542 APC

Audio Processing Core
FM & HD audio

OWNER'S MANUAL

www.solidynepro.com

Summary

Thank you for choosing us

Congratulations! The unit you have in your hands has been developed using the highest technology for digital audio. The 542APC processor is the "top of the line" in the Solidyne Broadcasting Processor series. It synthesizes more than 40 years of experience in the development of audio processors for broadcasting.

The 542APC introduces a great innovation in the field of FM processors: it's an audio processing core that runs audio processing software. The software defines the features (model) of the unit. Free upgrades for a same model can be downloaded indefinitely. And you can get a more advanced application that expands the performance of the processor, which is equivalent to having a new model... but without changing the hardware!

All control and settings of the 542APC are made by connecting the unit to a local network. Entering the IP address (shown on the front display of the processor) in any web browser, the user access the Control Panel: an intuitive environment designed to operate on touch screens. IP access allows the remote control when the unit locates at the transmitting plant (requires LAN access to the transmitter plant).

The processor can also be remotely operated using the 542-RM controller which features a 7 "touch screen to operate the unit and monitor transmission parameters in the studios.

The essentials can be adjusted from the OLED mini-display and the control wheel located at the front.

542APC includes an internal RDS encoder for sending text to the audience. And FM stereo encoder with dual MPX output.

542APC is not only an audio processor but also a measuring and control instrument. It incorporates an FM tuner and a real-time signal analysis stage that allows the user to monitor various aspects of the transmission and the processor itself (modulation depth, pilot tone level, RDS and others) as well as some audio quality features . The tuner can be set to any frequency, which allows you to monitor other radio stations.

Please read this manual carefully to get the best performance of the processor.

About this manual

Manual	April 2018
Firmware	5B – 1.30

Whats in the box

Inside the box you will find:

- ✓ *1 Solidyne 542APC processor*
- ✓ *This user's manual*
- ✓ *1 power cord (Interlock)*
- ✓ *1 telescopic antenna*
- ✓ *1 Guarantee card*
- ✓ *4 Rubber feet*

Please check that all these items are inside the box and that the equipment was not hit during the transfer.

Recommendations for the installing

The Solidyne 542APC is designed to be installed in a standard 483 mm (19 ") rack. Its height is one rack unit.

Four self-adhesive rubber feet are included for the case where the unit is placed on a table.

When mounting the equipment in a rack, use flat head screws with a flexible washer (plastic or rubber). First tighten the lower screws and then the upper screws, to prevent the weight of the unit from pushing the upper angles. Do not tighten the screws too much, a slight force when tightening them is enough. Excessive force on the screws can deform or even break the panel angles.

WARNINGS

AC Voltage	This unit works with **110/200 VAC**. The voltage selects from a switch located at rear panel. **CHECK THE VOLTAGE SELECTOR BEFORE PLUG TO THE AC OUTLET.**

CAUTION

To reduce the risk of electric shock, do not remove the enclosure covers. Internal parts do not require user maintenance. Contact qualified technical assistance.

Connect the supplied 3-prong power cord to a 3 hole AC outlet. Do not use adapters. **MAKE SURE YOU HAVE A PROPER GROUNDING.**

The admiration mark within a triangle that appears in this manual alerts the user to the presence of important operating and settings instructions.

The letter "i" within a circle in this manual alerts the user to the presence of important information, recommendations and advice.

1.1 Connections and basic settings

POWER ON

Before plugging to AC line, **check the position of the VOLTAGE SELECTOR** on the rear panel. 200-240V or 100-130V.

Use the supplied three-prong Interlok cable and ensure proper grounding. The unit has an on/off switch on the rear panel.

CONTROL

Some basic functions are configured using the front panel control wheel. Its use is very intuitive: **Turn the wheel** to choose options and change values. **Push the wheel** to confirm a value or access an option.

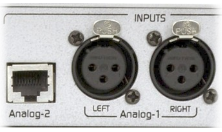
ETHERNET CONNECTION

All functions and settings of the 542APC are controlled from an internal WEB interface. Connect the 542 ACP's ETHERNET port to the LAN router, using a standard UTP cable. By default the unit works in DHCP mode. The router will assign it an IP address, which is displayed on the OLED screen of the processor.

WEB CONTROL

Using any network terminal, enter the IP address in a WEB browser. The browser will display the 542's internal WEB pages. The initial screen is a Status page. To change set and edit options, press the ADJUST MODE button located at the top right. Read this manual for details.

INPUTS

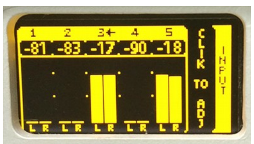


CONNECTION

Default active input is the balanced analogue with XLR connection. There is a second analog input on RJ45 (StudioHub(c) compatible) and digital inputs AES3 and IP streaming (optional). To switch the input proceed as follows.

SET UP

The analog inputs are set to operate at -18 dBfs with nominal level of +4dBu. When the mixing console works at 0VU, signal peaks at the 542APC input should be between -22 and -9 dBfs. The inputs gain can be adjusted from the front of the unit, as shown below.



INPUT SELECTION (from the frontal panel)

- ✔ Turn the navigation wheel until see the INPUT screen.
- ✔ Press the navigation wheel to access to the "active input" configuration.
- ✔ Turn the wheel to select an input (number and name of the input are shown eg "2: ANALOG2").
- ✔ Press the wheel to confirm the selection.
- ✔ The current active input is signalized by an arrow (input levels screen).

CHANGE THE INPUT GAIN (from the frontal panel)

- ✔ Turn the wheel until see the "enter SETUP MENU" screen. Press the wheel to enter.
- ✔ Turn the wheel to select "IN" and press to enter SETUP of inputs.
- ✔ Press the wheel to select "INPUT SEL". Turn the wheel to select the desired input. Press to confirm.
- ✔ Select "GAIN" by turning the wheel and press to activate the gain change.
- ✔ Turn the wheel clockwise to increase the gain and counterclockwise to decrease it. Adjust the gain so that the indicators show a peak value between -22 and -9 dBFS.
- ✔ Press the wheel to confirm the gain.
- ✔ Turn the wheel to select BACK and exit this menu.
- ✔ Turn the wheel to the last "BACK" icon to exit the SETUP menu. Changes are saved.

OUTPUTS

CONNECTION

The stereo encoder has **dual MPX** output with independent level adjustment. Connect an MPX output to the transmitter using 50 or 75 Ohm coaxial cable.

542 APC has two balanced analog stereo outputs: XLR and RJ45. Both outputs gives the same audio signal. From the WEB Panel, the de-emphasis curve (50/75uS) can be enabled/disabled for each output (for WEB-casting or link with repeaters).

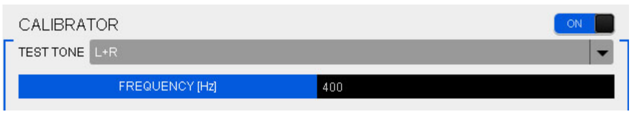
MODULATION LEVEL

The processor generates a calibration tone. To enable it follow this procedure:

Enter the **IP** in a WEB browser to access to WEB Control Panel.

Go to **ADJUST MODE**.

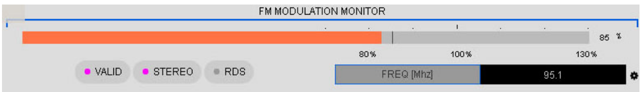
Go to the main menu at left and click on **Stereo Generator** option.



Turn on the **CALIBRATOR.** The input signal will be replaced by a sinusoidal tone. By default, the pilot tone is set at 9% and RDS at 4%.

To adjust the modulation proceed:

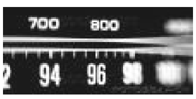
Adjust the MPX gain of the transmitter to achieve the desired modulation using the transmitter modulation meter (usually 100%). The default level for MPX output is 5 VPP. For fine tuning, use the MPX output level from the WEB Control Panel (IP access).



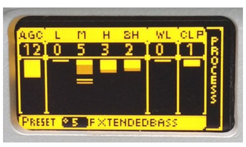
Tune the transmission with the internal tuner of the 542APC and verify 100% on the modulation meter.

Turn off the calibrator. The input signal is reestablished. Audio peaks will modulate exactly at the same level set with the calibrator.

PROCESSING



The radio station is on the air with the sound of the Solidyne 542APC. You can listen to the different default sound presets to find the sound that fits the needs of the station. Tune into a good audio system (or use good headphones). Or use the 542's internal tuner. Proceed as described below.



- ✔ From the front panel, turn the navigation wheel until the PROCESS screen is displayed. The current preset is displayed on the bottom line of the screen.
- ✔ Press the control wheel to change the preset.
- ✔ Turn the wheel to choose a preset and press again to confirm. The sound in the air will gradually change to the new setting.
- ✔ From the WEB Control interface, the current preset is changed from the PRESETS section located in the fixed monitoring area.

Factory presets can not be edited. The user has user memories to create their own settings, starting from one of the factory presets. This is done only from the WEB Control screens.

2.1 Installing

The unit can be mounted in a standard 19 "rack or on a table. In the latter case it is advisable to place the rubber feet in the base of the processor. Do not place the unit on an unstable surface or shelf; The unit may fall, causing injury to persons and damage to the unit.

The ambient temperature should be between 5ºC and 40ºC. Avoid direct sunlight on the processor or the proximity of heat sources.

The openings and grooves allow ventilation and air circulation. These openings must not be blocked or covered, so as not to interfere with the cooling of the internal components of the equipment.

The 542APC has internal protection against RF fields, which allows them to be mounted close to transmitters (AM or FM). Avoid the presence of strong electromagnetic fields (power transformers, motors, etc.).

2.2 About RJ45 in audio

With the advent of audio over IP (AoIP) various manufacturers adopt RJ45 connectors and shielded twisted pair cable to replace traditional audio connectors. A single RJ45 connector contains two balanced lines, reducing the number of connectors and the size of the rear panel. In addition, the use of structured cable facilitates installation in any city in the world due to the availability of components and tools used in data networks, avoiding welding.

At the end of the STP cable the connection to the audio device (microphones, speakers, audio interfaces) still requires standard audio connectors. For this, short lengths are used that are provided with female RJ-45 at one end and the necessary audio connector at the other end.

The balanced analog inputs of the Solidyne 542APC come with RJ45 connectors and are connected using shielded CAT-5 cable (STP). This simplifies the installation.

Figure 1: RJ45-to-audio terminals

All RJ45 audio connections are compatible with the **StudioHub** cables (http://www.studiohub.com/).

2.3 REAR PANEL

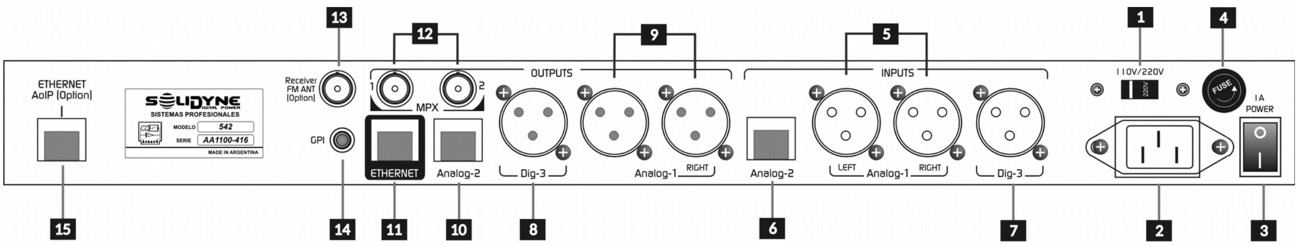


Figure 2- Rear Panel

2.3.1 Power supply

⚠	**CHECK** the position of the switch **AC VOLTAGE [1]** according to the correspondent voltage: 200/240V or 100/130V

The unit is switched on by a main switch 3. The mains voltage must be kept within a range of less than 10%. Otherwise use fast acting voltage stabilizers (ferroresonance or electronic). The unit has a general fuse 4 of 1A.

Do not use adapters that will void the power cord ground 2 Remember that the entire audio system must have a secure ground. It is recommended to follow the current regulations (Article 810 of the National Electricity Code (NEC) -USA- ANSI / NFPA No. 70-1984). These provide information for proper grounding.

Do not mix the power cord with the audio cables, especially with those that carry analog audio.

2.3.2 Analog audio connections

2.3.2.1 XLR inputs and outputs

542APC has one balanced stereo input 5 and one balanced stereo output 9 that uses XLR. The inputs are electronically balanced.

The analog XLR connects as is standard:

Balanced XLR

1 = GND
2 = Balancead signal (+)
3 = Balanced signal (-)

Unbalanced XLR

Inputs: Live = 2
GND = Joint 1 and 3

Outputs: Live (+) to pin 2; **leave pin 3 unconnected**.
GND = pin 1

⚠	Take special care to keep the phase in balanced wiring.

Use two pair shielded cable, preferably with double shielding. It is recommended to keep the length of cables less than 30 meters, although in special cases can reach 100 meters accepting a reduced loss at high frequencies.

2.3.2.2 RJ45 inputs and outputs

The balanced inputs **6** and unbalanced outputs **10** over RJ45 are connected using shielded twisted pair (STP) CAT5. While there is no standard for using RJ45 with audio signals, Solidyne devices are compatible with StudioHub (c) (USA) accessories widely used in radio stations.

The following table shows the signal distribution at the RJ45 connector.

