Audimax 362 HD

3 band audio processor
FM & HD audio

OWNER'S MANUAL

www.solidynepro.com

SOLIDYNE DIGITAL POWER

Ultima revisón

14 de Enero de 2016

What's in the box?

The Audimax 362HD packaging include:

- ✓ *1 Audimax 362HD Processor (include supports for rack montage).*
- ✓ *1 User's manual.*
- ✓ *1 Power cord.*
- ✓ *1 Guarantee agreement.*
- ✓ *4 self-adhesive rubber supports.*

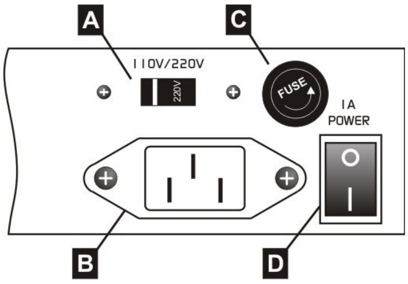
Generalities

Please read carefully the following recommendations:

- *Audimax 362HD processors are designed for rack montage in 19" standard racks. Also it can be used on a table or desktop, quitting the lateral supports.*
- *The room temperature must be between 5 and 40°C (41°/104° F). Avoid the direct solar ray incidence on the equipment. Avoid the proximity of heat sources and high electromagnetic fields (high power transformers, motors, etc). 462-MKII have internal protection against RF fields, that allows assembly it next to AM/FM RF amplifiers.*
- *The installation in very humid places or with saline atmosphere will have to be avoided, since they will can cause corrosion in the printed circuit board and electronic component.*

Chapter 1 – Installation

1.1 Power source



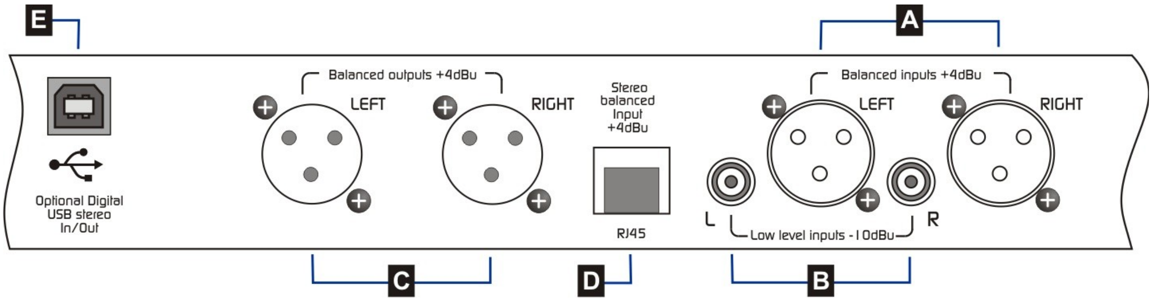
A – AC voltage. Verify the correct position of the AC120/220 voltage switch, located on the rear panel. The variation of the AC voltage must be smaller than 10%. Otherwise, use tension stabilizers of fast action.

B – Interlok

C – General fuse

D – The on/off switch.

1.2 Rear panel - wiring



Inputs and outputs are **electronically balanced**. Use shielded twisted pair audio cables for audio connections (microphone cable). Input and output connectors are XLR. Connect as the following:

1	GND
2	Signal (+)
3	Signal (-)

A XLR inputs Nominal level + 4 dBu. Inputs works in "bridging" mode, with impedance bigger than 10 KOhms.

B Unbalanced inputs using Phone connectors. Designed to work with signals of -10 dBV.

C XLR balanced outputs.

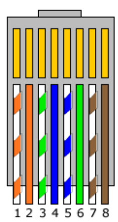
1.2.1 RJ-45

D Balanced inputs are available on an RJ45 connector.

Today many broadcasting mixers gives inputs and outputs on RJ45 for the analogical audio signals; and are wired using CAT-5 shielded twisted pair (STP). With the arriving of audio over IP (AoIP) several companies start to use RJ-45 and multipair shielded cable to replace to the different traditional audio connectors, in order to standardize all in one type of connector and cable.

In fixed installations, like radio stations, the use of connectors RJ45 presents several advantages respect to the traditional connectors: a unique cable is able to send both channels, reducing the quantity of wires nee/ded. Additionally, installation is easy and the tools needed are available in any city, due are the same used for data networks. And you don't need to solder anymore!

There is no standard to connect an RJ45 with audio signals; but Audimax 362HD are compatible with the accessories manufactured by StudioHub (USA), a brand widely used in radio stations. The following table shows the pinout. of RJ45 inputs.

RJ45	PIN	COLOR
	1 Left (+)	Orange & White
	2 Left (-)	Orange
	3 Right (+)	Green & White
	4 GND	Blue
	5 Reserved	Blue & White
	6 Rigth (-)	Green
	7 -15 (optional)	Brown & White
	8 +15 (optional)	Brown

At the end of the STP cable, the connection to the audio device (microphones, speakers, audio players) needs standard audio connectors. The RJ45 wiring offers short end cable with female RJ45 to any standard audio connector. Please refer to web site for more info.

1.2.2 USB input/output

E Audimax 362 HD / USB has an optional module that enables receiving and sending audio connected to a computer with a USB A/B standard cable.

Connect the unit to any USB port of a computer running Windows©. When connecting, Windows© recognizes the USB device and install the drivers. Additional drivers are not required. Audimax 362HD will appear on Windows like one stereo USB audio device and one stereo USB recording device.

To check the available sound devices, go to *"Windows Control Panel > Sound devices > Audio".* Here the default sound devices are defined. Remember to update the settings on the audio applications.

ABOUT USB DETECTION

Before connecting the USB cable to the computer, make sure that both the console and the computer have an effective ground by power cords. For safety, connect a tester in the range of 25VAC between the chassis of the PC and console and verify that the voltage is zero. Only then connect the USB. If there are differences of tension, the USB port on the console or the computer may result damaged.

We recommend do not change the USB cable to other USB ports, to avoid that Windows change the order of USB devices.

Windows 7 / 8.1,: Check that the audio recording device was properly recognized. If Windows 7 recognized it as "microphone device", the recordings will be mono (the same signal in both channels). To correct this: Control Panel → Sound → Record → choose the USB device (shown as USB microphone) and click [Properties]. Then select the tab "Advanced", display the menu of formatting options and choose a format for stereo recording (2 channels, 16 bits, 44100Hz).

