Support Department	APPLICATION NOTE			
Subject	**Prodys IP FAQ**			
Author	Pedro Parrilla	Ver.: 1.0.3	Creation date:	**24/07/2006**
Doc.	Prodys IP FAQ.doc	Pages: 36	Modification date:	**01/02/2008**

INDEX

INTRODUCTION

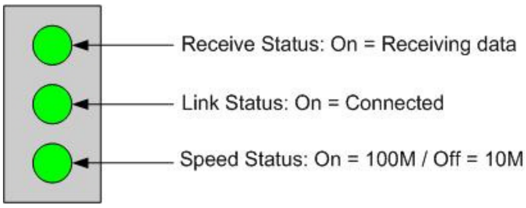
This chapter is aimed at providing a fundamental understanding of Prodys IP Codecs operation and at describing some solutions to common problems when using Prodys IP Codecs for the first time.

1.- ETHERNET CONNECTION

Prodys IP Codecs come equipped with an 10/100BaseTx standard Ethernet port with auto-negotiation in the starting stage.

When connecting to another Ethernet device, Prodys IP Codecs can detect automatically the network speed: 10 or 100Mbps. Next to the socket there are three LEDs that indicate different states for the connection and these are very useful in problem-solving situations.

LAN LED's:



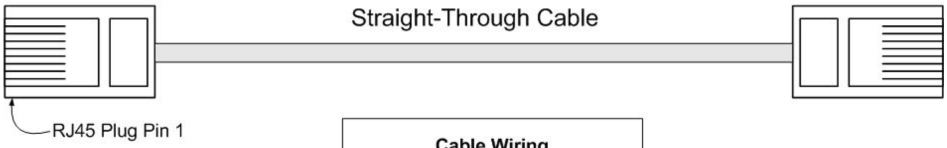
However, regarding the 'duplex' mode, Prodys IP Codecs can detect this automatically when starting, but this mode cannot be changed after that (on the fly) till the next starting. In addition, the unit will set full duplex mode by default when the unit is not connected to any Ethernet device when starting. For further information about this, please, read the Ethernet Overview.ppt, available in the Prodys site download section.

Prodys encourage the user not to use old 10Mbps half-duplex hubs with their IP Codecs, due to the low speed and performance of the half-duplex technology.

Note: Direct connection between a Prodys IP Codec and a PC needs "crossover" CAT5 cable. If you use a switch or hub normal CAT5 cables can be used!!!
Note: From version 5.2.1 onwards, it is possible to configure the speed and duplex settings to a fixed value (not auto-negotiation).

1.1.- CONNECTING TO A HUB OR SWITCH

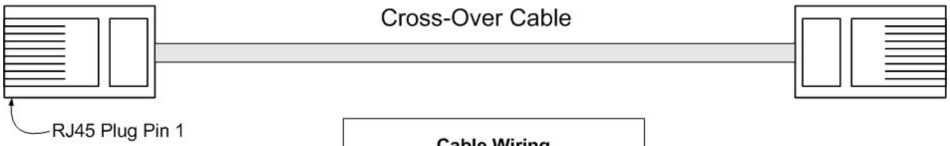
In the majority of cases you can simply connect the unit's LAN port to your Ethernet network's Hub or Switch using an Ethernet cable (CAT5). In this case you should use a standard 'straight-through' Ethernet cable (not a 'cross-over' cable). This kind of cable can normally be found in any IT shop. In any case, this cable is described in more detail below:



Cable Wiring	
Prontonet	**Hub**
Tx+ 1	1 Tx+
Tx- 2	2 Tx-
Rx+ 3	3 Rx+
Rx- 6	6 Rx-

1.2.- CONNECTING TO A PC

In some cases, such as when you configure the equipment, it is possible that you will want to connect the unit directly to a PC. In this case the PC must have a free Ethernet port to connect to and you must use a 'cross-over' Ethernet cable. Again, any good IT shop will stock these cables. This time the wiring is as follows:



Cable Wiring	
Prontonet	**PC LAN card**
Tx+ 1	3 Rx+
Tx- 2	6 Rx-
Rx+ 3	1 Tx+
Rx- 6	2 Tx-

2.- IP: PROTOCOLS, ADDRESSES, PORTS AND ROUTER CONFIGURATION

The operation of the Prodys IP Codecs for audio streaming offers two operational modes: UNICAST and MULTICAST.

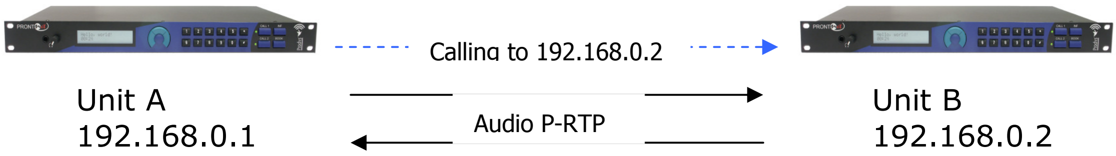
From version 5.2.1 onwards, Multi-Unicast is also available. With Multi-Unicast, point to multi-point connections are available. The same compressed audio (one encoder) can be sent to up to 10 different locations at the same time by means of Unicast connections. This solution overcomes the lack of Multicast support by some networks.

2.1.- UNICAST COMMUNICATIONS

The term UNICAST is used in the networking world to refer to the connection to a single destination. Applied to the Prodys IP Codecs, this is when a point-to-point connection is created between two units uni- or bidirectionally.

2.1.1.- ESTABLISHING A UNICAST CONNECTION

The procedure for establishing a connection is very similar to that of making an ISDN call but entering an IP address of the unit we wish to connect to. For further information about IP addressing, please, read the IP Addressing Overview.ppt file, available in the Prodys site download section.
The audio data connection can be bi-directional (as in an ISDN codec) in which case we need two connections, one in each direction. For this, the unit that receives and accepts the call will automatically call back to the originating unit and establish a reverse connection.

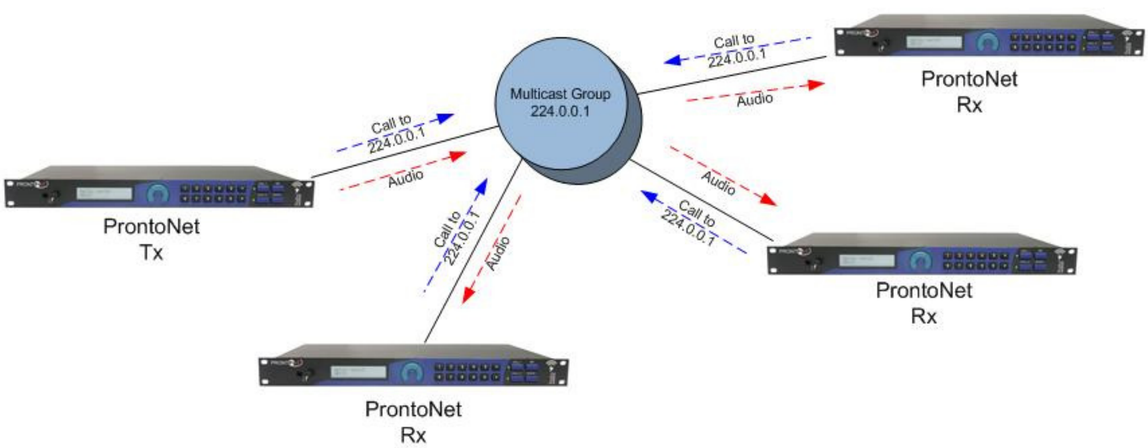


2.2.- MULTICAST COMMUNICATIONS

The term MULTICAST is used in the networking world to refer to the connection to multiple destinations. Applied to Prodys IP Codecs, this is when a point-to-multipoint connection is created between several units unidirectionally. For further information about multicast, please, read the Multicast Overview.ppt file, available in the Prodys site download section.

2.3.- ESTABLISHING A MULTICAST COMMUNICATION FROM THE PRONTONET

With MULTICAST the calls must be made from both ends. Both the sender of the data and all the receivers of the data must call to establish a connection to the multicast group. The multicast operation can be shown in the following diagram:



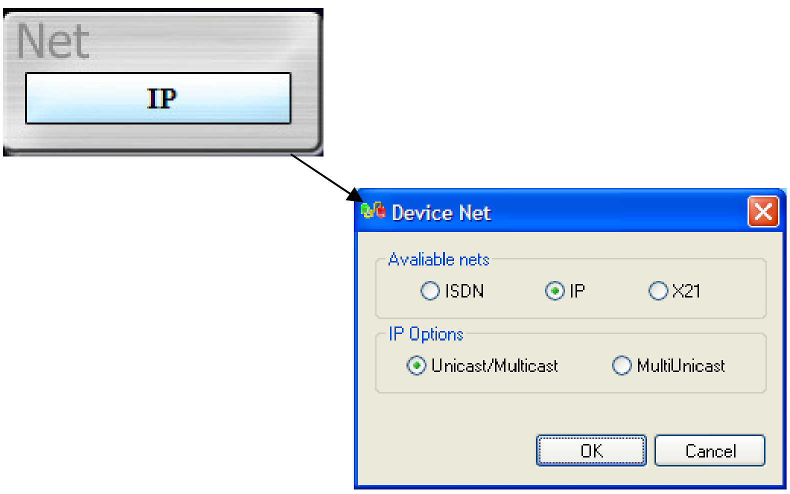
This can be done in any order, that is, calls can be set up first from the transmitter and then from each of the receivers, or the other way round.

- **Multicast considerations:**

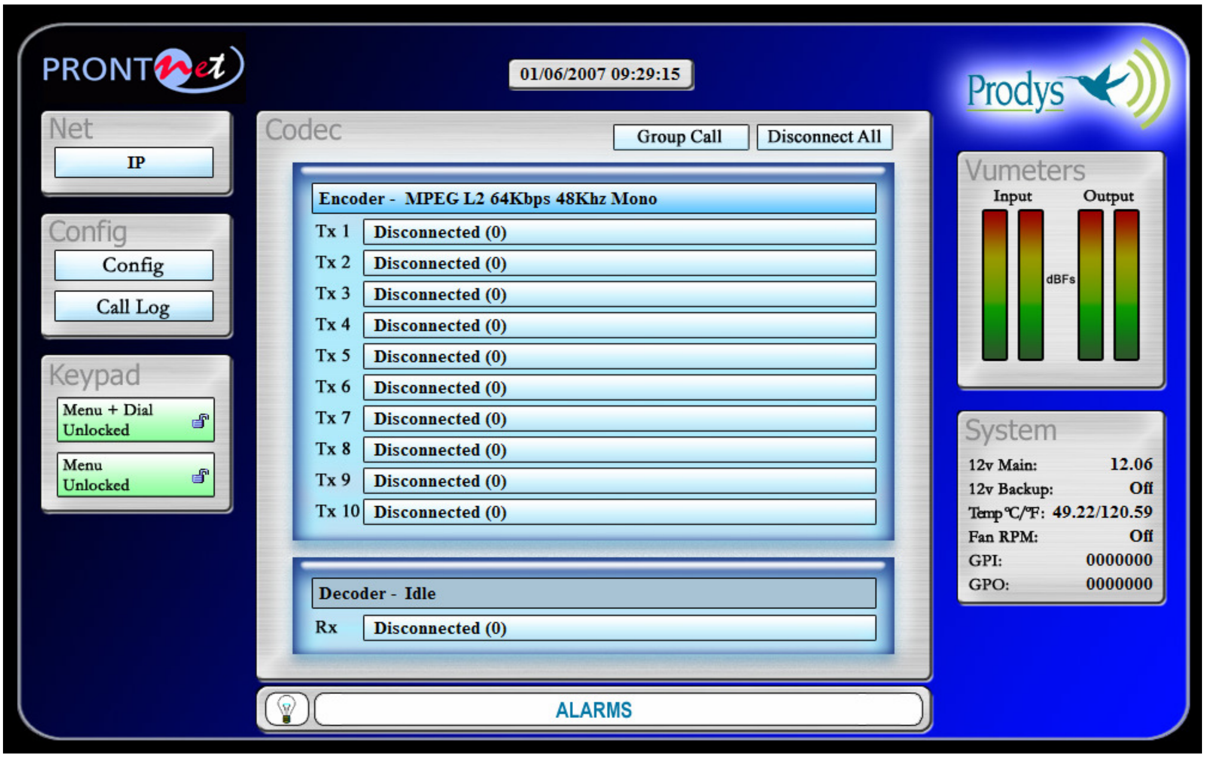
- Internet Protocol (IP) multicast is a bandwidth-conserving technology that reduces traffic by simultaneously delivering a single stream of information to thousands of corporate recipients and homes.
- Multicast is based on the concept of a group. An arbitrary group of receivers expresses an interest in receiving a particular data stream. This group does not have any physical or geographical boundaries—the hosts can be located anywhere on the Internet. Hosts that are interested in receiving data flowing to a particular group must join the group using IGMP. All this is done automatically by ProntoNet when establising a connection.
- Multicast traffic is rejected when going through the Internet, since most IP servers on the Internet do not currently support the multicasting part of the protocol, except when using VPNs, because VPN's encapsulates IP packets as unicast frames, so routers simply see an ordinary packet.
- All IP multicast group addresses will fall in the range of 224.0.0.0 to 239.255.255.255, but some of them are reserved, that's why the range of addresses from 224.0.1.0 through 238.255.255.255 are called globally scoped addresses.
- There should only ever be one transmitter connected to a MULTICAST group or else audio reception errors will occur.
- For transmitting MULTICAST audio the Prodys proprietary protocol "Prodys eXtended Real Time Protocol (PX-RTP) will be used.
- The ProntoNet transmitter cannot be in automatic encoding mode.
- To guarantee a constant delay all the units must synchronise their clocks. Each receiver will activate a clock-sync algorithm that adjusts its PLL (Phase-Lock Loop) .