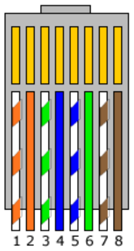
RJ-45	RJ45 PINOUT	
	PIN	CABLE COLOR
	1	Orange/White
	2	Orange
	3	Green/White
	4	Blue
	5	Blue/White
	6	Green
	7	Brown/White
	8	Brown

Table 1 – RJ45

RJ45 BALANCED INPUT	
PIN	CABLE COLOR
1 Left channel (+)	Orange/White
2 Left channel (-)	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 Reserved	Blue/White
6 Right channel (-)	Green
7 -15 (optional)	Brown/White
8 +15 (optional)	Brown

Table 2 – RJ45 balanced inputs

RJ45 UNBALANCED OUTPUTS	
PIN	COLOR DE CABLE
1 Left channel (+)	Orange/White
2 GND	Orange
3 Right channel (+)	Green/White
4 GND	Blue
5 Reserved	Blue/White
6 GND	Green
7 -15 (option)	Brown/White
8 +15 (option)	Brown

Table 3 – RJ45 unbalanced outputs

2.3.3 Digital audio connections

2.3.3.1 AES-3

The processor includes an AES-3 input and output. When using the digital input, it is also convenient to connect the analog inputs. If the signal is lost at the main input, the processor automatically switches to another input that has a signal. The default entry is selected from the "Inputs" menu, as explained below.

S/PDIF: *A S/PDIF signal can be connected to AES-3 by using an S/PDIF-to-AES3 adapter. The figure shows a compact adapter with XLR and BNC connectors.*

The AES-3 output is balanced, using a male XLR. The connections are the following:

XLR	Signal
1	GND
2	AES3 (1)
3	AES3 (2)

Table 4 – Standard AES-3 connection

2.3.3.2 AoIP input (optional)

The "Ethernet AoIP" port **11** allows receiving incoming digital audio streaming, used for transporting audio from Studios to Transmitting Plant. See "*2.8 - AoIP Connection*"

In the Studio, streaming can be generated using Solidyne DX816 mixing consoles; 2600 Series or the Solidyne ADA102 link.

2.4 GPI

General-purpose input (GPI) **14** switches the processing preset when microphones are activated in the Studio. In this way you can use a specific processing setting created for voices when the microphones are in the air. Preset presets for "01: VoiceSoft", "02: VoiceMid" and "03: VoiceLoud." Each radio station can create its own settings for voices.

The GPI uses an RCA type connector. Switching occurs when the input is energized (9-14 VDC). When this happens, the 542APC changes the current preset. When the voltage at the GPI input is zero, it returns to the previous preset.

Usually the 542APC is in transmitting plant. On 542APC models with option / AoIP the "Mic on Air" signal generated from Studios is sent directly when using Solidyne DX816 consoles; 2600 Series or Solidyne ADA102 link; Connected to the processor over Ethernet. For other consoles, the control can be solved using a data relay (Relay) of the link Study to Transmitting Plant.

When the processor is installed in the Studio, the switching can be resolved by connecting the air light voltage output to the GPI of the 542APC. This way, when the air light goes on, the equipment will switch to presets for the voice of speakers instead of the preset setting normally used in the air for music. This can be verified on the OLED screen of 542 which marks the preset used at any given time. CAUTION: Verify that the voltage delivered by the console is within the proper range (9-14 VDC).

2.5 MPX

The 542APC features **dual MPX output** 12, with independent level adjustment.

The output connectors are BNC type. 75-ohm coaxial cable (RG59 standard on CCTV) can be used for the connection. The length of this cable should be kept less than 25 meters.

When entering the transmitter via MPX, make sure that the internal pre-emphasis network of the transmitter is switched off (flat response 20 - 100 kHz). The pre-emphasis on 542APC is fixed at 50 or 75 microseconds.

It is very important to maintain an adequate distribution of ground connections. If in doubt, contact Solidyne describing the equipment and the connection diagram used.

2.6 RDS (OPCIONAL)

542APC has an internal RDS encoder. RDS (Radio Data System) is a system developed by the European Broadcasting Union (EBU). It allows adding additional information to a conventional FM signal, by including a data channel. Its main applications include:

1. The automatic tuning of a receiver to a radio network selected by the user. This allows the user to listen to the same program during a long trip by route, without having to manually tune the receiver to another transmitter of the same network, when the reception becomes deficient when leaving the area of coverage of a radio station.
2. The display on the screen of the receiver of the name of the station tuned, for example "Radio One", and the type of content that is receiving: news, magazine, sports, musical, varieties, religious, etc.
3. Automatic receipt of information related to traffic. When this feature is selected, news is prioritized over traffic, so that the receiver will automatically switch, within a same station chain, to the station that transmits information about the traffic, and once the information is completed, Tuner will return to the previously tuned station.

2.6.1 Programming the RDS

To configure and control the 542's RDS, download and install the software "Magic RDS" from the following link:

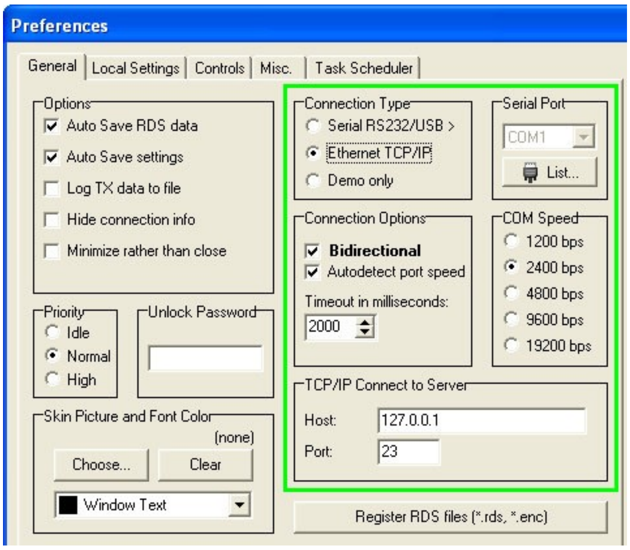
www.solidynepro.com/download/setuprds.rar

Magic RDS runs over Windows and access to the 542APC via IP through the ETHERNET port 11 connection. A LAN terminal is used to transmit data to the 542APC RDS encoder. The data can be static texts stored in the 542APC; or texts can changes dynamically, for example announcing the songs names. Dynamic text requires permanent TCP/IP access to the 542APC.

In the 542APC WEB Control Panel, the TCP port is configured in "System" tab (default 9762, see "3.3.12 System Settings"). Magic-RDS is configured in TCP/IP Ethernet mode with the HOST (IP) and port that were defined in the processor. To know about setting up and using Magic-RDS, install the application and consult the documentation.

2.6.1.1 Magic-RDS installation guide

1. Once downloaded, install and run **Magic RDS**.
2. Go to *Options → Preferences*
3. At *Preferences* choose the tab *General*.
4. Change ***Connection type*** *to "Ethernet TCP/IP"*.
5. In the field ***Port*** enter the TCP value that was defined at the 542APC. Default is 9762. The TCP for 542APC is set using the Control WEB panel, in the option "System" (see *"3.4.13 System settings"*).
6. In the field ***Host*** enter the 542's **IP**. By default the unit comes with dynamic IP enabled. The IP shows on the OLED screen at the frontal panel. To use RDS with dynamic text, that updates in real time (i.e. to show the song names) set the 542 with a fixed IP.



7. The name of the radio station enters in the field "Default PS" on the Magic RDS main screen.
8. Take in mind to change the default value of "PI" (Program Identification) to avoid conflicts with near broadcasters. Default value is FFFF (hexadecimal).

For more details about setting and use of RDS, please refers to the documentations included with Magic RDS.

2.6.2 Connecting the RDS to transmitter

542APC does not require a RDS connection. The MPX signal contains the RDS information, which is injected directly to the transmitter when the MPX output is connected.

The digital signal containing RDS information is transmitted at a rate of 1187.5 bit / s and modulates to a 57KHz sub-carrier, using amplitude modulation with suppressed carrier. This is added to the stereo multiplex signal that is sent to the input of the transmitter.

2.7 FM antenna

The BNC connector labeled "Receiver FM ant" **13** provides connection to the antenna of the internal FM receiver. An extensible telescopic antenna is supplied as standard. Depending on the installation conditions (location of the equipment, distance to the transmitting plant, etc.), an antenna (not supplied) installed outside the building may be necessary.

The internal receiver can be tuned at any station to analyze and display the level of modulation, audio levels, RDS information, among other aspects (see "*3.3.11 FM Analyzer Monitor*").

2.8 IP access

The **ETHERNET port** **11** allows the connection of the 542APC to a LAN to access the WEB Control Panel. The network router assigns an IP address to the processor (DHCP). This IP allows access to the internal web pages of the processor, using a computer connected to the same LAN and any WEB browser.

For details on use see *"3.3 - Control and settings via web"*.

2.8.1 Internet remote access

To access a device outside the local network, there are two methods:

If the device is directly connected to the Internet, enter the IP address provided by your ISP.

If the device is connected to another LAN (with internet access): configure the LAN router in order to redirect the incoming packets on port 80 to the local IP address assigned to the 542APC, and to redirect packets using port 9762 (or other port assigned to RDS).

ATTENTION! AVOID FIRMWARE UPDATES WHEN ACCESS TO THE DEVICE IS VIA INTERNET. TO UPDATE FIRMWARE REMOTELY, USE A SAFE POINT-TO-POINT CONNECTION (VPN).

2.9 Updates and upgrades

The 542APC is an Audio Processing Core on which runs an audio processing application (firmware) that defines the characteristics of the unit (model). By changing the software, you can change the characteristics of the processor. In addition, you can download free updates for the same model, indefinitely.

2.9.1 Updates

The successive updates of the same firmware can:

- ✔ Optimize processes
- ✔ Add new processes and functions.
- ✔ Improve or modify the graphical user interface.
- ✔ Add new processing settings (presets)

The current version and model are displayed on the OLED display start screen, and on the top line of the WEB status screen. To update the software version proceed as follows:

WARNING

DO NOT turn off or disconnect the unit during this operation. It can cause damage non reparable by the user.

- ✔ Go to www.solidynepro.com. In the UPGRADES section, look for the application "Discovery & Upgrade". Download the application.
- ✔ Extract the file (RAR) and install the application. The installer will create the "Solidyne" folder in the Windows Start menu.
- ✔ Download the appropriate update for your 542APC. Each firmware model has its own updates. You can not install updates of a model on a different model (for example, you can not install a firmware upgrade "FM 5 bands" on an APC processor running a 4-band firmware).
- ✔ Run the application (Start → Programs → Solidyne → Solidyne Discovery & Upgrade).

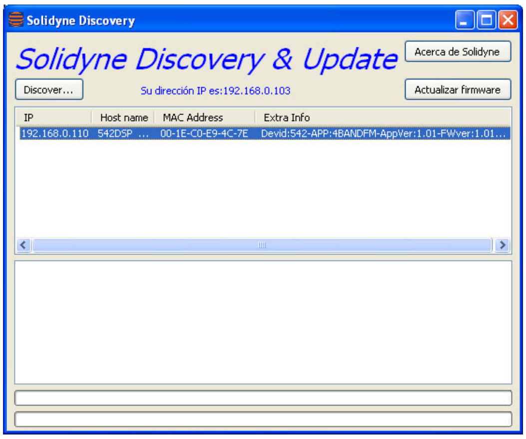


Figure 3: Update tool

- ✔ Press the "Discover..." button. A list will appear with Solidyne equipment connected to the network.
- ✔ Select the corresponding 542APC device and press the "Update firmware" button
- ✔ In the popup window select the downloaded firmware file and click "OK".
- ✔ The internal programming of the 542APC will be overwritten.

2.10 Streaming (optional)

2.10.1 Connections diagram

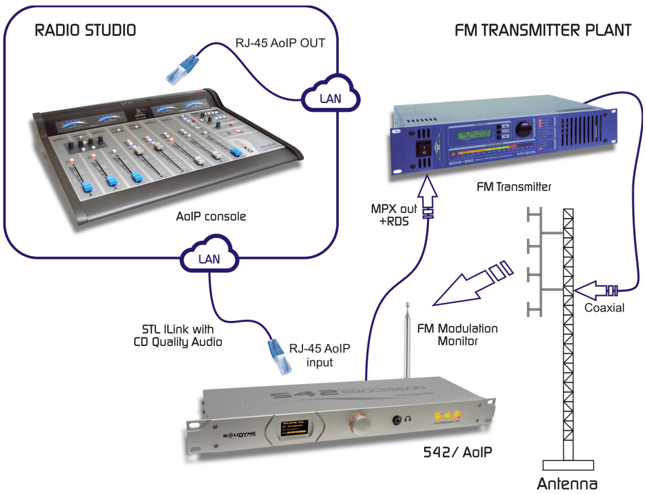


Figure 4: Basic diagram for streaming connection

2.10.2 Procedure

Step 1

Connect the **ETHERNET AoIP 15** port to the network using a CAT5 UTP cable. To configure the streaming options, access the Control Panel of the Ethernet-AoIP module. By default it comes in "Dynamic IP" mode, so it obtains an IP address via DHCP. Once the IP address is obtained, the green LED on the rear panel flashes (RJ45 ETHERNET AoIP).

ATTENTION: DO NOT CONFUSE THE IP ADDRESS OF THE AoIP MODULE (ETHERNET PORT AoIP 15) WITH THE IP ADDRESS OF THE EQUIPMENT CONTROL (ETHERNET Port 11).

The processor can also be connected directly to a modem-router, since it will usually also assign an IP via DHCP.

If the 542 AoIP module doesn't find a DHCP server, then it will search the network for a free IP address (this may take a few minutes).

Step 2

To know the IP address assigned to the 542 AoIP module, download the application **"Solidyne Discovery AoIP"** from the following link:

www.solidynepro.com\DW\IP.exe

The downloaded file is a self-executing ZIP. Running the file creates a folder called "Solidyne" which contains the necessary applications and instructions. Locate the leame-readme.txt document in that folder and follow the instructions.