1.3 Modo HD (High Definition)

HD mode is suitable for audio applications where the stage FM stereo coding (MPX) is not used. It is an important contribution to the technology as audio processors can be used in digital radio (HD Radio) or conventional radios inside the Recording Studio to record voices. It is also suitable for small AM radio stations.

The main feature of "HD mode" is that the pre-emphasis (75/50 microseconds) is disabled, achieving a brighter treble and rich sound. This enables the processor to be used in sound reinforcement installations (small rooms, auditoriums) in popular music concerts, discos, sound systems in malls, etc.

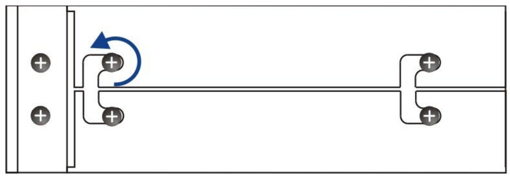
In HD mode, the processor connects using the audio outputs (balanced or USB).

When the unit is set to HD mode:

- *Pre-emphasis for FM is disabled (and therefore de-emphasis on audio outputs is disabled).*
- *A fast limiter is enabled over the audio outputs.*
- *The MPX output turns off.*

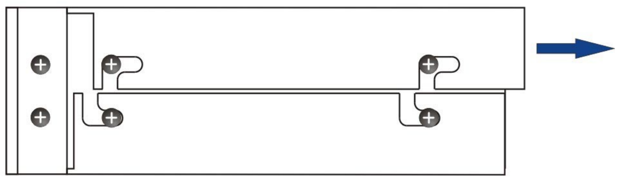
To change the working mode, you need to access to internal micro-switches. For this, remove the top cover following the indications:

1



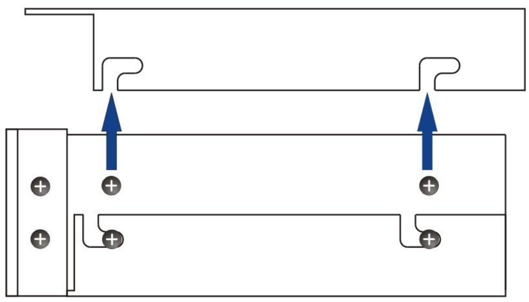
Slacken the screws on the top cover. No need to remove them.

2

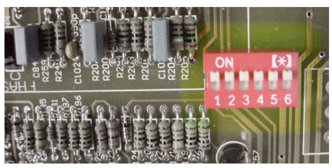


Pull the cover backwards.

3



Lift to remove the cover.



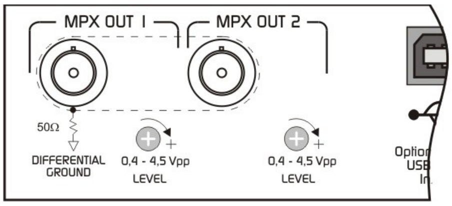
All switches 'ON' = FM Mode

All switches 'OFF' = HD Mode

The FM/HD mode is indicated by the LED "FM" (ON = FM, OFF = HD). By default the unit is set for 'FM Mode'.

1.4 MPX Output

Audimax 362HD has dual MPX output, with independent level control. The secondary output allows to connect a backup transmitter. Or a second transmitter to nocturnal emission.



The connectors are a BNC. For MPX output use 75 ohms coaxial cable (standard in CCTV). The maximum recommended length is 25 meters.

When use the MPX input of the transmitter, make sure that the internal pre-emphasis stage is disabled (flat response between 20 – 100 KHz).

Is important to have an adequate ground installation. In case of doubt, refers to your Solidyne dealer.

1.4.1 Humming

If some residual humming appears when the system is started up; turn off the unit. If humming disappears, check the 462 input connections. If the humming still here; unplug the MPX cable. If humming ends, this indicates that a problem occurs with the ground connections.

Nevertheless, this problem rarely can occur since the Audimax MPX output is differential type, where the BNC connector has floating GND to cancel possible humming.

1.4.2 Modulation level

The modulation depth is adjusted by changing the MPX output level. It is recommended to use a program material with voice and loud music (contemporary pop for instance).

- The level of audio on the console must peak at **0 VU**, in order to the AGC stage works in its point of equilibrium.
- Move all knobs to the center position.
- If the transmitter has modulation level, move to the minimun..
- Change the output level of 362's MPX until reach 75KHz of deviation (100%), measurred at an FM Modulation Meter (like Solidyne VA16) o using the instrument of the tranmiter.

Some countries allows non-recurrent peaks at 110% (I.e. FCC on USA).

n daily use processor probably modulation indicators show needle-highest peaks at 100%. This may be due to overshoot or ballistic responsive average values of sine wave and the indication is wrong with very processed audio material.

After adjusting the modulation level, you can proceed to customize the sound of the radio; adjusting 362HD Audimax controls. Obviously this does not affect modulation adjustment, because the processor delivers a constant level signal peak. To set the personality of sound in air, proceed as described in "2.2 - Audio settings".

1.5 ADITIONAL TECHNICAL INFORMATION Balanced and unbalanced I/O

In the following paragraphs you will find information about how to connect unbalanced outs to balanced inputs, and viceversa. This information will be of extreme utility, not only for Audimax 362HD, but also for other equipment of your radio. Different cases are decrypted.

1.5.1 Balancead lines

Balanced lines use three cables to transmit the signal: positive, negative and ground. The negative takes the same audio signal of that the positive but inverted 180°. GND corresponds to the shield of the cable, which rejects the electrostatic noise (switches, big motors, etc.). The fact that the audio signal is sent by two cables, inverted in one of them, practically annuls the noise induced on the cable, specially the electromagnetic one, caused by fluorescent tubes, AC lines, etc.

