User Manual

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PROCESAMIENTO DIGITAL Y SISTEMAS, S.L

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Chapter I

INTRODUCTION

Quantum is a powerful new-generation IP audio codec family that extends the technology introduced in the existing Prodys Prontonet codec family.

Quantum inherits also world-class IP technology from the Prodys Ikusnet Video Codec Family: automatic jitter buffer, adaptive bit rate, file recording and uploading and **bonding of up to 7 IP links** (Ethernet, 3G/4G, Wi-Fi) for secure audio streaming with the best possible audio quality.

A **Wireless Communication Module** is available with up to **four 3G/4G data links with bonding capabilities**. There are internal and external modules for better reception and versatile installation options.

In this manual, all the aspects regarding the functions, configuration parameters and operation modes of the Quantum family of codecs are fully explained.

I.1 About this manual

The **User Manual** is applicable to most of the common features provided by the Quantum family of codecs. For the specific aspects of one particular member of the family, please, refer to the specific model's hardware manual.

I.2 Contents

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

- **Chapter I – Introduction**
  Contents of the Manual.

- **Chapter II – What is Quantum?**
  This chapter is a brief summary of Quantum, its features and applications

- **Chapter III – Basic Management**
  This chapter is a quick guide aimed at the new users, so they can establish its first communications in a very simple way.

- **Chapter IV – Functional description**
  This unit can be controlled from its built-in web graphical interface, its front panel display menu, or from the ProdysControlPlus application. This chapter describes, in a comprehensive way, how the unit can be configured using either controlling interface.

- **Chapter V – Technical specifications**
  This appendix describes the technical specifications common to all devices of the Quantum Family.

- **Chapter VI – Updating the firmware**
  This appendix describes how to update the Quantum codec firmware.

- **Chapter VII – Detailed port description**

Chapter II

WHAT IS QUANTUM?

II.1 Introduction

The **Quantum** family of codecs is based on an FPGA/DSP platform, fully designed by Prodys to obtain the best performance in audio mixing, compression and transmission.

Quantum family of codecs comprises the following devices:

- Quantum XL
- V-Quantum
- Quantum W
- Quantum Lite
- Quantum ST
- QLST
- Quantum 3U
- Quantum One

In the following sections we will take a look at them.

II.2 Quantum XL

Quantum XL is the only portable audio codec in the market capable of mixing up to 5 mic/line mono inputs + 1 USB audio interface, and encoding and streaming 2 x stereo audio programs simultaneously.

Quantum XL has also the ability to record audio files for later transmission, and to send/receive files via FTP. When sending files, the unit supports fast FTP by aggregating the bandwidth of up to 7 different IP interfaces (four 3G/4G, one Wi-Fi and two Ethernet ports), thus making the file transfer much faster than a regular FTP transmission.

II.3 V-Quantum

V-Quantum is a device designed for audio contributions in different environments. It has two modes of operation:

- Commentary unit, sending its audio input to on-site infrastructure like OB-Vans or IBCs using audio-over-IP Dante or AES67 protocols.
- Codec mode, acting as an audio mixer (4 mic/line mono inputs + 1 USB audio interface) and transmitting two stereo audio programs directly to studio using IP networks.

V-Quantum also implements an H.264 video decoder.

II.4 Quantum W

Quantum W is a smaller version of Quantum Portable with 3 microphone/line inputs instead of 5 but with all the power of its bigger brother.

II.5 Quantum Lite

Quantum Lite is Prodys's handheld audio codec for professionals. This device enables extreme reliable field audio contributions over wireless networks by means of concurrent use of two UMTS /LTE network connections.

Quantum Lite can combine up to 4 uplink IP connections: 2x UMTS/LTE, 1x Ethernet & 1x Wi-Fi (optionally) for any scenario.

This device turns also off-line jobs in easy

tasks with embedded audio recording and transfer.

II.6 Quantum ST

Besides the portable versions of Quantum, there is a 1U rack mount model called Quantum ST whose main application is to receive, decode and play audio streams from remote Quantum units. But Quantum ST is not just a decoder, it is also capable of encoding audio back to the portable unit and provide audio feedback to the remote unit.

In addition to this, Quantum ST Duo version is also able to establish a second bidirectional stereo audio channel between the studio site and a remote Quantum codec.

This unit can be controlled from its built-in web graphical interface, its front panel display menu, or from the ProdysControlPlus application.

II.7 QLST

QLST is the ideal studio mate for the handheld Quantum Lite product.
QLST is a small size, single stereo codec with analogue or AES/EBU inputs and outputs.

II.8 Quantum 3U and Quantum One

Quantum 3U and Quantum One are the modular IP multicodec solutions with 14 codecs in a 3U chassis or 4 codecs in a 1U chassis. These multicodecs can also use Dante/AES67 for their audio inputs and outputs.

II.9 Quantum Family modes of operation

These are Quantum's four main modes of operation:

- **Real Time Ethernet transmission.** When a wired IP connection is available via an Ethernet connector, a bidirectional audio communication can be established between two Quantum units using a wired IP link.

  In the models with two Ethernet ports, it is also possible to use both for the audio transmission. This can be done in two ways:
  - Bonding mode, splitting up the bit-stream into 2 bit-streams. Those 2 sub-streams will be multiplexed at the receiving end by another Quantum codec.
  - Diversity, sending the same bit-stream through the two Ethernet ports.

- **Real Time 3G/4G transmission.** When no wired IP connection is available, a bidirectional audio communication between a portable Quantum unit and a rack Quantum unit can still be established using the wireless 3G/4G communication module. Quantum is able to aggregate bandwidth from up to four 3G/4G data links, dividing up (demux) the bitstream into N sub-streams that will be bonded (multiplexed) at Quantum receiving unit. Using this technology, the bandwidth provided by the mobile network operators can be widely enlarged. The Diversity approach is also possible using 3G/4G connections.

- **Wi-Fi connectivity:** When Wi-Fi networks are available, and the unit is equipped with the Wi-Fi module, one Wi-Fi network can be used for audio streaming. Take into account that the main purpose of the Wi-Fi link is not to act as a primary link, but to reinforce the other available links like 3G/4G or Ethernet interfaces.

- **Real Time Custom transmission.** When Ethernet, 3G/4G and Wi-Fi connections are available, a combination of up to 7 different interfaces can be used to stream the audio as well as to upload pre-recorded files. This technology allows the user to get as much bandwidth as possible by aggregating the capacity of each of the available links. This is done by a proprietary statistical algorithm that evaluates in real time the capacity of each communication link and balance the bandwidth usage of each interface based on that information. Diversity is also possible with any combination of different IP interfaces.

Chapter III

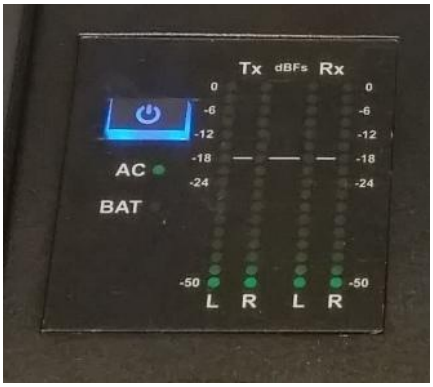
BASIC MANAGEMENT

This chapter is aimed at describing to new users how to start an audio connection over IP between two Quantum codecs. The usual application would be the connection of one portable Quantum codec to one rack mount Quantum ST, but any combination is possible. The following chapter explains how to connect one Quantum XL to a Quantum ST.

All screenshots in this chapter belong to the menu available for the touch panel in both Quantum XL and Quantum ST units, but they can be extended to the embedded web control page, given that the menu structure is exactly the same.

III.1 Turning on the units

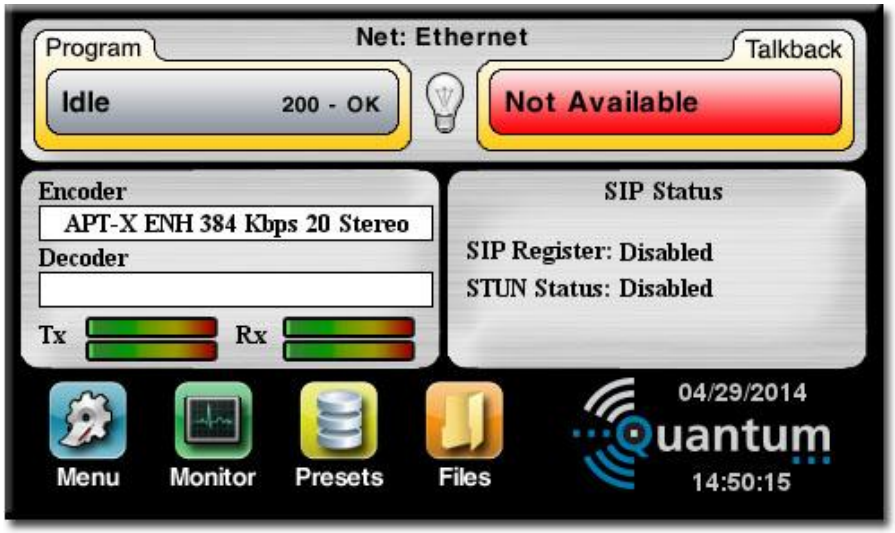
Quantum XL can be powered by its auxiliary external battery or can be connected to the external power adapter supplied with the unit. Quantum ST can be connected directly to any standard 110/220VDC power supply as it comes equipped with an internal Power Supply Unit (PSU).



Quantum XL has a button on the left side of the touch panel, near the VU-meters, that must be pressed for a few seconds for the unit to start. The same procedure must be followed to turn the unit off.

III.2 Main menu

Once the units have been turned on, the booting sequence will begin. The units will show the main menu on the touch panel screen or in the web browser similar to the one shown in the picture below:



NOTE: Take into account that the web control menu of Quantum XL codec is exactly the same as the touch panel menu, and the web interface on Quantum ST share most of the structure of Quantum XL codec so, from now on, on this chapter, we are will focus on showing the menu options for the web interface of the Quantum XL codec.

III.3 Control Web Interface

Prodys Quantum codecs can be controlled remotely using a 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.

Don't forget that it is necessary to set the IP addresses of the computer on which the web browser is running and of the codec to be controlled in the same IP network.

The control webpage is based in Html5 and can be open with any modern browser. In the following table the first operative version of the most common browser is listed:

Web Browser	Version	Release Date
Internet Explorer	10	Sep. 2012
Edge	12	Jul. 2015
Firefox	11	Jan. 2012
Chrome	16	Oct. 2011
Safari	7	Oct. 2013
Opera	12.1	Nov. 2012
Safari for IOS	6.1	Jan. 2013
Android Browser	4.4	Dec. 2013
Blackberry Browser	10	Jan.2013
Opera Mobile	12.1	Oct.2012
Chrome for Android	57	Mar. 2017
Firefox for Android	52	Mar. 2017
IE Mobile	10	Jun. 2012
Samsung Internet	4	Apr. 2016

To access a Quantum codec from the browser enter the IP address of the unit in the address bar and the following window will appear:

The user has to enter the username and password settings to get access to the main web page. The factory default configuration defines one user whose name is 'Administrator' with no password associated and with full access to all configuration settings and menus.

Quantum communicates to the web browser via port **TCP 80** to transmit/receive real time information (it is a dynamic web page).

When the user enters the main web page by entering a user name and a password, the following window will appear:

Quantum web page is arranged in three main areas:

- General Configuration area. ❶
- Control area. ❷
- Monitor area. ❸

III.4 Setting the IP parameters

We will split this chapter in two different types of interfaces: Wired (Ethernet) and Wireless (3G/4G). Wi-Fi can also be used as a streaming interface but it will be described in a different chapter.

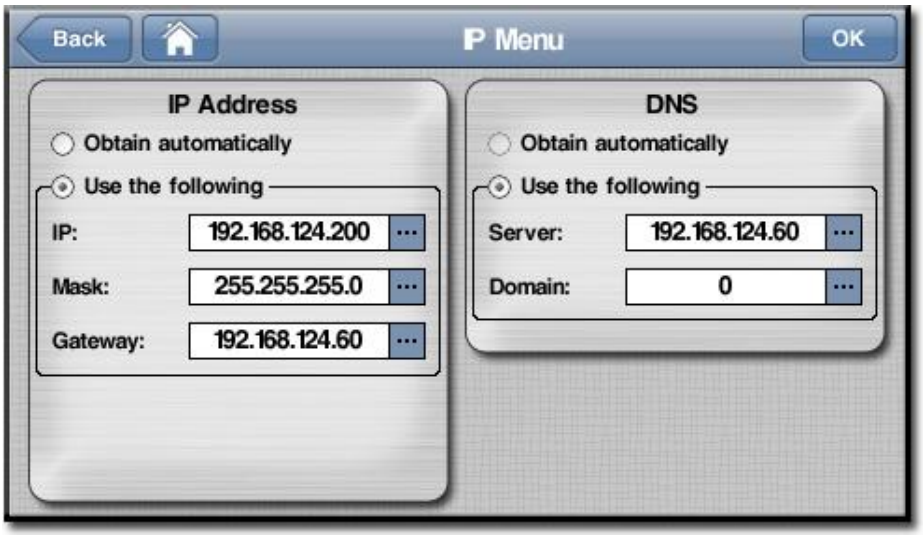
III.4.1 Wired IP connection (Ethernet)

All Quantum units (except Quantum Lite) come equipped with two Ethernet interfaces. Each Ethernet interface is fully TCP/UDP/IP standard, and can be connected to any standard IP network, be it local or part of a wide area network, such as the Internet.

First of all, the customer will have to set up all IP parameters. These settings will depend on the network topology. The assistance from the IT personnel might be necessary in some advanced networks.

To configure the IP parameters, click on the following icons of the front panel menu or the web page:

The following configuration window will appear:



There are two different options: Either to get the IP address automatically from a DHCP server when available, or to set up these IP parameters manually. In the example above, we are configuring them manually. Optionally, the DNS protocol can be configured. It allows using domain names instead of IP addresses when establishing calls.

NOTE: When the unit is configured to get the IP address automatically from a DHCP server, if no DHCP server is available in the network, the unit will try for a while to get it, indicating the message 'Negotiating DHCP' on the Program and TalkBack connection bars on the main menu page and, after that, the default IP address and netmask will be assigned automatically: 192.168.100.100 / 255.255.255.0.

LAN1: RJ-45 connector on the rear side; 10/100/1000BT Ethernet interface. Default interface for streaming and remote control.

This interface is always enabled and it is used for transmission and reception of control data (Web control, SDK remote control protocol, and ProdysControlPlus), regardless of the settings for the Streaming Network option (be it Wi-Fi, Ethernet, Custom…).

When 'Custom' is selected at the streaming network configuration menu, LAN1 can be selected as IP interface for bonding to other IP interfaces and thus to transmit and receive audio data. In this case, LAN1 will carry both audio and control data simultaneously.

When LAN2 interface is disabled and 'Ethernet' is selected as the streaming network configuration menu, LAN1 will transmit and receive all audio data for both the main program and the talkback audio streams. But, when LAN2 is enabled, and 'Ethernet' is selected as the streaming network configuration menu, LAN1 interface will only transmit control data: web control, ProdysControlPlus or any other IP data not related to the audio streaming.

LAN2: RJ-45 connector on the rear side; 10/100/1000BT Ethernet interface. It can be enabled or disabled by the user.

This interface will carry the streaming data (not control data) if the LAN2 interface is enabled in the configuration. When enabled it is available for bonding or diversity with other IP interfaces to stream audio data.

III.4.2 Wireless IP connection (3G/4G)

Quantum codecs can be equipped with internal 3G/4G modems, with external USB modems or with a Wireless communications module supporting up to 4 internal SIM card slots with an internal antenna system specifically designed to improve signal reception. Quantum units will aggregate the bandwidth from the different available wireless connections (bonding) or use them for transmitting the same stream in all of them (diversity).

The bonding mode allows the user to get as much bandwidth as possible by aggregating the capacity of each of the available links. This is done by a proprietary statistical algorithm that evaluates in real time the capacity of each

communication link and balance the bandwidth usage of each interface based on that information.

The wireless communications module is composed of a main box (it holds 4 3G/4G modems and the antenna system), the SIM module that holds 4 SIM card slots to insert the different SIM data cards, and the cable and connector that allows the user to hook this module up to the Quantum unit. The connector for this module is on the rear panel of the unit.

Before connecting the Wireless Communications module to the Quantum Portable device, the user must insert those SIM cards available for data connections into the SIM slots.

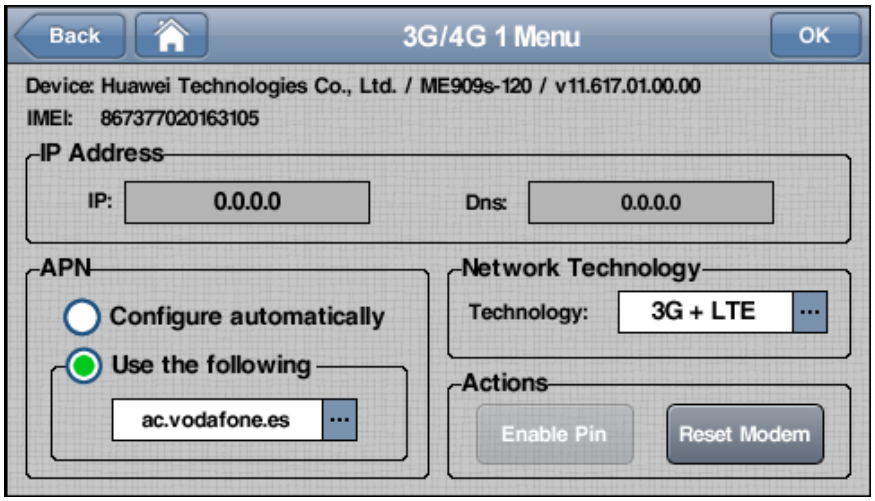
Each SIM slot in the SIM module has been assigned a number. That number must be taken into account when configuring each of the different 3G/4G data connections (APN, user and password) so that the SIM cards and the different configuration parameters match. Once the SIM cards have been inserted into the SIM slots, this module should be connected to the unit on its rear panel.

As soon as the user connects the module to the unit, it will be detected automatically and Quantum will ask for the PIN number of each of the available SIM cards, on either the web interface or the touch panel menu. Once the PIN number is entered, it will be checked and kept in memory till the 3G/4G module is disconnected or the unit is restarted.

Once the internal modems are up and running and, in order for the 3G/4G data connection to be used by the unit, the APN (Access Point Name), user and password settings for each mobile network operator being used must be configured by the user. To do that, click on the following icons of the front panel menu or the web page:

The configuration window similar to this will appear:

Just click on the corresponding icon to enter the APN configuration window:



In this configuration window the user has to set up the APN (Access Point Name) information, corresponding to the provider of the IP data connection for the 3G/4G modem under configuration. They can choose between setting the APN manually or to let the codec to look for the right APN in their database. This option can provide undesired results when APN values have been changed recently by the operator and the firmware has not been updated since, so it is always advisable to use the manual setting. The APN settings are widely available on the Internet for each 3G/4G data provider.

It is also possible to Enable or Disable the Pin number, and to reset the modem associated to that SIM card.

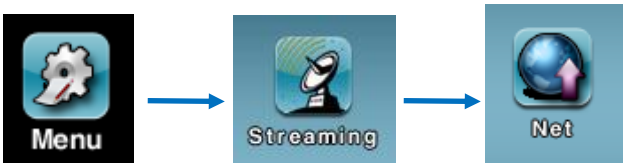
To end with and regarding the configuration of the 3G/4G connectivity, it is possible to configure the wireless link to make use of 3G + 4G (LTE) technology, or just limit the connectivity to one specific technology (either 3G or 4G/LTE).

Take into account that the step above must be followed for all internal modems that have been provided with a SIM card in its corresponding SIM card slot. Detailed information on the status of the different 3G/4G connections can be found at the 3G/4G section in the monitor menu.

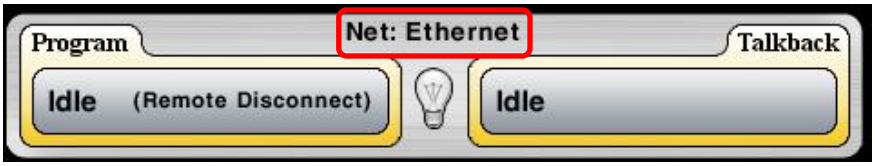
III.5 Selecting the interface

Quantum units support different data interfaces to transmit and receive audio. Before making the IP connection the user has to select what data interface will be used for the data connection: Ethernet (wired), 3G/4G (wireless), Wi-Fi (wireless) or custom (a combination of any). In this chapter we will only explain how to set the 3G/4G or the Ethernet interfaces to stream audio.

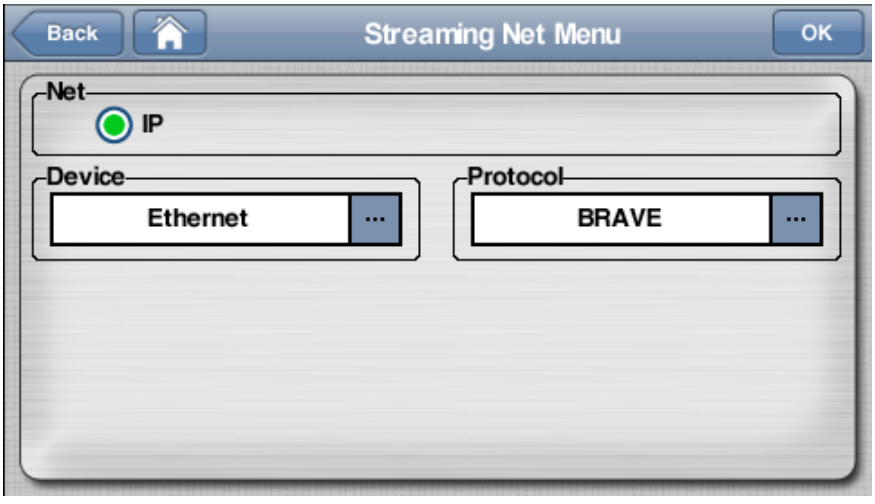
To select 3G/4G as streaming interface, please click on the following icons on the main menu:



Or use the shortcut:



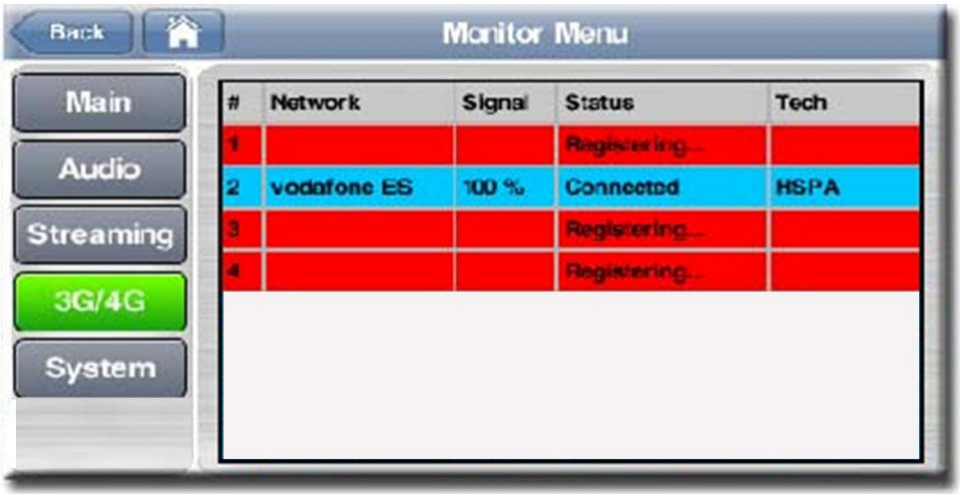
In the Streaming Net Menu, check the desired interface:



The transmission protocol can be selected between BRAVE, Prodys proprietary protocol allowing bandwidth aggregation and error recovery, MPEG-TS, ProntoNet V4 for compatibility with Prodys ProntoNet family codecs, SIP for compatibility with third party codecs and RTMP for connection with Internet audio servers.

IMPORTANT: SIP, ProntoNet V4, MPEG-TS and RTMP protocols are provided under license. They might not be availablefor some Quantum models.

If the selected interface is 3G/4G, please take a look at the 3G/4G Monitor Screen to check how many 3G/4G connections have been detected and are ready to be used. If the line corresponding to a SIM card does not show a blue colour, that connection will not be available for streaming. It is also important to note that the technology detected by the system for that link will determine the capacity of that connection.



Take into account that when a row shows a red warning line, most likely the reason why it is not connected is because the APN, user and password settings have not been properly configured for that provider on that SIM card, or that SIM card has not been enabled with a proper data plan to access the Internet.

NOTE: When Ethernet is selected as streaming interface, by default the Ethernet port labelled as LAN1 will be used for control and management whilst the secondary Ethernet port labelled as LAN2 will be used just for audio streaming. This configuration can be changed to use only LAN1 for control and streaming.

III.6 Establishing the IP connection

Once the IP settings, be it Ethernet or 3G/4G, have been properly configured, and the Program connection bar shows the word 'IDLE', the unit is ready to establish a call.
Please pay attention if LAN1 or LAN2 must be related to the proper Internet gateway according to LAN2 being enabled or not.