Step 3

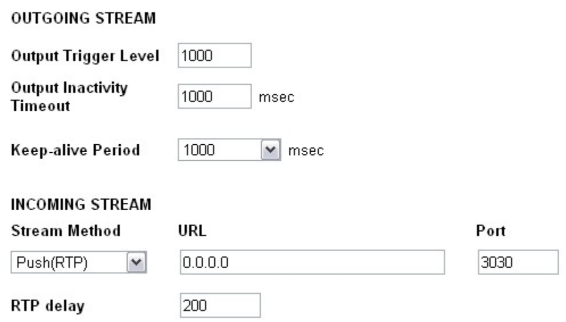
Open an Internet browser (in Firefox, Chrome) and enter the IP address obtained in Discovery AoIP. The 542 AoIP Control Panel will appear.

Step 4

At the transmitting plant, the 542APC works as a passive receiver and receives streaming from Studios. Only the port, the IP address and the transmission mode must be configured. By default, the device is set to Push RTP mode.

Step 5

Click the option “Settings → Streaming”.



In the field **INCOMING STREAM → “Port”** enter the port number assigned to the encoder. Press “Apply” to confirm the value.

Step 6

Note the size of the **input buffer (RTP delay)**, which is the memory that hosts the incoming streaming to avoid audio drops. These failures occur when the bandwidth is insufficient for the transmitted bit rate, and can be remedied by increasing the buffer size. But note that the larger the buffer size, the longer the delay is. The appropriate value depends on the network bandwidth and the transmitted audio format. The value of the buffer is expressed in milliseconds. For transmission in PCM times of 40 and 80 mS are appropriate. For compressed formats, higher buffer values are required because of the times involved in streaming decoding.

It is not necessary to set another parameter. The unit will decode the incoming stream. The audio format is defined in the encoder used in the Study.

2.10.3 Status screen

Site type: Displays the working mode. In this case it is a DECODER located in the Transmitter Plant, which behaves as a passive receiver receiving the streaming of audio generated from Studio ("Transmitter Decoder").

Stream mode: Displays the current configured event to start transmitting to the remote computer.

Keep-alive: Displays the current connection maintenance strategy.

Conection status: Status of the connection. If the connection is successful, the source IP is displayed in green (set from xxx.xxx.xx.xx). If there are problems in the connection, this field appears in red.

Incoming stream status: Incoming streaming status. In this case as the equipment is decoder, it is active.

Outgoing sream status: Status of outgoing streaming. As the device is unidirectional communication receiver, it appears inactive.

Audio input: Active audio input (analog or digital).

Audio format: In unidirectional communication the audio format is defined by the encoder. The decoder automatically detects the format of the incoming stream, which is displayed in this field.

Input/Output audio level: Real level of audio signals in dB.

Remote inputs: Status of control signals of remote unit (studio). When a line is active, the corresponding box will light green.

Relay 1... 4: Work mode of the relays.

Local inputs: Local inputs status.

Local Relays: Local relay status, which are commanded by the remote unit.

2.10.4 Ethernet-AoIP advanced settings

See the on line help in the WEB Control Panel of the Ethernet-AoIP stage.

2.10.5 Static IP

The advantage of working with a static IP address is that having a known IP facilitates access to the Ethernet-AoIP module in the future, in case you need to modify the initial configuration, avoiding the use of the "Solidyne Discovery AoIP" tool.

The default IP is 0.0.0.0 (dynamic IP enabled). To change it press the option "Configuration → Network" in the Control Panel of the Ethernet-AoIP stage

Usar SonicIP: If "Yes" the device will display its IP address through the audio output at startup. Default: "Yes".

IP address: Enter here the IP address for the device, for example: "0.0.0.0" for dynamic IP (DHCP / Bootp, IPzator, AutoIP). "192.168.0.12" for use on a LAN. Default: "0.0.0.0".

Network mask: Enter here the 4 values of the static IP mask, for example: "0.0.0.0" for a default network mask based on the IP address used. "255.255.255.0" for a class C network. Default: "255.255.255.0".

Gateway IP address: Enter the gateway IP address here: "0.0.0.0" there is no Gateway. "192.168.0.1" LAN Gateway.

The Gateway is needed when the device connects to others devices through WAN. Default: "0.0.0.0"

Primary DNS: Enter in this field the address of the Name Server (DNS) used to resolve URLs (eg www.radio.com). Example: "195.186.0.1" Default: "0.0.0.0".

Alternative DNS: In this field, indicate the IP address of an alternate DNS server in case the primary server is not available. Example: "195.186.1.111" Default: "0.0.0.0".

Syslog Address: Destination address for syslog messages sent by the BCL program via the SYSLOG command. Enter the IP of your syslog log machine if syslog messages are logged centrally. If you select 0.0.0.0, syslog messages are sent in broadcast mode. Default: "0.0.0.0"

Device Name for DHCP: Name of the 542APC device used when querying the Dynamic Host Configuration Protocol (DHCP) service. If left empty a name based on the MAC address will be created. Length: up to 15 characters.

Web server port: Defines the port where the web server of the Solidyne device can be found. If set to "0", the default HTTP port (80) is used.

2.10.6 External Control

When the "542APC / AoIP" is streamed from other Solidyne devices, data lines can be used to command special 542APC functions. The Streaming-AoIP module has four internal GPIOs. The encoder inputs control the processor switches. In the WEB Control Panel, go to "Configuration" in the main menu. The "I / O and Control" section allows to define the behavior of the GPIO.

2.10.7 About audio formats

542APC/AoIP supports the following audio streams:

- MPEG1 / MPEG2 (only half-duplex)
- PCM MSB/LSB first

The CODEC used is defined in the encoder hardware. The 542APC processor automatically recognizes most audio formats. Remember that the playback buffer size is an important parameter *(Settings → Streaming → RTP Delay)*. The value expresses the playback buffer in milliseconds.

MP3 low bitrate	400 mS
MP3 high bitrate	200 mS
PCM 44.1/48 KHz	40 mS

The optimum value depends on the audio format and the sampling rate. Small values minimize delay, but increase the chance of audio being chopped.

3.1 OVERVIEW

The basic settings can be controlled from the frontal panel. The unit must be connected to a local network via the ETHERNET port, to access to the Web Control screens using any WEB browser (Google Chrome is recommended).

3.1.1 Presets

The presets save the settings of the AUDIO PROCESSING steps, which define the sound of the radio station.

The 542APC has 16 factory presets and 16 user presets. The presets from 01 to 15 are factory settings, ready to go on the air immediately. There are various presets that cover different criteria and needs. Each of them has a name that describes it (Voice Loud, MaxBass, Maxludness, etc.).

A factory preset can be customized by copying it to a user's position for editing. By default the user settings are clones of the Loudness-1 setting. You can create a new preset by changing each of the processing steps, or by copying a factory preset to edit it (recommended).

System settings, such as input levels, MPX level, RDS, etc., are not stored in audio presets because they are global and independent of the preset selected.

3.1.2 Password

The user can assign a password of up to 8 characters (letters, numbers and signs) to prevent unauthorized persons from accessing the Web Control Panel. Each time the Web Control is accessed, the password will be requested.

By default the password is disabled (auto-login mode). If enabled, default password is 1234. **The basic settings that can be accessed from the frontal panel are not protected by the password.**

3.1.3 Modes of control

The 542APC can be controlled in several ways:

a) The essentials can be adjusted directly from the front panel of the rack.

b) Connecting to a local network, access the Web Control Panel by entering the IP address of the 542APC in any web browser on the LAN.

c) Lite Commander: It is an application that runs on a computer and accesses the 542APC (via Ethernet) to switch certain parameters (current preset, mono / stereo mode) according to a time schedule (see "3.5 Lite Commander").

d) The RM542 Remote Control hardware (optional) allows remote control of the processor using an Internet connection. It has a height of three rack spaces and a touch screen of 7 inch. Usually the remote control is installed in the Studios and also operates as Transmit Monitor.

Figure 5 - Remote Control Unit

3.2 FRONT PANEL

The front of the unit has an OLED display and a navigation wheel with pushbutton, which accesses the basic functions of the unit. Advanced setup options and audio processing settings are accessed from the internal WEB pages.

Figure 6: Start up

Turning on the 542APC will display the initialization screen (showing firmware version) for a few seconds. The display will show the current IP address, MAC, and current preset.

Turn the navigation wheel to scroll through different monitoring and adjustment screens. The handling is very simple:

- ✔ Rotate the wheel to choose options within a screen.
- ✔ Press the wheel to change an option.
- ✔ Press the wheel again to confirm the value.
- ✔ Rotate again to select another option, or press "BACK" to return to the previous screen.

3.2.1 INPUTS

The INPUT section shows the signal level present in each audio input. The active audio input is indicated by an ARROW next to the number. By default the active input is the analogue XLR.

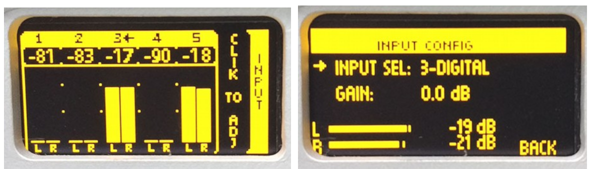


Figure 7: Entradas

3.2.1.1 INPUT SELECTION (from the frontal panel)

- ✔ Turn the navigation wheel until you reach the INPUT section.
- ✔ Press the wheel to access the "active input" setting.
- ✔ Turn the wheel to select the input (number and name of the input is shown eg "2: ANALOG2").
- ✔ Press the wheel to confirm the selection.
- ✔ The selected active input is indicated by an arrow on the level indicator screen (INPUT).

3.2.1.2 INPUT GAIN

Set the input gain to a range of +/- 14 dB. The gain for the analog inputs is set to -18 dBfs for a nominal input level of + 4dBu (AES K18). This represents a headroom of 18dB. To adjust the level, make sure that in normal use of the air console (0 VU) the peaks in the level meter of the 542APC are between -22 and -9 dBfs.

- ✔ Turn the navigation wheel until "enter SETUP MENU" is displayed and press to enter.
- ✔ Turn the navigation wheel to select "IN" and press to enter input SETUP
- ✔ Press the wheel to select "INPUT SEL". Turn the control wheel to the desired input, then press to confirm.
- ✔ Select "GAIN" by turning the wheel and press to activate the gain change.
- ✔ Turn the wheel clockwise to increase the gain and counterclockwise to decrease it. Adjust the gain so that the level indicators show a peak value between -22 and -9 dBFS.
- ✔ Press the wheel to confirm the gain.
- ✔ Turn the wheel to select BACK and exit.
- ✔ Turn the wheel to the last "BACK" icon to leave the SETUP menu. Changes are saved automatically.

3.2.2 OUTPUTS LEVELS

Figure 8: Outputs

Shows the audio level at each output. From the frontal panel, the output levels appears into the SETUP options.

3.2.3 PROCESSING

Figure 9: Processing

It shows the action of the AGC, the multiband compressor and the wideband clippers. It is possible to change the processing setting by pressing and turning the scroll wheel. To confirm a setting, press the wheel. The processing parameters can only be edited from the web control screens.

3.2.4 FM RECEIVER

The internal FM tuner allows you to tune in to any station and analyze various transmission parameters. There are three screens that display transmission parameters: FM MONITOR, FM MODULATION MONITOR and RDS ANALYZER.

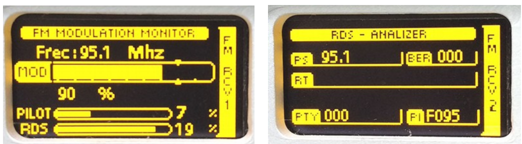


Figure 10: FM receiver

3.2.5 Headphones

Adjusts the level of the headphone output, and allows you to select the signal source from the following:

Figure 11: Headphones level

- Analog and digital audio inputs.
- The FM tuner.
- The AGC output.
- Audio processed (de-emphasized).

3.2.6 SETTINGS

To access to settings options, press the navigation wheel in the SETUP MENU screen:

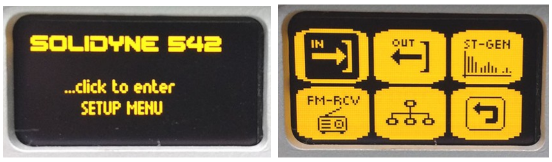


Figure 12: Settings

Basic settings are:

INPUTS: Set the current active input and the input gain.

OUTPUTS: Allows to set the levels for the audio outputs.

STEREO GENERATOR

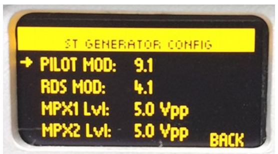


Figura 13: Stereo Coder

Allows to set the level for the pilot tone; moduation for RDS sub-carrier and the level for MPX-1 and MPX-2 (in volts peak to peak).

FM RECEIVER

Figure 14: FM Mode and tune

Allows to set the FM band type and tuned frequency.

ETHERNET

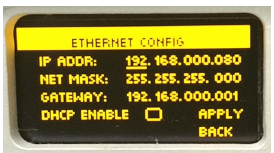


Figure 15: Ethernet

Allows connection to a LAN to access the internal WEB screens. By default, the device comes in DHCP mode. The home screen displays the current IP address.

3.3 Processing Presets

The factory presets was created to meet different needs of broadcasters. There are settings that prioritize sound quality, settings that prioritize the loudness, bass-boosted settings, treble enhancement, optimized for word intelligibility, and more. Presets from 01 to 03 are settings for voices. Presets from 04 to 13 are general purpose and are sorted by loudness level, from lowest to highest. The user can copy and edit a factory presets to create their own preset, as is explained below.

The following table shows the loudness of each preset referred to 100% modulation (400Hz@100% = 10 LU). The measurement was made with a 6-minute program material that combines fragments of music with different dynamics and different cuts of voices.