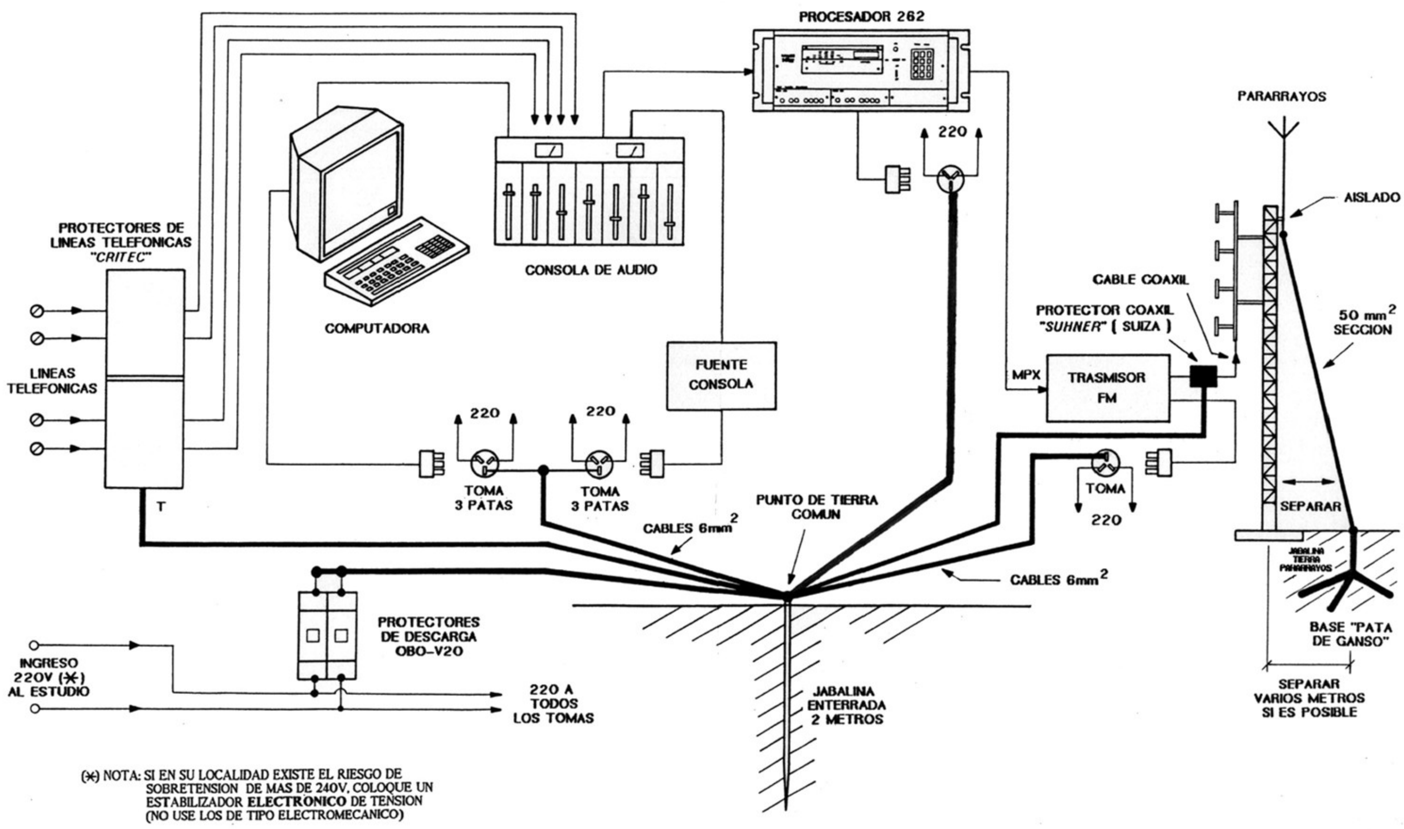
A balanced input amplifies only the **difference** between both positive and negative signal. Since the induced noise causes the same deviation in both cables, when reducing both signals the noise is eliminated (the (-) wire invert its phase, then, the induced noise is reduced to itself). In addition, equipments that uses balanced I/O are of professional category and uses line levels of +4dBu (1,23 Volts). Connectors are XLR or stereo plugs (one by channel).

1.5.2 Unbalanced lines

They are used in “home” and non-professional equipment. The audio is sent by a pair of cables; being the one alive and other the shield of the cable. These cables are much more **sensible to the noise**, mainly of electromagnetic type. They handle line levels of the order of -10dBv (0.36 Volts).

The connectors typically used on this kind of units are RCA o male plugs (TRS).

1.5.3 Diagrama de conexión a tierra recomendado



1.6 Work settings

1.6.1 Processor as FM stereo coder

It's normal use as audio processor for FM. Audimax 362HD is located at the transmission plant, connected as usual via MPX transmitter. The second MPX can be used to connect a backup transmitter.

The program signal input for analog inputs. Use the appropriate input (balanced or unbalanced) according to the characteristics of the console.

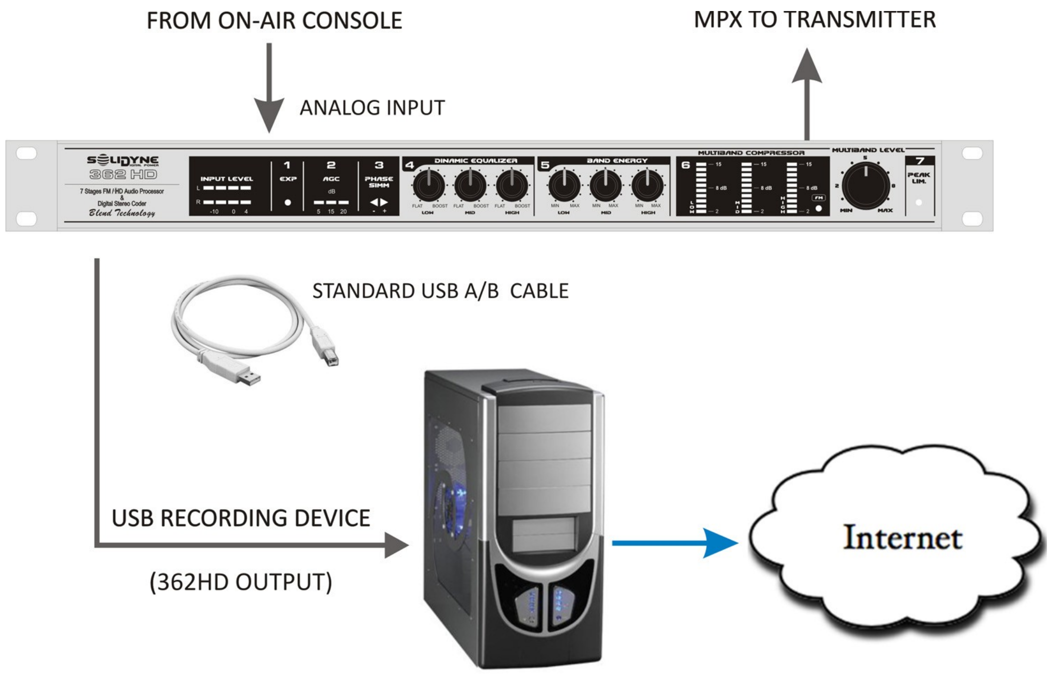
When the transmission station is far from the studio, the processor is located at the transmitter site and receives PGM audio via a point-to-point, which can be: RF audio; digital audio via optical fiber or UHF (see the link Solidyne ADA102 STL).

