



bw broadcast

# DSPXtra-HD

incorporating the

*Ariane*

RMS Leveller



## Digital Audio Broadcast Processor

Operational Manual  
Version 1.10

[www.bwbroadcast.com](http://www.bwbroadcast.com)

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## **WARRANTY**

BW Broadcast warrants the mechanical and electronic components of this product to be free of defects in material and workmanship for a period of one (1) year from the original date of purchase, in accordance with the warranty regulations described below. If the product shows any defects within the specified warranty period that are not due to normal wear and tear and/or improper handling by the user, BW Broadcast shall, at its sole discretion, either repair or replace the product.

If the warranty claim proves to be justified, the product will be returned to the user freight prepaid.

Warranty claims other than those indicated above are expressly excluded.

### **Return authorisation number**

To obtain warranty service, the buyer (or his authorized dealer) must call BW Broadcast during normal business hours BEFORE returning the product. All inquiries must be accompanied by a description of the problem. BW Broadcast will then issue a return authorization number.

Subsequently, the product must be returned in its original shipping carton, together with the return authorization number to the address indicated by BW Broadcast. Shipments without freight prepaid will not be accepted.

### **Warranty regulations**

Warranty services will be furnished only if the product is accompanied by a copy of the original retail dealer's invoice. Any product deemed eligible for repair or replacement by BW Broadcast under the terms of this warranty will be repaired or replaced within 30 days of receipt of the product at BW Broadcast.

If the product needs to be modified or adapted in order to comply with applicable technical or safety standards on a national or local level, in any country which is not the country for which the product was originally developed and manufactured, this modification/adaptation shall not be considered a defect in materials or workmanship. The warranty does not cover any such modification/adaptation, irrespective of whether it was carried out properly or not. Under the terms of this warranty, BW Broadcast shall not be held responsible for any cost resulting from such a modification/adaptation.

Free inspections and maintenance/repair work are expressly excluded from this warranty, in particular, if caused by improper handling of the product by the user. This also applies to defects caused by normal wear and tear, in particular, of faders, potentiometers, keys/buttons and similar parts.

Damages/defects caused by the following conditions are not covered by this warranty:

Misuse, neglect or failure to operate the unit in compliance with the instructions given in BW Broadcast user or service manuals.

Connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used.

Damages/defects caused by force majeure or any other condition that is beyond the control of BW Broadcast.

Any repair or opening of the unit carried out by unauthorized personnel (user included) will void the warranty.

If an inspection of the product by BW Broadcast shows that the defect in question is not covered by the warranty, the inspection costs are payable by the customer.

Products which do not meet the terms of this warranty will be repaired exclusively at the buyer's expense. BW Broadcast will inform the buyer of any such circumstance. If the buyer fails to submit a written repair order within 6 weeks after notification, BW Broadcast will return the unit C.O.D. with a separate invoice for freight and packing. Such costs will also be invoiced separately when the buyer has sent in a written repair order.

### **Warranty transferability**

This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BW Broadcast.

### **Claims for damages**

Failure of BW Broadcast to provide proper warranty service shall not entitle the buyer to claim (consequential) damages. In no event shall the liability of BW Broadcast exceed the invoiced value of the product.

### **Other warranty rights and national law**

This warranty does not exclude or limit the buyer's statutory rights provided by national law, in particular, any such rights against the seller that arise from a legally effective purchase contract. The warranty regulations mentioned herein are applicable unless they constitute an infringement of national warranty law.

## SAFETY INSTRUCTIONS

**CAUTION:** To reduce the risk of electrical shock, do not remove the cover. No user serviceable parts inside. refer servicing to qualified personnel.



**WARNING:** To reduce the risk of fire or electrical shock, do not expose this appliance to rain or moisture.



This symbol, wherever it appears, alerts you to the presence of uninsulated dangerous voltage inside the enclosure—voltage that may be sufficient to constitute a risk of shock.



This symbol, wherever it appears, alerts you to important operating and maintenance instructions in the accompanying literature. Read the manual.

### DETAILED SAFETY INSTRUCTIONS:

All the safety and operation instructions should be read before the appliance is operated.

#### Retain Instructions:

The safety and operating instructions should be retained for future reference.

#### Heed Warnings:

All warnings on the appliance and in the operating instructions should be adhered to.

#### Follow instructions:

All operation and user instructions should be followed.

#### Water and Moisture:

The appliance should not be used near water (e.g. near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool etc.).

The appliance should not be exposed to dripping or splashing and objects filled with liquids should not be placed on the appliance.

#### Ventilation:

The appliance should be situated so that its location or position does not interfere with its proper ventilation. For example, the appliance should not be situated on a bed, sofa rug, or similar surface that may block the ventilation openings, or placed in a built-in installation, such as a bookcase or cabinet that may impede the flow of air through the ventilation openings.

#### Heat:

The appliance should be situated away from heat sources such as radiators, heat registers, stoves, or other appliance (including amplifiers) that produce heat.

#### Power Source:

The appliance should be connected to a power supply only of the type described in the operating instructions or as marked on the appliance.

#### Grounding or Polarization:

Precautions should be taken so that the grounding or polarization means of an appliance is not defeated.

#### Power-Cord Protection:

Power supply cords should be routed so that they are not likely to be walked on or pinched by items placed upon or against them, paying particular attention to cords and plugs, convenience receptacles and the point where they exit from the appliance.

#### Cleaning:

The appliance should be cleaned only as recommended by the manufacturer.

#### Non-use Periods:

The power cord of the appliance should be unplugged from the outlet when left unused for a long period of time.

**Object and Liquid Entry:**

Care should be taken so that objects do not fall and liquids are not spilled into the enclosure through openings.

**Damage Requiring Service:**

The appliance should be serviced by qualified service personnel when:

- The power supply cord or the plug has been damaged; or
- Objects have fallen, or liquid has been spilled into the appliance; or
- The appliance has been exposed to rain; or
- The appliance does not appear to operate normally or exhibits a marked change in performance; or
- The appliance has been dropped, or the enclosure damaged.

**Servicing:**

The user should not attempt to service the appliance beyond that is described in the Operating Instructions. All other servicing should be referred to qualified service personnel.

**CE CONFORMANCE:** This device complies with the requirements of the EEC Council Directives: 93/68/EEC (CE Marking); 73/23/EEC (Safety – low voltage directive); 2004/108/EC (electromagnetic compatibility). Conformity is declared to those standards: EN50081-1, EN50082-1.



**WARNING:** This equipment generates, uses, and can radiate radio frequency energy. If not installed and used in accordance with the instructions in this manual it may cause interference to radio communications. It has been tested and found to comply with the limits for a Class A computing device (pursuant to subpart J of Part 15 FCC Rules), designed to provide reasonable protection against such interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, at which case, the user, at his own expense, will be required to take whatever measures may be required to correct the interference.



**CANADA WARNING:** This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications. Le présent appareil numérique n'émet pas de bruits radioélectriques dépassant les limites applicables aux brouillages radioélectrique édicté par le ministère des Communications de Canada.

## **FORWARD**

Thank you for your purchase of the DSPXtra-HD digital audio broadcast processor. The processor before you, DSPXtra-HD, has been specifically designed for digital audio broadcasting. This includes over-the-air digital broadcasting systems such as DAB, DAB+, HD Radio™ (IBOC) AM or FM, DRM, DRM+, FMeXtra™ as well as Internet based broadcasting (streaming).

Over the last decade the staff at BW Broadcast have observed broadcast processors from afar with fascination and intrigue and shared a keen interest in digital audio processing. In January 2002 we decided on a whim that it was time to have a go at designing our own digital audio processor. We knew that if we were going to design an audio processor we had to do it the BW way and make the processor the most cost effective fully featured all-in-one broadcast processor on the planet. The one aim, to make an audio processor that offered all of the features found in the other more costly processors but at a fraction of the price. That processor was the DSPX!

Why DSPX? Like most products the concept is conceived nameless. The name DSPX came about for several reasons. Firstly, the BW team couldn't decide on a name we liked so the concept became DSP processor X. As the project developed it soon became clear that the processor was to be a processor for all seasons and so in true algebraic fashion X represents many things. DSPX was born.

18 months later and it was time for a more advanced model. We bring you, the DSPXtra. The Xtra is the result of continued research and development and the inclusion of the award winning Ariane RMS leveller concept from TransLanTech sound of New York city.

DSPXtra-HD is built on DSPX-HD but with Extra added features. More DSP horsepower, improved distortion control, six bands of peak limiting and that all important award winning Ariane RMS leveller. The processing algorithm inside DSPXtra-HD is optimized for digital audio broadcasting and features linear filters for maximum transparency, advanced levelling and compression multi-band processing and precision low IMD codec-aware look-ahead limiting that reduces coding artifacts.

Once again, thank you for your purchase; we hope you enjoy the DSPXtra-HD!

## **BW Broadcast Team**

# INTRODUCTION TO THE DSPXtra-HD

The BW Broadcast DSPXtra-HD is a new generation of digital audio signal processor for digital broadcasting such as DAB, HD Radio or internet streaming. Using the latest multi-band DSP technology the DSPXtra-HD offers a versatile and powerful tool in creating a loud punchy on-air presence.

The DSPXtra-HD has been designed and built from scratch using a new approach to the design of a digital audio processor that incorporates the most up to date components. Cutting edge technologies allow the DSPXtra-HD to produce similar results to other processors in the market but in a simpler more cost effective way. The advances we have made have allowed us to pass the savings on to our customers.

## What's Under the Lid?

The DSPXtra-HD is driven by a fast 8 bit micro-controller which controls an array of specialised analogue and digital circuits: These include 24-bit A/D and D/A converters, analogue level control circuitry, 22 x 24 bit DSP's, an ethernet port, a trigger port, a USB port, an RS232 port, over 200 LEDs (metering), an LCD screen, 2 sample rate converters, a headphone jack and memory devices to hold the software and firmware.

## The Processing Architecture!

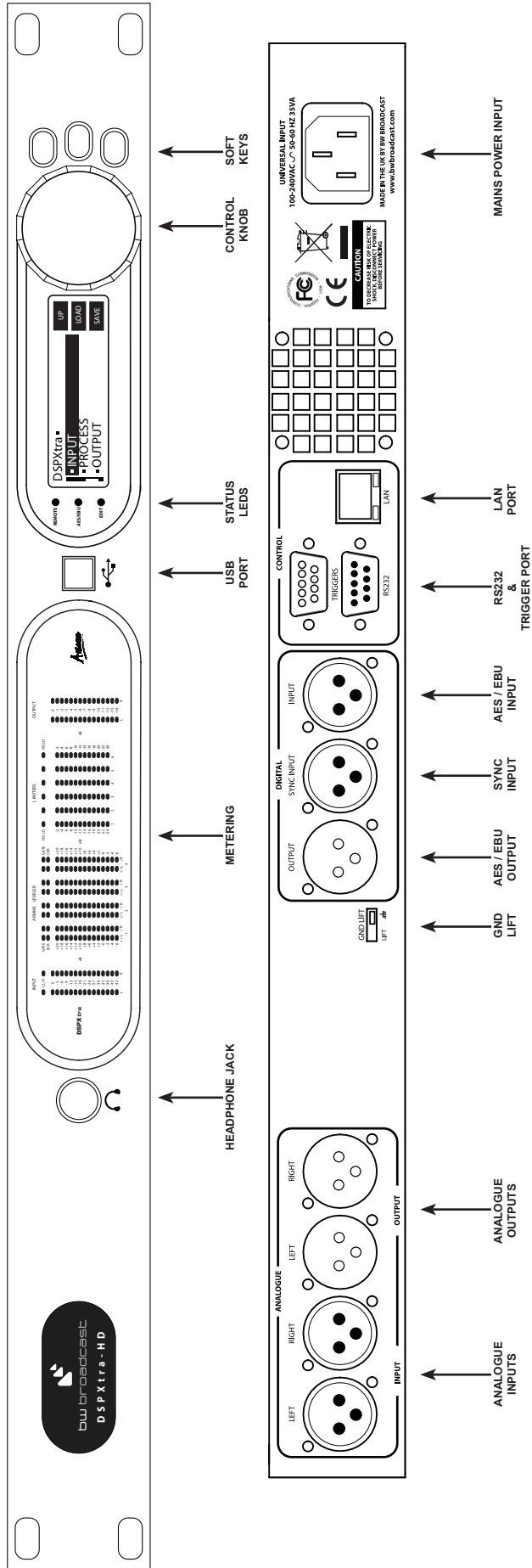
After input selection the 24-bit digital audio signal is passed through conditioning circuitry before being presented to the Ariane RMS levelling block. The four band RMS leveller corrects for input level variations and also improves consistency. The block also can provide stereo enhancement or restriction with its 'matrixed' mode of operation. The output of the AGC feeds the EQ and bass enhancement sections before being split into six bands by linear phase time aligned filters. The six bands are processed by dynamic audio limiters on each band. Multiband look-ahead limiting plus precision final look-ahead limiter ensures your signal is kept to a strict maximum while maintaining consistent and clear sound.

The easy to use front panel control system and LED metering display, afford the user with ease of use and setup.

Comprehensive control of every processing parameter is available to the user both from the front panel control system and by remote (computer) control.

At a fraction of the size, weight and price of its rivals, the DSPXtra-HD is a clear winner.

**Dynamic, fresh & innovative...**  
**... The DSPXtra-HD**

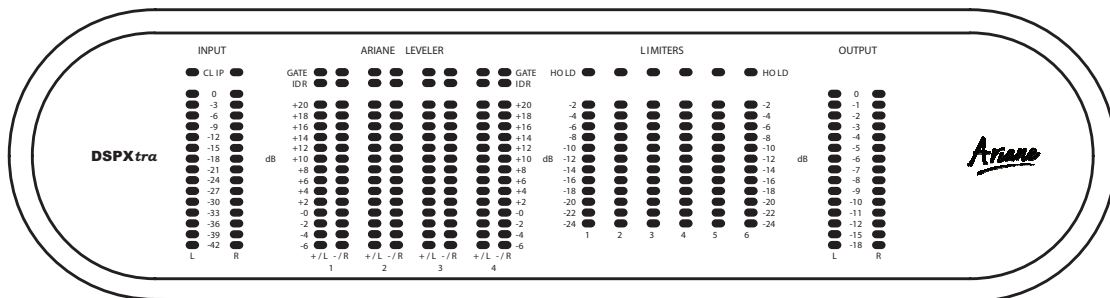


## DSPXtra-HD FRONT AND REAR PANELS



# DSPXtra-HD METERS

The DSPXtra-HD has several hundred LEDs that provide ‘always on’ instant IO and processing metering.



## I/O metering

The input meters show the level of the input audio. The meters are ‘hooked in’ to the DSP code after the input level selection and mode options. The clip LED’s represent the onset of the A/D convertors clip-point and these LED’s should not light under any circumstances. Adjust the input gain control of the DSPXtra-HD if they do.

The output meters represent the level in dB below full scale output. This output level is the peak output level of the processing and has nothing to do with the actual output level set by the analogue and digital output level options.

The output meters show a smaller dynamic range compared to the input ones. This reflects the smaller dynamic range of the audio once processed by the DSPXtra-HD. If we were to have used the same scale as the input metering we would not see a lot of activity on the LED’s, with all of the LEDs on most of the time.

The IO meters follows an approximation of the PPM level of the audio waveform while a floating ‘peak hold’ dot tracks the absolute value of the waveform

## G/R metering - Ariane leveller

The Ariane metering displays gain being applied with more leds indicating more gain, the metering having eight columns, two columns for each band. The scale is -6 to +20dB in 2dB steps.

When the leveller is operating in matrix mode the left hand column of each band (1 through 4) represents the Right plus Left component (aka the SUM signal) while the right hand column of each band represents the Left minus Right component of the audio (aka the DIFFERENCE signal).

When the leveller is operating in stereo mode the left hand column of each band represents the left channel and the right hand column represents the right channel.

When in bypass the metering returns to the 0dB gain position to indicate unity gain through the leveller (bypassed).

There are two other leds above each gain column. One is the gate for the band and the other is the IDR for the band. More information on the gate and IDR is contained in the processing setup section of this manual.

## G/R metering - Six band limiter

The six band limiter displays gain reduction from 0dB to -24dB in 2dB steps.

There is only one meter per stereo channel and the value shown is the largest gain reduction of the left and right channels. Under normal operation (with a stereo audio feed) this is fine but you may observe strange metering if the channels are not very balanced in level or you are using the DSPXtra-HD to process two separate mono feeds.

The red LEDs above each gain reduction meter indicate HOLD when lit. More information on HOLD function is contained in the processing setup section of this manual.

## STATUS LEDS

The DSPXtra-HD front panel contains three status LEDS.

**REMOTE:** Indicates that the DSPXtra-HD is currently talking to a remote computer. This will flash during an update of the firmware or remote control with the remote control application. **REMOTE** ●

**AES/EBU:** Indicates the presence of a valid AES/EBU signal connected to the digital audio input of the DSPXtra-HD. **AES/EBU** ●

**EDIT:** Indicates that you are currently editing a parameter. **EDIT** ●

## QUICK START

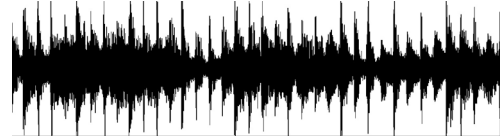
1. Install the processor into the rack.
2. Connect AC power to the unit and turn on the power.
3. Connect the analogue and / or digital audio inputs.
4. Select the analogue or digital input as the source of the processing with the 'INPUT SELECTION' parameter which can be found in the 'INPUT' menu. Apply audio and observe the input meters. For analogue inputs, adjust the 'INPUT' menus 'INPUT LEVEL CONTROL' so that the input meters do not clip. We recommend setting the mixing board or audio source to full level output (even clipping) prior to adjusting this control. This ensures that the processor's A/D converters will not clip under any conditions.
5. Connect the audio analog and/or digital outputs as required and set the output levels for the analogue and digital outputs to match the equipment that the processor is connected to.
7. Press the 'LOAD' soft key and select a preset from the list!