In order for the user to be able to establish an IP connection, the Program connection bar should show the word 'Idle', meaning that the streaming interface is ready to be used. Otherwise, the message 'Down' would appear in this bar:

When working with 3G/4G as streaming network, or when bonding different interfaces, there will be some circles representing each of the different interfaces as in the picture below:

One red circle means that the corresponding 3G/4G data link is not up and running. Grey means that the connection is fine. As long as one of them is up and running, the connection bars will show 'Idle', allowing the user to establish a connection. Otherwise, the connection bars will show the word 'Down' in a red background colour.

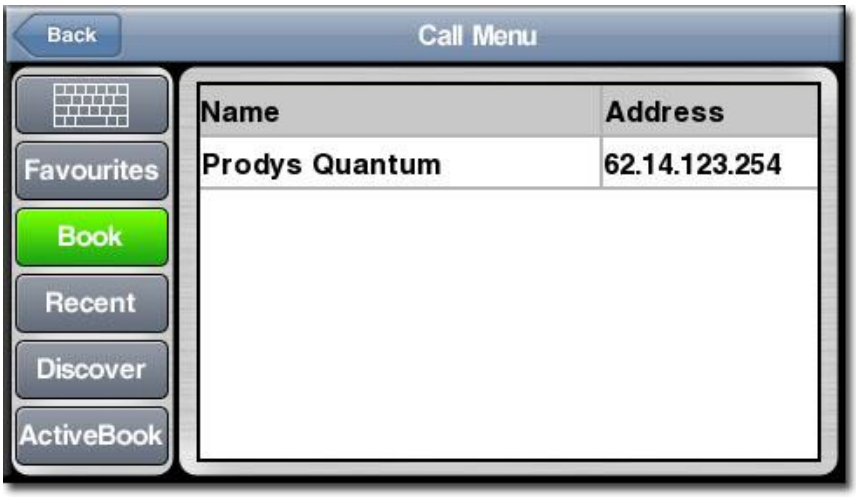
Using BRAVE transmission protocol Quantum units with double codec capacity can service two independent bidirectional stereo communications:

1. PROGRAM or CODEC 1
2. TALKBACK or CODEC 2

To establish the connection on the Program or Talkbalk line, just click on the corresponding connection bar. The dial window will appear as follows:

Just enter the destination IP address, the number of codec (1 or 2 if you are calling to a double codec), the type of call and the connection Profile and press the 'Call' button.

The destination address can be dialed manually pressing the "keyboard" box or can be chosen from the phone book, from the recently dialed addresses or from those units automatically discovered by Quantum in the same network. By pressing at the "Book" box in the Dial Window, the following window will be shown:



The user can either select one entry of the Favourites, of the whole Phone book or press on 'Recent' button to select the destination from a list of previous destinations. Click on 'Discover' to get a list of all compatible units automatically detected in the same local area network.

Once the destination has been either manually entered or chosen from any of the available lists, just click on the 'OK' button and the 'Call' button at the main Dialing Window and the connection will be established.

If the connection gets through, the connection bar for the connected line will light on green and will show the word 'Connected' and the duration of the connection:

This indicates that the Program line has been established and it is connected to the far end.

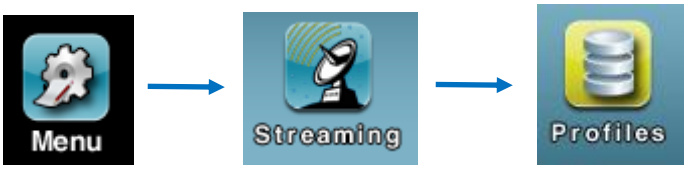
When establishing a connection over one or several wireless connections, the unit will make a call in all the IP interfaces, meaning that the user does not have to make a call for each data link, but just only one for all of them. As it can be seen below, the connection bar will show with a small circle in green what 3G/4G connections are properly connected when it is on green color.

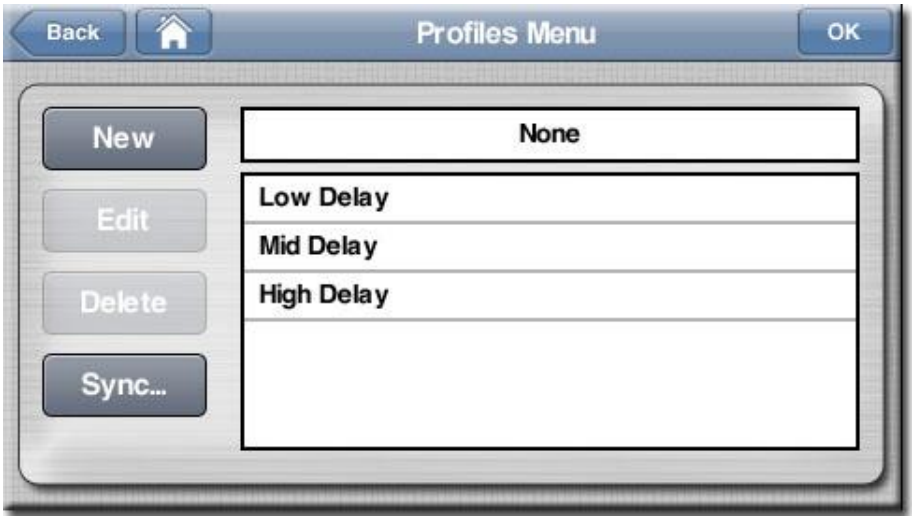
If the user attempts to establish a call and it does not go through, a brief description of the reason will be shown on the connection bar:

III.7 Setting a profile

The connection profiles can be configured by the user, and they define several different parameters for the transmission (the most important is the encoder algorithm) and some parameters that will be applied at the receiving end, being the most important the length of the jitter buffer (i.e. the delay). The longer the buffer, more protection against packet losses and jitter but longer delay. The shorter the buffer, less protection but shorter delay. The key point in being able to adjust this parameter from the calling end is that most likely the user who really knows what the capacity for the current IP link being used for streaming is will be the user that is establishing the call. If no profiles are selected, the buffer at the receiver end will keep its current configuration.

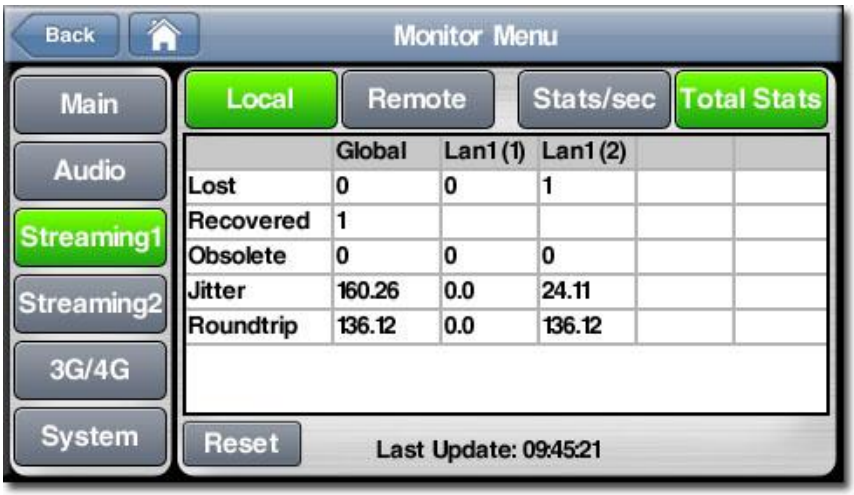
To enter the Profiles menu just click on the Profiles button at the main streaming menu:





III.8 Monitoring the status of the connection

Once the connection has been established successfully, the user can monitor the performance of the link by means of the Real Time Network Analyzer on the Streaming Tab of the Monitor Section:



This screen allows the user to access in real time and during the connection to some crucial parameters of the IP network performance, such as lost, recovered and obsolete packets, jitter, roundtrip... This information is offered for each interface involved in the communication and for the local and the remote terminal, so the user can monitor the remote end reception as well.

III.9 Finishing the connection

The communication might be stopped at any time by clicking the corresponding connection bar, either the Program or the Talkback bar.

Chapter IV

FUNCTIONAL DESCRIPTION

IV.1 Control Area. Communications

Thanks to BRAVE protocol, among other benefits, most Quantum units can service two independent bidirectional IP audio communications:

1. PROGRAM or CODEC 1
2. TALKBACK or CODEC 2

Each communication is operated and displays its actual status by means of a touch button on the control area of the main screen.

Possible communications statuses are:

- Disconnected *(channel idle; ready for next connection)*
- Connecting *(establishing the connection)*
- Connected *(streaming is ongoing)*
- Reconnecting *(resuming the connection)*
- Down *(Ethernet or 3G/4G modem not detected)*
- Negotiating DHCP *(acquiring IP address)*

In 3G/4G and Custom modes, the communication bars show a series of small circles, one for each of the participant interfaces in the communication. These circles are grey when the interface is detected but the link is disconnected; green when the link is active and red when the link is down. When a circle blinks is because communication is being established in that interface:

To establish the connection on the Program or Talkback line, just click on the corresponding connection bar. The dial window will appear as follows:

Just enter the destination IP address, the number of codec (1 or 2 if you are calling to a double codec), the type of call and the connection Profile and press the 'Call' button.

There are several types of call:

- **Two Way Symmetrical**: Bidirectional. Same connection profile in both directions.
- **Two Way Asymmetrical**: Bidirectional. Two different connection profiles, one per direction.
- **Tx One Way**: Transmission call. The connection profile sets the local encoder algorithm and the remote codec's reception buffer.
- **Rx One Way**: Reception call. The connection profile sets the remote encoder algorithm and the local codec's reception buffer.

IV.2 Configuration Menu

To access all main configuration menu click on the Menu icon on the left-bottom part of the main screen.

The configuration menu offers the following items:

- **Audio**: encoding parameters, audio input and output configuration.
- **Streaming**: network and transmission/reception parameters.
- **Interfaces**: IP configuration for the different IP interfaces.
- **Book**: phone book edition.
- **Users**: definition and configuration.
- **System**: system parameters.
- **Logout**: to return to login window.

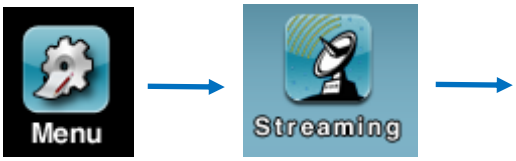
IV.2.1 Audio options menu

As the different codecs of the Quantum family have very different audio configurations, this menu is explained in the specific manual of each Quantum codec:

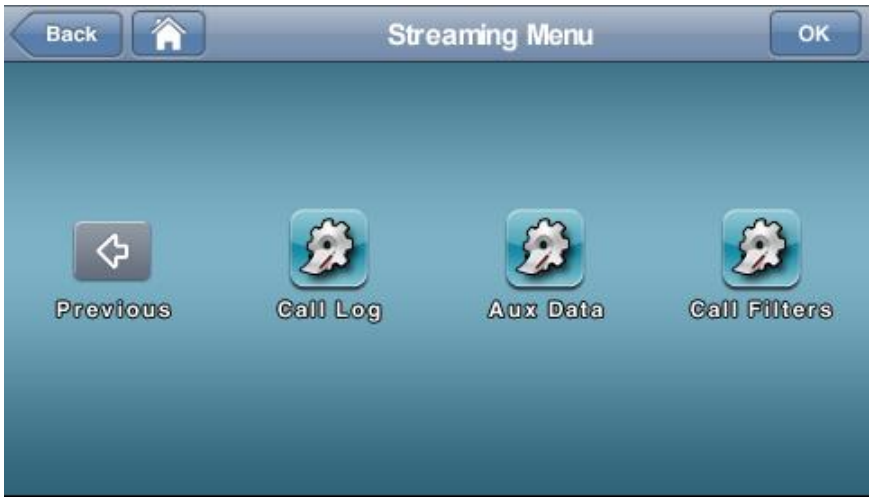
- "Quantum XL Hw and Reference Manual Rev x.x"
- "Quantum W Hw and Reference Manual Rev x.x"
- "Quantum ST Hw and Reference Manual Rev x.x"
- "Quantum Lite Hw and Reference Manual Rev x.x"
- "QLST Hw and Reference Manual Rev x.x"
- "Quantum 3U Hw Manual Rev x.x"

IV.2.1.1. Streaming options menu

Click the following icons on the main menu to enter the streaming configuration window:



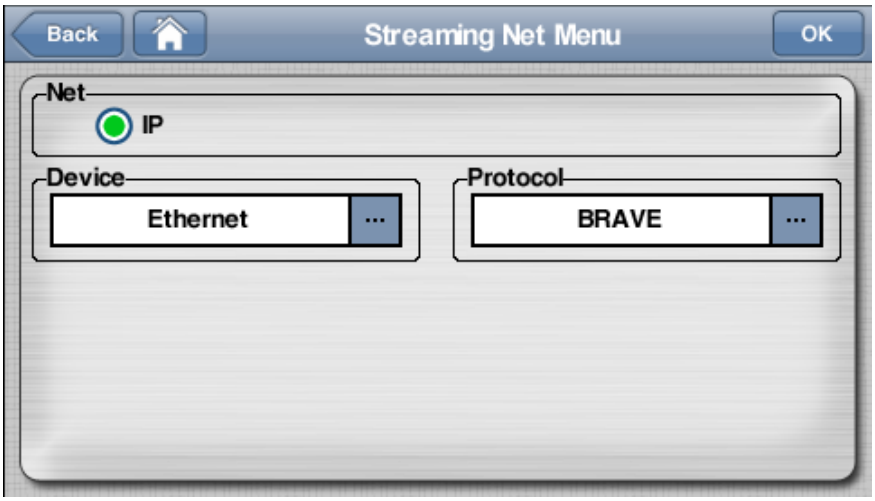
The user switches the streaming network (3G/4G, Wi-Fi or Ethernet) through this menu, as well as adapts some default parameters to specific network circumstances.



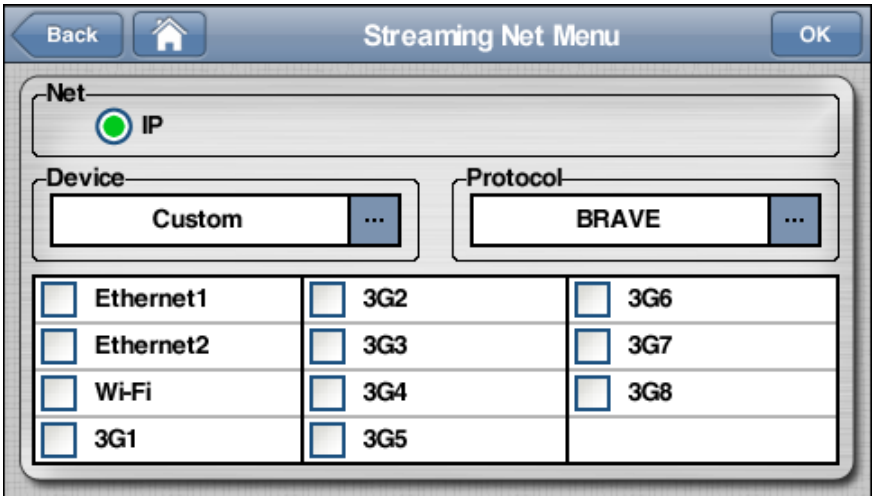
IV.2.1.2. **Network selection (Net)**

Within this menu the user chooses which communications interface to use: IP or ISDN. If IP is chosen, it is possible to select the Device: Ethernet, Wi-Fi, 3G/4G or Custom.

The transmission protocol can be selected between BRAVE, Prodys proprietary protocol allowing bandwidth aggregation and error recovery, MPEG-TS, ProntoNet V4 for compatibility with Prodys ProntoNet family codecs, SIP for compatibility with third party codecs and RTMP for connection with Internet audio servers.



If Custom mode is selected, a new menu appears with all the available interfaces that can be chosen for the transmission.



The more links are selected, the more bandwidth the unit will be able to get for the audio streaming, and the more secure the connection. Up to 7 interfaces (4 interfaces in Quantum Lite) might be linked together to get as much bandwidth as possible in Bonding mode, or to create redundant streams in Diversity mode.

With SIP protocol with SIP Diversity license, is possible to choose two interfaces for the transmission.

With ProntoNet v4 protocol, the Custom menu allows to choose which interface to use for the transmission. This ProntoNet v4 protocol only one transmission interface is possible.

IV.2.1.3. **Connection options**

This window permits the configuration of several parameters to control the way the calls are made:

- **Enable Hang-up Confirmation**. When the box is ticked, if the user touches the connection bar to end a communication, the codec will require the user confirmation before finishing the connection. This is useful to avoid accidental hang-ups.

- **1-Click call**. This option can be set independently for Program and Talkback channels. When this option is chosen, the codec will make a call just by pressing the connection bar, without the need to introduce any destination address or phone book entry. To set up this mode, an IP destination address, a call type and a profile have to be selected in the <Settings…> dialog.

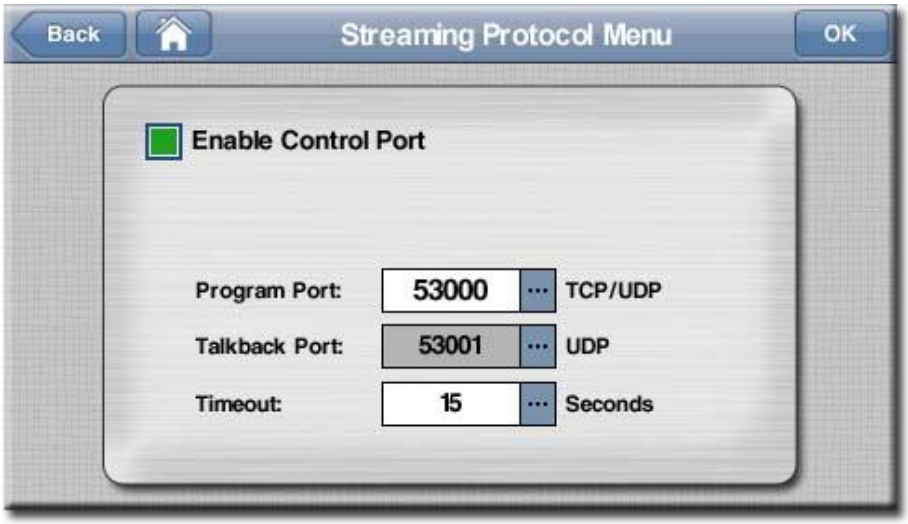
- **Connection On Startup**. This option can be set independently for Program and Talkback channels. When this option is chosen, the codec will make a call automatically just after the codec has started up. The destination IP address, the call type and the profile have to be selected in the <Settings…> dialog.

These two options are very useful to make it easier for some non-technical users to establish a connection.

IV.2.1.4. **Streaming Protocol selection (BRAVE)**

When BRAVE transmission protocol is selected, this menu is used to set IP port and protocol parameters. For most applications the default values are suitable.

- Enable Control Port *(TCP 53000, enabled by default [1])*
- Program Port *(UDP 53000 by default for PROGRAM channel [2])*
- Talkback Port *(UDP 53001 by default for TALKBACK channel [3])*
- Timeout *(the connection timeout is 15s by default [4])*



[1] Disabling the <Control Port>, means disabling the TCP protocol. Therefore any streaming connection should be request from both parties to meet the communication.

[2] Usually this number is more important for the receiving Quantum since the local router must provide “port forward” on this port number for the Quantum. This port must be “forwarded” for TCP & UDP. Please contact your IT specialist or network manager for details about this procedure.

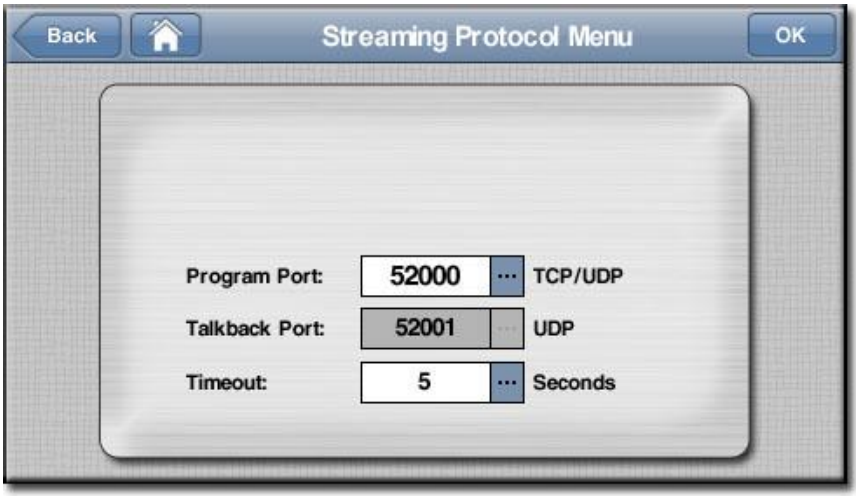
[3] Preceding remark is applicable for the Talkback channel too.

[4] This parameter applies for the detection of interrupted streaming connections. After a streaming connection has been involuntary interrupted, Quantum retries indefinitely to re-connection to its destination.. destination.

IV.2.1.5. **Streaming Protocol selection (ProntoNet V4)**

When Prontonet V4 transmission protocol is selected, this menu is used to set IP port and protocol parameters. For most applications the default values are suitable.

- Program Port *(UDP 52000 by default for PROGRAM channel [5])*
- Talkback Port *(UDP 52001 by default for TALKBACK channel [6])*
- Timeout *(the connection timeout is 5s by default [7])*



[5] Usually this number is more important for the receiving Quantum since the local router must provide "port forward" on this port number for the Quantum. This port must be "forwarded" for TCP & UDP. Please contact your IT specialist or network manager for details about this procedure.

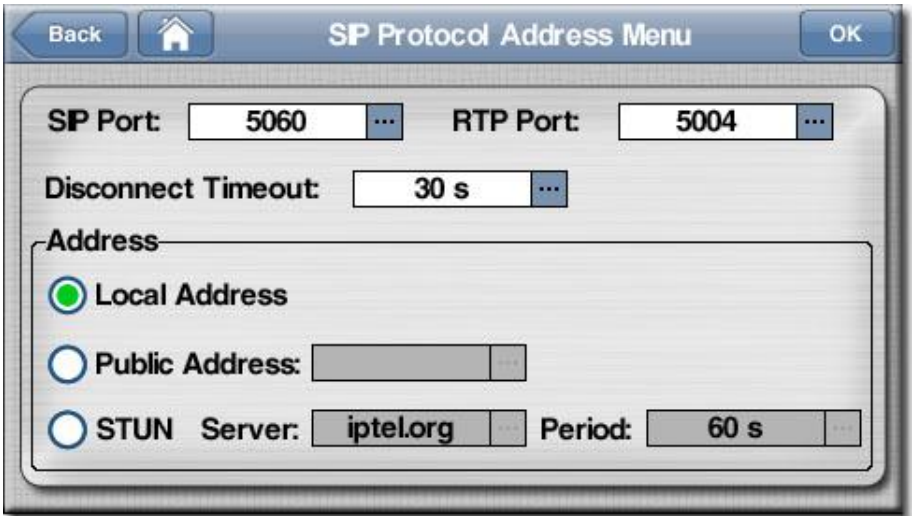
[6] Preceding remark is applicable for the Talkback channel too.

[7] This parameter applies for the detection of interrupted streaming connections. After a streaming connection has been involuntary interrupted, Quantum retries indefinitely re-connections to the destination.

IV.2.1.6. **Streaming Protocol selection (SIP)**

When SIP transmission protocol is selected, this menu contains the SIP configuration parameters.

The SIP protocol address menu permits the configuration of the ports and the SIP address involved in the communication.



- **SIP Port**: UDP 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. This port might be changed by the user if the SIP infrastructure has been set up to run SIP on a different port.

- **RTP Port**: UDP port dedicated to the RTP communication, this is the actual port for the audio streaming. By default the number 5004 is reserved.

- **Disconnect Timeout**: if no audio streaming is received when the Disconnect Timeout expires, the audio communication will be finished.

With regard to the address configuration, there are three different possibilities:

- **Local IP Address**: In case the SIP connection will be made on a LAN environment, the user should tick the 'Local Address' checkbox.

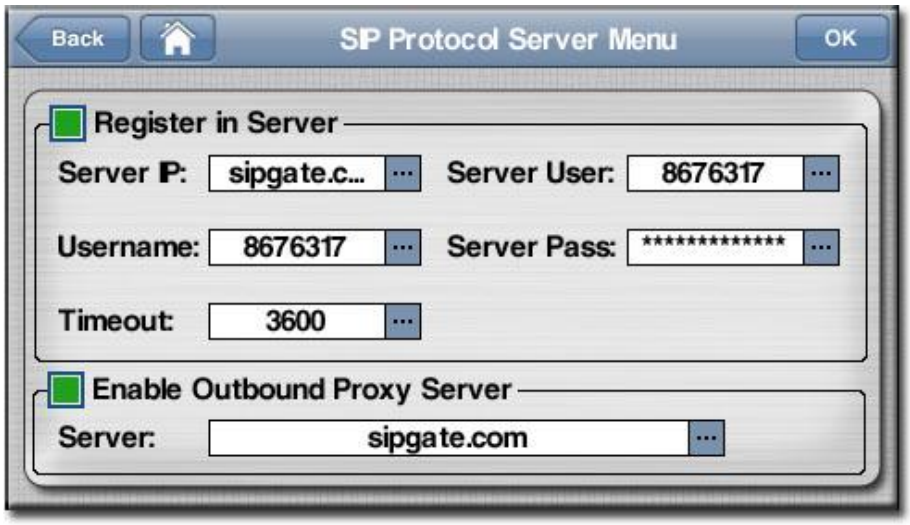
- **Public IP Address**: This parameter is necessary when making connections through a router to the Internet, and the public IP address is known by the user.[8] The user must copy the public IP address of his router into this box.