2.4.- MULTI-UNICAST COMMUNICATIONS

To select this feature, open the new 'NET' menu by clicking on the 'NET' button:



The main window will show all the available connections:



Depending on the compression mode, up to 10 Tx connections will be available:

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

As can be seen, there is an independent control bar for each connection. Each of the connection bars will show the line status in real time.

In addition, it is possible to make or hang up several connections at the same time by using the 'Group Call' and 'Disconnect All' buttons respectively.

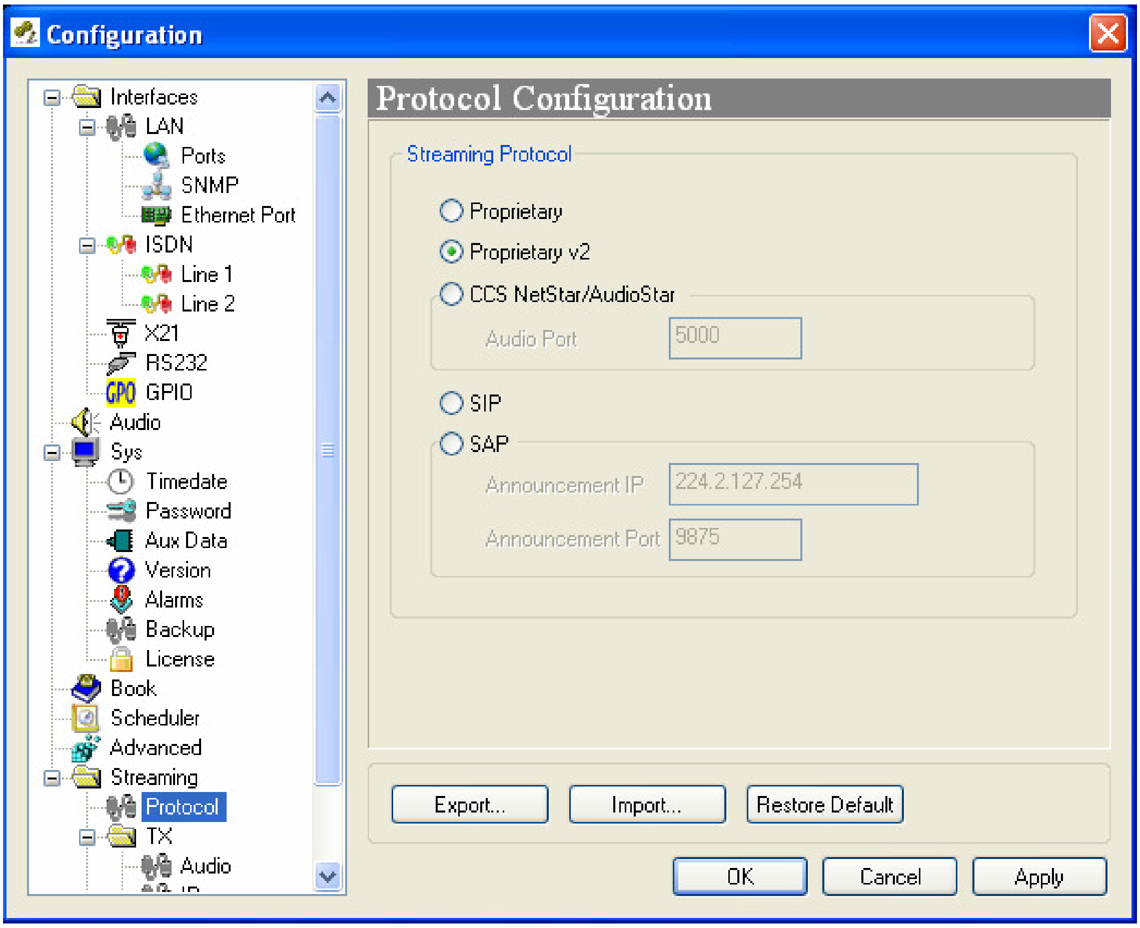
2.5.- PRODYS PROTOCOLS

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

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

- Prodys Auxiliary Data Protocol (P-AUXP). This protocol is based on UDP and used for transmitting/receiving auxiliary data. Available from version 4.8.0 on.
- Prodys U-bit Protocol (P-UbP). This protocol is based on UDP and used for transmitting/receiving the User Bit from the AES/EBU frame. Available from version 4.8.0 on.

From version 5.2.1 onwards, a new proprietary set of protocols (version 2) is available. This new set of protocols (version 2) is not compatible backward, and it introduces some new fields in some of the protocols previously listed so that it is possible to measure lost and disordered packets in real time and during the audio connection.



2.6.- SIP (EBU TECH 3326 STANDARD FOR AUDIO OVER IP)

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

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

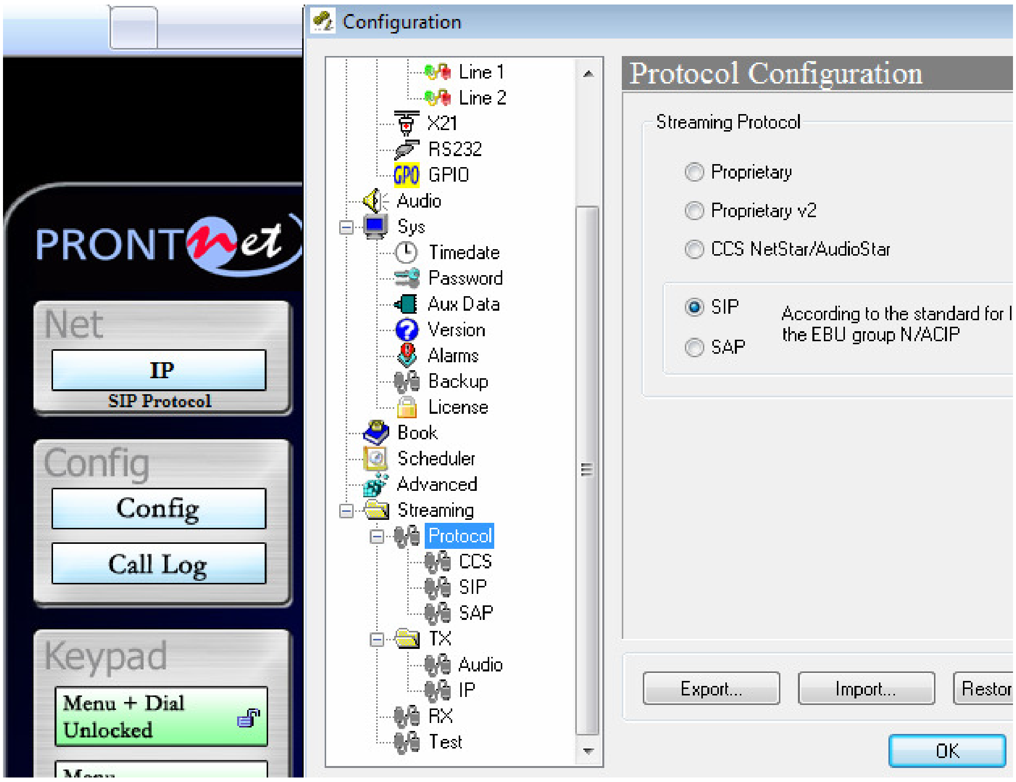
The physical location, often related to fixed IP addresses or subnets, is not further meaningful. In this sense, specially portable audio codecs will profit with an easy call procedure. Neither is required to agree ahead on the audio compression type and data stream rate, because the SIP Protocol manages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

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

2.6.1.- Enabling SIP protocol for Prodys IP codecs

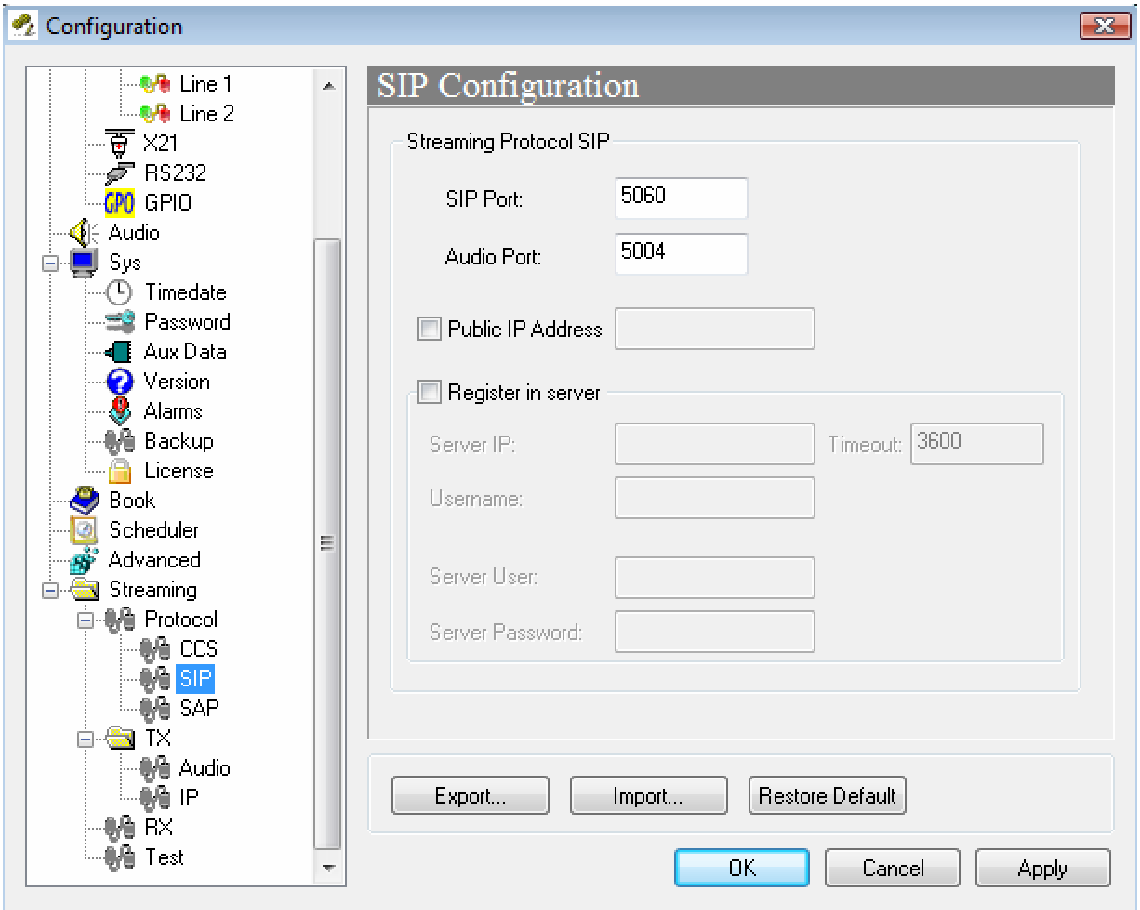
To enable SIP calls the codec administrator has to:

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



2.6.2.- Configuring SIP

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

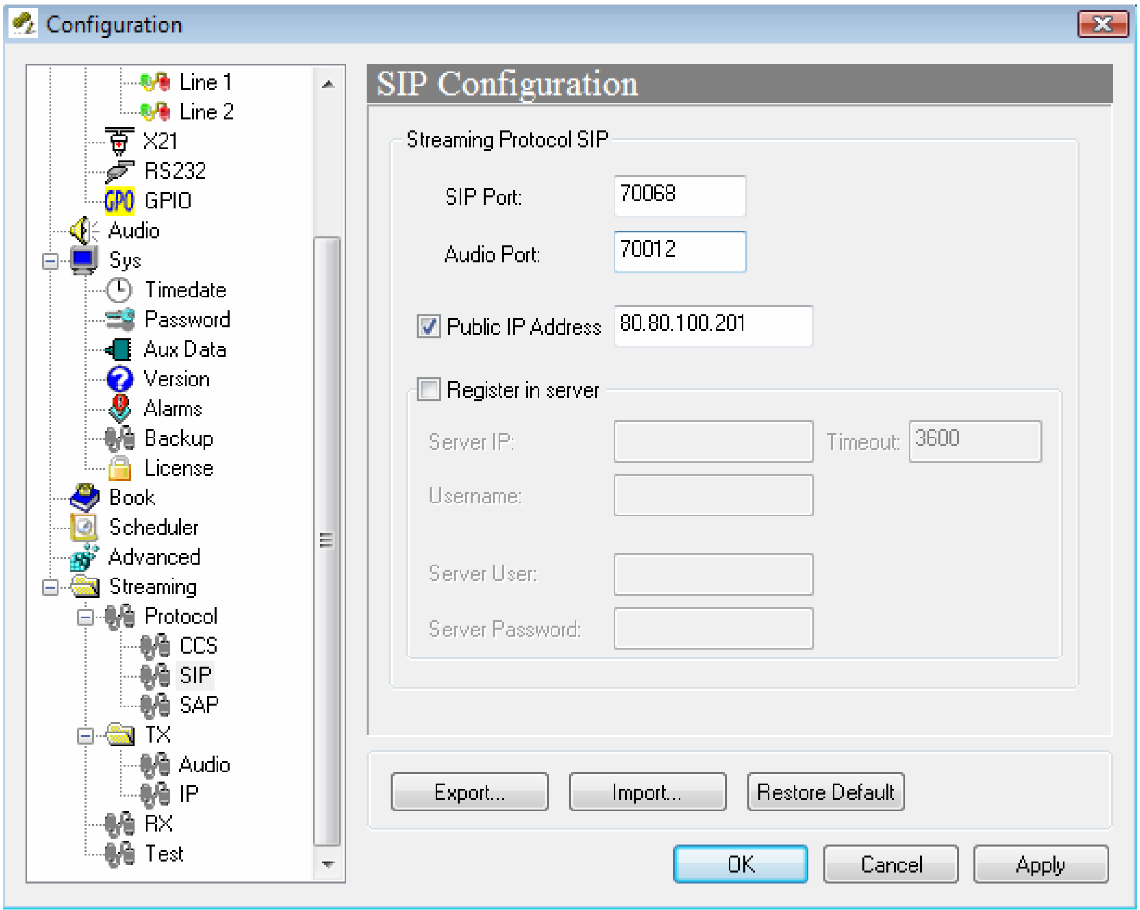


2.6.2.1.- Public IP Address

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

- The Public IP Address of the router's WAN interface must be configured.
- It is recommended to purchase static IP addresses access from Internet Service Provider.

If the codec is not going to connect through a router/firewall which translates from private to public IP address (NAT), it is not necessary to use this field, and should be left empty.

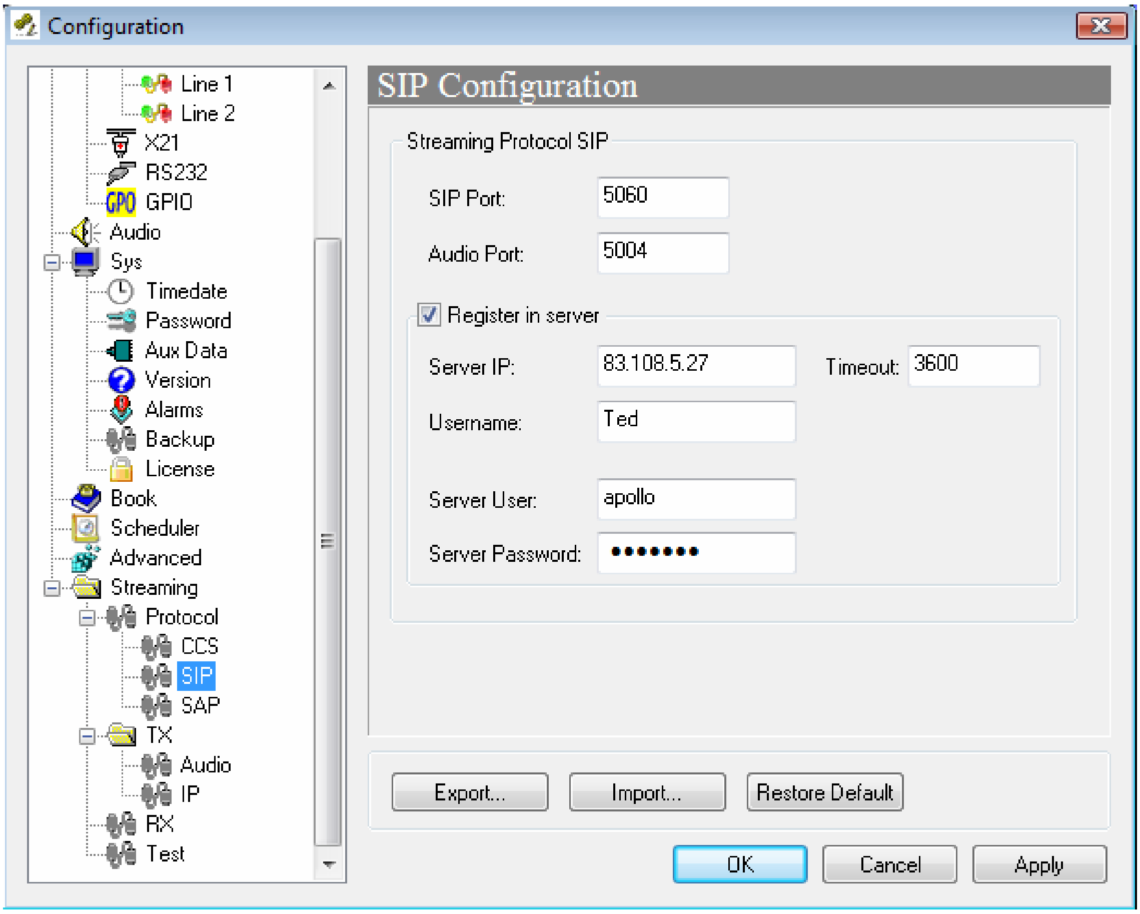


NOTE: There are different ways to discover automatically the Public IP Address from the codec which are expected to be included in the next firmware version.

2.6.2.2.- SIP Server Registration

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

- **Server IP**: IP address of the SIP Server. At this place the codec registers periodically the proper user information for full protocol support.
- **Timeout**: Period (in seconds) for refreshing the user information registered.
- **Username**: Your alias on the Internet regardless of your current IP address. By this alias you are identified for other SIP participants. This username will be sent along with user and password information to the SIP server.
- **Server user & password**: Most SIP servers requires authentication before proceeding registering of SIP users. This parameters should be assigned to you by the SIP Server administrator.



2.6.3.- Calling with SIP

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

- Peer to peer.
- Through SIP Server.

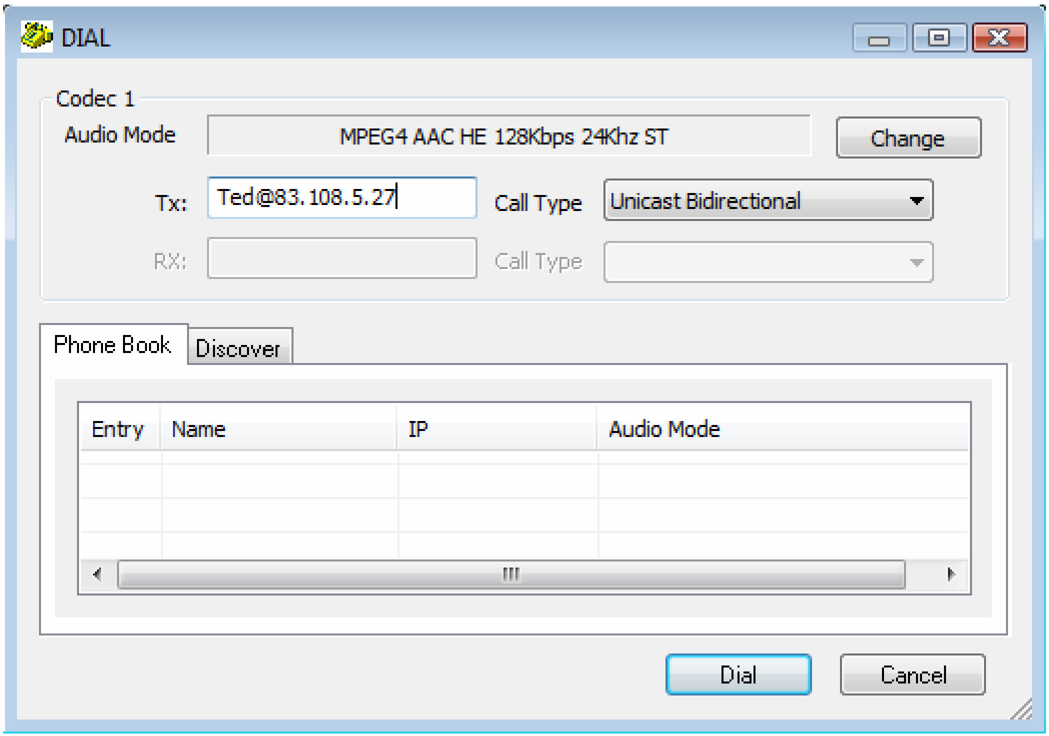
2.6.3.1.- Peer to peer SIP calls

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

2.6.3.2.- Calls through SIP Proxy Server

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

Please take into account that the address to dial should contain the SIP server address preceded by the registered name of the called party. It is the SIP server which redirects the call to the destination if it is registered. Please, see the example below.



In this example, this call will be headed to the Sip server at IP 83.108.5.27, and to the user registered as 'Ted' on that server.

NOTE: Please note that some SIP phones do not require the user to enter the IP/Domain name of the SIP Server when the unit has been configured to be registered on that SIP server. Only the name of the called part has to be entered to establish the connection. This functionality is expected to be included in next firmware versions.

There are several Public SIP servers available. Please, bear in mind that some of these servers act as Proxy SIP Servers, in the sense that not only the signalling but also the audio data goes thought them. This implies a longer audio delay in the communication, as well as a restriction on the supported compression modes for the audio connection.

2.6.4.- Port configuration on the router for SIP

Please, refer to Chapter 2.10.- Router configuration.

2.7.- OTHER NOT AUDIO NETWORK PROTOCOLS

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

-DHCP: IP address auto configuration.
-DNS: Domain Name Server.
-IGMP: This protocol deals with Multicast traffic.

-ICMP: IP control protocol.
-HTTP: A embedded web server allows the user to monitor/configure the unit from an Internet Explorer Web Browser.
-RIP2: Internal/alias IP address can be assigned.
-SMTP: Simple mail transfer protocol. Allows the user to send email alarm notifications.
-SNTP: Simple network time protocol. Time synchronization between units. UDP port number 123.
-SNMP: Simple network management protocol over UDP on port number 161.

