5060	5004
Codec WB	50061	50070	5160	5104
Codec WC	50121	50130	5260	5204
Codec WD	50181	50190	5360	5304

---

[28] Network Address Translation: The router translates private IP address into public addresses.

In order to complete the configuration and, only when establishing IP connections with Prodys Proprietary protocols[29], the codecs on the receiver site (in this example the "East" site) must register the public (external) addresses of the codecs at the "West" site. This is done with a BOOK entry at each codec.

Figure 9 shows an example of one BOOK entry for destination "Codec WA" at the mate codec of the "East" site.[30]
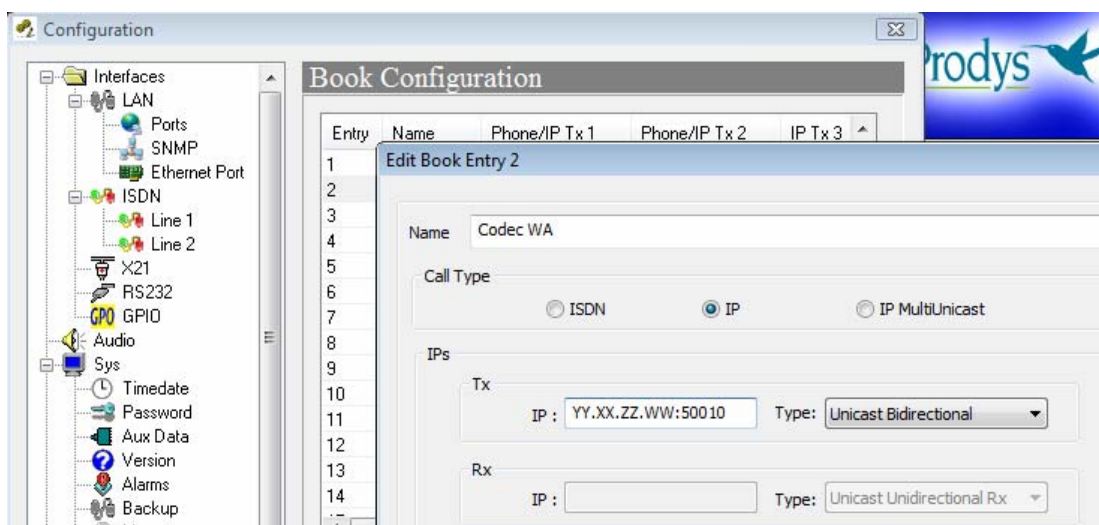
Note that each BOOK entry must meet the correct Base Port of the destination.

So far the settings for the "West" site. The whole procedure must be repeated for the "East site". This is:

- Port Forwarding provision on the eastern gateway. (It is not necessary to meet the same port ranges given at the "West" site. **NOTE: When calling with Prodys Proprietary Protocol V3 is not necessary to configure NAPT (Port Forwarding) in the router placed at the caller side, only at the callee side.**
- Port definition at each eastern codec according "East" gateway provision.

---

[29] Please note calling via the SIP protocol does not require keeping a registration of the call initiator inside the receiving codec BOOK.

[30] If all western codecs are registered at each eastern codec, any eastern codec is able to receive an incoming call from any western codec.

- BOOK entries at each western codecs for their mates when using Prodys Proprietary Protocols.

For initiating a call: the destination IP address must be dialled together with the proper destination Base Port.
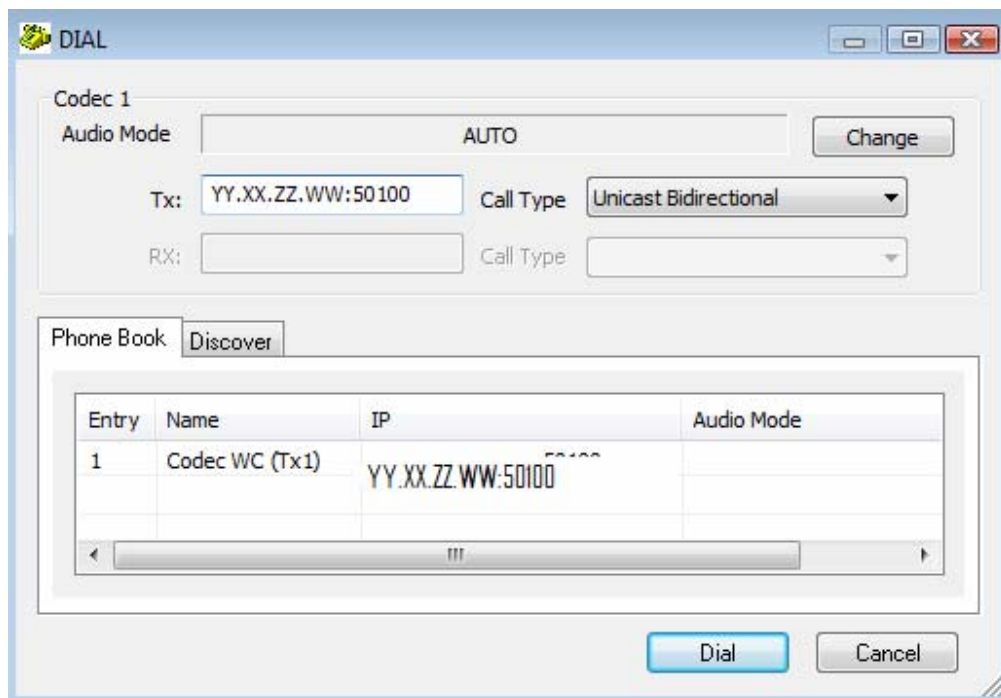
V.6.13 Other not audio network PROTOCOLS

The Prodys IP Codecs also support many other network protocols, which are listed next:

-DHCP: IP address auto configuration.
-DNS: Domain Name Server.
-IGMP: This protocol deals with Multicast traffic.
-ICMP: IP control protocol.
-HTTP: A embedded web server allows the user to monitor/configure the unit from an Internet Explorer Web Browser.
-RIP2: Internal/alias IP address can be assigned.
-SMTP: Simple mail transfer protocol. Allows the user to send email alarm notifications.

-SNTP: Simple network time protocol. Time synchronization between units. UDP port number 123.
-SNMP: Simple network management protocol over UDP on port number 161.

V.6.14 PRODYS default factory IPv4 configuration

Prodys factory default settings for the IP parameters are:

-IPv4 address: 192.168.100.100
-Netmask: 255.255.255.0

For further information about multicast, please, read the IP Addressing Overview.ppt file, available in the Prodys site download section.

The user can restore these default settings by pulling down DIP switch number 7 and power cycling the unit. After that, the user can establish different values for these parameters either from the front panel or from the web page. Do not forget to pull the DIP switch up in order to maintain these settings.

V.6.15 Forward Error Correction (FEC)

Streaming audio over Wide Area Networks, like Internet, means living with a small chance that some bits audio information are lost along their way to the far-end codec. Those drop-outs would be perceptible without any counter-measure.

Prodys IP Family enables, if required, the Forward Error Correction (FEC) protocol as per the RFC 2733 standard. Essentially this protocol streams redundant audio information over the network. The decoder, at the far-end, recovers the audio information either from the main or the redundant IP stream. The chance to loose audio information is now negligible.

Transmission of redundant information pays a penalty due higher IP bandwidth requirement. For example a <50% FEC> means a 50% of bandwidth overhead, and it is able to recover from isolated lost packets. Also a <100% FEC> is possible, which doubles the bandwidth requirements. This method clearly requires additional delay at the sender. However, it allows some bursts of two consecutive packet losses to be recovered. The <FEC Port> is applicable for the

SIP protocol and defines the UDP Destination Port for the redundant audio information.

The proprietary protocol V4 interlaces the redundant audio information with the main audio information on the same UDP Destination Port. Streaming resilience can be set within 24 user definition steps. The actual streaming overhead can range therefore from 4% up to 100%.

Following examples are given:
- A 128kbps MPEG Layer III with 50% FEC requires actually a 192kbps bandwidth.
- A 128kbps MPEG Layer III with 100% FEC requires actually a 256kbps bandwidth.

Prodys recommends starting trials with small FEC settings, because this is the best trade-off for the average WAN performance.

The user enables the FEC only at the Encoder side, since the Decoders takes automatically advantage of the FEC information without explicit indication.

Following examples are given for different streaming arrangements:
- The far-end codec has FEC enabled. The near-end codec has not. This arrangement makes the audio decoding at the near-end codec more reliable. The IP bandwidth requirement is asymmetric for both ends.

- Conversely, the near-end codec has FEC enabled. The far-end codec has not. This arrangement makes the audio decoding at the far-end codec more reliable. The IP bandwidth requirement is asymmetric for both ends.

- Both codes have FEC enabled. This arrangement makes the audio decoding more reliable at both ends. The IP bandwidth requirement is symmetric for both ends.

V.7 Prodys IP Family codec operating as an ISDN codec

The Prodys IP Family codec as ISDN codec is very similar to the Pronto 3, therefore those users that are familiar with the Pronto 3 will find it easy to operate the Prodys IP Family codec over ISDN.

The most important points that a user must have in mind for this mode are as follows:

1. **ISDN interface:** The ISDN interface of the Prodys IP Family codec allows connection to a basic rate ISDN line. This interface allows the user two bi-directional channels with a bandwidth of 64 Kbps in each direction. There is an additional channel of 16 Kbps that is used for signalling.   This is why a basic rate connection is sometimes represented as 2B+D. The Prodys IP Family codec identifies each B channel as Line 1 and Line 2 respectively.