PRESET	LOUDNESS (LU) 10LU = 400Hz @ 100%
01 VOICE SOFT	-0,8 dB
02 VOICE MID	0,1 dB
03 VOICE LOUD	1,4 dB
04 Soft Processing	-2,9 dB
05 DeepBass	-0,6 dB
06 XtendedBass	-0,5 dB
07 Vocal Music 1	-0,2 dB
08 Vocal Music 2	0,3 dB
09 High-Boost	0,3 dB
10 Loudness 2	1,6 dB
11 MaxLoudness 1	1,8 dB
12 MaxLoudness 2	2,1 dB
13 MaxLoudness 3	2,8 dB
14 BS412 Hi-End	ITU BS.412
15 BS412 Puch	ITU BS.412

At next the concept behind the presets are described. We recommend to rear carefully the following notes.

3.3.1 VOICE PRESETS

The VOICE SOFT/MID/LOUD presets are set to process voices. When the microphones are enabled in the Studio, the 542APC switches preset to process the audio with a setting designed for the voices (see 3.3.12.2 Input action).

The main difference of adjustments for the voice, regarding the adjustments made for the music are the times of attack and recovery times of the AGC, of the compressors and of the wideband limiter.

The wideband AGC must recover its gain fast enough to compensate, for example, a low-level telephone communication. This is achieved with the quick window settings, as if the overall WB AGC times are very fast it will cause constant and audible level fluctuations.

The hold threshold is set to values between 6 and 8 dB higher than the values used for music, to avoid constant level changes. To make this adjustment, program material was used with a combination of voices and cuts that alternate words and fragments of music.

The attack of the WB AGC must also be fast, but a very fast time will cause audible level changes because of screams or laughter that can occur in front of the microphone in some types of shows. The action of the AGC must go unnoticed.

The multiband AGC should not have a very wide correction range. Between 2 and 3 dB is sufficient for most purposes. The coupling between bands should not be greater than 3 dB to avoid imbalances in the spectrum, except for the bass band that tolerates differences of 4 to 5 dB. This is sometimes convenient as it helps to balance the presence of bass between different voices. Remember that you can use the MB AGC Dynamic EQ to define an equalization profile.

The attack times of the compressors must be fast, mainly in the LOW band, to contain the great impulses that take place in the attacks of the word, when the compressors begin to work. If the attack times are slow, excessive processing and clipping of the signal may occur at certain speaker inputs. In other words, the announcer will sound "saturated" or "dirty" for a brief moment, when he begins to speak (and after each pause greater than the recovery time).

On the recovery times there is more freedom of action, so they adjust according to the type of voices that the station handles. As a general rule, slow recovery times produce a "softer" and natural sound, while fast times increase the loudness but processing becomes more aggressive (greater compression/limitation).

With respect to the gains (Drive), the voices do not tolerate too much multiband compression. An excessive processing will sound unpleasant to the ear and will detract from the naturalness of the voices.

TAKE IN MIND

- When creating your own setting for the voices, be aware that the equalization should not be too far from the setting used for music. It is recommended that the density equalizer in the setting for voices maintain the equalization profile used in the processing setting for the music.
- When using GPI switching, we recommend customizing the settings used for voice and music so that the SUPER BASS and STEREO ENHANCER stages remain in the same state in both programs. Stereo Enhacer processing will have no effect on the voices, since they are monaural (the same signal on both channels).

3.3.2 SOFT PROCESSING

This is the softest preset for music. It is an adjustment of great dynamics and less loudness. It seeks to respect the balance of the orchestra, maintaining the dynamic expression, within the limitations of FM transmission.

For this the DRIVE and THRESHOLD controls of the multiband compressor are adjusted to produce less compression. The Density EQ works on the order of -4dB, resulting in a higher dynamic range at the expense of less loudness. Remember: excessive multiband compression can cause a spectral imbalance that is very audible in Jazz music, acoustic instrumentation and orchestras. Because of the characteristics of these musical genres, a good ear will judge the audio processing that would be appropriate in rock & pop music.

The times of the bands MID-2, Presence and HI require special attention, since there is much participation of solo instruments. The recovery times of the bands must be similar, otherwise modulations of the timbre in some instruments may occur.

The adjustment strengthens the bass slightly. The bass dynamic is maintained with ½ second recovery time for the LOW band.

High frequencies are not emphasized. It seeks to prioritize the warmth, "sharpness" and definition of the instruments over the "bright effect". For this reason the recovery time of treble band (HI) is relatively slow. Keep in mind that the listener can always emphasize the treble on your tuner (and usually does!).

Regarding broadcasters, this preset will not achieve high impact voices, with "fat" basses, because it works with little multiband compression. It is recommended to use GPI switching and use the Voice-Soft preset for voices.

3.3.3 DeepBass/XtendedBass

Both settings emphasize bass. DeepBass reinforces very low frequency present in the original material, at frequencies around 96 Hz, and maintains the "punch" by releasing the compression threshold of the LOW band and adjusting the limiter / clipper of that band to almost 100% (0.25dB). XtendedBass uses the "SuperBass" process to extend and reinforce low frequencies in the mid-bass area (100 - 300 Hz).

While the first adjustment reinforces low-performing speakers with good low-frequency response, the second is appropriate to force the presence of bass on systems with small, limited-response speakers.

3.3.4 Vocal Music

These settings are created with music in which the singer predominates in the foreground. To create the preset were used orchestrated pieces and music with rhythmic bases. The goal was to keep the vocal range "soft" to get "clean" and defined vocals.

The AGC MB has its range of action very limited, to avoid colorations and sudden changes of level due to the great dynamics that many of the songs with predominance of the singer have.

3.3.5 MaxLoudness

This presets provides the loudest sound. The main objective was to achieve a great sonority in the air, with the minimum possible distortion. In presets of maximum sonority the spectral balance can be altered, because frequencies of the average range are emphasized.

The dynamics are greatly reduced to achieve high levels of average energy. MaxLoudness settings are commonly used, as they respond well to different music styles. They use different clipping level for MPX (overdrive). MaxLoudness-3 was created with pop hits of the last decade, that have strong presence of electronic rhythmic bases and synthesizers, and very high levels of loudness according to the criteria used in the mastering process.

The combination of hard compression with high gain results in a compact sound, which may sound "rough" in the air. Many FM stations that prioritize sonority versus sharpness and definition are especially looking for this kind of sound.

3.3.6 Presets optimized for ITU BS.412

Presets "BS412 Hi End" and "BS412 Punch" are optimized to be used with the MPX ITU BS.412 power control.

Both presets can also be used with the BS.412 control disabled. The broadcasters regulated by BS.412 are free to use any of the factory presets, but the optimized presets make better use of the dynamic range that is generated by the relationship between bounded MPX power and 100% modulation.

3.4 WEB CONTROL PANNEL

Configuration options and audio settings are accessed via IP. To access it is necessary to connect the equipment to a LAN, using the Ethernet port. Entering the IP address in a web browser on the network accesses the control screens of the processor, which are internal WEB pages of the 542APC.

The current IP of the 542APC is displayed on the OLED screen on the front of the unit

Figure 16: WEB Control Pannel

From now on, the screenshots of the control interface are shown with **inverted colors**, to facilitate the printing of this manual.

3.4.1 Status

The home screen is a monitoring pannel that shows the status of following items:

- Audio processing
- Input signals
- Output signals
- Transmission

From the status screen you can not change settings or switch the processing preset. In order to edit parameters and settings, enter to ADJUST MODE from the upper right corn .

3.4.2 ADJUST MODE

The ADJUST mode allows the user to edit the parameters of the 542APC. Access to this mode may be restricted by password (see 3.3.13 - System Settings ").

In the ADJUST mode the monitoring area is always visible. It has two levels of work:

BASIC VIEW: Displays only basic device configuration options, and allows you to change the current preset.

ADVANCED VIEW: Displays all controls on the 542APC. In this mode you can access the audio process setting and advanced system settings.

WARNING:

- ✔ Always press SAVE PRESET after change a value in a processing stage. SAVE PRESET will flash when a change is not saved.

- ✔ When edit a system configuration parameter, SAVE SETTINGS must be pressed (main menu at left). The SAVE option remains flashing when a change was not saved.
- ✔ **The current preset is a system value.** If the currently used preset is changed, SAVE SETTINGS must be pressed for the change to be stored in the hardware. If the preset change is not saved, in case the processor is restarted, it will load the last stored configuration.

3.5 INPUTS

To access to the input settings, click on the main menu the icon indicated by dotted circle.

Here you can set the gain and the stereo balance for each input (analog XLR, analog RJ45, AES3, Streaming-AoIP y FM Receiver).

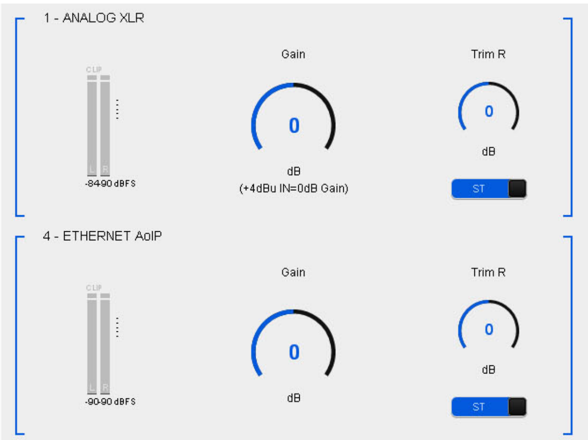


Figure 17: Analog inputs XLR and ETHERNET

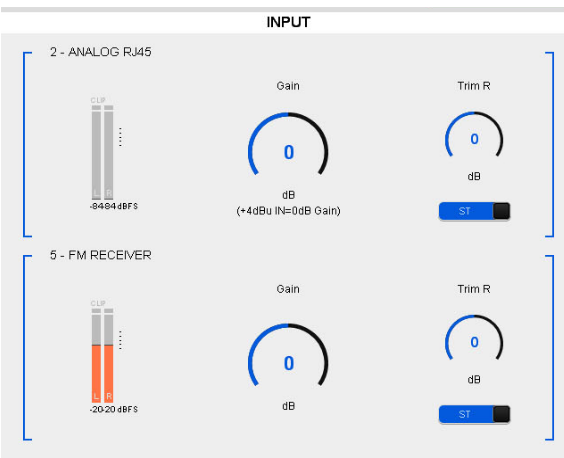


Figure 18: Analog input RJ45 and FM RECEIVER

The main input defines from the "MAIN SOURCE" drop-down menu. The 542APC fold-back feature switches to an alternative input in absence of signal. Up to two alternative inputs can be defined. If the signal at the main input fails, the unit switches to FAIL BACKUP 1. If this input does not have signal, 542 switches to FAIL BACKUP 2.

The switching takes place according to the following parameters:

LEVEL: defines the threshold below which the signal must remain to be considered as falling.

FAIL TIME: time the signal must remain below the threshold to be considered as falling.

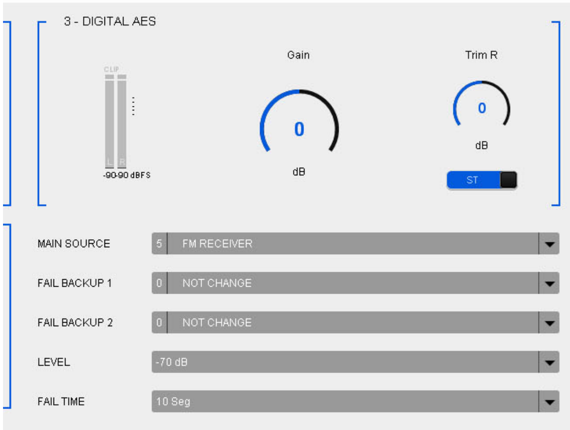


Figure 19: AES input and priority

3.6 OUTPUTS

The 542APC hardware has four audio outputs with independent level and d-emphasis.

- XLR analog output
- RJ45 analog output
- AES-3 output
- Streaming-AoIP

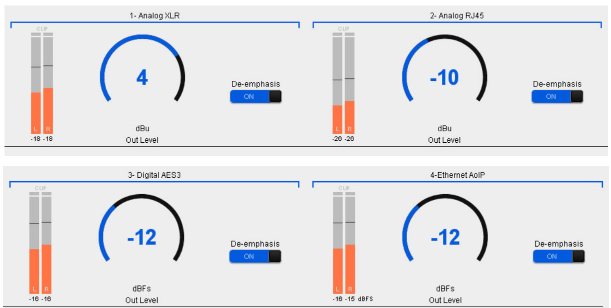


Figure 20: Outputs

3.7 STEREO CODER & RDS

542 APC includes stereo generator and RDS encoder. RDS programming is done with external software (see 2.6 RDS Encoder).

The configuration options of the stereo generator are accessed from the main menu, by clicking on the icon shown with the dotted circle.

3.7.1 MPX LEVEL

Figure 21: MPX Level

The two FM baseband (MPX) outputs have peak-to-peak (Vpp) calibrated level control. The MPX level will be adjusted to obtain 100% modulation at the transmitter.

3.7.2 Pilot Tone and RDS

Figure 22: Pilote Tone and RDS

Figure 23: Advanced settings

PILOT: Set the level of the pilot tone. At the factory, the pilot tone is set to 9%. In areas with high radiofrequency congestion, it can increase up to 12%, but keep in mind that if the Pilot Tone and RDS values are increased, the audio will be reduced for the same level of modulation.

RDS: Determines the modulation level of the RDS subcarrier (factory 4%).

MODE: The unit can be switched to mono or stereo. When this control is switched to MONO, the pilot tone is suppressed and the MPX component signal will always be MONO (see 3.5.3 - On monaural transmission).

Switching to mono can occur in several ways:

- From this control.
- Forced by preset (presets has a mono/stereo flag, see 3.4.2 - Preset Manager). In this case the control will indicate "Forced to MONO by preset".
- It can be switched manually from a computer using Lite Commander application (requires remote IP access).