2.8.- PRODYS IPV4 ADDRESSES

Prodys factory default settings for the IP parameters are:

-IPv4 address: 192.168.100.100
-Netmask: 255.255.255.0

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

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

2.9.- PRODYS PORTS FOR PRODYS PROPRIETATY PROTOCOLS (V1 & V2)

These are the **default** ports used by Prodys IP Codecs for their IP connections:

HeraFlash &
Prodys Control
UDP:50013

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

Audio Streaming
TCP 50019: P-RTCP (Control)
UDP 50021: P-RTP Unicast L1
UDP 50023: P-RTP Unicast L2
UDP 50025: PX-RTP Multicast L1
UDP 50027: PX-RTP Multicast L2
UDP 50037: P-AUXP Datos auxiliares (from version 4.8.0 on)
UDP 50039: P-UbP U-BITs (from version 4.8.0 on)

Test Streaming Tool
TCP 50033
UDP 50033

<u>External Protocol (P-XP)</u>
TCP 50031: Control Port
TCP 50035: Status Port

An ICMP 'ping' packet is sent every 3 seconds in order to check connection integrity. From version 4.8.0 on, this protocol has been removed.

HeraFlash: Prodys upgrading application.
ProdysControl: This application allows the user to control several and different units from the same screen and in real time.
Test Streaming Tool: This is an application built into the web page to measure the quality of the connection: bandwidth, delay, jitter, packet loss and packet disorder.

From version 4.8.0 on, it is possible to change these ports:

There are two different groups of ports:

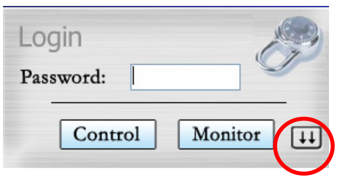
Web Server Port: By default, it is TCP port 80. This is the internal web server port.

Base Port: By default, it is 50011 for TCP and UDP ports. This is the first port of the range of ports used by the unit. From this base port, up to 30 ports should opened/forwarded. That is, if the base port is set to 50011, the range of ports goes from 50011 to 50041, both for UDP and TCP, should be opened/forwarded in the corresponding router/firewall (when required).

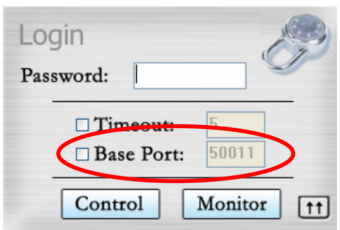
The following things should be taken into consideration when changing the base port:

1. To access the web page of the unit, the new port has to be indicated in the http address bar of the web browser after the IP address, separated by a colon:
   http://<IP>:<Port>>
   Example: 192.168.0.10:8080

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

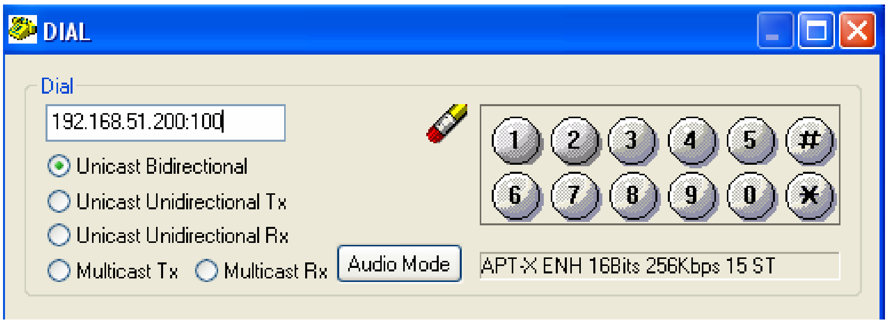


and specify the base port which we want to connect to:

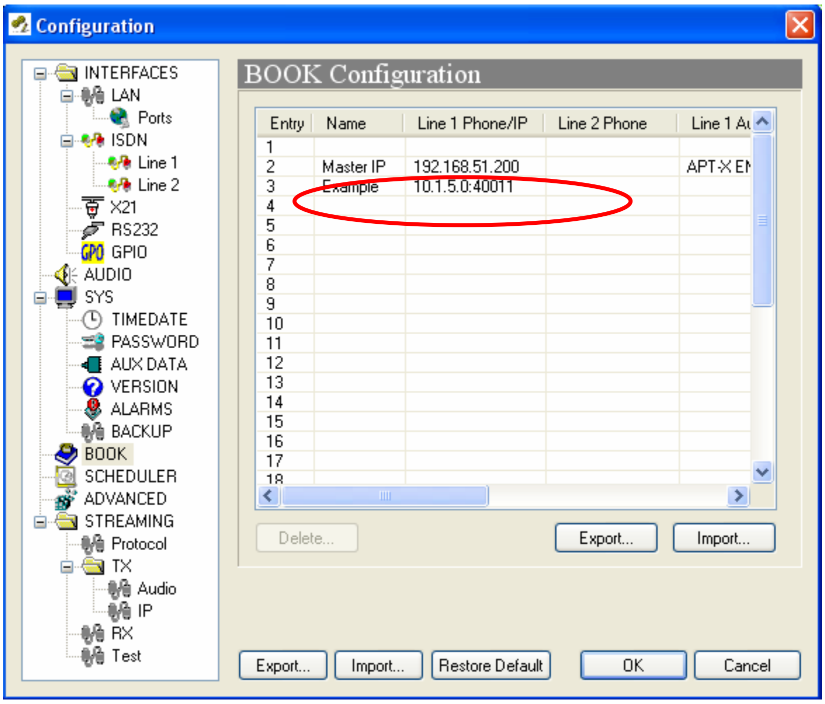


3. When making a call, in case that the receiver has changed the default base port, the user should indicate the address and the port to call to:

<IP>:<Port>



4. When establishing a bidirectional call with Prodys IP codecs, two connections are made automatically, one for each direction. For the receiver to call properly to the caller, it should have an entry on its phone book with the IP address of the caller and the base port of the caller. Example: if the caller has the IP address 10.1.5.0 and its base port is 40011, in the receiver, the following entry should be configured in its phone book:



2.10.- ROUTER CONFIGURATION

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

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

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

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

- The user should bear in mind that other protocols such us SNTP or SNMP use UDP ports for their communications, and that should be taken into account when configuring the routers/firewalls when required. That is, SNMP 161 UDP port should be enabled/forwarded in the corresponding router/firewall when trying to control the units via SNMP from outside our network. SNTP 123 UDP port should be configured only when one the units acts as a time server, and it is going to receive time requests from outside our network.

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

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

3.- EXAMPLES OF PORT SETUP FOR CALLING OVER IP

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

Prodys is ahead of the development of the current standard from the EBU group for audio distribution over IP, although we also support our own proprietary protocols for those cases where they better suit customer needs. Therefore, this chapter comprises both, configuration for Prodys Proprietary Protocols and for SIP/SAP/SDP/RTP protocols (complying to EBU Tech 3326 standard)[1]. From now on we will use only SIP to refer to all protocols included in this standard.

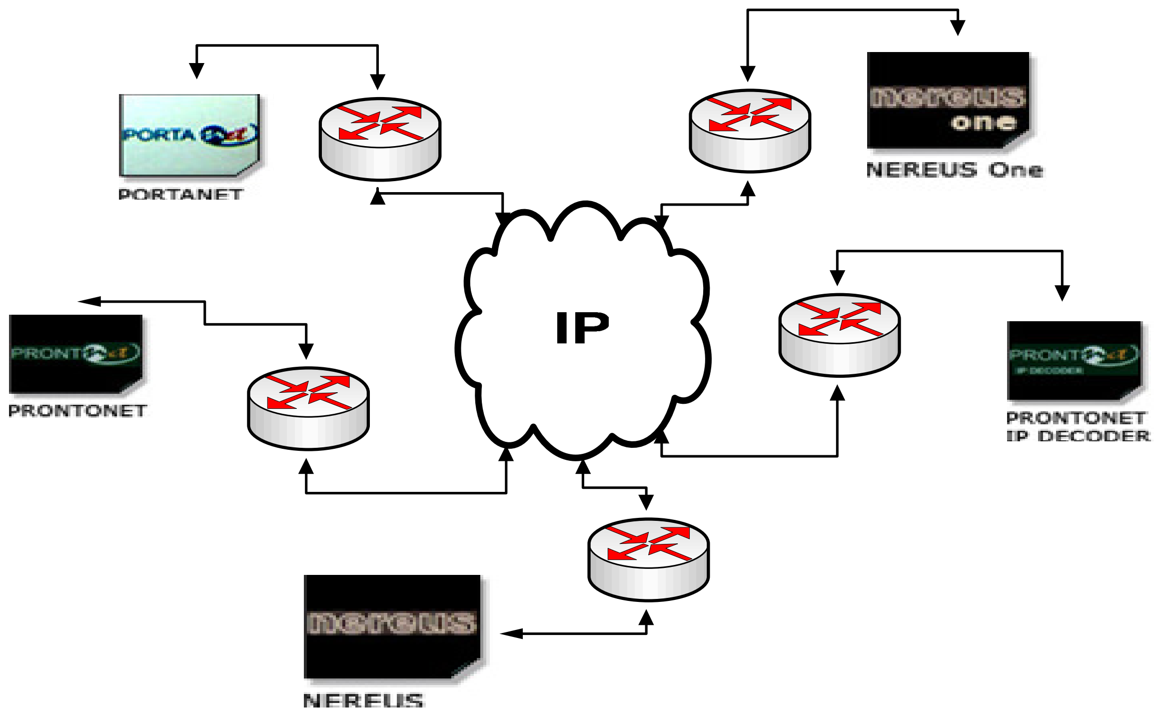
3.1.- DIFFERENT NETWORK TOPOLOGIES AND PORT CONFIGURATION

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

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

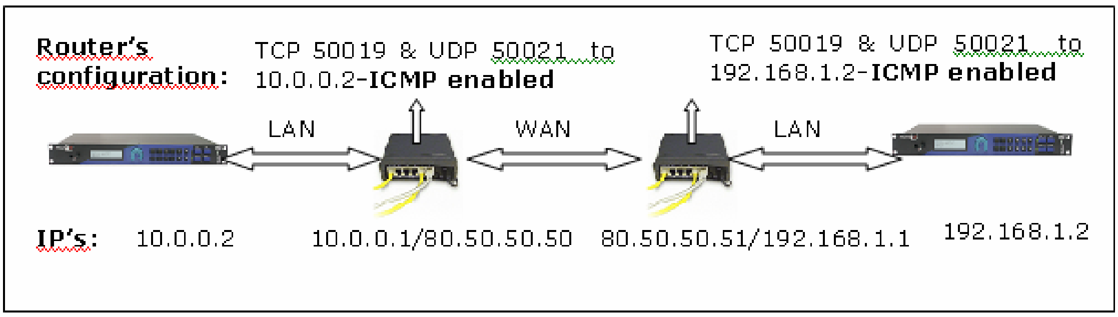
[1] SIP is the commercial name commonly adopted to refer to the EBU standard for audio over IP. This standard also comprises other protocols such us SAP, SDP and RTP.

- If only one single codec behind a router/gateway is to connect, you might refer to the "One-to-One topology" chapter below.
- If multiple codecs are deployed sharing the same gateway (e.g. Nereus or several ProntoNets on one site), the configuration description in "Array of codecs in a back-to-back configuration" section below must be taken into account.
- The third scenario would be that where only one codec (A) behind a router is to communicate with other codecs (B,C) at the remote site sharing a common router. In this case, the configuration detailed on "One-to-One topology" chapter would be applied to A, whilst B and C would require the configuration defined in "Array of codecs in a back-to-back configuration".



3.2.- ONE-TO-ONE TOPOLOGY

This section describes how to connect two codecs over an IP backbone or Internet in a one-to-one scenario where only one codec is placed behind the router/gateway.