2. **Dual Codec:** The Prodys IP Family codec can operate as "DUAL CODEC", which means it can establish two totally independent communications via each B channel at 64 Kbps. This is only possible when each B channel or communication line uses a MONO mode compression algorithm in this situation the Prodys IP Family codec will be able to use two Encoders and two Decoders independently. This limitation is imposed by the fact that there is only one stereo audio input and one stereo audio output.
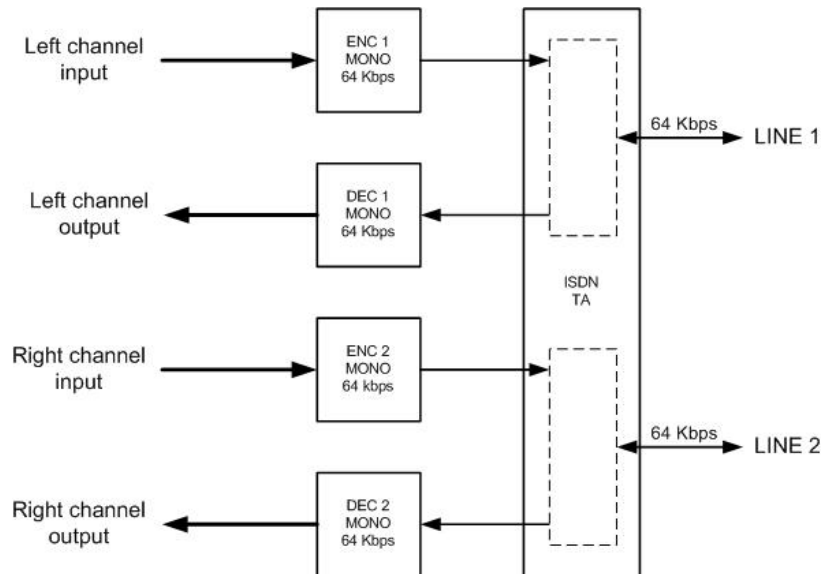
PRONTONET AS CODEC DUAL

> **The left audio I/O channel is always assigned to Line 1 and the right audio I/O channel is always assigned to Line 2.**

3. **Communications in Dual mode, Joint Stereo or Stereo at 64 Kbps**: When we configure Encoder 1 to Dual mode, Joint Stereo or Stereo at 64 Kbps, Encoder 2 is disabled since there are no free audio inputs or outputs to assign to it.
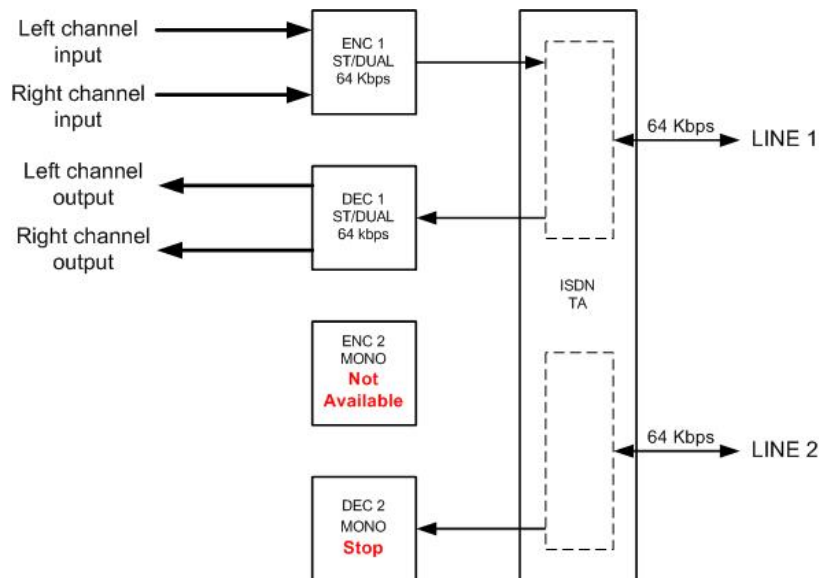
PRONTONET WORKING AT 64 Kbps IN STEREO/DUAL MODE

4. **Communications at 128 Kbps – Inverse Multiplex:**

Certain algorithms permit the compression of the signal at bit-rates other than 64Kbps, for example at 128Kbps. With the Prodys IP Family codec it is possible to send or receive audio compressed at 128 Kbps using both the B channels of the basic rate ISDN connection. This is done by using 'inverse multiplex' techniques that split the transmission of the full 128 Kbps over two channels of 64 Kbps and then sum the parts together again at the other end.

Inverse Multiplex is often known as 'bonding'
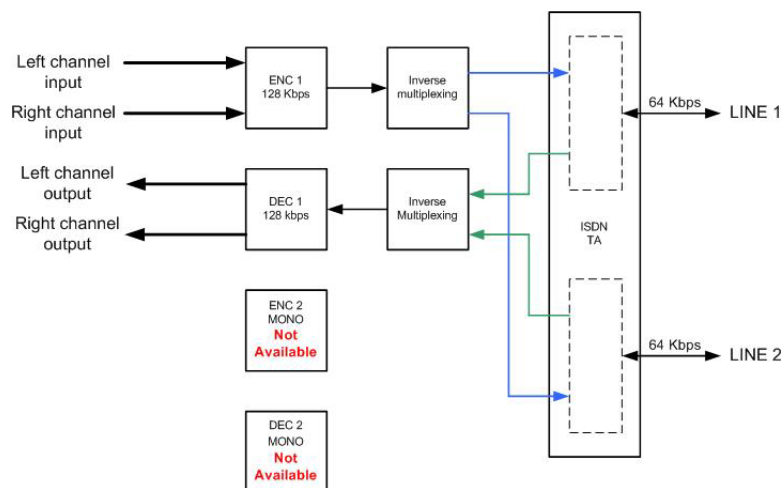
PRONTONET WORKING AT 128 Kbps

5. **Communications combining 128 and 64 Kbps**: It is possible to transmit encoding audio at 128 Kbps and to receive decoding audio at 64 Kbps or vice versa. Here below there are different examples of it.
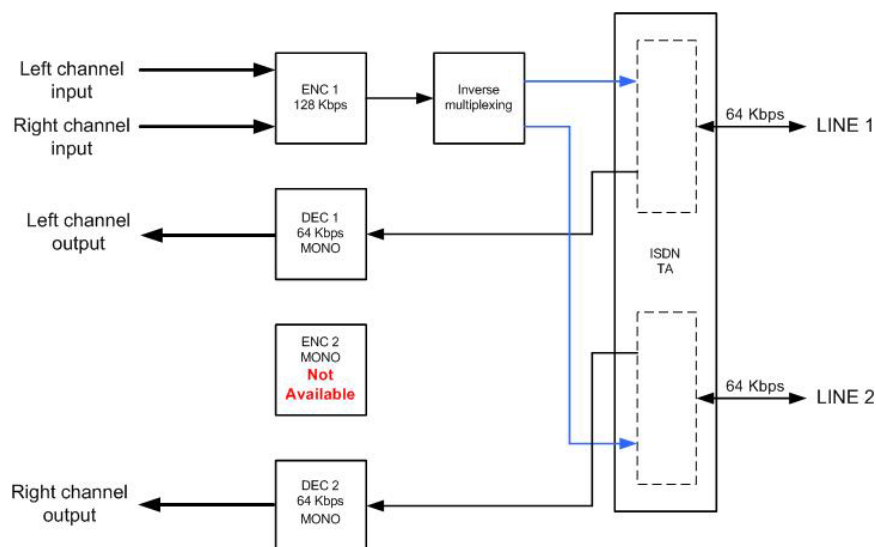
PRONTONET WITH ENCODER AT 128 Kbps & DECODER MONO AT 64 Kbps
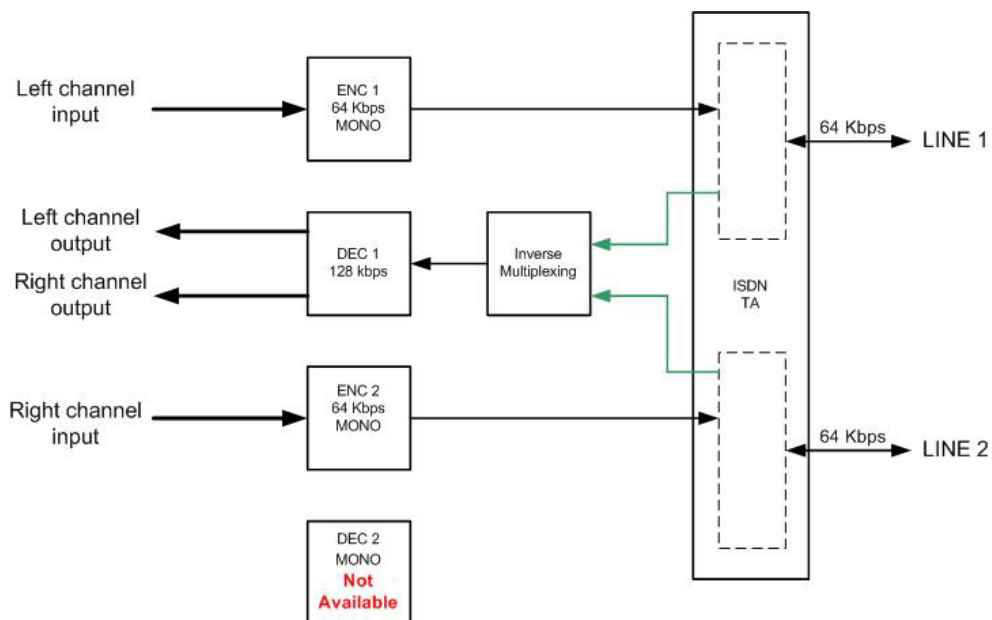
PRONTONET WITH ENCODER AT 64 Kbps & DECODER AT 128 Kbps
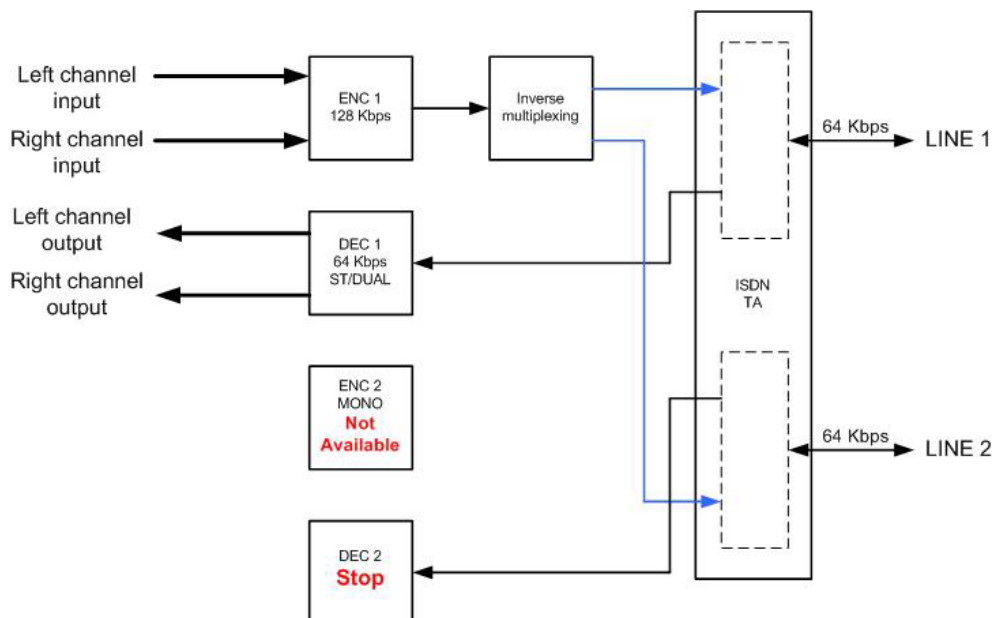
PRONTONET WITH ENCODER AT 128 Kbps & DECODER ST/DUAL AT 64 Kbps