Pilot Phase: Pilot Tone Phase Fine Tuning is an advanced control that improves stereo separation (see 3.5.2 - About channel separation).

SUB (L-R) adjust: Changes the level of the L-R module to compensate for mismatches that may occur in the coaxial-antenna assembly (see 3.5.2 - About channel separation).

3.7.3 Calibrator – Modulation level

542APC generates a calibration tone useful to set the modulation level. To adjust the modulation see 3.5 - Set the modulation level.

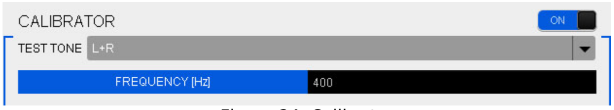


Figure 24: Calibrator

3.8 MPX power ITU-R BS.412

This stage manages the power of the MPX signal according to the recommendation of the International Telecommunication Union ITU-R BS.412 **for European countries**. It is accessed from the ADJUST mode by pressing in the main menu the icon indicated by the circle dotted in the figure.

> If the legislation in your country does not require this regulation, DO NOT enable this function, since it will reduce the loudness.

Recommendation ITU BS.412 was created to eliminate interference between adjacent FM channels; since the separation between channels in many European countries is 100 KHz. It was found that the source of the interference was the density of the program material produced by the current audio processing methods, which generate that the carrier is continuously modulated around 100%.

The BS.412 sets a maximum allowed value for the power in MPX, in order to reduce the density of the modulating signal, but maintaining the maximum modulation level at 75 KHz. There is then a double limit: maximum modulation peak and average power in MPX (integrated in 60 seconds). In other words, BS.412 forces broadcasters to decrease the volume by limiting the power allowed in MPX. As a result of this regulation, all the radios will sound in the air at the same level, without interfering with each other.

Recommendation BS.412 defines a maximum power level and a measurement algorithm, which defines the reference 0 dBr for the maximum MPX power value integrated in 60 seconds, with a tolerance of +0.2 dB.

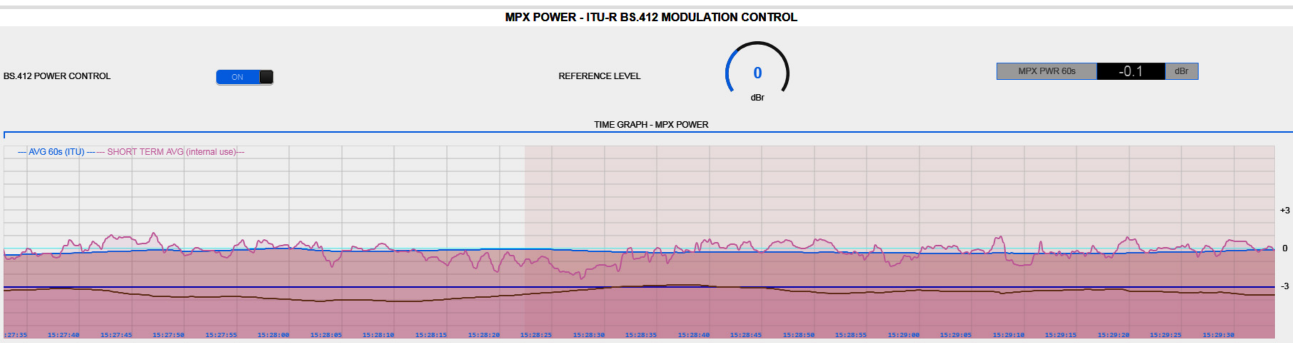


Figure 25: BS.412 MPX power time graph

Not all European countries apply the BS.412 recommendation in the same way. Some regulations allow levels higher than 0 dBr or have staggered objectives to implement the MPX level reduction until reaching 0 dBr. The BS.412 ITU modulation control in the 542APC has a control that allows the user to modify the MPX power reference level.

The timeline graph allows you to visualize the evolution of the MPX power in the last 120 seconds. Displays an integration curve of 60 seconds as established by BS.412; and a short time integration profile.

3.8.1 Audio presets and BS.412

All factory presets can be used with the MPX power control. The processor will always maintain the MPX level as allowed. When there is a power limit, it does not make sense to use high loudness presets; because the audio will be processed to increase the energy, and then processed to decrease it. One advantage of working under the ITU BS.412 regulation is that the war for loudness disappears, and broadcasters are able to use more dynamic range. The Solidyne 542 APC includes two factory presets optimized to work in conjunction with the ITU BS.412 power control, that take advantage of available dynamic range.

3.9 FM ANALYZER

The 542APC processor has an FM tuner that allows measurements on the received radio wave. Usually the tuned radio is the station on which the 542ACP is working, but the user can tune into any station and view its transmission parameters.

The "DOWNLOAD FM REPORT" option (at the bottom of the window) dumps the report into a .txt file. The report is saved in the browser's default download folder.

3.9.1 FM Tuner

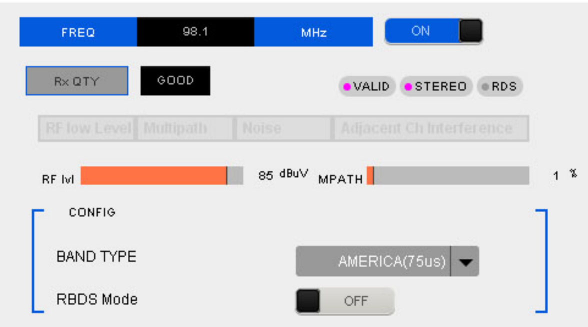


Figure 26: Tuner

The FM tuner is associated to the modulation monitor, that shows several aspects on the FM transmission, by the analysis of the RF signal of the tuned station.

VERY IMPORTANT
The measurements are valid only when the radio station is correctly tuned. If the tuned station is weak (RF Level less than 40dB) or has a lot of multipath distortion (greater than 10%) the measurements will not be valid. In this condition, the data fields will be darkened.

ON/OFF: Turn on/off the tuner. When the reception is bad or is not possible to tune the current dial, the tuner turns off.

FREQ: In this text field the user enters the radio station dial in MHz, using a dot for the decimals (eg: 99.5). To tune a station proceed:

- Enter the frequency into the field “FREQ”. Press ENTER to confirm.
- When ENTER is pressed, the dial updates. If the tuner was off, it turns on.

Reception Quality: Indicates if the reception of the tuned station is adequate to validate the measurements. In case the reception is inappropriate, the variables will be indicated out of range (RF low level, Multipath, Noise, Adjacent channel interference). When reception does not allow measurements, the data fields are dimmed.

RF LEVEL: Shows the RF level at the 542’s antenna. The RF level should be greater than 40dB for measurements to be valid.

Multipath: Percentage of distortion caused by propagation by multiple reflections of the radio wave. The level must be less than 10% for the measurements to be valid.

Band Type: Determines the type of FM band, which varies by region.

RBDS Mode: Activates RBDS, which is a modification of the RDS standard used only in the USA.

3.9.2 Modulation Meter

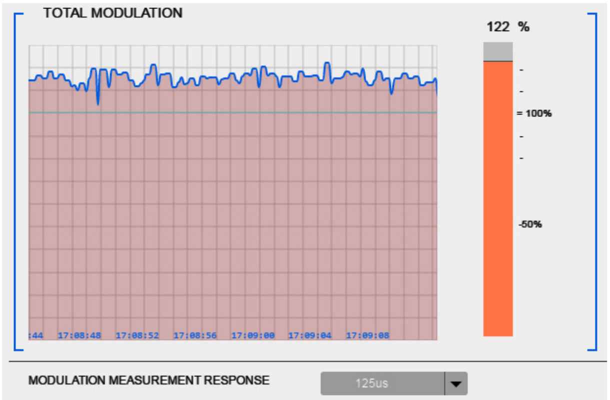


Figure 27: Modulation

TOTAL MODULATION: Indicates the FM modulation level measured on the tuned signal.

MODULATION MEASUREMENT RESPONSE: Measurement integration time. You can change between: Instant, 125uS; 250uS; 500uS and 1mS. 125uS is recommended for peak measurement. The appropriate integration for measurement to have legal validity varies according to the standards of each country (some allow integration of 1 mS).

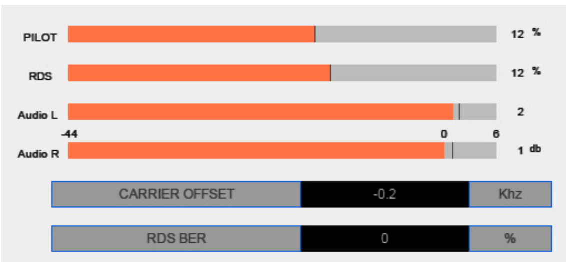


Figure 28: Transmission parameters

PILOT: Percentage of modulation of the pilot tone. The default value is 9%.

RDS: Modulation percentage of the RDS sub-carrier. The default value is 4%.

AUDIO L/R: Audio level, indicated in dBfs.

CARRIER OFFSET: Indicates the deviation of the carrier frequency, in KHz.

RDS BER (bit error ratio): This is a measure of the reception quality of RDS data. A value of 0% indicates that no errors were detected and 100% indicates that it is not possible to decode the RDS data.

RDS DATA: Displays texts transmitted by RDS.

Figure 29: RDS Data

3.9.3 Channel separation and distortion

The 542APC allows to measure on the tuned air signal, the channel separation and the harmonic distortion + noise. These measurements provide for the set: processor; stereo encoder; transmitter and antenna.

The measurements can be made remotely and without interrupting the air emission. The Solidyne technicians will be able to evaluate, detect and diagnose remotely diferent aspects of the installation and operation of the equipment and the chain of transmission.

To carry out the measurements, the program audio is interrupted and an audible tone of very short lenght plays on-air (about 250 milliseconds).

To run the measurement press on "*RUN new measurement*". The results are shown at left.

CHANNEL SEPARATION	
> 40 dB	Excellent
35 to 40 dB	Very good
30 to 35 dB	Good
25 to 30 dB	Fair
< 25 dB	Poor

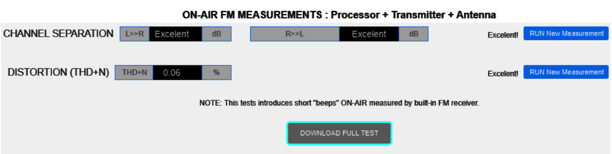


Figure 30: THD and Channel separation

542APC has advanced controls that allows to improve the channel separation. Please see *"3.11.2 - Measures and improve the channel separation"*.

3.10 Alarms and Logs

The alarm panel is accessed in ADJUST mode, by clicking on the main menu the icon indicated in dotted circle.

The 542 APC FM monitor can send alerts via email when one or several of the following events take place:

- Audio silence
- Overmodulation
- Low RF power

The alarm screen display the following options:

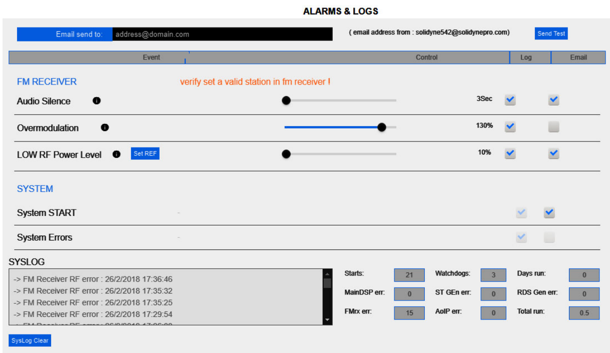


Figure 31: Alarms settings

⚠	The values are sensed based on the FM tuner. The station must be correctly tuned (see 3.4.6 - FM analyzer monitor).

E-mail send to: Enter a valid e-mail address. The alerts will be sent to that address. The "Send Test" option sends a test mail to the declared address.

If you require alerts to reach more than one email address, please note that you can use the automatic mail forwarding service of the email account.

The boxes "LOG" and "E-MAIL" enable the recording of each item in the internal memory or for sending alarms respectively.

Audio silence: Send an alert if a silence exceeding a limit established in seconds is detected. The value is adjusted with the slider on the right.

Overmodulation: Sends an alert if it detects that the modulation exceeds the value set for the alarms. The value is adjusted with the slider on the right. This value does not afect the transmission in any way.

LOW RF power level: Send an alert if a drop in the transmitted power is detected. To set the value, press "Set REF" under normal transmission conditions. Then using the slider indicate the percentage of power loss for which the alarm will be activated. This value does not afect the transmission in any way.

System START: Generates an alarm in case the equipment is restarted (for example due to power cut)

System Errors: For internal use. Only system events are logged in the SYSLOG.

SYSLOG: It is an event log housed in the memory of the processor. Reserved for use by personnel specialized in diagnostic tasks and technical support.

3.11 Calibrate the modulation

The pilot tone is factory set at 9% and the RDS carrier at 4%. Usually it is not necessary to change these values. If modifications are required, it must be done BEFORE performing the modulation adjustment. To adjust the modulation of the transmitter simply proceed as follows:

- ✔ Go to the Stereo Generator setup screen.
- ✔ Turn on the 542APC CALIBRATOR. The input audio will be replaced by a sinusoidal tone of 400Hz.
- ✔ Tune the transmission to the internal tuner of the 542APC and check the processor modulation meter for the 100% level. For fine tuning you can change the MPX output level from the WEB screen of the 542APC (the default level is 5 VPP).
- ✔ The modulation meter shows peak level. If the transmitter modulator has its own modulation meter, check the modulation there as well.

  NOTES
  The modulation indicator of the 542APC shows the modulation peak, while many transmitters have slow response meters. In that case, the indication of both instruments will be more similar the higher the peak density of the audio.