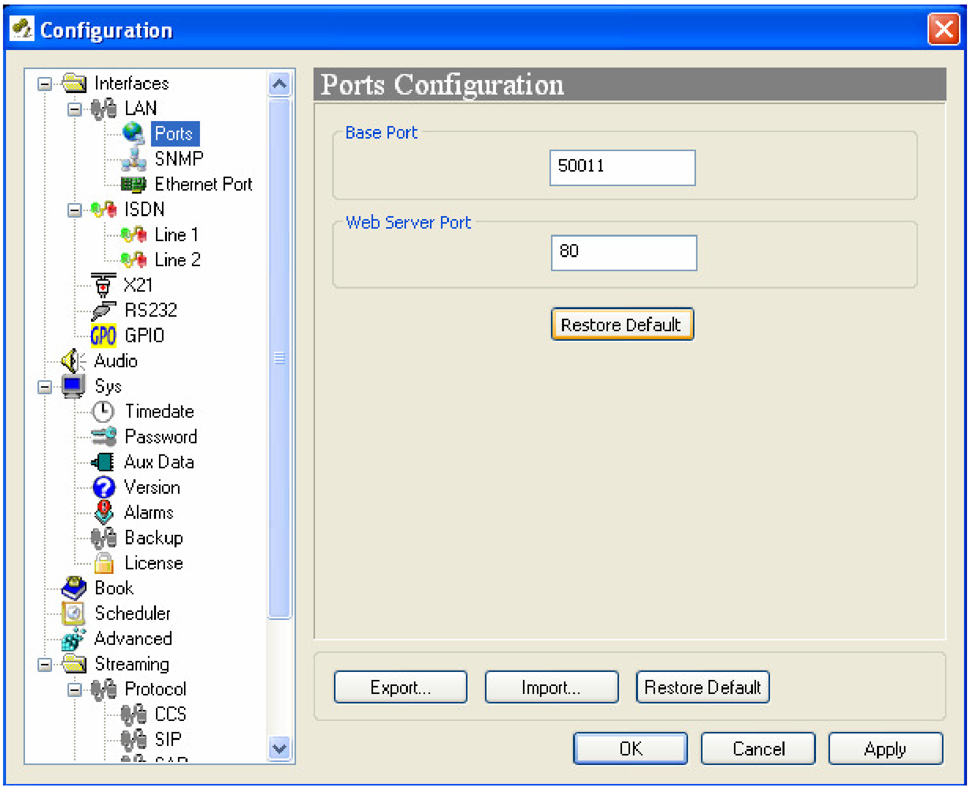
It is assumed one IP-gateway at each place (labelled as WEST and EAST). In this way, the WAN IP address of the router determines the external internet address of the codec behind this gateway.

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

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

Entry for router	First port	Last port	Local address	Map port
Codec West	80	80	10.00.2	80
Codec West	5004	5004	10.00.2	5004
Codec West	5060	5060	10.00.2	5060
Codec West	50011	50039	10.00.2	0 [2]

Please make sure that for both codecs the default port assignment is provided. Take into account that this router configuration only applies when the default ports for IP communications are used. In case you want to make sure that the default ports will be used for the connections, the user can restore the default port configuration from the port configuration window on the control web as shown in the picture below, by clicking on the 'Restore Default' button under the Web Server Port entry box[3].

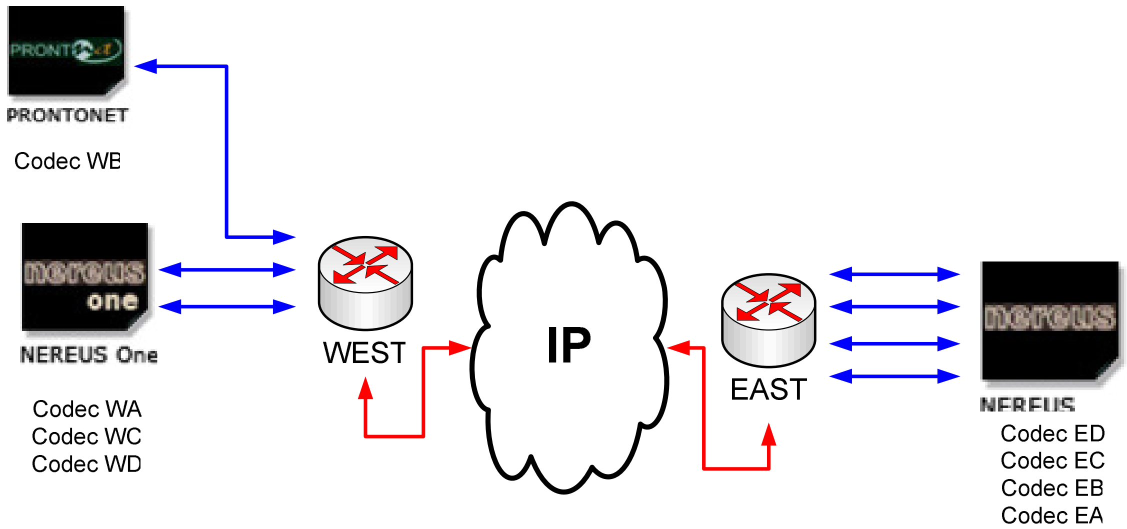


[2] Map port "0" means for most of the router configuration that the external port number matches the port number for the local area network. Given an example: an incoming request from the WAN interface to the port number 50021 will be forwarded to the codec on the same port number 50021.

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

3.3.- ARRAY OF CODECS IN A BACK-TO-BACK CONFIGURATION

This section describes an example of how to match an array of four Prodys Family codecs with a mate array of four Prodys codecs at an opposite place. The following arrangement allows to share up to four high quality audio stereo signals between both sites. The same procedure can be extended for any number of codecs inside the array.



It is assumed one IP-gateway at each place (labeled as WEST and EAST). In this way, the WAN IP address of the router determines the external internet address of all four codecs behind this gateway. Let us write a place holder for this example as:

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

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

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

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

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

Entry for West	First port	Last port	Local address	Map port
Codec WA	50001	50060	LL.YY.XX.AA	0 [4]
Codec WB	50061	50120	LL.YY.XX.BB	0 [5]
Codec WC	50121	50180	LL.YY.XX.CC	0
Codec WD	50181	50240	LL.YY.XX.DD	0
Codec WA	5004	5004	LL.YY.XX.AA	5004 [6]
Codec WA	5060	5060	LL.YY.XX.AA	5060
Codec WB	5104	5104	LL.YY.XX.BB	5104
Codec WB	5160	5160	LL.YY.XX.BB	5160
Codec WC	5204	5204	LL.YY.XX.CC	5204
Codec WC	5260	5260	LL.YY.XX.CC	5260
Codec WD	5304	5304	LL.YY.XX.DD	5304
Codec WD	5360	5360	LL.YY.XX.DD	5360

figure 6

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

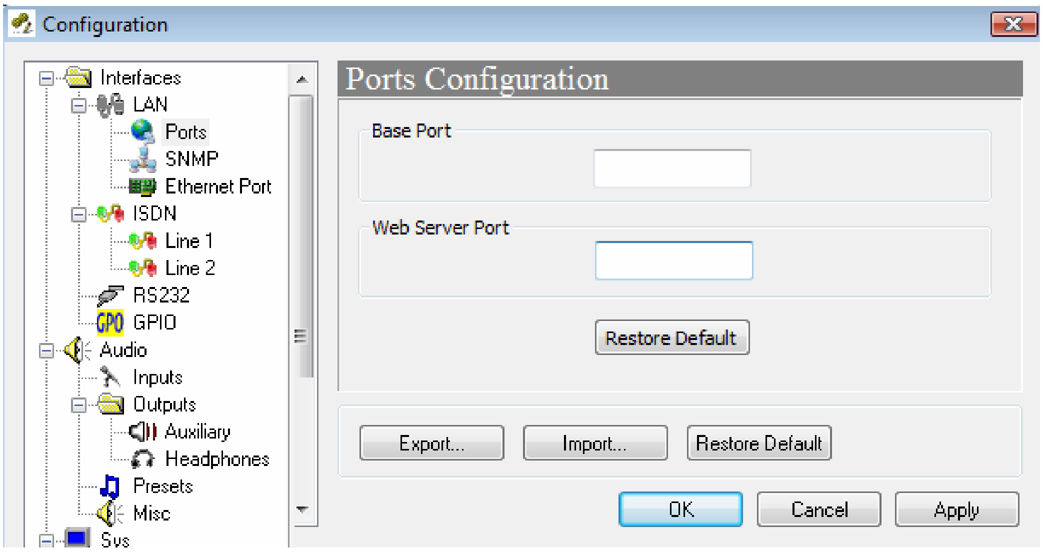


figure 7

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

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

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

In case of using SIP for establishing IP audio connections, the configuration window would be the following:

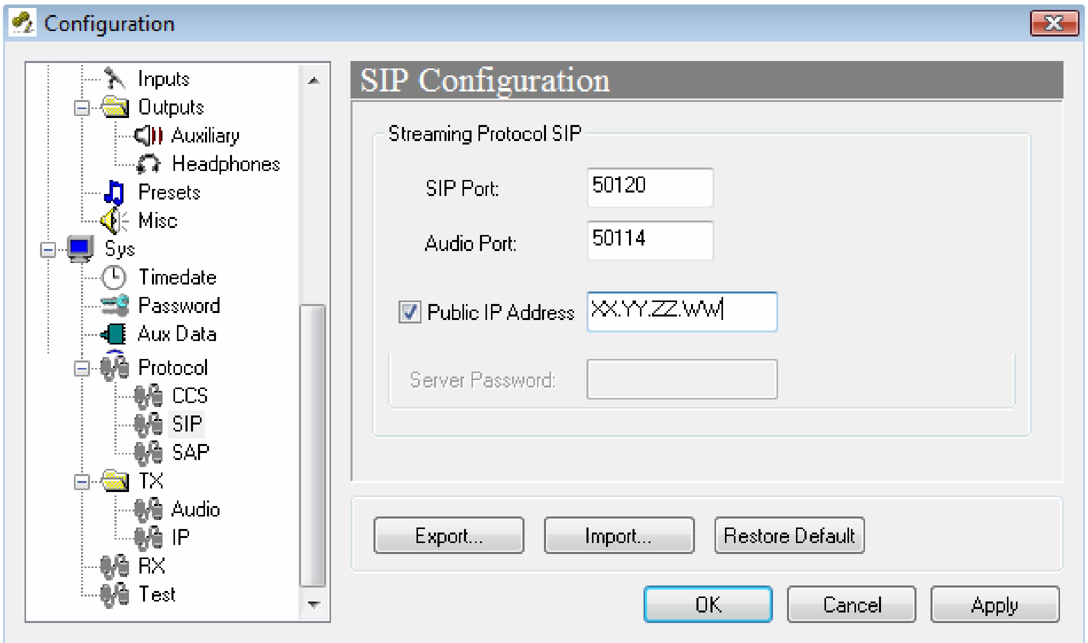


figure 8

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

NOTE: As per the current firmware version (5.5.0), it is mandatory to enable the Public IP Address option in the SIP configuration when making calls through routers/gateways with NAT[7], and to provide the public IP address of the router's WAN interface manually.

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

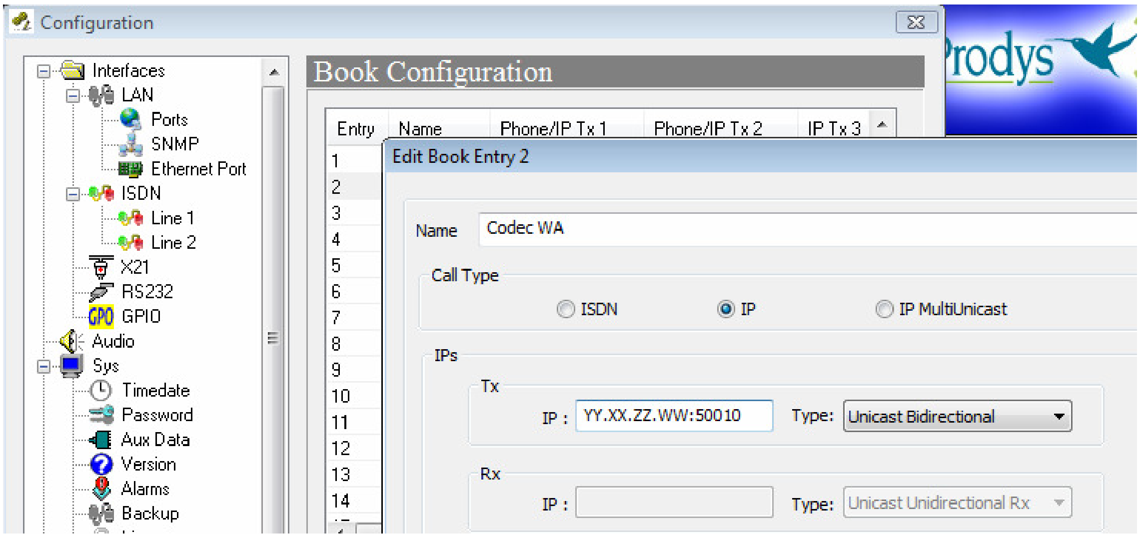
Entry for West	Web Server Port	Base Port	SIP Port	RTP Port
Codec WA	50001	50010	5060	5004
Codec WB	50061	50070	5160	5104
Codec WC	50121	50130	5260	5204
Codec WD	50181	50190	5360	5304

figure 9

In order to complete the configuration and, only when establishing IP connections with Prodys Proprietary protocols[8], the codecs on the receiver site (in this example the "East" site) must register the public (external) addresses of the codecs at the "West" site. This is done with a BOOK entry at each codec.