V.7.1 Establishing ISDN calls

The proper NET parameter must be selected for ISDN calls (see picture on section III.3.1). ISDN destination numbers are entered at the dial menu or an entry from the Address Book is selected using the BOOK key[31]. Bear in mind that the use of an entry in the Address Book will also reconfigure the Encoder and so is not limited to only making calls. We need to distinguish between calls at 64 Kbps and calls at 128 Kbps:

- Making calls at 64 Kbps

The user has to select an audio compression mode compatible with a 64kbps bearer channel. Then he has to type the destination ISDN number into the <Phone> box of the Line 1 or Line 2.

- Making calls at 128 Kbps

The user has to select an audio compression mode compatible with a 128kbps bearer channel. Then he has to type the destination ISDN number into the <Phone> box of Line 1.

The codec will dial first for the bearer channel assigned Line 1, then after automatic or manual response (see next section for details) of the first bearer channel, the second bearer channel is dialled by the codec towards the mate bearer channel of the same destination. After automatic or manual response (see next section for details) of the second bearer channel the audio streaming over a 128kbps capacity can go ahead.

---

[31] ISDN sub-addressing is possible apending the ISDN number to dial with the ASCII char (version 5.5.0 onwards).
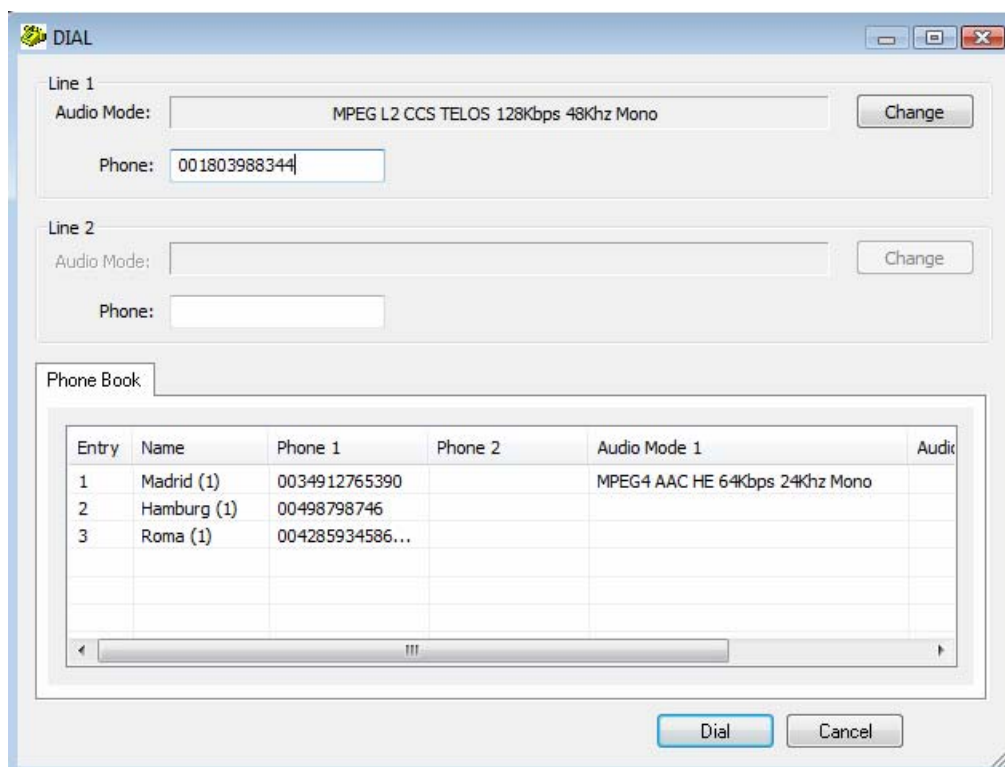
V.7.2 Receiving calls via ISDN

The user must bear in mind the ISDN configuration according section IV.1.5 for both, Line 1 and Line 2. Any inconsistent configuration might prevent the expected operation (eg. Manual anser, incoming call filter, etc.).

V.7.3 Restrictions in ISDN communications

As we have already stated, the Prodys IP Family codec can work as a DUAL CODEC over ISDN, meaning it can operate with two independent lines. However, this is only possible when two MONO communications are used. The use of Dual, Joint Stereo or Stereo modes creates certain restrictions on the system that are in place to avoid conflicting situations.

1. DUAL, JOINT STEREO or STEREO modes are only available on Encoder 1.

2. When Encoder 1 is configured to DUAL, JOINT STEREO or STEREO, Encoder 2 is not available.

3. **The Prodys IP Family codec will connect the line but the audio will not be decoded in the following cases**:

   a. The line 2 is connected and the Decoder 2 is FRAMED. Prodys IP Family codec receives a call in the line 1 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

      Line 1 will connect but the audio will not be decoded since Prodys IP Family codec working as a DUAL CODEC only allows two MONO communications. The display shows an error code to indicate the situation.

   b. Prodys IP Family codec receives a call in the line 2 and detects that the audio is encoded in DUAL, JOINT STEREO or STEREO mode at 64 Kbps.

      Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows an error code to indicate the situation.

   c. The line 1 is connected and the Decoder 1 is FRAMED in DUAL, JOINT STEREO or STEREO mode at 64 Kbps. Prodys IP Family codec receives a call in the line 2.

      Line 2 will connect but the audio will still not be decoded since these modes are not allowed on Line 2. The display shows "**STOP**" to indicate the situation.

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V.8 Prodys IP Family codec operating as an X21 codec

It is mandatory to select match the clock speed of the X21 network supporting the communications as shown in the picture.
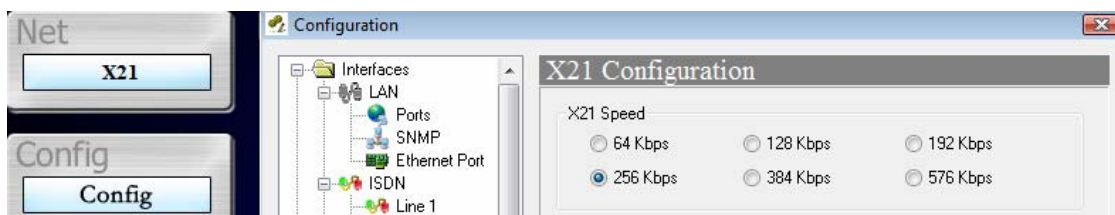
For starting an audio transmission and reception over the X21 interface the user must click the <Connection> bar from the local and the remote codec. Since no call establishing protocol takes place over the X21, the user must start and end the connection over the X21 network manually.
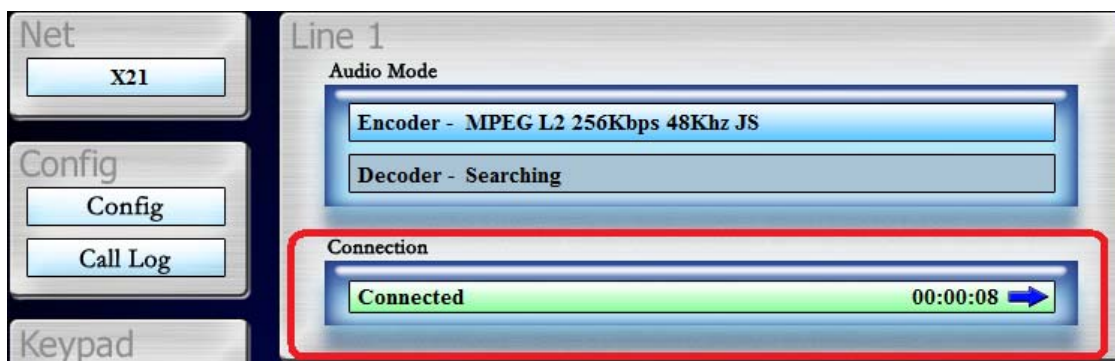
V.9 Net Backup function

A Prodys IP Family codec, streaming audio over IP or X21 networks might switch the communication to stream over an ISDN network if the default network fails. This option is available in <Net Backup> menu and increases the availability of the audio stream to the remote site.

> It is recommended to disable the <Net Backup> function if the ISDN network is not in place, or the codec is not yet completely configured, in order to avoid unsuccessful attempts to start backup calls.