  For the modulation measurement on the 542APC to be valid, the signal reception on the tuner must be optimal.

- ✔ Turn OFF the CALIBRATOR. It resets the input audio, which will modulate peaks at the level set with the calibrator.

3.11.1 About modulation peaks

The FCC (USA) recommendations for FM radio stations are followed in many countries. Recommendation 73.268 indicates that the frequency modulation should be kept as high as possible, but not exceeding 100% in frequent recurrence peaks ("In no case is to exceed 100% on peaks of frequent recurrence"). This indicates that in momentary (and not frequent) peaks can be exceeded the 100% of modulation staying within the legal framework. The 542APC processor is designed to meet this FCC standard, allowing it to slightly exceed 100% on non-recurring peaks. When adjusting the modulation, check the regulations of your country.

When choosing and adjusting the audio processing presets, keep in mind if it is a single program for the whole day; or whether the automatic switching by microphone is activated, or by hourly range. In the first case, a commitment adjustment must be used compatible with all types of voices and music that the station handles. Automatic switching of presets eliminates compromises as each preset will be optimal for that type of music or voice. This also reduces the auditory fatigue associated with radio stations that use rigid processors, without computed control.

3.11.2 Measures and improve the channel separation

In an FM installation, the separation between channels in the transmission is affected by the coupling of the set coaxial cable - antenna. How much stereo separation lost depends on the quality of the elements and its correct installation 1.

542APC includes pilot tone phase correction and gain adjustment for the module (L-R). It allow to compensate partially the coupling of the transmission systems, improving the separation of channels in the transmission.

To make the adjustment proceed as follow:

1. On the FM Monitor Analyzer screen, perform a channel separation measurement. The higher the value obtained in dB, the better the separation. If the indication is greater than 40 dB (Excellent) it is not necessary to make the adjustment.

The measurement of channel separation introduces short audible tones on the air. It is necessary that the station is correctly tuned so that the measurements are valid (see 3.9 - FM analyzer monitor).

2. On the screen STEREO GENERATOR, increase 0,1 dB the value "SUB(L-R) Adjust".

3. Repeat the measurement for the channel separation and check if the value increased.

   If instead of increasing decrease, go to the next step (4).

   If the separation increased, repeat point 2 by increasing "SUB (L-R) Adjust" by 0.1 dB and checking again if the channel separation continues to improve. So on until it does not improve anymore.

4. If instead of increasing (improving) the channel separation of the worst of the two values decreases, then REDUCE by 0.2 dB the value of "SUB (L-R) Adjust".

5. Measure the channel separation again, verifying that it has now increased. Repeat the operation in steps of 0.1 dB until it no longer improves.

   If the maximum channel separation obtained is not "Excellent", the phase correction can be tested. Change the "Pilot Phase" control in 0.5 degree steps in one direction and in another to see if any additional improvement can be obtained.

6. Once the best possible value has been achieved (with good transmitters and antennas it should be Excellent >40 dB) the setting must be saved in internal memory by pressing SAVE in the left menu.

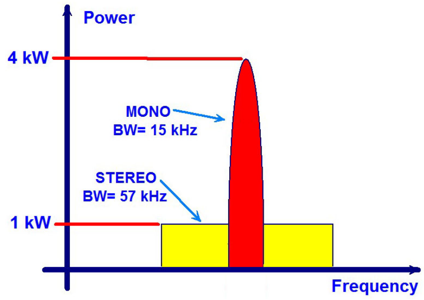
No values better than -40 dB are measured because above that separation it has been shown that the ear (even using headphones) is unable to perceive any improvement in the sensation of stereo space.

1 – *FM stereo separation degradation as a function of antena system VSWR*, by Peter Onnigian, Jampro Antenna Company, for 31º convención AES.

3.11.3 About mono transmission

Monaural transmission, although not a very common practice, is used by some FM stations located in cities with a lot of congestion (and few controls and regulation) in the electromagnetic spectrum. When the broadcast content has predominance of the spoken word (magazzines, journalistic, news, sports) the transmission in stereo does not bring significant advantages, since it is only relevant for the transmission of music.

Mono FM transmission increases the coverage area of a station, and significantly reduces interference. Transmitter power, which in stereo mode is scattered over a large bandwidth, in mono concentrates in just a quarter of the bandwidth, increasing its effective power 4 times in the air (See NAB Engineering Handbook). This increases the range and eliminates the interference of nearby radios and multipath distortion in large cities.



This technical advantage of monaural transmission can be exploited to combat interference, improving reception and radio coverage in times when talk shows must compete for the audience. The station can transmit mono at certain times and in stereo at other times.

It is even possible to switch the transmission to mono whenever the microphones are activated, so that music and commercial messages are always output in stereo. In this case, if the voice of the speakers is preceded by a musical background, it is desirable that this background music is mono, so that when the microphones are enabled, stereo / mono switching is inaudible (even in headphones).

Switching to mono can occur in several ways:

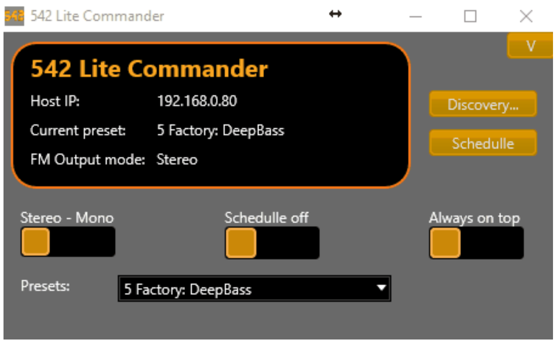
- Forced by a change of preset (the presets has a mono / stereo attribute, see 3.4.2 - Preset Manager).
- Manual switching from a computer using the Lite Commander application (requires remote IP access to the 542APC).

3.12 Lite Commander

Lite Commander is a tool for Windows that allows you to program changes of presets and switching the mode of transmission from stereo to mono, according to a schedule. The mono transmission option is useful in areas of high congestion in the radio spectrum, as it extends the coverage area of the radio. This trick can be used in news programs or sports broadcasts (see "3.5.3 on mono transmission").

Lite Commanrder is free download from http://www.solidynepro.com/DW/lite.exe. Installation and use instructions are installed with the software.

The application connects to the processor via IP, so the computer running Lite Comander must have access to the IP of the 542APC.



4.1 Input conditioning

This option is available only in ADVANCED MODE. There are process appliqued on the active audio input.

4.1.1 Pre-filter and pre-emphasis

Figure 32: Pre-filtering

LOW PASS: It is a high pass filter Chebyshev type. The cut-off frequency can be changed between 0 (off) and 40 Hz (default 20 Hz). The purpose of the filter is to eliminate subsonic audio signals, since they do not contribute anything to the music when it falls into a zone of almost no auditory sensitivity. However, they have a pernicious effect that produces an unpleasant sensation: the saturation of the amplifiers and the speakers (by excessive excursion of the cone).

HI PASS: High frequency cut filter applied to the audio band.

PRE-EMPHASIS: Adjusts the pre-emphasis curve according to the corresponding standard in each country (Europe = 50uS; USA, ASIA, Latin America = 75uS)

4.1.2 Expander

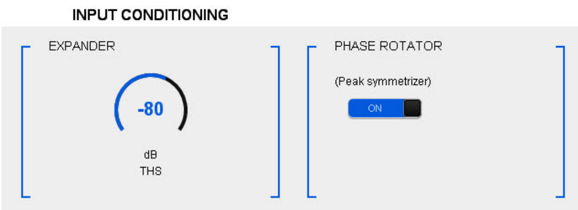


Figure 33: Expander and symmetrizer

The object of the expander is to improve the signal-to-noise ratio in the air. Multiband compression, while increasing loudness, reduces the S / R ratio. This effect would be annoying because, if not for the action of the expander could be heard the floor of noise in prolonged pauses that normally are generated with the word. The threshold is the point at which the expander begins to reduce its gain, as the signal level decreases. It is expressed in dBfs.

4.1.3 Phase rotator

It's a known fact that, by a particular arrangement of the vocal cords, the sound emission of human voice are asymmetric triangular pulses. The three cavities that filter and form these formants, to obtain the vocal sounds, do not modify this intrinsic characteristic of the human voice. The whole word spoken and still sung is strongly asymmetrical. This creates a significant reduction in the energy of the audio signal, particularly when passing through a compressor. This is because a compressor adjusts its compression level to the highest peak, regardless of its polarity. In this way when a polarity is set at 100%, the opposite polarity hardly exceeds 50%, due to the asymmetry.

It is a known phenomenon that music tends to sound stronger than the human voice, after passing through a compressor. This is because the musical sounds are symmetrical, while the human voice is not. To correct this anomaly WITHOUT INTRODUCING ANY CHANGE IN SOUND QUALITY, peak symmetrizers are used.

This technique, based on a discovery by Dr. Kahn, acquires international validity with Eng. Bonello's work, particularly that published in the Journal of AES, Vol. 24, which describes, for the first time, the theory of Its operation.

4.1.4 Stereo enhancer

Any FM station that transmits with stereo field enhancement, stands out against conventional stereo transmissions for having a "wider" sound. This effect contributes to increase the loudness perceived by the listener. When listening with headphones, the enhancement is more noticeable.

Stereo enhancement uses a stereo expansion algorithm to simulate surround sound in two-channel systems. Hearing has better response to phase changes between ears below 2000Hz, so if you increase the phase difference you get a wider stereo image.

There are two control parameters:

Figure 34: Stereo enahacement

Cut Freq (Hz) – Define the cut-off for the low-pass filter, which determinate the range of frequencies of the processed signal that is added.

Intensity: Level of the processed signal added to the dry signal.

> When voice processing switching is used, it is recommended that the music and voice presets used have the STEREO ENHANCER in the same condition (enabled or disabled).

4.2 Equalizer

4-band parametric equalizer. Its action is not canceled by the density equalizer in the multiband compression stage, because the parametric equalizers can work with specific frequency ranges, while the compression bands work with wider frequency ranges.

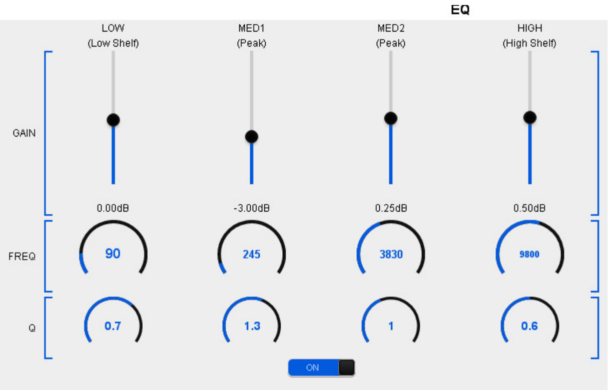


Figure 35: 4-bands parametric EQ

The parametric EQ is used for point corrections or fine adjustments, keeping in mind other adjustments in the processing chain, mainly the density equalizer and the dynamic equalizer. To determine the overall curve or equalization profile, use the dynamic EQ.

4.2.1 SUPERBASS

This control reinforces mid-bass in the range of 100 - 300 Hz, improving the presence of bass in small speakers. Unlike the equalizer, which emphasizes the components present in the signal, the Superbass reinforcement synthesizes and adds harmonics generated from the lower frequencies.

FREQ: This is the cutoff frequency of the low-pass filter. Defines the frequency range used to generate harmonics.

INTENSITY: Determines the amount and strength of harmonics generated

SUB-GAIN: Level of synthesized basses that are added to the signal.

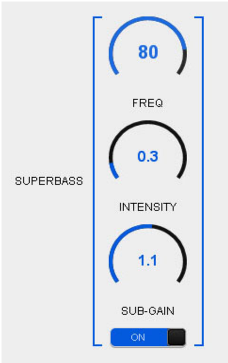


Figure 36 - Superbass

When using preset switching for voices, the settings for used music and voice should have Superbass in the same condition (enabled or disabled).

4.3 Automatic Gain Control (WB-AGC)

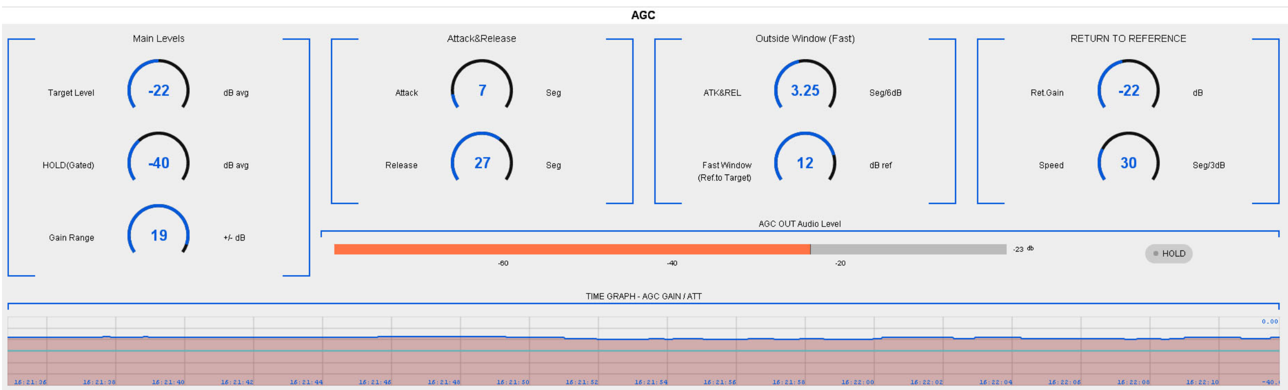


Figure 37: Automatic Gain Control (AGC)

The wideband AGC works with the full-range audio signal. Its function is to compensate for differences in level in the input signal of the processor, so that it reaches the following stages with constant level. The time constants of the AGC are a very important adjustment, and they change according to the characteristics of the sound material. Example: An optimized setting for voices will require different tempos to another one created with music.