# INTRODUCTION TO AUDIO PROCESSING

Most audio processors use a combination of compression, limiting and clipping to 'funnel' the dynamic range down, reducing the peak to average ratio in each stage. A cascaded arrangement of compressor, limiter and clipper produces the best results. The first stage of processing usually operates in a slow manner, the processing getting progressively faster and more aggressive as the audio passes through the chain. The instantaneous peak clipper or look-ahead limiter is the final stage of the chain and sets the final peak level.

The images below illustrate a section of audio as it passes through a typical audio processor.

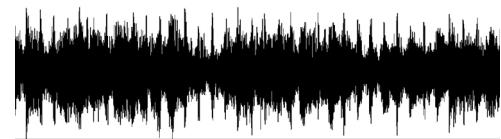
The first image to the right is an unprocessed section of audio.



The images that follow represent compression of the input waveform, followed by limiting and then finally peak clipping.

## Compression

Compression reduces the dynamic range of the audio waveform slowly in a manner similar to a trained operator riding the gain. Compression is usually performed on the RMS level of the audio waveform and the ratio of compression is usually adjustable. Compression is usually gated to prevent gain riding and suck-up of noise during silence or quiet periods.



## Limiting

Limiting is a faster form of compression that employs faster time constants and higher ratios to produce a denser sound while controlling peaks based upon the peak level of the audio waveform. Excessive limiting can create a busier packed wall of sound effect.



## Clipping

Clipping the audio waveform will not produce any audible side effects if performed in moderation. Excessive clipping will produce a form of distortion that produces a tearing or ripping sound. Clipping can also be used as an effective method of high frequency peak control when used in conjunction with distortion controlling filtering.



## Look-ahead limiting

Often used instead of a clipper in systems that feed bit rate reducing audio codecs, look-ahead limiting examines the audio waveform and prepares a gain control signal in advance of the delayed audio waveform arriving. This prevents overshoots while minimising distortion. A look-ahead limiter behaves in the same way as a soft clipper. Competent look-ahead limiters will usually be of the multi-band variety.

# THE TECHNICAL AND THE PHILOSOPHICAL BACKGROUND OF THE ARIANE DIGITAL AUDIO LEVELLER

(Or, Why Automatic Level Control Doesn't Have to be Processing's Weak Link!)

## What is Levelling?

Levelling is the automatic process of reducing the long-term dynamic variations in level of an audio source. Also called Automatic Gain Control or, AGC.

## Why Use Levelling?

Levelling is required for multiple purposes, but the most important is to force a given program's level variations to best conform to the requirements of human hearing of the consumer in their expected listening environment. Average levelling control is done on the basis of the audio signal's average level over time, while peak levelling control is based upon the highest instantaneous value. While peak amplitude value can be determined by simple measurement of the audio's absolute value at any given moment in time, average control can only be derived by analysis over a specified period of time.

## Levelling for effect

Levelling can be for artistic purposes, or simply as a control utility. These two separate purposes are distinction are not usually made as most processors' effect is taken as inseparable. The Ariane was created specifically to allow control without any effect. Automatic level control can be implemented in ways that are very audible. For our purposes, however, levelling is being considered purely for its merit as a utility for the maintenance or improvement of audible perception. The Ariane has been designed to have little or no sound effect of its own. The Ariane's level-managed output will allow the following dynamics processing to work with an extremely consistent signal, so that the expected effect will itself be more consistent and predictable.

## Levels and Human Hearing

It is accepted among many learned students of human perception that human hearing, complex as it is, is more sensitive to the long-term average of sound level than it is to instantaneous peaks. The human ear/brain perception system integrates sound levels over time. That is, a sound of a given level that is sustained for a length of time will sound louder than a shorter time burst of the same level. Depending upon duration and intensity, a particular sound can actually be lower in level than another, yet be perceived to sound louder. The sensitivity of human hearing to time as well as amplitude is critical when considering how to control listening levels over time.

## RMS Average

One very useful way to determine average level is called Root Mean Squared, or RMS for short. This is a mathematical process where the audio's absolute value is squared, then an average of this signal is taken over a given length of time, then the square root is taken of that average. The result is a control signal that corresponds to the power level of the audio over time, rather than its peak level.

RMS measurement, with its requisite integration, corresponds nicely with human hearing. By using an RMS system of measurement and control, a level controller can work remarkably well in conjunction with human hearing. Thus, by choosing to analyse our signal with RMS detection, we can build a simple system whose control is much less distracting to the human ear than a simple peak-based control system.

## The Nature of Program Audio

The creation of programs and presentations of audio generally requires the mixing and level adjustment of multiple sources. It is the nature of 'raw' unprocessed audio sources (such as live microphones) to be wide ranging in their levels, consistency and other qualities. These sources are the ones that most obviously will require a leveler's automatic assistance. Yet it is also the nature of audio that some sources, such as produced music, will be very consistent in level and quality, even while their initial absolute level may be unknown. These sources will not require as much control, if any, as the unprocessed material. What is most typical of all, is that these multiple types of sources may need to be conformed to each other before their final combined presentation to the end user. Taking advantage of coincidental properties of human hearing and program audio, it doesn't take a genius to note that the audio for programming should be presented in a way that humans find most easy to accept. While extreme loudness variations are disturbing, corrective control systems themselves can add effects that are equally if not more disturbing. In other words, in audio levelling, the cure can be worse than the illness. By seeking out and utilizing only those aspects of audio control that are utilitarian AND pleasing to the ear, while keeping at arm's length those methods of audio control that are generally found to be distracting, and then matching this knowledge base with an understanding of the nature of typical program audio, an intelligent control system can be designed that best meets the needs of successful, listenable programming over the long term.

## Windowing Release

While we humans respond to average level in determining loudness, we also crave detail. That is, minute varia-

tions create the 'flavour' of the sound, and help distinguish, for example, a flute from an oboe. Details 'live' in the fast variations that are mostly confined to the peaks, so preserving the fast variations while making level corrections only relatively slowly maintains this audio flavour. In the overall scheme of audio processing, the less we rely upon fast peak control and the more we rely upon slow average control, the more life-like our audio will be. If confining our wide-range level changes to a slow control will reduce the destruction of audio detail, then stopping the control altogether totally eliminates any possibility of detail eradication.

### **How And Why Windowing Release works**

In a simple level control, when audio gets too loud it is turned down, and when it is too soft it is turned up. But what if we had a third possibility, i.e., if it is at the proper level, do nothing?

*This is what windowing release is all about.*

First presented in the CBS Laboratories "Audimax" level controller of the 1960s, the concept of windowing release (which CBS called "Platforming" and is also described as 'hysteresis release') is an elegant way to have your cake and eat it, too. The basic idea is that if the acceptable (and desirable) short-term variations of the audio remain within a fixed range, the audio level is considered to be correct. The control system should make NO adjustment until the audio moves away from the correct level. By determining a decibel range within which any variation in the audio's level is considered to be acceptable, we can set our equipment to then make changes only when it leaves this range. This 'do nothing' range could be as small as a fraction of a decibel, or as wide as we want to make it, though practical experiments have shown the useful range is from about one to fifteen decibels.

Should the audio's energy go outside either the top or bottom thresholds of our imaginary 'window', the control system will briefly engage to correct the audio level, moving the window with it. Once our window of acceptable variation is set at the newly appropriate level, the system once again disengages control and does nothing. The result is that audio levels can be aggressively controlled, yet not sound it. All the detail, the instantaneous variations, the 'flavour' remains, even though the audio is held at a consistency of level that the human ear typically finds to be very comfortable. The sound is extremely natural, mainly because our system spends most of its time doing nothing. Multiple Band ('multi-band') AGC level controller that uses a single, wide-band system will not alter the spectral balance of the controlled audio in any way. This is a good and admirable goal. But a wide-band AGC system has a serious flaw: a highly-audible control side effect called spectral intermodulation. When one instrument that is confined to a limited portion of the spectrum (such as a bass drum) is higher in energy level than the rest of the spectrum combined, the level intensity and rhythmic variation of that instrument will dominate the control and 'modulate' the AGC's automatic level variation over the entire spectrum. When the rhythmic characteristics of the lower-level instruments (such as strings) are sustained and held longer, the modulation effect is clearly obvious. It is most annoying when the modulating instrument is at one extreme of the spectrum where the ear is least sensitive while the modulated instruments are in the middle of the most ear-sensitive area of midrange frequencies. Various solutions have been attempted this "hole-punching," such as slowing down the attack and release of the wide-band system, or desensitizing the low frequency sensitivity of the control loop. Using a windowing release also helps. But these 'solutions' only slightly reduce the fault, they do not fix it.

The solution is to control different portions of the spectrum with some degree of independence. In most cases it can be shown that, for a given amount of dynamic gain change, by splitting the audio into several frequency bands any control 'artifacts' are much less audible to the human ear than using a single band.

There is however, a potential problem with multi-band processing that can make it a deal-breaker: "spectral skewing." Spectral skewing is what happens when the operating characteristics of the multiple bands cause them to drift way from each identical gains when differing material comes along at differing levels. The effect can be deadly: a song can have a different spectral balance depending only upon how hard the processor is driven. Song fades turn muffled or shrill, depending on processor settings. The solution to the spectral skewing problem is to make each band's detection and control systems react with the same characteristics, i.e., reaction times (attack and release), compression ratios, thresholds, etc., and, importantly, set the crossover frequencies such that there is a similar amount of energy in each band. This is the approach undertaken with the Ariane, and spectral skewing very effectively becomes a non-issue. Any resultant spectral shift will in most cases be unnoticed and in some cases, considered an improvement.

### **Feedforward control**

Historically, automatic gain controllers have used a detection scheme located at the output of the processor which was fed back to control the gain at the input. This is a self correcting system, called "feedback" or "closed-loop." Until accurate and stable linear-to-logarithm (and vice versa) converters became common, feedback processing was the only economical option. The problem with feedback processing is that the control range is limited and non-linear from one end of its control range to the other. This makes feedback best suited only for relatively small amounts of variable gain, i.e., less than 20 dB. This is OK for peak limiting, where the

range of control is usually less than 15 dB, but for the purpose of a modern AGC this range is too 'limited.' A "Feedforward" (or "open loop") detector analyses the signal before the variable gain element and creates a control signal that can accurately mirror level variations in the incoming audio. With a feedforward system the effective range is many decades of decibels, and its linearity of control is virtually perfect from one extreme of gain to the other. For an initial AGC for audio of an unknown level, a feedforward system makes great sense. In the Ariane, the control range is well over 40 dB, and the control characteristics are the same at both extremes.

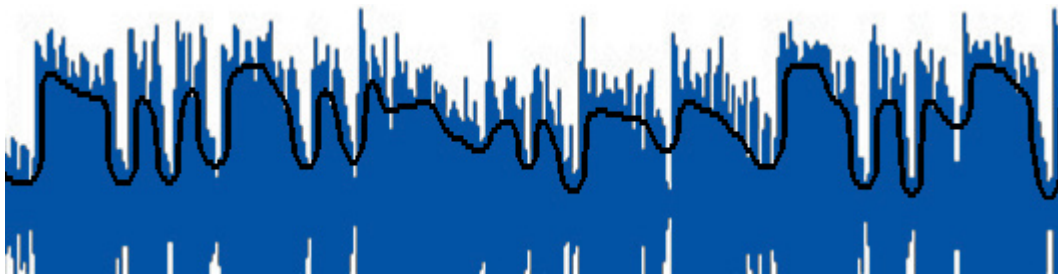
### Introducing The Ariane

The Ariane is a:

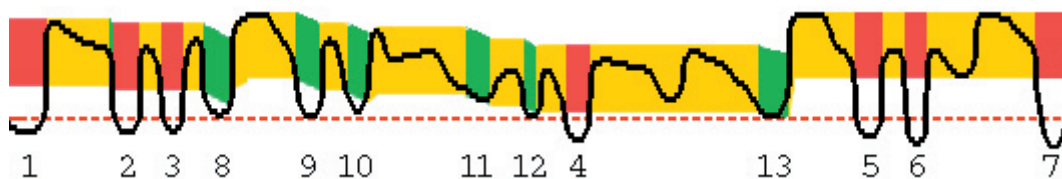
- High-quality digital,
- Two-channel,
- Multi-band,
- RMS-derived,
- Windowing release,
- Feedforward control...

automatic levelling processor for general purpose audio level management.

Each of these features of the Ariane were chosen to solve a particular problem that has manifested itself in previously available levelling solutions. Processors with one or more of the above features have long been available, but the Ariane is the first leveller to bring this unique and significant combination of features together in a unified design.



The above waveform chart demonstrates the peak level of a piece of typical music audio, in blue, superimposed with its RMS energy value, the black line. Note that the RMS level does not closely follow the peak level but, rather, represents the short-term average density of the audio. Now let's superimpose the Ariane control system upon that same RMS waveform.



The horizontal red, dashed line represents the Gate threshold. Whenever the RMS energy goes below that level (numbers 1 through 7, above), the release is gated, and the Ariane's gain holds steady, with no change. When this is true, the broad horizontal band is red (the vertical dimension of the multi colour band represents the Ariane's "window").

When the audio's RMS energy drops below the window, but still remains above the Gate threshold, the window releases (numbers 8 through 13), and the window band is green, indicating an increase in gain.

Whenever the RMS energy stays inside within the range of the window, the band is Yellow, and the audio level is unchanged, just as it was when it was gated.

In both charts, time goes from left to right, and higher amplitude is upward; in the lower chart, the broad multi-colour band's position vertically indicates more gain the lower it goes.

It's easy to see that the Ariane can be adjusted to spend most of its time holding steady, with no level change.

*This is the little bit of 'magic' that makes the Ariane sounds so good!*

## SOURCE MATERIAL QUALITY

The DSPXtra-HD has the ability to substantially improve the quality of your ON-AIR digital broadcast. However it can only work with what you provide it. The best performance will be obtained when the processor is fed with very clean source material. DSPXtra-HD can improve audio somewhat (especially intelligibility) through dynamic multi-band re-equalisation, but it can't improve audio quality in terms of distortion - poor quality source material will only sound poorer after processing.

We strongly advise against the use of MP3's and any other loss-less compressed audio formats for audio storage. Linear formats are always to be preferred. Compressed audio formats employ frequency masking data reduction techniques to reduce the bit-rate. Through re-equalisation the DSPXtra-HD can violate the frequency masking characteristics codec relied on, "revealing" distortion that was inaudible prior to the DSPXtra-HD processing.

Additionally, as digital audio broadcasting systems employ perceptual coding as part of the transmission system, using compressed sources will further decrease audio quality. If audio is stored in the play-out system in compressed format, this already reduced audio information is subjected to yet another coding cycle for transmission, often with very low target bitrate. The end result is lower audio quality and increased coding artifacts, than there would be if an uncompressed audio sources were used and cascading codecs avoided.

## FINAL LOWPASS FILTERING

The DSPXtra-HD has a built-in lowpass filter adjustable from 4kHz to 20kHz that limits the amount of high-end spectrum and energy delivered to the codec. Codecs are especially sensitive at higher frequencies, where most of the annoying artifacts like "metallic high-end", "phasing" or "squishing" occur. Reducing the amount of high-end puts less stress on the codec and can lower the amount of codec induced artifacts. The actual setting will depend on the type of codec and the bitrate used. Lower bitrates require more audio information to be thrown away by the codec, so lower filter settings would have to be employed to reduce artifacts. If you're using higher bitrates, you can set lowpass filter more towards 20kHz without causing excessive artifacts. As a rule, low pass filter frequency should be set at or below codec's half sample rate frequency (i.e. 15 kHz for codec running at 32 kHz sample rate).

## OUTPUT LEVEL SETTING

DSPXtra-HD has an adjustable output level that can be set to drive subsequent equipment with full scale digital audio level. However, caution should be made here if the following equipment using perceptual coding (codec) does not take into account internal codec overshoots. Overshoots can occur in the codec when presented with the tightly peak-controlled audio such as the one produced by DSPXtra-HD. If the codec does not provide headroom internally, these overshoots would be clipped leading to increased distortion and artifacts. Therefore we would advise setting the output level of the processor lower than 0.0 dBfs to leave the headroom in the transmission system for codec overshoots. The actual output level setting would depend on the amount of codec overshoot occurring in the system, which depends on the type of codec used and how aggressively you're processing audio with DSPXtra-HD. Generally, codecs that use spectral band replication (SBR) technology (such as aacPlus, HDC, Windows Media 11, etc.) have more overshoots than the codecs that don't, especially in the higher frequencies.

If you're using DSPXtra-HD to process for HD Radio (IBOC), DAB+, DRM or web-streaming, a good starting point to set the output level would be -2.0 dBfs. This would ensure maximum audio quality at the expense of slightly reduced level (which the users can always compensate with volume control on their radios).



# THE DSPXTRA-HD AND ITS PROCESSING STRUCTURE

The DSPXtra-HD broadcast audio processor can be used for processing audio prior to digital radio broadcast or Internet streaming. Digital radio encompasses DAB, HD RADIO (IBOC), DRM, FMeXtra and other radio based broadcasting systems. The DSPXtra-HD can also be used effectively for audio post production and mastering, ideal for giving CD's that HOT sound. It is also possible to use the DSPXtra-HD for other audio level control/ equalization applications such as night clubs and bands. However, this manual will only be referring to the use of the DSPXtra-HD for digital radio processing.

Before we discuss the processing structure in full we would like to tell you a little about the final peak limiting stage of the DSPXtra-HD. It employs look-ahead limiting as opposed to clipping. Look-ahead limiting produces less artifacts than conventional clipping, so will reproduce the original audio more accurately with less bits of digital information because it is not wasting bits encoding non-audible clipping artefacts. Look-ahead limiting produces less harmonic distortion but produces IM distortion if over driven resulting in a packed, busy sound. DSPXtra-HD employs multi-band look-ahead limiting followed by precision final look-ahead limiter.