- **STUN Server & request period**: It is possible to use the STUN protocol to get the public IP address of the unit when connecting to a WAN network by using a NAT enabled device such as a modem or a router, and the user is not aware of that address. Just by entering the IP address of a STUN server, the unit will get that information from it. The 'Request Period' defines how often the unit will ask for this information and refresh the public IP address value. This information might change quite often depending on the type of connection.

The next step to follow for the SIP configuration is to set up the SIP Server configuration parameters[9]:

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

[9] A SIP account must have been enabled in the corresponding SIP server for this to work. Please ask for these parameters to the corresponding IT department or SIP provider.



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

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

- **Server user & Server password**: Some SIP servers require authentication before proceeding to the register of SIP users.

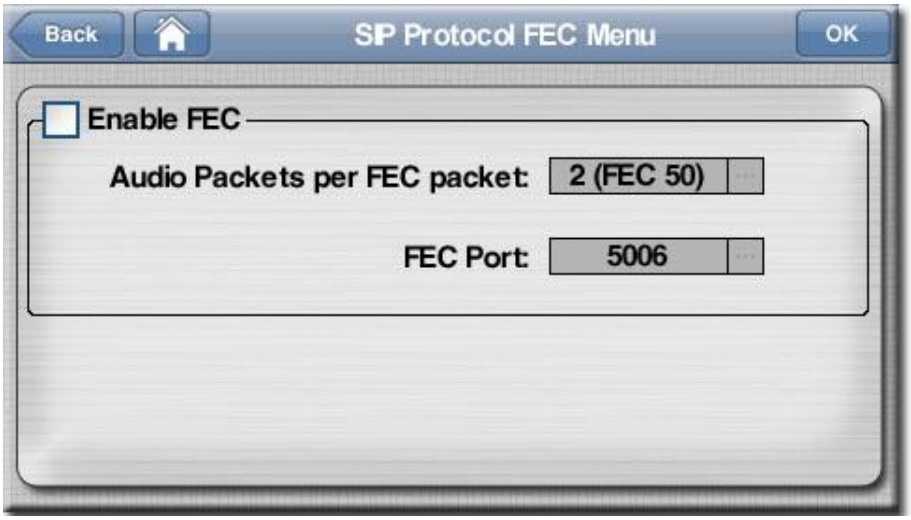
- **Timeout**: Time in seconds before the user information registered at the previous server is flushed. Therefore the information about the SIP user is updated from time to time.

It is also possible to enable an **outbound proxy server** for the SIP communications.

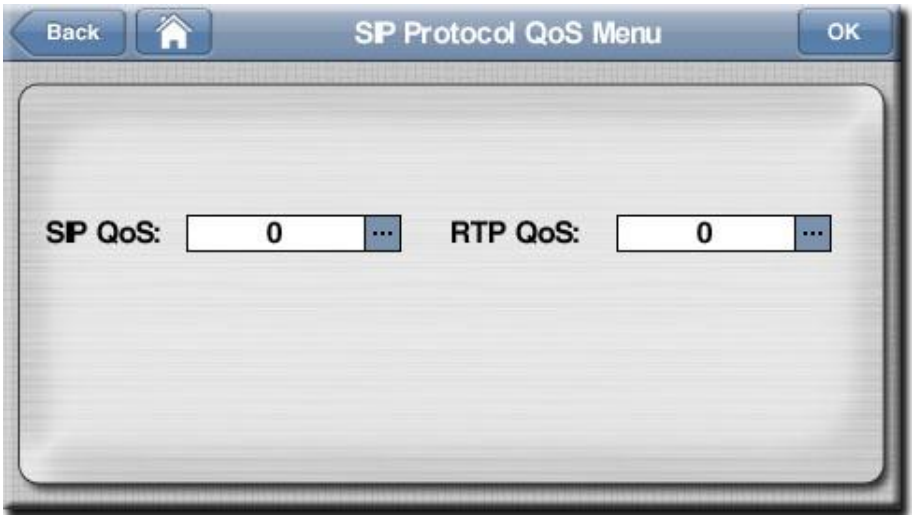
The SIP protocol FEC menu allows the user to enable **Forward Error Correction (FEC)**. The FEC function enables redundant information streaming applicable to the outgoing audio stream. Therefore the steaming bandwidth actually increases but the reception at the remote Quantum is more reliable against packet losses. FEC increases the delay and bandwidth of the audio communication, but it makes the connection more robust.

This function is convenient when detecting some few random packets losses. Typical application scenarios are public IP networks like Internet over ADSL or Wi-Fi access points.

The FEC port and the number of audio packets per FEC packet can be selected here.



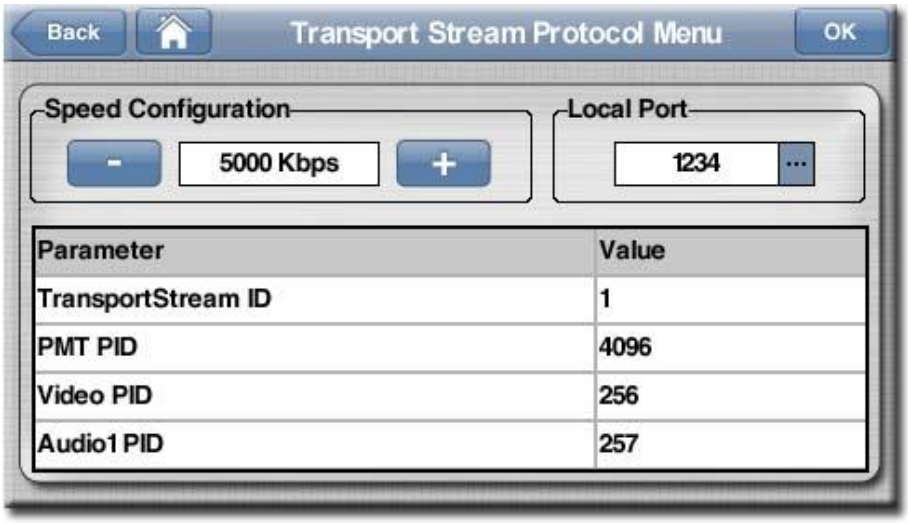
Finally, SIP and RTP QoS can also be set in the QoS menu. SIP and RTP QoS are numerical values between 0 and 255 to establish the quality of service of the SIP signalling packets and the RTP audio packets.



IV.2.1.7. Streaming Protocol selection (MPEG-TS)

When MPEG-TS protocol is selected, this menu is used to set some MPEG-TS parameters. These parameters are:

- Speed of the Transport Stream.
- Local port. Its default value is 1234.
- Transport Stream ID.
- PMT PID.
- Video PID.
- Audio1 PID.



NOTE: Quantum streams audio data encapsulated in MpegTS format over UDP protocol. Please check that the remote codec support MpegTS over UDP.

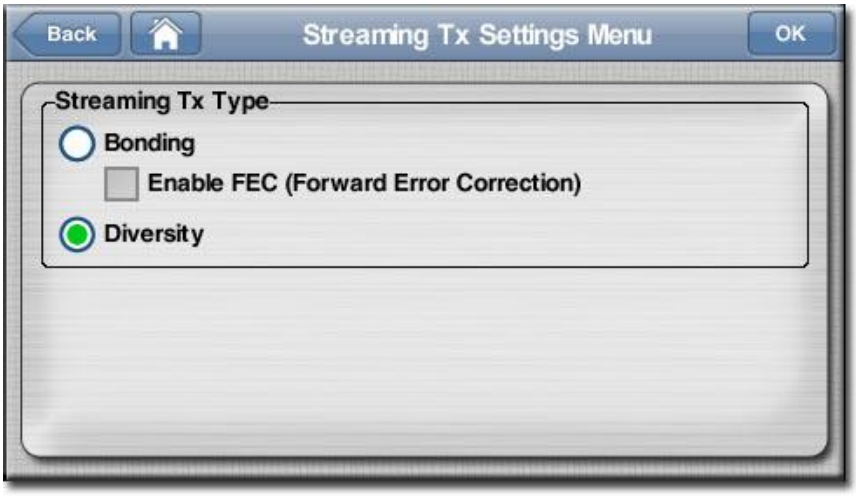
IV.2.1.8. Streaming TX parameters (BRAVE)

This menu allows the user to choose between Bonding and Diversity types when working with BRAVE protocol. Independent configuration variables are available for PROGRAM and TALKBACK channels.

- **Bonding:** it consists of the bandwidth aggregation of all the transmission interfaces available at that moment. The traffic is sent using a dynamic load balance algorithm.

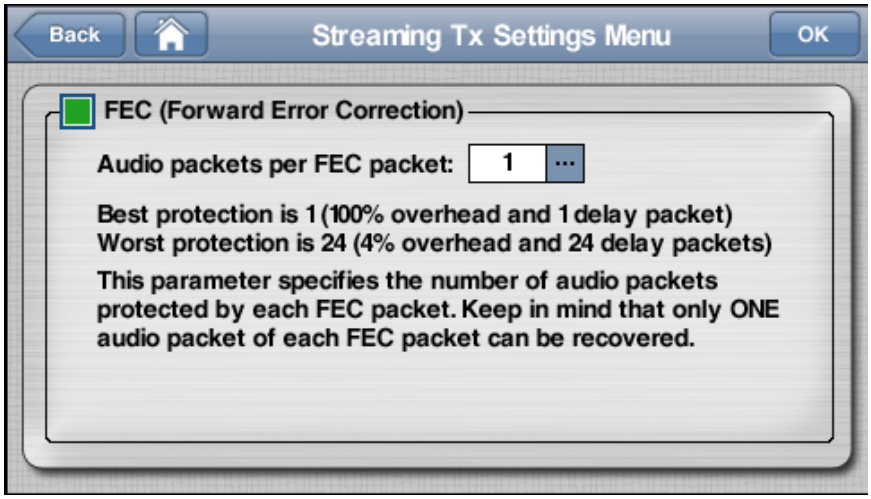
- **Diversity:** this is a redundancy scheme where the traffic is sent simultaneously in all transmission interfaces available.

When bonding type is chosen, it also allows the activation of Forward Error Correction (**FEC**).



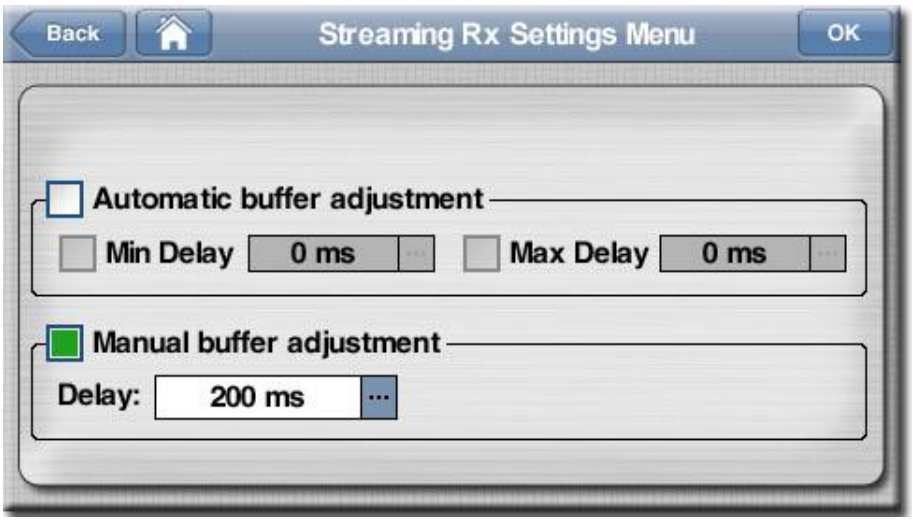
IV.2.1.9. Streaming TX parameters (ProntoNet V4)

This menu allows the user to configure the FEC for protocol ProntoNet V4.



IV.2.1.10. **Streaming RX parameters**

This menu configures the jitter buffer. This configuration affects the overall communication delay. Independent configuration variables are available for PROGRAM and TALKBACK channel.



The buffer size can be set manually or the automatic adjustment algorithm can be chosen.

With automatic adjustment the jitter buffer adapt itself automatically to the jitter of the network. The automatic jitter buffer is able not only to enlarge, but it is also capable to shorten (decimation of samples) in order to recover the initial delay when the jitter of the network decreases and the connection gets back to a more stable and consistent status.

The user has the possibility to set the range of delay where the jitter buffer will stay in. If a maximum delay is set, the buffer will never grow above that value. If the minimum delay is set, the buffer will never be shorter than that value, and that value will be the initial configuration.

The maximum buffer size is 20 seconds. The user has to take into consideration that this size includes the encoding/decoding delay. This means that this would be the total delay for the communication if the network adds no delay.

These parameters can be changed during the audio connection.

IMPORTANT: this configuration will be overridden if the caller uses a profile when establishing the call. Take into account that the user can define different configurations/profiles that affect the operation of the jitter buffer at the receiving end. To avoid this type of operation, the user should choose 'None' as the selected profile when establishing the call and the receiving end will use the configuration defined by the user in this configuration section.

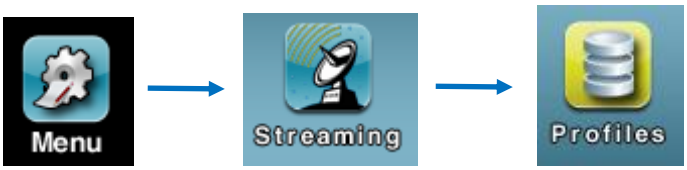
IV.2.1.11. Connection Profiles

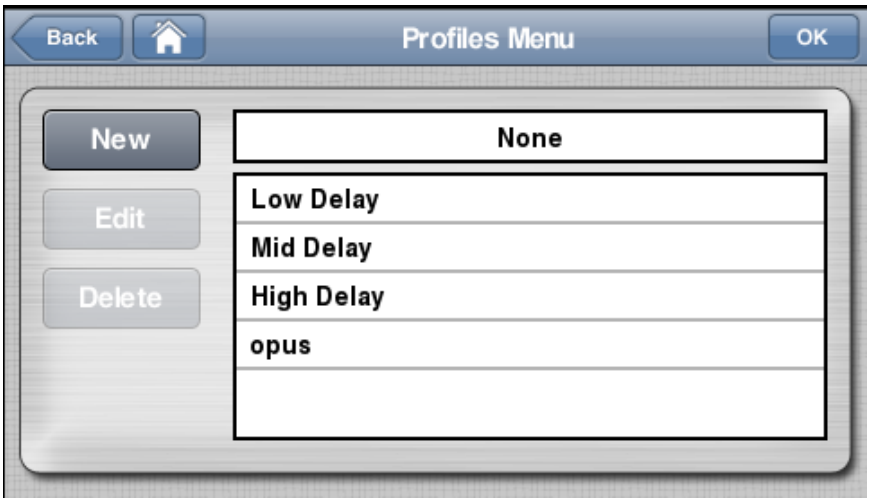
A connection profile is a user-configurable set of parameters that will be enabled whilst the duration of an audio connection. The values set in the profile are no longer valid when the connection finishes. The profile is selected in the Call Menu window, along with the call type and the destination:

One bidirectional communication can use two different profiles (asymmetrical), one for each direction, or can use the same profile (symmetrical) in both directions.

If no profiles are selected, the buffer at the receiver end will keep its current configuration.

To enter the Profiles menu just click on the Profiles button at the main streaming menu:





This menu allows the user to create, edit or delete connection profiles.

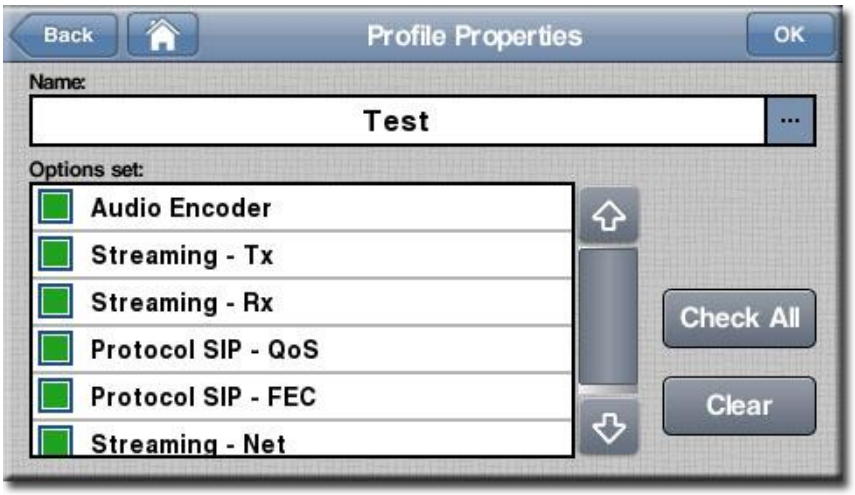
Profiles can include a wide variety of parameters, related both with transmission and reception:

- **Audio Encoder**: Establishes the encoding algorithm of the transmitted audio streams, program and talkback.
- **Streaming Tx**: Parameters related with the transmission, for example FEC.
- **Streaming Rx**: Defines the jitter buffer size of the remote codec which is going to receive the stream. The buffer at the receiving end is usually the main contributor to the global delay of the communication. The longer the buffer the more protection against packet losses and jitter but the longer the delay. The shorter the buffer, the less protection but the shorter the delay. The key point in being able to adjust this parameter from the calling end is that most likely the user who really knows what the capacity

for the current IP link being used for streaming is will be the user that is establishing the call.

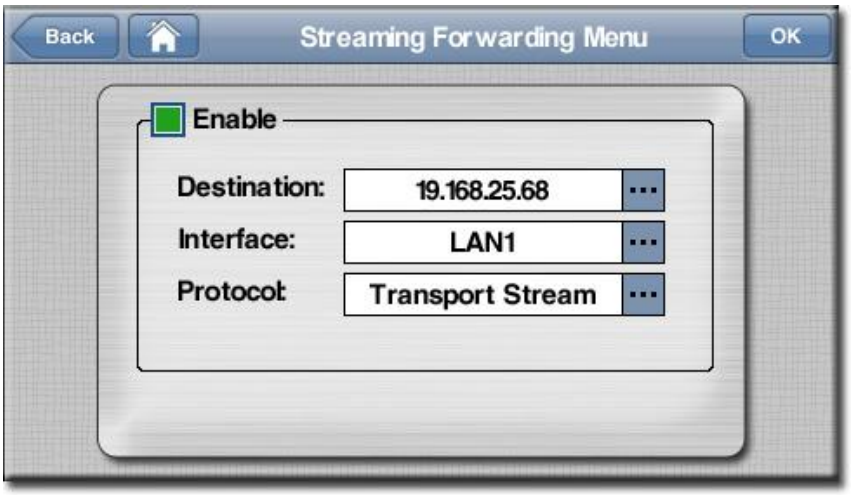
- **Protocol SIP – QoS**: Sets the RTP Quality of Service (QoS) parameter.
- **Protocol SIP – FEC**: Sets the FEC parameters for the SIP communications.
- **Streaming – Net**: The transmission protocol can be chosen between BRAVE, SIP and any other available protocols.

This menu also allows the user to define the name of the profile.



IV.2.1.12. **Forwarding**

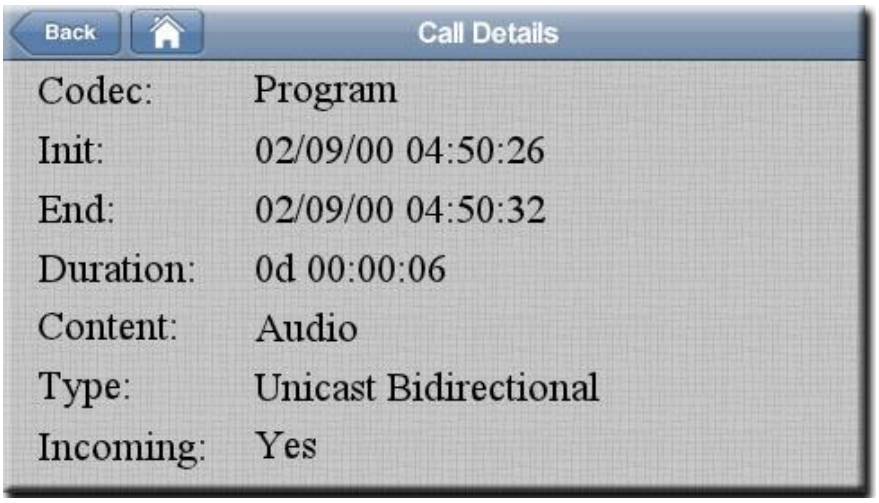
When the Streaming Forwarding license is activated, it is possible to resend the incoming stream to another destination using MPEG-TS or RTMP protocol over UDP (port UDP 1234). The destination address and the Ethernet port involved (LAN1 or LAN2) can be selected.



IV.2.1.13. **Call log**

The call log stores the information about input and output calls, including start/end date and time and IP address.

More information is available clicking on the entries.



The call log can be exported to an XML file.

IV.2.1.14. **Call Filters**

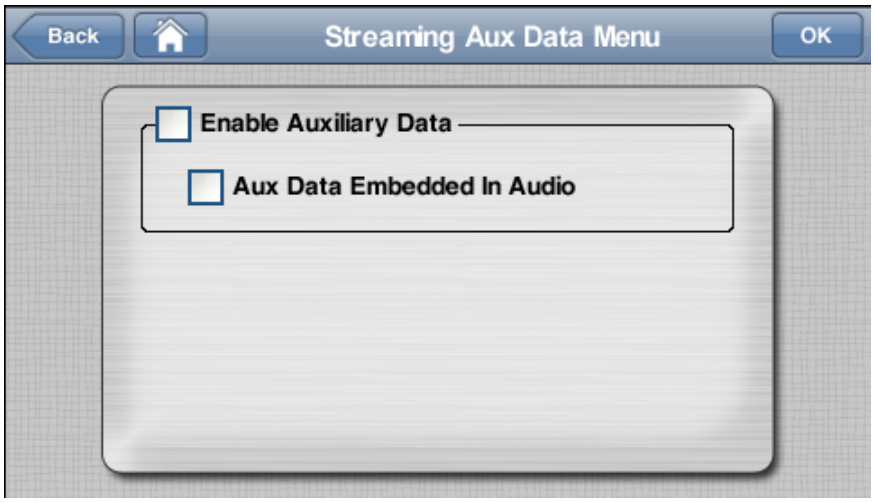
There is a new option in Streaming menu. The Call Filters provides two lists to filter the calls into the device: White List and Black List.

If the White List is enabled, only the numbers or IP addresses included in the list are allowed as incoming calls.

If the Black List is enabled, the numbers or IP addresses included in the list are rejected as incoming calls.

IV.2.1.15. **Auxiliary data**

The Streaming Aux Data Menu allows establishing auxiliary data channels associated to each communication using the RS232 serial ports.



It is also possible to embed the auxiliary data in the audio streams when the compression format allows this possibility.

IV.2.2 Interfaces options menu

In this menu several communication interfaces can be configured: Ethernet, Wi-Fi, 3G/4G services for audio communications, and RS232 and GPIO for the transmission of auxiliary data. There's also an option for controlling the bandwidth consumption in each interface.

IV.2.2.1. **LAN configuration**

The LAN1 interface is always enabled and used for transmission and reception of control data (Web control, SDK remote control protocol, and ProdysControlPlus), regardless of the settings for the Streaming Network option (be it Wi-Fi, Ethernet, Custom…).

When 'Custom' is selected at the streaming network configuration menu, LAN1 can be selected as IP interface for diversity or bonding to other IP interfaces and thus to transmit and receive audio data. In this case, LAN1 will carry both audio and control data simultaneously. If 'Custom' is selected but LAN1 is not selected, LAN1 will only carry control data.

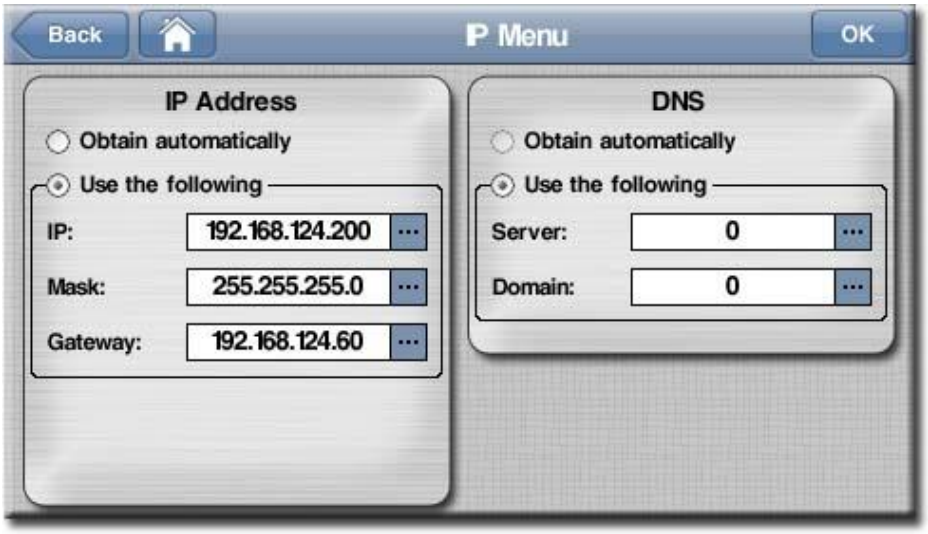
When 'Ethernet' is selected as the streaming network configuration menu and LAN2 interface is disabled, LAN1 will transmit and receive all audio data for both the main program and the talkback audio streams. But, when LAN2 is enabled, and 'Ethernet' is selected as the streaming network configuration menu, LAN1 interface will only transmit and control data (web control, ProdysControlPlus or any other IP data not related to the audio streaming.