Main Line
IP / X21

ProntoNet
MASTER

ProntoNet
SLAVE

Backup configuration
according the phone
book - index 1

ISDN CALL
when Backup
is ON

Backup On
when ISDN
CALL is
received

ISDN

BOOK EDITOR [01]
{ISDN}IP ENC1 ENC2

Backup Line
64 / 128 Kbps
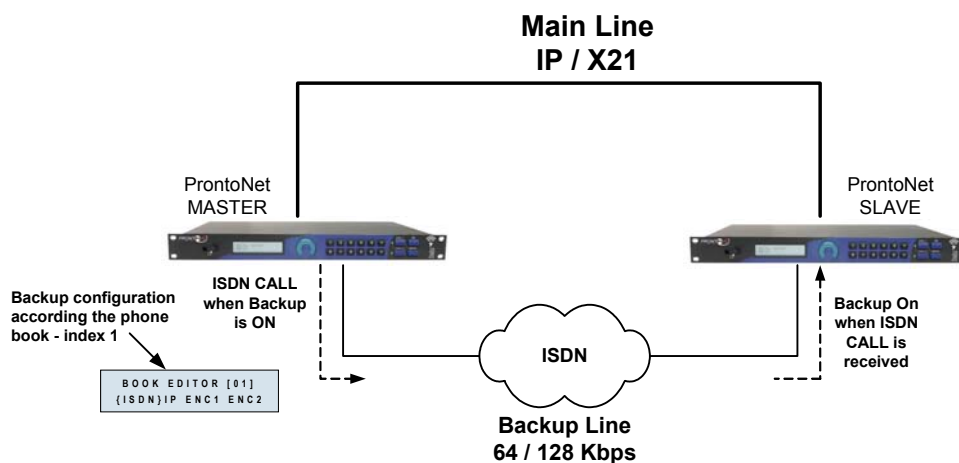
V.9.1 Net Backup MASTER Configuration

The Net Backup operation requires the user to define forehand one codec as the Net Backup MASTER, while defining the communication's mate codec as Net Backup SLAVE.

Once the backup option has been enabled, the codec configured as MASTER monitors continuously the main path (IP or X21) for a faulty performance. Once the quality of the main path undergoes a user defined threshold the communication is switched to stream over the ISDN line.

As soon the main path's performance exceeds again the minimum threshold defined by the user, the MASTER returns to streaming back the audio over the main path.

Either of two performance measurements could trigger the Net Backup function:
- <Decoder Framed Lost>: The <Timer Down> defines the delay time in seconds after interruption before switching to the ISDN network. The <Timer Up> defines the delay time in seconds to determine that the main network is operational again.
- <Drop-outs Number>: This is a performance measurement for the decoded audio stream. The parameters <Number> and <Window Time> defines the amount of transmission errors allowed in a timeframe, before switching to the ISDN network.
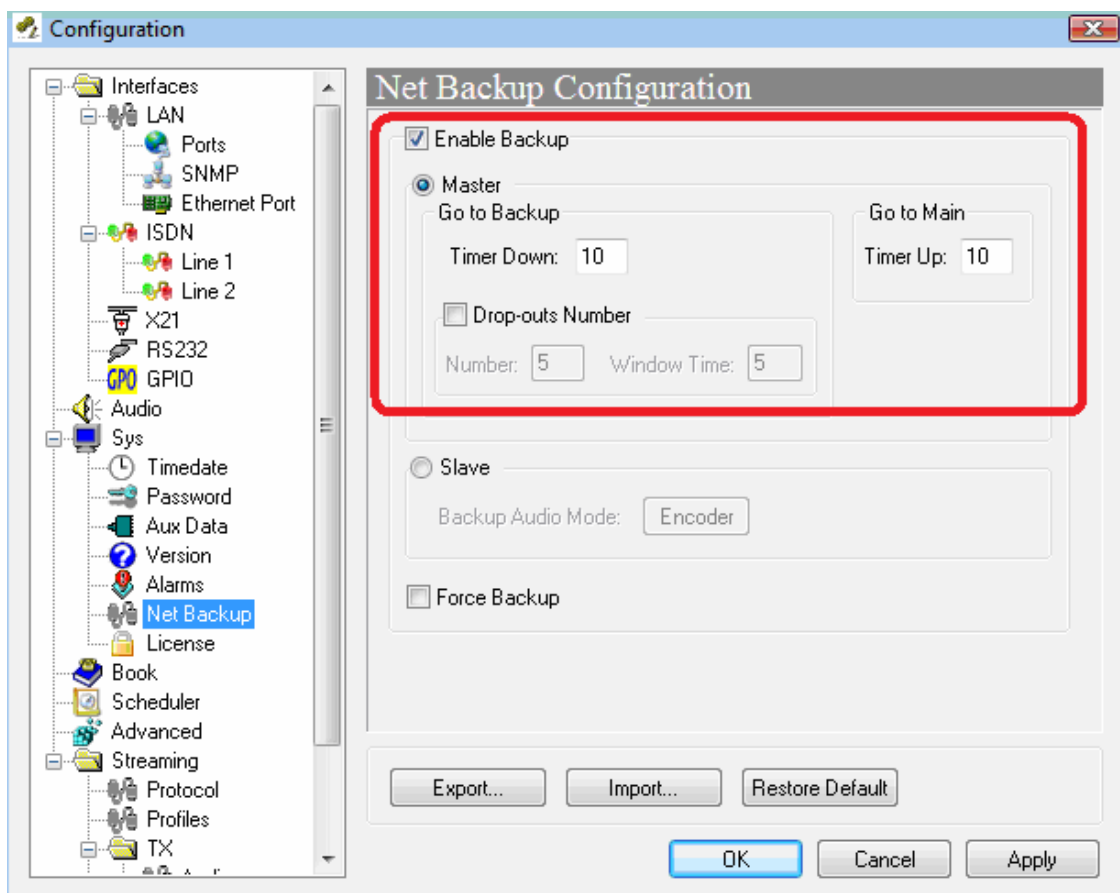
The Prodys IP Family codec MASTER uses the information stored in the index 1 of the phone <BOOK> to configure the unit and to call to the stored numbers.

V.9.2 Net Backup SLAVE Configuration

A Prodys IP Family codec defined for Net Backup SLAVE operation just reacts automatically on the MASTER request to start a new communication over ISDN.

The SLAVE codec will drop any ongoing communication over the main network (IP or X21) if an ISDN ringing is detected at this interface. The audio streaming over ISDN will progress.
As opposed the SLAVE codec will drop any ongoing ISDN communication if the MASTER is requesting to switch again to the main network.

In order to prevent any other ISDN request but the MASTER backup call it is recommended to put in place the proper filter configuration for in-dialling ISDN calls for Line 1 and 2.
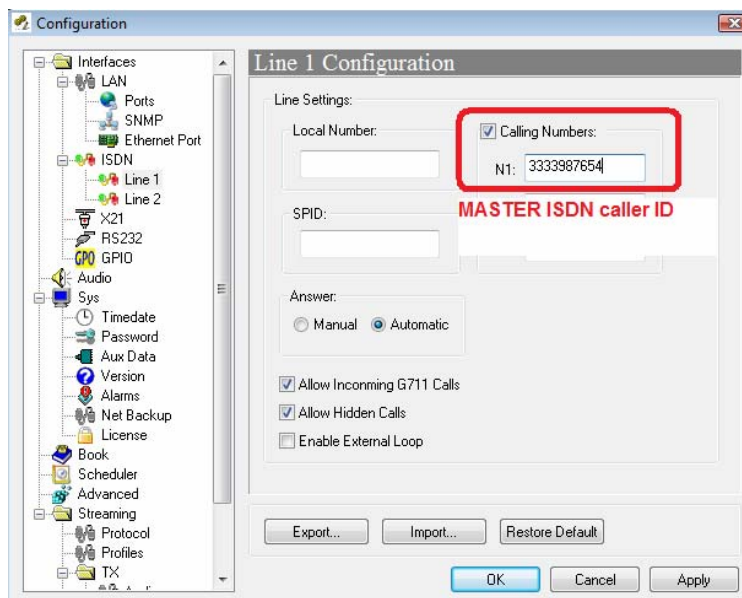
The user might define forehand the compression algorithm to use over the ISDN Net Backup path by the SLAVE codec (obviously those must fit into a 64kbps or 128kbps channel).
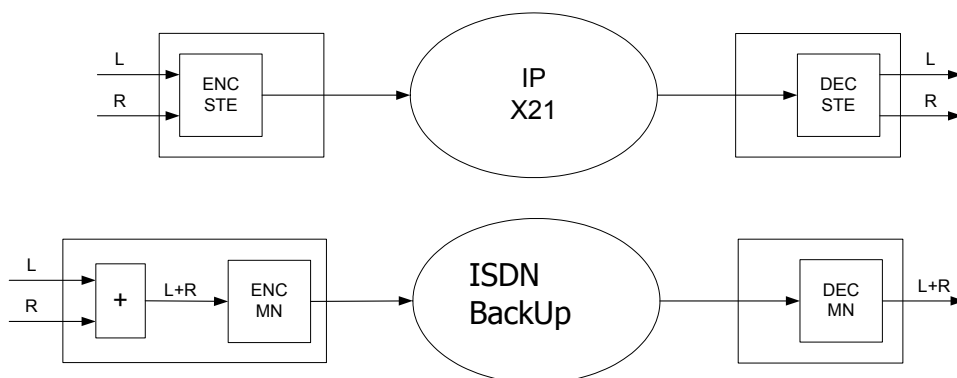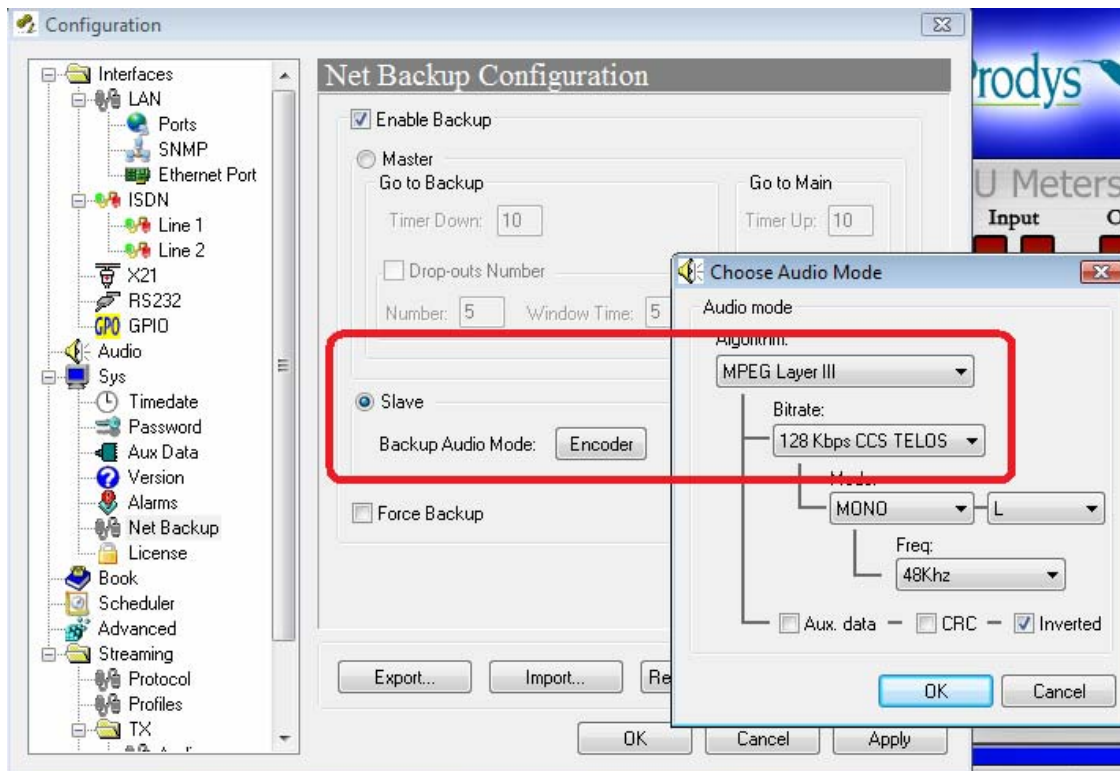
V.10 Device Backup function