4.3.1 Target level

Target level is the value at which the AGC adjusts the input signal, amplifying it when it is less than that value, or attenuating it when it is higher. The output of the AGC always tends to the target level. The variation of the output level of the AGC is displayed dynamically in a time chart (AGC Out).

The **gain range** set the degree of attenuation or amplification that the AGC can apply to the signal. For example: if the gain range is 20dB, the AGC can compensate for variations of up to 40dB in the input signal.

If the hold threshold falls within the gain compensation range, the signal freezes, does not compensate. For example: if HOLD is -40dB, target -22dB and range is 20dB; The AGC could compensate for a -44dB signal. But if the signal drops to -44dB the AGC freezes. The gain compensation will occur from the hold threshold.

4.3.2 Hold

The AGC is of the triggered type. If the input signal drops abruptly, the AGC does NOT change its gain, but "freezes" its current value, remaining in that state until the signal exceeds the "hold" threshold. The value does not remain frozen indefinitely. While the signal remains below the "Hold" threshold; the AGC level will slide to the target level, with the slope defined by the "Return to reference" value.

Without this feature, the AGC would continually compensate for the input level and in long silences would start to raise background noise, because in the absence of signal the AGC would increase its gain to the maximum possible. With the triggering technique this drawback is eliminated.

On the other hand, it is possible to adjust it to preserve part of the dynamic range in that music that is characterized by great changes between pianos and forts. That is to say: if after a strong passage there is a subtle input of an instrument, the AGC will be frozen in its previous level, allowing the volume contrast.

4.3.3 WB-AGC attack time

It is the time it takes for the AGC to reduce its gain when the input signal increases. As a general rule, it can be said that the attack time must be slow to prevent the AGC reacting with transient volume increases (a laugh or a musical beat). For the voice, times of about 15 sec are adequate, while for music values of 20 sec or slower work well.

The attack time only takes effect while the signal level moves into the "Outside Window". When the signal level drops out from "Outside Window", the AGC goes into quick response mode (See below for Outside Window).

Note that when the signal level increases sharply, during the attack time the signal is contained by the multiband compressors, which act strongly until the AGC adjusts its gain. Depending on the processing stage settings, a very slow AGC attack time can cause excessive signal compression (especially in the voices), which generates a very audible unwanted effect.

4.3.4 WB-AGC recovery time

When the input signal lowers its level, the AGC starts to increase its gain to compensate for the level of the input signal. The time taken by the AGC to compensate for the reduction of the input level is called the recovery time. Remember: The function of the AGC is that the signal reaches the following process stages with a stable level, independent of the output level of the mixing console.

This recovery time takes effect while the signal level is inside the "Outside Window" range. When the signal level falls out of that range, the AGC switches to using the fast response times.

Wideband AGC (WB-AGC) times interact with multi-band AGC (MB-AGC) times, as both controls changes the signal level at the same time. When the MB-AGC is active, the gain variation at the output of both AGC's will be faster than that indicated by the WB-AGC controls, depending on the range of action in which the multiband AGC is working.

4.3.5 Outside window (fast)

Defines a range or "window" of levels that determines the behavior of the WB-AGC depending on the input level. While the input signal remains within that range of levels, the WB-AGC works with the main attack and recovery times. If the input signal drops out of this window, the WB-AGC reacts using the fast times defined in "Outside Window" (ATK / REL) until the input signal returns to the "Outside Window" range. Once the signal returns to those values, the WB-AGC continues to operate using the main attack and recovery times until the signal reaches the Target Level.

The "Outside Window" range is determined by the Fast Window value, which is expressed in dB referring to the Target Level. The level window will be:

From [Target Level – Fast Window] to [Target Level + Fast Window]

Example: *Fast Window* = 12dB; Target Level = -22dBfs; then the Outside Winws will be **-34dBfs to -10dBfs**

The fast reaction times are determined by the ATK&REL value and expressed in sec/6dB. Following the above example, if ATK & REL is 1 sec/6dB and the input level drops to -42dBfs, the WB-AGC will take 1 second to increase the input level to the "Outside Window" range (-34dB). Once that level is reached, the WB-AGC is still working but with the main attack and recovery times.

	TIPS

- The attack and recovery times of the WB-AGC must be carefully adjusted so that its action is not evident. If the attack time is excessively long, the action of the WB-AGC may be noticeable (level reduction may be noted). If the recovery time is very slow and the attack time is very fast, when someone shouts (or a cough or a laugh) the WB-AGC will reduce the level abruptly and then it will take time to recover its gain. The effect will be similar to "someone turned down the volume of the radio".
- For music, the recovery time should be long. If it is very short, its action will become evident, and the volume contrasts (the dynamics of music) will be completely lost.

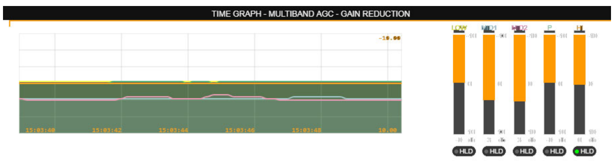
4.4 Multiband AGC

While the Wide Band Automatic Gain Control (WB-AGC) performs an adjustment of the overall level of the program signal; the multi-band AGC (MB-AGC) performs a more precise level control on each band, which allows:

- **Optimize the level in each frequency band.** The Multiband AGC can react faster than the WB-AGC to contain or reinforce the signal in each band. Depending on how the bands are leveled, and the spectral balance of the material, this increases the loudness.
- **Print to the sound a consistent equalization profile** that will remain stable regardless of the characteristics of the program signal.
- **Avoid excessive action of limiters** when one or more bands have high signal levels.

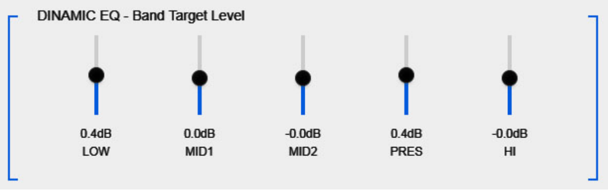
The multiband AGC stage (also known as "levelers") controls the level in each frequency band, being able to attenuate the level of a band to avoid that the compressor of that band works in excess; or increase it to reach the compression threshold. For this, the MB-AGC defines a target level for each band, and compensates the gains for the signal to remain at the target level.

The MB-AGC indicators show the gain compensation of each band, which is also shown in a timeline chart.



4.4.1 Dynamic EQ (Target levels)

Like the broadband AGC, **each band is compensated according to a target level**. When the signal in a band is greater than the target level, the AGC decreases the gain in that band; while increasing it when the signal is below the target level.



The maximum gain compensation applied is bounded by the "ACTION RANGE" control. The value indicates in dB, in each band, the maximum gain variation that will be applied to approximate the signal at the target level. For example: ± 3dB limits the gain variation by 6dB, as the signal will be attenuated or increased by up to 3 dB as appropriate.

If all target values are 0dB, the average spectral balance of the music is maintained. If a target level is defined above zero, that band will be emphasized since it will always tend to have more energy than the others. Values below zero attenuate the presence of that band. In this way, the target levels of the MB-AGC allow to define an equalization profile, based on the crossing frequencies of the filters (125Hz, 800Hz, 2.5KHz and 8KHz).

Unlike a conventional equalizer, which emphasizes or attenuates frequency ranges regardless of signal characteristics; the MB-AGC behaves like a dynamic EQ because it will reduce the gain of a band if its level is above the target level, or amplify it when the level is lower. This technique balances differences between different program materials because if a material has strong bass, the MB-AGC attenuates the bass band, but if the bass are weak, it strengthens them, making the presence of bass is homogeneous over time.

4.4.2 Band link

In order for the MB-AGC to not generate imbalances in the spectral balance, which can occur due to excessive correction on one band in relation to the others, **all the bands are linked to the action of the MID-1 band**. Each band has a maximum deviation value (in dB) from the current MID-1 level. No band may have a level difference with respect to MID-1 greater than the maximum allowable deviation.

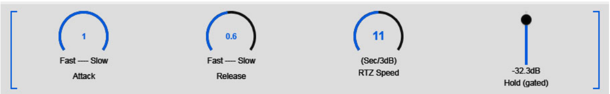
When MID-1 changes its level, it can "drag" other bands even though they are at the target level. For example, suppose the allowed correction for the LOW band is 3dB and for the rest it is 2dB. If MID-1 changes 4dB, it will drag 1dB to the LOW band (it maintains 3dB difference) and 2dB to the other bands (it maintains a difference of 2dB).

The user sets the maximum allowable correction for each band in the "BAND LINK" section.

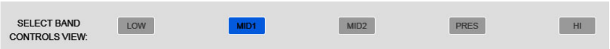
For music with vocal predominance and for microphones, the MB-AGC correction range should be narrow, about 2dB, except for the low band which can manage differences of 3 to 4dB (suitable values for balancing voices and materials with different bass levels).

4.4.3 Attack, release and hold

Each band has its Attack, Release and Hold times. These times interact with wideband AGC times.



The controls of time works over the band selected in the buttons located at right.



In general, the MB-AGC attack and recovery times are adjusted to work faster than the times used in the WB-AGC. The WB-AGC makes global level corrections, while the MB-AGC works on more abrupt level variations within a range of frequencies.

4.5 Multiband Compressor/Limiter

The goal of multiband compression is to increase the power in the audio signal over the entire audio spectrum. Below is a brief overview. Further details on how different multiband processing settings affect music and the word.

The multiband compression works with 5 fixed bands:

LOW:	20 Hz – 125 Hz
MID-1:	125 Hz – 800 Hz
MID-2:	800 Hz – 2.500 Hz
Presence:	2.500 Hz – 8.000 Hz
HIGH:	8.000 Hz – 15.000 Hz

The compressors work as limiters, because the compression ratio is very high, on the order of 60: 1, but with soft knee. The soft knee causes compression at the threshold point to be applied gradually.

The sense of dynamic range is obtained by adjusting the attack and recovery times. This happens because many signal peaks that pass through the compressors during the attack time, also pass through the clipers, since these work with high thresholds. The presence of these attacks in the final signal is fundamental for the perception of dynamic range and the sensation of "clarity" and "naturalness" of the sound

Each band has five values that define its action:

- ✓ Attack time
- ✓ Rlease time
- ✓ Hold
- ✓ Threshold
- ✓ Drive

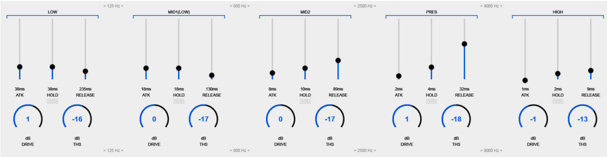


Figure 38: Multiband Compressor/Limiter

Keep in mind that both compressor and density equalizer settings vary depending on the type of sound material, so there is no single setting 100% optimal for all cases. This is why it is convenient to switch the processing when the microphones are on the air, or depending on the kind of music emitted at different times.

4.5.1 Attack times (ATK)

It is the time it takes for the band compressor to act, after the signal exceeds the threshold. Slow attack times generate more "impact" and presence in that band (especially in the low bands) but will be greater the action of the band limiter (soft clipers). This is because the sound attack "passes" through the compressor, and reaches the limiter with a high level. Limiters contains the peaks with soft clipping; a technique much harder than soft knee compression. With fast attack times, clipping is avoided, but very fast times can produce a sound that is too "boring"; without dynamics and therefore unnatural.

Attack times are set for each band. The characteristics of the material to be processed again come into play. In general terms, some musical styles such as rock and pop, tolerate more clipping, ie slower attack times. This provides a great sense of dynamic range (sound depth and impact of percussion). For orchestral, jazz, piano, it is convenient to use fast attack times.

4.5.2 Release times

It is the time it takes the compressor to return to a 1:1 (linear) ratio after the signal falls below the threshold.

Release times are also key adjustments to optimize the dynamic range feel. In general terms; if the release time is slow the compressor acts practically at all times; attenuating the level of the band. The attacks have less incidence, because they are attenuated during the release time. With very slow release times the compressor acts in a similar way to a leveler, maintaining the dynamics but without generating a great increase of loudness.

For example: electronic music requires fast release time in the low band, to reinforce the punch of techno drum. The compressor must release so that each beat is affected by the attack time.

In the treble band the same thing happens. If the release time is very slow, the attack has no effect and the treble (i.e. a Hi-Hat) lose impact. Faster release times increase the presence of treble and overall "brightness," but can produce a harsh sound for some musical styles.

4.5.3 Hold

When a signal exceeds the threshold, the compressor decreases its gain. The HOLD time maintains the degree of compression regardless of the audio level. After the hold time, the signal is released according to the recovery time; or remains compressed if the level is above the threshold. The longer the hold time the softer the compression, but the loudness decreases.

4.5.4 Thershold (THS) and Drive

Both controls affect the degree of compression applied to that band.

"Drive" amplifies the signal before the compressor. If this value is increased and the threshold is maintained, the signal will be more compressed, and the energy in that band will increase.

The **threshold** value changes the level at which the compressor starts to act. Lowering the threshold increases the compression, but decreases the energy in the band, as the compressor attenuates more signal peaks. To compensate the attenuation caused by the increase in compression, the level of that band must be raised in the Density Equalizer.

4.6 Density EQ and Clippers

The "density equalizer" allows to re-configure the spectral balance of the sound, determining in the final mix the incidence of each band.

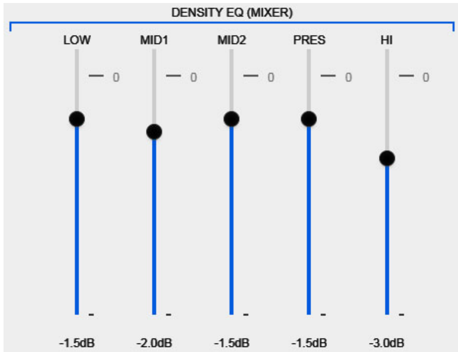


Figure 39: Density EQ

Note that the higher the level of a band in the Density EQ, the closer it is to the threshold of the limiter, which increases the loudness at the expense of reducing the dynamics with soft clipping, which generates a more "harsh" sound.