The LAN2 interface will only carry the streaming data (not control data) if the LAN2 interface is enabled in the configuration and 'Ethernet' streaming mode is selected, or if 'Custom' streaming mode has been selected and LAN2 has been enabled for bonding with other IP interfaces.

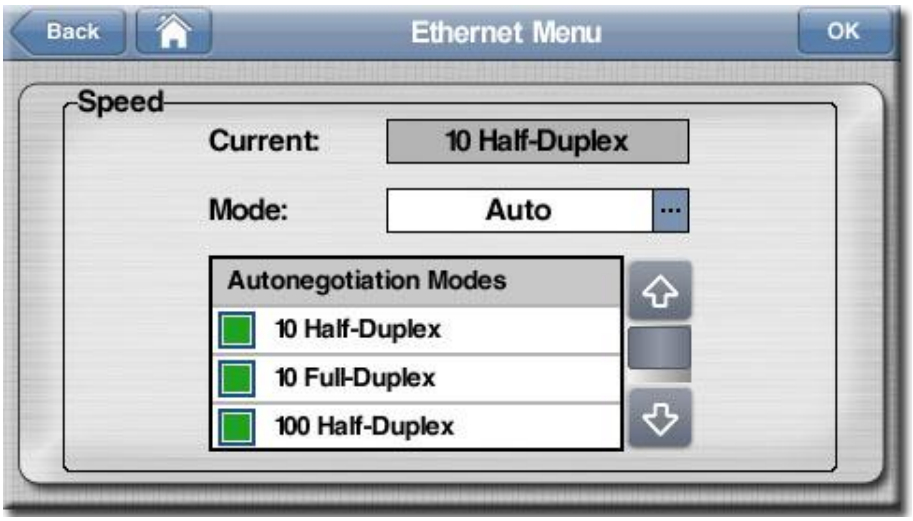
IV.2.2.2. **LAN1 configuration**

All configuration parameters related to the Ethernet port will be established here: IP address, IP mask, IP gateway, DNS and DHCP.

The LAN1 interface is used for all IP communications when the LAN2 interface is not enabled. Once LAN2 is enabled, LAN2 will carry all IP data related to the PROGRAM and TALKBACK streams, whilst the LAN1 interface will only carry control IP data, like the web control.



With the Ethernet menu it is possible to set speed and duplex configuration:



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.

At the top of the window, and in case the unit is set to 'Auto' mode, the 'Current' box will indicate what operation mode the unit has adopted after the negotiation.

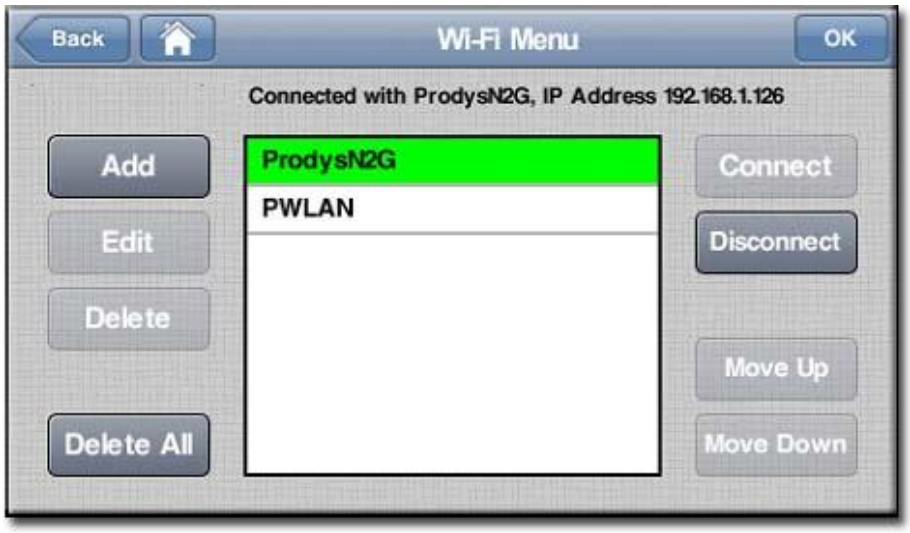
IV.2.2.3. **LAN2 configuration**

IP address, IP mask, IP gateway, DNS and DHCP parameters are defined here. This interface is restricted to the PROGRAM and TALKBACK streaming if this LAN2 has been enabled; keeping just the web control access on LAN1.

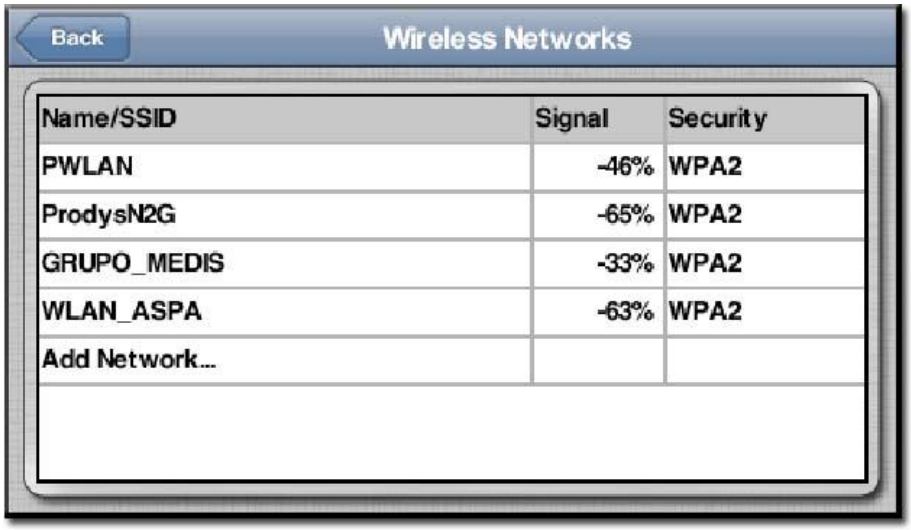
IV.2.2.4. **Wi-Fi configuration**

Quantum units can be equipped with an external USB Wi-Fi module. This chapter describes how to configure this interface. The user has to take into account that one thing is to configure this interface to connect to a Wi-Fi network, and another thing is how to use this interface in terms of streaming. For example, this interface might be defined as the main and only IP interface to stream the audio or for file uploading, or it might be used together with other IP interfaces to get more bandwidth (bonding) or more redundant streams (diversity).

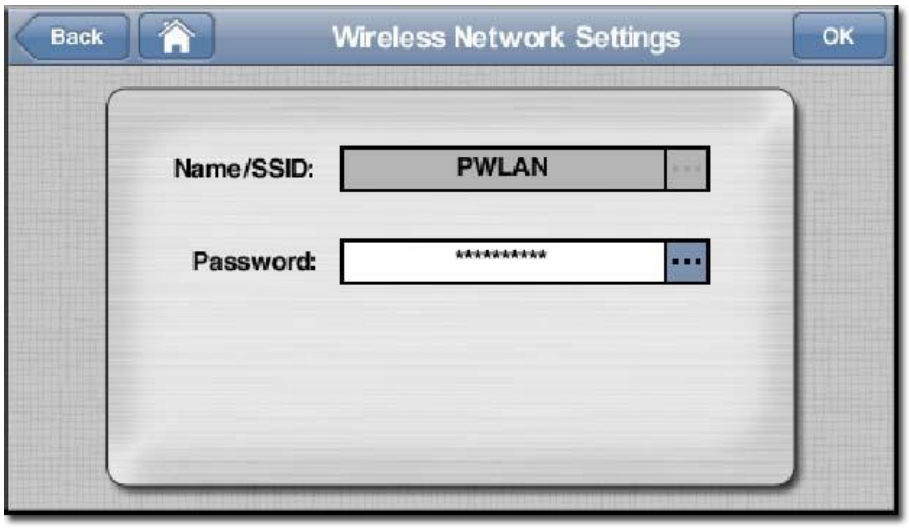
Please take a look at the picture below. It shows the main Wi-Fi configuration menu:



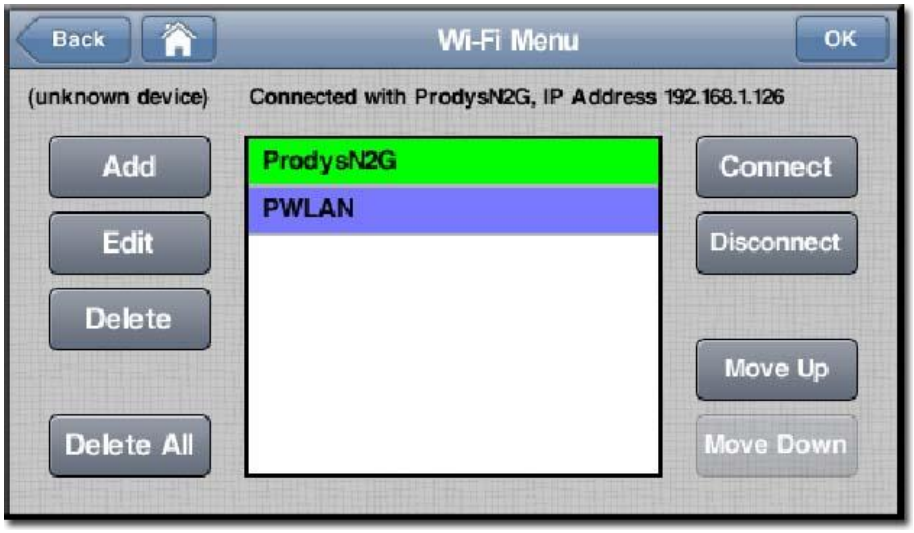
From this menu the user can select what Wi-Fi network will connect to. By pressing the Add button it is possible to access the Wireless Networks Menu where the different Wi-Fi networks automatically detected by the unit will appear, so that the user can just select any of them and add them to the list of Wi-Fi networks on the main Wi-Fi menu. The type of encryption and the quality of reception will be shown for each of the available networks.



Once in the main Wi-Fi menu, just clicking on any of the Wi-Fi networks available in the list, the user will edit the password settings for that specific network.



Once all networks in the list have been configured with their corresponding security settings, in order for the unit to connect to any of those networks, the user has to click on the desired network and press the Connect button. The current Wi-Fi network in use will be shown on a green background colour, whilst the selected Wi-Fi network will be shown on a purple background colour.



In the picture above, ProdysN2G is the current Wi-Fi network in use, whilst the PWLAN network is currently selected by the user. By clicking on the Disconnect button the user can disconnect the unit from the current Wi-Fi network.

To delete a network, just select one network and press the Delete button. The system will ask for a confirmation.

To end with, it is also possible to define the priority of the Wi-Fi networks. The unit will try to connect first to the Wi-Fi network at the top of the list. The user can change the order of the different networks in the list by means of the Move Up and Move Down buttons.

Once all networks in the main list have been defined (password configured), and the user has selected one of the available networks, just by pressing on the Connect button the unit will try to connect to that network.

NOTE: In order for the user to configure the Wi-Fi interface, this interface must be selected as one of the IP interfaces to be used for the streaming, be it alone or along with others, like 3G/4G links for example.

IV.2.2.5. 3G/4G network's configuration

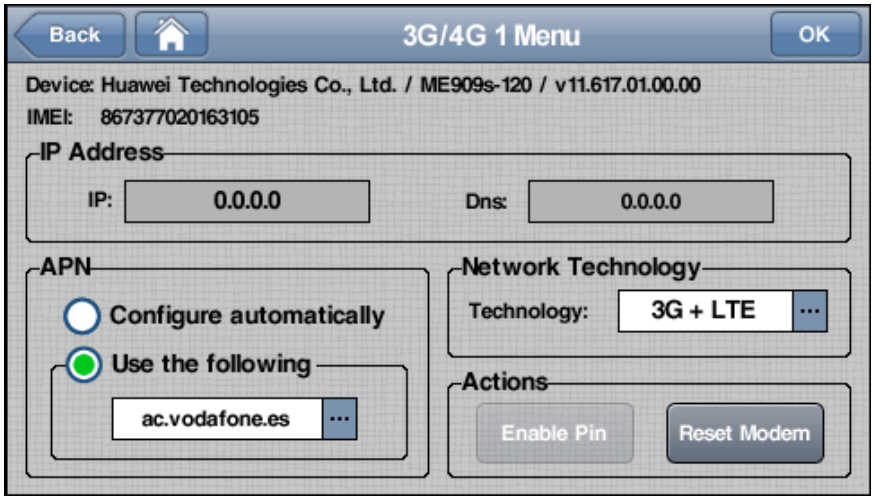
Quantum units can stream over a bundle of up to four 3G/4G modems (4 internal SIM card slots) or two modems for Quantum Lite.

Different service providers can be used for this purpose, so every 3G/4G module must be configured according to the data access settings for that specific provider.

In this configuration window the user has to set up the APN (Access Point Name) information, corresponding to the provider of the IP data connection for the 3G/4G modem under configuration. The APN settings are widely available on the Internet for each 3G/4G data provider.

Another option is to choose the automatic configuration option.

Take into account that the step above must be followed for all internal modems that have been provided with a SIM card in its corresponding SIM card slot. Some more detailed information on the status of the different 3G/4G connections can be found at the 3G/4G section in the monitor menu.



The network technology is also configurable. The possible values are:

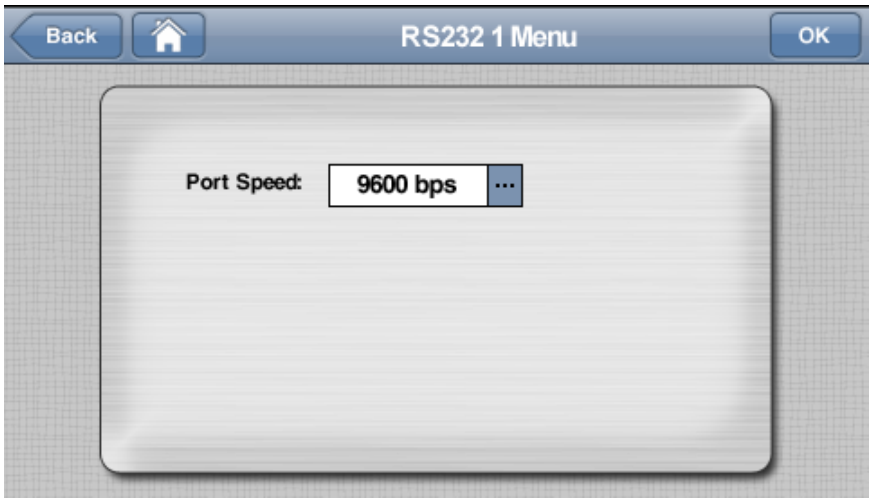
- **3G+LTE**: The modem will try to connect first to a 4G signal. In case there is no one available, it will try with 3G.
- **3G Only**: The modem will try to connect only to 3G networks.
- **LTE Only**: The modem will try to connect only to 4G networks.

Please take into account that the step down from 4G to 3G is possible during the connection, but the step up from 3G to 4G is not usually unavailable while streaming. If the user knows that the system switched over 3G at some point but there is 4G reception, just hung up the current connection and let the system switch over 4G again.

This window also offers the IMEI value of the modem, on the top part of the window.

IV.2.2.6. **RS232 configuration**

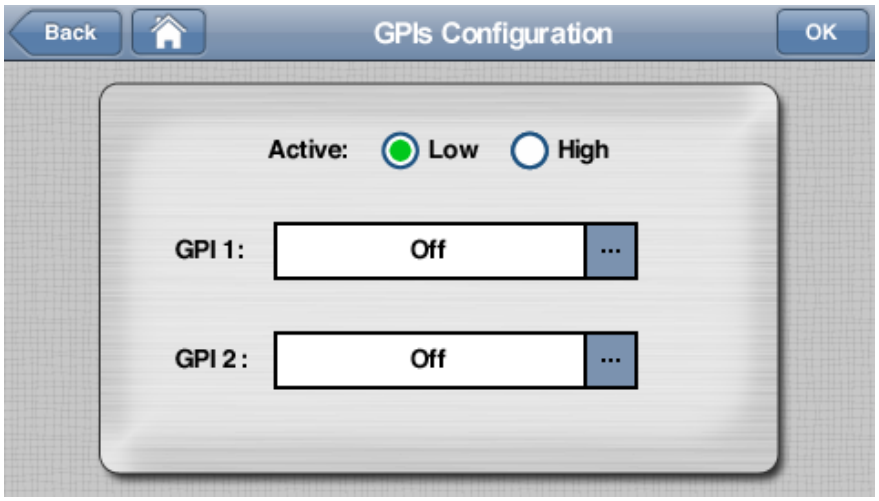
An auxiliary serial data channel can be established using the RS232 interface. The range of speeds ranges from 300 bps to 9600 bps. When the serial interface is enabled, the user can embed auxiliary data coming from the serial interface (RS232) into the audio stream. This data will be available at the output of the serial port at the receiving end.



IV.2.2.7. **GPIOs**

From this menu the GPIO port is configured. The amount of GPIO available might be different from one codec to another of Quantum Family, but the different functions for the GPIO are shared among them.

It is possible to choose the polarity of the GPIs and GPOs. They can be active Low or active High.

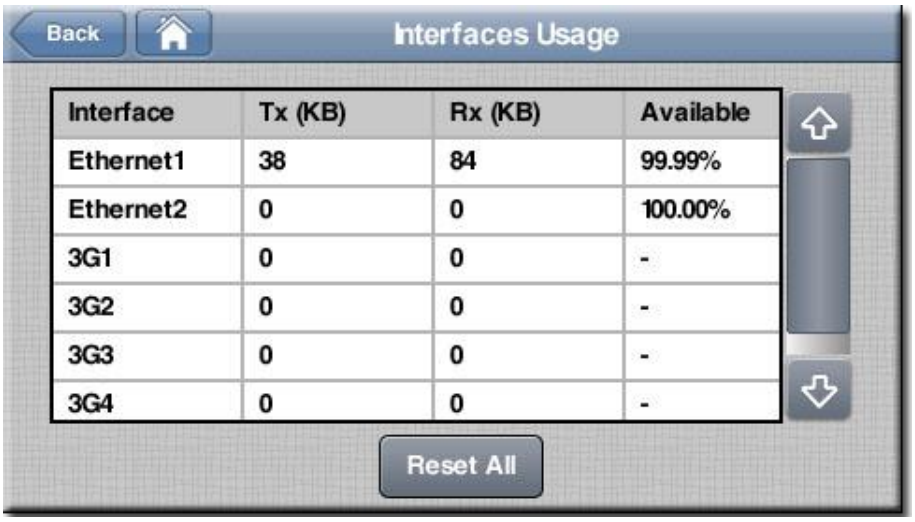


INPUTS	
Transparent	Under this configuration, the state of the input will be present in its homologous output in the Quantum codec connected on the other end. This allows the user to get remote contact closures, a typical application in the broadcast market.
Program Call/Hang-Up	With the same input the user can establish and finish the program connection (call when the input is activated, hang-up when it is deactivated). The destination address should be selected from the book.
Talkback Call/Hang-Up	With the same input the user can establish and finish the talkback connection (call when the input is activated, hang-up when it is deactivated). The destination address should be selected from the book.
Reboot	Allows the user to reboot the unit upon the activation of a GPI.
Disable Fan	This control enables or disables the internal FAN
Load Preset	Allows the user to load a preset upon the activation of a GPI.
Acknowledge Alarms	When this input is activated, the alarms will be acknowledged.
Program Call	When this input is activated, the codec will proceed automatically to connect the Program (codec 1) line. The user can select which entry of the book will be used for this purpose.
Program Hang-up	When this input is activated, the codec will proceed automatically to disconnect Program (codec 1) line.
Talkback Call	When this input is activated, the codec will proceed automatically to connect the Talkback (codec 2) line. The user can select which entry of the book will be used for this purpose.
Talkback Hang-up	When this input is activated, the codec will proceed automatically to disconnect Talkback (codec 2) line.

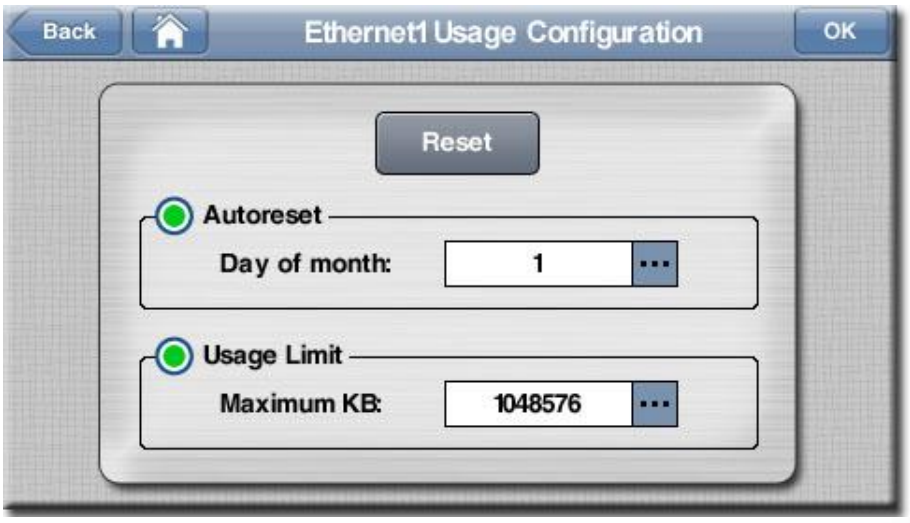
OUTPUTS	
Transparent	Under this configuration, the state of the output will be the same as the corresponding input of the Quantum codec connected on the other end.
Alarm Active	The output will be activated when one alarm is activated.

IV.2.2.8. Usage

The Usage menu allows the user to get a real-time measure of the bandwidth consumption (Rx and Tx) for each of the different IP interfaces.



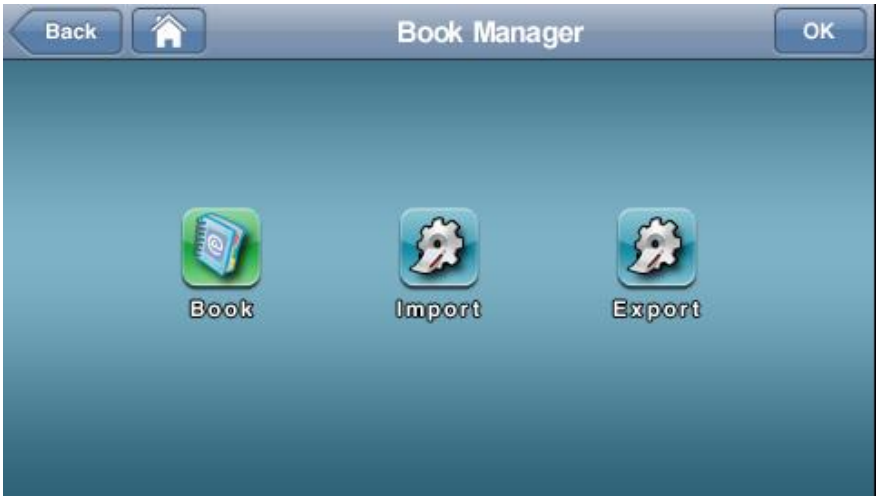
By clicking on any of the interfaces shown in the table, the following configuration screen will be shown.



In this screen it is possible to reset the statistics for that interface, set an option to auto-reset to the day of the month selected (very useful for monthly data plans of some mobile operators) and set a limit to the consumption (this is also useful for some data plans with a limitation in the speed or bandwidth).

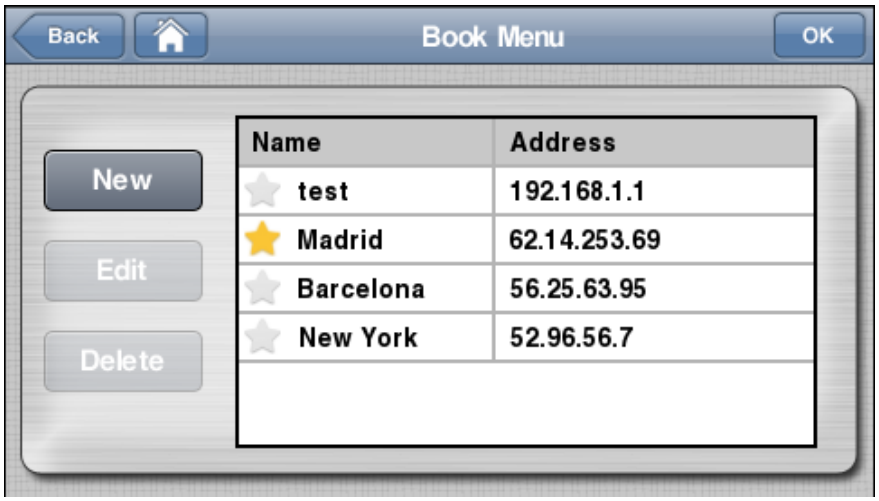
IV.2.3 Book manager

Favorite addresses to call can be stored thru this path. Then direct dialing is possible to any stored destination.