## THE PROCESSING PATH

### Input selection and conditioning

The DSPXtra-HD offers the user input selection, gain control and a selection from a range of stereo/mono options. The audio is then routed through defeatable high pass and phase rotating filters.

A silence detector provides automatic primary to secondary input failure switching.

### Ariane RMS leveller.

For a transparent input levelling function, the DSPXtra-HD employs an RMS detected multi-band leveller Ariane. With an award winning windowing gating concept, the Ariane improves consistency and presents a uniform audio signal to the following stages. This important stage is described in more detail throughout this manual. Ariane leveller can also provide stereo enhancement through its matrix mode of operation.

### Bass enhancement

The DSPXtra-HD offers three forms of bass enhancement.

1. A 12dB/Octave shelving filter with up to 12dB of gain.
2. Bass tune control.
3. A peaking filter that can be set to provide up to 6dB of gain on 1 of four frequencies with a choice of 4 Q's. This can be thought of as a simple bass parametric.

### Xover

The DSPXtra-HD employs linear-phase time aligned digital FIR filtering to split the audio spectrum into 6 bands while maintaining sonic transparency.

### Multi-band Limiters

Each band has its own dynamic peak limiter. Multiple time constant based detectors with built in adjustable hold and delay functions significantly reduce distortion.

### Mixer

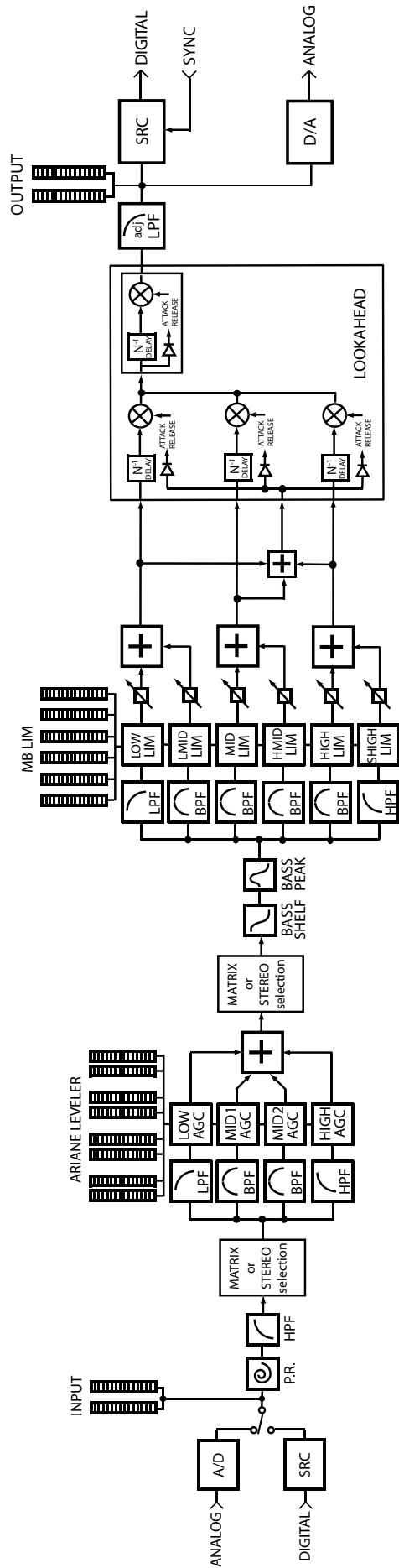
The six bands are 'virtually' mixed together at this stage. In truth, the six bands have become three. The three bands are fed off into the two peak processing paths.

### Look-ahead limiter

The DSPXtra-HD look-ahead limiter is one of the most sophisticated available in a broadcast processor. Processing is performed in three bands for maximum transparency and clarity followed by additional precision look-ahead limiter. A lowpass filter is provided to compensate for the artifacts generated in the codec following DSPXtra-HD.

### Output selection, processing and routing

The DSPXtra-HD allows the user to select where each processing path is routed to and provides output level controls. The digital output sample rate can also be configured to a variety of settings.



DSPXtra-HD PROCESSING BLOCK DIAGRAM

## **MENU SYSTEM OVERVIEW**

This section presents an overview of the four main menus and their submenus and any parameters that are contained in them.

**INPUT:** Contains all the controls that affect input selection, level control and signal conditioning.

**PROCESS:** Contains all the controls that affect the processing.

**OUTPUT:** Contains all the controls that affect output selection, level control and signal conditioning.

**SCHEDULE:** Contains all the real time clock controls for switching presets (Dayparting).

**SYSTEM:** Contains all the system controls (non processing) such as remote control and security.

## USER LEVELS

The DSPXtra-HD has the ability to restrict the use of some of the more advanced processing parameters by selecting a beginner mode. This mode control is located at the top level of the processing menu tree, the options being beginner and advanced.

Selecting advanced will allow access to every menu and processing parameter while beginner 'hides' certain processing controls and menus. The menu structure diagram on the next page illustrates what is hidden when in beginner mode.

It is worth noting that when in beginner mode the 'HIDDEN' processing parameters are still active, but defaulted to a safe value that works well with the processing parameters that are left on and selectable. Another reason for doing this is to stop the scenario of setting an advanced control to a certain value and then switching to beginner mode and having no way of fully controlling the processing because the 'HIDDEN' parameter is left in a state that makes the audio sound bad. One downside of this is that if you make a preset in advanced mode and then switch to beginner mode the sound will change as the hidden parameters default to safe values. To avoid issues like this we suggest starting with beginner mode and then switching to advanced mode to refine the preset if necessary. We see no reason why you would want to do this in reverse but we thought it best to mention it so that when switching to beginner mode you don't wonder why you have just lost your preset that you just spent an hour working on. One good reason for using the save preset facility at various stages when making a new preset!

## ARIANE:

The Ariane leveller has three modes of operation, Matrix, Stereo and bypassed. While we are discussing the menu system we thought it prudent to mention that the displayed processing options for the Ariane leveller depend on what mode is selected. The DSPXtra-HD only displays the controls for the currently selected option.

# MENU STRUCTURE

USER LEVEL (Disables some menu options when beginner mode selected)

## MENU: INPUT

- INPUT SOURCE
- STEREO MODE
- ANALOG INPUT LEVEL
- RIGHT TRIM
- INPUT FAIL SWITCH
- HIGH PASS FILTER
- PHASE ROTATOR

## MENU: PROCESS

### MENU:ARIANE

- MODE (Enables one of three Ariane modes below)
- MENU:XOVER (Advanced mode)
  - B1/B2 XOVER
  - B2/B3 XOVER
  - B3/B4 XOVER
- MENU: MATRIX (Ariane mode 1)
  - MENU:L+R
    - IDR
    - GATE (Advanced mode)
    - B1RELEASE
    - B2RELEASE
    - B3RELEASE
    - B4RELEASE
  - MENU:L-R
    - IDR
    - GATE (Advanced mode)
    - B1RELEASE
    - B2RELEASE
    - B3RELEASE
    - B4RELEASE
  - ATTACK (Advanced mode)
  - B1<B2 COUPLING (Advanced mode)
  - B3>B4 COUPLING (Advanced mode)
  - B1 L-R MUTE (Advanced mode)
  - L-R RESTRICTION (Advanced mode)
  - L-R INJECTION
- MENU: STEREO (Ariane mode 2)
  - IDR
  - GATE
  - ATTACK (Advanced mode)
  - B1RELEASE
  - B2RELEASE
  - B3RELEASE
  - B4RELEASE
  - B1<B2 COUPLING (Advanced mode)
  - B3>B4 COUPLING (Advanced mode)
  - CHANNEL COUPLING (Advanced mode)

### MENU:BYPASS (Ariane mode 3)

## MENU: ENHANCE

- DEEP BASS
- BASS TUNE
- PEAKING BASS FREQ
- PEAKING BASS Q
- PEAKING BASS GAIN

## MENU: MULTI-BAND LIMITERS

- MASTER LIMITER DRIVE
- MENUS: BANDS 1-6
  - DRIVE
  - THRESHOLD (Advanced mode)
  - LIMITER ATTACK
  - LIMITER DECAY (Advanced mode)
  - COMPRESSOR ATTACK (Advanced mode)
  - COMPRESSOR DECAY
  - HOLD (Advanced mode)
  - DELAY (Advanced mode)

## MENU: MIXER (Advanced mode)

- BAND1MIX
- BAND2MIX
- BAND3MIX
- BAND4MIX
- BAND5MIX
- BAND6MIX

## MENU: MB LOOKAHEAD

- DRIVE
- SHELF EQ
- B1-2 ATTACK (Advanced mode)
- B1-2 DECAY (Advanced mode)
- B3 ATTACK (Advanced mode)

- B3 DECAY (Advanced mode)
- B4 ATTACK (Advanced mode)
- B4 DECAY (Advanced mode)
- MENU: FINAL LOOK-AHEAD
  - LP FILTER
  - DRIVE

## MENU: OUTPUT

- MENU: ANALOG
  - OUTPUT LEVEL
- MENU: DIGITAL
  - OUTPUT LEVEL
  - SAMPLE RATE
  - HEADPHONE LEVEL

## MENU: SCHEDULE

(X) REPRESENTS 0-7

- TIME
- DAYPARTING ON/OFF
- DAYPARTS 1-4
  - DP(X) ON/OFF
  - DP(X) START TIME
  - DP(X) TIME ON (LENGTH)
- DAYPARTS 5-8
  - DP(X) ON/OFF
  - DP(X) START TIME
  - DP(X) TIME ON (LENGTH)

## MENU: SYSTEM

- LCD CONTRAST
- LED CONTRAST
- TRIGGER PORT
- FULL LOCK
- OUTPUT LOCK
- SERIAL PORT
- REMOTE SOURCE
- MENU: LAN CONFIG
  - IP
  - DG (DEFAULT GATEWAY)
  - SM (SUBNET)
  - MA1 (MAC ADDRESS PART1)
  - MA2 (MAC ADDRESS PART2)
  - PORT
- MENU: ABOUT
  - VERSION
  - CONCEPT
  - HARDWARE
  - CTRL SYS.
  - PROCESSING
- BOOTLOAD

## PROCESSING PARAMETERS

**User Level:** This parameter allows you to select between beginner and advanced modes. Beginner mode restricts access to certain menus and controls.

The **'INPUT'** menu contains all of the options and parameters relating to the control and conditioning of the audio inputs.

**Source** This parameter allows you to select the between the analog and digital inputs as the source for the processing.

**Input Mode:** This parameter allows you to select different mono options as well as the default Stereo option. There is also the ability to swap the left/right channels.

**Analogue Input A/D Clip Level:** Allows you to set the analogue input level with reference to the DSPXtra-HD's A/D converter. This would normally be set to +24dBu if driving the DSPXtra-HD from professional audio equipment. This would translate to the wide-band AGC being driven to 12dB gain reduction by a 0VU (+4dBu) audio signal. Make sure that the input audio meters NEVER show clipping under any conditions.

**Right Trim:** This parameter allows you to adjust the right channels gain in small increments to BALANCE out any small gain discrepancies between the left and right channels. The range is +/- 3dB.

**Input Fail Switch:** This parameter allows you to turn on the silence detection system and specify a silence time. When enabled this parameter will switch the input source from the primary to the secondary source when a predetermined length of silence occurs.

**High pass filter:** This parameter allows you to select from a variety of high pass filters. You can select from 20Hz, 30Hz, 40Hz, 50Hz and 60Hz. You also have the ability to bypass the high pass filter with the 'OFF' option. The high pass filter can be used to reduce rumble on vinyl or can be effective in removing low frequency energy that is imperceptible but could cause significant band 1 gain reduction.

**Phase Rotator:** This parameter if enabled will help to reduce vocal distortion in aggressive presets by reducing asymmetry in the voice which would otherwise put more workload on the final peak limiting stages. We recommend enabling this option if you want competitive loudness; otherwise leave it off as the phase rotation process does colour the sound slightly, although this coloration is often used for artistic effect.

The **'PROCESS'** menu allows access to all of the processing blocks that make up the DSPXtra-HD. There are only sub-menus inside the **'PROCESS'** menu. The submenus are laid out in the same configuration as the signal path through the DSPXtra-HD.

### **'ARIANE AGC'**

The ARIANE AGC in the DSPXtra-HD like any automatic gain control device is designed to correct input level fluctuations and provide a consistent level to the other DSPXtra-HD processing blocks that follow it. The Ariane AGC in the DSPXtra-HD is designed to operate in an undetectable, unobtrusive manner similar to a trained operator controlling a mixing desk. The Ariane AGC can operate in 'sum and difference' matrix mode or conventional stereo mode, with or without channel coupling.

**Mode:** This parameter selects which mode of operation to use, Matrix, Stereo or Bypassed.

The **'XOVER'** menu provides the user with another tool in creating their sonic signature.

**'B1-2 XOVER'** Sets the B1-2 Xover point to one of five frequencies.

**'B2-3 XOVER'** Sets the B2-3 Xover point to one of five frequencies.

**'B3-4 XOVER'** Sets the B3-4 Xover point to one of five frequencies.

The **'MATRIX'** menu contains the controls for the matrix mode of operation (SHOWN IN MATRIX MODE)

The **'L+R'** menu contains the controls for the L+R (SUM) section of the matrix mode of operation.

**'IDR'** Sets the range in dB of the window gating system. The processing will turn off while the

audio is inside this window. The window can be set between 1 and 15dB.

**'GATE'** The gate function prevents 'suck-up' of noise during periods of silence or low level audio. The level can be adjusted to turn on when the input drops to a level from -15dB to -35dB below the reference level. After 10 seconds of being gated the gain reduction will return to the 0dB reference point.

**'B1 RELEASE'** Controls the release/decay rate of band 1 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B2 RELEASE'** Controls the release/decay rate of band 2 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B3 RELEASE'** Controls the release/decay rate of band 3 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B4 RELEASE'** Controls the release/decay rate of band 4 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

The **'L-R'** menu contains the controls for the L-R (difference) section of the matrix mode of operation

**'IDR'** Sets the range in dB of the window gating system. The processing will turn off while the audio is inside this window. The window can be set between 1 and 15dB.

**'GATE'** The gate function prevents 'suck-up' of noise during periods of silence or low level audio. The level can be adjusted to turn on when the input drops to a level from -15dB to -35dB below the reference level. After 10 seconds of being gated the gain reduction will return to the 0dB reference point.

**'B1 RELEASE'** Controls the release/decay rate of band 1 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B2 RELEASE'** Controls the release/decay rate of band 2 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B3 RELEASE'** Controls the release/decay rate of band 3 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B4 RELEASE'** Controls the release/decay rate of band 4 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**ATTACK** Controls the attack rate of the Ariane AGC, the time the AGC takes to respond to an increase of input level. The ATTACK time can be varied between 1 and 10 with smaller numbers indicating faster attack time.

**'B1<B2 COUPLING'** Sets the amount of coupling between the band 2 gain reduction system and the band 1 gain reduction system. This is adjustable from 0dB to 15dB with an option for no coupling (OFF) This coupling is uni-directional, band 2 into 1.

**'B3<B4 COUPLING'** Sets the amount of coupling between the band 3 gain reduction system and the band 4 gain reduction system. This is adjustable from 0dB to 15dB with an option for no coupling (OFF) This coupling is uni-directional, band 3 into 4.

**'B1 L-R MUTE'** Mutes the output from the B1 L-R processing stage. This can be set to ON or OFF. Can help reduce multi-path when enabled. More on this in this manual.

**'L-R RESTRICTION'** couples the L-R control system to the L+R control system so that the L-R gain

reduction can not increase any more than a certain dB above the L+R gain reduction. This prevents build up of an excessive amount of stereo content. This is adjustable from 0dB to 15dB with an option for no coupling (OFF)

**'L-R INJECTION'** Sets the level of the L-R output in relation to the L+R output. This control can set the maximum level of L-R energy in the output audio. This is adjustable from 0dB to -15dB with an option for no L-R (OFF) which creates a mono output signal.

The **'STEREO'** menu contains the controls for the stereo mode of operation (SHOWN IN STEREO MODE)

**'IDR'** Sets the range in dB of the window gating system. The processing will turn off while the audio is inside this window. The window can be set between 1 and 15dB.

**'GATE'** The gate function prevents 'suck-up' of noise during periods of silence or low level audio. The level can be adjusted to turn on when the input drops to a level from -15dB to -35dB below the reference level. After 10 seconds of being gated the gain reduction will return to the 0dB reference point.

**ATTACK** Controls the attack rate of the Ariane AGC, the time the AGC takes to respond to an increase of input level. The ATTACK time can be varied between 1 and 10 with smaller numbers indicating faster attack time.

**'B1 RELEASE'** Controls the release/decay rate of band 1 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B2 RELEASE'** Controls the release/decay rate of band 2 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B3 RELEASE'** Controls the release/decay rate of band 3 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B4 RELEASE'** Controls the release/decay rate of band 4 of the AGC, the time the AGC takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 with smaller numbers indicating faster release times.

**'B1<B2 COUPLING'** Sets the amount of coupling between the band 2 gain reduction system and the band 1 gain reduction system. This is adjustable from 0dB to 15dB with an option for no coupling (OFF) This coupling is uni-directional, band 2 into 1.

**'B3<B4 COUPLING'** Sets the amount of coupling between the band 3 gain reduction system and the band 4 gain reduction system. This is adjustable from 0dB to 15dB with an option for no coupling (OFF) This coupling is uni-directional, band 3 into 4.

**'CHANNEL COUPLING'** Couples the Left channel control system to the Right channel control system so that the either channels gain reduction can not increase any more than a certain dB above the other channel gain reduction, helping to preserve stereo balance. This is adjustable from 0dB to 15dB with an option for no coupling (OFF)

The **'BYPASS'** menu contains no entries (SHOWN IN BYPASS MODE)

The **'ENHANCE'** menu contains the low frequency enhancement filters which are used to provide bass enhancement and help overcome the bass reduction effect of multi-band compression.