Prodys equipment reliability meets strict customer's requirement. Even so, the <Device Backup> function is an option[32] aimed increase the service availability of the audio stream over IP according to an N+1 protection topology.

This function requires de deployment of one redundant Prodys IP Family codec[33] as a "hot-swap spare part" for one or several operational codecs sharing the same sub-network.

The codec enabled as <Device Backup> will remain assigned to this job up to disabling or an N+1 protection switch happens.

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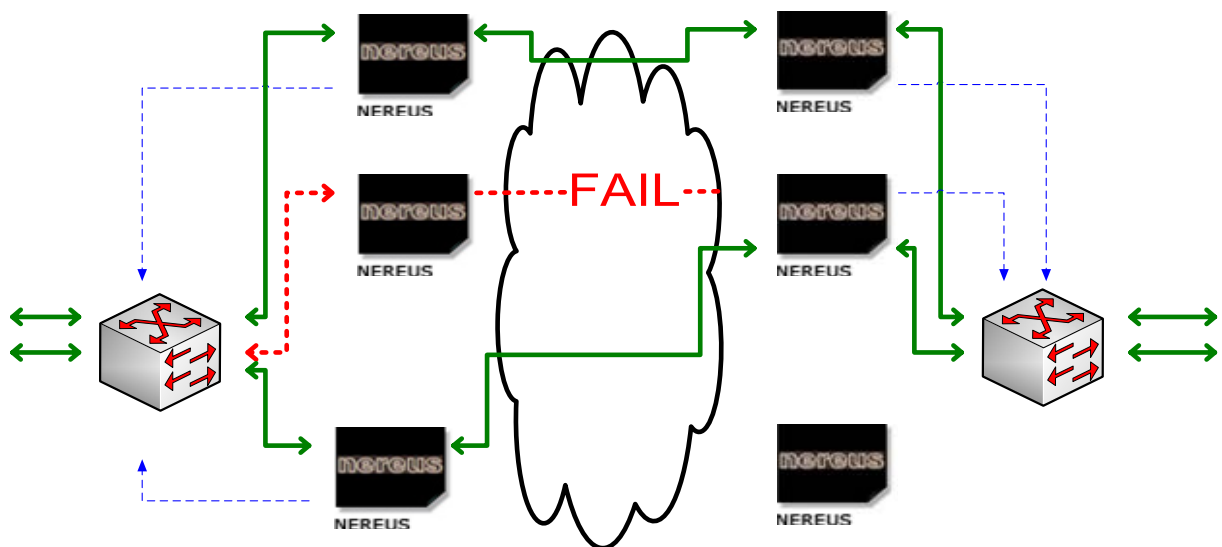
[32] The Device Backup function is an optional feature which has to be licensed by Prodys. The user is kindly requested to contact our sales@prodys.net for further information. Please refer to section IV.2.8 for licensing advanced features.

[33] Any Prodys IP Family codec could stand as a <Device Backup> for any other type of Prodys IP codec. This is, essentially a ProntoNet could backup any Nereus codec module or vice versa. Obviously any mismatch of the I/O should be taken into account.

Following parameters must be defined for the Device Backup codec:

- **<Enable Backup>:** Assignment of the current codec as Device Backup codec.

- **<Timer Down>:** A service outage of any supervised codec triggers a timer. This parameter defines de delay in seconds before taking over the affected service on the Device Backup codec.

- **<Drop-outs Number>:** This is a performance measurement for the decoded audio stream. The parameters <Number> and <Window Time> defines the amount of transmission errors allowed in a timeframe, before taking over the affected service on the Device Backup codec.
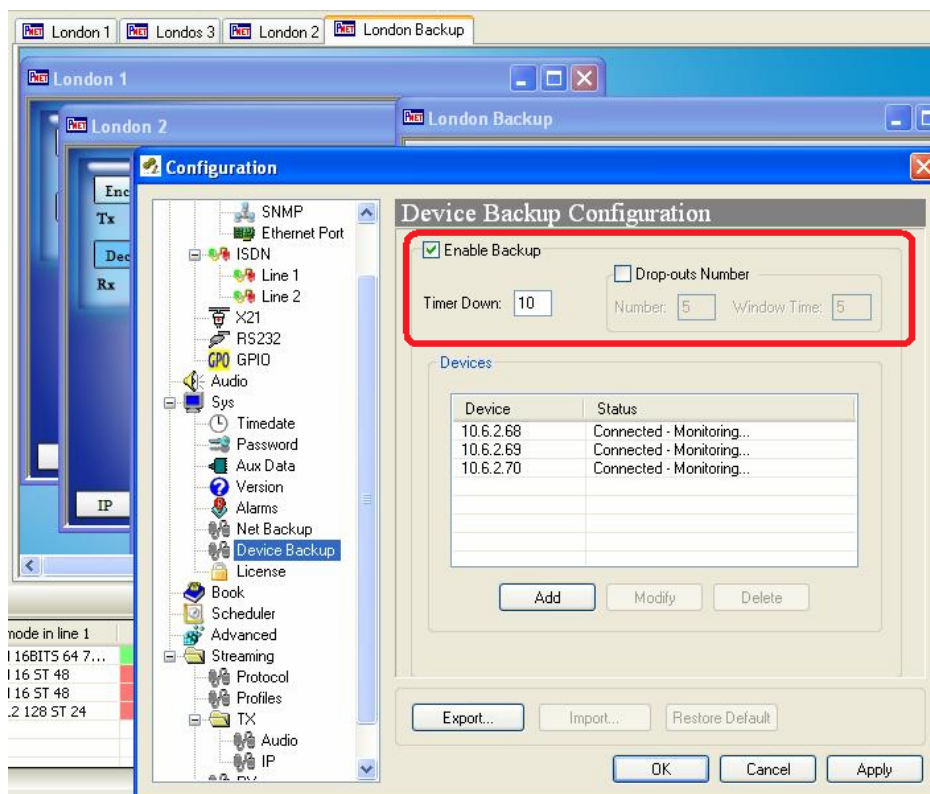
V.11 Addition of the codecs to be monitored

The Device Backup codec must share the same sub-network with the codecs to be monitored. The user has to define the IP Address for any of the codecs to be monitored at the Device Backup codec.

> Removal of any single codec from this monitor list is also possible; further, this operation is mandatory before powering down any monitored codec.