1.6.1.1 USB for external streaming

Requires Solidyne Audimax 362 HD / USB. Using USB is independent from MPX. This makes it possible to use the same audio processing on-the-air and on the webcast. The processor receives the program signal by the analog inputs, and outputs the processed signal to USB and MPX. A computer receives the audio processing via USB, for streaming.

MPX is connected, as usual, to the FM transmitter; while the USB is used to streaming, using a computer, the processed audio.

It is recommended to install in the computer that manages the streaming any remote access software.



Standard connection for FM with USB for streaming

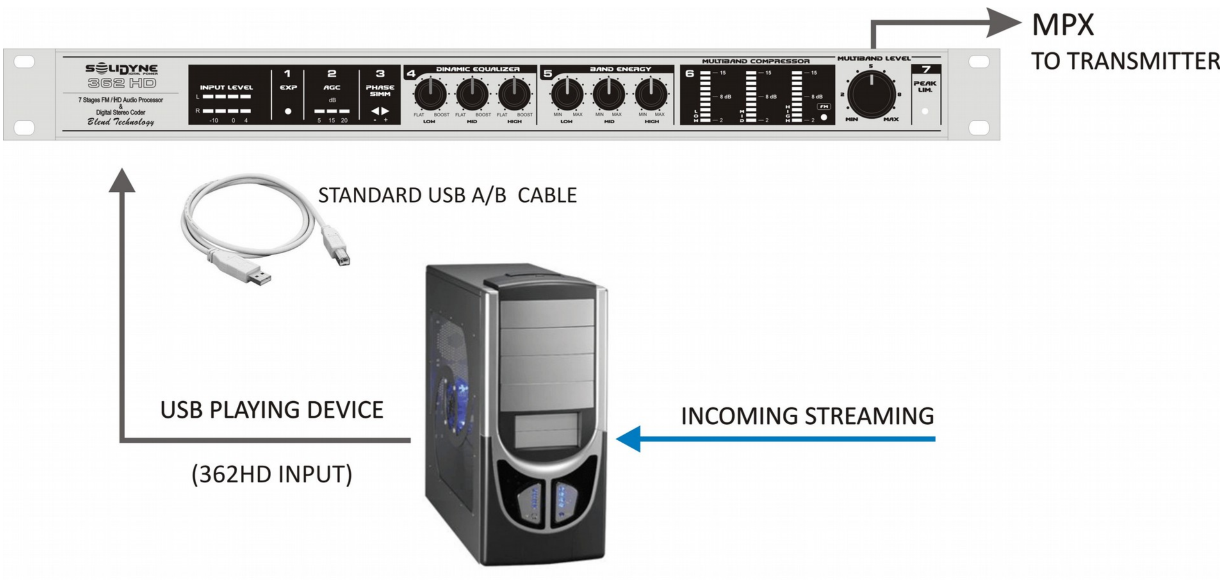
1.6.2 Retransmission of *incoming streaming*

Requires Audimax 362 HD/USB and an external computer. It is used, for example, to signal repeater stations.

The external computer receives streaming audio from the main station (head station), either via the Internet or LAN. Decodes it and sends it as PCM signal to the Audimax 362HD/USB processor through the USB connection. The audio is processed and sent through the MPX outputs to the FM transmitter.

The following diagram shows the signal flow for this configuration.

We recommend installing any software on the remote computer access to streaming encoder monitoring.



retransmission of a *streaming*

1.6.3 As audio processor

In his mode HD (High Definition) HD 362 Solidyne Audimax can be used in applications requiring narrow the dynamic range of the signal, or an increase in the sound; and do not involve transmission of FM (MPX is disabled; see "1.3 HD mode").

Examples:

- Radio stations with digital transmission (HD Radio).
- *Hi Fi Streaming*

  Here all Audimax 362 HD features become essential. The peak control takes advantage of the digital range, it is possible to adjust the level of the streaming audio with a minimum safety margin (2 to 3 dB, required to prevent digital clipping in overshoot) which enables transmission over the Internet with high loudness levels.

- Pre-processing of audio in mobil units**.**
- **Sound reinforcement** for small shows; rehearsal hall.

  When the Audimax operates in HD mode, a soft clipper on audio outputs is coupled; to limit peaks that cross the multiband compressor. This ensures that no overshoot they reach the power amplifier stages.

- Rooms, auditoriums, conference rooms, background music.

  The combination of Automatic Gain Control (AGC) and multi-band dynamic compression, allow to achieve consistent and clear sound; with uniform loudness at all times, regardless of the characteristics of the input signal.

Chapter 2
Using the processor

2.1 MPX level

The first adjustment is the modulation level on the transmitter (please see “1.4.2 – Modulation level”).

2.2 Audio settings

Audimax 362HD was designed to offers an easy and very intuitive operation. You don’t need to have specialized knowledge to start up the processor and to adjust the sound settings. Simply begins placing all knobs to the center position. That is all! You will be on-air with a great sound. Soon you will have to customise the sound according to the musical style of your radio, for which we recommends you to read the following explanations kindly.

2.2.1 Input level

Audimax 362HD has an automated control for the input level. The rear panel presents balanced inputs (+4dBu) and unbalanced inputs (-10 dBV). Make sure to connect the mixing console to the appropriate inputs according its nominal output level.

Automatic input gain control (AGC) eliminates variations in characteristics of the operation of the mixing console level, and compensates for differences in level of the recorded material. That is, if the level from the console remains very low for some time, the processor will compensate its input to maintain a constant output level. If the output level of the console exceeds constantly 0VU; AGC attenuate the signal. In this way the program signal reaches the processing steps with constant level and sound on-the-air is consistent, always with the same degree of processing.

> To verify that the input level is appropriate, observing the three LEDs on the AGC, located on the front panel operate properly: The first LED (green) should be always on in the presence of music or voice; the second (yellow) flashing audio peaks or always on. The third LED (red) should never be turned on permanently.