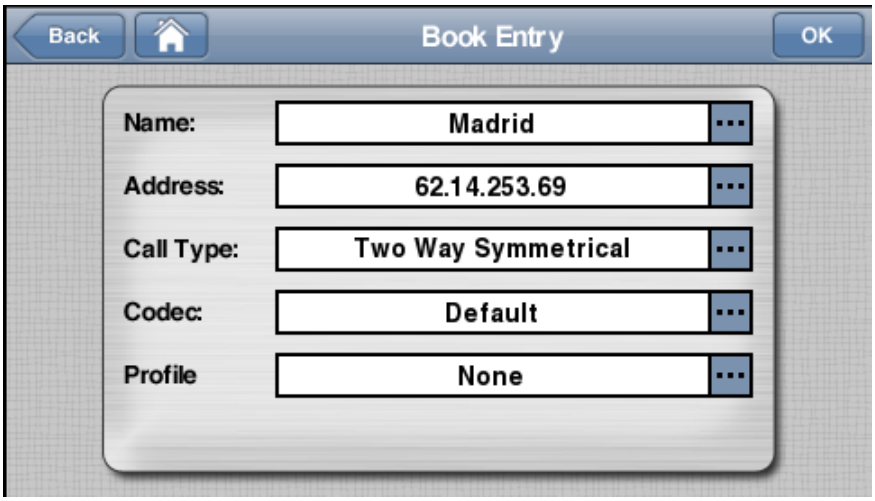


IV.2.3.1. **Book**

This menu allows the user to add, modify or delete different phone book entries.



When the star on the left side of the name is highlighted, that book entry becomes part of the Favourites list in the Call menu.
The book entries are composed of name, address, call type, codec (1 or 2) and profile.



IV.2.3.2. **Import and Export Book**

This two options permit to save and load the book entries in XLM format. With these two options is possible to copy a book from one codec to another.

IV.2.4 User options menu

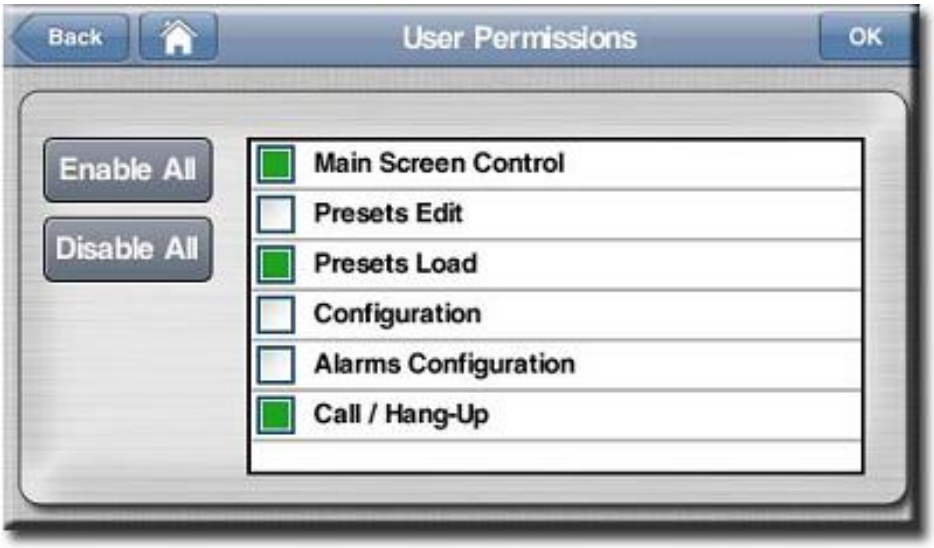
It is possible to establish different access levels to the codec defining different users[10]. Each user consists of a name and a password that has to be used to log into the codec web interface or the touch panel menu. Please take a look at the picture below with the login window:

IV.2.4.1. Creating a new user

To create a new user click on the 'New' button and enter a name. Once the name is entered and the OK button is clicked, the following options appear:

[10] By default the user Admin is enabled without password or operation restrictions.

The <Permissions> button opens a dialog for operation's restriction for this user. The following example is a typical profile for basic users.



Green colour stands for activation. Those checkboxes that have been enabled will allow this user to access the corresponding menu/functions. Those that have not been enabled will appear as greyed-out in the corresponding menu, or they will not appear in the user interface.

IV.2.5 System menu

The system menu screen offers a number of options for the codec maintenance. The following pictures show the different menu options. Take into account that the different options have been spread out over several different windows that can be travelled by clicking on the Next button. Take also into account that not all the options are available for all the Quantum models.

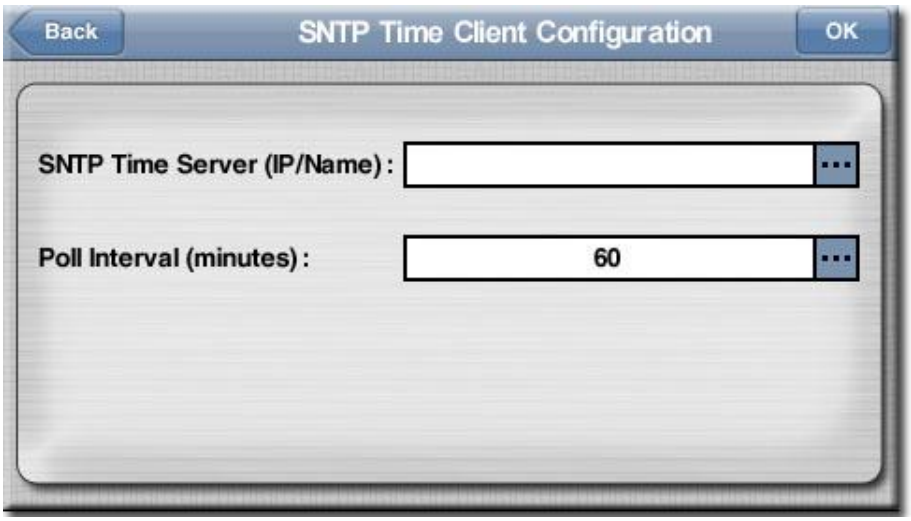
The options available in this menu are:

- **Name**: This option permits to assign a name to the codec. This name will be used for several different IP services to identity the unit, like the HTML web interface, or the SNMP monitor protocol.

- **Version**: Shows firmware versions of the codec processors.

- **Date/Time**: Date and time settings. This option also allows the unit to synchronize time and date by using **SNTP protocol** (Simple Network Time Protocol). SNTP operates always in the client-server mode and for this reason Quantum codecs can work as SNTP server or SNTP client.

When the codec is configured as a time client, it's only necessary to include the IP address of the SNTP time server.



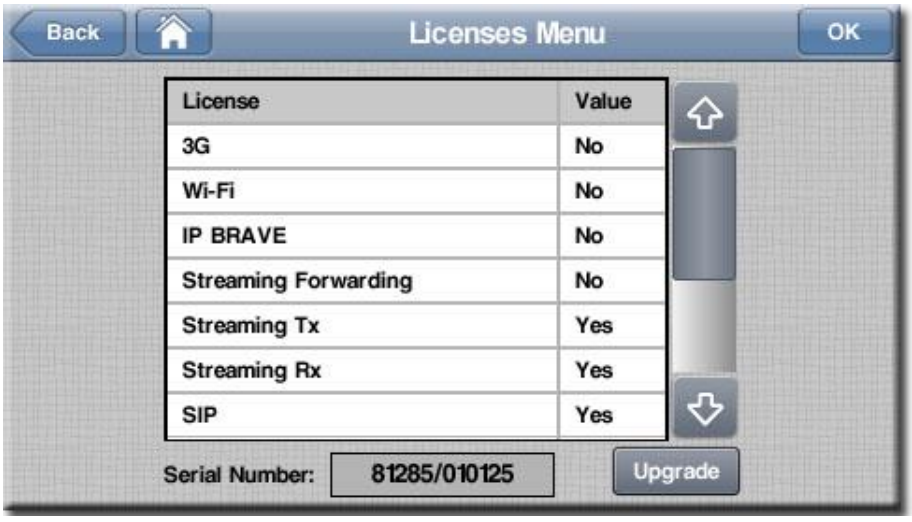
- **Default Configuration**: This option allows the user to restore the default configuration. User can exclude from this restoration the following configuration sections:

    - LAN Settings
    - Phone Book
    - APN list
    - Alarms
    - Presets
    - Profiles
    - Statistics
    - Users
    - Wi-Fi settings

This is very useful in case the user just want to restore some parts of the factory default configuration.

- **Licenses**: Licenses management. Accessing through the web browser, it's possible to upgrade the unit with new licenses. These licenses will be provided by Prodys as a file with extension "hlf".

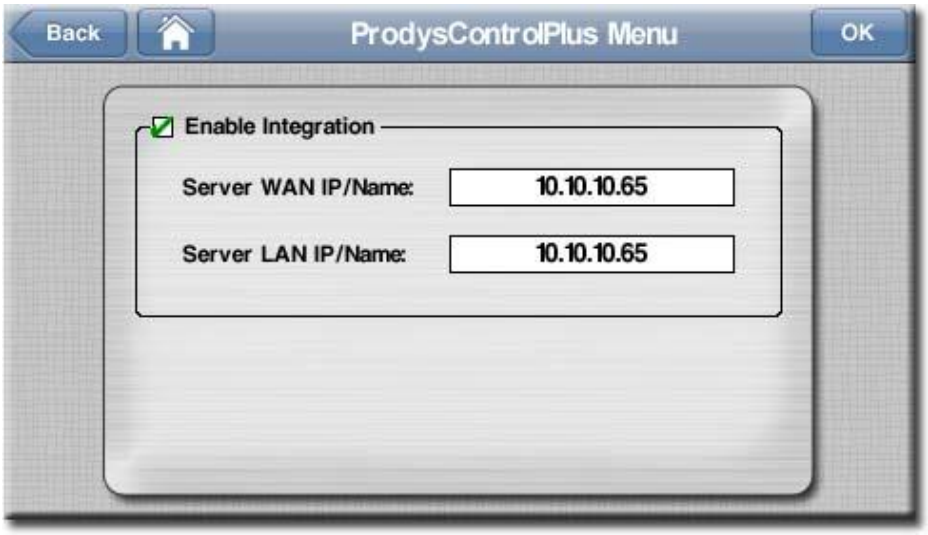
For more information about the different licenses available at any given moment please contact Prodys Sales Team at sales@prodys.net.



- **Save Log**. This option is only available when controlling the unit via the web browser. The file generated by the system contains some information about the internal operation of the system that might help Prodys Support

department to understand the scenario where the customer is using the unit.

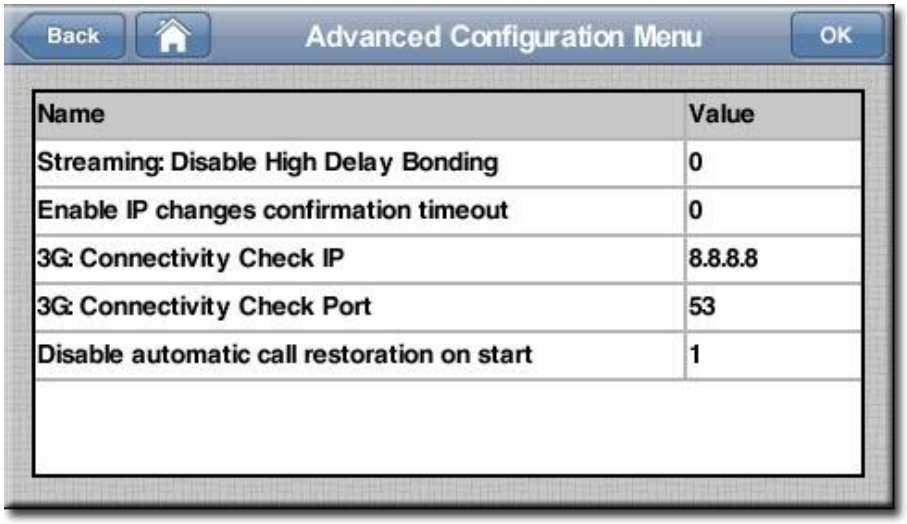
- **Reboot**: To reboot the codec.
- **Import/Export Configuration.** Only through web browser. Recording of a configuration file and loading of a configuration file.
- **Enable Autologin**: to activate and deactivate the login screen for each user.
- **ProdysControlPlus**: Enables ProdysControlPlus software integration.



NOTE: This functionality allows the user to control the unit remotely, even though that unit being controlled is behind a firewall or even connected to the internet just via 3G or 4G. This establishes a kind of NAT Traversal communication between the PC running the ProdysControlPlus application and the remote unit that allows that PC to access the unit remotely. If the unit is in the same network as the PC running the ProdysControlPlus Server, just enter the Server LAN IP address. If the PC running ProdysControlPlus Server is in another network across the Internet, just enter the public IP address of the router which that computer is connected to.

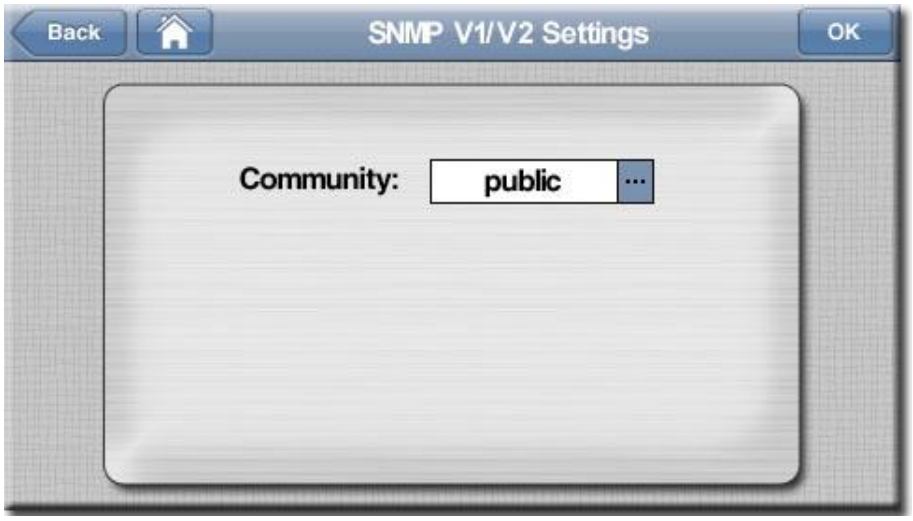
Take into account that the remote unit will traverse any NAT/Firewall so it will allow the remote PC to access and control it remotely, but the NAT/firewall at the PC side must configured so that ports TCP 54000, 54001, 54002, 54003 and 54004 are opened and forwarded to the PC running ProdysControlPlus. However, this does not have to be done at the remote side, where it is much more difficult or even impossible (for example when connecting via 3G/4G) to access and control the firewall.

- **Advanced**: advanced configuration options.

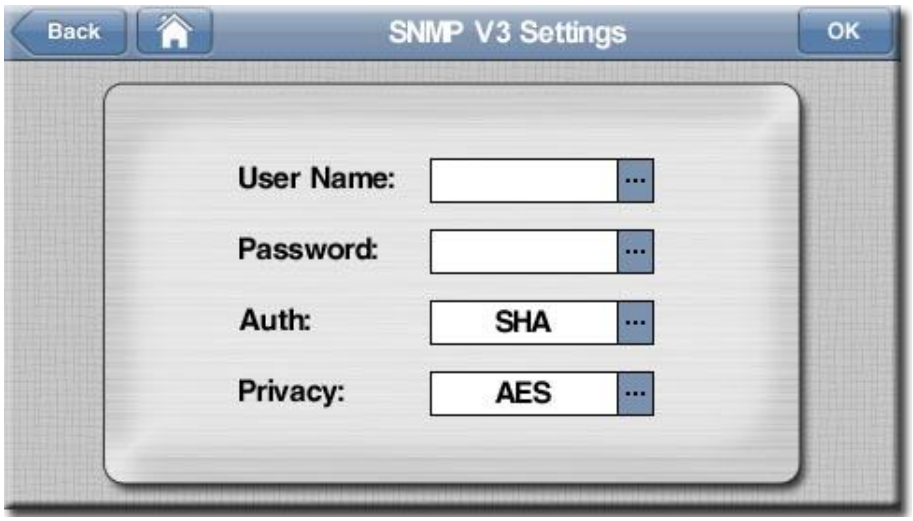


- **Touchscreen**: This option allows the user to calibrate the touch screen.
- **SNMP**: SNMP configuration for versions v1, v2 and v3.

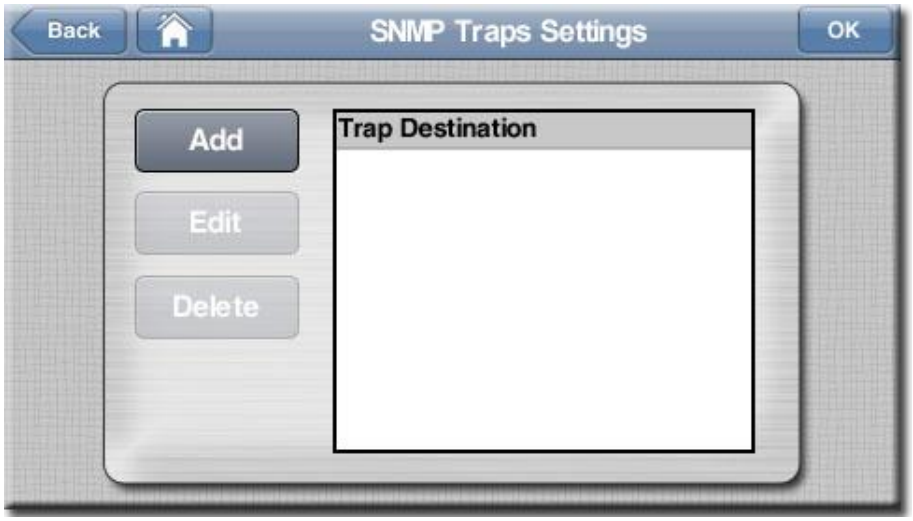
For versions v1 and v2 is possible to set the SNMP community:



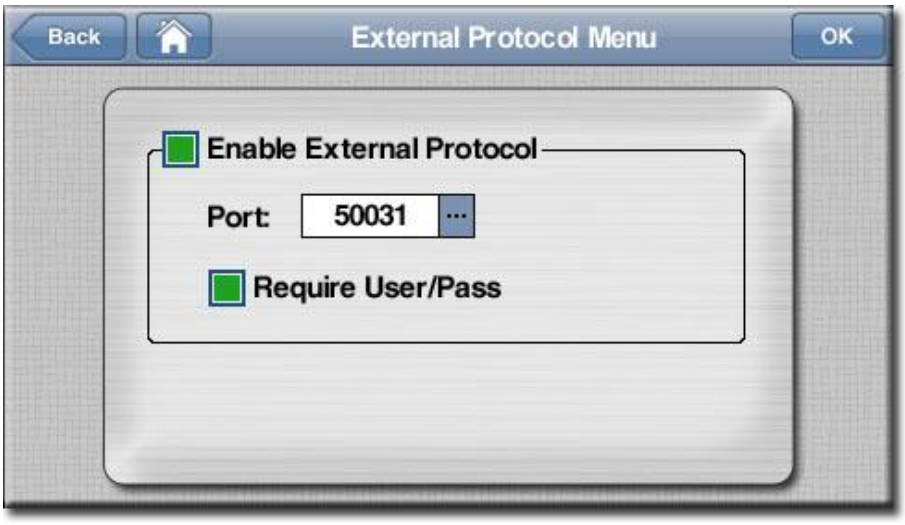
For version 3 it is possible to set the user name and password as well as the authentication and the privacy options:



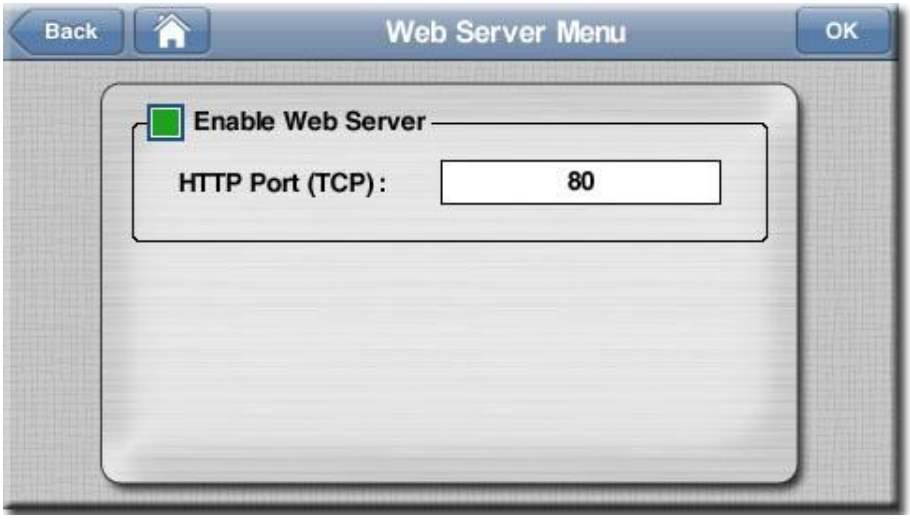
It is also possible to activate the SNMP trap's delivering in the event of an alarm. In the SNMP traps setting dialog, one or several trap destinations can be set.



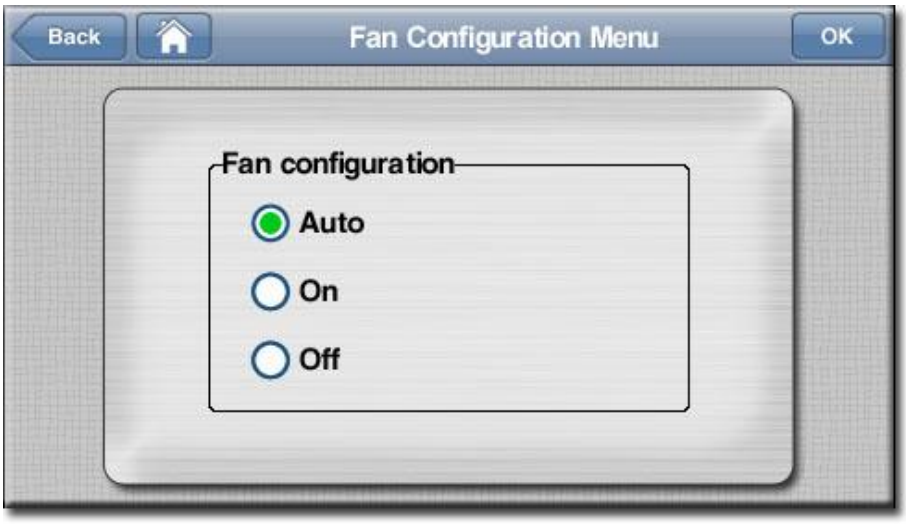
- **External protocol**: The Quantum Software Development Kit provides a tool **to manage the Quantum units from external applications** or devices other than ProdysControlPlus. In this way, it is possible to customize the control of the codecs and integrate them in a global management system. The port used for this external management can be changed. More information in the Quantum SDK User Manual.



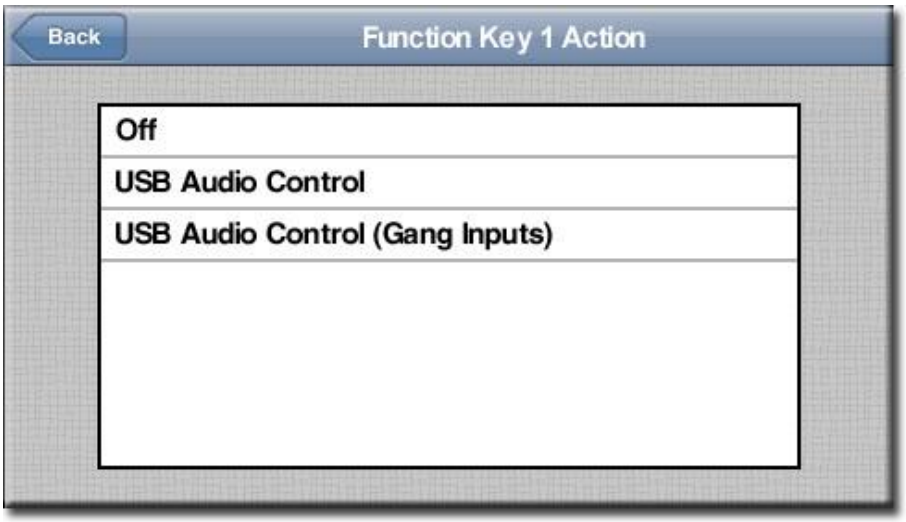
- **Web server**: the web server can be deactivated to avoid remote control of the codec. It is also possible to change the default port (80).



- **Fan**: Permits enabling or disabling the fan of the unit. AUTO means that the Fan will start working when the temperature surpasses a threshold. This is the best configuration to preserve fan lifespan. It is also possible to enable or disable the fan regardless of the temperature.