**'DEEP BASS'** A 12dB/octave shelving bass equalizer that provides between 0 and 12dB of bass boost. Use this control with caution as too much low frequency bass boost can cause loss of mid-bass because the very low frequency bass, often inaudible on many receivers dominates the gain reduction of the BAND 1 AGC and limiter. A setting of 6dB is a good compromise and starting point.

**'BASS TUNE'** Adjusts multiple points in the dynamics control system, allowing you to control the 'flavour' of the bass.

**'PEAKING BASS EQUALIZER'** A pseudo parametric style bass equalizer control that allows you to sweet tune the bass. Four frequencies, amplitudes and Q's are provided giving you 64 different bass



curves to select from. Frequencies selectable: 60Hz, 76Hz, 95Hz and 120Hz. Q's selectable: 0.4, 1, 2 and 4. Gains selectable: 0, 1.5dB, 3dB, 4.5dB, 6dB.

The **'MULTI-BAND LIMITERS'** peak limit each of the bands to prevent distortion in the processors final peak control system.

**'MASTER LIMITER DRIVE'** Sets the drive into the multi-band xover that generates the six bands of audio. This control allows a +/- 6dB adjustment.

#### **'B1-6'**

**'DRIVE'** Controls the drive into the limiter. The drive can be increased or decreased by up to 12dB.

**'PEAK ATTACK'** Controls the attack rate of the limiter, the time the limiter takes to respond to an increase of input level. The attack time can be varied between 1 and 10 which corresponds to 1mS to 200mS on a semi-exponential scale.

**'PEAK DECAY'** Controls the peak release/decay rate of the limiter, the time the limiter takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 which corresponds to 10mS to 1000mS on a semi-exponential scale.

**'AVG ATTACK'** Controls the average attack rate of the limiter. This control is a modifier control that scales from the peak attack control. The attack time can be varied between 1 and 10 which correspond to peak attack time / 2 to peak attack time / 1000 on a semi-exponential scale. The AVG attack control determines the dynamics of the dual time constant system and how control is shared between the peak and average circuits.

**'AVG DECAY'** Controls the average release/decay rate of the limiter, the time the limiter takes to respond to a decrease of input level. The DECAY time can be varied between 1 and 10 which corresponds to 200mS to 5000mS on a semi-exponential scale.

**'HOLD THRESHOLD'** The HOLD function prevents 'suck-up' of noise and helps reduce IM distortion by allowing the limiter to rest during periods of silence or low level audio. The level can be adjusted to turn on when the input drops to a level from -20dB to -40dB. The HOLD can also be switched off.

**'DELAY'** This function provides a delay before the peak decay circuit is activated. This can be used to reduce distortion when faster release time constants are employed by reducing the gain hunting between neighbouring transients. The range available is 1mS-500mS on a semi-exponential scale.

**'MIXER'** menu. Each band can be adjusted over a small range to provide small EQ changes. These controls are limited in range to prevent excessive drive into the final peak limiting stages and excess distortion being introduced. A solo mode is provided to aid in the setting up of parameters.

**BAND1MIX:** -3dB to +3dB of level adjustment is available.

**BAND2MIX:** -3dB to +3dB of level adjustment is available.

**BAND3MIX:** -3dB to +3dB of level adjustment is available.

**BAND4MIX:** -3dB to +3dB of level adjustment is available.

**BAND5MIX:** -3dB to +3dB of level adjustment is available.

**BAND6MIX:** -3dB to +3dB of level adjustment is available.

The **'MB LOOKAHEAD'** menu contains the controls that affect the look-ahead limiter.

**'DRIVE'** Controls the drive into the multi-band look-ahead limiter.

**'B1-2 ATTACK'** Controls the average attack rate of the B1-2 look-ahead limiter. This defines the energy distributed into the secondary time constant circuit. The attack time can be varied between 1 and 10 which corresponds to 10mS to 2000mS on a semi-exponential scale.

**'B1-2 DECAY'** Controls the average decay rate of the B1-2 look-ahead limiter. This defines the decay rate of the secondary time constant circuit. The decay time can be varied between 1 and 10 which corre-

sponds to 10mS to 2000mS on a semi-exponential scale.

**'B3 ATTACK'** Controls the average attack rate of the B3 look-ahead limiter. This defines the energy distributed into the secondary time constant circuit. The attack time can be varied between 1 and 10 which corresponds to 10mS to 2000mS on a semi-exponential scale.

**'B3 DECAY'** Controls the average decay rate of the B3 look-ahead limiter. This defines the decay rate of the secondary time constant circuit. The decay time can be varied between 1 and 10 which corresponds to 10mS to 2000mS on a semi-exponential scale.

**'B4 ATTACK'** Controls the average attack rate of the B4 look-ahead limiter. This defines the energy distributed into the secondary time constant circuit. The attack time can be varied between 1 and 10 which corresponds to 10mS to 2000mS on a semi-exponential scale.

**'B4 DECAY'** Controls the average decay rate of the B4 look-ahead limiter. This defines the decay rate of the secondary time constant circuit. The decay time can be varied between 1 and 10 which corresponds to 10mS to 2000mS on a semi-exponential scale.

**'LP FILTER'** A high frequency filter adjustable over a range of 4-20 kHz. This control can be used to tame the artifacts of subsequent perceptual coding, especially when used with very low bitrates.

The **'FINAL LOOKAHEAD'** menu contains the controls that affect the final look-ahead limiter.

**'LP FILTER'** A high frequency filter adjustable over a range of 4-20 kHz. This control can be used to tame the artifacts of subsequent perceptual coding, especially when used with very low bitrates.

**'DRIVE'** Controls the drive into the final look-ahead limiter.

**The 'OUTPUT' menu contains all of the options and parameters relating to the control and conditioning of the audio outputs.**

The **'ANALOG'** menu contains the controls relevant to the analogue outputs.

**'OUTPUT LEVEL'** Controls the output level of the analogue output. Range is -12dBu to +24dBu.

The **'DIGITAL'** menu contains the controls relevant to the AES/EBU digital outputs.

**'OUTPUT LEVEL'** Controls the peak output level of the digital output. Range is -12dBFS to 0dBFS.

**'RATE'** This parameter sets the output sampling rate for the AES/EBU digital output. The available rates are 32 KHz, 44.1 KHz, 48 KHz, follow digital input rate and follow external sync rate.

The **'HEADPHONE LEVEL'** sets the output level of the front panel headphone port. Range is 0 to 25 with higher numbers equating to more volume. The headphone port follows the output mode and de-emphasis settings of the analogue outputs.

**The 'SCHEDULE' menu contains all the controls for the dayparting (REAL TIME CLOCK) preset switching.**

**'TIME'** Sets the time and day of the DSPXtra-HD's Real Time Clock.

**'DAYPARTING ON/OFF'** Enables or disables the scheduling.

**'DP(X) ON/OFF'** Enables or disables an individual daypart schedule.

**'DP(X)'** Sets the preset to switch to when this daypart is triggered.

**'DP(X) START'** Sets the start time day and time of the daypart. There is also an 'ALL days' option.

**'DP(X) LENGTH'** Sets the length in minutes that the daypart will run for.

**The 'SYSTEM' menu contains all the system controls (non processing) such as remote control and security**

**'LCD CONTRAST'** Sets the contrast of the front panel LCD screen. The range is 0 to 25.

**'LED CONTRAST'** Sets the brightness of the front panel LED meters. The range is 1/4, 1/2, 3/4 and full brightness.

**'TRIGGER PORT'** This enables or disables the rear panel trigger (remote) port. The options are enabled and disabled. More information is available in the trigger port section of this manual.

**'FULL LOCK'** This enables or disables the security code lock. The options are enabled and disabled. More information is available in the code lock section of this manual.

**'OUTPUT LOCK'** This enables or disables the output code lock. The options are enabled and disabled. More information is available in the code lock section of this manual.

**'REMOTE SOURCE'** This selects the serial/USB port or the NET/LAN port as the remote control method. The default option is Off.

The **'LAN CONFIG'** menu contains the controls relevant to the LAN/NET port.

**'IP'** Sets the IP address of the LAN port.

**'DG'** Sets the default gateway of the LAN port.

**'SM'** Sets the subnet mask of the LAN port.

**'MA1'** Sets the first half of the MAC address of the LAN port.

**'MA2'** Sets the second half of the MAC address of the LAN port.

**'PORT'** Sets the port number of the LAN port.

**'ABOUT'** DSPXtra-HD version number and design credits.

**'BOOTLOAD'** This option is used to FLASH update the software and firmware inside the DSPXtra-HD. Further information on using this option is described in the documentation supplied with the upgrade.

# SETTING UP THE PROCESSING ON THE DSPXTRA

This section has more detailed information on setting up the DSPXtra's processing.

Because the DSPXtra contains another product in its own right, the Ariane, we have decided to start with discussing that section of the processing and its setup. This also seems logical because it is the first part of the processing chain after the audio has been routed through the high pass and phase rotating filters which we will get out of the way below before moving on to the background, workings and setup of the Ariane.

## High pass filter

The high-pass filter has five selectable cut off frequencies and a bypass option. Most users will bypass the high-pass stage but there are several cases where enabling the high-pass has an advantage. The first of those is stations that mainly play vinyl recordings. Vinyl recordings can suffer from low frequency rumble and the high pass filter reduces the effect of these low frequency rumbles. By removing the very low frequency content from the program material you can often improve the rest of the bass sound from the processor. The theory is that most people can't hear or speakers can't produce the very low frequency bass. By removing this sub-sonic bass more room is made in the processed waveform for frequencies that can be heard. Another reason that is given is that this very low frequency bass can dominate the band 1 AGC and limiter, especially after bass enhancement has been carried out. The low frequency shelving filters used in processors have much higher gains at 20Hz than say 50Hz where most people can hear and speakers reproduce bass. The processing stages will respond to this amplified 20Hz content even though most people won't ever hear it when listening to your radio station.

## Phase rotator

This parameter if enabled will help to reduce vocal distortion in aggressive presets by reducing asymmetry in the voice which would otherwise put more workload on the final limiting stages. Human speech (particularly male) can be very asymmetric compared to music and the phase rotator helps to bring symmetry to the audio waveforms. We recommend enabling this option if you are after maximum loudness; Conservative formats such as classical may prefer to leave it off as the phase rotation process does colour the sound slightly, although this coloration is often used for artistic effect.

## The Ariane RMS leveller

The first decision you need to make is which operating mode to use. The Ariane has three modes: Stereo, Matrix and bypass. Each mode has its own capabilities and purposes.

**Stereo** In the Stereo mode, the Ariane is a traditional Left/ Right two channel AGC, with a single set of adjustments for both channels. The two channels' dynamic operation can be set such that control of both channels is coupled together completely, allowed to operate independently, or with a user defined amount of inter-channel control coupling.

**Matrix** In Matrix mode, the audio is converted before processing to a Left-plus-Right or 'mono sum' signal, and a corresponding Left-minus-Right or 'difference' signal. Matrix mode allows for stereo enhancement and optimization for any stereo broadcast medium (such as stereo FM, AM or Television) that ultimately is transmitted in sum and difference. There are separate controls for the sum and difference signals. After processing, the Ariane's matrix signals are reconverted to Left and Right stereo audio for output.

Once you have determined the appropriate operating mode to suit your needs, you could simply load one of the Ariane's factory presets to get rolling (there are several presets for each mode). You can then start your fine tuning... and the fun begins!

## Main Processing Controls

All three operating modes have a common set of controls that allow you to adjust the Ariane to best fit your needs, plus a few controls that are unique to each mode, to further refine that mode's operation.

The Ariane's basic processing adjustments are the IDR, Gate and Release settings, which are common to all three modes. They set up the general static operating parameters which the Ariane will use to determine how best to dynamically control your audio.

## IDR

IDR or Instantaneous Dynamic Range, is the main setting to adjust the amount of control action you wish your Ariane to have. The IDR setting is the 'size' of the RMS energy window in the windowing-release system. As long as the short-term variations of the RMS energy in the audio remain within this range, the Ariane will stop making any changes to the gain in that band. For this reason, the IDR control is the most powerful and sonically-influential setting in the Ariane. IDR can be adjusted over a range of from 1 to 15 decibels in one-dB steps, with the lower numbers (a smaller 'window') being more aggressive and the higher numbers (bigger 'window')

sounding more natural. An IDR setting of 5 dB is a good compromise between control inaudibility versus a useful amount of control.

### Gate

The Gate threshold control determines the point below which the Ariane will not seek to increase audio gain. Audio that falls above this threshold is considered valid, and fair game to be brought up toward operating level. The Gate adjustment can be set anywhere from a low of -35 dB below 0dB nominal reference operating level, to as high as -15 dB below 0 dB, in one-dB steps. The higher the Gate threshold, the less any low level material will be boosted.

### Attack

The Attack time adjustment sets how quickly the Ariane will decrease gain to the incoming signals that go above the upper threshold of the window. The adjustment is from 1 to 10, with 1 being the fastest, and 10 slowest reaction time. The attack time is the same for all bands.

### Release

The Release time adjustment sets how quickly the Ariane will add gain to the incoming audio signals that go below the lower threshold of the window (but above the Gate threshold). The adjustment is from 1 to 10, with 1 being more aggressive, and 10 being very open and natural. The default setting is 4. Although Release settings can be set differently for each band, we recommend keeping them all at the same value to ensure good dynamic and spectral tracking between bands.

### Band-to-Band and Channel-to-Channel Coupling/Restriction

The flexibility of the Ariane is enhanced by various coupling and restriction controls. The Low frequency band (B1) can be restricted from ever adding more gain than is user-specified, with respect to the dynamic gain of the Low Mid band (B2), using the "B1<B2" menu setting. Likewise, using the "B3>B4" menu setting, the High frequency band (B4) can be restricted from adding more gain than is user-specified, with respect to the dynamic gain of the High Mid (B3) band. When enabled, the restriction controls make it impossible for the Low and High bands to ever operate with more than a specified amount of gain beyond that of the adjacent band (note: they are always free to operate with less gain). The restrictions are one-way, setting a relative limit on how much gain can be added in the extreme bands (B1, B4), all the while the dynamic gain controls in the adjacent bands (B2, B3) are unaffected. A good example to explain the need and setup of such controls would be if a radio station regularly makes remote broadcasts near heavy traffic, which typically has an excess of very low frequencies. With the B1<B2 restriction turned off, the Low band (B1) is free to bring up the level of any rumble that may occur above the Gate threshold. Normally, the Gate threshold can be set to -25 dB or even lower, and normal background noise is not a problem. But should the traffic rumble go above the Gate threshold it would be considered by the Ariane to be valid audio, and could potentially be increased nearly to the level of the other audio. This is a very unnatural-sounding and potentially problematic situation. By setting the B1<B2 coupling to, for instance, 4 dB, the gain in the Low Band (B1) will only go as high as 4 dB greater than the gain occurring in the Low Mid Band (B2), but no more. As long as this condition continues, the B1 gain will track the dynamic gain of B2, plus 4 dB, but no greater. And having 4 dB more B1 gain than in the B2 band will not sound nearly so noticeable as 15 to 20 dB more gain! The settings for the B1<B2 and B3>B4 band gain restrictions can be set to 'Off' or to allow anywhere from 15 dB to full coupling ('0 dB').

### Mode-specific Controls

There are various coupling/restriction and other miscellaneous settings that are mode-specific, as follows:

#### Matrix mode:

**L-R B1 Mute** By reducing the level of low frequencies in the L-R channel, a subjective improvement in cleanliness can be made. You essentially give up nothing: human hearing is decidedly weak in detecting low frequency directional cues, and any reduction of material in that region will most likely not be missed by even the most critical listeners.

**L-R Restriction** You can control the L-R band gains with reference to the L+R bands similar to the way the Low band can be coupled to the Low Mid band. The settings are "Off", where the L-R is totally independent of the L+R, or any degree of coupling, from 15 dB up to 0 dB where the L-R gains are completely locked together to the L+R's.

**L-R Injection** The amount of processed L-R audio can be controlled before it is matrixed with the L+R to re-create Left and Right at the Ariane's outputs. Normally, this setting should be about -5 dB, but it can be set higher for a bigger stereo effect, or lower for reduced coding artifacts. As you might guess, this setting is somewhat subjective.

#### Stereo

**Channel Coupling/Restriction** The two channels can be coupled together, in an adjustable amount, from full coupling ("0 dB") to the relatively independent 15 dB, or "Off," no coupling at all. The upside is that you can control channel imbalances. The downside is that you risk occasionally shifting the centre

image if Release and/or IDR controls are set up aggressively.

### **Bass enhancement**

The frequency contouring effect of multi-band audio processors often leaves the bass lacking a little. The summation of the bands tends to give a boost to the presence frequencies and leaves the bass sounding a little thin. This effect can be compensated somewhat by enhancing the bass prior to multi-band processing.

The DSPXtra has two types of bass enhancement filter. A low frequency shelving boost filter and a peaking bass equaliser.

The shelving filter has a 12dB/octave slope and can be adjusted to provide between 0 and 12dB of bass boost. Use this control with caution as too much low frequency boost can cause loss of mid-bass because the low bass, often inaudible on many receivers dominates the gain reduction of the BAND 1 AGC and limiter. A setting of 6dB is a good compromise and starting point.

The peaking bass equalizer is a pseudo parametric style bass equalizer control that allows you to sweet tune the bass. Four frequencies, amplitudes and Q's are provided giving you 64 different bass curves to select from. Frequencies selectable: 60Hz, 76Hz, 95Hz and 120Hz. Q's selectable: 0.4, 1, 2 and 4. Gains selectable: 0, 1.5dB, 3dB, 4.5dB, 6dB. A starting setting of 95Hz, Q of 1 and gain of 4.5dB warms the bass up quite nicely but you are free to experiment to get the bass sound you're after.

Additionally, bass tune control allows control of the 'flavour' of the bass by adjusting various points in the bass dynamic control system.