2.2.2 Customizing the sound

All adjustments are made from seven knobs located in the front panel.

The stage **Dynamic Equalizer** is a 3-bands dynamic audio equalizer that allows reinforcing certain frequencies, usually the bass, to have a sound with ‘punch’, ideal for the car-stereos. Unlike the conventional EQ’s, whose action is lost for high levels of modulation, this EQ emphasizes its action when higher is the modulation.

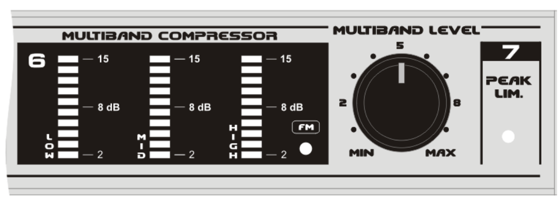
To adjust the dynamic EQ, make sure that the multiband compressor is operating. For this, see the LED’s meters located at the right; the first LED’s must be illuminated.

In these conditions, turns the Dynamic Equalizer knobs (Low, Mid and Hi). Turning the knobs to the left, the response of the dynamic EQ will be flat. Turning the knobs to the right, a boost in this band takes place. When you boost a band, the action of the compressor for this band increases.

NOTE: Dynamic Equalizer make changes over the attack times of mutiband compressor.

The **Band Energy** section increases the peaks density for the band, achieving high loudness signals. This control changes the recovery times of the audio compressors. It must be settled according to the musical style and the kind of sound that you’re looking on-the-air: compact with high loudness and “punch”, turning the knobs to the right (fast recovery times); or more soft and smoothly, with the knobs at the left (slow recovery times).

Multiband Compressor is the core of the processing system, constituted by 3 independent audio compressors that works on 3 frequency bands: Low, Mid and Highs.



The Multiband control **changes the input gain of the compressors**. This control affects more radically the output level of the Audimax 362HD. Turning this knob to the left you will obtain a smoother sound, with little processing and therefore with less energy. Turning it to the right, you will increase the processing and the energy of the sound.

Take in mind that with an excessive processing you will obtain a very hard sound on the air, with high energy, but too compressed (smaller dynamic range) and with less definition (more clipping).

Normally a suitable level is obtained when the indicators LED's of the multiband compressor act without light the red LED.

> ***Very important note:*** *In order to the Dynamic EQ and Band Energy take effect, is necessary that the multiband compressor works. That is to say, in all bands the first LED must be lights, and still better, the second too. If the compressors don't works, the others controls don't take effect on the sound, since they are linked to the multiband compressor.*

Chapter 3
Theory of audio processors

***NOTE:** in order to complement the study of this subject is recommended to visit our WEB (www.solidynepro.com). In the DEMO section there are a Power Point presentation called Audio Processors. It has a complete Technical Appendix that it analyzes how the audio processing increases the coverage area of the FM stereo transmission.*

3.1 A brief story...

From mid of the 1930 decade, when appears the first compressors and expanders units, to the present time, all chains of audio for broadcasting incorporate devices whose function is to alter the dynamic range of the sound. The advance of the technology improves these devices during the '70s.

The compressors, expanders and audio limiters were gaining in efficiency and complexity. In the beginning, its main parameters (attack and recovery times, thresholds, etc.) were fixed by design or by the operator, through the device's controls. In the '70s, these functions begin to be automatic, based on the characteristics of the audio signal, but having at the same time a control on their action to be able to customize the sound.

When five or more devices are grouped in a same equipment, they begin to be denominated: AUDIO PROCESSORS.

Since 1970, **Solidyne** introduces important advances in this field, like the invention of a control technique based on FET's with guided gate (see publication in Rev. Tel. Electrónica, September/70). They follow diverse publications, having particular international relevance the work published in June/76 at the *Journal of the Audio Engineering Society*, New York, U.S.A. where a new concept was introduced, which persist to the present time: **PHSICOACUSTIC PROCESSING**.

This new technique is the base for all the modern audio processors for broadcasting use. The necessity to process the phase to make symmetrical the human voice waveform is another one of the techniques that Solidyne has introduced internationally (see mentioned article AES). Today, Orban, Omnia, Aphex, etc use our ideas.

The concept of psychoacoustic processing is simple in essence, although of complex accomplishment. It consists of analyzing the way in which the sound is perceived by our ear, considering diverse investigations and developed acoustic models.

The brain uses to process audio data, the information that arrives through 30,000 nervous fibers, originating at the Basilar membrane. Then, it will be possible to be computed the auditory reactions and to be governed all the aspects of the audio processing. This way, the electronic system works transforming the original signal into another one, of greater energy and greater quality of sound. Then, it will be possible to reduce the dynamic range of the audio signals, to eliminate the peaks, and even, to clip them partially to increase its energy.

If this were made directly, obeying to purely electronic concepts of efficiency, the quality would be degraded and the sound would be very poor. If, however, the psycho acoustic concepts are applied, and factors like the aural masking, the pre and post pulse inhibitions, the Hass effect, the reflections at the ear pinna, the aural models of Dr Karjalainen, etc; it will be possible to create a new generation of processors that allow to important increases of energy, increasing at the same time the sensation of "Perceived Sound Quality".

At the light of these discoveries the psychoacoustics processing was defined in these terms:

PHSYCO ACOÚSTIC PROCESSING is the technique that allows to increase the range of AM or stereo FM transmission, by increasing the energy of the audio signal, and also increasing the "quality of sound" perceived by the listener.

Nevertheless, it is fundamental throughout this process, to maintain very low the audio distortion produced by harmonic and IM components. This happens because the psychoacoustic processing MODIFIES the waveform of the complex signal of audio, but IT DOES NOT DISTORT IT. Since the distortion concept, in this context, implies the existence of a sound that offends the ear, sounding unnatural.

This is because the psychoacoustic processing obtains that the ear accepts like of better quality than the original one, to certain modifications of the waveform. But it does not mean an "anesthesia" to the ear, to avoid perceiving the distortions due to deficiencies in the quality of the electronic circuits of the processors.