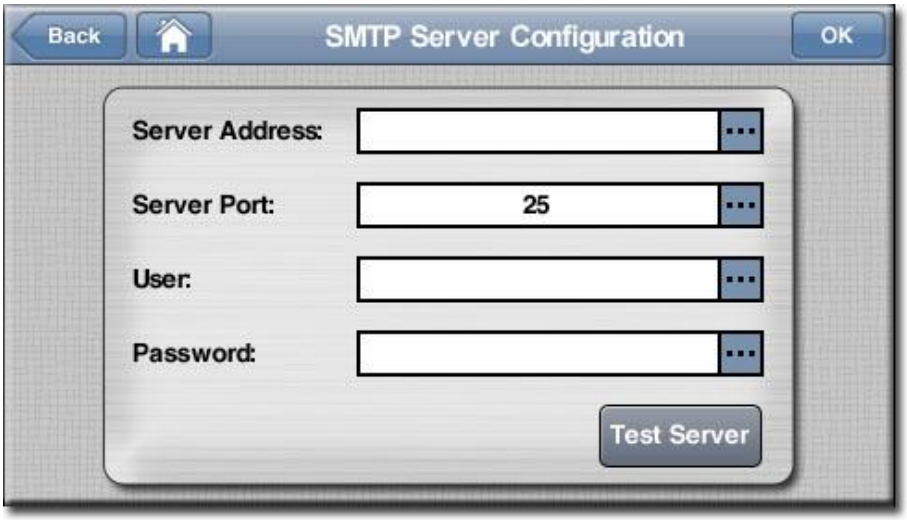


- **Function keys:** It is possible to define the functionality of F1 and F2 keys in Quantum XL, W and V-Quantum. If "USB Audio Control" is chosen, pressing the function key will turn the 4th and 5th rotary controls into the gain controls of the USB audio. If "USB Audio Control (Gang Inputs)" is chosen, the 4th and 5th rotary controls will be coupled.

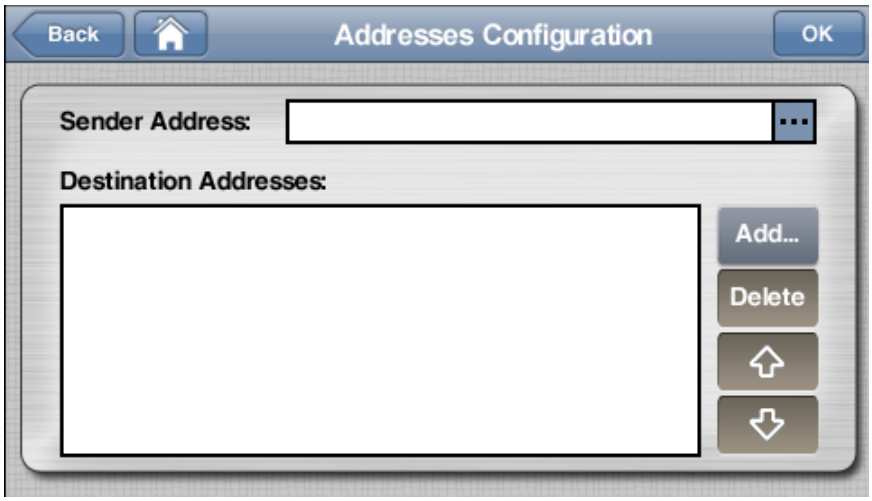


- **E-mail notifications:** this option activates the alarms notification using emails. Once this possibility is activated, an email server, the list of recipients and the severity level have to be introduced.

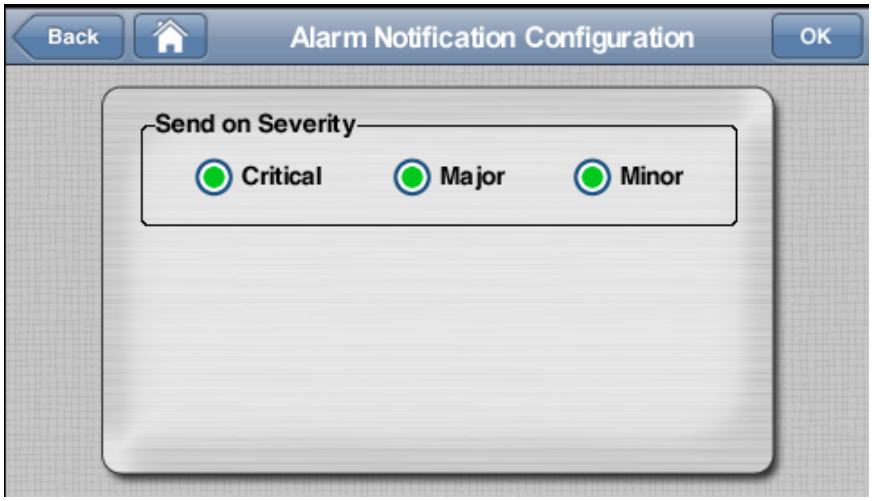
To configure the email server it is necessary to fill the address, port and login data. It is also possible to try the email delivery sending a test email clicking on the "Test Server" button.



A list of recipients can be created on the "Addresses Configuration" window. Also the name of the sender can be configured.



The alarms severity level can be used to filter the email notifications. This is configured on the "Alarms" window. There are three levels: Critical, major and minor.

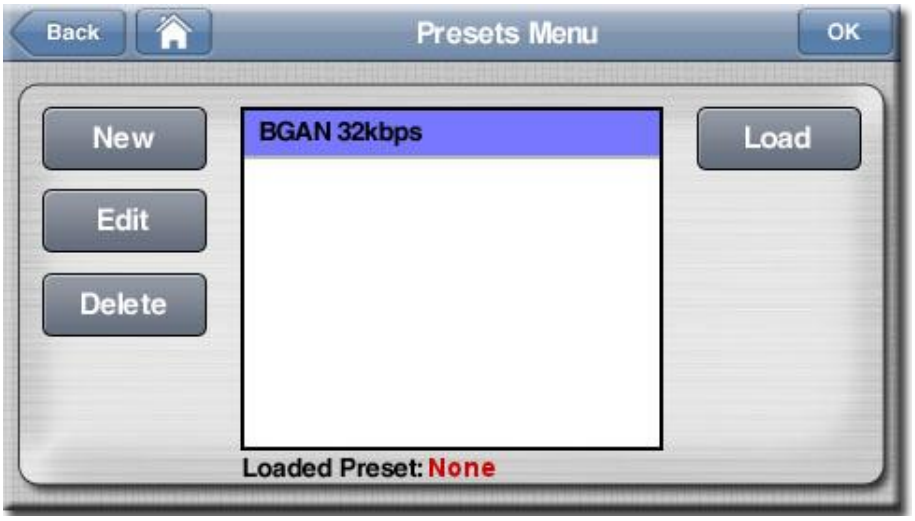


- **Blink:** this option makes the SYS led blink in the front panel of a Nerea card. This option is useful to identify in a chassis which card is being controlled by the web browser.

IV.3 Preset menu

From the main screen the <Presets> menu gives access for its administration. Presets are customer configurations for the audio encoders which can be turned active with a simple click.

Not only the encoder parameters can be configured; also streaming specific parameters like 3G/4G or Ethernet, bit rate automatic adaption or FEC can be defined in advance. For this reason, this is a very convenient configuration option for non-technical users.



- New *(Create a new preset)*
- Edit *(Edit an existing preset)*
- Delete *(Delete an existing preset)*
- Load *(Loading a preset means turning this configuration active)*

IV.4 Monitor menu

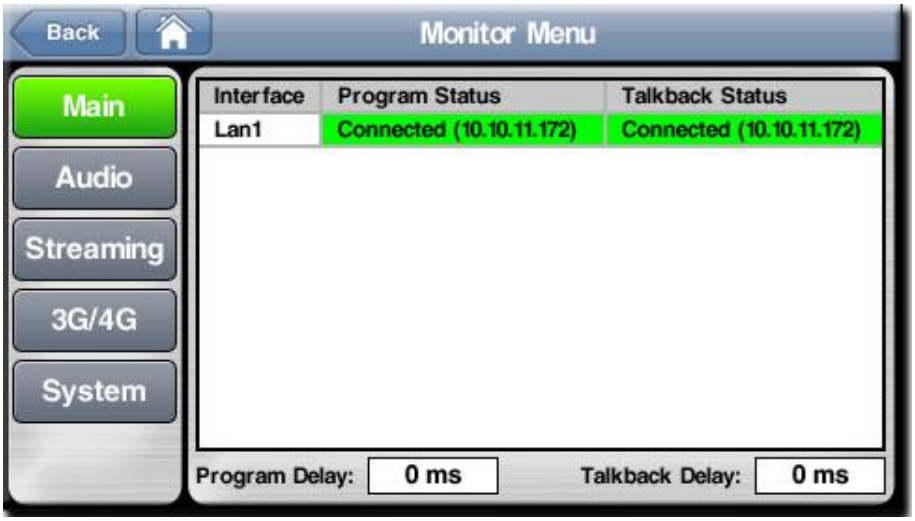
The user can check in real time many different parameters regarding the streaming operation and network performance thru the monitor menu.

The monitor menu is divided into six sections for specific information:

- Main *(Line Status, buffers)*
- Audio *(TX /RX VU-meters, audio compression status)*
- Streaming *(Complete statistics of the transmission)*
- 3G/4G *(Availability and strength of 3G/4G networks)*
- System *(local IP address, temperatures, feeding voltage)*

IV.4.1 Main monitor screen

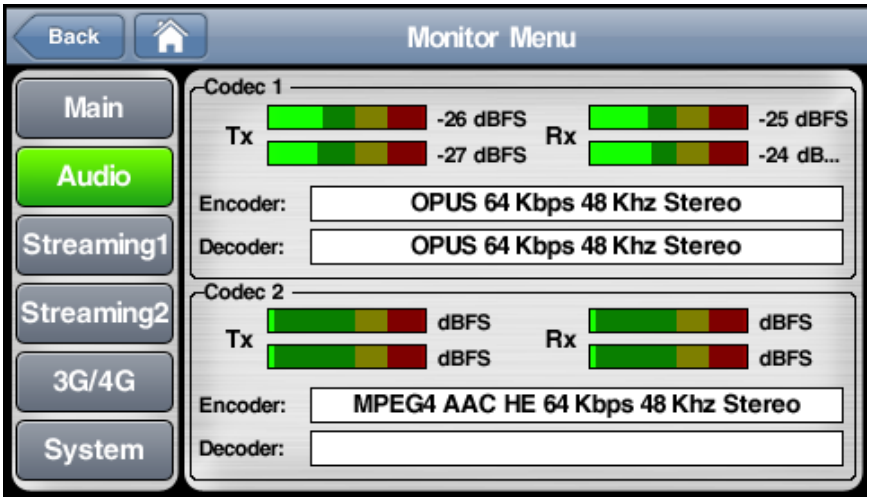
This screen shows the status of the lines contributing to the PROGRAM and TALKBACK communications. This status may be *Connected / Reconnecting / Disconnected / No physical line*.



This screen also shows the delays of Program (Codec 1) and Talkback (Codec 2) communications.

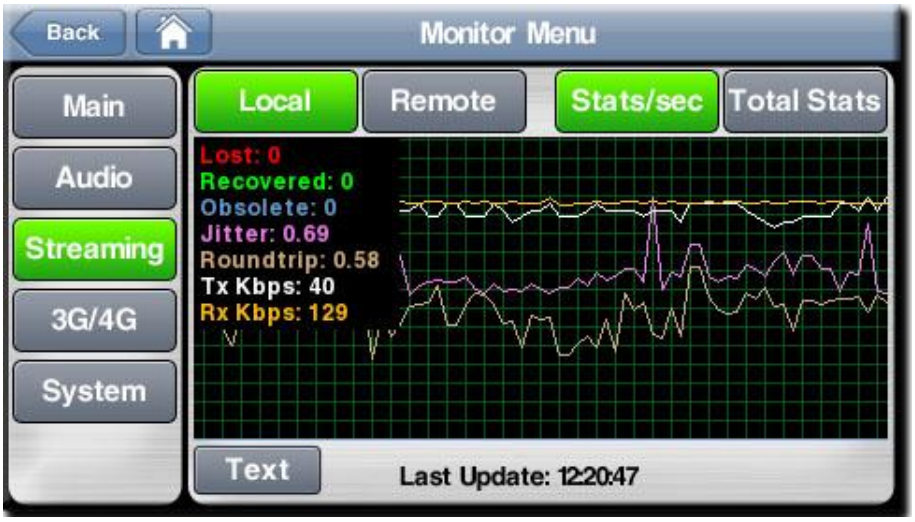
IV.4.2 Audio monitor screen

L & R audio VU-meters for the Program (Codec 1) channel and the VU-meter of the talkback (Codec 2) channel give indication of the transmitted and the received audio level in dBFS.

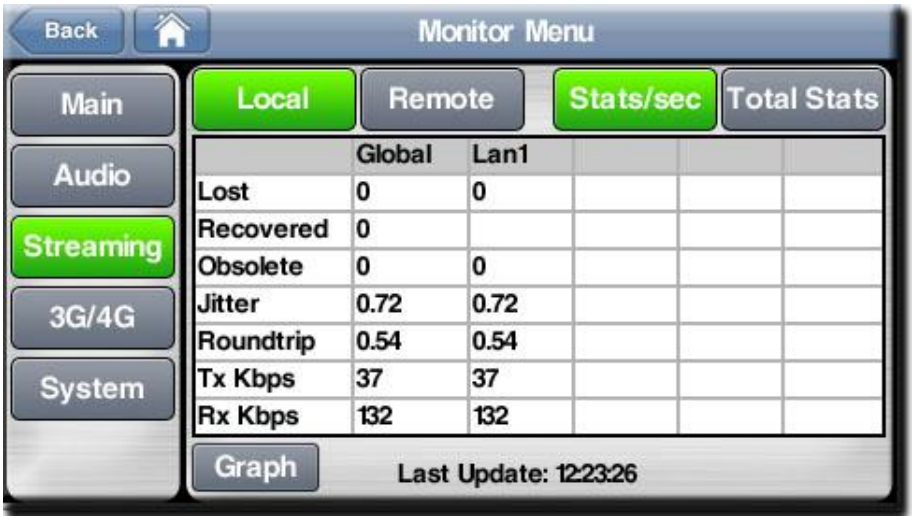


IV.4.3 Streaming monitor screen

The <Streaming> monitor displays the foregoing performance indications for both communication parties. In double codecs there is a separate screen for each communication.
Data can be shown in text or in graphical mode.



The **<Local>** field indicates the performance of the incoming stream, while the **<Remote>** field indicates the measured performance at the remote destination. Therefore the local user gets a valuable feedback about the reception quality at the destination.



The statistic data can be offered in two different ways:

- If the **<Stats/sec>** button is selected, the data appearing on the screen are an average value measured every second.
- If the **<Total Stats>** button is selected, the data shown are the accumulated values from the beginning of the communication.

The statistical data is presented for the whole communication ("Global" column) but also for each interface through which the communication is done in case 3G/4G or Custom bonding is used.

The **<Lost>** field accounts every streaming IP packet which does not arrived to the receiver. A large account will lead to a poor audio quality.

The **<Recovered>** packets field gives some feedback about the number of packets that the codec is able to recover.

The **<Obsolete>** packets field accounts every streaming IP packet which does not arrived to the receiver in time. It gives an indication that our RX buffer is not big enough.

The **<Jitter>** field indicates the highest jitter peak measured for this connection; it is a network impairment indication. Jitter is a time measurement in milliseconds and gives indication about instant changes of the streaming delay; low jitter usually indicates an optimal network performance.

Quantum's streaming buffer takes care about this network impairment automatically.

Some remarks about jitter:

1. *Some milliseconds of jitter are unavoidable for any streaming network.*
2. *Large jitter peaks over >200ms could mean a poor network performance, depending of the type of connection.*
3. *The jitter peaks adds additional delay to the streaming delay.* [11]

The **<Roundtrip>** field shows the time, in milliseconds, that a packet takes to go from the transmitter to the receiver, and back.

<Tx Kbps> and **<Rx Kbps>** show the bit rate of the transmitted and received streams. These values are interesting in the 3G/4G and Custom communications to know how the stream is divided into the several interfaces.

IV.4.4 3G/4G monitor screen

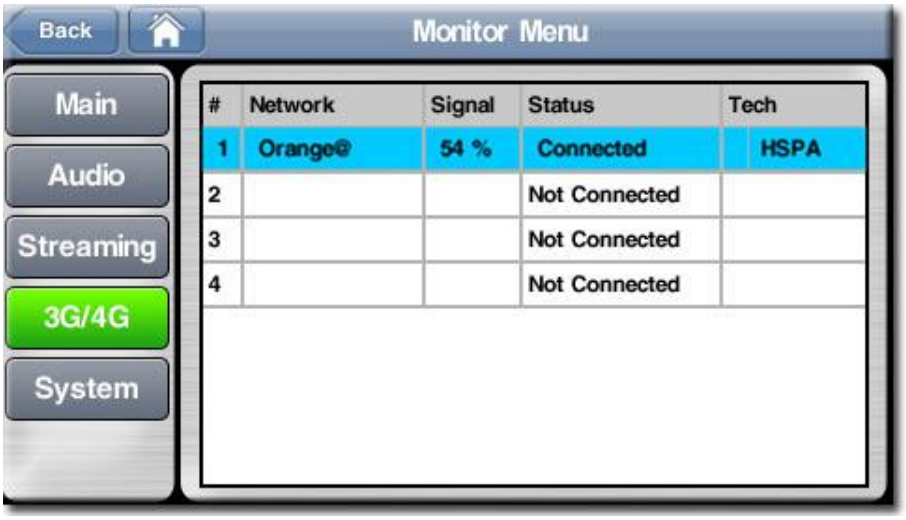
This monitor screen shows information about the 3G/4G connections available on the unit:

- Network Provider.
- Percentage of signal strength. This information is useful to estimate the quality of the link.
- Status of the connection: Connected is the only status that allows the system to stream audio over that 3G/4G data link. Any other status will indicate that the unit is either trying to register in the 3G/4G network or trying to connect to that link. NOTE: Please bear in mind that most of times, when the 3G/4G connection cannot be establish and the corresponding row in the 3G/4G monitor table shows the word registering

[11] The overall communication delay, encoder→decoder, is build up by the encoding delay plus the buffer depth. The buffer depth in turn is related to the maximum jitter.

or connecting in red color, it is most likely a problem with the 3G/4G configuration (APN, user and password), or the SIM card has not been enabled yet for data transmission or it lacks from credit to do so.

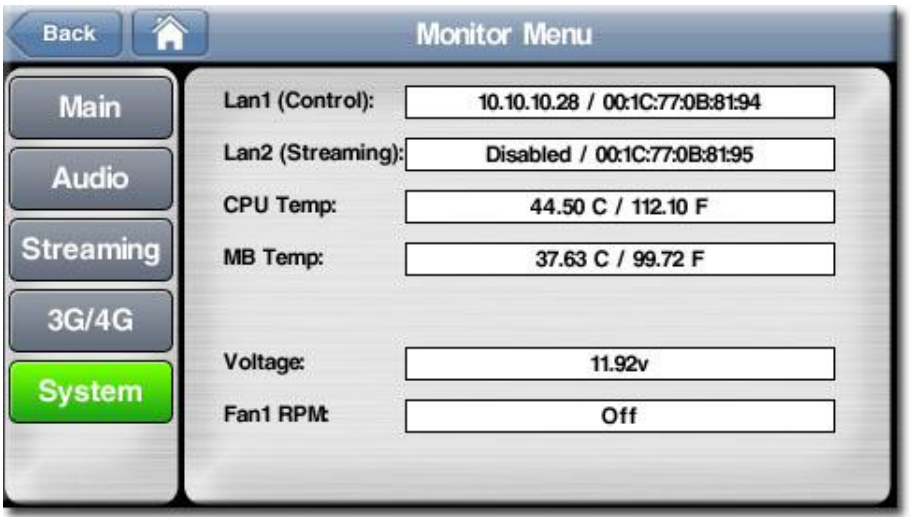
- 3G/4G Technology detected: This is the type of technology that has been detected by the unit. It could be 3G (HSPA, HSPA+, UMTS, LTE, etc).



it is possible to access the configuration screen of each 3G/4G modem from the Monitor 3G/4G screen. You only have to click on the line interface that you want to access.

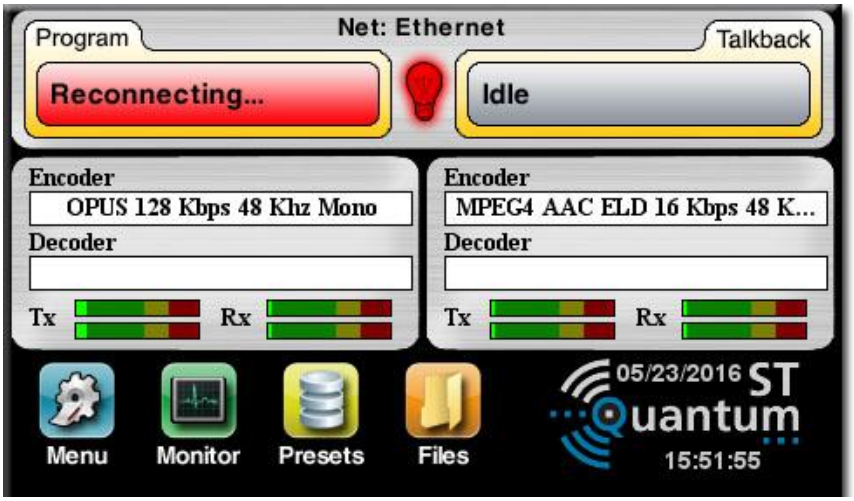
IV.4.5 System monitor screen

The System monitor screen shows some important information about the internals of the unit, like the temperature, or the voltage.



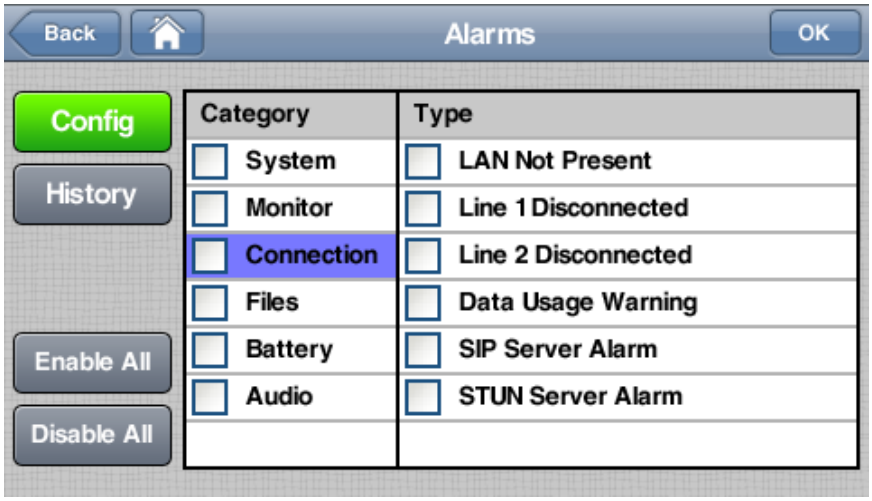
IV.5 Alarm menu

Quantum is able to check and notify many alarm events. By means on a red lamp Quantum warns the user about the occurrence of one of these alarm events.



When there is an active alarm, the background color of the title bar changes from blue to red.

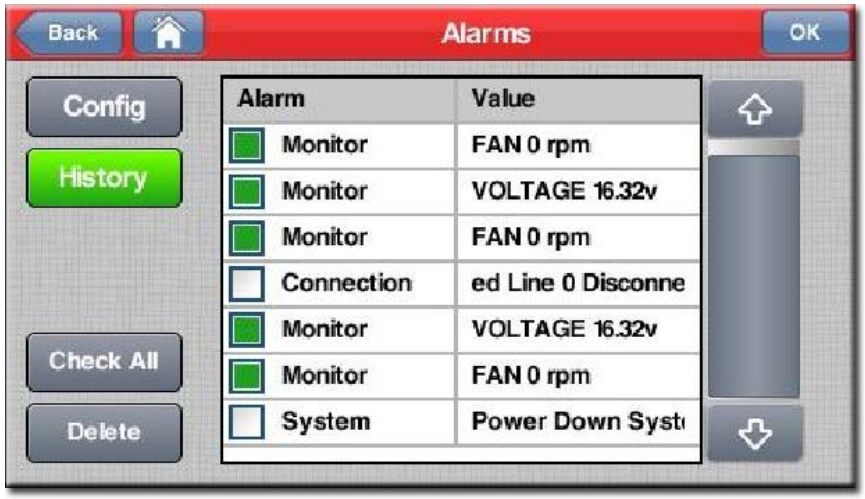
The alarm menu is opened by clicking on the lamp between the Program and the TalkBack bars.



Inside each alarm category, there is a list of possible alarms that can be selected individually clicking in its checkbox, or globally, clicking in the Category checkbox.
When not all the alarms inside a category are selected, the category checkbox is half-marked:

Clicking in the alarm type, outside the checkbox, a new window appears to select the severity level assigned to that alarm: critical, major or minor.

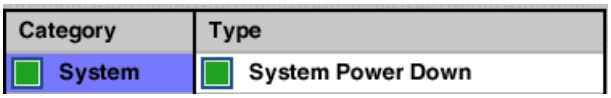
The alarm history submenu displays the alarm log as they happened.