[7] Network Address Translation: The router translates private IP address into public addresses.
[8] Please note calling via the SIP protocol does not require keeping a registration of the call initiator inside the receiving codec BOOK.

Figure 9 shows an example of one BOOK entry for destination “Codec WA” at the mate codec of the “East” site.[9]



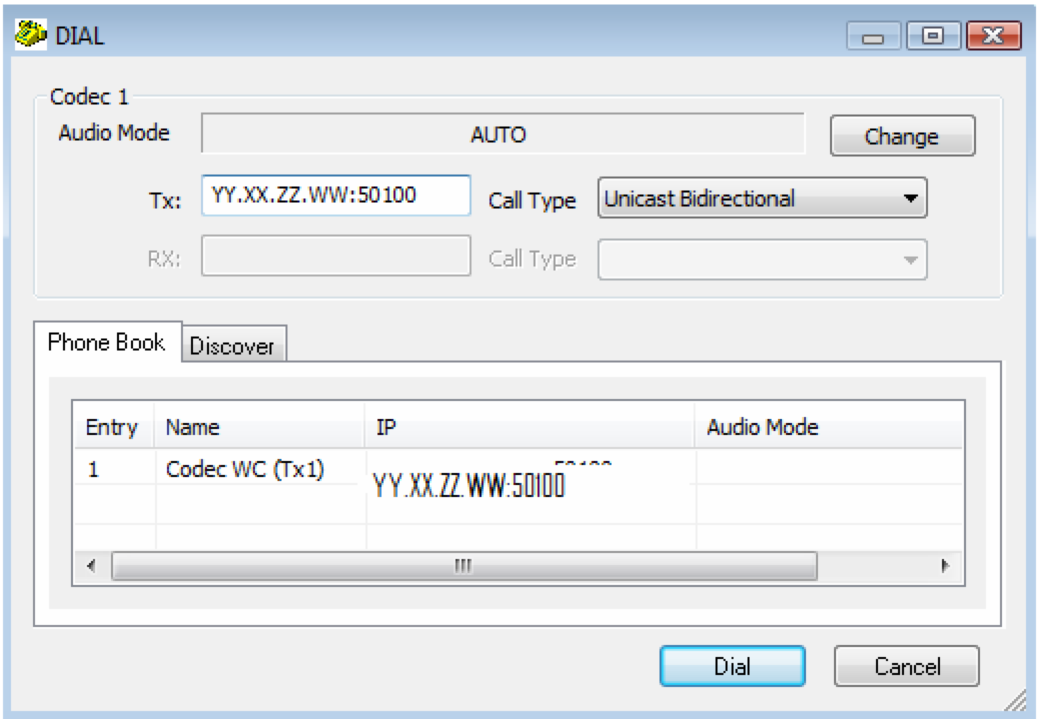
Note that each BOOK entry must meet the correct Base Port of the destination.

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

- Port Forwarding provision on the eastern gateway. (It is not necessary to meet the same port ranges given at the “West” site.
- Port definition at each eastern codec according “East” gateway provision.
- BOOK entries at each western codecs for their mates when using Prodys Proprietary Protocols.

For initiating a call: the destination IP address must be dialled together with the proper destination Base Port.

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



4.- CONTROLLING THE UNIT FROM THE WEB PAGE

All Prodys IP codecs units come with the following IP settings:

IP-address: 192.168.100.100
Subnet mask: 255.255.255.0

1. Configure your PC's TCP/IP settings to:

   IP-address: 192.168.100.X …(for example 192.168.100.1)
   Subnet mask: 255.255.255.0

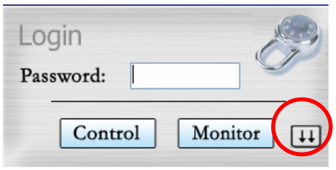
2. Open your Microsoft Internet Explorer:

   Type in the unit's IP address 192.168.100.100 into the Internet Explorer's address bar.

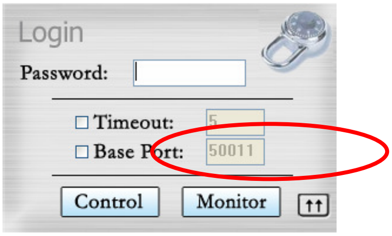
3. First time you connect to a Prodys IP Codec via Internet Explorer you will be asked to install the OCX file. Click on "Install", then click on "OK" in the "Login" window.

4. Select whether to control and monitor, or only monitor the unit with the corresponding button. The password is empty till the user set a different one.

1. When the default ports have been changed, click on the advance features button of the login window:



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



5. Once the web page appears, the user can change the IP settings of the codec to those required by the user's network. In this case, the computer IP settings should be configured accordingly.

Note: If you can't access the codec web page make sure that
a) The codec and the PC have the right IP-settings. If you do not know the codec IP address, make a factory default reset by restarting the unit with Microswitch 7 pulled down. Do not forget to pull up the switch again after reset!
b) You are using the right cable (crossover CAT-5 cable when connected directly without using a switch or hub).

4.1.- WHAT ARE S-CLUSTER OCX FILES?

OCX stands for ***OLE Control Extension***, an independent program module that can be accessed by other programs (like Microsoft Internet Explorer) in a Windows environment. OCX controls end with a .ocx extension. S-Cluster OCX files are installed on your computer when you access the IP Codec web page. They are installed on the following path (It depends on the Windows version and on the type of IP Codec):

C:/WINDOWS|WINNT/Prodys/Unit-name-version/Unit-name.ocx.

For example, to access ProntoNet version 4.5.0, the PC will store the following ocx file under Windows XP SP2:

C:/WINDOWS/Prodys/ProntoNetv4.5.1/ProntoNet.ocx.

These modules (OCX files) are used by Microsoft Internet Explorer to provide the user with graphical user interface to control Prodys IP Codes.

These modules (OCX files), use another files, like graphics, which are stored in some folders placed behind the root one, such us

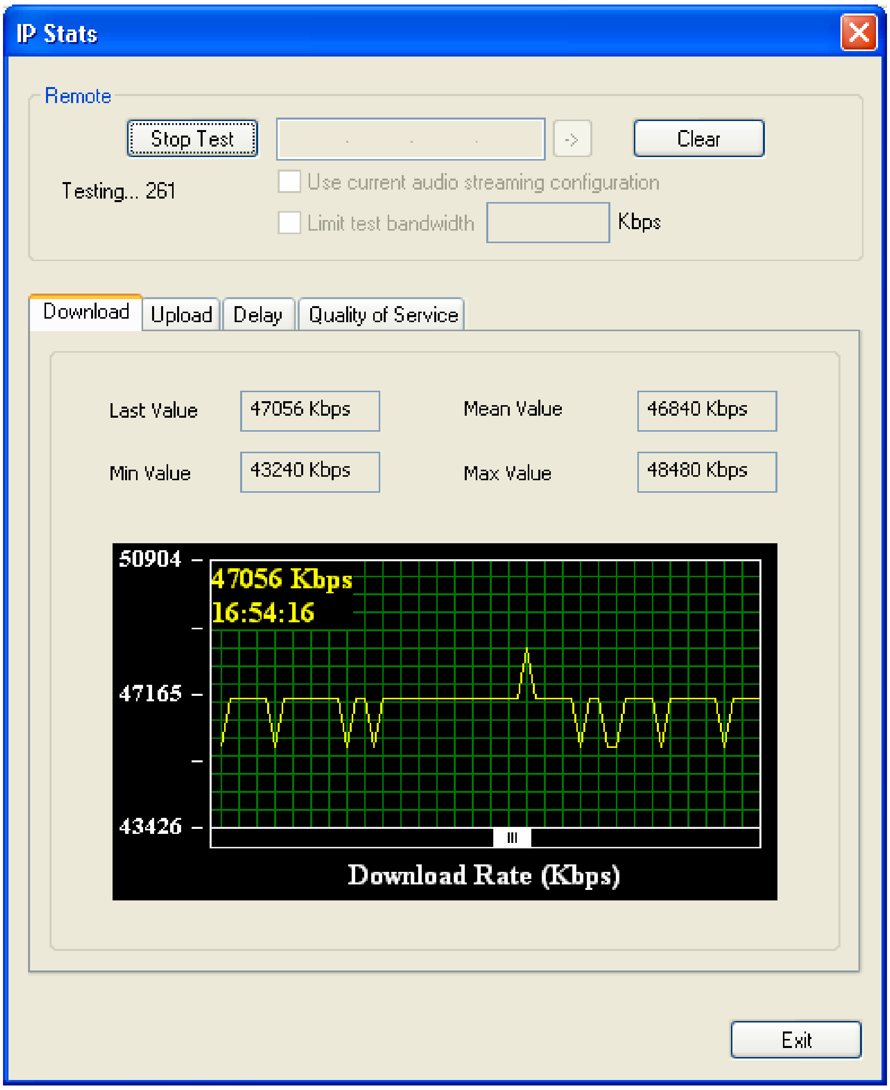
C:/WINDOWS|WINNT/Prodys/ProntoNet, for example.

5.- MEASURING THE IP LINK BEFORE CALLING

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

5.1.- TEST STREAMING TOOL

Access this tool by pressing the "SYS" button on the web interface, and select STREAMING – Test. Then, type in the IP address of the remote end unit and click on Start Test. By clicking on the different tabs, the user can get the information related to upload and download bandwidth, delay and quality of service. To stop the test, click on the 'Stop Test' button.

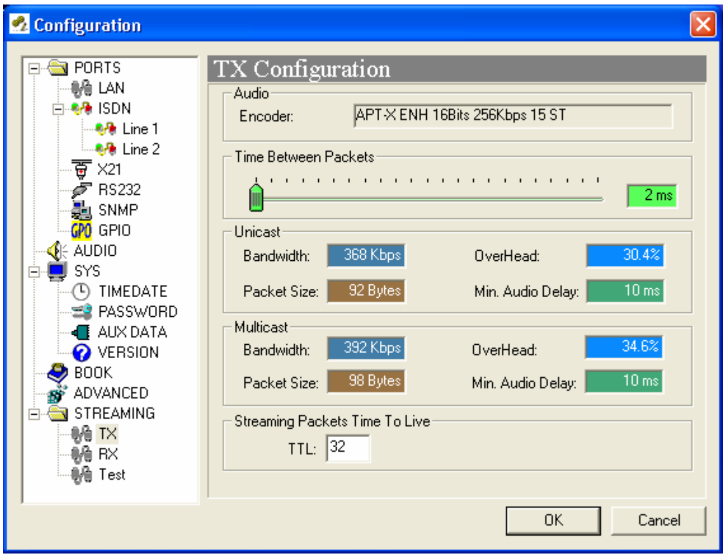


This information help you to decide which compression mode to use, and adjust the streaming parameters, based on a truly accurate knowledge of the quality of the connection, thus obtaining the best quality in the real-time audio transmission. Once selected the compression mode, with the measured jitter and bandwidth, the user is ready to adjust the 'Jitter Correction Buffer' in the reception side, and the 'Time Between Packets' in the transmission side.