Both, <Manual> IP address typing or addition by means of the <Discover Devices> menu are possible.
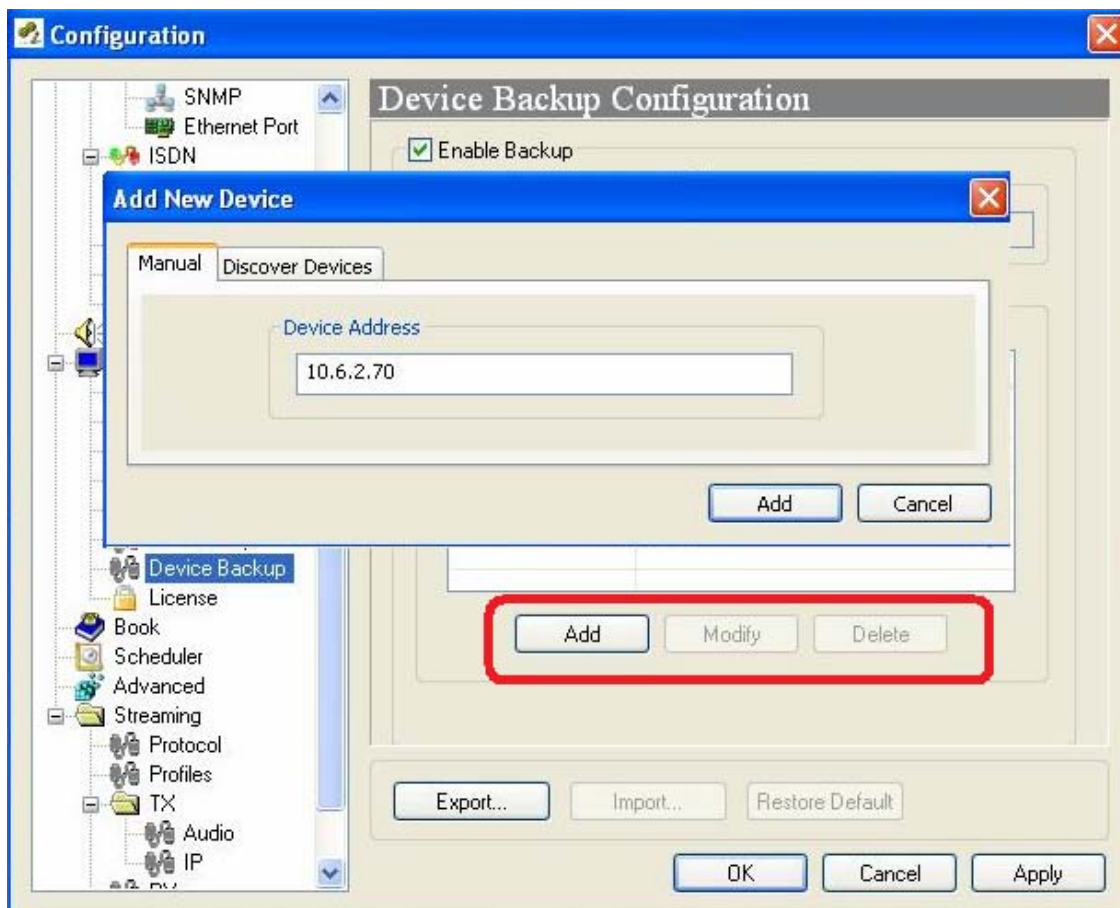
V.11.1 Protection switch N+1 by the Device Backup codec

After configuration the Device Backup codec remains checking periodically for sanity of operation for all codecs defined to be monitored. The Web Browser Control page for the Device Backup codec displays <Device Backup Supervisor>.

The Device Backup codec records also periodically the current configuration of the monitored codecs. In case one codec drops out of service all those service parameters are not lost.
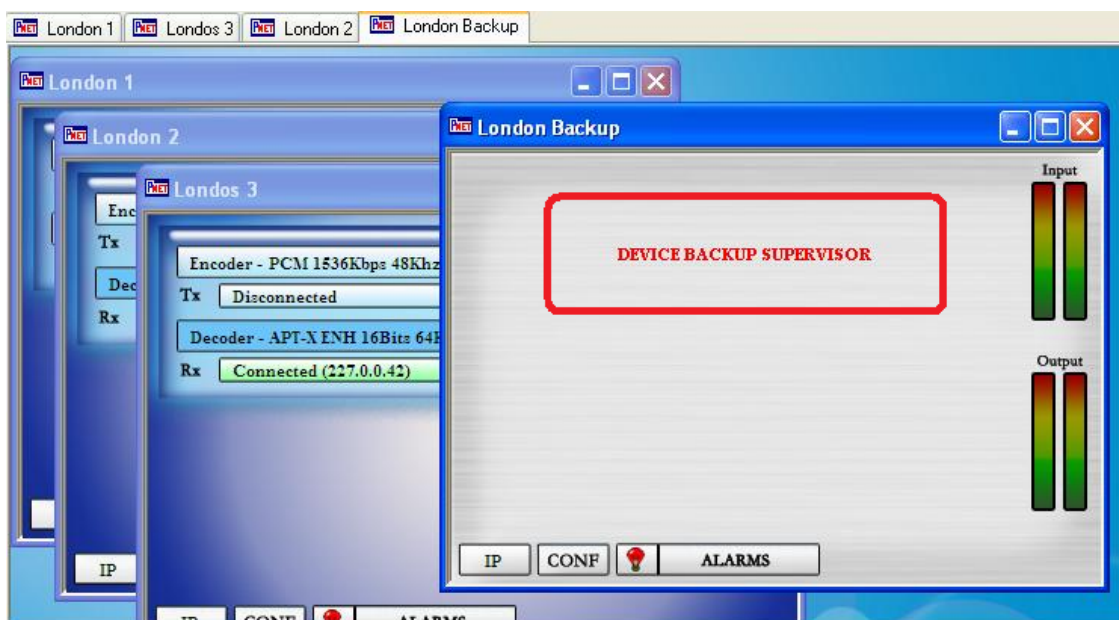
Upon the defined performance threshold is undergone by one of the monitored codec the Device Backup takes over the job. The Device Backup codec dials into the far-end codec of the former communication.

> It is remarked that the Device Backup codec keeps its IP address[34] even after a protection switch took place. Therefore it is unlikely that any <ping> to the codec out of service returns a successful acknowledge.

---

[34] Due restrictions of most network gears (routers and Ethernet switches) it is not a recommended policy to exchange the IP address and the MAC address for a short reaction time.
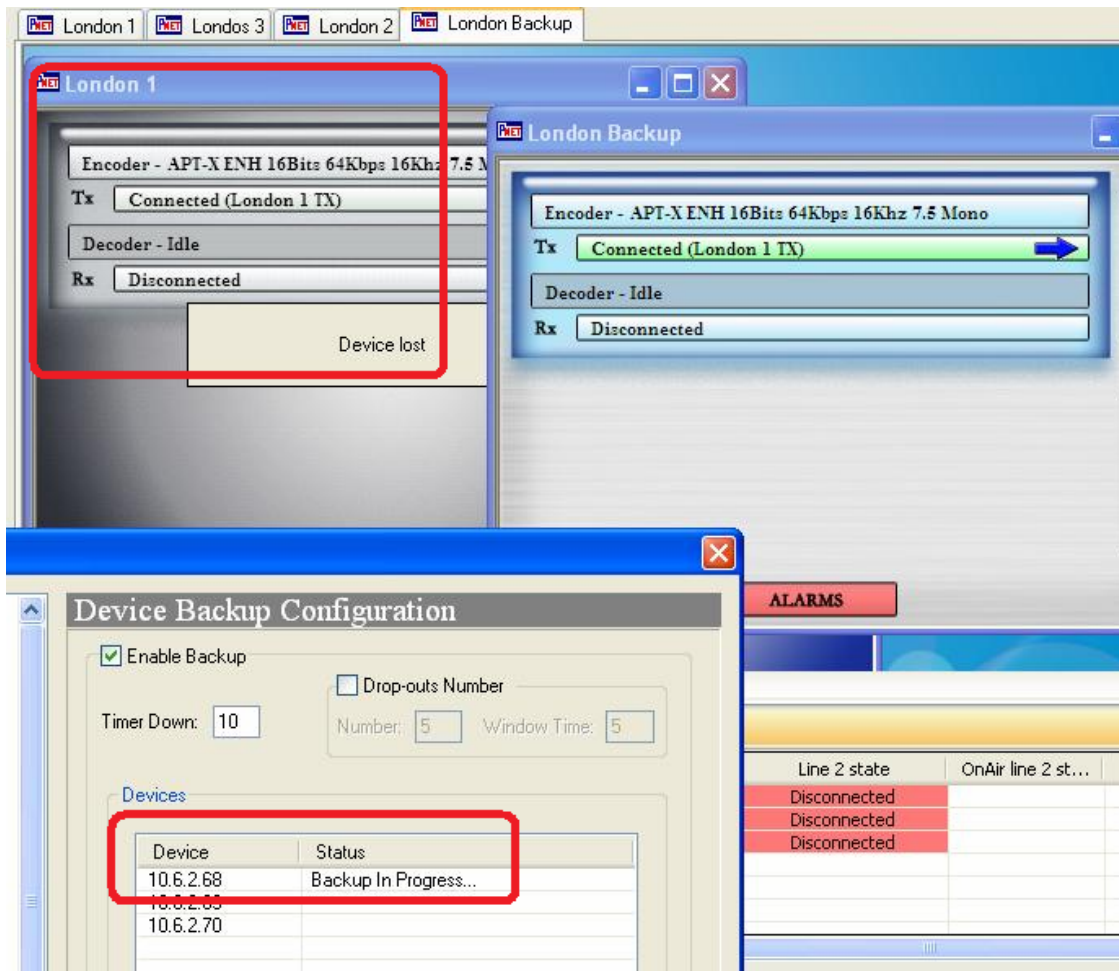
The Device Backup codec recovers automatically the idle condition by either replacing the codec out of service, the network performance threshold are met or by stopping the ongoing communication to the far-end codec.

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APPENDIX A: TECHNICAL SPECIFICATIONS

VI.1 Compression

- G722.
- G711 A/µ Law.
- MPEG 1,2 layer II (ISO/IEC 11172-3 /13818-3).
- MPEG 1,2 Layer III (ISO/IEC 11172-3 /13818-3).
- MPEG 2 AAC LC (ISO/IEC 13818-7).
- MPEG 4 AAC LC, LD & HE (ISO/IEC 14496-3).
- Standard and Enhanced aptX^{TM}.

MPEG decoding with Prodys' codecs has Error Concealment tools enabled. There are some scenarios where FEC mechanisms cannot be used due to bandwidth or delay restrictions. For those cases, a method to minimize the impact of packet losses in form of artifacts on the decoded audio might be used.