IV.5.1 Alarms definition and configuration

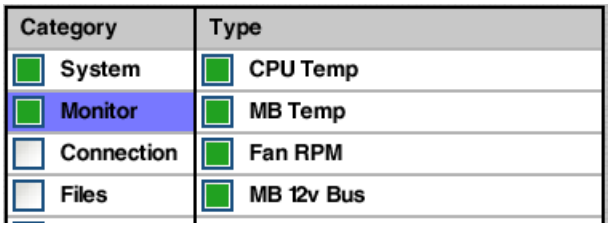
IV.5.1.1. System

This category of alarms is related to power problems:



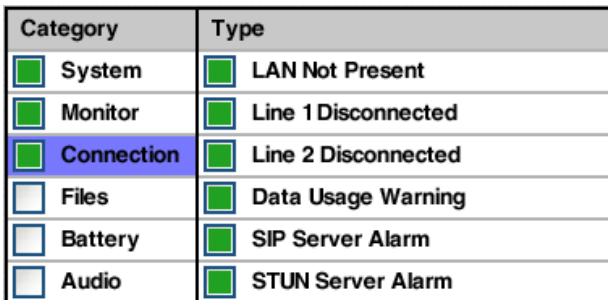
- **System Power Down:** The codec has lost power supply. This alarm is ALWAYS activated when the device is switched off, regardless of the motivation. It is useful to know if the codec is switched off and for how long.

IV.5.1.2. Monitor



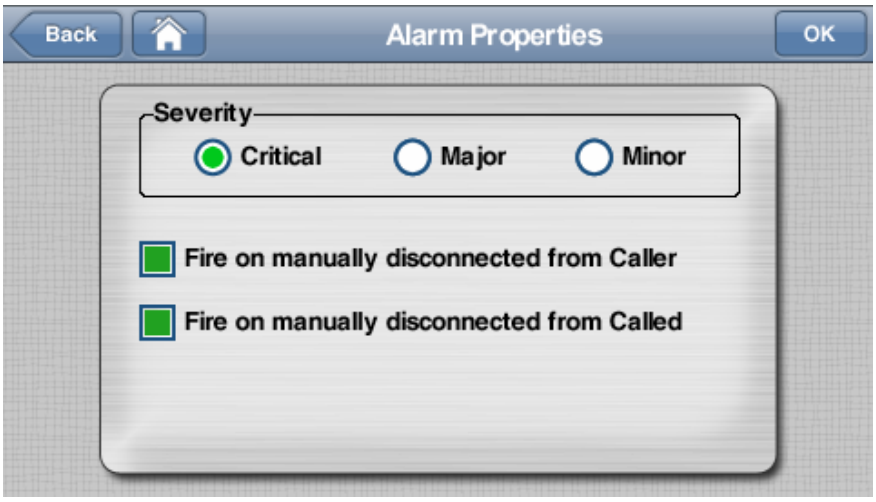
- **CPU Temp**: Active when CPU's temperature is > 70ºC.
- **MB Temp**: Active when motherboards' temperature is > 70ºC.
- **Fan RPM**: Active when fan's revolutions per minute are < 1100.
- **MB 12v BUS**: Active when bus voltage is < 12vdc.

IV.5.1.3. Connection



- **LAN Not Present**:
- **Line 1 Disconnected**: Active whenever the line is accidentally disconnected. If a "Fire on manually disconnected" option is selected, it will be also active

when the line is manually disconnected from the selected end. The origin of the disconnection can be selected by the user in the Alarms Properties menu.



- **Line 2 Disconnected**: Same as with line 1.
- **Data Usage Warning**: Active if usage is less than 5% to reach the limit established by user in Usage Configuration Menu.

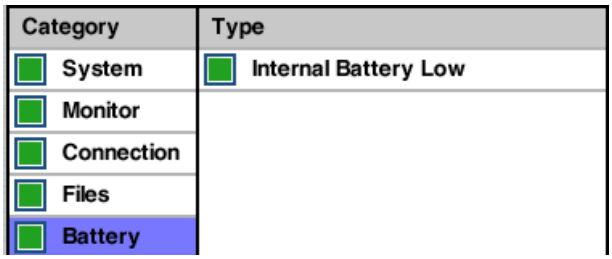
- **SIP Server Alarm**: Active if SIP server connection is lost.
- **STUN Server Alarm**: Active if STUN server connection is lost.

IV.5.1.4. **Files**

Category	Type
System	Low Free Space
Monitor	Upload Failed
Connection	AutoRec Tx Storage Not Ready
Files	AutoRec Rx Storage Not Ready
Battery	Upload Can't Connect
Audio	Download Failed
	Download Can't Connect

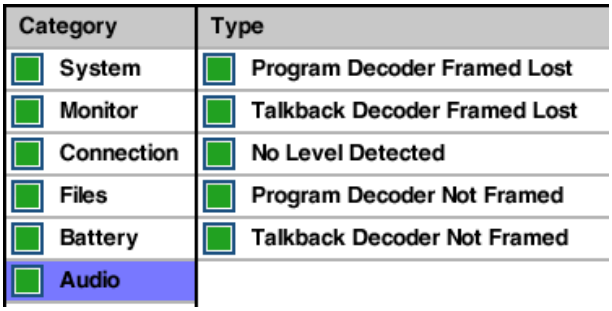
- **Low Free space**: Active when free space is <= 10% of the capacity of the selected memory device.
- **Upload Failed**: Active when the upload process fails.
- **AutoRec Tx Storage Not Ready**: Active if "Automatically record input signal when call connected" option is selected in Files->Rec menu and the selected memory device is not available.
- **AutoRec Rx Storage Not Ready**: Active if "Automatically record incoming stream when call connected" option is selected in Files->Rec menu and the selected memory device is not available.
- **Upload Can't Connect**: Active if the upload destination device is not available.
- **Download Failed**: Active if the download process fails.
- **Download Can't Connect**: Active if the communication with the server or the remote Quantum device from which the files have to be downloaded is not possible.

IV.5.1.5. **Battery**



- **Internal Battery Low**: Active when the battery is not charging and the charge percentage is <= 20%.

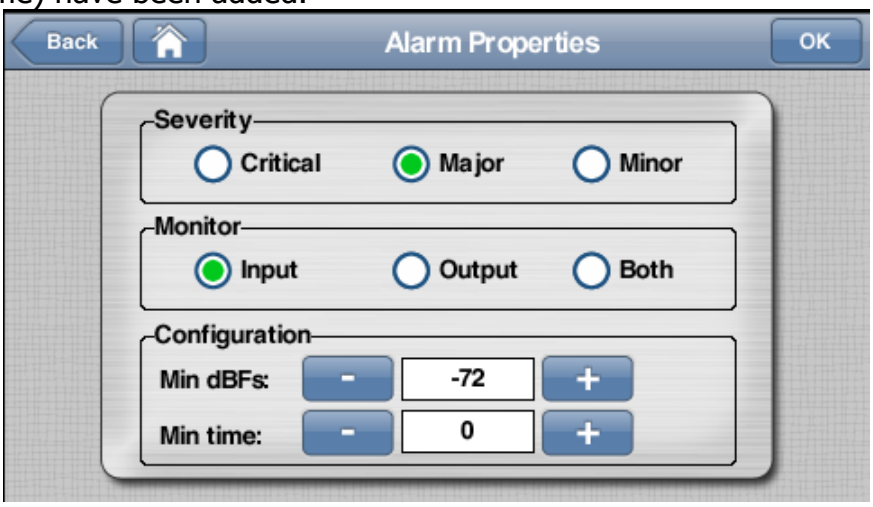
IV.5.1.6. **Audio**



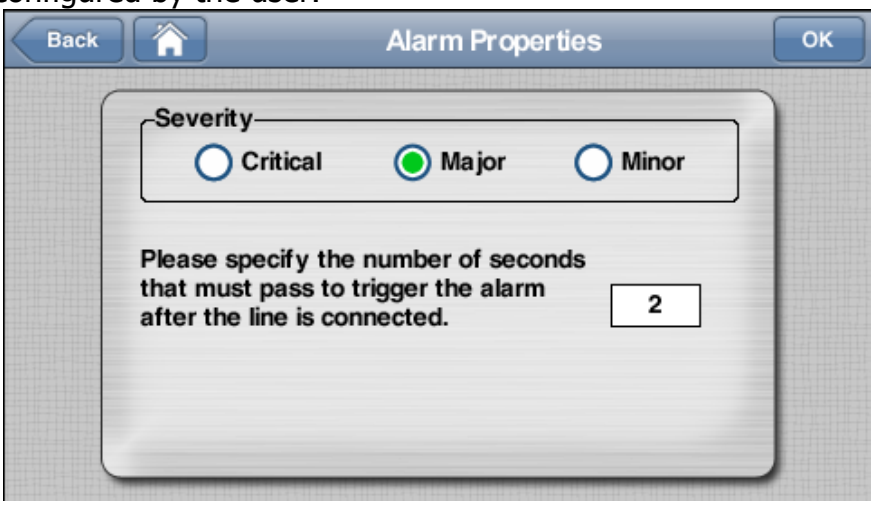
- **Program (Codec 1) decoder framed lost**: Active when a decoder loses synchronization and also when the codec is powered off. The alarm will show date and hour of the synchronization loss. The alarm will be maintained

active until the codec is synchronized again after rebooting. If synchronization is not possible, the alarm will be finished.

- **Talkback (Codec 2) decoder framed lost**: Same, but associated with Talkback or Codec 2.
- **No level Detected**: This alarm detects the absence of audio in input or output. The configuration is done through Severity selection menu, in which monitoring (input, output or both) and thresholds (minimum audio level and waiting time) have been added.



- **Program (Codec 1) Decoder not framed**: Active if, once the codec is connected, the synchronization is not achieved after a time window, in seconds, configured by the user.



- **Talkback (Codec 2) decoder not framed**: Same, but associated to Talkback or Codec 2.

IV.6 File menu

Quantum is a live contribution system, but the File menu adds further powerful capabilities to this product.

Users might perform the following actions:

- **Recording off-line** audio for later edition and transmission. The audio will be uncompressed and the resulting file will have the .wav extension.
- **Recording a backup file** of important contributions while streaming the live event to the studio.
- **Uploading** recorded files via standard FTP to a file server or to another Quantum system. When sending files from Quantum to Quantum ST (not to a standard FTP server), for example, bandwidth aggregation of up to 7 different IP interfaces is allowed, thus providing very fast file uploading. Once files reach Quantum ST at the remote end, the studio codec can be configured to upload any incoming files automatically to a FTP server that might be located in the same internal network, or can be read from a computer located in the same LAN.
- **Downloading** files from studio for later edition.
- **Sharing local files** to any Windows computer (Windows Samba protocol compatible).
- **Audio File Backup**: When this option is selected, a file previously selected will play continuously (loop) when a line drop is detected.

IV.6.1 File recording menu

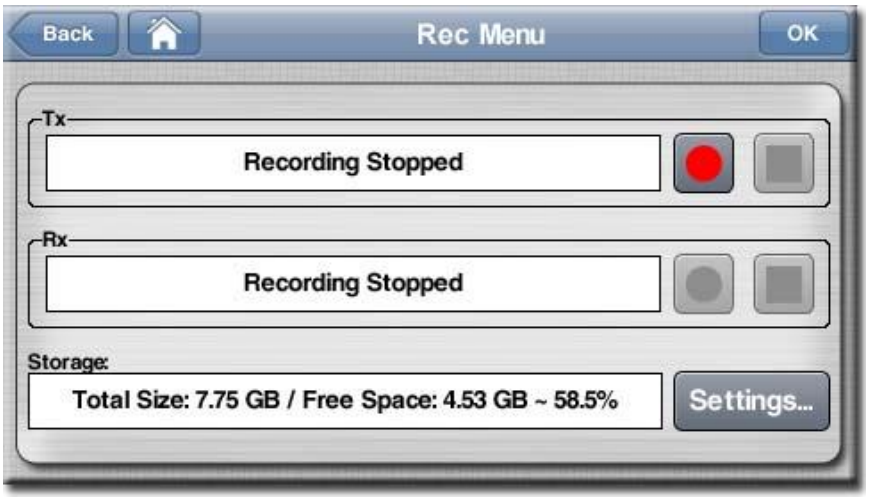
There are two different sources of recorded data:

- The PROGRAM channel content.
- The incoming stream from a remote codec.

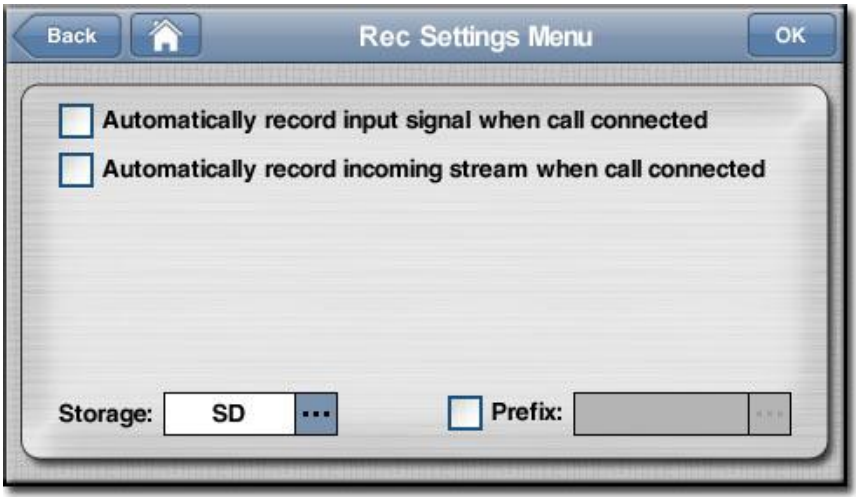
Both sources can be selected, even simultaneously, to create new recorded files. The format of the audio file will be always uncompressed audio, with .wav extension.

Three actions might be controlled from this menu:

- To start the file recording in the available storage system, press on the button with the red circle. The file will get a name automatically, but that name can be edited later on by the user.
- Stopping the recording is performed by pressing the button with the grey square on it.



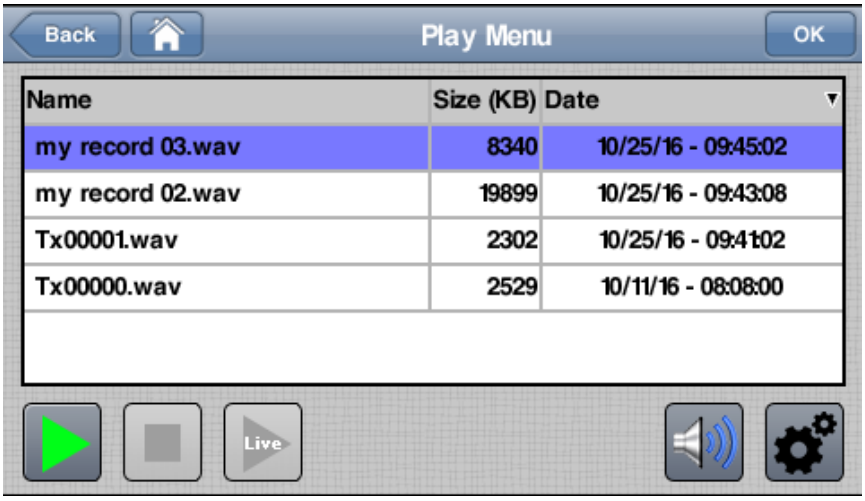
- Set the storage device and the automatic recording options pressing 'Settings' button. In this menu it is possible to enable the 'Automatically record input signal when call connected' option that sets the unit to record all audio contents being streamed on the line on a file. If 'Automatically record incoming stream when call connected' option is selected, the received stream will be automatically recorded in .wav format.



In this menu it is also possible to add a prefix to the generated audio files.

IV.6.2 File play menu

To play a file, just select one among the available files and use the play and stop buttons on the tool bar at the bottom to start and stop playing the file.

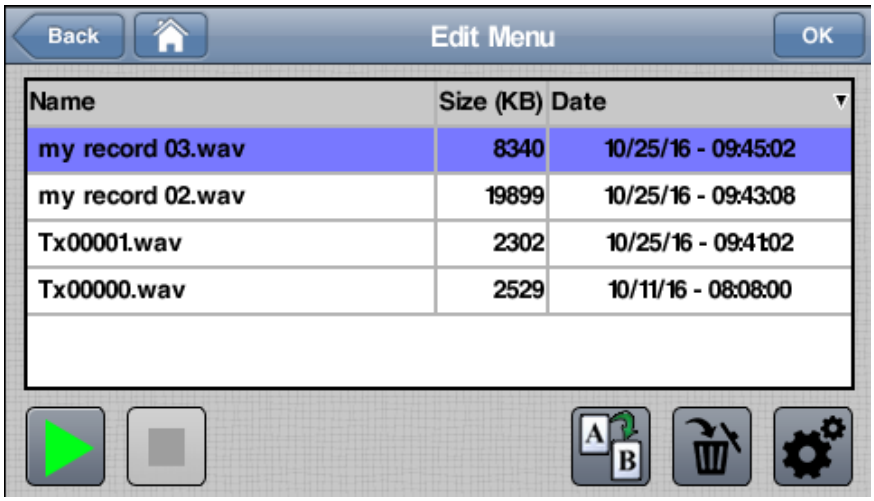


This menu also offers the possibility of playing and mixing a file during an audio transmission, pressing the play-live button. The gain of the file playback can be controlled pressing the speaker button on the right side.

The play menu provides also one icon for selecting the storage device.

IV.6.3 File edit menu

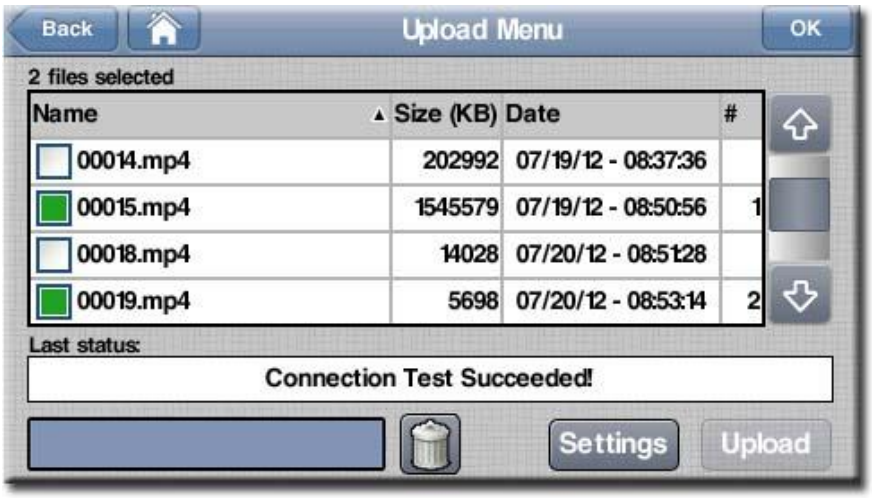
The edit menu makes possible to rename or delete the selected file using the buttons on the right side. The storage device can also be selected.



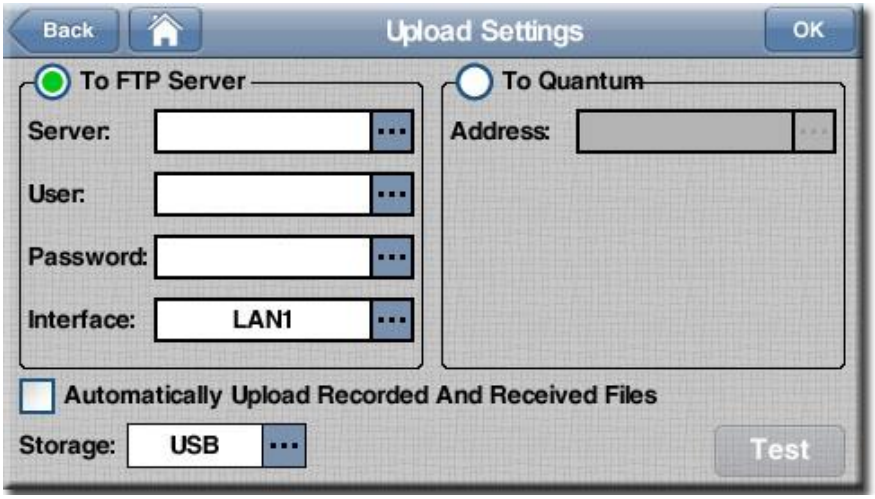
In portable codecs it is possible to make a top-and-tail edition of the recorded files using the touchpanel.

IV.6.4 File upload menu

This menu gives access to the embedded FTP client which can upload via FTP protocol one or several files. The order of the files in the list can also be configured.



Any file accessible by the unit in any of its storage means might be uploaded to an FTP server or to the internal storage of a remote Quantum. This can be configured in the <Settings> submenu.



In the <Settings> submenu the uploading destination can be chosen between an FTP server and a remote Quantum.

If a FTP server is chosen, the FTP address and the user access rights must be filled in. Those are permanently valid unless modified by the user. It is also possible to specify which communication interface is going to be used: LAN1, LAN2 or 3G/4G. If 3G/4G is chosen, the transmission will be done through a single 3G/4G data wireless connection, the first available following an increasing

order, as it is not possible to upload any files to an FTP server using 3G/4G 'bonding' (bandwidth aggregation).

If FTP upload to a remote Quantum unit is chosen, it is sufficient to indicate the IP address of that codec. The communications interface will be the one being in use by the codec at the time, so this mode does allow the usage of 3G/4G bonding or even "custom" bonding, thus performing bandwidth aggregation and resulting in a much faster file transfer than with any conventional IP link.

Once the file is in the remote Quantum unit, a second FTP upload can be configured to transfer automatically any incoming files to a FTP server automatically.

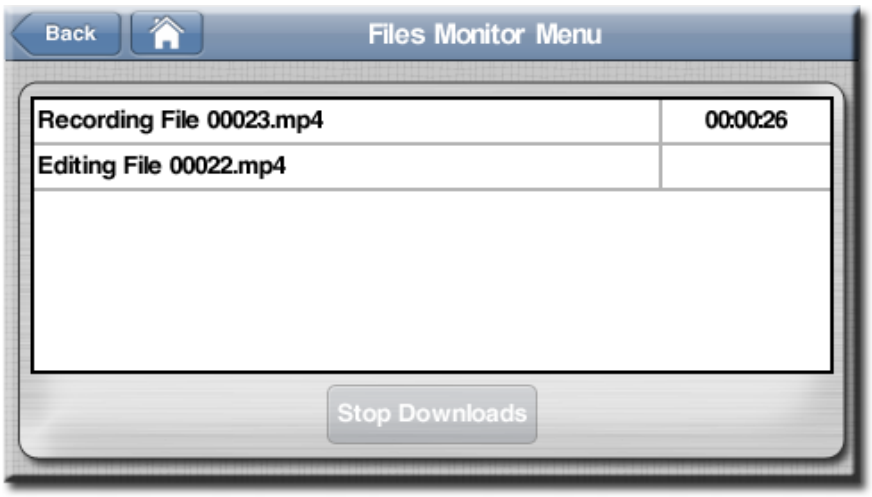
In fact, the option 'Automatically upload recorded and received files' enables the automatic transmission of any files that have been recorded or received from other Quantum units. The file transmission will be performed according to the Upload Settings menu.

The storage device where the files are to be selected can be chosen also in this FTP Upload Settings menu.

IV.6.5 File Monitor

The File Monitor Windows allows the user to monitor any activity regarding files, like uploads, transmission, reception or recording.

The File Monitor window will show and describe the different operations carried out in real time regarding files.



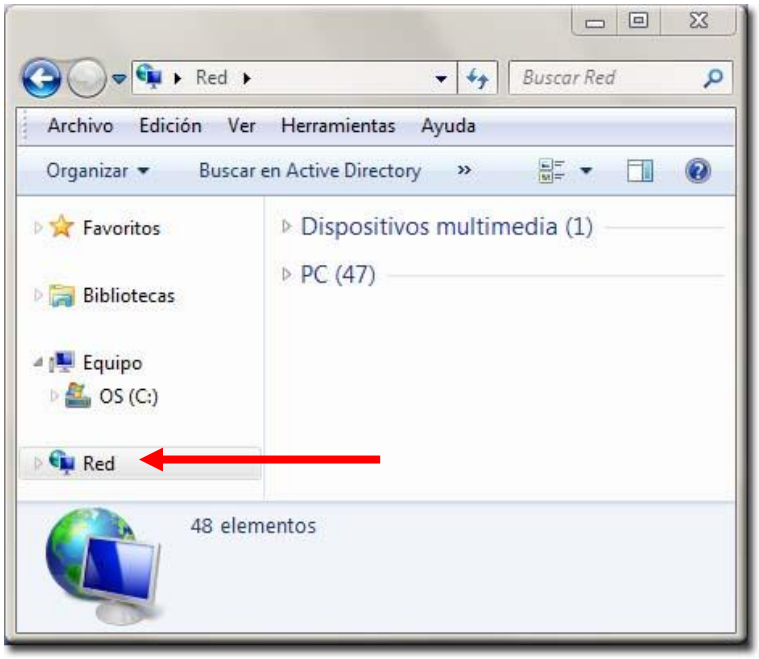
IV.6.6 Network share

It is possible to share files stored on a Quantum codec on a local network so they can be accessed from a Windows compatible computer for reading, deleting or copying. The different media storage systems available on the unit will be shared as a regular Windows network drive. To do this, you must enable the "Enable file sharing" option in Network Share menu.