5.2.- STREAMING TX: TIME BETWEEN PACKETS (TBP)

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

Access this by pressing the "SYS" button on the web interface select STREAMING – TX. You will be presented with a screen similar to below[10]:



From this window, the user can know beforehand, the bandwidth and delay of the connection. With a low delay algorithm such us apt-X, set to its lowest delay setting and with the algorithm data rate set to 256Kps, the network bandwidth required is 368Kps i.e. an overhead of 30.4%. This gives a minimum audio delay of 10ms end to end (add this to any delays from network switches, routers etc.), but the maximum bandwidth for this compression mode settings.

- So, 'Time Between Packets' parameter, is directly related with the packet size and therefore with the occupied bandwidth (overhead) and delay. Therefore, the appropriate value for this parameter is a trade off between delay and bandwidth:
  The larger the block size (TBP), the higher the delay, but the smaller the required bandwidth and overhead (more efficient use of IP packets), and vice versa.
- If there is no delay requirements in the connection, it is always advisable to select the highest value, since bandwidth will be smaller.

[10] This is an example screen for ProntoNet.

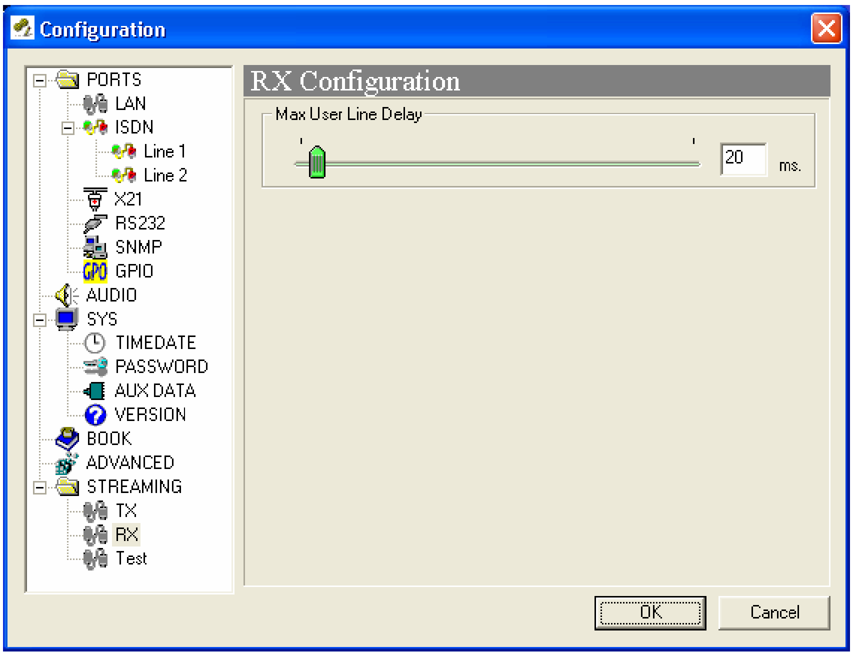
Also, the more frames over the network (with small TBP values, frames decrease in size and increase in number), the more likely it is that the jitter will grow.

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

5.3.- STREAMING RX: JITTER CORRECTION BUFFER

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

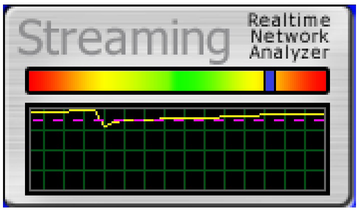
- "Jitter Value" is the difference between the maximum and minimum delay. If the jitter value is 0 it means that the delay is constant and it is not necessary to adjust the jitter correction buffer in the reception side. If the delay is not constant, it is necessary to adjust the buffer size in order to guaranty not audio drops, even when the delay reaches the maximum value.
- The unit of this buffer is miliseconds, and the range is 0-500. The delay grows as many as the number of miliseconds the buffer has been set to.
- The "jitter value" can me measured by the streaming tool commented before.
- **Rule: Max User Line Delay + TBP >= Jitter value.**



5.4.- REAL TIME NETWORK ANALYZER

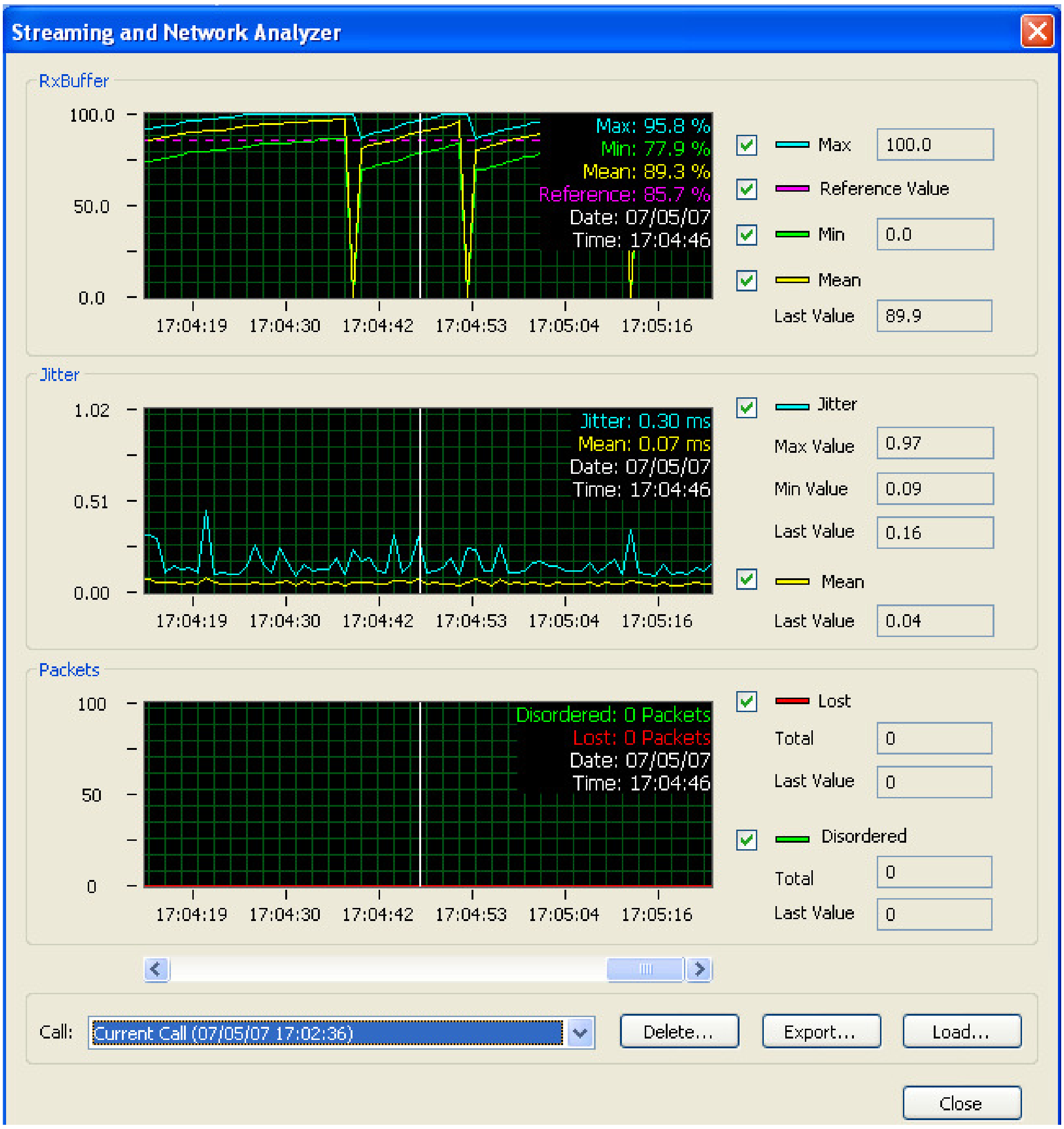
From version 5.2.1 onwards, it is possible to monitor in real time and during the audio connection some performance parameters in the network, such as jitter, lost and disordered packets. This information will be saved separately for each connection in RAM memory. Up to 24 hours of data can be stored.

Once the connection is established, the user can access the 'real time network analyzer' by clicking on the 'buffer occupation graph', in order to get information related to:



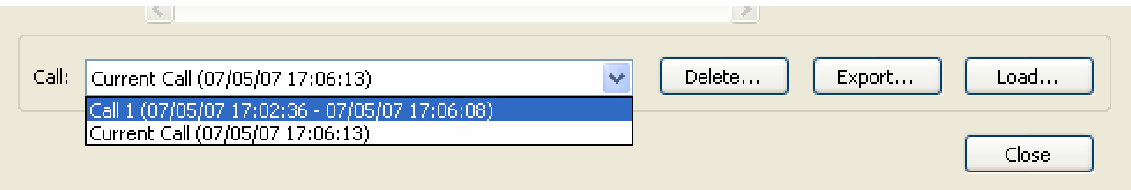
-Rx occupation: With average, maximum and minimum usage. Very low percentage of buffer occupation will cause audio interruptions and drop-outs.
-Jitter: High jitter values will match with low buffer occupation.
-Lost and disordered packets.

To be able to get all this information, Proprietary Protocol V2 should be configured as audio protocol. This protocol is not compatible with the previous one (version 1). Proprietary protocol version 1 will allow the user to get information about jitter and buffer usage, but will not allow the user to obtain information about lost and disordered packets.



All the information is displayed in different graphs, synchronized with each other, so that the user can move through all the data very easily.

Data from each connection is stored independently so that it is possible to access data from connections other than the current one. In addition, it is possible to delete, export or import data from any previous call.



6.- FAQS

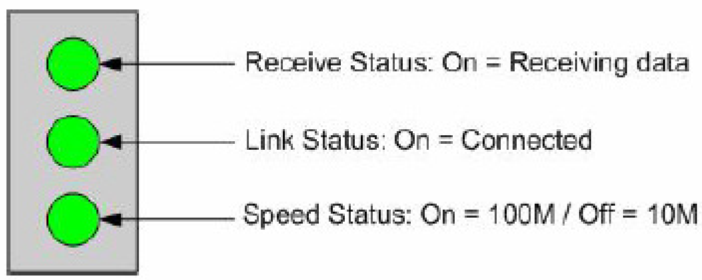
This chapter is aimed at providing some solutions to common problems when using Prodys IP Codecs for the first time.

6.1.- WEB PAGE CANNOT BE ACCESSED

1. The codec and the PC have the right IP-settings. If you do not know the codec IP address, make a factory default reset by restarting the unit with Microswitch 7 pulled down. Do not forget to pull up the switch again after reset!

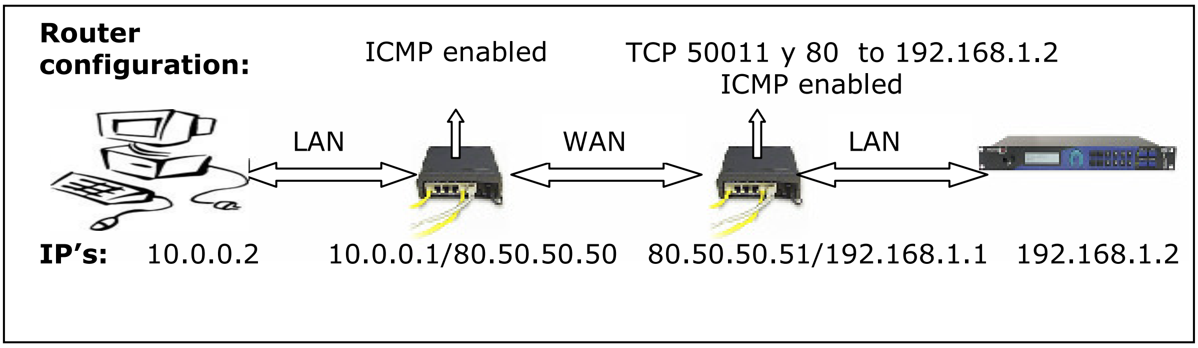
2. You are not using the right cable (crossover CAT-5 cable when connected directly without using a switch or hub).

3. The cable or the connector are faulty. There are some LEDs at the rear panel to check the Ethernet link:



4. The IP address of the Codec and the PC from which we are trying to access the web page are not in the same subnet, even although they are connected to the same LAN. The default factory settings for the IP address and netmask are 192.168.100.100 and 255.255.255.0 respectively. The user must change the IP settings in the computer or in the Codec to match the same network.
   To change the IP address of the Codec, the user can use the control keypad or the web page. In the menu, it is set by selecting CONF-PORTS-LAN. The IP settings can be entered manually or they can be obtained automatically when the unit starts from a DHCP server.
   Once the IP setting on the PC and the Codec are configured properly, you can check that IP connectivity exists by typing the following command at the command prompt of the operating system: C:>ping 192.168.100.100 ↵. This tool will inform the user whether there is IP connectivity between the PC and the Codec or not.