### **Multi-band limiters**

The multi-band limiters drive can be adjusted over a +/- 12dB range. Increasing the drive will increase the level of limiting and with it on air loudness, above a certain level of drive no more loudness will be obtained and all that will happen is you will generate higher levels of IM distortion and the sound will take on a busy packed texture. You may also observe higher levels of high frequency noise when the band 3 and 4 drives are increased. We don't usually find much use for drives above +6dB but more may be required if other settings are adjusted to compensate. In any case, observe the peak limiter meters for a good indication of how much drive to use. We don't recommend more than 12dB of gain reduction especially on bands 2, 3 and 4. Gain reductions of 4-8dB are a good compromise between loudness and quality.

The multi-band limiters have a threshold control and care should be taken when adjusting it as distortion in the following peak limiting stages can result if the threshold is set too high. The range is +/- 6dB.

The multi-band limiters in the DSPXtra are of the dual time constant variety. There is an attack and decay to handle the peaks and an attack and decay to handle the average level of limiting. Understanding how the two time constants interact is imperative if you want to make major changes to how each bands limiter reacts. We have included some scope screen captures to illustrate things a little clearer. The peak and average function can clearly be seen in the images.

Traditionally audio limiters have two time constants, an attack, the time it takes the limiter to respond to a signal above the threshold and a decay or release which is the time it takes to respond to a drop in level. In a traditional audio limiter the attack time is usually set to somewhere in the region of a few milliseconds and the decay time considerably longer at somewhere in the hundreds of milliseconds. This is not the most optimum solution because transients that last only a few milliseconds will reduce the level of the waveform for hundreds of milliseconds, reducing loudness and creating audible pumping effects.

The solution is multiple time constants where one set of time constants can be set to handle the fast peaks and another to handle the average level of limiting. Fast transients will release in a faster less noticeable way and won't punch holes in the sound in a way that single time constant limiters can. The secondary slower time constant circuit will not have much effect on the audio waveform when hit with a transient because the higher attack time, generally in the hundreds of milliseconds will not allow a build up of energy. In the case of a sustained envelope of audio above the threshold the multiple time constant will attack as normal with the peak time constant but the sustained energy will also charge the secondary slower circuit. When the audio energy falls away and the circuit goes into release the peak decay will dominate until it reaches a point where it hands over to the slower secondary time constant for a slower rate of decay. The illustrations show this to good effect, where transients have a fast release but multiple or sustained transients build up energy in the secondary circuit which acts as a platform for the peak to release to. The secondary circuit's platform can be thought of as the average level of limiting. Having this fast peak responding circuit ride on top of the average circuit creates many advantages, limiter transparency, less chance of pumping and greater loudness. By setting the time constants appropriately we can have the multiple time constant based detectors work as peak handling, average handling or the opti-

imum setting of a balance of the two.

The peak attack time should be set to the desired attack time required from that limiter. The range is 1-10 which corresponds to 1 to 200mS on an exponential scale. The peak decay time should be set to the desired peak decay time required for transients. The range is 1-10 which corresponds to a decay time of 10 to 1000mS.

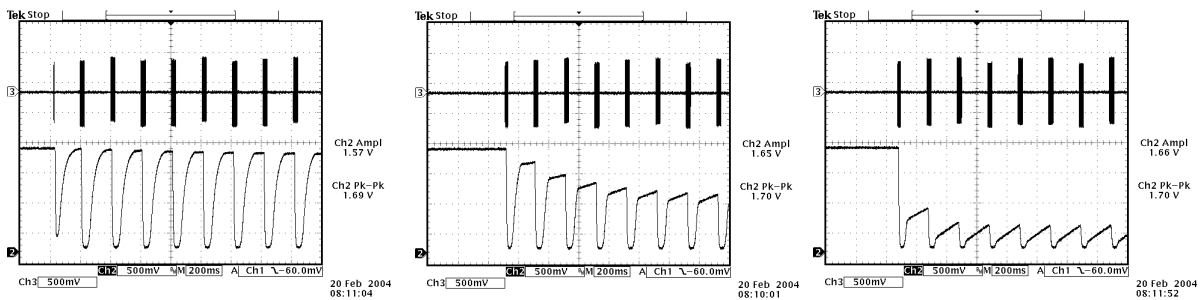
The average attack time is perhaps the most important control in the dual time constant detector as it sets the balance between peak and average energy in the detector. With smaller numbers more energy is transferred into the average circuit and a higher platform level is created so more time will be spent releasing at the slower average rate. Higher numbers offer slower attack times for the averaging part of the detector and this has the effect of lowering the average platform level and allowing the peak part of the circuit to dominate with its faster release times.

The Average decay time can usually be viewed as the nominal release time of the detector, similar to a standard single time constant limiters release time.

To recap, the peak attack time and average decay time play the same sort of role as that of a standard conventional single time constant based limiter.

The peak decay time sets the decay time for fast usually inaudible transients and the average attack time sets the ratio of peak to average control and defines the position of the platform that the peak circuit releases to.

#### Limiter control signals response to tone bursts

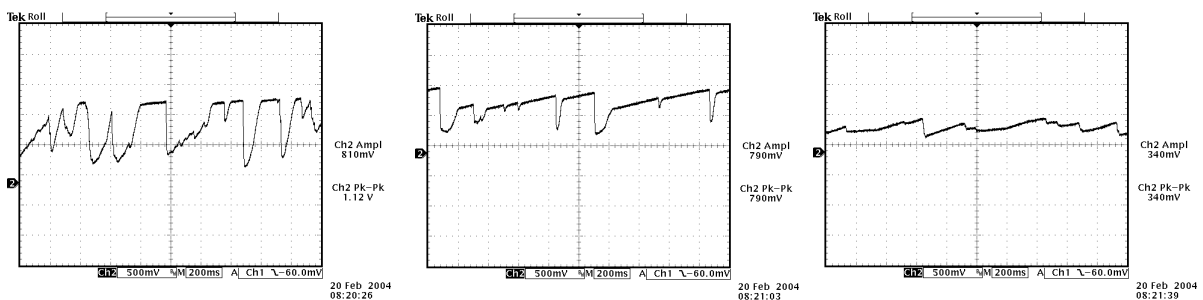


Peak time constants dominating control due to a very high setting of average attack

Peak time constants dominating to a lesser degree due to a high setting of average attack

Peak time constants dominating to a much lesser degree due to a lower setting of average attack

#### Limiter control signals response to program material



Peak time constants dominating control due to a very high setting of average attack

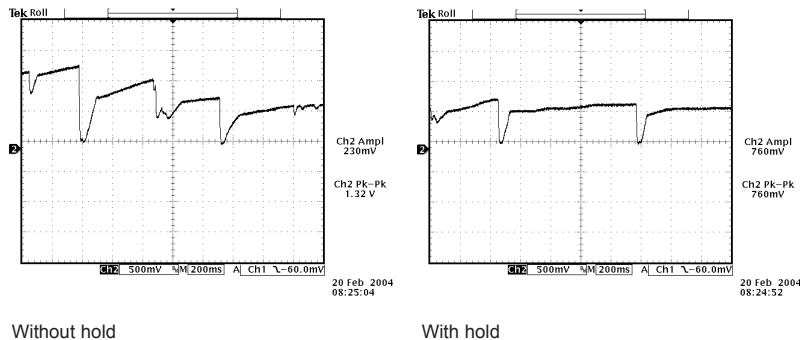
Peak time constants dominating to a lesser degree due to a high setting of average attack

Peak time constants dominating to a much lesser degree due to a lower setting of average attack

#### Hold

The multi-band limiters have an extra mechanism called HOLD which works in the same way as the GATE control in the AGC stages. Unlike the GATE control the HOLD control has no extra associated controls like return to rest levels and speeds. The hold feature when triggered will HOLD the average platform level at its last value before the hold was triggered. This feature can help reduce IM distortion, reduce pumping effects and avoid suck up of noise. The HOLD feature can be adjusted over a level of -20 dB to -40dB with an option for 'OFF'.

Unless you have a specific reason to turn it off we recommend a setting of around -32dB but this may need to be increased or decreased depending on drive levels into the limiter and how much effect you want the hold control to have on the audio. If set too low you will rob yourself off some loudness, if set too high you will lose some of the benefits of the mechanism by letting the limiter release too far before gating.



### Delay

The delay control sets the amount of time the processor will hold the gain reduction for before releasing at the peak decay rate. Careful setting of this control allows us to speed up the peak decay time without introducing the audible effects of faster decay times. Setting this control to 1 is a negligible amount of delay before decay which is equivalent to the delay being switched out of circuit. As you go up through the bands you will need lower delay numbers to avoid causing audible problems being introduced by the delay circuit. We have found that bands 3 and 4 work well with settings of 2-4, band 2 settings for 3-6 and band 1 settings of 5-8. Setting the control to 10 introduces almost half a second of delay before decay so make sure you don't use the higher settings of this control on anything other than band 1.

If you are unsure about this control we suggest you set it to 1 to turn it OFF.

### The mixer

The post limiters mixer in the DSPXtra is not strictly a mixer but a band output level control where small EQ changes can be made. It has been called the mixer as most other processors have a mixer at this position and our VIRTUAL mixer does the same job. Bands 1 and 2 do mix together at this point, so do 3 and 4 and also 5 and 6 so I suppose you could call it a half-mixer. The six bands have become three.

Be careful when making large EQ changes at this stage because there is no peak control prior to the clipping system. It is easy to overload the clipping stages by setting these controls all to large positive values. The control range for each band of +/- 3dB is purposely restricted for the above reasons.

### Look-ahead limiter

A look-ahead limiter is used in the DSPXtra to provide final peak control. This is not just a simple single band peak limiter. The DSPXtra look-ahead limiter works in three bands to maximise the quality of the processing while minimising audible artefacts followed by precision final look-ahead limiter.

Each of the three bands has its own length delay line optimised to match that of the dual time constant gain control circuit controlling that bands level. By realignment of delays the look-ahead limiter like all of the DSPXtra processing maintains linear phase. You have access to the secondary (average responding) time constants for each band and these controls allow an element of control over the texture of the look-ahead limiters processing. The look-ahead limiter has its own fixed internal time constants which 'RIDE' piggyback on the secondary time constants that you have access to. These fixed peak time constants are optimised for the band in question to preserve transparency and peak control and are matched to the delay lines.

The look-ahead limiter has its own drive control adjustable over a +/- 6dB range. Like the multi-band clippers the 0dB reference point for the drive was chosen as a compromise between loudness and quality. As you increase the drive more loudness will be obtained but at the expense of IM distortion which will start to make the audio sound busy and packed.

The adjustable time constants in the look-ahead limiter also play a big role in affecting on air loudness and these need to be adjusted carefully to prevent pumping when driving the look-ahead limiter more aggressively. Like the dual time constants in the multi-band limiters you can adjust so that the peak time constants have most of the control or the secondary ones have most of the control, or a balance, which works best. To lessen the effect of the secondary time constants set the attack to 10 and the decay to 1. This minimises energy distributed into the secondary time constant circuit. The peak time constant circuit will dominate and control the audio. If we were to do this with each of the bands of the look-ahead limiter we would significantly increase the loudness but more business and IM distortion would be introduced to the audio especially under higher levels of drive. If we were to flip the settings of the secondary time constants around to distribute most of the energy into the second-



any time constant circuit we would see an improvement of audio quality at the expense of reduced volume due to the longer release time constants being used. As previously pointed out a balance is best and that balance will vary with each band of the look-ahead limiter.

Great care is needed in setting the final look-ahead drive control. This control needs to be adjusted carefully and only you can make the decision on the balance between loudness and quality. As you increase the drive you will obviously obtain more loudness but at the expense of distortion and "busy" sound. There is a fine line between artistic distortion and distortion that your listeners will find uncomfortable to listen to, especially for extended periods of time. We also suggest taking into account that generally perceptual codecs don't like aggressively processed audio, leading to increased coding artifacts and codec overshoots.

Another control that affects coding artifacts is the lowpass filter, which removes high frequency energy from the audio - area that codecs are most sensitive to. As a general rule the filter should be set at or below half sampling frequency of the following codec. In any case, you should experiment with it as different codecs will respond differently to high-end energy.

### **Auditioning individual stages of the DSPXtra processing**

When setting up the DSPXtra it is sometimes useful to hear the effects of the adjustments on that particular processing stage. By setting the other processing blocks up in a certain way you can make it easier to hear the effects of the one you are adjusting.

For example, to listen to just the action of the multi-band limiters bypass the Ariane stage and back down the multi-band and final clipper drives. To hear just the clippers do the same but raise the limiter thresholds to +6dB and back the drive down into the limiters to -6dB. You can then up the drive into the clippers with the clipper drive controls and mixer controls to here just the clippers working.

## GETTING THE SOUND YOU WANT

While the DSPXtra-HD can help you obtain the sound that you want we must always take into account the limitations presented to us by the transmission channel. The biggest problem we have is the maximum peak level that can be handled by that transmission channel, and for digital broadcasting services that is 0 dBfs.

The trade off in any audio processor is loudness vs. quality. The mark of how good a processor is, is how loud the processor can go while maintaining sufficient quality. It is up to you where this loudness / quality trade off point is set. This point is also usually market and format dependent.

In the effort to squeeze as much bass and high frequency energy into the peak limited channel we must make compromises. Bass takes up a lot of room in the waveform and pursuit of a 'mega bass' type sound will leave you less room for high frequencies. When processing aggressively we usually will have to accept a certain level of bass distortion in making room for high frequencies or we will have to accept a certain level of high frequency distortion if our desire is lots of clean loud bass.

If your aim is a cleaner sound and a slight loss of loudness is not important then it is easier to get the tonal characteristic you're after without distortion. Lower final limiter drives will provide you with clean bass and crystal clear razor sharp high frequencies. The choice is yours.

### More LOUDNESS

Loudness can be increased in several ways.

#### Ariane AGC:

Speed up the release times, making them faster.

Decrease the IDR settings letting the release come into play more often.

Increase the GATE thresholds allowing the release times to release for longer.

Reduce the band and channel coupling ratios letting each bands energy not be restricted by its neighbour.

While it is possible to create a strange response in the Ariane it is hard to produce distortion because whatever gets through the AGC is dealt with by the following peak limiters. Pumping can be introduced by 'maxing out' the controls. For example, IDR 1dB, RELEASE 1 and GATE OFF. It is usually OK to max out one of the controls if the others are set accordingly.

#### Multi-band Limiters:

Slow down the peak attack times, letting more through to the peak limiter.

Speed up the release times of the average release time constants.

Slow down the average attack time so that the peak time constants dominate the control signal providing faster control.

Reduce the HOLD level or turn the HOLD level off.

Reduce the DELAY control to lower numbers.

Increase the limiter thresholds, letting more through to the peak limiter.

Reduce the band and channel coupling ratios.

Extra loudness can be obtained by working on only a single or a couple of the above suggestions. You are likely to run into trouble if you 'crank up' all of the above settings. You are likely to generate excessive distortion in the final limiter and generate a fatiguing sound if you're not careful. Less can be more! Make small changes and compare against the settings of the factory presets if you find you have lost your way somewhere.

#### Peak limiter:

Increase the look-ahead limiter drive

Increase the attack time constants, slowing down the attack of the secondary time constant circuits.

Decrease the decay time constants, speeding up the decay of the secondary time constant circuit.

WE SUGGEST ONLY SMALL MODIFICATIONS FROM FACTORY PRESET SETTINGS IF YOU ARE MODIFYING LOTS OF THE PARAMETERS. IF YOU ARE ADJUSTING ONLY A COUPLE FROM THE ABOVE SUGGESTIONS THEN YOU PROBABLY HAVE A BIT MORE LEE-WAY. IT IS VERY EASY TO LOSE YOUR WAY ONCE YOU START 'CRANKING' LOTS OF DIFFERENT SETTINGS.

### More CLARITY

We can obtain extra clarity and quality in several ways.

**Ariane AGC:**

Slow down the release times.

Increase the IDR settings letting the release come into play less often.

Decrease the GATE thresholds allowing the release times to release less often.

Increase the band and channel coupling ratios, letting each bands energy be restricted by its neighbour.

**Multi-band Limiters:**

Speed up the peak attack times, letting less through to the peak limiter.

Slow down the release times of the average release time constants.

Speed up the average attack time so that the average time constants dominate the control signal providing slower control.

Increase the HOLD level.

Increase the DELAY times.

Decrease the limiter thresholds, letting less through to the peak limiter.

Increase the band and channel coupling ratios, preserving the spectral and channel balance of the original program material

**Look-ahead limiter:**

Increase the look-ahead limiter drive.

Decrease the attack time constants, speeding up the attack of the secondary time constant circuits.

Increase the decay time constants, slowing down the decay of the secondary time constant circuit.

WE SUGGEST ONLY SMALL MODIFICATIONS FROM FACTORY PRESET SETTINGS IF YOU ARE MODIFYING LOTS OF THE PARAMETERS. IF YOU ARE ADJUSTING ONLY A COUPLE FROM THE ABOVE SUGGESTIONS THEN YOU PROBABLY HAVE A BIT MORE LEE-WAY. IT IS VERY EASY TO LOSE YOUR WAY ONCE YOU START 'TWEAKING' LOTS OF DIFFERENT SETTINGS.

**More BASS**

We can obtain more bass in several ways.

**Bass Enhancement:**

Increase the shelving filters gain. Listen carefully when adjusting because you can cause suck out of the mid and upper bass by excessive low frequency boost.

Increase the peaking filters gain.

Decrease the peaking filter Q factor.

Sometimes more AUDIBLE bass can be achieved through more peaking gain and less shelving gain as the peaking filter operates in the region where the ear is more sensitive. Lots of shelving gain can rob the loudness in this ear sensitive region by the gain being reduced by the less audible low frequency material that may have more level, controlling the gain reduction of the band 1 AGC and limiter.

**Ariane AGC:**

Speed up the band 1 release.

Reduce the IDR setting.

Reduce the band 2 to 1 coupling control, letting band 1 increase its gain more often.

**Multi-band Limiters:**

Increase the band 1 limiter drive.

Slow down the peak attack time for band 1, letting more through to the peak limiter.

Speed up the release time of the average release time constant.

Slow down the band 1 average attack time so that the peak time constant dominates the control signal providing faster control.

Decrease the band 1 HOLD level or turn it off.

Decrease the band 1 Delay time

Increase the band 1 limiter threshold, letting more through to the peak limiter.

Reduce the B1<2 coupling control so that band 1 has independent level control and is not coupled to band 2.

**Peak limiter:**

Increase the look-ahead limiter drive.

Increase the band 1 look-ahead limiter attack time constant slowing down the attack of the secondary time constant circuit.

Decrease the band 1 look-ahead limiter decay time constant speeding up the decay of the secondary time constant circuit.

WE SUGGEST ONLY SMALL MODIFICATIONS FROM FACTORY PRESET SETTINGS IF YOU ARE

MODIFYING LOTS OF THE PARAMETERS. IF YOU ARE ADJUSTING ONLY A COUPLE FROM THE ABOVE SUGGESTIONS THEN YOU PROBABLY HAVE A BIT MORE LEE-WAY. IT IS VERY EASY TO LOSE YOUR WAY ONCE YOU START 'TWEAKING' LOTS OF DIFFERENT SETTINGS.

### **More TREBLE (HF)**

We can obtain more high frequency energy in several ways.

#### **Ariane AGC:**

Speed up the band 4 AGC release time, making it faster.

Lower the IDR setting.

Increase the GATE threshold, allowing more gain to be applied to low level HF waveforms.

Reduce the B4<3 coupling control so that band 4 has independent level control and is not coupled to band 3.

#### **Multi-band Limiters:**

Increase the band 4 limiter drive .

Slow down the peak attack time for band 1, letting more through to the clippers.

Speed up the release time of the average release time constant.

Slow down the band 4 average attack time so that the peak time constant dominates the control signal providing faster control.

Decrease the band 4 HOLD level or turn it off.

Decrease the band 4 delay time.

Increase the band 4 limiter threshold, letting more through to the clippers.

Reduce the B4<3 coupling control so that band 4 has independent level control and is not coupled to band 3.

#### **Look-ahead limiter:**

Increase the look-ahead limiter drive.

Increase the band 4 look-ahead limiter attack time constant slowing down the attack of the secondary time constant circuit.

Decrease the band 4 look-ahead limiter decay time constant speeding up the decay of the secondary time constant circuit.

WE SUGGEST ONLY SMALL MODIFICATIONS FROM FACTORY PRESET SETTINGS IF YOU ARE MODIFYING LOTS OF THE PARAMETERS. IF YOU ARE ADJUSTING ONLY A COUPLE FROM THE ABOVE SUGGESTIONS THEN YOU PROBABLY HAVE A BIT MORE LEE-WAY. IT IS VERY EASY TO LOSE YOUR WAY ONCE YOU START 'TWEAKING' LOTS OF DIFFERENT SETTINGS.

### **More STEREO ENHANCEMENT**

We can obtain more stereo enhancement by employing the Arianes matrix mode of operation. This mode splits the audio into 4 bands of SUM and difference signal. In other words a mono signal and a signal containing the stereo information. By enhancing this stereo signal (the difference channel) we are able to increase the spatial effect. This must be done carefully to avoid problems such as multi-path when the DSPXtra-HD is processing for FM broadcast. The sophisticated controls in the Ariane section of the DSPXtra-HD are more than capable of handling this problem, if set correctly. To increase stereo (spatial) energy, make sure the Ariane is operating in matrix mode.

Set the L-R IDR to lower numbers

Set the Gate to higher numbers

Set the RELEASE for each band to lower numbers.

These settings will primarily affect how the L-R (spatial) energy is processed but there are some other controls that also have a part to play but are much more important when it comes to worrying about multi-path when the DSPXtra-HD is processing for analogue FM. These controls are the L-R restriction and L-R injection.

The L-R restriction control sets how much extra gain can be applied to the L-R signal in relation to the gain being applied to the L+R signal. This prevents low level L-R (spatial) energy from being 'sucked up' too much causing strange stereo effects. This control restricts the amount to a ratio between the L+R and L-R which is different to how the normal GATE control works.

The L-R injection level affects how much of the processed L-R signal (after the 4 bands are recombined) is fed to the demultiplexor which recreates the original left and right channels. If this control is set at 0dB then full processed spatial energy is demultiplexed with the L+R mono signal. At the other end of the scale setting the control to OFF will inject zero energy into the demultiplexor leaving only the original L+R mono signal which after demultiplexing leaves only a mono signal. Try it and see. You will notice that you can achieve quite a lot with this control. By processing the L-R to increase the stereo effect and then reducing the maximum level with the injec-

tion control we can create a consistent amount of stereo enhancement.

We recommend the L-R restriction control being set somewhere between -6 and -12dB. With -12dB being for more stereo effect as the L-R gain is able to increase -12db past the L+R gain. A setting of 0dB ties the L-R gain to the L+R gain which effectively turns off stereo enhancement.

We recommend the L-R injection control being set to somewhere between -6 and -3dB. Having the setting at less than unity (0dB) helps reduce multi-path, and the reduction in stereo effect can be more than made up for with the proceeding controls. Setting this control too low will minimise the stereo effect to the point of creating a mono signal.

You may think that lowering the settings of the injection and restriction controls has a similar effect. The difference between setting the L-R restriction to 0dB and L-R injection to OFF is that the former disables enhancement but the latter actually creates mono even if the input to the Ariane is stereo.

### **Less STEREO ENHANCEMENT**

Set the L-R processing controls to the same as that of the L+R controls (for consistency in the processing)

Set the L-R restriction control to a higher number to restrict too much gain being applied past the L+R's processing.

Set the overall level of the L-R with the injection control. Lower numbers are less stereo.

### **Stereo Enhancement and Audio Coding**

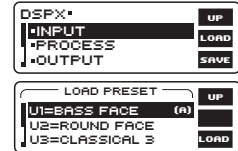
Perceptual codecs often exploit the audio redundancy that exists between stereo channels in order to reduce the amount of data needed to be coded. These techniques are mostly known as joint-stereo and lately, parametric stereo (PS). Excessive stereo width can put more burden on that part of the coding process, that is designed to handle usual amounts of stereo width in music. Therefore, excessive stereo enhancement can result in more artifacts after coding. When processing for coded audio (especially low bitrate) it is often desirable to restrict and even reduce stereo width of source audio as this can improve audio quality and reduce coding artifacts.

## MANAGING PRESETS (FRONT PANEL CONTROL)

The DSPXtra-HD has an assortment of factory presets and provision for 8 user ones. While the factory presets may not suit your tastes you will generally find one that serves as a good starting point in creating your own custom preset. The preset facilities of the DSPXtra-HD are accessed from the three intuitive soft keys.

### Selecting a preset.

First press the 'LOAD' button and then select the preset you want with the control knob. Factory presets are prefixed with Fx where x is a number. User presets are prefixed with Ux where x is a number between 1 and 8. Once you have selected the preset you want to load you simply press the 'LOAD' button again. You can audition various presets by simply scrolling through the preset list and hitting 'LOAD' over each one you wish to listen to.

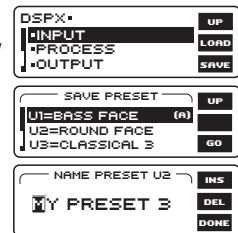


### Comparing a preset

When making processing adjustments it is often desirable to compare against the preset you are adjusting. For example you may wish to modify a factory preset and save it as a user preset. You select a factory preset and load it, making it active. You like the factory preset but want to increase the bass slightly and possibly reduce the drive into the main clipper to reduce distortion. You could modify both processing parameters and then press the 'LOAD' button. The middle button of the softkeys will change to B. Pressing this button will reload the saved preset allowing you to compare before and after your changes. The previously marked 'B' button will have now changed to a 'A' button which if pressed will return you to the adjusted preset. The other option you have if you don't press 'UP' is the 'LOAD' button, reloading the saved preset and discarding your changes. By using the button you can easily make processing adjustments quickly and hear instantly if the change is to your liking. It is very easy to forget where you are sonically so the comparison feature is very useful. You can also use the facility to make one processing parameter change at a time, adjusting it, discarding it or saving it to the preset. You can then repeat the comparison process until you are happy with all of your processing changes.

### Saving a preset

To save the current active settings to a user preset you simply press the 'SAVE' button. You can then select the user preset with the control knob and press the 'GO' button. A new screen is presented to you allowing the name of the preset to be changed. 'INSERT' and 'DELETE' hot keys are provided to speed things up. Once you are happy with the name you press the 'DONE' soft key to save the preset to the DSPXtra-HD's memory. For speed, saving the current settings to the same active preset is as simple as pushing the same button three times as 'SAVE' selects the presets, 'GO' confirms the preset to save and 'DONE' confirms the name entry stage.



### Exporting a preset to a PC

This is handled by the remote control application.

### Importing a preset from a PC

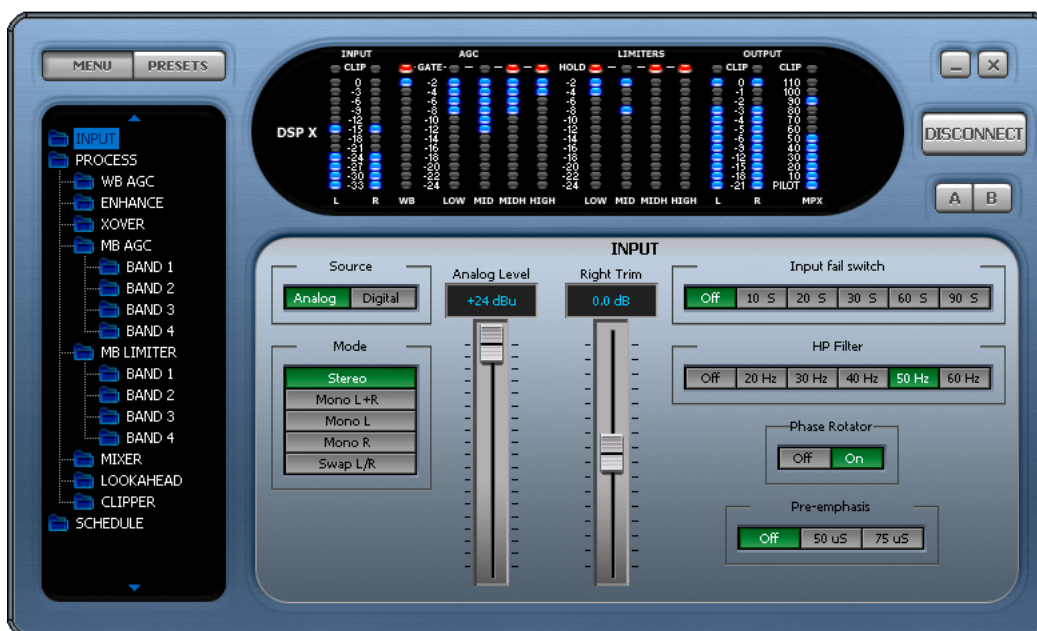
This is handled by the remote control application.

## REMOTE CONTROL OF THE DSPXTRA-HD

In addition to the front panel LCD control system the DSPXtra-HD has a USB port, a serial port and an ethernet/LAN port. These allow remote control of the DSPXtra-HD through a remote application program running on a windows based PC. The remote control program is available for download from [www.bwbroadcast.com](http://www.bwbroadcast.com)

The USB port on the front panel of the DSPXtra-HD is a USB/Serial convertor module. The USB signals from your computer are converted to serial commands which the DSPXtra-HD controller understands. This is transparent to the user but as far as the DSPXtra-HD is concerned the USB port is a serial device. The remote control application on your computer will select the USB device through a virtual comm port driver (VCP) which is a driver that you will need to install on your computer in order to connect to the DSPXtra-HD over USB. VCP driver Installation instructions are contained in Appendix A of this manual.

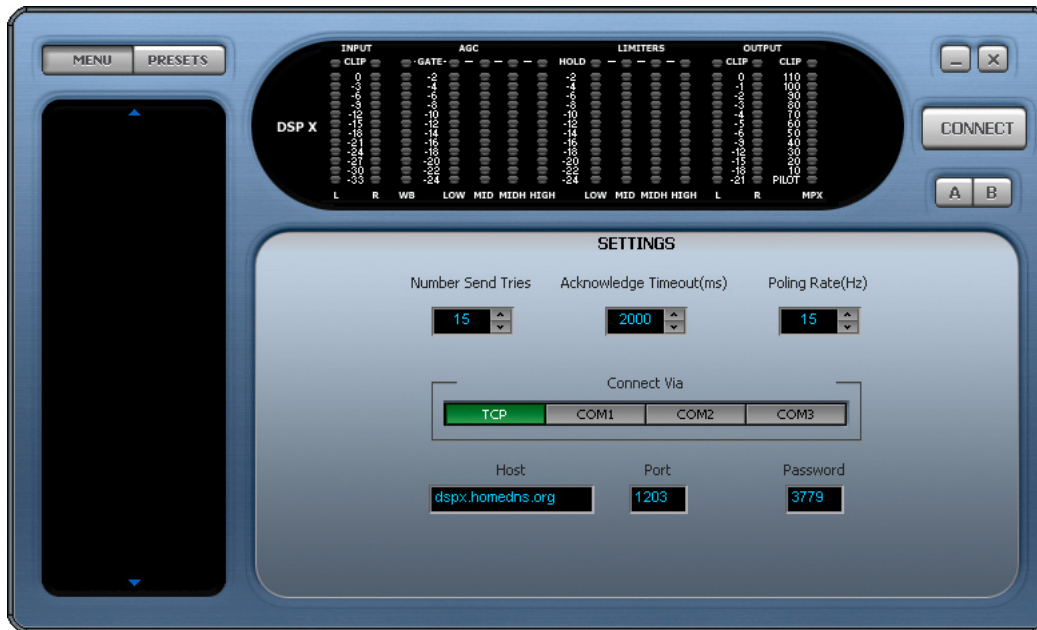
This section of the manual will refer to only serial and lan ports as the rear panel serial port and front panel USB port are the same to the DSPXtra-HD. they are only differentiated by the comm port they each have on the host computer. The DSPXtra-HD internally connects the back panel serial and front panel USB ports together in software. you can only use one at any one time. Connecting both to your computer may result in errors.



Remote application

The DSPXtra-HD can only talk to the serial system (RS232/USB) or the ethernet/LAN system at any point in time so you will need to select which one of the two remote control methods you wish to use by selecting the appropriate option from the remote source parameter which is contained in the system menu accessible from the DSPXtra-HD's front panel.

If connecting via an ethernet connection you will need to set the IP address or Hostname that the DSPXtra-HD is connected at and also include the PORT number that the DSPXtra-HD has been set to use. The default port that BW use is 1203. You can leave it as is unless you have a reason to change it. Your network administrator can help you with this.



Remote application connection screen

### CONTROL OF THE DSPXtra-HD BY RS232 or USB (SERIAL)

If you wish to use the RS232 or USB port to control the DSPXtra-HD follow the steps below

1. Connect the supplied serial cable to the rear RS232 port or a USB cable to the front USB port
2. Navigate to the 'REMOTE SOURCE' parameter (also in system) and select the serial option
3. Run the DSPXtra-HD remote application and you will be presented with a connection screen (see image). Select the COM port on your computer that you have plugged the serial cable into or select the virtual comm port corresponding to the USB cable.
4. A password needs to be entered, regardless of password settings on the DSPXtra-HD itself. A password still needs to be entered even if the passwords on the DSPXtra-HD have been disabled.
5. Click connect on the application and you should receive a 'please wait' box while the information is retrieved from the DSPXtra-HD. Once connected you are then free to control the DSPXtra-HD with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtra-HD remote application does not connect or disconnects after a few seconds then it could be that the password is incorrect. The default password for the DSPXtra-HD is 3779. You are free to change these on the DSPXtra-HD itself (see information on password control elsewhere in this manual)

### NET/LAN PORT

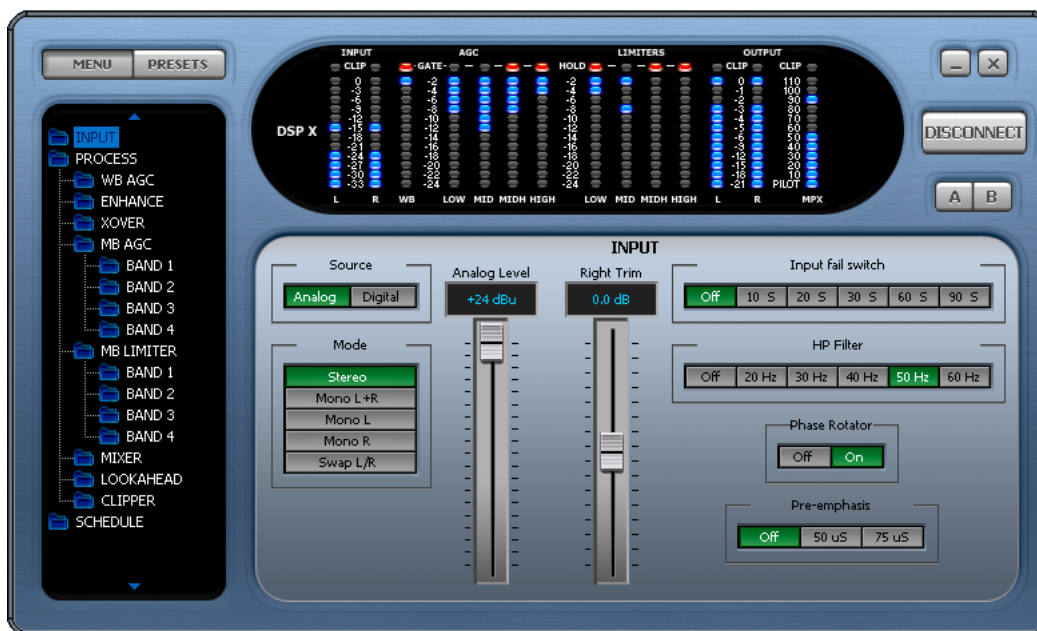
The DSPXtra-HD is equipped with a NET/LAN port for ease of remote control, setup and monitoring.