Considering that to obtain an excellent processing is necessary, at the moment, to use between 7 and 10 stages of processors, the distortion of each stage must be smaller than 0.01%. Greater distortion values, will lead inexorably to a degradation of the sound quality. You must remember that has been demonstrated (Journal of AES, Vol. 29,4,p.243), that is possible to measure distortions of 0.05% through a common loudspeaker (distortion bigger than 3%). This demonstrates that one distortion do not mask another one. A practical rule is, then:

ALL DISTORTION INTRODUCED IN THE AUDIO CHAIN OF THE TRANSMITTER, THAT EXCEEDS 0.05%, COULD BE LISTENED BY THE AUDIENCE, EVEN THROUGH RECEIVERS THAT HAVE VALUES OF DISTORSION 50 TIMES GREATER.

This, of course, is not a novelty for the conscious audio engineers around the World.

Therefore, the line of SOLIDYNE processors has distortion values smaller than 0,02%.

Few controls, easy adjustment...

The psychoacoustic processors created by Solidyne, have 70% of their functions automatically fit, under the control of the audio program. But they also present the essential controls for customize the sound of the radio, that you can adjust.

Audimax 362HD has a great advantage: it does not have critical adjustments. This means that in any position of his controls always it sounds well. The adjustment can be made then by inexpert people. Simply taking all the controls to the center position, you have an excellent sound on the air. From there customize the sound so that the radio sounds as you desired (this it is a question of personal taste).

3.2 Audimax 362HDHD overview

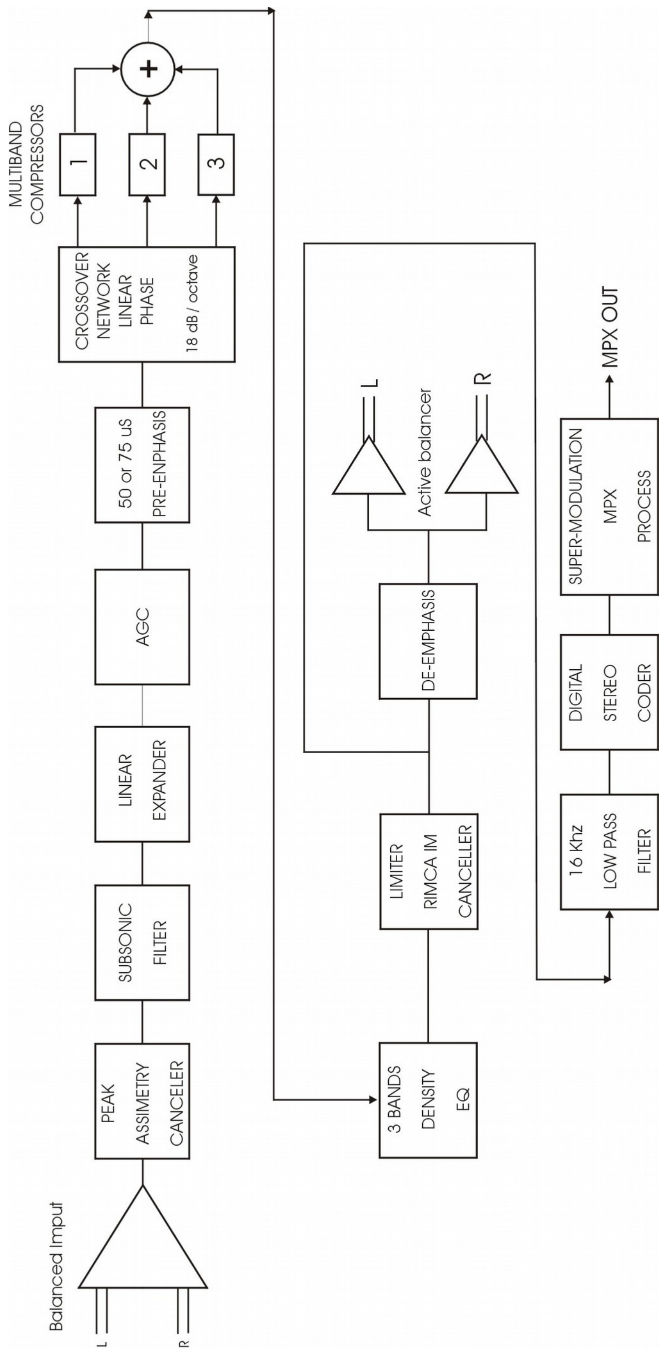
3.2.1 Introduction

The Audimax 362HD has **7 processing stages** and the **stereo coder**. Its main characteristic is the ease of use, because it doesn't require a specialized technician to start up and to adjust the unit; and neither it has the critical "Input level" control, since an automatic system adapts the input gain to the output level of any audio console, avoiding the more frequent operational errors. His low cost is appropriate for low-power FM radios, as well for recording studio of high power radio stations. The **362-IT** model is an excellent processor for **WEBcasting** (radio on Internet).

Operating in a FM radio, the Audimax 362HD **increases the reach of the transmission**, improving the covered area between 30 and 50%, obtaining an impressive audio quality that will distinguish your radio station. The AudiMax sound is *smooth* and *warmth,* with the classic characteristics of the analogical processes of high technology.

Audimax 362HD works with **3 bands** and it's fully controlled by **VCA** (Voltage Controlled Amplifiers). The stereo coder stage uses digital synthesis with 16X oversampling, a technology developed by Solidyne that guarantees ultra-low distortion and high channel's separation, not requiring any readjustment during all its life utility. MPX output is differential type, cancelling the residual humming.

3.2.2 Blocks Diagram



3.2.3 PROCESSING STAGES

Stage 1: Peak Asymmetry Canceller

It is known that, by a particularity disposition of the vocal cords, the sonorous emission that these generate are asymmetric triangular pulses. The three cavities that filter and shape these formants, to obtain the vocal sounds, do not modify this intrinsic characteristic of the human voice. All the spoken word and still sung is strongly asymmetric.

This creates an important reduction of the energy of the audio signal, particularly when it pass through a compressor, because the compressor sets its compression level for the greater peak, does not concern its polarity. In this way, when a polarity is fit to the 100%, the opposite polarity hardly surpasses 50%, due to the asymmetry. The fact that the music sounds louder than the human voice, after pass through a compressor is a phenomenon well known. This is because the musical sounds are symmetrical, whereas the human voice is not.

In order to correct this problem, WITHOUT INTRODUCING ANY ALTERATION AT THE SOUND QUALITY, peak asymmetry canceller is used.