VI.1.1 BANDWIDTH (KHz)

Legend:
x = Not available in the standard.
#,*,- = Not implemented

- **MPEG 1,2 LAYER II (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	4	X	X	4,9	x	x	7,3	#	#
64	10,7	4	4,8	11,7	4,9	6,1	11,3	7,3	11,3
128	20	10,7	16,3	15	11,7	13,6	11,3	11,3	11,3
192	20	14,5	20	15	15	15	x	x	x
256	x	20	20	x	15	15	x	x	x
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	X	x	X
256	X	x	X
384	X	x	X

- **MPEG 1,2 LAYER III (ISO/IEC 11172-3 /13818-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	8,1	#	#	8,2	#	#	8,1	#	#
64	15,2	8,1	8,1	15	8,2	8,2	11,3	8,1	8,1
128	18,2	18,2	18,2	15	15	15	11,3	11,3	11,3
192	20	20	20	15	15	15	x	x	x
256	20	20	20	15	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
32	7,5	#	#
64	7,5	7,5	7,5
128	7,5	7,5	7,5
192	x	x	X
256	x	x	X
384	x	x	X

- **MPEG 2 AAC LC (ISO/IEC 13818-7)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	15,8	7,5	8,3	14	8	12	11,3	8,3	10,5
128	20	15,8	15,8	15	14	15	11,3	11,3	11,3
192	20	15,8	15,8	15	14	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

- **MPEG 4 AAC LC (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	18	7,5	8,3	15	8	12	11,3	8,3	10,5
128	20	18	20	15	15	15	11,3	11,3	11,3
192	20	18	20	15	15	15	x	11,3	11,3
256	20	20	20	x	15	15	x	11,3	11,3
384	x	20	20	x	15	15	x	x	x

Bit Rate	Fs=16 KHz		
	Mono	Stereo/Dual	JStereo
64	7,5	7	7
128	X	7,5	7,5
192	X	x	X
256	X	x	X
384	X	x	X

- **MPEG 4 AAC LD (ISO/IEC 14496-3)**

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	10,6	#	#	11,8	#	#	11,3	#	#
128	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
192	18,4	10,6	13,6	15	15	15	11,3	11,3	11,3
256	20	18,4	18,4	15	15	15	11,3	11,3	11,3
384	20	18,4	18,4	15	15	15	-	-	-

- **MPEG 4 AAC HE (ISO/IEC 14496-3)**

Bit Rate	Fs=24KHz			Fs = 16 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
24	12	#*	#	12,3	#	#
32	16,8	#	#	15	#	#
48	16,8	12	15	15	12,3	15
56	16,8	12	15	15	12,3	15
64	20	16,8	20	15	15	15
128	20	20	20	x	15	15

- **aptX^{TM} SD & ENH 16 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz		Fs=16 KHz	
	MN	ST/Dual	MN	ST/Dual	MN	ST/Dual
64	x	x	X	x	7.5	X
128	x	x	15	7.5	X	X
192	20	x	X	x	X	X
256	x	x	X	15	X	X
384	x	20	X	x	X	X

- **aptX ENH 20 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
160	x	x	15	x
320	x	x	X	15
240	20	x	X	X
480	x	20	X	X

- **aptX ENH 24 bits**

Bit Rate	Fs=48KHz		Fs = 32 KHz	
	MN	ST/Dual	MN	ST/Dual
192	x	x	15	x
384	x	x	X	15
288	20	x	X	X
576	x	20	X	X

- **G722**

Bit Rate	Fs=16KHz MN
64	7.6

- **G711**

Bit Rate	Fs=8KHz MN
64	3.8

VI.2 IP Protocols and compatibility

- DNS
- HTTP
- ICMP
- IGMPv2
- IPv4 /TCP /UDP :RFC 791, RFC 793, RFC 768, RFC 1112
- RIPv2
- RTP
    - o :RFC 3550, RFC 3551, RFC 2250, RFC 3119,
    - o :RFC 3190, RFC 4184, RFC 3555, RFC 3640
- SAPv1 :RFC 2974
- SDP :RFC 4566, RFC 3264 (EBU Standard Tech 3326)
- SIPv2 :RFC 3261
- SNMPv2
- SNTP
- FEC :RFC2733

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Appendix B

APPENDIX B: DISCONNECT CODES

VII.1 ISDN Disconnect Codes ITU Q.931(decimal value)

These disconnect codes are usually more meaningful for the codecs with a direct connection to the Public ISDN Network. It is noticed that private PBX usually filters the Q.931 disconnect code to the codec.

Code	Meaning	Code	Meaning
00	Normal Disconnect	38	Network out of order
01	Unassigned (Unallocated number)	39	Permanent frame mode connection out of service*
02	No route to specified transit network	40	Permanent frame mode connection operational*
03	No route to destination	41	Temporary failure
04	Send special information tone*	42	Switching equipment congestion
05	Misdialled trunk prefix*	43	Access information discarded
06	Channel unacceptable	44	Requested circuit/channel not available
07	Call awarded and being delivered in an established channel	46	Precedence call blocked*
08	Preemption*	47	Resource unavailable, unspecified
09	Preemption, circuit reserved for re-use*	49	Quality of service unavailable
16	Normal call clearing	50	Requested facility not subscribed
17	User busy	53	Outgoing calls barred within CUG*
18	No user responding	55	Incoming calls barred within CUG*
19	User alerting, no answer	57	Bearer capability not authorized
20	Subscriber absent*	58	Bearer capability not presently available
21	Call rejected	63	Service or option not available, unspecified
22	Number changed	65	Bearer capability not implemented
23	Redirection to new destination*	66	Channel type not implemented

25	Exchange routing error*	69	Requested facility not implemented
26	Non-selected user clearing	70	Only restricted information bearer capability available
27	Destination out of order	79	Service or option not available, unspecified
28	Invalid number format	81	Invalid call reference value
29	Facility rejected	82	Identified channel does not exist
30	Response to STATUS ENQUIRY	83	A suspended call exists, but this identity does not
31	Normal unspecified	84	Call identity in use
85	No call suspended	98	Message not compatible with call state, or not implemented
86	Call having the requested call identity has been cleared	99	Information Element non-existent or not implemented
87	User not a member of CUG*	100	Invalid Information Element contents
88	Incompatible destination	101	Message not compatible with call state
90	Non-existent CUG*	102	Recovery on time expiry
91	Invalid transit network selection	103	Parameter not implemented*
95	Invalid message, unspecified	111	Protocol Error, unspecified
96	Mandatory Information Element is missing	127	Interworking, unspecified
97	Message type non-existent or not implemented		

VII.2 IP Disconnect Codes Prodys Proprietary Protocols:

Cause Code	Meaning
0	Initial Status.
1	No answer from the remote unit.
2	Call rejected. The remote unit can be busy.
3	Normal Disconnect from the remote unit.
4	Normal Disconnect from the local unit.
5	G711 with AES /EBU not available
6	Different Protocols
7	Connection Dropped
8	Different Nets
9	Different Ports
10	Unable to resolve name

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VII.3 IP Disconnect Codes SIP Protocol:

Code	Meaning	Code	Meaning
100	Trying	416	Unsupported URI Scheme
180	Ringing	420	Bad Extension
181	Call Being Forwarded	421	Extension Required
182	Call Queued	423	Interval Too Brief
183	Session Progress	480	Temporarily Unavailable
200	OK	481	Call/Transaction Does Not Exist
202	Accepted	482	Loop Detected
300	Multiple Choices	483	Too Many Hops
301	Moved Permanently	484	Address Incomplete
302	Moved Temporarily	485	Ambiguous
305	Use Proxy	486	Busy Here
380	Alternative Service	487	Request Terminated
400	Bad Request	488	Not Acceptable Here
401	Unauthorized	491	Request Pending
402	Payment Required	493	Undecipherable
403	Forbidden	500	Server Internal Error
404	Not Found	501	Not Implemented
405	Method Not Allowed	502	Bad Gateway
406	Not Acceptable	503	Service Unavailable
407	Proxy Authentication Required	504	Server Time-Out
408	Request Timeout	505	Version Not Supported
409	Conflict	513	Message Too Large
410	Gone	600	Busy Everywhere
411	Length Required	603	Declined
413	Request Entity Too Large	604	Does Not Exist Anywhere
414	Request URI Too Long	605	Not Acceptable
415	Unsupported Media Type		

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APPENDIX C: IP PORTS DETAILED DESCRIPTION

There are two main groups of ports, Management and Streaming and Audio Control ports:

VIII.1 Management Ports

HeraFlash &
Prodys Control

UDP 50013: This port is used by ProdysControl Management Software and the HeraFlash upgrading application their communication with Prodys IP units.

TCP 50017: This is the port used by ProdysControl to monitor/control the unit from the PC running ProdysControl Client application.

In case of remote upgrading of remote control via ProdysControl, if a firewall is in between, these ports should be opened/forwarded to the corresponding IP codec.

Web Page

TCP 80: This port is used for the access to the web login page via HTTP (Internet Explorer). Once logged in, the OCX control web takes over and manages the connection.

TCP 50011: Dynamic Web Page (OCX). This is the port used in the communication between the PC running Internet Explorer and the IP codec, once the user has logged into the control web page.

In case of remote controlling the unit either from the web interface or the ProdysControl Management Software, and a firewall is in between the IP codec and the PC, these ports should be opened/forwarded.

Test Streaming Tool

TCP 50033
UDP 50033

Test Streaming Tool: This is an application built into the web page to measure the quality of the connection: bandwidth, delay, jitter, packet loss and packet disorder. In case of trying to measure these parameters between two units with firewalls in between, these ports should be opened/forwarded to the corresponding IP codecs.

External Protocol (P-XP)

TCP 50031: Control Port
TCP 50035: Status Port

The external protocol allows the user to build its own management application based on commands either via the serial port or via the IP interface. These commands are defined in the ProntoNet SDK User Manual available on the Prodys Web under the 'Downloads' section.

VIII.2 Streaming and Audio Control Ports

TCP 50019: P-RTCP (Control) (v1,v2): This port is used to control any IP audio streaming communication with Prodys Proprietary Protocols V1 or V2.