### CONTROL OF THE DSPXtra-HD BY THE NET/LAN PORT

If you wish to use the NET/LAN port to control the DSPXtra-HD follow the steps below

1. Connect a cat 5 cable to the RJ45 port on the DSPXtra-HD and plug this into your network hub/switch. A Xover cable can be used to connect directly to a PC if you don't have a switch or hub.
2. . Navigate to the 'REMOTE SOURCE' parameter contained in the system menu on the DSPXtra-HD and select the Network option.
3. Run the DSPXtra-HD remote application and you will be presented with a connection screen (see below) . Select the Ethernet option.
4. A password needs to be entered, regardless of password settings on the DSPXtra-HD itself. A password still needs to be entered even if the passwords on the DSPXtra-HD have been disabled.
5. Click connect on the application and you should receive a 'please wait' box while the information is retrieved from the DSPXtra-HD. Once connected you are then free to control the DSPXtra-HD with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtra-HD remote application does not connect or disconnects after a few seconds then it could be that the password is incorrect. The default password for the DSPXtra-HD is 3779. You are free to change these on the DSPXtra-HD itself (see information on password control elsewhere in this manual)





Remote application connected

### PASSWORD ACCESS

The DSPXtra-HD contains two levels of password control, a high level password which blocks access to all areas of the DSPXtra-HD and an 'Output' level password that allows access to all areas of the DSPXtra-HD except the output menus that contain the output mode and level settings. The 'Output' level password could be given to programme controllers to adjust the processing knowing that the transmission will remain compliant as there is no way for the user to adjust the peak output level of the DSPXtra-HD.

These passwords can only be set from the DSPXtra-HD front panel and are located in the system menu. The password box is located on the connection screen to the right of host and port input boxes.

The default (factory shipped) passwords for the two locks are '3779'. The remote application will always default to this when it is run, unless you change it. Some users may find the output lock set to '0000' Try this is 3779 does not let you access the output menus.

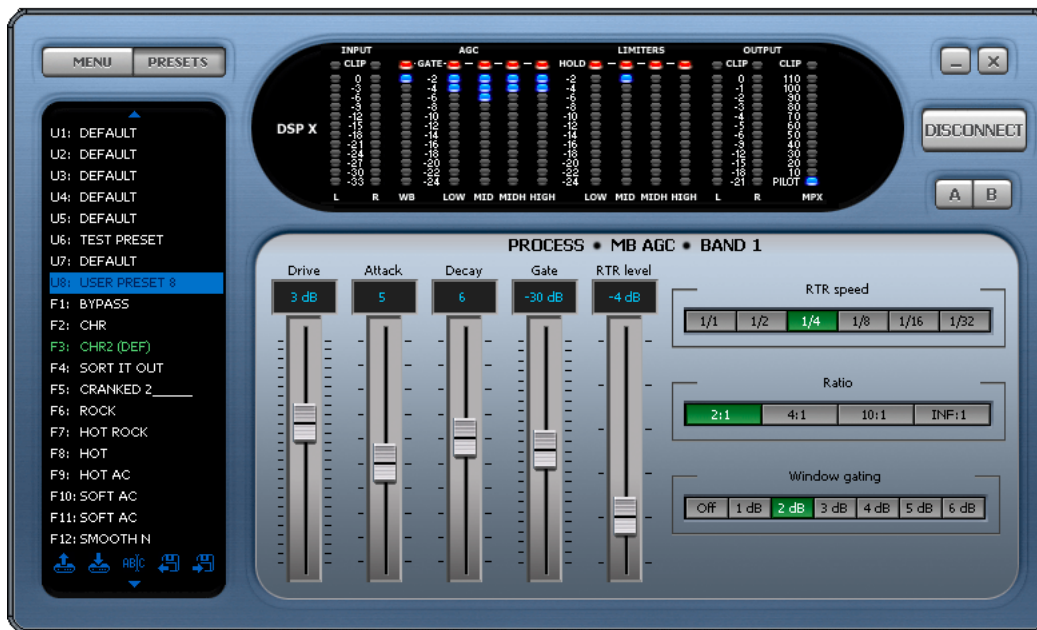
When the DSPXtra-HD is connected the LED's will show activity and the main controls window should show processing controls, depending on what option is selected on the menu tree located to the left of the screen. The DSPXtra-HD remote control application has three windows. The left hand contains the navigation/preset window. The top shows the LED metering while the bottom right contains the main controls window that is populated with the appropriate controls for the part of the processing that is selected in the menu tree.

At the top left of the application you have the menu/preset toggle buttons. These change the contents of the left hand window from the DSPXtra-HD menu tree to the preset list. The top right of the application contains the minimise and close icons, the connect / disconnect button and the A/B buttons which will be covered shortly.

### NAVIGATING THE PROCESSING STRUCTURE AND MAKING PROCESSING ADJUSTMENTS

Navigating through the processing structures of the DSPXtra-HD is very simple. When connected click the menu button (top left) if not already depressed. You should then see the menu tree in the left hand window (see above image).

You can now navigate through the menu tree and see the controls that are contained in that menu appear in the main controls windows. The example above shows the controls that are contained in the input menu. Changing the processing is as simple as adjusting the sliders and buttons.



Preset window shown

## WORKING WITH PRESETS

The DSPXtra-HD remote application makes it easy to load, save and change presets. Click the preset button (top left) if not already depressed. You should then see the preset list in the left hand window (see image).

### Understanding the preset list

The preset list contains all of the presets contained in the DSPXtra-HD. The user presets are prefixed U1 to U8 and the factory presets from F1 onwards. You may need to use the scroll arrows to view all of them as they won't all fit in the window at one time.

It is important that you understand the following terminology and how the various presets are displayed in the window if you want to use the preset window correctly and efficiently.

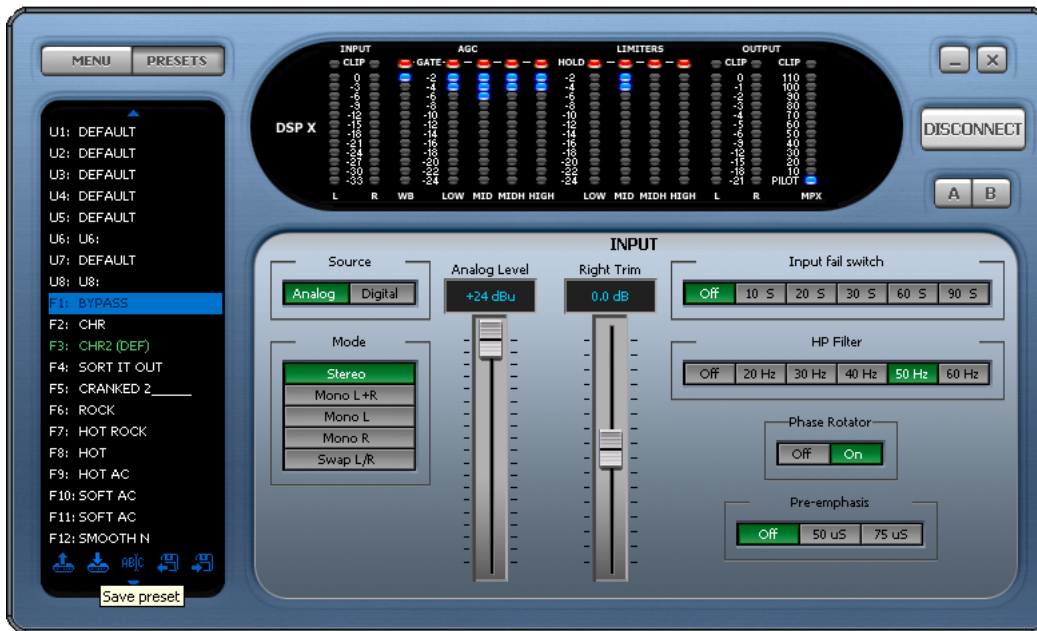
The currently 'on air' preset is always marked in green.

The preset marked in green will also have a label appended to the preset name. This can be (DEF), (TR) and (DP) and these stand for the default preset, daypart and triggered presets.

If you are not using dayparting or the external trigger port the default preset will always be the 'on air' preset and it will be marked in the preset list with a (DEF) which appends to the preset name. If the scheduler (dayparting) or the remote trigger interface has changed the preset the (DEF) marked preset may not be the one that is on the air. There are two additional identifiers to mark these occurrences. (DP) for Daypart and (TR) for remote trigger. If a daypart of trigger occurs the (TR) or (DP) will appear next to the name and the preset name will change to green to indicate that it is 'on air' and has overridden the (DEF) default preset. When the daypart or trigger finishes control will always return to the default (DEF) preset.

It is possible to have A (DEF)(TR)(DP) situation where a remote trigger forces 'on air' a user preset which has also been triggered by a daypart and that preset happened to be the default preset. Unlikely but possible.

Various preset operations are possible including changing the default preset, saving a preset to a user preset location, changing the name as well as PC file operations to backup or share presets with other DSPXtra-HD users. To perform an operation you will need to select a preset by clicking on to the name in the list. This will highlight the preset with a blue bar. This does not change the preset or affect anything on the air. All this blue selection bar indicates is that this is the preset that we want to perform an operation on. We have two methods of performing the operation on the preset. The first method is to click one of the icons at the bottom of the window. These are from left to right, Load preset, Save preset, Rename preset, Load from PC, Save to PC. The other method is to right click over the preset where you will be presented with a drop down menu containing the same options.



Preset icons

### Load preset

Load preset will change the default preset to the one selected. This usually means that this preset will become 'on air'. The exception to this is when the default preset is being overridden by a daypart or remote trigger. In this case the DSPXtra-HD will 'on air' the selected preset when the daypart or trigger hands back control to the default preset.

### Save preset

Save preset will save the current on-air preset to the highlighted user preset position. You can not write over a factory preset.

### Rename preset

Rename preset allows the user preset name to be edited. You are restricted to 15 characters.

### Saving presets to PC

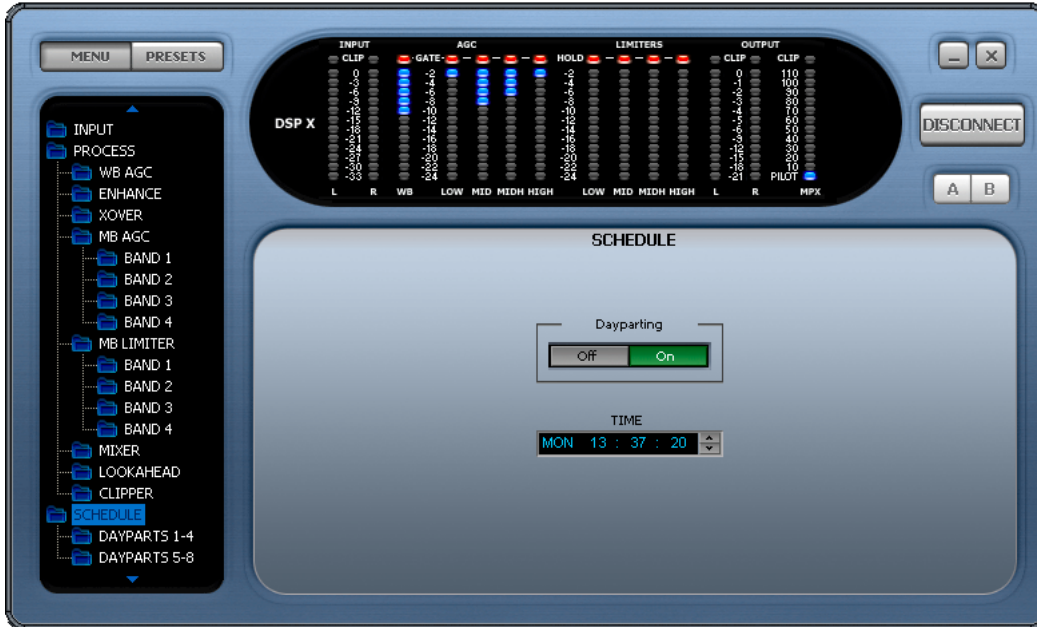
This option will pop up the standard windows save dialog box. You can select a file name and location for the preset to be saved under. The preset that is saved is the currently highlighted (in blue) preset, not the one that is currently 'on-air'.

### Loading presets from a PC

This option will pop up the standard windows load dialog box. You can browse to and select a preset file to be loaded into the DSPXtra-HD. The preset location that is loaded is the currently highlighted (in blue) preset, not the one that is currently 'on-air' You can only load into a user preset.

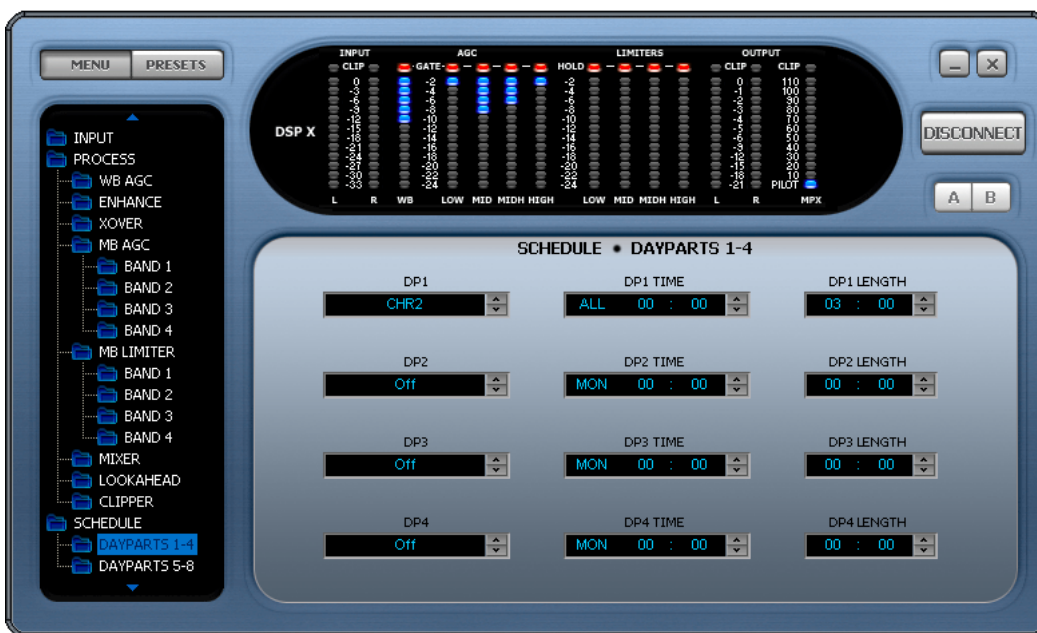
## SCHEDULING WITH THE REMOTE APPLICATION

Using the menu tree to change the processing is quite simple and really doesn't need a lot of explanation. The scheduling screens that controls the dayparting may appear daunting so we are going to give you a quick guide to using it. There are three menu locations for controlling scheduling. The first location is shown in the screen shot below and it contains the ON/OFF and time setting control. The two menu locations below it access two banks of four dayparts which make up the 8 dayparts contained in the DSPXtra-HD.



Scheduling

On the daypart windows you have four dayparts. Each has three parameters. The left hand box contains the name of the preset that you want this daypart to switch to. This box also has the ability to turn the daypart OFF by clicking down on the arrows until you reach the off option. If off is currently selected you can click up to rotate through the user and factory presets. The middle box contains the time and day that the daypart will start at. To select day, hour or minute click on the appropriate part of the box before using the up and down arrows. The day part of the time also has an ALL option. This means that the daypart will occur on every day. The right most box contains the length of the daypart in hours and minutes. Like the start time of the daypart you will need to click into the appropriate part of the box before clicking the up/down arrows.



Scheduling

The dayparts can be layered so that one can override another. Let's say the default preset was U1:MAIN PRESET and this was on the air all of the time. We want to change the preset from 7AM to 10AM every day of the week to F2:CHR and then from 10AM to 12PM we want U4:NEW PRESET and then back to F2:CHR until 5PM.

Rather than setup the dayparts as

**DPO: F2:CHR - ALL 07:00 - 03:00** (factory preset 2 to run from 7am everyday for 3 hours)  
**DP1: U4:NEW PRESET - ALL 10:00 - 02:00** (user preset 4 to run from 10am everyday for 2 hours)  
**DP2: F2:CHR - ALL 12:00 - 05:00** (factory preset 2 to run from 12pm everyday for 5 hours)

We could instead setup the dayparts as

**DPO: F2:CHR - ALL 07:00 - 10:00** (factory preset 2 to run from 7am everyday for 10 hours)  
**DP1: U4:NEW PRESET - ALL 10:00 - 02:00** (user preset 4 to run from 10am everyday for 2 hours)  
which saves a daypart position.

By carefully selecting the default preset and overlaying dayparts we are able switch presets significantly more than you first think you will be able to.

### **A/B COMPARISON FEATURE**

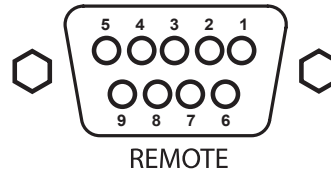
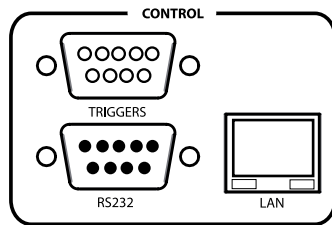
The Remote application has two buttons labelled A and B that are located just below the connection button. These buttons allow you to compare changes you have made to the processing against the saved preset. When you load a preset the buttons should be greyed out but as soon as you make any processing changes these buttons will become active. By selecting the B button you can temporarily revert back to the saved preset. During this time all the processing controls will grey out to indicate you are in a compare mode. To return to the settings that you have been adjusting click the A button and the processing controls will 'un-grey' If at any time you want to revert to the saved preset and lose your adjustment just reload the preset from the preset selection window.

The A/B feature makes it easy to build up your own presets by being able to easily compare before and after processing adjustments. We hope you find it useful.

## REMOTE TRIGGER PORT

The system menu contains the remote trigger port option from where it can be enabled or disabled.

If enabled the remote trigger port on the processor allows you to select any of the first 8 user presets by pulling one of 8 pins on the trigger port socket low. The rear panel trigger port socket is a 9 pin male D-type whose connections are shown below.



### REMOTE TRIGGER PORT PIN-OUT

<b>PIN 1</b>	USER PRESET 4
<b>PIN 2</b>	USER PRESET 3
<b>PIN 3</b>	USER PRESET 2
<b>PIN 4</b>	USER PRESET 1
<b>PIN 5</b>	EARTH RETURN
<b>PIN 6</b>	USER PRESET 8
<b>PIN 7</b>	USER PRESET 7
<b>PIN 8</b>	USER PRESET 6
<b>PIN 9</b>	USER PRESET 5

The trigger port socket contains an earth return connection pin for the 8 opto-isolated trigger pins. When the trigger pins are connected to the earth return pin they will change the currently active preset to the user preset triggered by that pin. If more than one pin is pulled low at the same time the pin with the lowest number will take priority. E.G. if all pins are pulled low trigger 1 will take priority. Once the trigger pin disconnects from the earth return connection the processor will return processing to the normally active preset.

Relays, contact closures, open collector and other hard wiring arrangements can be used to perform the appropriate connection between the trigger port pin and the earth return pin.

If you wish to trigger a factory preset you will need to copy that factory preset to a user preset first.

## SECURITY CODE LOCKS

The system menu contains the security code lock options from where they can be enabled or disabled.

The DSPXtra-HD has two code locks, the main full lock and the output lock. When enabled the full code lock prevents editing of the DSPXtra-HD parameters. When enabled the output lock restricts access to the output menu section of the DSPXtra-HD. This is useful when you want to stop someone from being able to adjust peak output levels which could cause your broadcast to be non-compliant with your regulatory bodies transmission specs. For example you may not want a programme director to be able to adjust the multiplex level into a transmitter but you do want them to be able to adjust the processing.

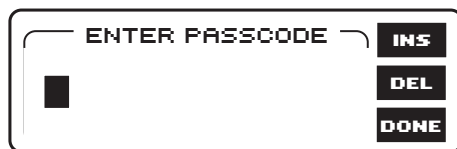
The security code locks if enabled will engage after 4 minutes of no activity on the front panel control system. This time has been chosen as a compromise between having the system lock you out when adjusting processing parameters and a short enough period of time to lock the unit after you walk away from the DSPXtra-HD.

The DSPXtra-HD is factory shipped with the factory default lock codes of 3779. Some users may find that the output lock is set to a default of '0000' so try this if you cant access the output menus with 3779.

**Enabling the security code lock feature:** Navigate to the system menu and set the code lock option to enabled. Confirm the current password. The lock will engage after four minutes of no front panel control system activity.

**Disabling the security code lock feature:** Navigate to the system menu and set the code lock option to 'disabled'. Confirm the current password.

**Changing the lock codes:** Once locked the DSPXtra-HD will not allow processing adjustments (or access to the output menu). The lock code can be entered with the use of the rotary encoder and the GO soft key can be pressed to confirm entry and proceed. If unsuccessful you will be required to try again. If successful the DSPXtra-HD will allow you to re-confirm the code or change it. This is where you may want to change the factory default code if you have not already done so. The soft keys will allow you to confirm the change or accept the previous code and the DSPXtra-HD should then be unlocked.



Entering the pass code



Confirming or changing the pass code

### Forgotten DSPXtra-HD lock code:

Contact DSPXtra-HD support and request the procedure to reset the DSPXtra-HD lock code back to factory default. In resetting the DSPXtra-HD code lock you will also wipe clean your user presets. We recommend that you export (back-up) your user presets to be on the safe side.

The easiest thing is not to forget the unlock code or to leave it at factory default. The factory default code is easy to remember, just punch in the first four letters of your favourite audio processor on a telephone keypad.

## CLOCK BASED CONTROL (DAYPARTING)

Your processor contains a battery backed up real time clock that can maintain the current time and date even when the power has been removed. This allows users to switch between presets at specific times of the day or week. This is very useful on a multi-format radio station where one processing preset may not suit all of the formats of music that are broadcast.

The easiest way to control the dayparting is with the remote control application which is described elsewhere in this manual but the use through the front panel menu system is described here.

The schedule menu contains the following options:

- The TIME
- Daypart ON/OFF control
- Dayparts 1-4
- Dayparts 5-8

### Setting the time (the processor's system clock)

Setting the time is quite simple. Select the Day, Hour, Minute or Seconds and rotate the Knob until you get to the desired setting. The Seconds can not be adjusted, only reset to 0 seconds as the knob is rotated.

There is also a clock calibration parameter, which allows a +/- 3 second correction factor to be applied at midnight each day to account for real time clock inaccuracies.

**Daypart ON/OFF** enables or disables the dayparting.

The **dayparts 1-4 and 5-8** options drop you down into two further menus. Each containing four dayparts. For each daypart you can enable or disable it and with the same control set the preset to switch to when the daypart triggers (when the daypart start time matches the system clock).

You can also set the start time (trigger) of each daypart and set the length that the daypart shall be (the time the trigger shall remain in force). The start time has a day option and this can be set to ALL which would mean that the daypart would trigger every day at the specified time. If the length is set so that the trigger will carry across midnight then the trigger will stop at midnight. TRIGGERS DO NOT CARRY ACROSS DAYS.

The dayparts can be layered so that one can override another. Let's say the default preset was U1:MAIN PRESET and this was on the air all of the time. We want to change the preset from 7AM to 10AM every day of the week to F2:CHR and then from 10AM to 12PM we want U4:NEW PRESET and then back to F2:CHR until 5PM.

Rather than setup the dayparts as

**DPO: F2:CHR - ALL 07:00 - 03:00** (factory preset 2 to run from 7AM everyday for 3 hours)  
**DP1: U4:NEW PRESET - ALL 10:00 - 02:00** (user preset 4 to run from 10am everyday for 2 hours)  
**DP2: F2:CHR - ALL 12:00 - 05:00** (factory preset 2 to run from 12pm everyday for 5 hours)

We could instead setup the dayparts as

**DPO: F2:CHR - ALL 07:00 - 10:00** (factory preset 2 to run from 7AM everyday for 10 hours)  
**DP1: U4:NEW PRESET - ALL 10:00 - 02:00** (user preset 4 to run from 10am everyday for 2 hours)  
which saves a daypart position.

By carefully selecting the default preset and overlaying dayparts we are able switch presets significantly more than you first think you will be able to.



# SPECIFICATIONS

**Frequency Response (Bypass Mode):**  $\pm 0.10$  dB, 2.0 Hz - 20 kHz or as set with the low pass filter.

**Noise:** Output noise floor will depend upon on the processor settings but is governed by the dynamic range of the A/D Converter. The dynamic range of the digital signal processing is 144 dB.

**Processing Sample Rate:** 48 kHz - 768 kHz depending on processing stage.

**Processing Resolution:** Internal processing has 24 bit (fixed point) or higher resolution.

## Analog Audio Input

**Configuration:** Stereo.

**Impedance:**  $> 10k$ , load impedance, electronically balanced.

**Nominal Input Level:** Software adjustable from 0 dBu to +24 dBu peak.

**Maximum Input Level:** +24 dBu.

**Connectors:** XLR female Pin 1 chassis ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.

**A/D Conversion:** 48KHz 24 bit 128x oversampled delta sigma converter with linear-phase anti-aliasing filter.

**Filtering:** RFI filtered.

## Analog Audio Output

**Configuration:** Stereo.

**Source Impedance:** 10 Ohm, electronically balanced and floating.

**Load Impedance:** 600 Ohm or greater, balanced or unbalanced.

**Output Level (100% peak modulation):** Adjustable from -12 dBu to +24 dBu peak, into 600 Ohms or greater load, software-adjustable.

**Signal-to-Noise:**  $\geq 90$  dB unweighted (Bypass mode, de-emphasised, 20 Hz–15 kHz bandwidth, referenced to 100% modulation).

**L / R Crosstalk:**  $\leq -70$  dB, 20 Hz - 20 kHz.

**Distortion:**  $\leq 0.01\%$  THD (Bypass preset) 20 Hz - 20 kHz bandwidth.

**Connectors:** XLR male. Pin 1 chassis ground, Pins 2 (+) and electronically balanced, floating and symmetrical.

**D/A Conversion:** 48KHz 24 bit 128x oversampled.

**Filtering:** RFI filtered.

## Digital Audio Input

**Configuration:** AES/EBU Stereo, 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.

**Sampling Rate:** 32, 44.1, 48, 88.2, or 96 kHz, automatically selected.

**Connector:** XLR female. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110 ohm impedance.

**Filtering:** RFI filtered.

## Digital Audio Output

**Configuration:** Stereo per AES/EBU standard.

**Sample Rate:** Internal free running at 32, 44.1 or 48 KHz selected in software. Can also be synced to the AES/EBU digital input at 32, 44.1, 48, 88.1 or 96 kHz, as configured in software.

**Connector:** XLR-type. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110 ohm impedance.

**Output Level:** Software adjustable from -12.0 to 0.0 dBFS.

**Filtering:** RFI filtered.

## Remote Control

**USB Port:** B type connector (front panel) \*

**Serial Port:** DB9 (rear panel) 19200 kbps

**Ethernet Port:** 10 Mbit/sec on RJ45 female connector.

**Remote Control (trigger port):** DB9 opto-isolated and floating. Eight pull low inputs.

## Other

**Voltage:** 100–240 VAC, 50–60 Hz, 35 VA.

**Connector:** IEC. Detachable 3-wire power cord supplied.

**Grounding:** Circuit ground is independent of chassis ground, and can be isolated or connected with a rear panel switch.

**Dimensions (W x H x D):** 44mm x 482mm x 200mm.

# DSPXtra-HD v1 preset sheet

	Preset 1	Preset 2	Preset 3	Preset 4
<b>INPUT</b>				
SOURCE				
MODE				
ANALOG LEVEL		dBu	dBu	dBu
INPUT FAIL				
HP FILTER		Hz	Hz	Hz
PHASE ROTATOR				
<b>PROCESS</b>				
<b>ARIANE</b>				
<b>MATRIX MODE</b>				
L+R				
IDR		dB		
GATE		dB		
B1 RELEASE				
B2 RELEASE				
B3 RELEASE				
B4 RELEASE				
L-R				
IDR		dB		
GATE		dB		
B1 RELEASE				
B2 RELEASE				
B3 RELEASE				
B4 RELEASE				
ATTACK				
B1<B2 COUPLE		dB		
B3<B4 COUPLE		dB		
L-R B1 MUTE				
L-R RESTRICTION		dB		
L-R INJECTION		dB		
<b>STEREO MODE</b>				
IDR		dB		
GATE		dB		
ATTACK				
B1 RELEASE				
B2 RELEASE				
B3 RELEASE				
B4 RELEASE				
B1<B2 COUPLE		dB		
B3<B4 COUPLE		dB		
CHANNEL COUPLE		dB		
<b>XOVER</b>				
B1-2		Hz	Hz	Hz
B2-3		Hz	Hz	Hz
B3-4		Hz	Hz	Hz
<b>ENHANCE</b>				
DEEP BASS		dB	dB	dB
BASS TUNE				
BASS PEAK GAIN		dB	dB	dB
BASS PEAK FREQ		Hz	Hz	Hz
BASS PEAK Q				
<b>MB LIMITER</b>				
MASTER DRIVE				
B1<2 COUPLING		%	%	%
B2>3 COUPLING		%	%	%
B3>2 COUPLING		%	%	%
B3>4 COUPLING		%	%	%
L/R COUPLING		%	%	%
<b>BAND 1</b>				
DRIVE		dB	dB	dB
THRESHOLD		dB	dB	dB
LIMIT. ATTACK				
LIMIT. DECAY				
COMP. ATTACK				
COMP. DECAY				
HOLD		dB	dB	dB

	DELAY							
<b>BAND 2</b>								
	DRIVE		dB		dB		dB	dB
	THRESHOLD		dB		dB		dB	dB
	LIMIT. ATTACK							
	LIMIT. DECAY							
	COMP. ATTACK							
	COMP. DECAY							
	HOLD		dB		dB		dB	dB
	DELAY							
<b>BAND 3</b>								
	DRIVE		dB		dB		dB	dB
	THRESHOLD		dB		dB		dB	dB
	LIMIT. ATTACK							
	LIMIT. DECAY							
	COMP. ATTACK							
	COMP. DECAY							
	HOLD		dB		dB		dB	dB
	DELAY							
<b>BAND 4</b>								
	DRIVE		dB		dB		dB	dB
	THRESHOLD		dB		dB		dB	dB
	LIMIT. ATTACK							
	LIMIT. DECAY							
	COMP. ATTACK							
	COMP. DECAY							
	HOLD		dB		dB		dB	dB
	DELAY							
<b>BAND 5</b>								
	DRIVE		dB		dB		dB	dB
	THRESHOLD		dB		dB		dB	dB
	LIMIT. ATTACK							
	LIMIT. DECAY							
	COMP. ATTACK							
	COMP. DECAY							
	HOLD		dB		dB		dB	dB
	DELAY							
<b>BAND 6</b>								
	DRIVE		dB		dB		dB	dB
	THRESHOLD		dB		dB		dB	dB
	LIMIT. ATTACK							
	LIMIT. DECAY							
	COMP. ATTACK							
	COMP. DECAY							
	HOLD		dB		dB		dB	dB
	DELAY							
<b>MIXER</b>								
	B1 MIX LEVEL		dB		dB		dB	dB
	B2 MIX LEVEL		dB		dB		dB	dB
	B3 MIX LEVEL		dB		dB		dB	dB
	B4 MIX LEVEL		dB		dB		dB	dB
	B5 MIX LEVEL		dB		dB		dB	dB
	B6 MIX LEVEL		dB		dB		dB	dB
<b>MB LOOKAHEAD</b>								
	DRIVE		dB		dB		dB	dB
	LOW ATTACK							
	LOW DECAY							
	MID ATTACK							
	MID DECAY							
	HIGH ATTACK							
	HIGH DECAY							
<b>FINAL LOOKAHEAD</b>								
	LP FILTER		kHz		kHz		kHz	kHz
	DRIVE		dB		dB		dB	dB
<b>OUTPUT</b>								
	LEVEL							

# APPENDIX A

## Installing USB VCP Drivers

To install the VCP drivers for the processor under Windows XP and XP SP1, follow the instructions below:

If a device of the same type has been installed on your machine before and the drivers that are about to be installed are different from those installed already, the original drivers need to be uninstalled.

Download the latest available VCP drivers from the [www.audio-processor.com](http://www.audio-processor.com) website and unzip them to a location on your PC.

If you are running Windows XP or Windows XP SP 1, temporarily disconnect your PC from the Internet. This can be done by either removing the network cable from your PC or by disabling your network card by going to the "Control Panel\Network and Dial-Up Connections", rightclicking on the appropriate connection and selecting "Disable" from the menu. The connection can be re-enabled after the installation is complete. This is not necessary under Windows XP SP 2 if configured to ask before connecting to Windows Update. Windows XP SP 2 can have the settings for Windows Update changed through "Control Panel\System" then select the "Hardware" tab and click "Windows Update".

Connect the device to a spare USB port on your PC. This will launch the Windows Found New Hardware Wizard. If there is no available Internet connection or Windows XP SP 2 is configured to ask before connecting to Windows Update, the screen below is shown. Select "No, not this time" from the options available and then click "Next" to proceed with the installation. If there is an available Internet connection, Windows XP will silently connect to the Windows Update website and install any suitable driver it finds for the device in preference to the driver manually selected.



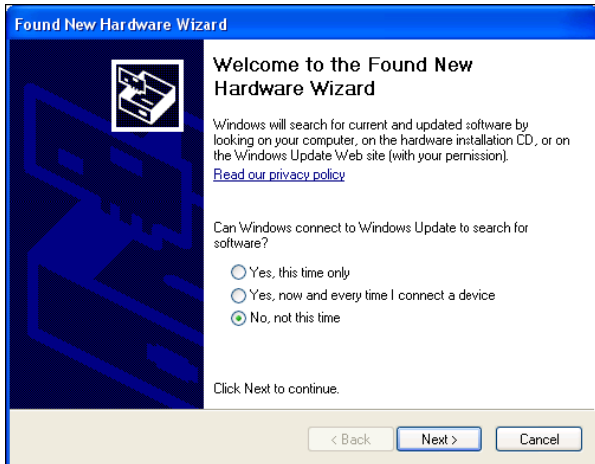
Select “Install from a list or specific location (Advanced)” as shown below and then click “Next”.



Select “Search for the best driver in these locations” and enter the file path in the combo-box (“C:\VCP Drivers” in the example below) or browse to it by clicking the browse button. Once the file path has been entered in the box, click next to proceed.



If Windows XP is configured to warn when unsigned (non-WHQL certified) drivers are about to be installed, the following screen will be displayed. Click on “Continue Anyway” to continue with the installation. If Windows XP is configured to ignore file signature warnings, no message will appear.



The following screen will be displayed as Windows XP copies the required driver files.



Windows should then display a message indicating that the installation was successful. Click “Finish” to complete the installation. This has installed the serial converter. The COM port emulation driver must be installed after this has completed.



The following screen will be displayed as Windows XP copies the required driver files.



After clicking “Finish”, the Found New Hardware Wizard will continue by installing the COM port emulation driver. The procedure is the same as that above for installing the serial converter driver.

Open the Device Manager (located in “Control Panel\System” then select the “Hardware” tab and click “Device Manger”) and select “View > Devices by Type”. The device appears as an additional COM port with the label “USB Serial Port”.





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