This technique, based in a discovery of the Dr Leonard Kahn, acquires international validity with the work of Oscar Bonello, published at the Journal of AES, Vol.24,5 in which it is described, for the first time, the theory of its operation.

The peak asymmetry canceller is in essence an all-pass network, a class of not minimum phase network. That is: a network whose transference function has zeros in the right semi plane. This network has a full flat response to frequency; only its phase response is function of the frequency. This phase rotation, which must compliment very particular conditions, is responsible of the peak symmetry of the audio signals. Signals that by their nature are symmetrical (like most of the musical instruments), are not modified by this processor.

This processor, by itself, allows to increase between 3 and 5 dB the final power broadcast by your transmitter (it is to say that it increases by a factor of TWO the average power transmitted). Numerous tests have been made in different countries, to verify, in real conditions, these results.

Stage 2: Input Expander

The expander, previous to the compression process, is an excellent resource to increase the signal/noise ratio of the original program. This is advisable, since the compression process, when reducing the high level passages, consequently increases the relative level of the passages of low level, and therefore the noise. This is a forced consequence of the compression process that has particular effect in the increasing of the ambient noise of the microphones. To avoid this, Solidyne processors incorporate a linear expander, previous to the compressor stage.

The concept of linear expander implies an expander that works within a very wide range of signals, below a threshold value. That means it always expands within that range, for any level of signal. That is to say that their curve of transference, based on the input level, is a straight line (from there the "linear" name). This implies that by each 10 dB that the input level reduces, the expander will reduce, for example, 13 dB. This happens for any input value, below the threshold. Then if the input is reduced in 30 dB, the output will do it in 39 dB; that is to say that the background noise has been reduced in 9 dB. This way, the expander compensates the increase of the noise that the compressor, like undesired effect, will increase.

At this point, maybe you will be thinking that DOES NOT HAVE SENSE to make an expander of the signal and soon to compress it. You will think, perhaps, that an effect cancels to the other. But it's not true for two reasons. First: the different attack and recovery times. Second: multiband compressors have elevated threshold, whereas the linear expander has a very low threshold and a linear behavior below the threshold. It means that the actions do not cancel, because both processes **are not complementary**.

The linear expander, to optimize its behavior, has instantaneous attack and a fast recovery times. Here is where the psychoacoustic concept "post-pulse hearing inhibition" is used. This allows using an expander with a quick recovery time, so that the ear does not perceive it. The broadband compressor that follows the expander has a very slow recovery time. Therefore, with impulsive signals, as the audio program, does not exist any cancellation effect.

Another advantage of using a linear expander previous to the processing is that an excellent audible sensation of dynamic range is obtained. In fact, recent studies have demonstrated that the audible sensation produced by the level variations of an audio signal, is related to the changes happened in the first 50 milliseconds, and is little dependent of the reached final value. This implies that an expander in the short term is perceived like a great dynamic range, whereas the power sensation (and even the coverage area of the radio transmitter) is related to the average energy, which depends of the compression of the energy level.

You can see that they are two concepts different. With audio processors of conventional design, the expander and the compression were antagonistic concepts. This does not happen in the field of the psychoacoustics processors.

Stage 3: Level Input Control

The Audimax 362HDHD has an automatic control for the input gain. Manual adjustments are not necessary. The **AGC** (Automatic Gain Control) guarantee that the audio signal enters to the delicate multiband compressors always with the same level, avoiding variations on the transmitted signal.

The AGC is designed to work with input levels from –10 dBu to +15 dBu, which qualifies to the AudiMax to work with all types of audio consoles, from DJ's mixers to professional broadcasting consoles!

Stage 4: Multiband Compressors

The purpose of the multiband compressors is to increase the perceived loudness sensation. The human voice and music will sound more solid, with better dynamic balance. Still more, the increase of the average energy of the audio signal is very considerable, increasing the coverage area of the radio for A.M. and FM transmissions (for more info visit www.solidynepro.com).

Multiband technology bases on the studies of Stevens (ref 1.2.3) about the loudness of each band frequency and the studies of Zwicker (ref 4) about its relation with the Critics Bands of the human ear. The integration time of the ear to reach the maximum loudness is of the order of 200 milliseconds (ref 5). This time must carefully be incorporated to the controls of the loudness compressors, to obtain the desired effect. The ear will perceive a greater loudness when the band compressors increase the relative loudness level.

The processor Orion 462 has frequency splitters with Butterworth filters of 18 dB/octave that divides the program signal in four frequency bands: low, low-middle, high-middle, high. In this form, the multiband compressor in independent form processes each bank of frequencies. This way is possible:

1. **To increase the total energy**, by the use of fast compressors for bass and ultra-fast for treble. If the bands were not divided, the compressors with so fast recovery time would produce a disagreeable sound effect; the percussion of low frequencies would modulate the high notes. And the high notes of an instrument would as well modulate the low tones, of violoncello, for example.

2. **To increase the perceived loudness**. This is because most of the modulation capacity of a transmitter is generally devoted to low frequency signals, of less than 160 Hertz. Nevertheless this information contributes very little to the loudness sensation, due to the reduced sensitivity of the ear for those frequencies. Therefore is desirable to increase the level of the medium and high frequencies. But this cannot be obtained by simple equalizing, because the sound balance would be destroyed. On the other hand, the peaks of high frequency would saturate the transmitter. The compression in separated bands allows increasing between 6 and 12 dB the energy for high frequencies without altering the tone balance; in fact, the frequency response continues being totally flat.

3. **To improve the audio quality**. Processing completely eliminate the "flat sound" sensation, perceived when a sonorous material is compressed, by means of fast compressors. This is obtained, additionally to the division in bands, using attack times appreciably elevated. This allows that very short peaks of the audio signal arrive freely to the following processor (peak clipper), that eliminates them, but maintaining the psychoacoustic sensation of power associated with the audio peaks.

Stage 5: Dynamic Equalizer

The Dynamic EQ is a 3 bands audio equalizer that acts over the threshold of the multiband compressor.

This technology operates in 3 bands (low, middle and high) modifying the density of energy (instead the level) of each band. Is formed by complementary filters of 18dB/octave carefully designed to obtain flat response. This built-in equalizer offers an enormous flexibility. In example: is well-known that the use of EQ at the console output has an adverse effect in the sound quality, since the more a frequency band is emphasized, grater is the action of the audio compressor (previous to the transmitter) for that band. Equalize a band implies to unbalance the entire audio spectrum. It doesn't happen with DENSITY EQ, since its action is coordinated with the following stages. The boosting of a frequency band is translated then in a correlative modification of the multiband compressor threshold, to carry out the new equalization.

