D-PRO-6HD

User's manual

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Section 1

Description

1 Description

This product is a fully multiband digital audio processor ideal for general Dtv and multimedia applications optimized for DAB+ and HD Radio broadcasting.
All audio processing is realized using DSPs, allowing for stable performance over time and extending functionality with firmware upgrades.
The audio processing chain comprises a two bands AGC which feeds the six bands limiter-compressor, followed by a final clipper-limiter operating at 96KHz, optimized for distortion masking, thus obtaining a cleaner and clipping artifacts free sound.

1.1 List of changes

1.0.0.0 28/07/2015	first release
1.1.0.0 10/09/2015	updated to new firmware 3.0 features
1.2.0.0 13/01/2016	updated to new firmware 3.1 features
3.2.0.0 14/09/2016	updated to new firmware 3.2 features
3.2.1.0 08/03/2021	updated to new firmware 3.2.1 features

1.2 Warnings

Before attempting any operation, please follow the safety instructions contained in the following paragraph.

The producer declines any liability for damage to people or things due to non-compliance, even if partial, of the following indications

• Ensure that the supply voltage corresponds to what is indicated on the apparatus.

• Ensure that the electrical system is equipped with a ground connection.

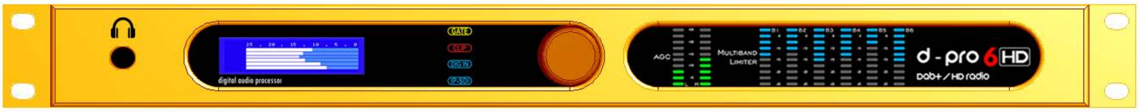
• Use only power sockets and cables with ground connection

• Disconnect power before attempting any operation inside the device.

• The power cutting device is the power cord, so this should be easily accessible and the socket must be positioned close to the apparatus.

• Any operation involving the access to internal parts must be performed only by trained service personnel.

1.3 Front panel



Front panel indicators and rotary knob

Headphone stereo jack 6,3mm (1/4")

- GATE: led is on when level is below gate threshold
- CLIP: the analog input level is too high.
- DIG IN: AES/EBU digital input is selected as audio source
- IP-SDI: Network IP is selected as audio source as audio input.

- AGC: input AGC gain level
- MULTIBAND LIMITER: multiband limiter compression level

Menus navigation and parameter editing is done through the rotary knob.

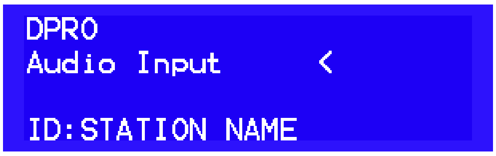
1.4 Rear panel

Connectors on rear panel

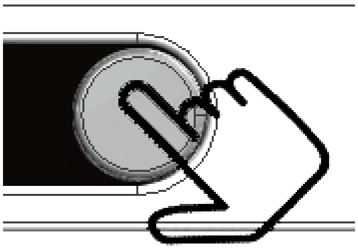
- DIG OUT (XLR M) AES/EBU output
- DIG IN (XLR F) AES/EBU input
- LEFT IN (XLR F) Left analog input
- RIGHT IN (XLR F) Right analog input
- LEFT OUT (XLR M) Left channel analog output
- RIGHT OUT (XLR M) Right channel analog output
- CTRL (DSUB 25F) Control port (option)
- RS232 (DSUB 9F) RS232 connector
- USB DRIVE (USB-A) USB connector for fallback audio on external drive (option only with ip audio decoder)
- LAN (RJ45) LAN connector for remote control or streaming audio module (option)

1.5 Editing and navigation

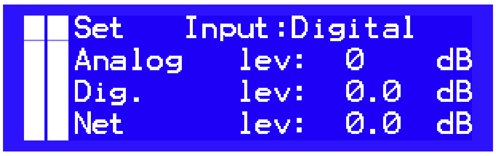
Menu navigation and parameters editing is done by the front panel knob.
On the main menu which appears after firmware boot, is possible to select the submenu for the various settings.



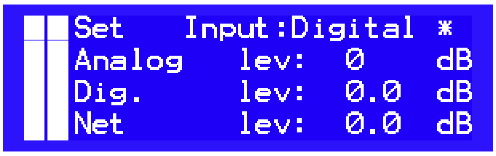
Pressing the knob, will enter the selected submenu.



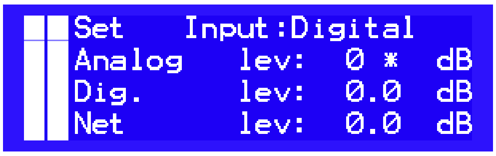
Once entered in the desired sub-menu, if there are adjustable parameters, you can proceed to editing



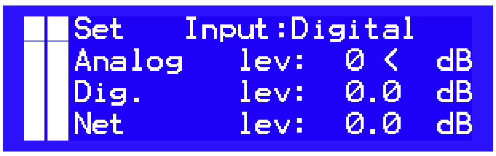
Pressing the knob, an asterisk "*" will appear next to the first editable element. This will not happen if, on the current window, there are no editable parameters.



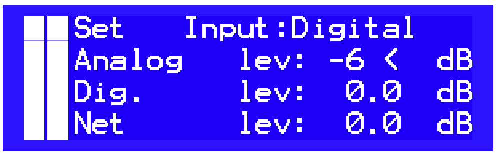
Turn the knob to position the "*" cursor on the other editable elements of the current window.



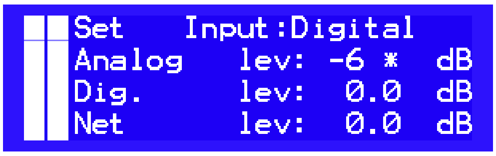
If you want now to change the analog input sensitivity, press the knob again. A "<" symbol will appear next to the parameter to indicate that this is being edited.



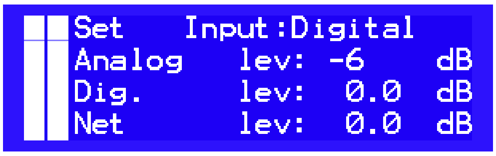
Turning the knob, you now will modify the selected parameter's value.



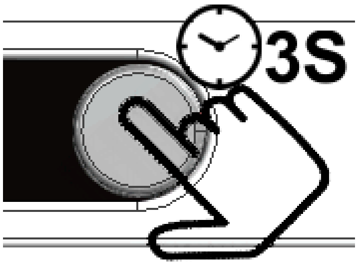
Once set the desired value, press the knob again. The cursor will switch to "*" again, indicating that items navigation mode is now active and other items can be selected and edited as described before.

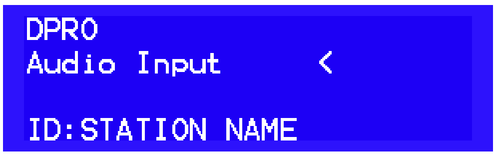


If editing is however terminated, press the knob again and the "*" cursor will disappear and you can rotate the knob to navigate to other windows.



Inside any submenu, pressing the knob for more than 3 seconds, will cause a jump to the main menu.





When pressing the knob for 1 seconds, will appear the Headphone Level regulation.

Section 2

Front panel display

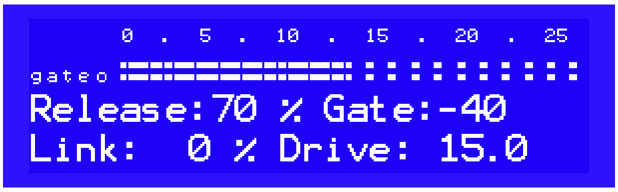
2 Front panel display

Following are listed all menus and functionalities available on the current firmware release.

2.1 Audio processor

Into this section are comprised all functions and menus related to the audio processor.

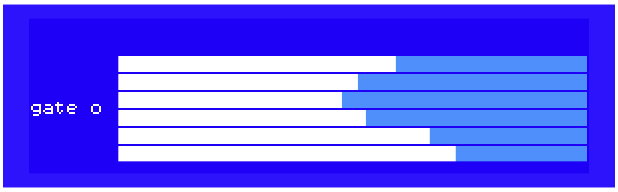
2.1.1 AGC



This window contains the settings for the dual band automatic gain control (AGC) block, placed at the head of audio processing chain.
The automatic gain control performs a pre-leveling of the signal applied to the following compressors/ limiters, keeping their working point almost constant.
At the right side of the bar which indicates the level of pre-amplification applied to the input signal, is placed the AGC status indicator: if "gate" indication is off, the input level is above the gate threshold and the AGC and compressors/limiters are operating correctly, if "gate" indication is on, it means that the input signal is below the minimum level so the AGC release is slowed down and the AGC will slowly raise its gain.