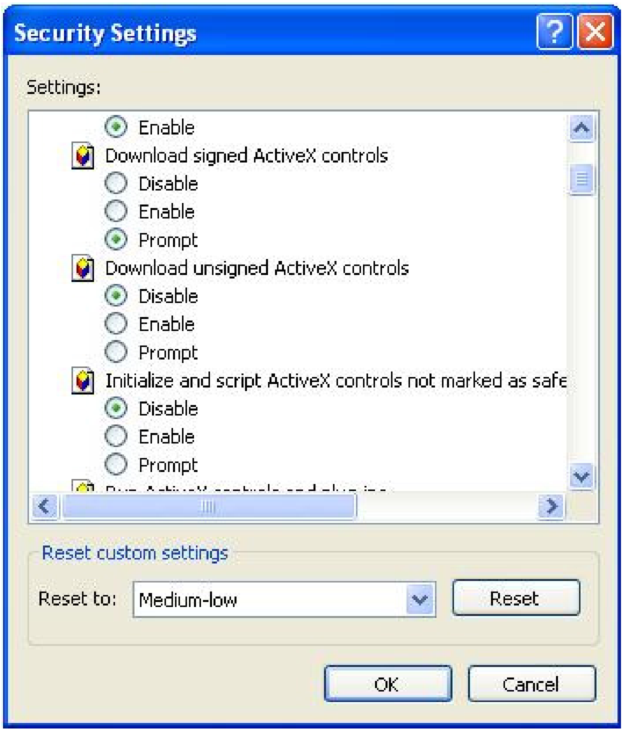
5. The PC and the Codec are not connected to the same LAN, but connected through a router. This connection could be, for example, a connection through the Internet with DSL routers. In this case, to access the codec web page remotely from the PC, it is necessary to enable the ICMP traffic at both sides **(only with versions prior to 4.8.0)**. In addition, the corresponding router should open the following TCP ports: 50011 and 80 (these are the default ones, so these ports might be different depending on th configuration), and forward this traffic to the Codec IP address.



6.2.- MICROSOFT INTERNET EXPLORER BLOCKS OCX INSTALLATION WHEN TRYING TO ACCESS THE CODEC WEB PAGE.

The first time the user accesses the codec web page, an OCX file has to be downloaded and installed in the computer. This is done automatically unless the web browser disables it. So, depending on the configuration of the web browser, the following message can appear when first accessing the web page:

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



Windows Vista: En caso de que el usuario tenga algún problema con la instalación del OCX sobre Windows Vista cuando accede por primera vez a la página web de control del equipo, por favor deshabilite el User Access Control (Control de Acceso de Usuario), intente de nuevo acceder a la página web de control y vuelve a habilitar el UAC en caso que lo considere oportuno. Esto deberá acerse cada vez que se actualice el firmware del equipo.

Cada version de firmware lleva un OCX nuevo, que requiere instalación sólo la primera vez que se accede a la página web de control para esa versión. Si la unidad es actualizada, dependiendo de la configuración de caché del navegador Internet

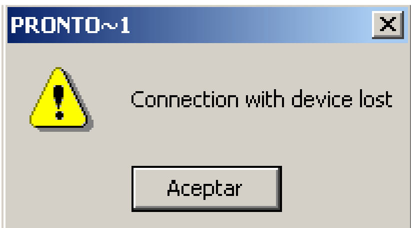
Explorer, podría haber problemas con el acceso a la página web, dado que la página que el navegador nos muestra podría ser la almacenada de la versión anterior, que no puede conectar con el equipo, y daría un error de versión incompatible. Para evitar esto, se puede pulsar la tecla F5 que intenta refrescar la página sin pasar por cache, mostrando así la correcta, que intentaría la instalación del nuevo OCX. Otras veces esto no es suficiente, siendo necesario entrar en 'Opciones de Internet' en el menú del Internet Explorer, pulsar sobre 'General' y una vez dentro de este menú seleccionar 'Borrar ficheros temporales'.

6.3.- MICROSOFT WINDOWS VISTA BLOCKS OCX INSTALLATION WHEN TRYING TO ACCESS THE CODEC WEB PAGE.

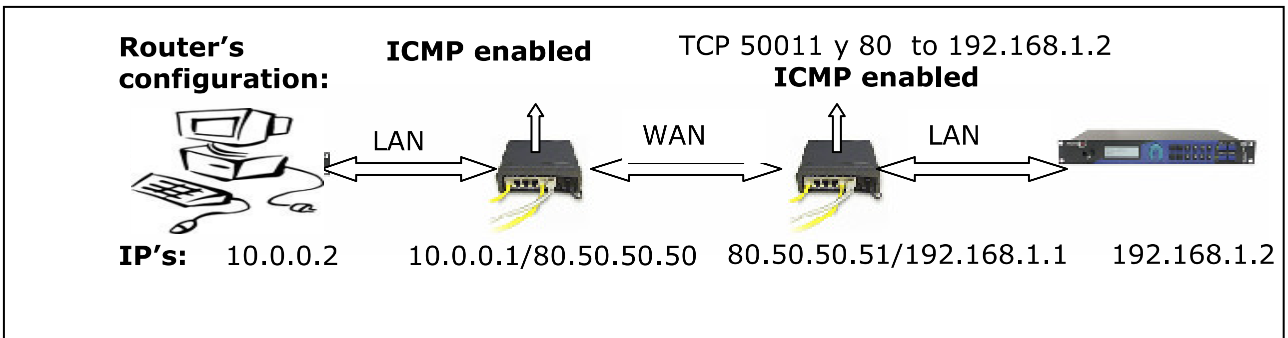
Should the user experience a problem when downloading the OCX file when first accessing the web page of the unit, please disable UAC (User Access Control) on Windows Vista. Once the OCX file has been installed in the computer, UAC can be enabled again.

6.4.- WHEN ACCESSING THE PORTANET WEB PAGE, THE CONNECTION BETWEEN THE PC AND THE PORTANET IS LOST AFTER A FEW SECONDS

1. **Versions prior to 4.8.0:** In some operating systems like Windows XP SP2, there is a firewall enabled by default which disables the ICMP traffic to and from the PC. In this case, the PC cannot maintain the connection with the web page, and it is lost after a few seconds. To avoid this situation the user must enable the ICMP traffic in the firewall.

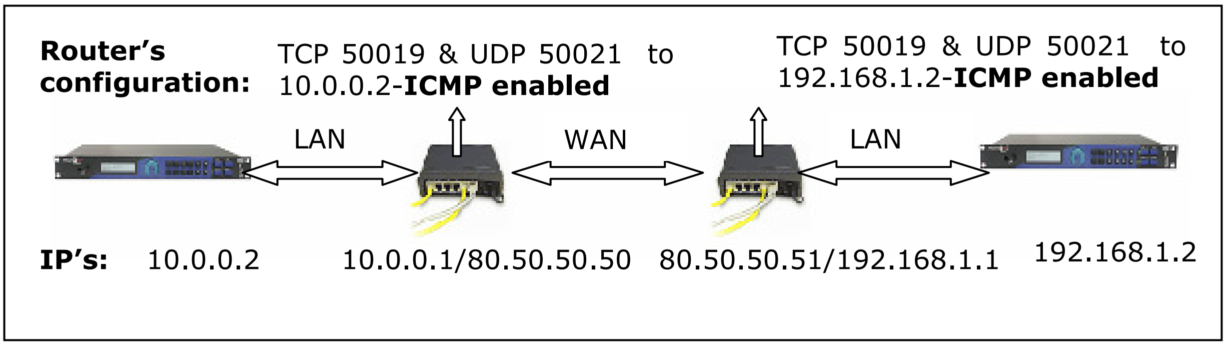


2. When the connection between the Codec and the computer is made through a router, that is, the connection is made over a WAN or the Internet, the user must configure the router at the Codec side to let pass through traffic coming from ports TCP 50011 and TPC 80 (these are the default ones, so these ports might be different depending on th configuration), and also, the ICMP traffic **(versions prior to 4.8.0)** at both sides. Traffic from those ports should be also forwarded to the IP address of the Codec.



6.5.- UNICAST COMMUNICATION PROBLEMS

To establish a unicast communication between two Prodys IP Codecs, the following ports are used (**by default**): TCP 50019 to control the communication and UDP 50021 to send/receive audio data. The typical problem is when this communication is being established across different networks through two routers one at either end. In this case, these ports (or the new ones when the base port is changed) on the routers have to be opened and forwarded to the Codec IP address. The ICMP traffic has to be also enabled in the routers.



Note: From version 5.2.0 onwards, it is possible to obtain network related parameters such as jitter, lost or disordered packets in real time during the audio connection. To be able to get all this information, Proprietaty Protocol V2 should be configured as audio protocol. This protocol is not compatible with the previous one (version 1) so it could lead to confusion given that it would not be possible to establish a connection between ProntoNet when the protocol version does not match.

6.6.- WHEN CONNECTING TWO PORTANETS IN UNICAST, THERE IS NO AUDIO AT ONE END

To establish a unicast communication between two Prodys IP Codecs, the following ports are used (**by default**): TCP 50019 to control the communication and UDP 50021 to send/receive audio data. If the communication is made through a router/firewall and the control port is opened, but the data port is closed, it would be possible to establish the connection, but not to receive the audio data. In a bidirectional communication, we could even receive audio only at one end, if only one of the two routers is properly configured.

6.7.- NO AUDIO WHEN CONNECTING TWO PORTANETS USING MULTICAST

To establish a multicast communication between two Prodys IP Codecs, the UDP 50025 port is used (**by default**) to send/receive audio data. The typical problem is when this communication is being established across different networks through two routers, one at either end. In this case, these ports (or the new ones configured by the user) have to be opened and forwarded to the corresponding Codec IP address. The ICMP and multicast traffic have to be also enabled in the routers (**only for versions prior to 4.8.0**).

Take into account that only a small part of the Internet called Mbone supports multicast traffic, so to send multicast traffic over the Internet, a technique called 'tunneling' must be used. VPN networks can be used for this purpose. For a brief introduction to VPN, please, take a look at the document called VPN.ppt, available in the Prodys site download section.

Note: From version 5.2.1 onwards, it is possible to obtain network related parameters such as jitter, lost or disordered packets in real time during the audio connection. To be able to get all this information, Proprietaty Protocol V2 should be configured as audio protocol. This protocol is not compatible with the previous one (version 1) so it could lead to confusion given that it would not be possible to establish a connection between ProntoNet when the protocol version does not match.

6.8.- INTERRUPTIONS TO AUDIO WHEN CONNECTING TWO CODECS

1. A decisive factor in real time audio streaming is the 'jitter', or delay variation. To deal with the jitter in the connection, the Prodys IP Codecs provide a tool which allows the user to modify the size of the reception buffer, and so, to compensate for the jitter. The maximum value for this buffer is 500 msc (10 sc. from version 4.7.1 on). This buffer has to be configured from the web page to be, after adding to the TBP value, at least the same as the 'jitter' in milliseconds.

   The 'jitter' can be measured from the 'Test streaming' tool provided from the web page. See chapter Streaming Rx.

2. The audio interruptions could be due to a reduction on the bandwidth in the IP connection. With the 'Test Streaming' tool from the web page, the user can measure the download and upload bandwidth between two Prodys IP codecs, such as PortaNets. Once the user knows the available bandwidth, it is possible to select the proper bit rate for the compression mode. See chapter Test Streaming Tool

   The user can obtain information about the delay of the encoder/decoder process for any particular mode, and the actual bandwidth which will be required for that mode, from the web page, in the 'audio' tab. In addition, in the non-block modes, like PCM or apt-X, it is possible to modify the size of the frames from 2 to 24 msc. In the rest of the modes, this size is fixed and determined by the corresponding standard. The larger the block size, the higher the delay, but the smaller the required bandwidth (more efficient use of IP packets), and vice versa. See chapter Streaming Tx.