In this form, its action extends to the range of sounds of very high intensity, where the conventional EQ's are inefficient, due the excessive compression.

Stage 6: Energy Bands

Band Energy controls increase the peak density of the audio bands, obtaining signals with very high loudness. Knobs act on the recovery times of the compressors. Each band have a different recovery time, of variable range.

"Energy Bands" adjusts according to the music's style that the radio manages. Turning the controls to the left you will have a smooth sound (long recovery times); whereas turning them to the right you increases the energy for the bands, which will produce a more "hard" sound with great "punch" and "sharp" highs (fast recovery times). As example we say that for melodic music, classic (academic), etc., in which there is no noticeable rhythmical support, agrees not to emphasize too much the Energy Band controls, that is to say, to use long recovery times. For Rock & Pop is advisable to increase the energy bands so that the sound has more "punch".

Stage 7: Stereo Coder

The sum of the signals is sent to the **stereo coder**. It uses digital technology to generate the MPX signal. This technique, created by Solidyne, allows to obtain a perfect coder with distortion 10 times below the audibility threshold and channel separation better than 75 dB.

It's based on the oversampling concept, that divides the audio signal in 16 samples that are processed separately at 38 x 16 = 608 KHz. Due to this elevated sampling rate, the anti-alias filters works over 500 KHz, eliminating the "phase rotation" effect at 53 KHz. With this new solution and the use of advanced technology in each part of the circuit, residual components of distortion below -90 dB are obtained.

It is described separately in this manual, the way to do measurements and reception tests of the stereo coder (see Chapter 4).

MPX processing

The studies about the modulation on an FM transmitter indicate that when the transmitter is modulated by stereo MPX signal appears a new effect, not present on the original audio signal.

This effect, denominated in U.S.A. *MPX Interleaving* (also known as *peak correlation*), determines that the modulation peak in MPX does not coincide with the modulation peak of the stereo signal, considered in independent form.

This means, in simple terms, that if the peaks of channels L and R are limited separately so that MPX signal never over modulate, during most of the time, the modulation capacity of the transmitter will be wasted. And this happens because signal MPX is the sum of L+R but also includes the 38 KHz sub-carrier. According to the relation of phase between these three elements, there will be different peak values for the interleaving. This phenomenon indicates that is possible to increase the modulation without increasing the deviation of 75 KHz of the transmission, taking advantage of the modulation capacity that normally is wasted.

Audimax 362HD processor uses a MPX Processing technology named Super Modulation. This processing consists on a system that controls the peaks, operating at 608 Khz, eliminating the peaks in MPX base band signal and filtering them so that there are not left residual components.

REFERENCES

1.- S. S. Stevens, The measurement of loudness, ASA Journal, Vol.27, pg. 815.

2.- S. S. Stevens, The direct estimations of sensory magnitudes-loudness; American J. Psychol. 69, 1-25, 1956.

3.- S. S. Stevens, Concerning the form of the loudness function; ASA Journal, Vol. 29, pg 603-606, 1957.

4.- E. Zwicker – Flottrop – Stevens; critical bandwidth in loudness sumation, ASA Journal, Vol. 29, pg. 548-557, 1957.

5.- Stanley Gelfand, Hearing, pg. 392, Edited by M. Dekker, N. York, 1990.

6.- Oscar Bonello. NEW IMPROVEMENTS IN AUDIO SIGNAL PROCESSING Journal of the Audio Engineering Society, Vol. 24 № 5. USA, 1976

7.- Oscar Bonello PC CONTROLLED PSYCHOACOUSTIC AUDIO PROCESSOR, 94th Audio Convention, Berlin March 1993

8.- Oscar Bonello, Burst Masking (Enmascaramiento por Ráfaga) Anales del II Congreso Iberoamericano de Acústica, Madrid, octubre 2000

9.- Oscar Bonello, Multiband Audio Processing and Its Influence on the Coverage Area of FM Stereo Transmission, Journal of Audio Engineering Society, New York, March 2007

Chapter 4 – Technical Specifications

Input
XLR3 connector, self-adjusted level
Level: -10 dBu to + 15 dBu
Z= 600 / 10 Kohms, balanced

Output
Balanced, + 4 dBu Z= 600 / 10 Kohms, with de-emphasis

MPX Output
600/10 Kohms, factory set level to
standard 4 Vpp. Differential output to cancel hum loops between
transmitter and studio ground

Frequency Response
20 - 16.000 Hz +/- 0,5 dB
measured below compression & limiter threshold

Harmonic Distortion
Below 0,02 % @ 30-15.000 Hz

Noise
Below - 90 dBA ref 100 % modulation

Stereo Separation
> 75 dBA

Subsonic Filter
Chebyshev 2nd order, 15 Hz

Asymmetry Cancelling
5:1 cancelling effect, using Khann-Bonello method

Expander
10:1 slope, 100 uS attack time
AGC (wideband)
VCA controlled, 30 dB range

Multiband Compressors
3 bands, 18 dB/octave, linear phase crossover
Compressors: 30 dB full range, 5:1 slope Automatic attack time / Release
controlled by Energy panel controls
IM Cancelled Clipper
IM attenuation > 30 dB below 250 Hz
Dynamic EQ
0 - 12 dB dynamic boost at Low, Mid and High Frequency

Processing Power
7 stages of processing devices

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

Dimensions
19" rack mount. Module one (44,4 mm)

Pilot tone stability
+/- 0,002 % (+/- 0,5 Hz)

STEREO CODER SPECIFICATIONS
Measured from internal Stereo coder
jumper to MPX out

Audio input
2 Vpp for 100 % MPX output (4 Vpp)

Frequency Response
15 Khz/5 order elliptic LP filter
20-15.000 +/- 1 dB
Attenuation at 19 Khz > 50 dB

Harmonic Distortion
Less than 0,01 % THD at 1 Khz
Below 0,015 % 20-10.000 Hz

Signal to Noise Ratio
Better than 85 dBA with reference to
100% modulation

Stereo Separation
Better than 50 dB @ 20-10.000 Hz
Typical > 60 dB at 1 Khz
38, 57, 76 & 95 KHz suppression
Below - 70 dB