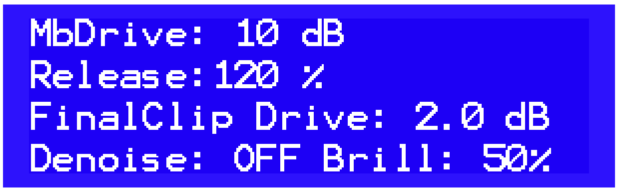
Release (...): AGC release time setting (40% fastest, 200% slowest).
Gate: Gate level serring. Below this level, the AGC and multiband limiter release is slowed down.
Link: set the maximum compression difference between the bass and middle/highs band.
Setting this value to zero, the bass compression level will be totally independent and this will cause also a mild auto-equalization.
Setting this value to 100%, it will force the bass compressor to have a gain never greater than the middle/highs band giving a sound very close to the original.
Drive: indicates the AGC input driving level, thus determining the AGC maximum gain .

2.1.2 Compression levels



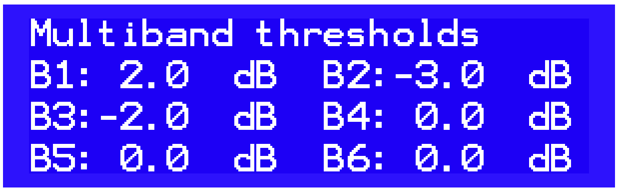
In the window, the gate status and compression-limiting levels of the multiband limiters are shown.

2.1.3 Multiband limiter parameters



MbDrive: multiband limiter drive level.
Release: Multiband limiter release time.
FinalClip Drive: set the wideband final clipper driving level. Higher driving level corresponds to higher loudness, but will cause higher distortion.
Denoise: Denoiser. If activated, it will monitor the high frequency content of the audio input signal, reducing the band 6 gain when HF input level is too low. It can be used to reduce noise coming from old tape recordings.
Brill: Brilliance high band (>6KHz) shelf equalizer, increasing the value produces an increase in the high frequencies.

2.1.4 Multiband limiter thresholds

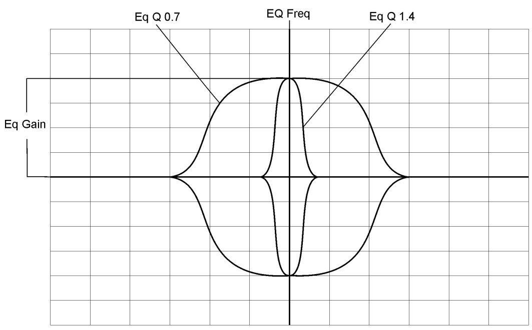


All six-bands limiters thresholds are adjustable in order to apply a dynamic equalization according to the user's taste. Lowering the limiting threshold of any band will result in attenuating the level of that band, while raising the threshold, will amplify its level.

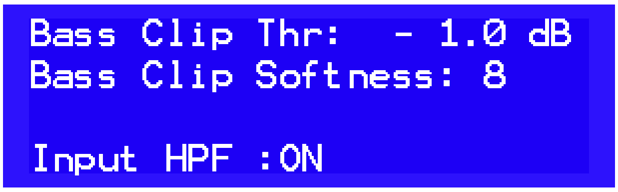
The six bands cutoff frequencies are :
B1: 100Hz and below
B2: 400Hz center frequency
B3: 800Hz center frequency
B4: 1600Hz center frequency
B5: 3200Hz center frequency
B6: 6400Hz and above

2.1.5 Bass parametric eq

BassEqFreq: parametric equalizer center frequency
BassEqGain: gain (boost) of parametric equalizer.
BassEq Q: Q-factor (bandwidth) of parametric equalizer.
Punch enh (0..2): punch enhancer (0 off, 2 max). Give some nice "analog-style" bass enhancing.



2.1.6 Bass clipper/limiter

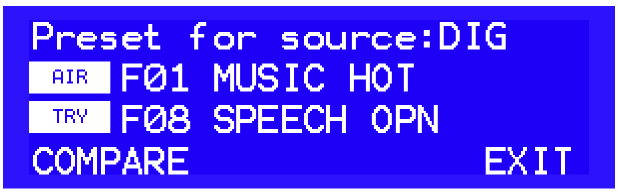


Bass clip Thr:bass clipper threshold.
Bass clip softness: bass clipper curve selection.
Bass clipper will have a curve progressively softer with increasing value.
A softer curve reduces the distortion introduced by the clipping in contrast, setting to zero (corresponding to the hardest curve), will produce a bass rich in harmonics.
Input HPF (ON/OFF): enables / disables the 30Hz high pass filter for infrasonic component reduction.

2.1.7 Loading a preset

The "AIR" field shows the on-air preset name, while, in the "TRY" field, can be specified the name of a new preset to load. Once selected the desired preset, go on the "COMPARE" label which will start to flash.
When "COMPARE" is flashing, pressing the front panel knob will cause the "TRY" preset to go on-air while the previously on-air preset will go to "TRY", allowing for comparison.

If Source Dep.Presets function was enabled, it will be possible to specify a different preset for each audio source selected so, in this case, the source for which the current preset is used will be displayed.



It is possible to change the audio source from Audio Input menu.
If necessary, make any editing you want to the factory preset (any preset starting with Fxx) selected, then save the preset to an user memory (Uxx).
Once saved, the preset will become the audio processing preset for the selected audio source, or the general preset used for all sources, depending on whether or not the function "Source Dep.Presets" was enabled.
A modified, but not saved, preset will be marked by the "MOD" prefix. This preset will be overwritten if a new preset is loaded and then edited.

Mode F00: BYPASS, has the input level as the only editable parameter and this is the same both for the Analog / AES EBU source and NetAudio.
In the case where it is necessary to put have the processor in bypass mode when any of the two sources are selected, it will be possible to compensate the differences in audio levels, if any, adjusting their input levels from the audio input menu.
If one of the sources is the Analog channel, we recommend to set the input level of the AD converter in coarse 6dB steps, then use the bypass input level for fine tuning of Analog source, then adjusting the level of the other channel, netaudio or AES EBU, from the audio input menu

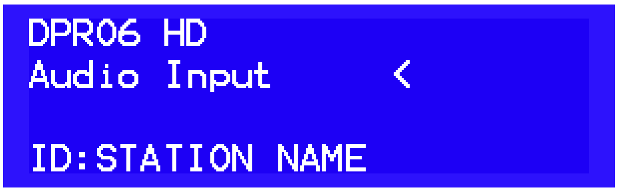
2.1.8 Saving a preset

Curr Preset name shows the name of the preset which is actually on air The name can be changed following the editing rules described before. When name editing is finished, the "*" cursor will be positioned to the right side of the name string and will allow to move on the other editable elements on the window.
Once the preset name has been edited, you can choose an user file to store it, then go to "SAVE" button which will start to flash. Pressing now the knob will result in saving the preset to the specified user file.

2.1.9 Bypass

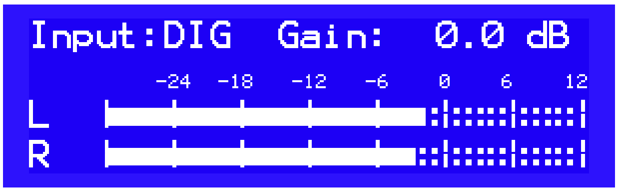
If the selected process is "F00: BYPASS", the only editable parameter will be the input level.
No other parameter is used, so the other editing windows will be hidden.

2.2 Audio input setup



In this section are present all controls related to audio sources and automatic audio fallback

2.2.1 Audio levels monitor and input level setting



In the current window is displayed, with greater accuracy, the input level of the selected channel.

The input level of the selected channel can be adjusted, but the channel will not be put on air. To change the on air input channel, the following window must be used.

2.2.2 Audio source setup

Audio input: Analog, analog input
Digital, AES/EBU input
NetAudio, streaming audio input.

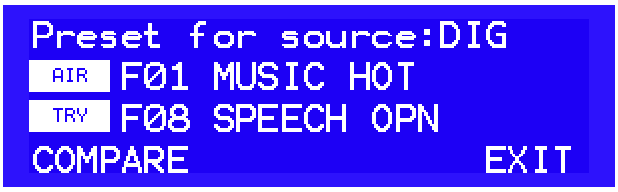
Analog lev: audio input attenuator level. This adjustment sets the input level to the A / D converter. The nominal level is indicated in the bars:

The initial adjustment of the input level should be made so that the maximum analog input level is equal to or only slightly above that limit.
In no case the front panel "CLIP" led should light.

Dig. lev: AES/EBU level adjustment
Net lev: streaming module level adjustment.

The adjustment of the individual channels is in 0.1dB steps.

2.2.3 Audio processor preset selection

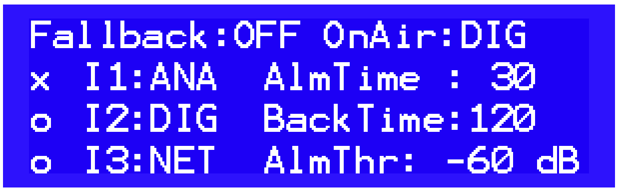


If Source Dep.Presets function was enabled, it will be possible to choose a different preset for each audio source, which can be selected in the previous menu, otherwise this window will not be visible. Once selected a preset, it can then be edit into the Audio Processor section.