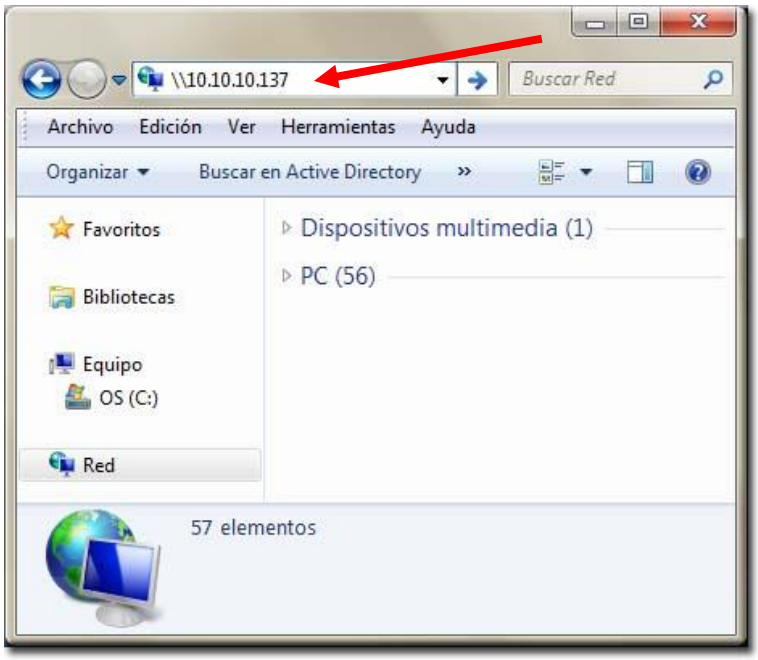
The procedure for accessing the Quantum storage unit from a computer running Windows connected to the same local network is:

1. Open a Windows Explorer application window:

2. Click on the Network area, on the window's left side:



3. Write the Quantum IP address on the dialog bar, preceded by two backslashes.



4. The Storage folder will appear in the right area of the Windows Explorer window. From this moment on the contents of the external storage unit of the codec will be available from this application.

IV.6.7 Audio File Backup

When this option is enabled, a file previously selected will play continuously (loop) when a line drop is detected.

The file to be played can be selected from the files present in the storage device or can be uploaded to that device from the computer controlling the codec.

IV.6.8 File download

It is also possible to download a file from an FTP server and store it in a memory device inserted in the Quantum codec.

Chapter V

AUDIO ENCODING ALGORITHMS

There is a wide variety of audio encoding algorithms implemented in the codecs belonging to the Quantum Family. This variety allows a better adaptation to the different scenarios where requirements of bandwidth, delay and quality can be very different.

Algorithms	Standard
PCM	
G711 A/μ Law	
G722	
MPEG 1,2 layer II	ISO/IEC 11172-3 /13818-3
MPEG 4 AAC LC	ISO/IEC 14496-3
MPEG 4 AAC LD	ISO/IEC 14496-3
MPEG 4 AAC ELD	ISO/IEC 14496-3
MPEG 4 AAC HE	ISO/IEC 14496-3
Enhanced apt-X	
OPUS	

MPEG and OPUS decoding make use of Error Concealment tools. There are some scenarios where FEC mechanisms cannot be used due to bandwidth or delay restrictions. For those cases, a method like Error Concealment to minimize the impact of packet losses in form of artifacts on the decoded audio is the perfect complement.

V.1.1 BANDWIDTH (KHz)

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

V.1.1.1. **G711**

Bit Rate	Fs=8KHz
	Mono
64	3.6

V.1.1.2. **G722**

Bit Rate	Fs=16KHz
	Mono
64	7.5

V.1.1.3. **MPEG 1,2 LAYER II**

Bit Rate	Fs=48KHz			Fs = 32 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	4	x	x	5,16	x	x
64	10,6	4	4,8	11,7	4,9	6,1
80	14,4	x	x	13,6	x	x
96	14,4	6,1	6,1	15,1	6,1	6,1
128	20	11,3	16,3	15,1	11,7	13,6
160	20	14,5	20	15,1	13,6	13,6
192	20	14,5	20	15,1	15,1	15,1
256	x	20	20	x	15,1	15,1
384	x	20	20	x	15,1	15,1

Bit Rate	Fs=24 KHz			Fs=16 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
32	7,2	#	#	7,5	#	#
64	11,3	7,3	11,2	7,5	7,5	7,5
80	11,3	9,4	11,2	7,5	7,5	7,5
96	11,3	11,2	11,2	7,5	7,5	7,5
128	11,3	11,3	11,3	7,5	7,5	7,5
160	11,3	11,3	11,3	7,5	7,5	7,5

V.1.1.4. MPEG 4 AAC LC

Bit Rate	Fs=48KHz			Fs = 32 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	20	7,5	10,5	15	8	12
80	20	7,5	10,5	15	8	12
96	20	7,5	10,5	15	8	12
112	20	7,5	10,5	15	8	12
128	20	20	20	15	15	15
160	20	20	20	15	15	15
192	20	20	20	15	15	15
256	20	20	20	x	15	15
384	x	20	20	x	15	15

Bit Rate	Fs=24KHz		
	Mono	Stereo/Dual	JStereo
64	11,2	8,2	10,5
80	11,2	8,2	10,5
96	11,2	8,2	10,5
112	11,2	8,2	10,5
128	11,2	11,2	11,2
160	x	11,2	11,2
192	x	11,2	11,2
256	x	11,2	11,2

V.1.1.5. MPEG 4 AAC LD

Bit Rate	Fs=48KHz			Fs = 32 KHz			Fs=24 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
64	13,5	#	#	13,9	#	#	11,2	#	#
80	13,5	4	5,4	13,9	4,9	4,9	11,2	4,8	4,8
96	13,5	4	5,4	13,9	4,9	4,9	11,2	4,8	4,8
128	16	13,5	14,2	14,9	13,9	14,1	11,2	11,2	11,2
160	16	13,5	14,2	14,9	13,9	14,1	11,2	11,2	11,2
192	16	13,5	14,2	14,9	13,9	14,1	11,2	11,2	11,2
256	20	16	16	14,9	14,9	14,9	11,2	11,2	11,2
384	20	16	16	14,9	14,9	14,9	#	#	#

V.1.1.6. MPEG 4 AAC HE

Bit Rate	Fs=48KHz			Fs = 32 KHz		
	Mono	Stereo/Dual	JStereo	Mono	Stereo/Dual	JStereo
24	12	#	#	12,3	#	#
32	16,8	#	#	15	#	#
40	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
80	20	16,8	20	15	15	15
96	20	16,8	20	15	15	15
112	20	16,8	20	x	15	15
128	20	20	20	x	15	15

V.1.1.7. MPEG 4 AAC ELD

Bit Rate	Fs=48KHz		
	Mono	Stereo/Dual	JStereo
16	7	#	#
24	13.1	#	#
32	16,9	#	#
40	16,9	7	7
48	18,4	13,1	15,4
56	18,4	13,1	15,4
64	20.2	16,9	18,4
80	#	16,9	18,4
96	#	18,4	18,4
112	#	18,4	18,4
128	#	20.2	20.2

V.1.1.8. Enhanced Apt-X 16 bits

Bit Rate	Fs=48KHz		Fs = 32 KHz		Fs=16 KHz	
	Mono	St/Dual	Mono	St/Dual	Mono	St/Dual
64	x	x	x	x	7.5	x
128	x	x	15	x	x	7.5
192	20	x	x	x	x	x
256	x	x	x	15	x	x

384	x	20	x	X	x	x

V.1.1.9. **Enhanced Apt-X 24 bits**

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

V.1.1.10. **OPUS**

Bit Rate	Fs=48KHz	
	Mono	ST/Dual
8	4	x
12	4	x
16	6	x
20	8	x
24	12	x
28	12	x
32	20	12
40	20	12
48	20	20
56	20	20
64	20	20

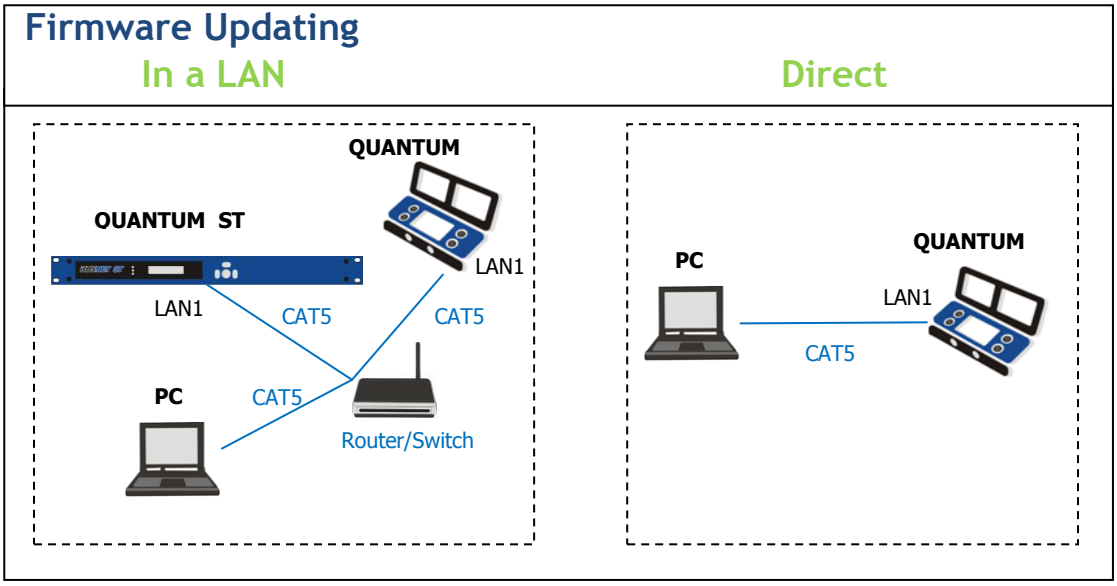
Chapter VI

UPDATING THE FIRMWARE

To update the firmware of a Quantum unit follow these steps:

1. Make sure your Quantum and the computer to be used for the update are connected to the same LAN. Of the two Ethernet interfaces available in Quantum, the update operation is always done through the labeled LAN1:

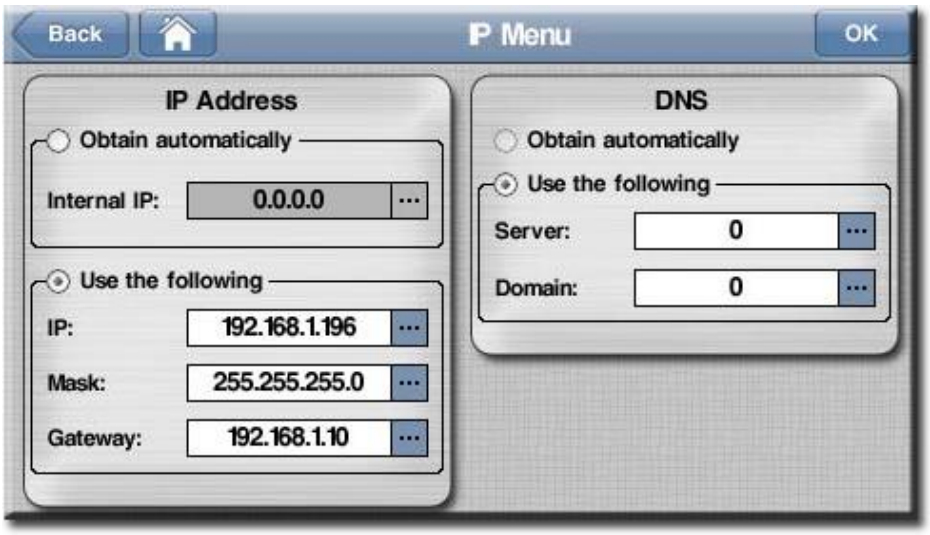
2. If you have not set up a LAN network, it is also possible to connect the computer and the Quantum LAN1 port directly through a standard or crossover CAT-5 cable. The Ethernet port on the computer automatically adjusts to the type of cable.



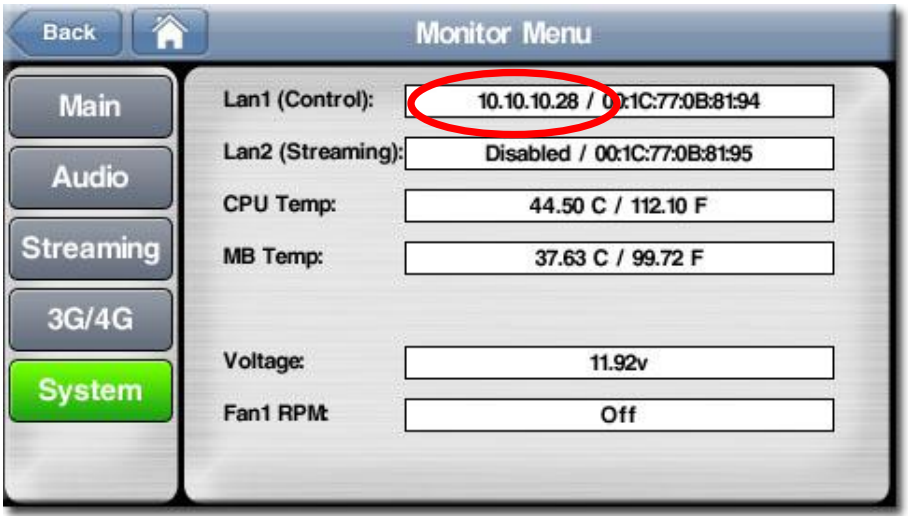
3. Make sure the Quantum LAN1's (Control) and the PC's IP addresses belong to the same subnet (IP address segment). If this is not the case and you need to modify Quantum's IP address, follow these steps:

    a. Enter the IP menu by clicking the following icons on the touch screen menu or web page:

    b. The codec's IP address can be entered manually or can be obtained automatically:



i. If automatic IP address is chosen, a DHCP server will provide Quantum with an appropriate IP address value for the subnet where it is connected. To know the value of the address assigned, the monitoring system screen can be consulted:

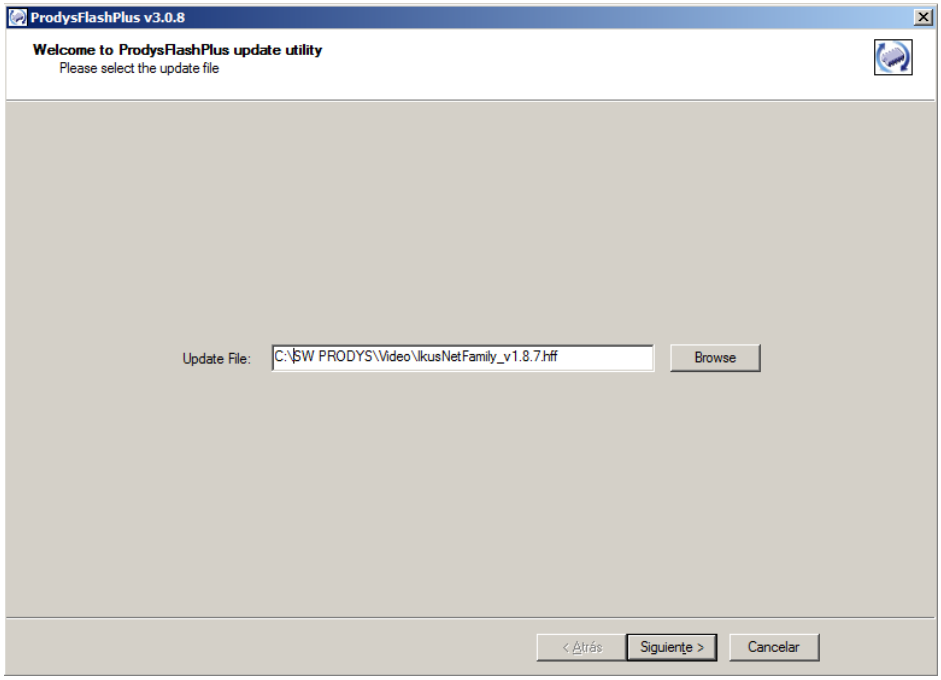


ii. If the manual option is chosen, you must enter values for IP address and mask consistent with the subnet where the PC is.

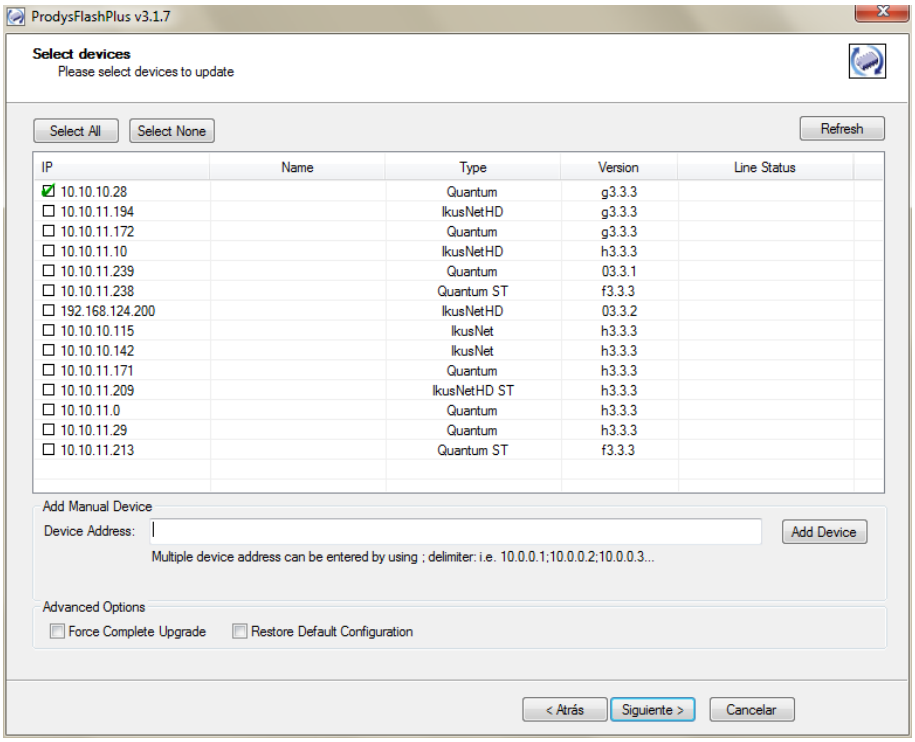
4. Ensure that the PC is able to access the codec opening the control page from the Internet Explorer by entering the LAN1 IP address in the address bar.

5. Start up the ProdysFlashPlus application.

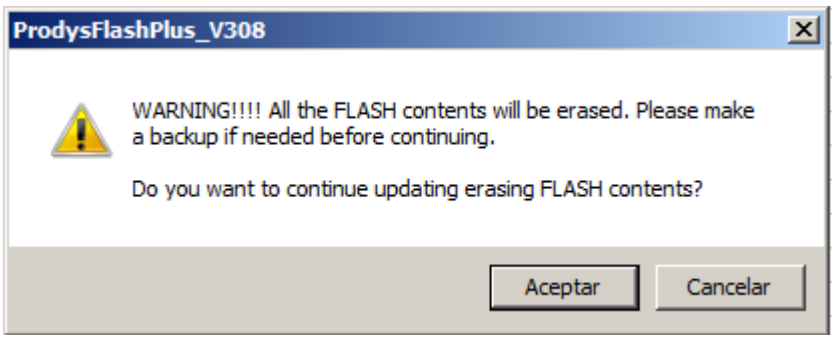
6. Click on browse and select the file with the new firmware version.



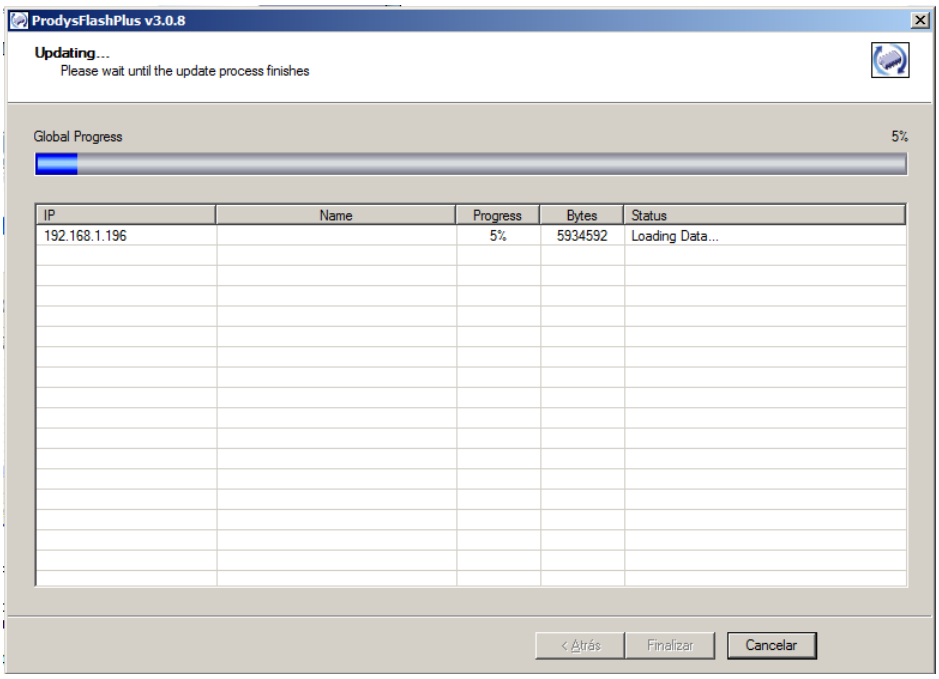
7. All the units connected to the computer's network will be detected automatically. Look in the first column the IP address configured on the unit's LAN1. Select the codec checking the box. If you are upgrading more than one codec, select all the boxes for the codecs you want to upgrade.



- 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.

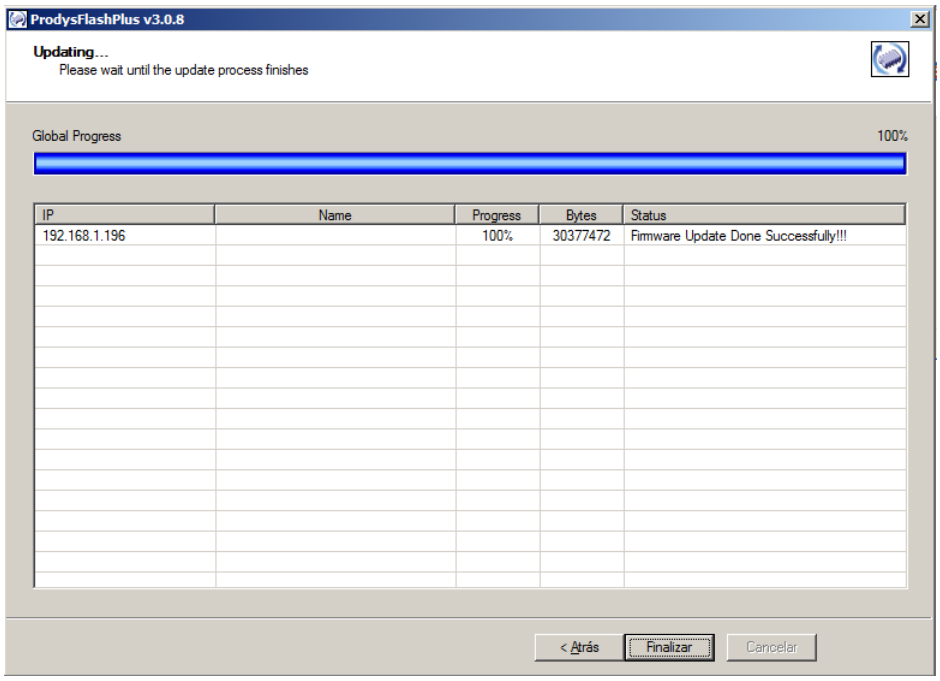


- By clicking on next, the upgrading process will start.



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


ProdysFlashPlus v3.0.8
Updating...
Please wait until the update process finishes
Global Progress
100%
IP
Name
Progress
Bytes
Status
192.168.1.196
100%
30377472
Firmware Update Done Successfully!!!
< Atrás
Finalizar
Cancelar

Chapter VII

DETAILED PORTS DESCRIPTION

There are two main groups of ports: Management and Streaming ports:

VII.1 Management Ports

ProdysFlashPlus

UDP 50013: This port is used by the ProdysFlashPlus upgrading application to allow the communication with Prodys IP units.

ProdysControlPlus

TCP 54001: This port is used for accessing the codec through the ProdysControlPlus Management Software.

Web Page

TCP 80: This port is used for web control via HTTP (web browser).

VII.2 Streaming Ports

TCP 53000: P-RTCP (Control): This port is used to control any IP streaming communication. This port can be disabled. In this case, there will be a lack of control in the status of the connection at the far end, and both ends will have to initiate the connection.

UDP 53000: P-RTP Unicast PROGRAM audio and auxiliary data channel: This port is used to carry the audio and auxiliary data over IP for unicast communications on the PROGRAM line.

UDP 53001: P-RTP Unicast TALKBACK audio channel: This port is used to carry the audio data over IP for unicast communications on the talkback channel.

Type of Ports	DEFAULT PORTS			
Management Ports	**Web Page** TCP 80	**Prodys Control Plus** TCP:54001	**Prodys Flash Plus** UDP:50013	**External Protocol (API/SDK)** TCP: 50031
Streaming[12] Ports by Protocol	TCP 53000 UDP 53000 UDP 53001		UDP 5060 UDP 5004	
	Prodys BRAVE		SIP	

[12] Default streaming ports: They can be changed by the user.