4.6.1 Band clippers

The band limiters are soft clippers that stop the peaks that pass through the compression stage due to attack times.

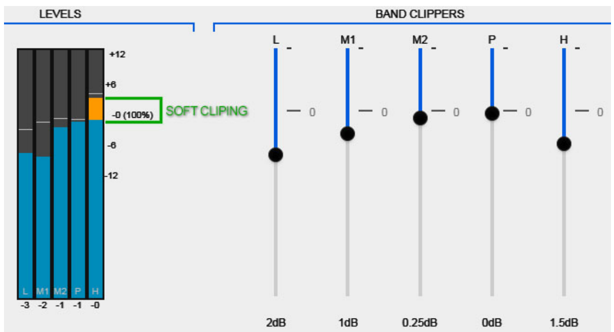


Figure 40: Band limiters

The controls adjust the clipping threshold. The indication "0" (zero) corresponds to 100% of the audio modulation. If a threshold is set below zero, that band will never be able to modulate 100% by itself; although the modulation can reach 100% in the sum of the bands. If a threshold is set at zero or above zero, it does not mean that the band will always reach 100% modulation, as that depends on the signal level present in that band, which is determined by the "Density EQ".

The Level Indicators show the level present in each band and the amount of limitation applied. The signal level in a band can never exceed the threshold of the limiter. Above the limiter threshold, the level indicator shows the amount of clipped signal in orange.

Usually the band limiter is set to 100% or up to 1 dB below zero, with the exception of the low band threshold usually set between 2 and 6 dB below 100%, to leave headroom to the other bands in the final sum.

4.6.2 WB Limiter

The sum of the bands occurs after the band limiter. Although the bands were limited to 100%, the sum of the limited signals will generate new peaks in the resulting signal, which will be above 100%.The function of the wide band limiter is to stop the peaks produced in the sum of the bands, limiting them to 100% of audio level. It is a "look ahead" type with a delay time of 2 mS. The limiter "predicts" the attenuation necessary for the signal to remain at 100% and acts with zero attack time, which prevents harmonic distortion. The threshold is fixed at 100% of the audio modulation. The input gain adjustment (DRIVE) allows amplifying the band mix, increasing the action of the wide band limiter.

“Shape” control

The wideband limiter increases the loudness for limiting levels of up to 6dB. Above 6dB, no significant increase in loudness is achieved, and instead unwanted effects in the audio are generated, due to the effect of intermodulation: the sound is heard "overprocessed", "without depth". In order to work with higher wideband limiting levels, minimizing undesired effects, 542APC implements advanced WBL control that modifies the limiter's behavior by introducing soft clipping for high energy levels. The user can adjust the degree of action of the advanced control, but the advanced control only works when the peak density is high, which allows its action to be masked to the ear.

4.6.3 Composite clipper

At the end of the processing chain, the SUM (L + R) and DIFFERENCE (L-R) signals are clipped separately with a soft clipper. The threshold is set at 100% modulation. The "DRIVE" control is decisive, since it applies gain to the addition and subtraction signals before the clipper. If DRIVE = 0 the clipper has practically no incidence; but 1dB of gain implies 1dB of clipping in the signal.

Values of 0.5 to 1dB are tolerable for most cases. Values greater than 1dB can be used but generate a harsh sound, which some identify as "FM style". Use this technique only if it is necessary to achieve high levels of loudness on- air.

4.7 System settings

Figure 41: System settings

4.7.1 System status

Shows the active hardware modules.

4.7.2 Input action (GPI VOICE)

Enables remote switching of the preset via the GPI input. Remote switching allows changing the processing preset when the microphones are enabled in the Studio. It can occur in two ways:

- Via the GPI connector located on the rear panel (see 2.4 GPI)
- On 542APC / AoIP models, the "Mic on Air" signal generated in Studios is sent directly when Solidyne DX816, 2600 Series or Solidyne ADA102 is connected to the processor by IP.

The "PRESET TO" field defines the preset to be used when the microphones are in the air (default: VOICE-LOUD)

4.7.3 RDS Remote connection

Defines the TCP port to receive communication from the RDS software. The default TCP port is 9762. To change it, enter the new value and press APPLY.

The **Solidyne-Magic RDS** software is required to control the RDS encoder (www.solidynepro.com/download/setuprds.rar)

4.7.4 NETWORK

Network settings. By default, the device comes in DHCP mode. The user can assign a static IP address. The assigned IP address is reported in OLED display of the rack.

4.7.5 SECURITY

An access password is assigned here. The password will be requested to enter the ADJUST MODE in the WEB Control screens. The "DEFAULT VISUALIZATION" option predefines the user level in which the WEB Control Panel starts: Monitoring, Basic Editing or Advanced Editing.

By default, password access is disabled. If enabled, the default password is 1234.

4.7.6 TECHNICAL REPORT

It generates a report with parameters of operation and configuration of the processor, to be sent to the technical support. The report is saved in the browser's default download folder.

4.7.7 SYSTEM TIME

When the unit is connected to a LAN and accessed from a computer, it synchronizes the date and time, which is used to record system events in the technical reports.

4.7.8 SYSLOG

It is an internal system event log, which may eventually be required for technical support.

4.8 Manage the presets

The unit has 16 processing presets, and 16 free memories for the user to create their own settings. Factory presets can not be edited. All "empty" user presets have the same values as "Loudness-1".

Figure 42: Presets

CURRENT PRESET is the active preset, the processing applied to the sound being transmitted to the air. When the current preset is changed, press the SAVE SETTINGS icon (in the menu on the left of the WEB Control Panel) so that the change is registered in the equipment. If SAVE is not pressed, the unit will return to the last stored preset if it is restarted.

REFERENCE PRESET is to load a second preset and put it on the air, in order to compare its sound against the current setting. By pressing “COPY REF TO CURRENT”, the reference preset can override the current preset when it is a user memory.

4.8.1 Create a preset

To create or modify a preset, access the ADVANCED view in the ADJUST mode.

To create a new preset, the user can directly edit any of the user's positions; or copy a factory preset and then edit (recommended). Only the CURRENT PRESET can be edited. In REFERENCE mode, editing is disabled. The preset loaded as a reference is used to compare the sound and to visualize the values of the processes.

To copy a factory preset proceed:

- ✔ Load the preset that you can copy in the field REFERENCE PRESET.
- ✔ Load the destination user’s preset in CURRENT PRESET.
- ✔ Press COPY REF TO CUR. The values from REFERENCE PRESET overwrites the CURRENT PRESET.

4.8.2 Preset Manager

The Preset Manager is accessed from the main menu of the WEB Control panel. It allows you to perform the following actions:

- Export a preset to a file on the hard disk.
- Import a preset from disk / pendrive
- Copy factory presets to user memories.
- Rename presets.
- Assign the "MONO" flag to a preset. When this preset is aired, it will cause the transmission to switch to MONO.
- Assign to a preset the VOICE flag (set for voice)

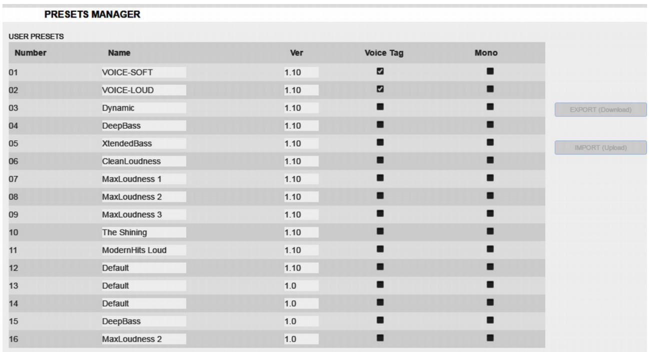


Figure 43: Presets Manager

4.8.2.1 Export/import presets

To export a preset, proceed as follows:

- Select in the list the preset to be exported.
- Press the EXPORT button (Download)
- The preset will be exported to a file named [preset_name] .542. One file is generated for each preset.
- The files are saved in the default DOWNLOAD folder of the Web Browser.

Section 5 Technical specifications

INPUTS

Stereo balanced XLR3 connector, 600 ohms
Nominal level +4 dBu. Max level +24 dBu, Software adjusted

Stereo balanced on RJ45 (compatible StudioHUB)
Nominal level -10 dBu to +24 dBu, 600 ohms

Digital AES-3 input transformer balanced
Nominal level -18 dBFS Adjust from -24 to 0 dBFS
Sample rate 44.1 KHz – 48 Khz - 96 KHz

Optional **stereo AoIP** for LAN Ethernet or Internet input

RF Input: Digital Receiver for FM Monitor and Audio Analyzer

VOICE/ MUSIC change: **GPI** = +5V ... +15V for VOICE preset (On-Air microphones). Voice/Music switch received from console in AoIP models

OUTPUTS

Analog Balanced on XLR connector
+4 dBu; Z= 600, Max +18 dBu, Flat frequency response.

Analog unbalanced RJ45 output (compatible StudioHUB)
+4 dBu/600 ohms.

Digital AES-3 transformer balanced 0VU at -12 dBFS

MPX-1 & MPX-2 for FM transmitters (Normal & Emergency)
0 - 5,5 Vpp Independent level software controlled, Z=50 Ohms

Differential output, BNC connector, floating ground 50 ohms
Allows 45 dB canceling buzz & noise due to ground loops
Protected for electrical storms 2 KV overload

Optional **AoIP** processed digital output in /AoIP models. It allows for direct audio streaming or for AoIP connection between different Studios

IN/OUT Control

Automatic fold-back to switch the input in case of absence of signal in the main input.

Frequency response

ANALOG BAL = 20 – 16 KHz +/- 0,3 dB
AES to AES = 20 – 16 KHz +/- 0,3 dB
Measured below compression & limiter threshold

Harmonic distortion

ANALOG BAL = Below 0,005 % @ 1 KHz
AES to AES = Below 0,002 %

Dynamic Range

ANALOG BAL to ANALOG BAL = 95 dBA
AES to AES = 110 dBA

Stereo separation

> 80 dBA

Subsonic filter

Chebyshev 4th order, Selectable: OFF - 40 Hz

Asymmetry cancelling

5:1 cancelling reduction using Khann-Bonello method

Expander

Software controlled with user settings

Multiband compressors

From 4 to N bands depending of software version used
Linear Phase crossover
Software adjustable automatic attack and release time

Multiband compressors

From 5 to N bands, scalable by firmware.
Linear Phase crossover
Software adjustable automatic attack and release time

EFFECTS

Super BASS effect and Stereo Enhancer is standard in all software versions

Linear limiters with predictive technology (Look Ahead) replaces the old clipper systems giving the audience a full clean sound

PROCESSING

#bands, #stages, features, etc are 100% dependent on software version used. It can work in FM, AM, HD, TV & Streaming modes

Standard 542APC works in five bands FM mode 0-50-75 uS

Latency (typical) 9 mS

POWER

115 V / 230 V (rear switch selected) 50/60 Hz, 20W

DIMENSIONS

19" rack mount. Module one (44,4 mm) // weight 3 Kg Net; (4 Kg for courier freight)

DSP STEREO CODER

DIMENSIONS

Two MPX outputs with individual remote level control by LAN or Internet

Differential output, BNC connector, floating ground 50 ohms. Allows 45 dB canceling buzz & noise due to ground loops

Level of each output adjustable from 0 to 5,5 Vpp

FREQUENCY RESPONSE

20-15.000 +/- 0,2 dB, plus 16 Khz/linear phase filter

Attenuation at 19 Khz > 80 dB

THD

From 30-12.000 Hz, below 0,01 % Measured using Belar Digital Stereo decoder DSD-1A and Tektronix Spectrum Analyzer

S/N

Better than 85 dBA with reference to 100% modulation. Measured using Belar Digital Stereo decoder model DSD-1A19" rack mount. Module one (44,4 mm) // weight 3 Kg Net; (4 Kg for courier freight)

STEREO SEPARATION

>65 dB at 1 Khz

38 Khz SUPRESSION

Below -80 dB Ref 100% modulation

57, 76 & 95 Khz SUPRESSION

Below -80 dB Ref 100% modulation

PILOT TONE STABILITY

+/- 0,002 % (+/- 0,5 Hz)

INTEGRATED RDS ENCODER

RDS / RBDS SIGNAL

Conforms to CENELEC EN50067 / EN 62106 / Control interface based on ASCII commands and UECP protocol

Built-in weekly scheduling

RDS SIGNAL BANDWIDTH

+/- 2.4 kHz (50 dBc)

SPURIOUS SUPPRESSION

>90 dB

HARMONICS SUPPRESSION

>80 dB

CLOCK REFERENCE

Pilot Tone

19KHz PILOT PLL LOCK BANDWIDTH

+/- 2 Hz

DATA CONNECTOR: ETHERNET PORT

RJ45 connector for TCP/IP LAN Ethernet Text features include dynamic PS, parsing, scrolling, fixed messages, scheduling and reading from HTTP.

DATA PORT SPEED

2400 – 9600 BPS

SUPPORTED SERVICES

PI Program Identification, M/S Music/Speech, PS Program Service, PIN Program-Item Number, PTY Program Type, ECC Extended Country Code, TP Traffic Program, RT Radiotext, AF Alternative Frequencies, TDC Transparent Data Channels TA Traffic Announcement, IH In House Applications, PTYN Program Type Name, ODA Open Data Applications, DI Decoder Identification, CT Clock-Time and Date, EON Enhanced Other Networks information