The loaded preset will be assigned to the audio source currently on air.

2.2.4 Audio fallback setup (audio backup)

The processor can be set to automatically switch to an auxiliary input source if the main input audio level is below a minimum threshold.

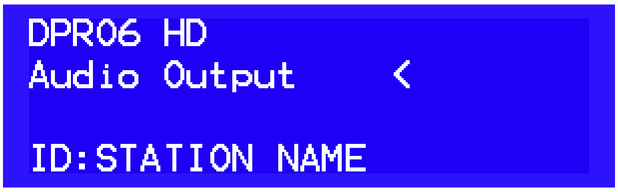


Fallback (ON/OFF): Enable / Disable automatic audio fallback on backup source.
OnAir: audio source actually on air
I1 - I2 - I3: Channel list. The channel are listed by order of priority, with maximum priority assigned to channel 1.
When an high priority channel will become unavailable, it will be replaced by a lower priority one until its recovery.
AlmTime: Alarm timeout
Back Time: Back timeout. It is the time for which the main audio source must be above the minimum threshold before it can be set back on air.
AlmThr: minimum audio level threshold

The main audio source is considered valid If the level of both channels L / R are above the alarm threshold. If a single channel is less than the alarm threshold for a period longer than the alarm timeout, the backup source will be put on air.
When the main audio source will recover, once elapsed the "Back Timeout", it will be put back on air.

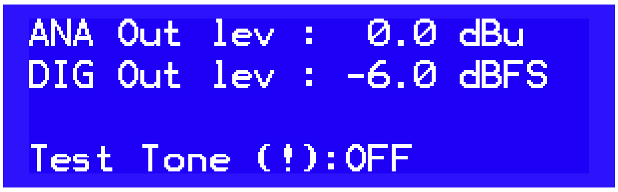
Changes will be applied upon exiting the menu.

2.3 Audio output setup



This section contains the settings relating to the DIG OUT - L R OUT audio outputs.

2.3.1 Audio output regulation



ANA LR Out : audio output level on rear XLR Left-Right OUT connector.
DIG LR Out : output audio level on rear XLR DIG OUT connector
Test tone (!): when activated, will put on air in all outputs a 400Hz tone 0dBFS.

2.3.2 Loudness limiter

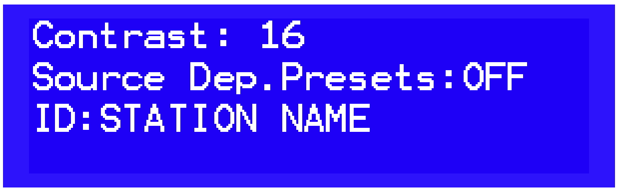
The loudness limiter can equalize the perceived loudness among different audio program sources such as live shows, commercials, films, music videos.
It operates according to the ITU-R BS1770 standard.

Itu1770 limiter (ON/OFF): enable / disable loudness control.
Reference: loudness limiter reference setting
dblim: level of limiting applied

Reference level is referred to an output level of 0dB. Other SDI, Analog, or AES/EBU Output level settings will be added to the reference value set.
Eg: SDI output level = 0.0dB, Loudness Reference = -23dB LKFS, results in 0.0 -23 = -23dB LKFS on air
SDI output level = -10.0 dB, Loudness Reference = -23dB LKFS, result in -10 -23 = -33dB LKFS on air

2.4 System setup

2.4.1 Display, Source Dependent Presets, Device ID



Contrast: display contrast adjustment.
Source Dep.Presets: enable/disable audio preset selection depending on audio source.
ON: if the function is enabled, a different audio processing preset can be specified for each audio source.
OFF:if the function is disabled, the same preset will be used for all audio sources.

ID: device identifier. It is a 16 characters long label which will be shown in the main menu screen, useful for identification when multiple devices are installed.

2.4.2 GPI

In this menu it is possible to remotely select the channel to be broadcast (optional).
Selection is possible via GPI, serial and Network card.

GPI status: status of the 8 available ports, from 7 to 0.

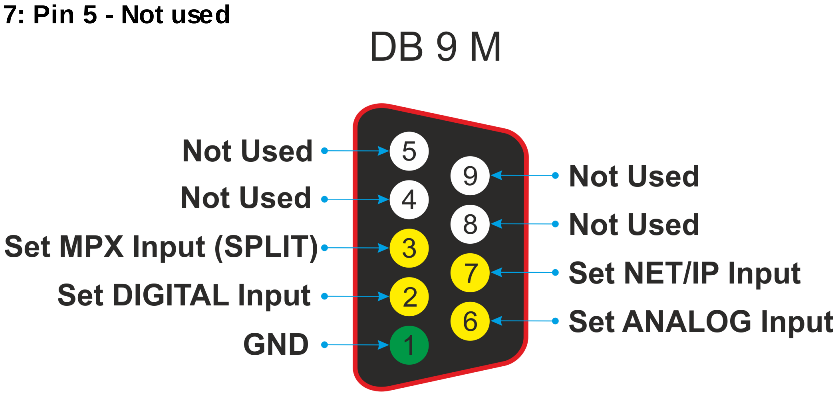
Split ctrl port: enable the function and select the communication mode.
OFF: disabled function.
GPI: active function on CTRL port.
Serial: active function on RS-232 port.
Network: active function on Ethernet port.

Invert split status: only in FM versions

Rear Connectro GPI (CTRL)

GPI port status:
0: Pin 6 - Set ANALOG Input

1: Pin 2 - Set DIGITAL Input
2: Pin 7 - Set NET/IP Input
3: Pin 3 - Not used
4: Pin 8 - Not used
5: Pin 4 - Not used
6: Pin 9 - Not used
7: Pin 5 - Not used



The inputs are activated by grounding the relative pin or by providing a voltage from 5 to 12VDC, in this case the jumpers placed on the board inside the device will have to be moved from the "CLOSE" to "VIN" position.
The selection of the channels through the GPI port takes place with exclusive priority on a single channel, if more inputs are activated at the same time the channel is restored before the selection

Example: the Analog channel is on the air, pin 2 is activated by broadcasting the Digital channel, at the same time pin 7 NET / IP is also activated, since the selection of two channels is not valid at the same time, the Analog channel returns to air .

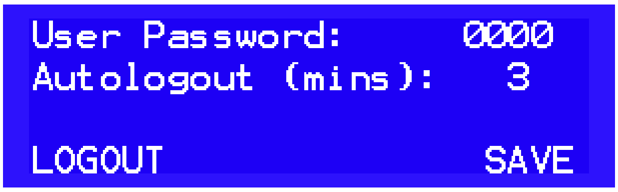
Network commands from Lan network:
By selecting Network on "Split ctrl port" commands can be sent via LAN using the UDP protocol with standard ethernet card, the TCP protocol with IP Streaming option card.

Network commans by Lan:
By selecting Network on "Split ctrl port" commands can be sent via LAN using the UDP protocol with standard ethernet card, the TCP protocol with IP Streaming option card.

[AB010103] select analog input
[AB010200] select digital input
[AB010301] select NET/IP input
[AB010002] o [AA010100] return on main channel

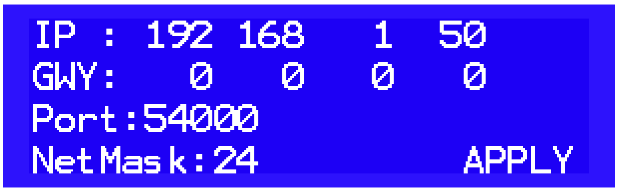
The selection command must be repeated continuously otherwise the device re-enters the main channel.

2.4.3 PassCode Setup



User Password: in this section the user can enter a numeric password that will be required for accessing the device. Setting "0000" (default value) will disable the password prompt.
Autologout (mins): Automatic lock timeout. If no action is done on the front panel knob within the specified time interval, the device will enter into lock state and the user will need to re-enter the password to log in again.
LOGOUT: immediate logout and lock.
SAVE: Saves any changes made to the numeric password or the duration of the automatic logout timing.

2.4.4 Ethernet setup



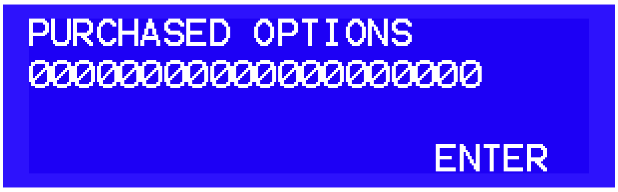
IP: current ip address
GWY: gateway address
Port: UDP port number
NetMask: netmask in CIDR notation.
APPLY: apply and save changes.

This page is visible only if the network module is installed.

2.4.5 System info

Sys: operating firmware release
Proc: audio processor coder firmware release
Hw: hardware release
BL: MCU bootloader release
SN: serial number

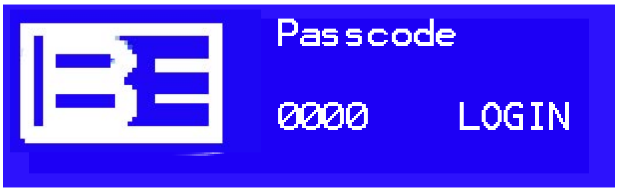
2.4.6 Options



In this window the user must enter the code to enable the optional features.
Once entered the code for the desired option, go to the "ENTER" button and, when it will flash, push the front panel encoder.
The device will verify the code and, if found correct, will activate the requested option.

2.4.7 Login window

To protect the device from unauthorized operation, a security code can be enabled by accessing the System menu, Set password.
When the device is turned on (or the auto logout time is elapsed), the login page will be shown where the pass code is to be entered.

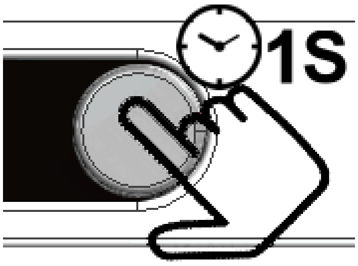


LOGIN: when the correct pass code is entered, will log in user.

Pass code lost or forgotten: after the 20th unsuccessful login attempt, an unlock code will be displayed.Call customer service and tell this code in order to get back your pass code.

2.5 Headphones Level

The headphone level can be adjusted using the front panel knob.
Pressing the knob for more than 1 second, will show the adjustment level menù.



After 3 seconds the level menù will close automatically.

Section 3

Programmming software

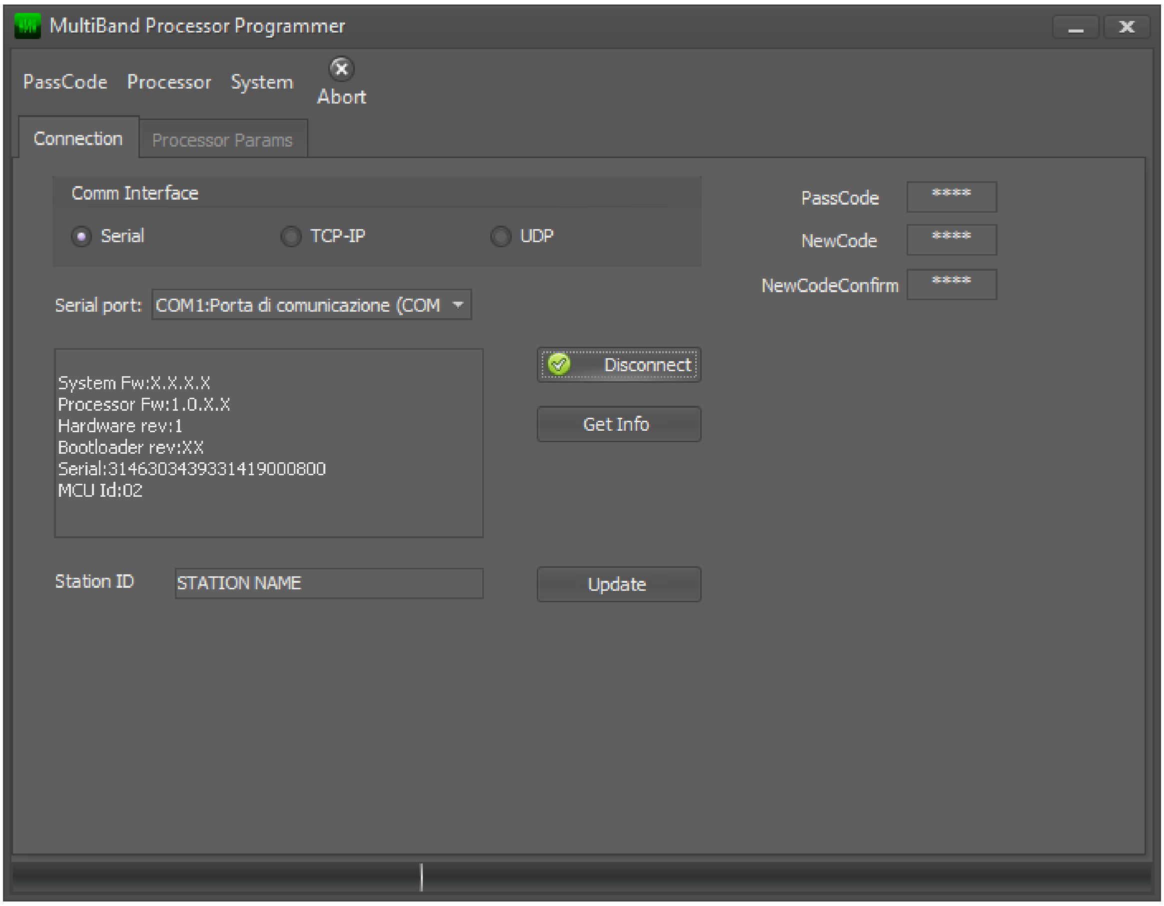
3 Programmming software

A PC based softwareMultiband Processor Programmer, compatible with 32 and 64 bits Windows Xp, Windows 7, Windows 8.X, Windows 10, is provided for processor remote control and setup.

Connection is done through:
- serial port 9600 baud 8N1
- ethernet port TCP-IP (if optional streaming module is installed)
- ethernet port UDP

3.1 Main window

In the main window the type of connection can be selected, via COM port or through Ethernet port using TCP-IP (needs streaming module) or UDP (needs ethernet module). Once the communication mode is set up and the processor connected, press "Connect". If communication is correct, the program will read all internal settings and device information.

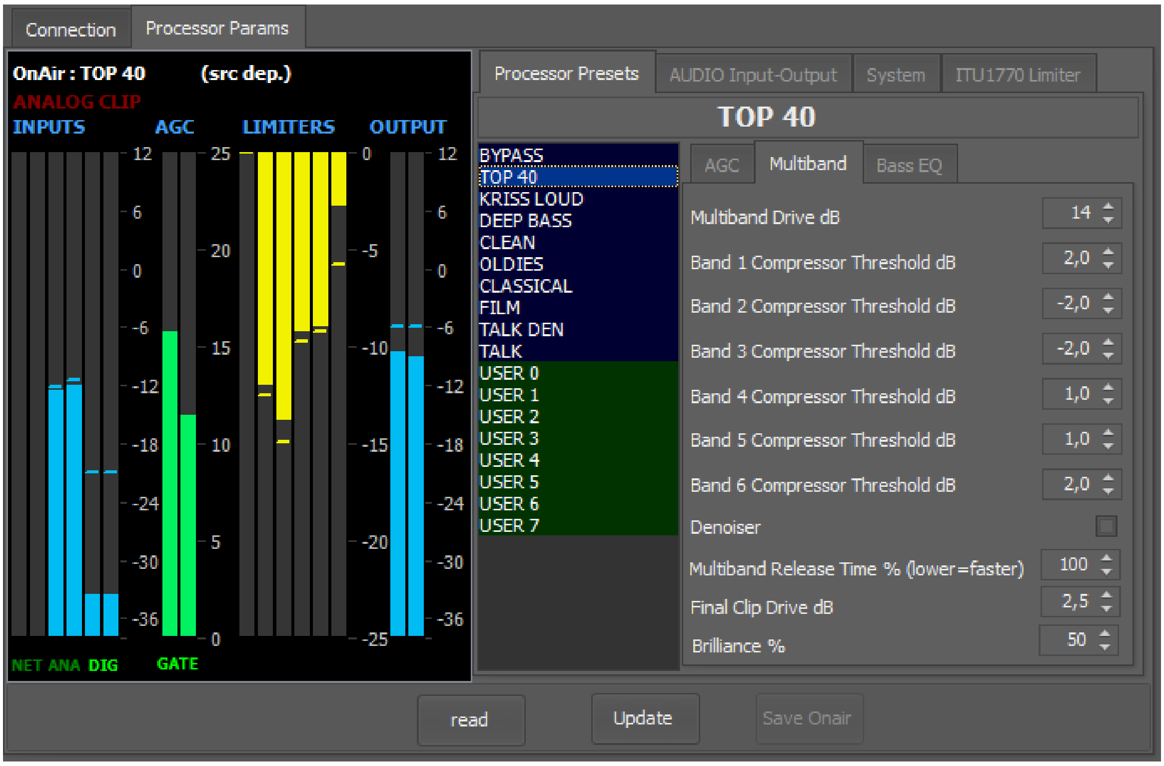


3.2 Processor Params

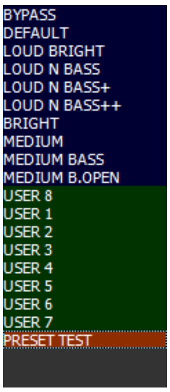
This section contains the audio process controls and presets, setting of the audio input and output sources and system parameters.

3.2.1 Processor Presets

On this page are shown the input levels, the audio input currently on air, the presets list and the audio processing parameters.



All presets are listed and, with a double click, is possible to recall a preset and put it on air. Factory presets are displayed in blue background, user definable presets in green and, in orange, a modified preset not yet saved.

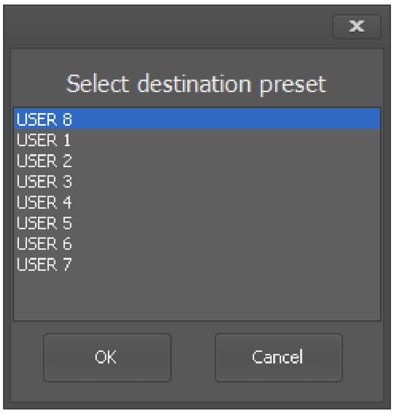


The modified preset (in orange) will be usable until a new preset is loaded and then modified.

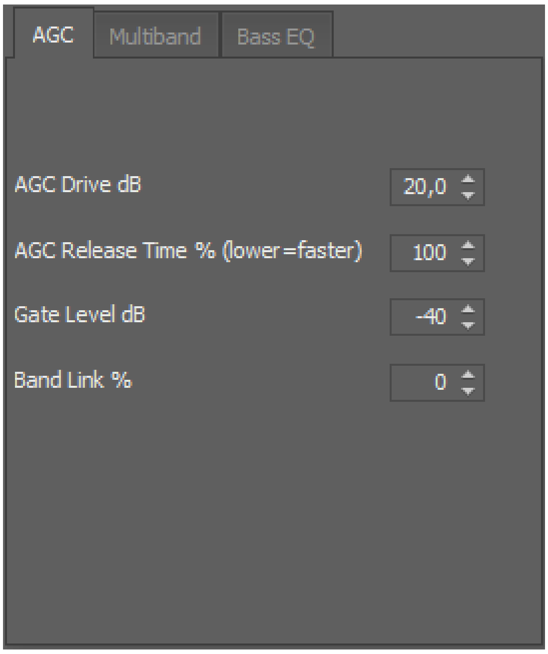
On the same page are present the audio processing parameters. For a detailed description of each parameter, please refer to the audio processor section:
Processor Drive, Gate Level
AGC Enabled, AGC Speed
Compression, Master Release
Band Link, Brillance, Presence
Bass Lim Thr, Bass Lim Presence, Bass Clip Threshold, Bass Clip Softness
Bass EQ, Bass EQ Q, Bass EQ Freq
Expander Level, Stereo Expander Enabled

Read: read the parameters setup from the processor
Update: send the modified audio processing parameters to the processor: when a factory or a user preset is modified, a temporary preset will be created. The **Save Onair** button will be enabled to save the preset once changes are terminated. The preset name can be directly changed in the preset name editor.

Then,pressing the **Save OnAir** button, it will be possible to select the destination user preset.



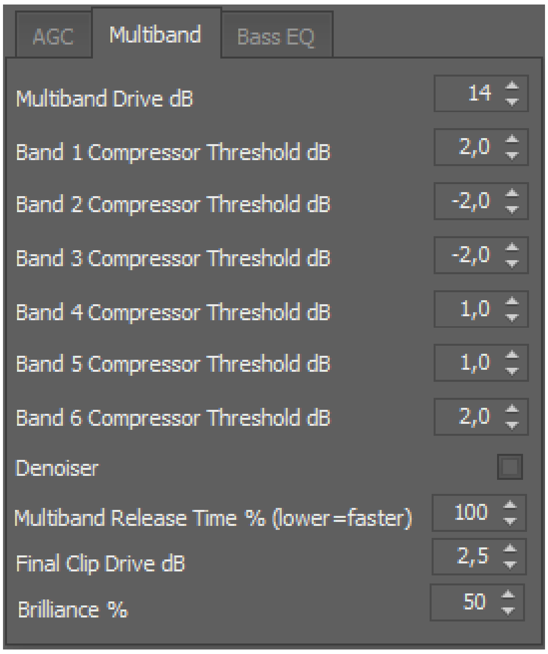
3.2.1.1 AGC



This window contains the settings for the dual band automatic gain control (AGC) block, placed at the head of audio processing chain.
The automatic gain control performs a pre-leveling of the signal applied to the following compressors/limiters, keeping their working point almost constant.
At the right side of the bar which indicates the level of pre-amplification applied to the input signal, is placed the AGC status indicator: if "gate" indication is off, the input level is above the gate threshold and the AGC and compressors/limiters are operating correctly, if "gate" indication is on, it means that the input signal is below the minimum level so the AGC release is slowed down and the AGC will slowly raise its gain.

AGC Drive: indicates the AGC input driving level, thus determining the AGC maximum gain .
AGC Release Time: AGC release time setting between 40% fastest and 200% slowest.
Gate Level dB: Gate level serring. Below this level, the AGC and multiband limiter release is slowed down, the signal below this threshold it is no longer considered processable.
Band Link %: set the maximum compression difference between the bass and middle/highs band, increasing this value the two band link together.
Setting this value to zero, the bass compression level will be totally independent and this will cause also a mild auto-equalization.
Setting this value to 100%, it will force the bass compressor to have a gain never greater than the middle/highs band giving a sound very close to the original.

3.2.1.2 Multiband



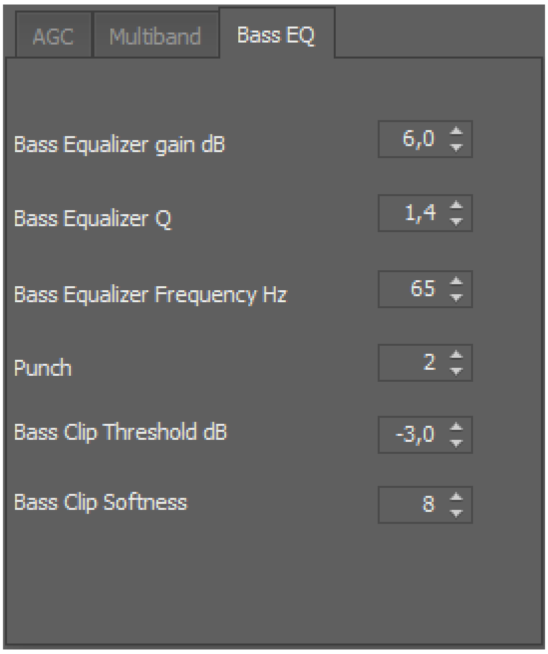
All six-bands limiters thresholds are adjustable in order to apply a dynamic equalization according to the user's taste. Lowering the limiting threshold of any band will result in attenuating the level of that band, while raising the threshold, will amplify its level.

Multiband Drive dB: indicates the multiband limiter driving level, thus determines how much the multibands will work and so re-equalize the program.

The six bands cutoff frequencies are :
Band 1 Compressor Threshold dB: 100Hz and below
Band 2 Compressor Threshold dB:: 400Hz center frequency
Band 3 Compressor Threshold dB:: 800Hz center frequency
Band 4 Compressor Threshold dB: 1600Hz center frequency
Band 5 Compressor Threshold dB: 3200Hz center frequency
Band 6 Compressor Threshold dB: 6400Hz and above

Denoise: Denoiser. If activated, it will monitor the high frequency content of the audio input signal, reducing the band 6 gain when HF input level is too low. It can be used to reduce noise coming from old tape recordings.
Multiband Release Time %: Multiband release time setting between 40% fastest and 200% slowest, making the multiband faster increases the loudness.
FinalClip Drive dB: set the wideband final clipper driving level. Higher driving level corresponds to higher loudness, but will cause higher distortion.
Brilliance: Brilliance high band (>6KHz) shelf equalizer, increasing the value produces an increase in the high frequencies.

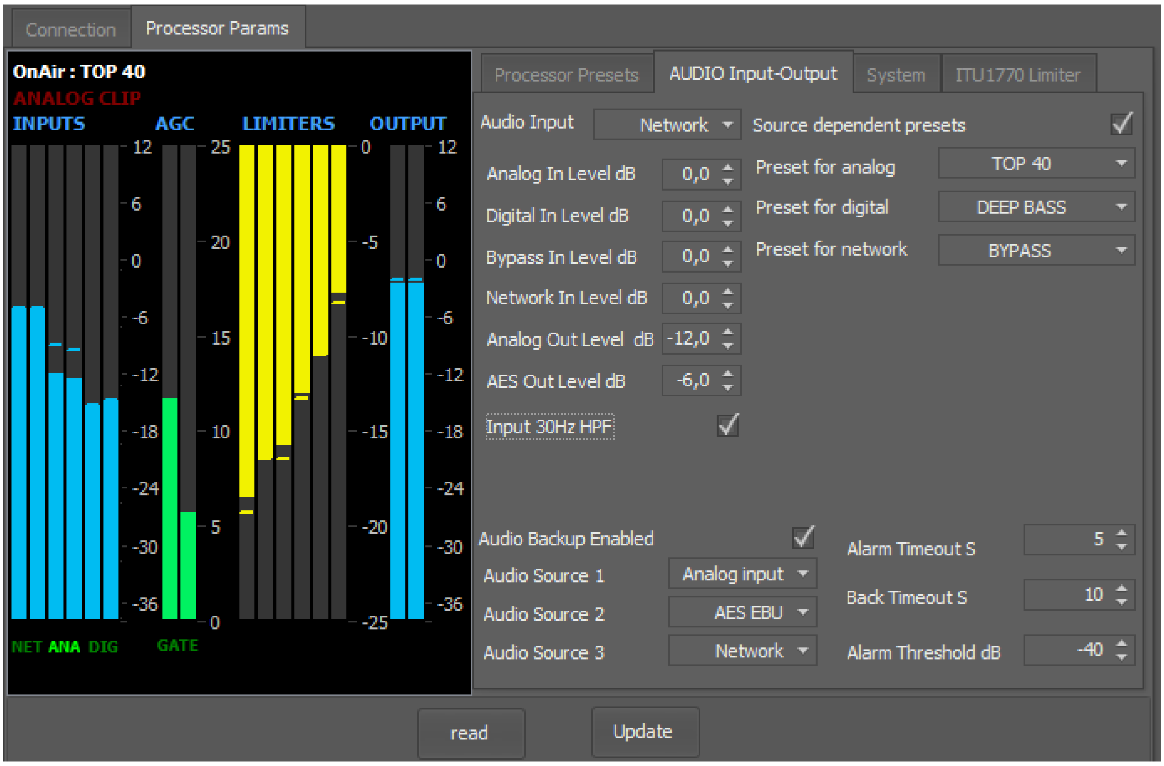
3.2.1.3 Bass EQ



Bass Equalizer gain dB: gain (boost) of parametric equalizer.
Bass Equalizer Q: Q-factor (bandwidth) of parametric equalizer, increasing this value the filter becomes more selective around the center frequency.
Bass Equalizer Frequnecy Hz: parametric equalizer center frequency
Punch: punch enhancer (0 off, 2 max). Give some nice "analog-style" bass enhancing.
Bass Clip Threshold dB: bass clipper threshold, the threshold beyond which the bass clipper begins to cut.
Bass Clip Softness: bass clipper curve selection.
Bass clipper will have a curve progressively softer with increasing value.
A softer curve reduces the distortion introduced by the clipping in contrast, setting to zero (corresponding to the hardest curve), will produce a bass rich in harmonics.

3.2.2 Audio input-output

This page allows for selection of on air audio input, changing audio output parameters, setting up the audio fallback and the source dependent presets.



For a detailed description of the following parameters, please refer to audio input setup, audio output setupand source dependent presets

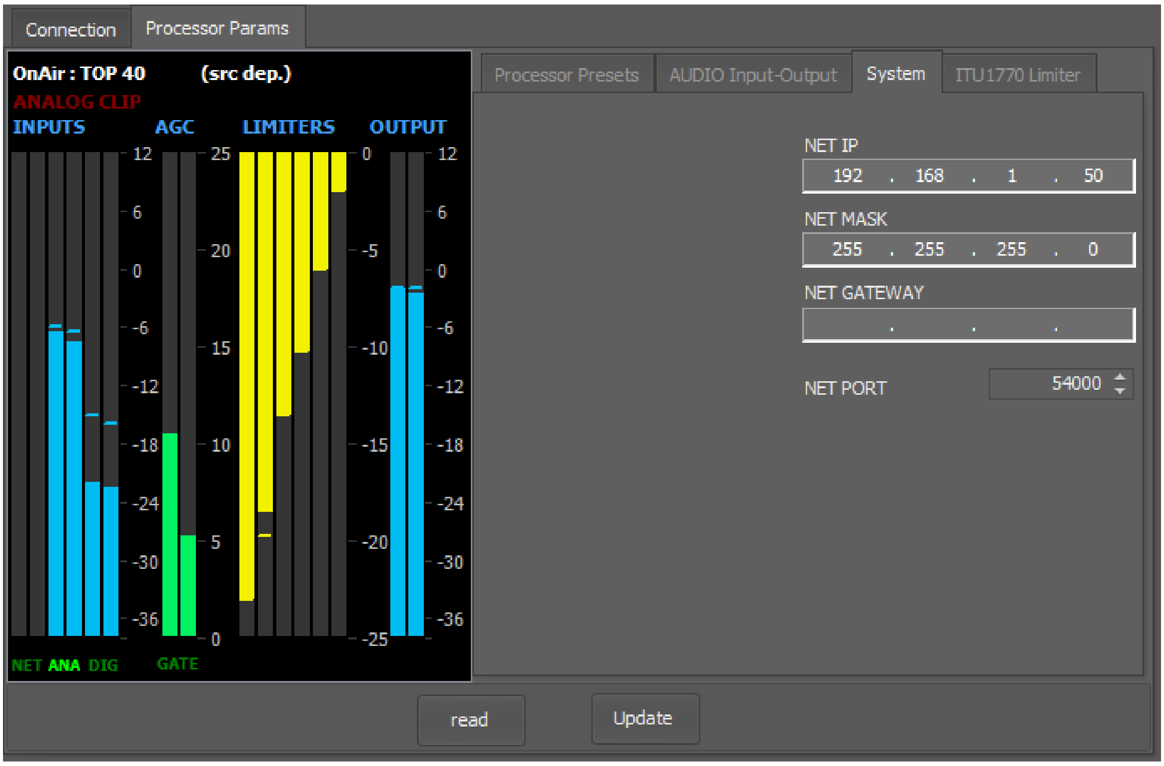
Audio Input, Analog InputLevel, Digital InputLevel, Network InputLevel
Bypass InputLevel
Audio OutputLevel
Audio Backup Enalbled
Source dependents presets

Input 30Hz HPF: enables / disables the 30Hz high pass filter for infrasonic component reduction.

Read: read parameters from the processor
Update: send changes to the processor

3.2.3 System setup

In this page it is possible to insert and adjust the network parameters.



The network parameters are related to the optional network module only. If the streaming module was installed, network configuration will be done through the web server of the module.

For a detailed description of the following parármeters, please refer to System setup

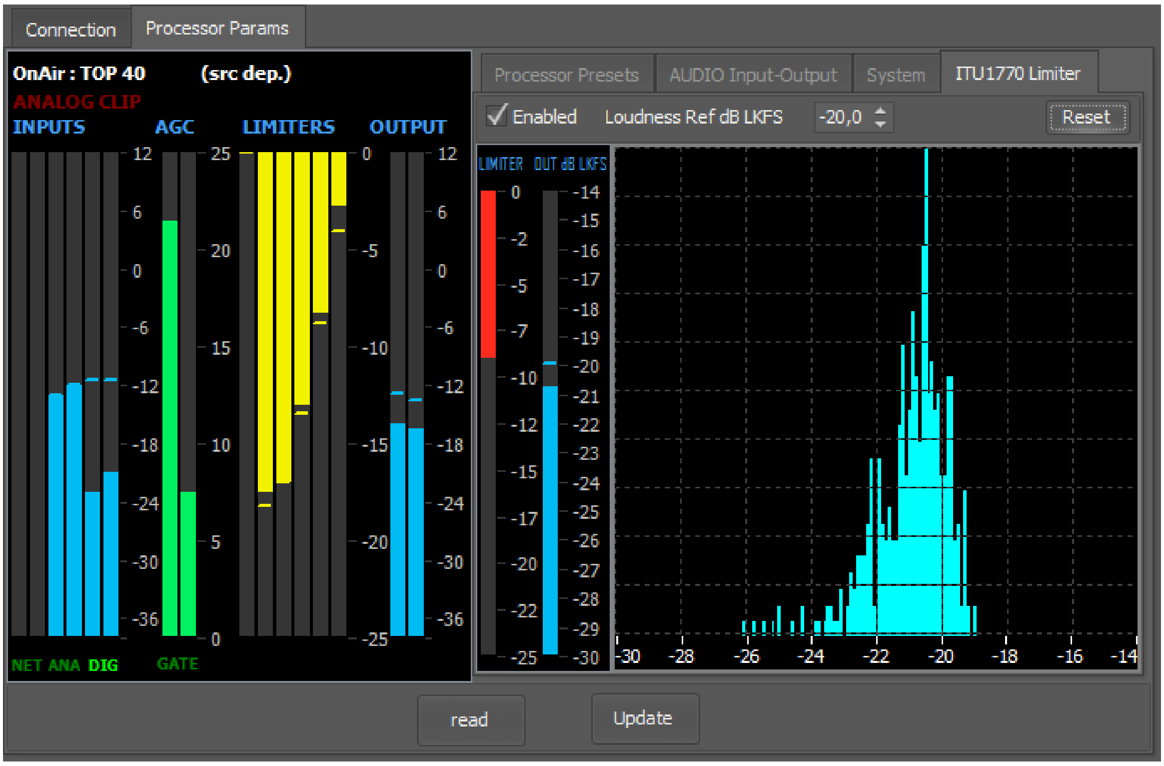
NET IP, NET MASK, NET GATEWAY, NET PORT

Read: read parameters from the processor
Update: send changes to the processor

3.2.4 Loudness limiter

In this window is possible to set the loudness limiting reference level, according to the ITU-R BS.1770 standard. The limiting level is shown on the red bar labeled "LIMITER". The "OUT dB LKFS" bar shows the current loudness level with a 400mS integration time.
On the histogram window is shown the loudness level integrated over time. Pressing the "Reset"button will start a new measure and clear all previous values.

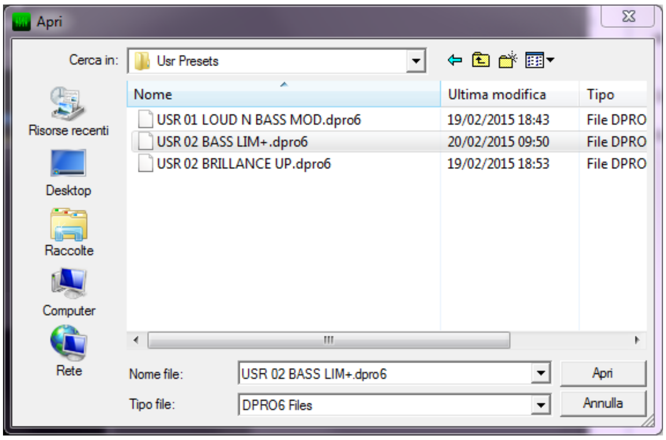
For a detailed description of the following parartmeters, please refer to Loudness limiter section.



3.3 Loading and saving a preset

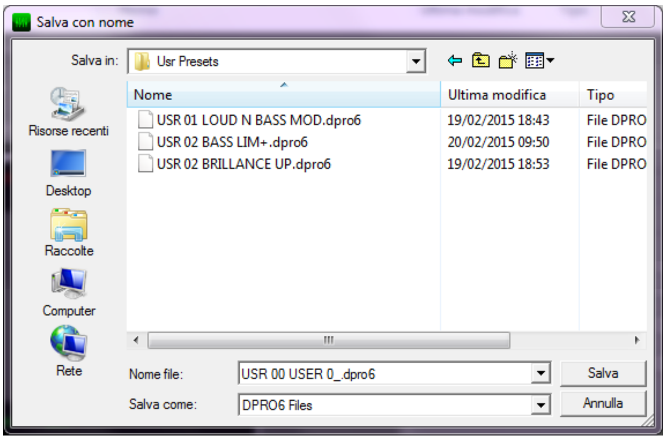
All the audio presets customized by the user can be saved in files, this preset can be transferred to another DPRO or saved in an external drive.
The remote control software allows to load and save the audio processing presets through the File>Processor menu:

Save Usr Presets to Disk



the on air preset will be saved to disk.

Load Usr Presets to Onair

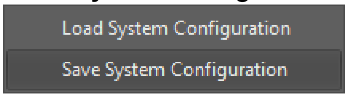


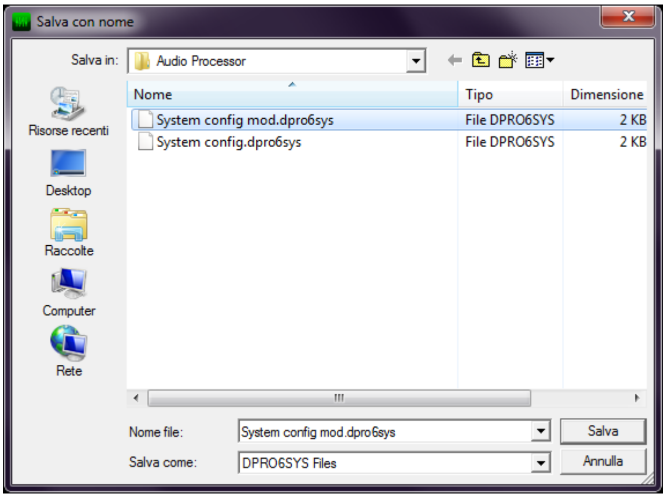
the preset loaded will be put on air.

3.4 Loading and saving system

The remote control software allows to load and save the system configuration through the "System" menu, and this includes all parameters listed under "System" and "Audio Input-Output" tabs.

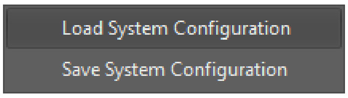
Save System config to Disk

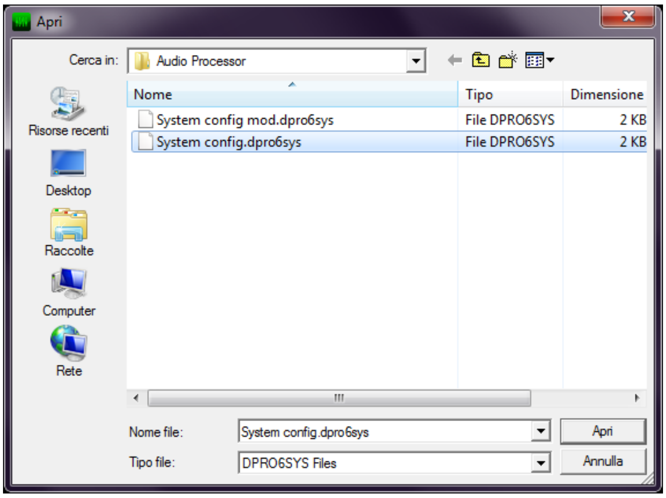




the system configurationi will be saved to disk.

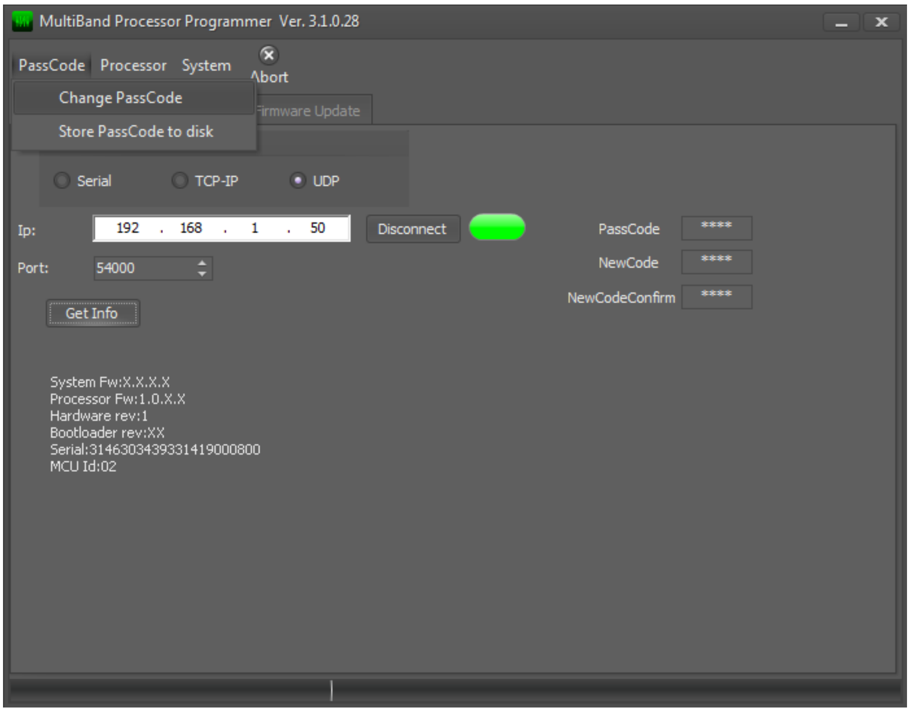
Load System config from Disk





the system config preset loaded in.

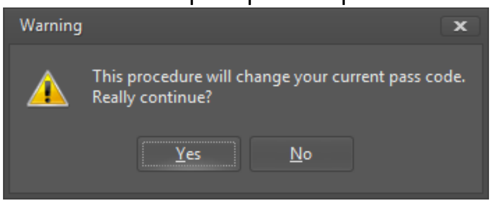
3.5 Passcode setup

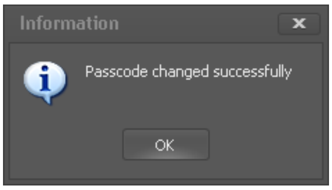


To protect the device from unauthorized operation, a security code can be set. The default code is 0000 (security disabled), but it can be changed through the following procedure:

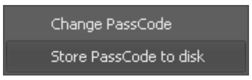
Put the current Passcode into the PassCode edit box (0000 by default).
Put a new code into the "New Code" and, the same again, into the "New Code Confirm" editor.
Go to Menu->PassCode->**Change Passcode**

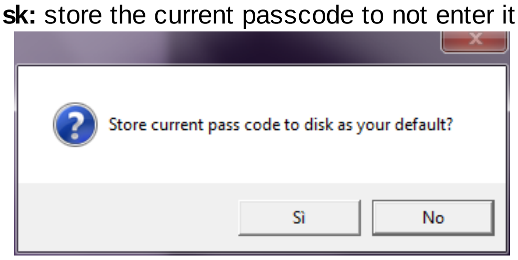
The user will be prompted for passcode change confirmation, then the passcode will be changed.





Store Password to disk: store the current passcode to not enter it each time the program is used.





Section 4

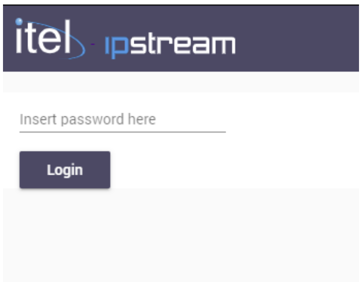
Streaming audio module

4 Streaming audio module

The module adds audio streaming decoding or encoding capabilities for receiving or trasmitting streaming HLS / MP3 / AAC + shoutcast servers by HTTP/HTTPS using TCP / IP connectivity.
The audio output is transferred in digital format to audio processor / stereo encoder, thus achieving the highest audio quality.

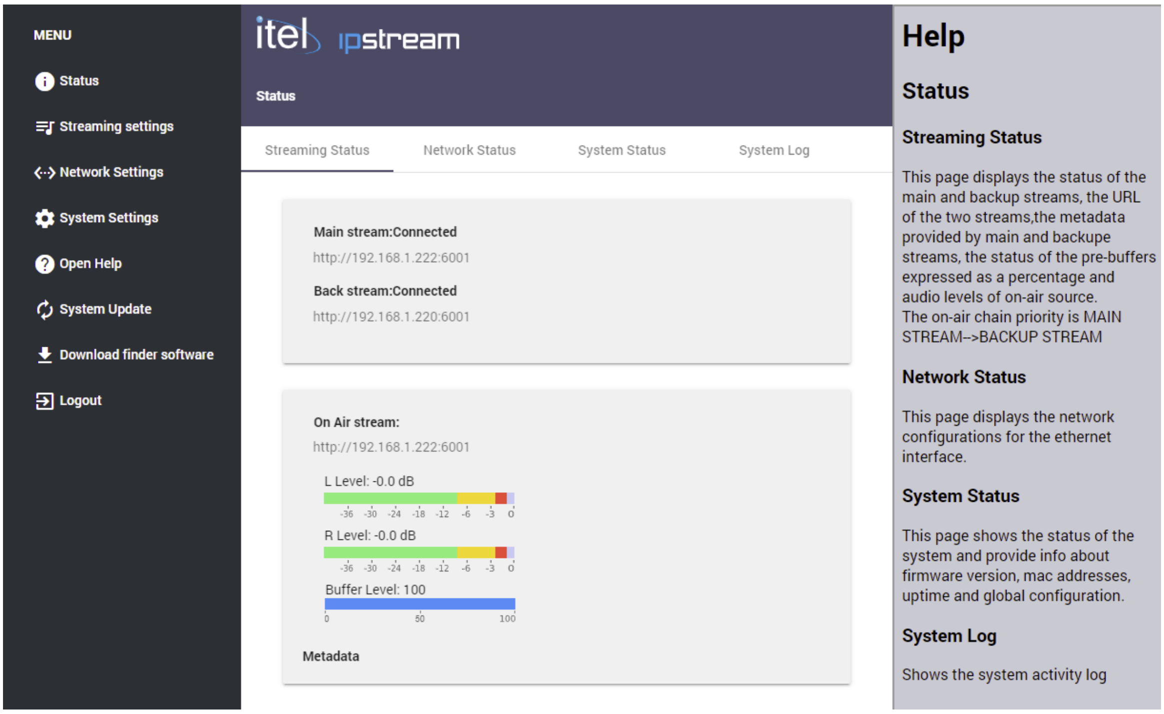
4.1 Decoder audio Stream

Connect the apparatus to your LAN and turn it on. Open a web browser on a PC from which the apparatus can be reached and type on the web browser's address bar: http://192.168.1.100.
The audio module internal webserver should then load the login default page.



No password set by default, click on the login button and enter the main page.

Go to the "Streaming settings" menu and set one or more URLs from which to receive the audio stream.
Save current configuration by pressing the "Apply settings" button.



The module adds audio streaming decoding capabilities for receiving streaming MP3 / AAC + / HLS shoutcast servers or by using TCP / IP connectivity.
The audio output is transferred in digital format to audio processor / stereo encoder, thus achieving the highest audio quality.
An helpful guide for all settings appears on the side of the configuration page.

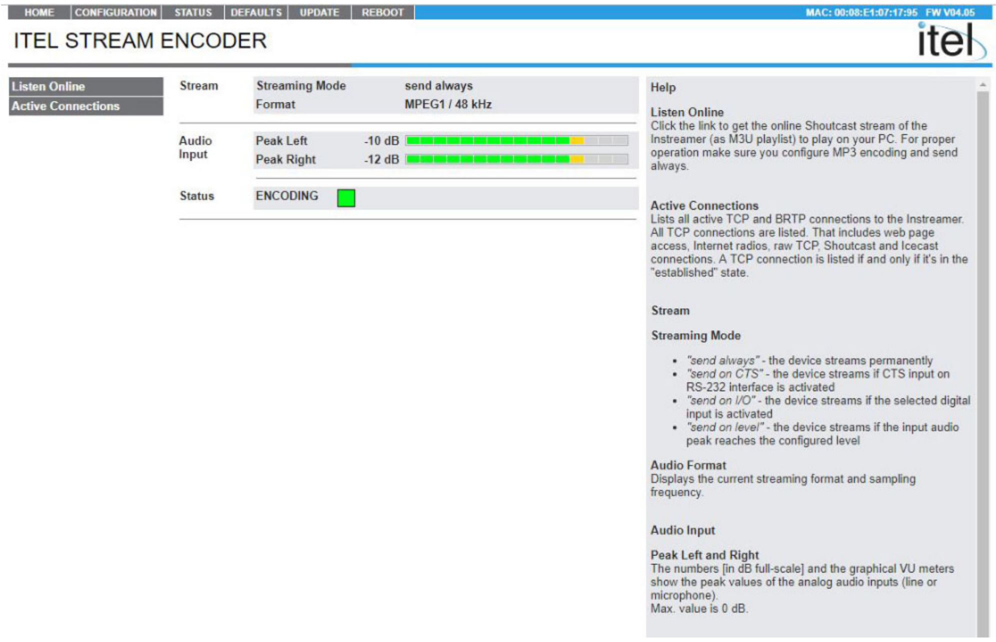
On the audio processor, now select the audio input "NetAudio" and the streaming audio received by the module should be put on-air.

Adjust, if necessary, the input level of audio input "netaudio." If you intend to use any audio processing during streaming audio playback, this setting can also be omitted since the processor will provide for automatic leveling.
The preset for the network input can be chosen into the next window, after "netaudio" input has been selected into the current.

If you want to use more than a single audio source, for using the integrated automatic audio fallback, proceed to its selection and its associated audio processing preset.

Finally, turn the audio fallback on.

4.2 Encoder audio Sream



The audio streaming encoder module has the task of generating a streaming stream in the IP network, various encoding formats allow the transmission of the audio generated by the processor over the network to shoutcast / icecast servers, even in low latency, using RTP, UDP, Multicast or TCP / IP connectivity.

AvailableCodec:
- MP3 fino a 320kbps, (VBR e CBR),
- PCM 16bit @8, 16, 22.05, 24, 32, 44.1, 48 kHz
- G.711, uLaw, aLaw

An helpful guide for all settings appears on the side of the configuration page.

4.3 Quick setup

The default IP address is **192.168.1.100**.
If the IP is unknown, connect the the processor directly to a PC network card, then use the supplied utility "module_finder.exe" to recover its IP address.

Connect the apparatus to your LAN and turn it on. Open a web browser on a PC from which the apparatus can be reached and type on the web browser's address bar: http://192.168.1.100.
The audio module internal webserver should then load the default page.
Go to the "CONFIG" menu and set one or more URLs from which to receive the audio stream.
Save current configuration by pressing the "SAVE & REBOOT" button, wait for the restart of the module.

Configure the network settings of the module: SETUP -> NETWORK
Save by pressing the "SAVE & REBOOT" button, wait for the restart of the module.