UDP 50021: P-RTP Unicast L1 (v1,v2): This port is used to carry the audio data over IP for unicast communications on line 1 with Prodys Proprietary Protocols V1 or V2.

UDP 50023: P-RTP Unicast L2 (v1,v2): This port is used to carry the audio data over IP for unicast communications on line 2 (only available in double IP mode) with Prodys Proprietary Protocols V1 or V2.

UDP 50025: PX-RTP Multicast L1 (v1,v2,v3): This port is used to carry the audio data over IP for unicast communications on line 1 with Prodys Proprietary Protocols V1, V2 or V3.

UDP 50027: PX-RTP Multicast L2 (v1,v2,v3): This port is used to carry the audio data over IP for unicast communications on line 2 with Prodys Proprietary Protocols V1, V2 or V3.

UDP 50037: P-AUXP Auxliary Data (from version 4.8.0 on) (v1,v2): This port is used to carry the Auxiliary Data in case the unit is configured to send this data outside the audio frame.

UDP 50039: P-UbP U-BITs (from version 4.8.0 on) (v1,v2,v3): This port is used to carry the User bit of the AES/EBU frame when the AES/EBU audio input with U-Bit is selected as audio input source.

TCP 51000: P-RTCP (Control) (default for V3): This port is used to control any IP audio streaming communication in Prodys Proprietary Protocol V3. This port can be disabled. In case this protocol is disabled, there will be a lack of control in the status of the connection at the far end.

UDP 51000: P-RTP Unicast L1 & Auxiliary Data (default for V3): This port is used to carry the audio data and auxiliary data over IP for unicast communications on line 1 in Prodys Proprietary Protocol V3. Both audio and auxiliary data are multiplexed into the same stream in Protocol V3.

UDP 51001: P-RTP Unicast L2 & Auxiliary Data (default for V3): This port is used to carry the audio data and auxiliary data over IP for unicast communications on line 2 in Prodys Proprietary Protocol V3[35]. Both audio and auxiliary data are multiplexed into the same stream in Protocol V3.

TCP 52000: P-RTCP (Control) (default for V4): This port is used to control any IP audio streaming communication in Prodys Proprietary Protocol V4. This port can be disabled. In case this protocol is disabled, there will be a lack of control in the status of the connection at the far end.

UDP 52000: P-RTP Unicast L1 & Auxiliary Data (default for V4): This port is used to carry the audio data and auxiliary data over IP for unicast communications on line 1 in Prodys Proprietary Protocol V4. Both audio and auxiliary data are multiplexed into the same stream in Protocol V4.

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[35] Line 2 over IP is only available in 'double mode'.

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UDP 52001: P-RTP Unicast L2 & Auxiliary Data (default for V4): This port is used to carry the audio data and auxiliary data over IP for unicast communications on line 2 in Prodys Proprietary Protocol V4. Both audio and auxiliary data are multiplexed into the same stream in Protocol V4.

UDP 5060: SIP port (SIP): This port is the default port defined by the SIP standard. This port carries the data for call flow control.

UDP 5004: RTP port (SIP, RTP): This port is the default port defined by the RTP standard. It carries the audio data over IP for unicast communications when SIP or RTP are selected as Streaming Protocols.

Appendix D

APPENDIX D: UPDATING THE FIRMWARE

From version 5.2.1 and by using HeraFlashPlus version 2.5.0 or higher, the upgrading can be performed over WAN networks, such as Internet.

From version 5.4.1 onwards, all Prodys IP codecs will share the same firmware file, making it more easier to upgrade several different units to the latest firmware version.

To upgrade a unit with a firmware version lower than 4.7.1, please upgrade the unit with the latest version of HeraFlashPlus to the firmware version 5.4.1, and then upgrade again from version 5.4.1 to the final one. Otherwise, the unit will not upgrade properly.
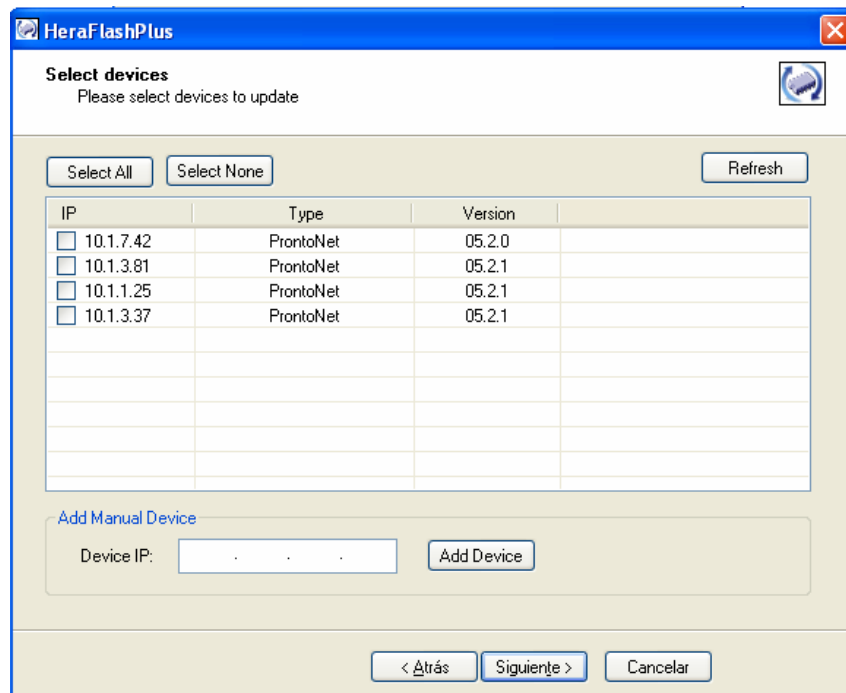
To update the firmware of a defined unit follow these steps:

1. Start up the HERAFlashPlus application.

2. Click on browse and select the file with the new firmware version.

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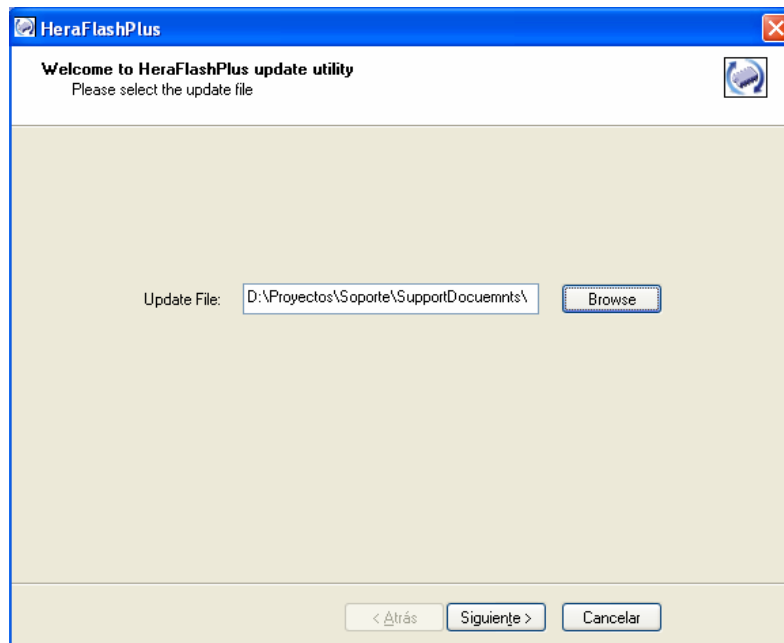
3. All the units connected to the computer's network will be detected automatically. Select which units will be upgraded.

- Press next to start the update process. You will see a warning message telling you that all the contents of the Flash will be erased and that you must back-up this data BEFORE updating the device.

- By clicking on next, the upgrading process will start.
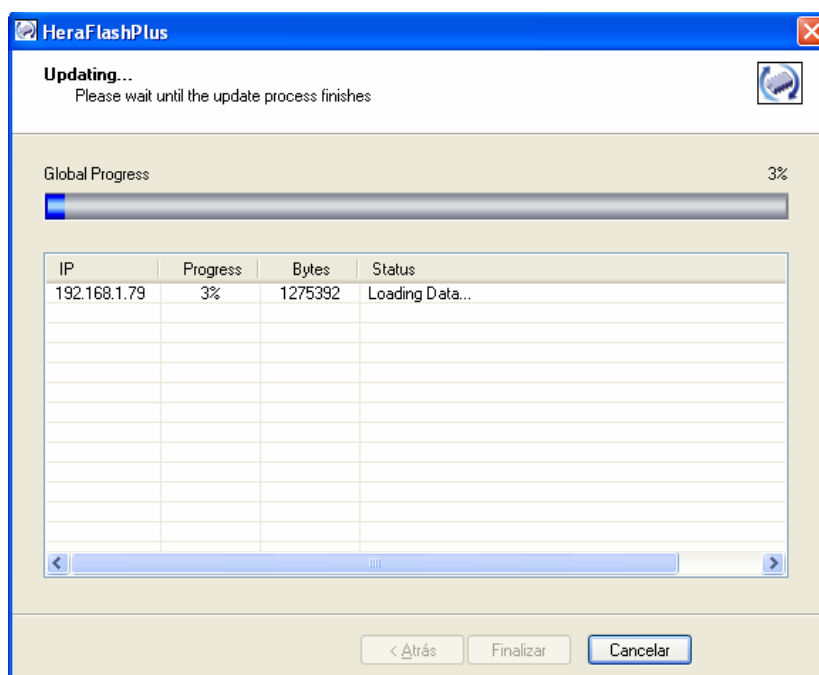
The update process will begin, the Flash will be erased and a progress bar will tell you more info and alert you when it is finished.
- You are able to abort the update process until the recording process begins. At this point, the cancel button is disabled. In any case, even if a power failure or a similar failure takes place, it is always possible to return to start the recording process again.

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- If everything has gone correctly, at the end of the recording, you will be shown the following message: