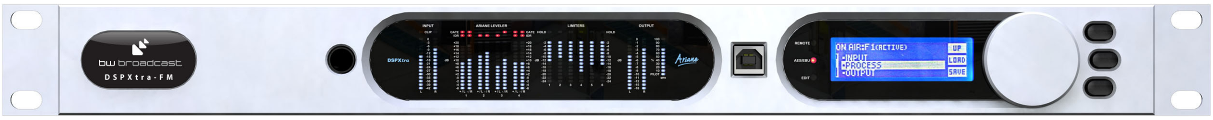
bw broadcast

DSPXtra

incorporating the

Ariane

RMS Leveller



Broadcast Audio Processor

Operational Manual
Version 2.30

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.

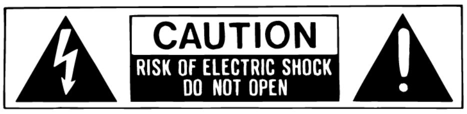
Warranty transferability
This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BW Broadcast.

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.

1.3 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 present appareil numerique n'emet pas de bruits radioelectriques depassant les limits applicables aux brouillage radioelectrique edicte par le ministere des Communications de Canada.

Forward

Thank you for your purchase of the DSPXtra-FM digital audio broadcast processor.

Over the last decade the staff at BW 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-FM. 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-FM is built-on DSPX-FM 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.

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

BW Broadcast Team

Introduction To The DSPXtra-FM

The BW Broadcast DSPXtra-FM is a new generation of digital audio signal processor that can be used to process audio ready for FM and digital broadcasting such as DAB, HD Radio or internet streaming.

Using the latest multi-band DSP technology the DSPXtra-FM offers a versatile and powerful tool in creating a loud punchy on-air presence.

The DSPXtra-FM 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-FM 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-FM 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.

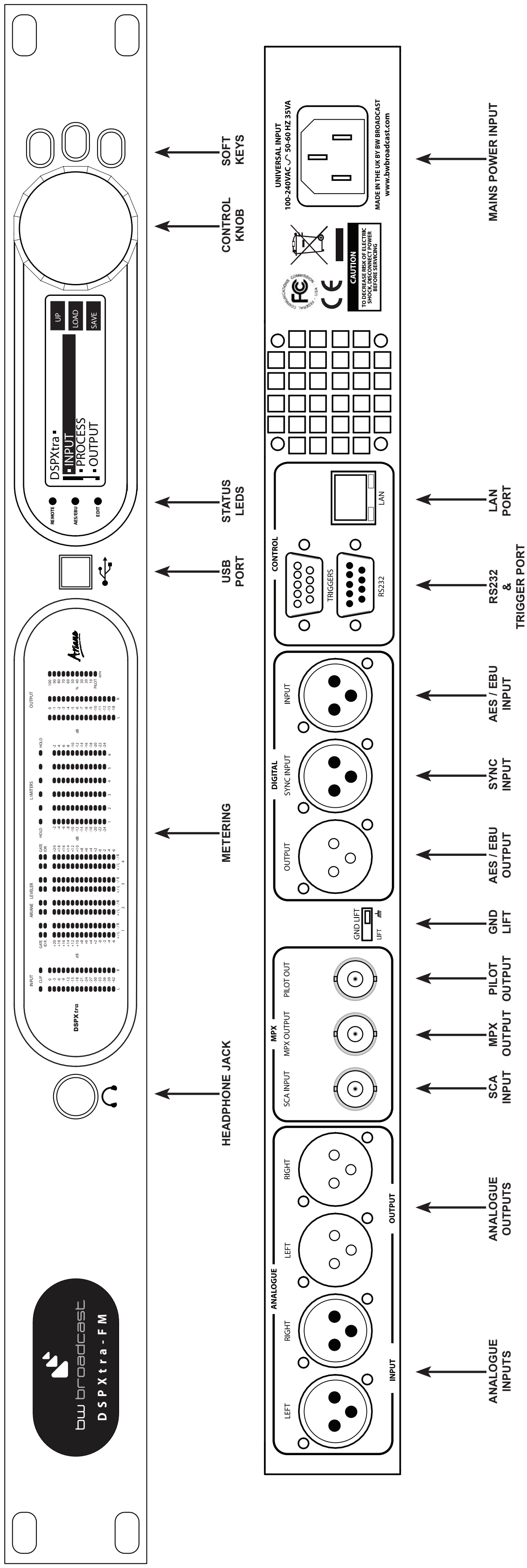
The Processing Architecture!
After input selection the 24-bit digital audio signal is passed through conditioning circuitry before being presented to the Ariane RMS levelling block. The four band RMS leveller corrects for input level variations and also improves consistency. The block also can provide stereo enhancement 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. The unique dual processing paths allow simultaneous processing for FM and digital radio. Look-ahead limiting and distortion cancelling clipping ensures your signal is kept to a strict maximum while maintaining crystal clear sound. A supersonic sample rate DSP stereo encoder provides MPX generation with fantastic stereo separation.

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-FM is a clear winner.

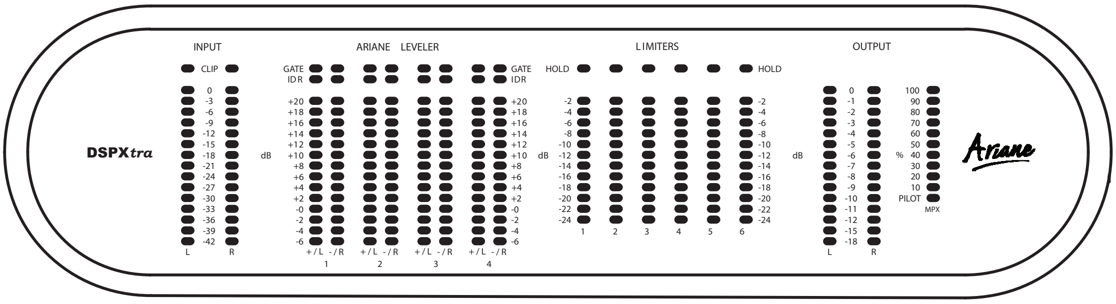
Dynamic, fresh & innovative...
... The DSPXtra-FM



DSPXtra-FM FRONT AND REAR PANELS

DSPXtra Meters

The DSPXtra 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 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. 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 multiplex output metering represents the composite outputs peak level. This is a representation of the output in relation to the peak composite level of the processing and not the actual level set by the multiplex output level control.

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 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-FM front panel contains three status LEDS.

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

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

EDIT: Indicates that you are currently editing a parameter.

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 converter will not clip under any conditions.

FM use:

5. Select the pre-emphasis setting for your region (input menu). 75 µs for USA and 50 µs for Europe.

6. Connect the audio outputs as required and set the output level and de-emphasis settings for the analogue and digital outputs to match any external links, stereo encoders, or transmitters that require left and right audio inputs or an AES/EBU input. Make sure the output mode is set to FM for the output in question.

7. If you are using the MPX Output (preferred), navigate to the ‘STEREO’ menu and adjust the ‘MPX OUTPUT LEVEL’ to match the transmitter (or link device) that follows the processor. Adjust for 100% modulation with audio.

8. Select a factory preset (see Managing presets).

DR use (DAB/HD RADIO/DRM/FMeXtra/STREAMING):

5. Set the pre-emphasis setting to OFF from the input menu.

6. Connect the audio outputs as required and set the de-emphasis for the analogue and digital outputs to OFF, MODE to DR and set the output levels for the analogue and digital outputs to match the equipment that the processor is connected to.

7. Select a factory preset (see Managing presets).

8. Navigate to the look-ahead menu and adjust the SHELF EQ control to suit. This sets the brightness of the DR outputs sound.

Dual use: FM + DAB/HD RADIO/DRM/FMeXtra/STREAMING:

5. Select the FM pre-emphasis setting for your region (input menu). 75 µs for USA and 50 µs for Europe.

6. Connect the audio outputs as required and set the de-emphasis for the analogue and digital outputs to OFF, MODE to DR and set the output levels for the analogue and/or digital outputs to match the DR equipment that the processor is connected to (EG: codec/computer).

7. If you are using the MPX Output, (preferred) navigate to the ‘STEREO’ menu and adjust the MPX ‘LEVEL’ Output to match the transmitter (or link device) that follows the processor. If one of the analogue or digital outputs needs to feed another piece of equipment that cant take a composite MPX input then make sure you select FM mode for that output with the appropriate de-emphasis setting to match the corresponding piece of a equipment in the FM chain.

8. Select a factory preset (see Managing presets).

9. Navigate to the look-ahead menu and adjust the SHELF EQ control to suit. This sets the brightness of the DR outputs sound.

NOTE: The front panel headphone jack connects to the analogue outputs so the sound may be excessively bright if pre-emphasis is engaged and de-emphasis on the analogue outputs is set to off.

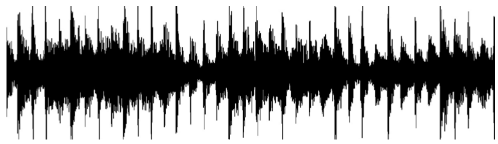
Introduction To Audio Processing

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 a 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 Arianes 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 leveller'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 variations 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.

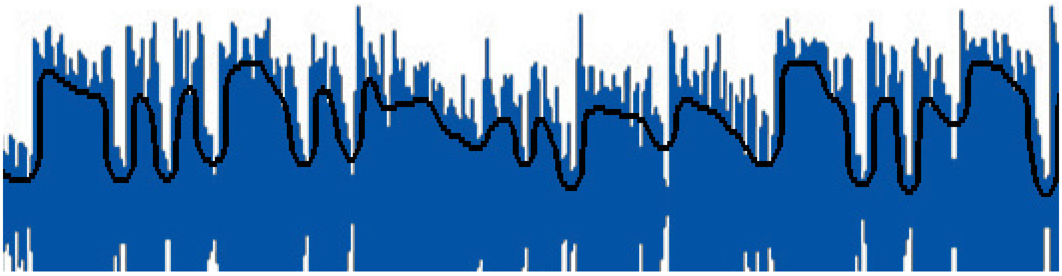
Introducing The Ariane

The Ariane is a:

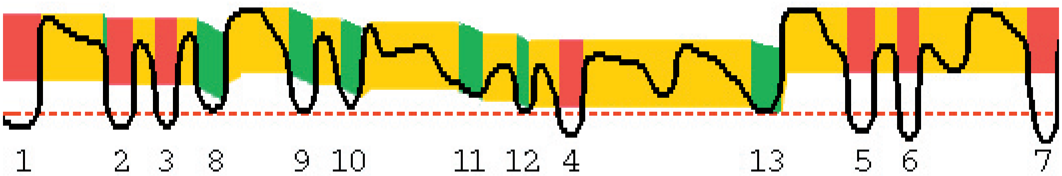
- High-quality digital,
- Two-channel,
- Multi-band,
- RMS-derived,
- Windowing release,
- Feedforward control…

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 multicolour 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-FM has the ability to substantially improve the quality of your ON-AIR broadcast. However the DSPXtra-FM can only work with what you provide it. The best performance will be obtained when the DSPXtra-FM is fed with very clean source material. After dynamic multi-band re-equalisation is performed poor quality source material will sound poorer when processed with the DSPXtra-FM.

We strongly advise against the use of MP3's and other compressed audio formats for audio storage. If you must use compressed audio we advise rates of 256 Kbps and higher. 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-FM can violate the frequency masking characteristics of the bit reduction process, creating distortion that was inaudible prior to the DSPXtra-FM processing.

Pre-emphasis

If you are using the DSPXtra-FM to process for FM broadcast you will need to enable the pre-emphasis filter in the DSPXtra-FM. Even though your STL or transmitter may contain pre-emphasis we recommend disabling it, letting the DSPXtra-FM handle the pre-emphasis for the transmission system. The DSPXtra-FM uses sophisticated processing methods to limit the high frequency energy of the pre-emphasis curve while maintaining a 'bright' sound. Using de-emphasis and then pre-emphasising again will only degrade performance and possibly casue overshoots, resulting in lower average deviation.

The exception to the rule is when the DSPXtra-FM is feeding discrete left and right outs to a compressed audio STL. Bit rate reduction codec's do not cope with pre-emphasis very gracefully and artifacts will be generated. The best option in this case is to de-emphasise the output of the DSPXtra-FM prior to the STL system. At the transmitter site the pre-emphasis can be enabled in the transmitter to restore the processed signal back to normal prior to transmission.

The best solution is always to locate the processor at the transmission site. This way overshoots are minimised and quality is maintained.

The DSPXtra-FM And Its Processing Structure

The DSPXtra-FM broadcast audio processor can be used for processing audio prior to broadcast on FM and digital radio services. Digital radio encompasses DAB, HD Radio (IBOC) and other radio based broadcasting as well as internet radio, also known as streaming. The DSPXtra-FM 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-FM 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-FM for FM and digital radio processing.

Before we discuss the processing structure in full we would like to tell you a little about the final peak limiting stages of the DSPXtra-FM. The DSPXtra-FM employs dual output paths for peak control. Your processing application may need you to configure the DSPXtra-FM in a certain way. Selecting the wrong output path and or not configuring the other settings that affect it may seriously downgrade your audio quality.

The first peak control path is known as 'FM' as it is typically used when processing signals for FM broadcast. It employs distortion controlled clippers to limit the peaks of the signal. Distortion controlled clipping is the best method for preserving as much high frequency energy as possible, important when the high frequency loss characteristics of the FM broadcast de-emphasis curve is taking into account. Distortion controlled clipping produces harmonic distortion which if used moderately can produce a sizzling bright sound but can result in a ripping or tearing sound if used excessively (overdriven).

The second peak control path is known as 'DR' (digital radio) and is the desired method of peak processing when the output feeds a codec that employs 'bit rate reduction compression techniques'. The 'DR' path 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 artifacts. Look-ahead limiting produces less harmonic distortion but produces IM distortion if over driven resulting in a packed, busy sound.

The DSPXtra-FM can be configured so that each peak control path can be routed to either of the digital or analogue outputs. The stereo encoder is always fed with the 'FM' path. One popular configuration for FM radio stations is to use the DSPXtra-FM to process their FM broadcast and to have the 'DR' path feed their digital radio service or web stream, each service optimally processed for that medium. We suggest that digital radio services always use the 'DR' path but you are free to experiment with both options.

The Processing Path

Input selection and conditioning

The DSPXtra-FM 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, phase rotating and pre-emphasis filters. A silence detector provides automatic primary to secondary input failure switching.

Ariane RMS leveller.

For a transparent input levelling function, the DSPXtra-FM 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 levller can also provide stereo enhancement through its matrix mode of operation.

Bass enhancement

The DSPXtra-FM 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-FM 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.

Distortion controlled clippers (PEAK CONTROL PATH 1)

The DSPXtra-FM clipping algorithms peak limit (clip) and linear phase filter the audio in three bands for maximum distortion control before being fed to the final clipper stages.

Look-ahead limiter (PEAK CONTROL PATH 2)

The DSPXtra-FM 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. A cut shelving filter is provided to compensate for the effect of pre-emphasis when the DSPXtra-FM is used to process FM signals at the same time.

Output selection, processing and routing

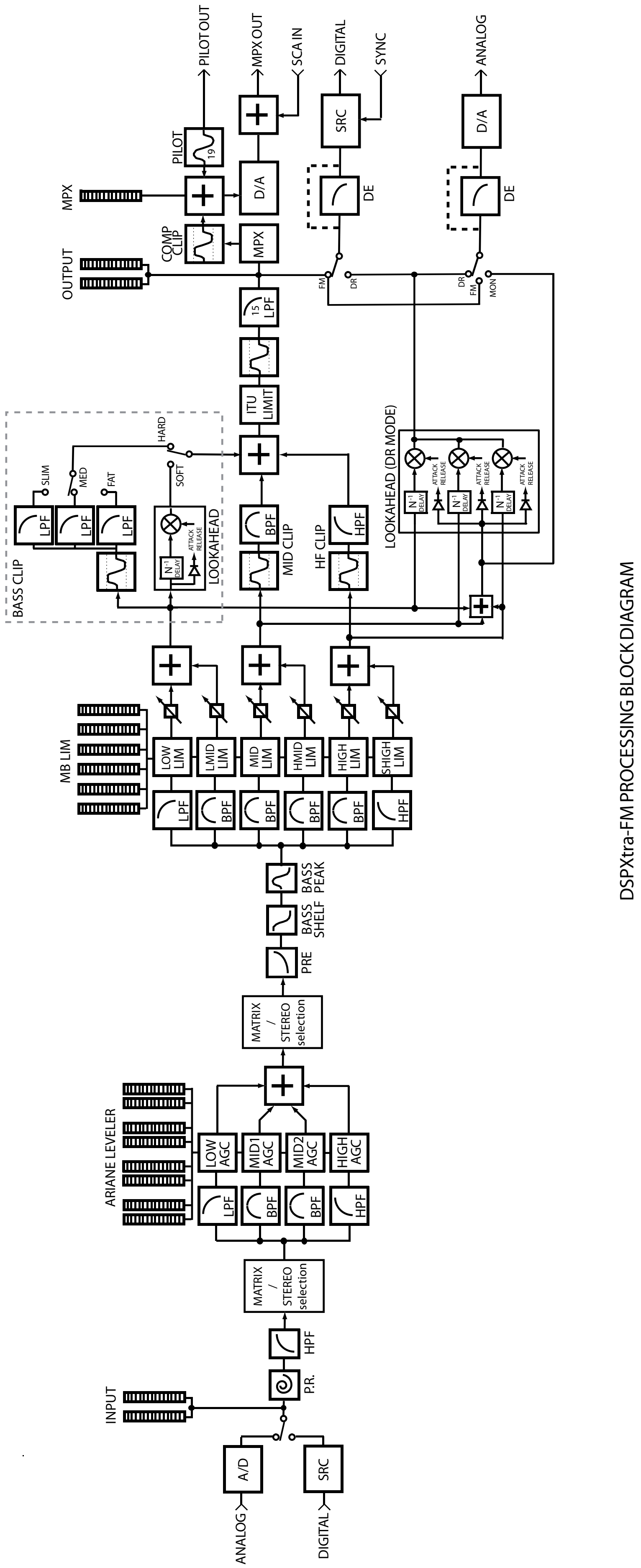
The DSPXtra-FM allows the user to select where each processing path is routed to and provides output level controls. A de-emphasis option is provided on both the analogue and digital outputs. The digital output sample rate can also be configured to a variety of settings.

The analogue output also has one extra path that can be routed to its outputs. This is a lower latency (delay) path that bypasses the final clipper stages and reduces delay by more than 4mS. This can be used to help with talent (DJ's) who can not get to grips with the delay of the main processing path. This output if selected should only be used as a studio monitor signal and should not be used ‘on the air’ as the peak clipping stage in the monitor processing path is not over-sampled or anti-aliased.

The analog and MPX outputs will be affected by the ITU limiter, if it is engaged. The purpose of this limiter is to comply with the ITU BS.412 standard. The standard calls for limiting of the power of the composite signal being broadcast. When activated, the ITU limiter will drastically reduce the loudness of your signal.

Stereo encoder

The DSPXtra-FM's DSP stereo encoder takes its inputs from the FM path of peak processing. The audio fed to the stereo encoder does not pass through any de-emphasis circuits. The stereo encoder is highly over-sampled and offers superb stereo performance. A composite clipping function is provided for those who wish to use it, as well as a pilot protection filter.


DSPXtra-FM PROCESSING BLOCK DIAGRAM
ANALOG
DIGITAL
A/D
SRC
INPUT
P.R.
HPF
MATRIX / STEREO selection
ARIANE LEVELER
LPF
BPF
BPF
HPF
LOW AGC
MID1 AGC
MID2 AGC
HIGH AGC
MATRIX / STEREO selection
PRE
BASS SHELF
BASS PEAK
MB LIM
LPF
BPF
BPF
BPF
BPF
HPF
LOW LIM
LMID LIM
MID LIM
HMID LIM
HIGH LIM
SHIGH LIM
BASS CLIP
LPF
LPF
LPF
SLIM
MED
FAT
HARD
SOFT
LOOKAHEAD
N-1 DELAY
ATTACK RELEASE
BPF
MID CLIP
HPF
HF CLIP
LOOKAHEAD (DR MODE)
ITU LIMIT
15 LPF
OUTPUT
FM
DR
MON
MPX
COMP CLIP
MPX
PILOT
19
PILOT OUT
D/A
MPX OUT
SCA IN
DE
SRC
DIGITAL
SYNC
DE
D/A
ANALOG

User Levels

The DSPXtra-FM 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!

Menu System Overview

This section presents an overview of the four main menus and thier 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.

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**
- **PRE-EMPHASIS**

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: LOOKAHEAD LIMITER**
  - **DRIVE**
  - **SHELF**
  - **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: CLIPPERS**
  - **MULTI-BAND CLIP DRIVE**
  - **BASS CLIP LEVEL**
  - **BASS CLIP TYPE**
  - **BASS CLIP SHAPE** (Advanced mode)
  - **MID CLIP LEVEL**(Advanced mode)
  - **HF CLIP LEVEL** (Advanced mode)
  - **HF CLIPPING**
  - **MAIN CLIP DRIVE**
  - **CLIPPER FINESSE** (Advanced mode)
  - **COMPOSITE CLIPPER** (Advanced mode)
  - **ADVNACED:**
    - **CLIP DISTORTION CONTROL**
    - **MAIN CLIP FINESSE**
    - **OVERSHOOT CONTROL**
    - **ITU LIMITER**

MENU: OUTPUT
- **MENU: ANALOG**
  - **OUTPUT LEVEL**
  - **MODE**
  - **DE-EMPHASIS**
  - **HEADPHONE LEVEL**
- **MENU: DIGITAL**
  - **OUTPUT LEVEL**
  - **MODE**
  - **RATE**
  - **DE-EMPHASIS**
- **MENU: STEREO**
  - **LEVEL**
  - **PILOT LEVEL**
  - **PILOT PROTECTION**
  - **ITU LIMITER**
  - **PILOT OUTPUT**

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-FM's A/D converter. This would normally be set to +24dBu if driving the DSPXtra-FM 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 <u>NEVER</u> 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 could otherwise cause carrier shift (AFC LOOP) problems in older FM transmitters. We suggest you leave the high pass filter off unless you have a reason to turn it on.

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 clipping 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.

Pre-emphasis: This parameter allows you to activate pre-emphasis when the DSPXtra-FM is being used to process for FM broadcast. The available options are 50 µs, 75 µs and OFF.

The 'PROCESS' menu allows access to all of the processing blocks that make up the DSPXtra-FM. 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-FM.

'ARIANE AGC'
The ARIANE AGC in the DSPXtra-FM like any automatic gain control device is designed to correct input level fluctuations and provide a consistent level to the other DSPXtra-FM processing blocks that follow it. The Ariane AGC in the DSPXtra-FM 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 clipping 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 peak clipping 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 **'LOOKAHEAD'** menu contains the controls that affect the look-ahead limiter which serves as the peak limiting method for the DR mode of operation.

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

'SHELF EQ' A high frequency shelving filter with a variable cut response. Adjustable over a range of 0-17 which corresponds to a cut of 0dB to -17dB measured at 15 KHz. This low-pass shelving filter is used to compensate for the effects of pre-emphasis when the DSPXtra-FM has pre-emphasis enabled (FM applications). The shelf control can also be used to tame high frequency energy when the box is configured for DR applications and no pre-emphasis is enabled.

'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 corresponds 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.

The **'CLIPPER'** menu contains the clipping controls that form the final peak limiting stages of the DSPXtra-FM's FM mode of operation.

'MULTI-BAND CLIP DRIVE' Controls the drive into the multi-band clippers that precede the main clipper. Adjustable over a -6dB to +6dB range.

'BASS CLIP LEVEL' Controls the clip level of the mix of Bands 1 and 2**.** The clip level range is -6dB to 0dB referenced to the main clippers output level.

'BASS CLIP TYPE' Controls the action of the bass clipper. Hard provides maximum punch by using the filtered harmonics of the clipped bass signal to create the illusion of more bass. Soft employs look-ahead techniques to dynamically control the bass clip level. This creates a more natural soft clip action which may be preferred for some formats. The soft clip option can raise the bass to 100% modulation as opposed to the hard option which does not allow bass to exceed the bass clip threshold. The soft option will add almost 5mS extra delay to the audio path.

'BASS CLIP SHAPE' Controls the shape of the bass clip transition. Fat is square topped traditional hard clipping and med and slim smooth out the clipping transition slightly and help reduce distortion a little.

'MID CLIP LEVEL' Controls the clip level of band 3**.** The clip level range is -12dB to 0dB referenced to the main clippers output level.

'HF CLIP LEVEL' Controls the clip level of band 4**.** The clip level range is -12dB to 0dB referenced to the main clippers output level.

'HF CLIPPLING' This control redistributes high frequency peak control between the six band limiters and the HF clipper**.** The range is 0 to 17 with 0 equating to full control by the limiters and higher numbers producing less control by the limiters on a frequency dependent basis, leaving control by the HF clipper. This control is similar to a variable de-emphasis control in the limiters side-chain control signals.

'MAIN CLIP DRIVE' Controls the drive into the main output clipper that defines the systems peak clipping ceiling. Adjustable over a -6dB to +6dB range.

'COMPOSITE CLIP Controls the drive into the composite clipper which effectively sets the amount of composite clipping. The range of composite clipping is -0.5dB to +2dB.

The **'ADVANCE'** menu contains the controls that fine tune final clipping

'MAIN CLIPPER DISTORTION CONTROL' Controls the distortion reduction effect of the distortion controller in the DSPXtra-FM's back-end clipping system. The range of multi-band clipping control is 1 to 10. Setting this control to 1 virtually defeats the mechanism, while higher numbers will progressively make the mechanism work on reducing the distortion and keeping the clarity and cleanliness of your on-air sound.

'MAIN CLIPPER FINESSE' Another distortion controlling mechanism that helps to reduce IMD in the final clipper. The range is 1-10 with 10 producing the most distortion control. A setting of 1 effectively bypasses this control and makes the clipper perform similar to the one on V1. This control is very subtle and may not appear to do a lot on some program material while a lot on others. HINT: Overdrive the main clipper to hear the effect of this control and then back the drive back down after the finesse control is set to your taste.

'OVERSHOOT CONTROL' Controls how much the main clipper will pass overshoots to the composite clipper. The range of overshoot control is 1 to 10 with lower numbers allowing more overshoots through to the composite clipper.

'ITU LIMITER' Sets the threshold of the integrated ITU limiter or turns it off. When activated, the unit will comply with the ITU BS.412 standard that requires limiting the power of the broadcast MPX signal. This limiter will drastically reduce the loudness of your signal and should be used only if the regulations in your country require you to comply with ITU BS.412. Otherwise set this parameter to off. The thresholds are calibrated in dB relative to the ITU BS.412 reference.

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.

'MODE' This parameter sets the output mode of the analogue and headphone outputs. The available options are FM, DR and MON representing the distortion controlled clipping path, the look-ahead limiting path and the lower latency (delay) talent (DJ) monitor path.

'DE-EMPHASIS' This controls the de-emphasis setting of the analog and headphone outputs. Options are Off, 50 μs (Europe) and 75 μs (USA)

'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 **'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.

'MODE' This parameter sets the output mode of the digital output. The available options are FM and DR representing the distortion controlled clipping path and the look-ahead limiting path respectively.

'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.

'DE-EMPHASIS' This controls the de-emphasis setting of the digital output. Options are Off, 50 μs (Europe) and 75 μs (USA).

The **'STEREO'** menu contains all the controls relevant to the DSP stereo encoder that generates the multiplex signal.

'OUTPUT LEVEL' Controls the output level of the composite MPX output. Range is 0dBu to +12dBu.

'PILOT LEVEL' This parameter sets the level of the composite signals 19 Khz pilot tone. The adjustable range is 6% to 12% and an OFF setting for mono applications.

'PILOT PROTECTION' Activates a narrow notch filter that protects the pilot in the multiplex signal. If you are using lots of composite clipping, this filter will protect the pilot region from being contaminated with harmonic products. As a consequence, you might need to readjust the MPX output level slightly as activating this filter may increase overshoots slightly (how much will depend on the amount of composite clipping used).

'ITU LIMITER' Sets the threshold of the integrated ITU limiter or turns it off. When activated, the unit will comply with the ITU BS.412 standard that requires limiting the power of the broadcast MPX signal. This limiter will drastically reduce the loudness of your signal and should be used only if the regulations in your country require you to comply with ITU BS.412. Otherwise set this parameter to off. The thresholds are calibrated in dB relative to the ITU BS.412 reference.

'PILOT OUTPUT' This controls the 19KHz pilot output reference signal which is available on the DSPXtra-FM back panel. Options are enabled and disabled**.**

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

'TIME' Sets the time and day of the DSPXtra-FM'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 ful.

'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-FM version number and design credits.

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

Setting up the Processing on the DSPXtra-FM

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

Because the DSPXtra-FM 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. Some processing experts believe that removing the very low frequency content from the program material improves the rest of the bass sound from the processor. The theory is that most people can't here or speakers can't produce the very low frequency bass. They believe that 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 here 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.

Some radio transmitters suffer from AFC bounce and overshoot when driven with high levels of very low frequency bass. If your transmitter suffers from this phenomenon you may need to turn your modulation down to accommodate these overshoots. The high-pass filter in the DSPXtra-FM can cure this problem by removing the very low frequency content from the program material

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 clipping 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 can 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 In FM broadcast, the transmission of low frequency information in the stereo subcarrier has been known to exacerbate multi-path interference. By reducing the level of low frequencies in the L-R channel, a subjective improvement in reception 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 Arianes outputs. Normally, this setting should be about -5 dB, but it can be set higher for a bigger stereo effect, or lower for reduced multi-path interference. 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-FM 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 clipping stages can result if the threshold is set too high. The range is +/- 6dB.

The multi-band limiters in the DSPXtra-FM 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 is takes the limiter to respond to a signal above the threshold and a decay or release which is the time is 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 optimum 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.

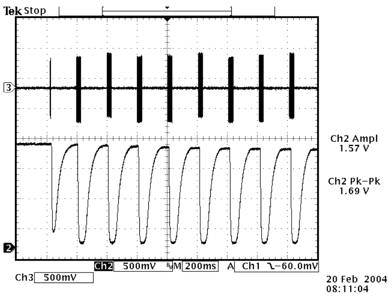
The average attack time is perhaps the most important control in the dual time constant detector as it sets the balance between peak and average energy in the detector. With smaller numbers more energy is transferred into the average circuit and a higher platform level is created so more time will be 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.

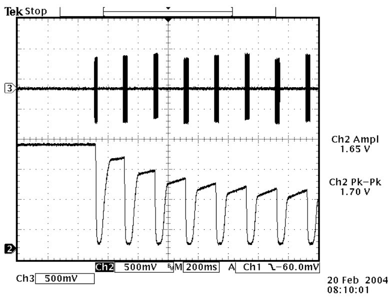
To recap, the peak attack time and average decay time play the same sort of role as that of a standard conventional single time constant based limiter.

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.

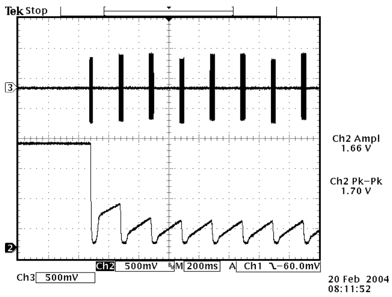
Limiter control signals response to tone bursts



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

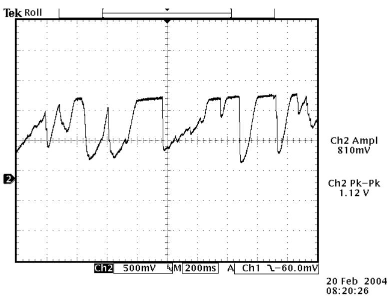


Peak time constants dominating to a lesser degree due to a high setting of average attack

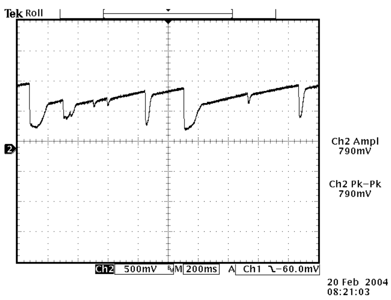


Peak time constants dominating to a much lesser degree due to a lower setting of average attack

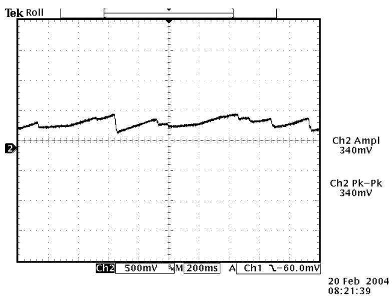
Limiter control signals response to program material



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



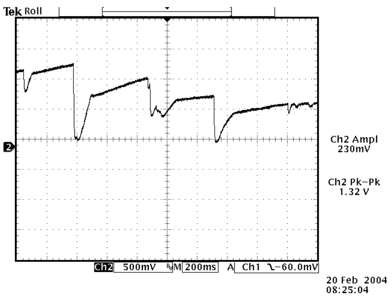
Peak time constants dominating to a lesser degree due to a high setting of average attack



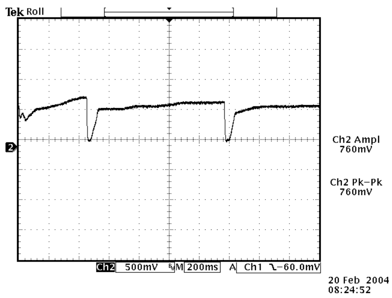
Peak time constants dominating to a much lesser degree due to a lower setting of average attack

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.



Without hold



With hold

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-FM 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.

The multi-band clipper
While the outputs of the multi-band limiters are peak limited they suffer from overshoot. The audio waveform can pass through before the limiter has time to 'attack' the signal. This is not a design flaw in the limiter but a required response. If you remember from our earlier discussions about audio processing you will recall that the processing can sound more natural and dynamic if a limiter lets sharp transients through rather than clamping down the whole audio signal whenever a transient occurs. These small transients can audibly dominate the limiting if the attack time of the limiter is too fast. It is much better to let these small transients through and deal with them in the next stage of processing.

The multi-band clipper that follows the multi-band limiters is designed to deal with these limiter overshoots and clip them to a pre-defined level. You may also recall from our earlier discussions on processing that clipping transients and overshoots is pretty much inaudible if done in moderation. The combination of the multi-band limiters and multi-band clippers provides us with the perfect answer. The limiters control the peaks of the audio but suffer from overshoot, the clippers then provide us with a true defined peak ceiling. Because the limiters precede the clippers we won't suffer from clipper overload induced distortion because the clippers are being fed with audio that has a pretty constant peak level. We also gain the advantage of more dynamic natural sound from the limiters because we are not worrying about overshoot and can set the attack times of the limiters higher than what would have been possible without the multi-band clippers.

Radio stations have a desire to be competitively loud and clipping is the easiest and most effective way in gaining loudness in a processor. While clipping is effective there is only so far you can push a clipper before noticeable distortion occurs. We can push this boundary of distortion back further by filtering out some of the distortion post clipper. By filtering after the clippers we are able to significantly reduce audible distortion.
The DSPXtra-FM multi-band clipper has three clippers and three post clipper filters. Bands 1 and 2 sum to serve the bass-clipper. Band 3 and 4 serve the mid-clipper and band 5 and 6 serve the HF-clipper. Low pass, band pass and high pass filters are used respectively.

The DSPXtra-FM has several controls that relate to the multi-band clippers. The first is a drive control and is self explanatory. It is a ganged level control that works in conjunction with the mixer controls. The 0dB drive level is a reference point that we choose that drives the multi-band clippers at a level that will produce a competitive amount of loudness. You may want to increase this if your goal is maximum loudness, listening carefully for distortion at the same time or decrease it when you do not want to process so heavily. The mixer level controls and multi-band limiters peak attack times and thresholds will have an effect in how much drive gets through to the multi-band clippers so you may need to compensate with the multi-band clipper drive should you adjust these.

Each of the clippers has a threshold control and these thresholds are referenced against the main output clippers clip level. For example, setting the bass-clipper at -6dB would allocate half of your available modulation level for the bass (mix of bands 1 and 2) and leave the remaining half for the sum mix of the mid-clipper and the HF-clipper. If these two clippers' clip thresholds were set so that they didn't add up to more than 50% modulation, say -12dB and -12dB then the main clipper would have no work to do as all of the peak control would be done with the multi-band clippers. By having defined peak clipper outputs we know that even with summation our peak level can only be the sum of each of those peak clipper outputs. In practice it is best to let the main output clipper do some of the work as a greater level of HF energy can be maintained. The best use of the multi-band clippers is to control the bass energy fully and to keep the mid-range and HF energy from causing excessive clipping distortion in the final clipper. We will discuss the bass-clipper next but before we do we would like to recommend mid and HF-clipper clip thresholds of between -8 and -3dB. Higher numbers produce more brightness but at the expense of greater distortion in the final clipper. A balance of multi-band clipping and final clipper clipping produces the best results.

Bass clipping
Most competent processors have a bass-clipper prior to the final clipper. The purpose of the bass-clipper is to keep low frequency energy to a pre-determined level to allow for the summation of the other bands. Without the bass-clipper the bass signal can push the mid and HF audio waveforms into the final clipper creating audible IM distortion, the worst type of distortion. By restricting the bass to a certain level the mid and HF energy has its own reserved space in the summated waveform and we reduce the likelihood of bass generated IM distortion.

The downside to bass clipping is you are restricting the bass to a lesser level than what it would be without it. The upside is that moderate levels of bass clipping won't cause a large loss of bass loudness and should have minimal audible artefacts.

When bass-clipper is being driven more aggressively you will start to notice distortion generated. This distortion can be used to actually give the illusion of more bass, especially on smaller radios that are incapable of producing the lower frequency fundamental bass waveform. This can be viewed as an upside of bass clipping. You need to decide what level of bass clipping is acceptable to your format, both in creating room for summation from the other bands and making the punch/distortion trade-off. We have been discussing a conventional bass-clipper configuration and this is referred to as the hard bass-clipper option in the DSPXtra-FM.

There is one more bass clipping option and that is known as the ‘SOFT’ option and uses look-ahead limiting. Look-ahead limiting produces a soft-clipping function on the bass and this significantly reduces distortion in the bass-clipper. There are a couple of down sides to this option. The first is latency (delay) as the DSPXtra-FM needs extra time to look-ahead to make the decisions to control the waveform before it arrives. The second is we don't get the bass punch feature we spoke of earlier as there is less harmonic distortion generated. What we do get is cleaner bass and the ability to use the look-ahead calculation time to modify the bass clip level dynamically to let the bass fill in to 100% modulation when there is no mid or HF content taking up waveform real estate. When you set the bass clip level in soft mode you are not setting the maximum level of bass as in hard mode but actually setting the maximum level it can be turned down to in the presence of mid and HF energy.
For example, if the bass clip level was set to -4dB and the audio waveform only contained bass, the bass-clipper level would raise to 0dB and you would obtain 100% modulation with the bass. If the audio waveform had mid or HF content the bass-clipper level would dynamically reduce to make room for the mid and HF content but could only reduce by an amount equal to that of having a fixed bass-clipper threshold -4dB. This maximum amount of

reduction feature stops mid and HF energy from over controlling the bass.

You can think of the bass-clip threshold in the same way regardless of what mode you use. Just think of it as creating space for the other bands.

What mode should I use? If you can live with the delay then try both and see what sound you prefer. Talk radio and softer formats such as easy listening usually sound better with the soft option. Dance and urban formats can benefit from the added punch generated by the controlled distortion with the hard bass-clipper. If you prefer the soft option but the extra delay makes it uncomfortable for your DJ's then you can consider using the lower delay monitor output from the DSPXtra-FM as a studio feed.

The final clipper

The final clipper, used in the FM processing path is a sophisticated highly over-sampled peak limiter that incorporates distortion controlling techniques and has an embedded 15 kHz low-pass filter. This section of processing is the last line of defence in the processing and is also the most critical part in the loudness/quality trade-off. While each of the proceeding processing stages play a part in reducing the peak to average ratio of the audio waveform none has the same effect on the peak to average ratio as the final clipper.

Great care is needed in setting the final clipper 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. 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 that you make final clipper drive adjustments in tandem with the multi-band clipper drive as what is taken from or added from one can usually be made up for with the other.

The final clipper now has an additional control to help reduce IMD distortion. This clipper finesse control is an additional program dependent mechanism that helps to reduce distortion by analysing the the level of IMD distortion and attempting to lower it by controlling how much the low frequencies can push the higher frequencies into the clipper. The control is very subtle and its range has been limited to restrict the amount of control, preventing pumping and a loss of loudness which would undo what we want to use the clipper for which is gaining loudness.

You may not notice the effect of this control on all program material. When adjusting the clipper finesse control we recommend that you turn the final clipper drive up past the point that you have it set at. This will make the effect of the finesse control much more obvious and allow you to find the setting that sounds best for your format. Once the clipper finesse control is set you can back down the final clipper drive to the point that sounds best knowing that the clipper finesse control has been set correctly to help keep the distortion down on difficult program material.

There is also an overshoot control that trades off distortion between final clipper and composite clipper. Letting more overshoots to be handled by composite clipper can sound a touch nicer, albeit at the expense of slight contamination of the output spectrum.

Look-ahead limiter

A look-ahead limiter is used in the DSPXtra-FM to provide peak control for the DR path of processing. This is not just a simple single band peak limiter. The DSPXtra-FM look-ahead limiter works in three bands to maximise the quality of the processing while minimalising audible artefacts.

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-FM 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 minimalises 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 secondary 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.

One of the most important features of the look-ahead limiter is the shelf filter control. This is effectively an adjustable low pass filter that is used to tame high frequencies. The shelf is needed when the DSPXtra-FM has its pre-emphasis control engaged, for several reasons. The first being that the DR processing path usually serves a broadcast medium that does not have de-emphasis so outputting pre-emphasised audio will not sound very nice. Secondly, the pre-emphasised audio will dominate the peak control of the look-ahead limiter and create spectral modulation pumping. The shelf allows us to compensate for the effect of this increased high frequency energy restoring a more natural tonal balance to the DR processing path. The shelf controls range is 0 to -17dB with negative numbers equating to more cut. The numbers are the gain reduction being applied at 15 KHz. As an example a processor setup for DUAL use that had its pre-emphasis set to 75uS would require a shelf cut of between -17dB and -14dB to restore a more natural tonal balance and 50uS pre-emphasis would probably require -15dB to -12dB. Even if the processor does not have pre-emphasis enabled you may find that you may want to experiment with small cut figures of between -4dB to -1dB to tame the high frequency energy that may have built up in the multi-band processing.

We need to draw your attention to one other control that can have an effect on the look-ahead limiter and which may not be immediately obvious. The HF clipping control (located in the clipper menu) redistributes control of high frequency energy from the multi-band limiters into the following processing stage. For the FM path this HF energy gets handled by the HF clipper but in the case of the DR path and its look-ahead limiter this high frequency energy can modulate the rest of the audio waveform and introduce a pumping type sound. If you are using the processor to process for DUAL services (FM and digital) then you can use the shelf control to compensate for pre-emphasis. If you are not using pre-emphasis (non FM use) then do check the HF clipping control if you are experiencing excessive HF energy and/or pumping from the look-ahead limiter.

ITU BS.412 limiting

The ITU BS.412 standard recommends that the power of baseband composite signal integrated over any 60 seconds interval, does not exceed the power in a sine wave that modulates the carrier to +/- 19 kHz. This requires significant reduction in loudness compared to the usual requirement of maximum +/- 75 kHz peak deviation only. The ITU limiter in the DSPXtra , when activated, will reduce the loudness of your broadcast signal and maintain its power at the level required by the BS.412 standard. It will do so with the slow limiter to avoid noticeable pumping which is the consequence of the standard requiring measurement of the pre-emphasized composite signal and not employing any frequency weightning to make it more natural sounding.

When activated, the ITU limiter will reset the Multiband Clipper Drive, Main Clipper Drive as well as Composite Clipper Drive to their default values. Keep in mind that even if you then turn the ITU limiter back off, those values will not return to the values they were before you tuned on the ITU limiter, but will remain at their factory defaults. This means that if you have a custom preset that changes the values of those parameters, turn the ITU limiter on and then back off, it will not sound the same as the mentioned parameters will be reset to their factory values.

The ITU limiter in DSPXtra will preserve the sound texture of your preset (excluding the distortion and effects of the main clipper). However, as the ITU limiter reduces the power of your signal anyway, you may wish to relax the processing and actually make use of the increased rms-to-peak ratio. We would therefore advise raising the MID and HF clipper thresholds to -1 dB and BASS clipper threshold to -3 dB. You may also want to increase the limiter peak attacks to let more of the transients through.

Composite clipper

The composite clipper in the DSPXtra-FM's stereo encoder is highly over-sampled and allows you to gain an extra dB or two of modulation loudness when using the multiplex output to drive your FM transmitter. The range of the composite clipper is -0.5 to 2dB.

Here at BW Broadcast we prefer to use the multi-band and final clipper to generate the required amount of loudness because composite clipping introduces stereo crosstalk. When we do use the composite clipper we prefer to use small amounts of no more than 1dB. A pilot protection filter is provided which will protect the pilot region from contamination by composite clipping products.

GETTING THE SOUND YOU WANT

While the DSPXtra-FM 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. For FM broadcast this is +/- 75 KHz for 100% modulation and for digital services, 0dBfs.

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 be whilst 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 clipper 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 clippers.
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 clippers.
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 clippers 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 some where.

Final clippers: (FM USE)
Increase the multi-band clipper drive.
Possibly switch the bass clipper to hard mode.
Raise the bass, mid and HF clip levels.
Increase the final clipper drive and reduce its finesse control to a lower number.

Look-ahead limiter: (DR USE)
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.
Adjust the shelf control to obtain the most suitable HF/loudness balance.

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 clippers.
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 (lower thresholds) .
Increase the DELAY times.
Decrease the limiter thresholds, letting less through to the clippers.
Increase the band and channel coupling ratios, preserving the spectral and channel balance of the original program material.

Final clippers: (FM USE)
Decrease the multi-band clipper drive
Possibly switch the bass clipper to soft mode
Lower the bass, mid and HF clip levels to prevent less distortion in the final clipper while carefully listening for distortion generated in the multi-band clipper.
Decrease the final clipper drive
Increase the clipper finesse control

Look-ahead limiter: (DRUSE)
Decrease 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.
Adjust the shelf control to obtain the most suitable HF balance.

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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 and band 2 limiter drive .
Slow down the peak attack time for bands 1 and/or 2, letting more through to the clippers.
Speed up the release time of the average release time constant of those bands.
Slow down the band 1 and band 2 average attack time so that the peak time constant dominates the control signal providing faster control.
Decrease the band 1 and 2 HOLD level (higher thresholds) or turn it off.
Decrease the band 1 and 2 Delay time.
Increase the band 1 and 2 limiter threshold, letting more through to the clippers.

Bass clipper: (FM USE)
Increase the multi-band clipper drive.
Increase the bass clip level.
Set the bass clip mode and shape to suit the tone of the bass you prefer.

Look-ahead limiter: (DRUSE)
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.

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More TREBLE (HF)
We can obtain more high frequency energy in several ways. When processing for FM we need to use distortion controlled clipping to preserve as much of the high frequency content as possible, which will be removed by the de-emphasis curve in the listeners radios. The ‘HF CLIPPING’, band 5 and/or 6 ‘PEAK ATTACK’ and ‘THRESHOLD’ controls governs the amount of high frequency control distortion controlled clipping that is performed.

Digital radio users should avoid high HF clipping settings as they put extra workload on the look-ahead limiter. Dual use (FM and DR) users can compensate for the HF clipping control with the look-ahead shelf control.

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 5 and 6 limiter drives.
Slow down the peak attack time for band 5 and 6, letting more through to the clippers.
Speed up the release time of the average release time constant of those bands.
Slow down the band 5 and 6 average attack time so that the peak time constant dominates the control signal providing faster control.
Decrease the band 5 and 6 HOLD levels or turn them off.
Decrease the band 5 and 6 delay time.
Increase the band 5 and 6 limiter threshold, letting more through to the clippers.

HF clipper: (FM USE)
Increase the multi-band clipper drive.
Increase the HF clipper level.
Set the HF clipping control to higher numbers which shifts control from the limiters to the distortion controlled HF clipper.

Look-ahead limiter: (DRUSE)
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.

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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-FM is processing for FM broadcast. The sophisticated controls in the Ariane section of the DSPXtra-FM 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-FM 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 demultipexing leaves only a mono signal. Try it and see. You will notice that you can achieve quite a lot with this control and it is quite important when we consider multi-path. Unlike other processors we can use this control to control our maximum level of L-R energy in the stereo subcarrier. By processing the L-R to increase the stereo effect and then reducing the maximum level with the injection control we can create a consistent amount of stereo enhancement but with less level in the subcarrier which results in less multi-path.

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.

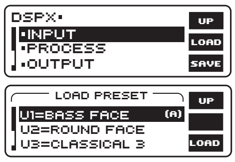
Less STEREO ENHANCEMENT

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. Don’t go too heavy on this setting as you could actually end up turning down any stereo content in the original program material when you are only trying to restrict enhancement.

Managing Presets (front panel control)

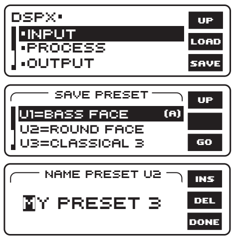
The DSPXtra-FM has an assortment of factory presets and provision for 8 user ones. While the factory presets may not suit your tastes you will generally find one that serves as a good starting point in creating your own custom preset. The preset facilities of the DSPXtra-FM are accessed from the three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

Factory Presets

The factory presets in the DSPXtra are not supposed to be de facto standards by any means but are starting points for you to create your own user presets. It is impossible to create presets that will suit every format and market. What is right for one market is not usually right for another. The staff at BW will be able to help you refine your sound further if none of the factory presets meet your requirement.

V2 contains the following factory presets.

PRESETS. These presets are optimised for FM use and most rely on reasonably high settings of the 'HF CLIPPING' control to maintain brightness through HF distortion controlled clipping. The look-ahead shelf control is usually set to lower numbers to compensate for the high level of HF introduced by pre-emphasis. This prevents the DR outputs from sounding too bright when the processor is being used in a dual processing capacity.

If you are using the DSPXtra for just HD/DAB, streaming or another digital medium that does not employ pre-emphasis like analogue FM and want to use one of these presets then we recommend setting the pre and de-emphasis parameters to OFF, lowering the HF clipping control and increasing the look-ahead shelf parameter closer to 0dB. The look-ahead shelf parameter adjustment assumes you will be using the DR mode on whichever output you choose to use, analogue or digital.

F1 BYPASS
This preset force gates the AGC's setting them to unity gain. The limiter and clipper thresholds are raised and drives are appropriately set so that the peak input to the DSPXtra matches the peak output of the DSPXtra.

F2 AC MA
A nice balance for lighter formats where loudness is not everything. Matrix mode in the Ariane.

F3 AC ST
A nice balance for lighter formats where loudness is not everything. Stereo mode in the Ariane.

F4 AC V.LOUD
A more competitive version of the above preset. Matrix mode in the Ariane.

F5 CHR MA
This preset enhances the bass and treble and provides a good starting point for pop music formats in competitive markets. Matrix mode in the Ariane.

F6 CHR ST
This preset enhances the bass and treble and provides a good starting point for pop music formats in competitive markets. Stereo mode in the Ariane.

F7 CLASSICAL1
Name speaks for itself. AGC operating in 2 band mode (through coupling), High IDR's and slow releases. Limiters lightly used just for protection purposes. No bass enhancement. Stereo mode in the Ariane.

F8 CLASSICAL2
Same as above but slightly faster release and lower IDR's help bring up quieter passages a little. Slight bass enhancement and EQ add a little life to the mix while still keeping as much of the original dynamics as possible. A little bit more competitive than the previous preset. Stereo mode in the Ariane.

F9 COUNTRY1
A nice balance. Easy to listen to with a nice tone. Stereo mode in the Ariane.

F10 COUNTRY2
Slightly more competitive with stereo enhanced through Matrix mode in the Ariane.

F11 DANCE MA
For stations who play club music and like there bass. Not the loudest but has a very solid low end! Matrix mode in the Ariane.

F12 DANCE ST
For stations who play club music and like there bass. Not the loudest but has a very solid low end! Stereo mode in the Ariane.

F13 DARE 2 TRY
A wall of destruction, hence the name. Really only suitable for demonstrating loudness. Heavy levels of clipping employed. Matrix in the Ariane.

F14 HOT MA
Nice EQ of bass and HF provide a good starting point for 80's formats. Matrix mode in the Ariane.

F15 HOT ROCK MA
More aggressive version of ROCK preset. Louder and a bit more alive. Matrix mode in the Ariane

F16 HOT ROCK ST
More aggressive version of ROCK preset. Louder and a bit more alive. Stereo mode in the Ariane

F17 HOT ST
Nice EQ of bass and HF provide a good starting point for 80's formats. Stereo mode in the Ariane.

F18 JAZZ ST
Somewhere between AC and Classical the Jazz preset maintains the original dynamics while adding a little life to the original material. Stereo mode in the Ariane.

F19 MELLOW ST
Controlled bass and highs with this preset will make your sound roll. Stereo is used in the Ariane.

F20 ROCK MA
Lots of HF and clipping going on here. Smash it up baby! Matrix is used in the Ariane.

F21 ROCK ST
Lots of HF and clipping going on here. Smash it up baby! Stereo is used in the Ariane.

F22 TALK
Fast releases and higher IDR's keep the voices where they should be, in the mix, but without pumping. Stereo is used in the Ariane.

F23 URBAN
A pleasing level of HF and Bass. Competitive but not too over-processed. Matrix is used in the Ariane.

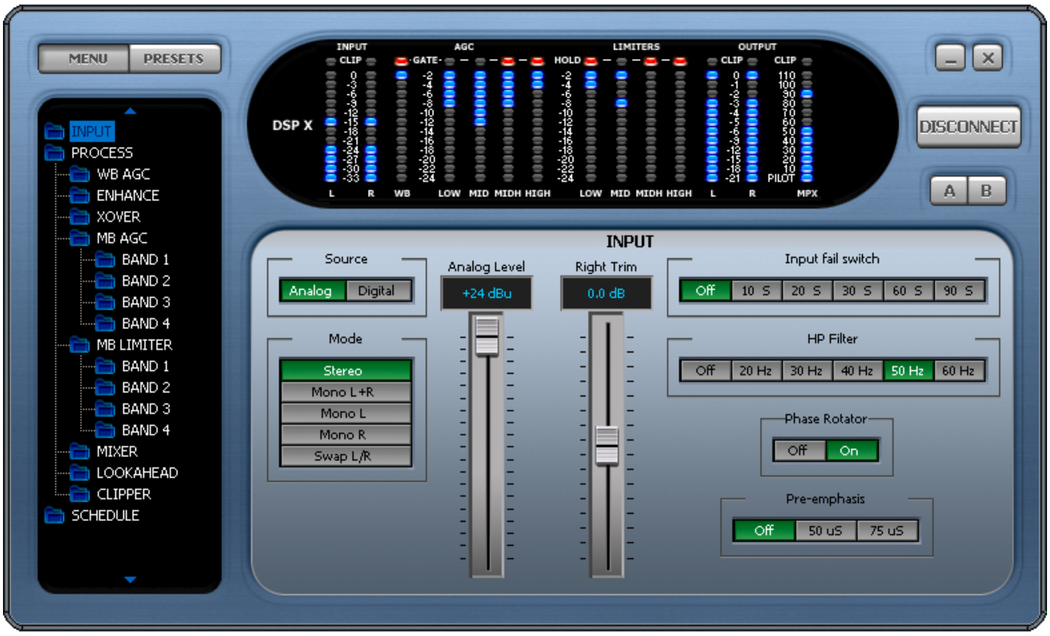
F24 WIDE OPEN MA
Maximum effect settings in the Ariane enhances the L-R energy to create a massive spatial effect. Be care-ful of multi-path if suing this unmodified, as the stereo subcarrier energy is huge. Matrix mode used in the Ariane.

Remote Control Of The DSPXtra-FM

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

The USB port on the front panel of the DSPXtra-FM is a USB/Serial convertor module. The USB signals from your computer are converted to serial commands which the DSPXtra-FM controller understands. This is transparent to the user but as far as the DSPXtra-FM 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-FM over USB. VCP driver Installation instructions are contained in Appendix A of this manual.

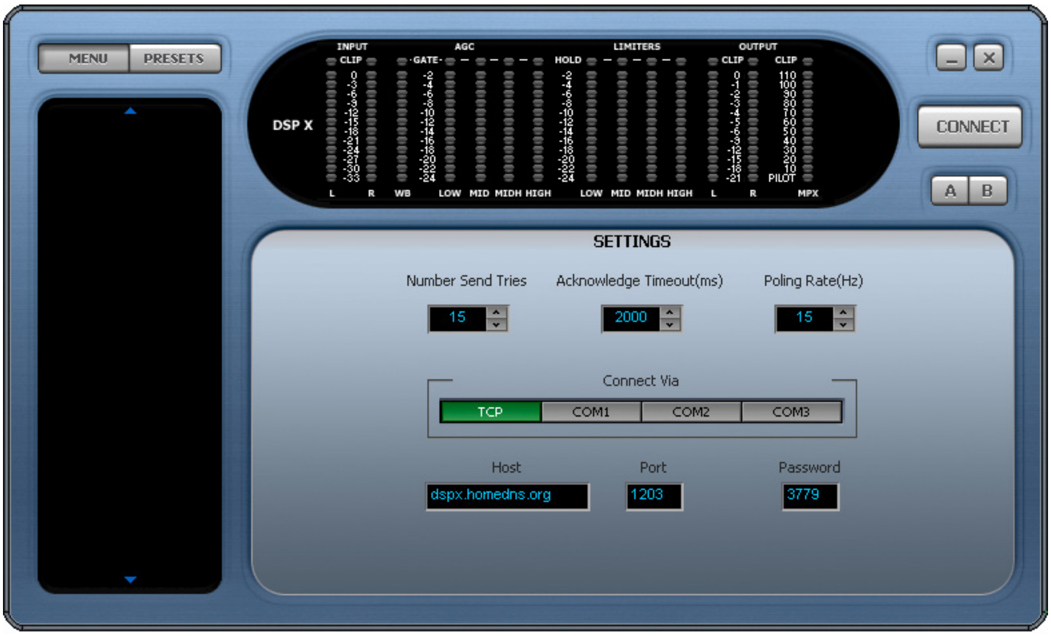
This section of the manual will refer to only serial and lan ports as the rear panel serial port and front panel USB port are the same to the DSPXtra-FM. they are only differentiated by the comm port they each have on the host computer. The DSPXtra-FM internally connects the back panel serial and front 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-FM can only to 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-FM's front panel.

If connecting via an ethernet connection you will need to set the IP address or Hostname that the DSPXtra-FM is connected at and also include the PORT number that the DSPXtra-FM 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.

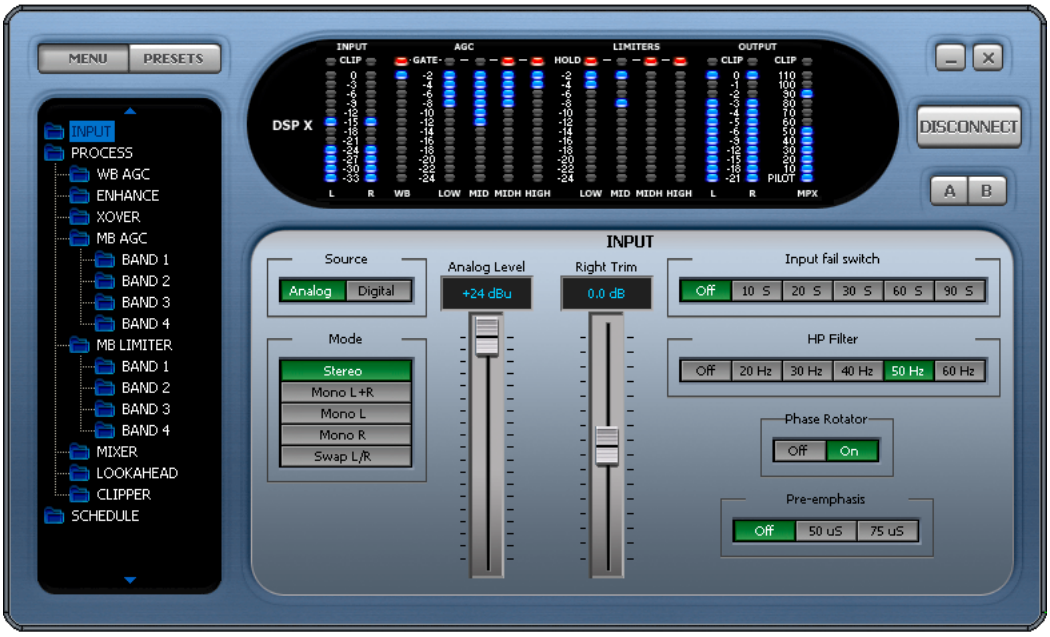


Remote application connection screen

CONTROL OF THE DSPXtra-FM BY RS232 or USB (SERIAL)
If you wish to use the RS232 or USB port to control the DSPXtra-FM 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-FM 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-FM itself. A password still needs to be entered even if the passwords on the DSPXtra-FM 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-FM. Once connected you are then free to control the DSPXtra-FM with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtra-FM 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-FM is 3779. You are free to change these on the DSPXtra-FM itself (see information on password control elsewhere in this manual)

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

CONTROL OF THE DSPXtra-FM BY THE NET/LAN PORT
If you wish to use the NET/LAN port to control the DSPXtra-FM follow the steps below
1. Connect a cat 5 cable to the RJ45 port on the DSPXtra-FM 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-FM and select the Network option.
3. Run the DSPXtra-FM 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-FM itself. A password still needs to be entered even if the passwords on the DSPXtra-FM 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-FM. Once connected you are then free to control the DSPXtra-FM with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtra-FM 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-FM is 3779. You are free to change these on the DSPXtra-FM itself (see information on password control elsewhere in this manual)



Remote application connected

PASSWORD ACCESS

The DSPXtra-FM contains two levels of password control, a high level password which blocks access to all areas of the DSPXtra-FM and an 'Output' level password that allows access to all areas of the DSPXtra-FM 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-FM.

These passwords can only be set from the DSPXtra-FM 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.

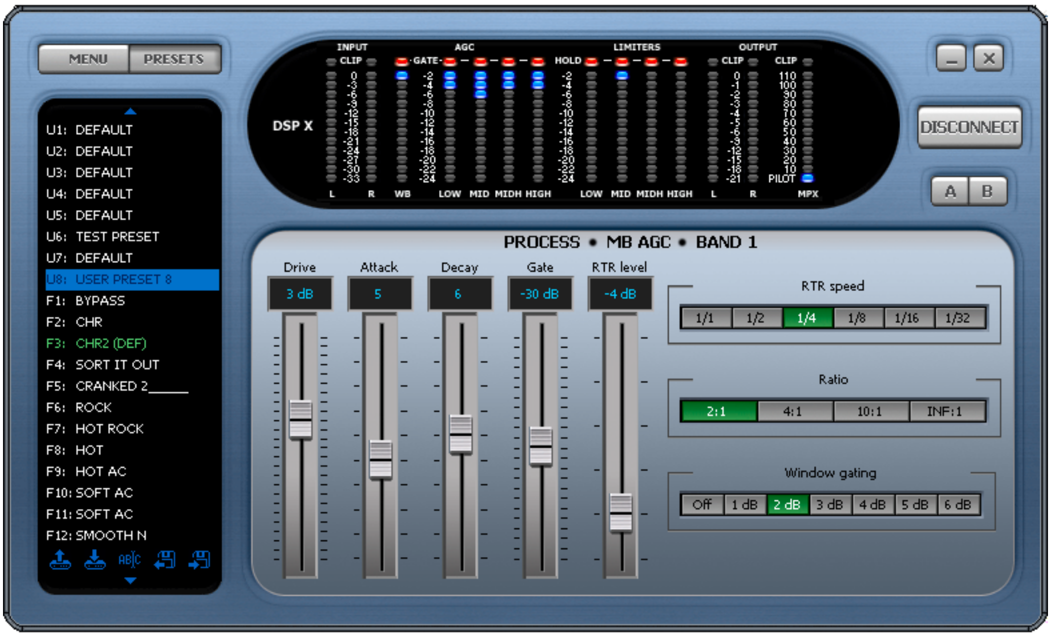
When the DSPXtra-FM is connected the LED's will show activity and the main controls window should show processing controls, depending on what option is selected on the menu tree located to the left of the screen. The DSPXtra-FM remote control application has three windows. The left hand contains the 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-FM 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-FM 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-FM 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-FM. 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.

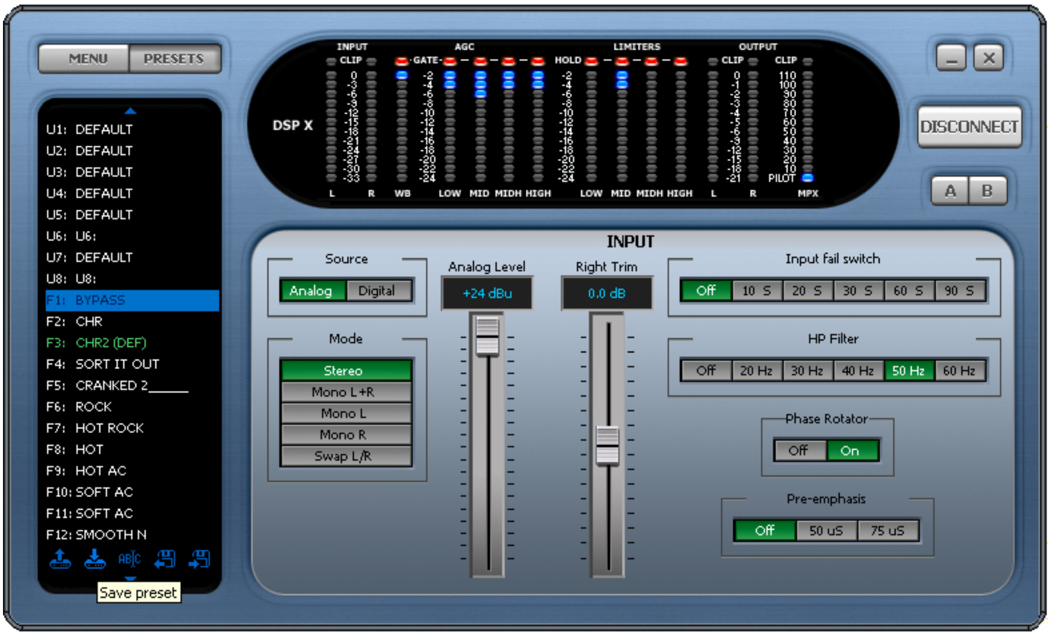
The currently 'on air' preset is always marked in green.

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-FM 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 if 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-FM 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.

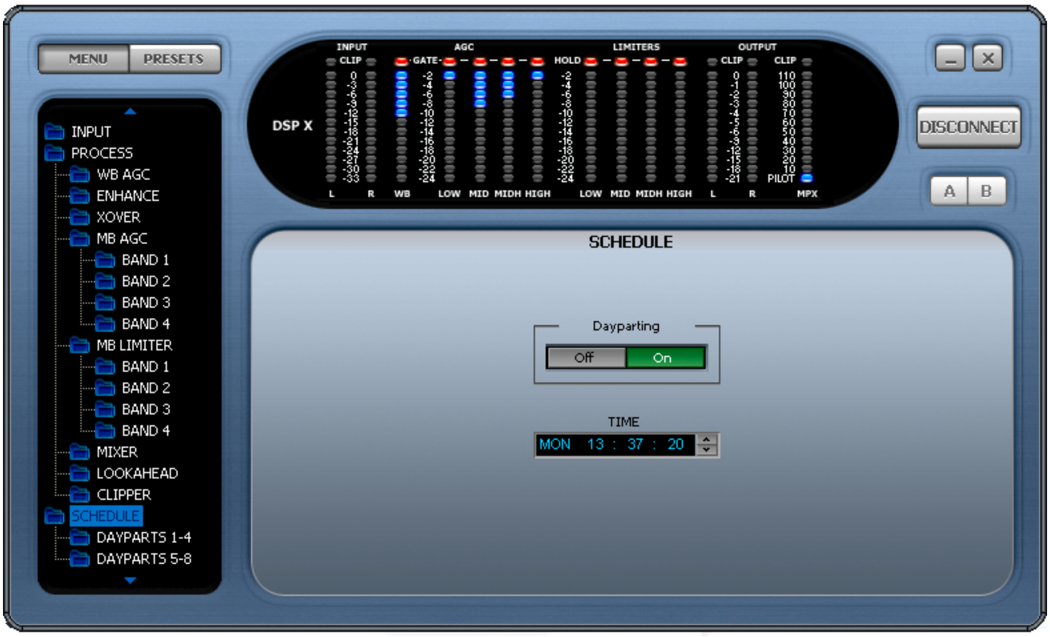
Rename preset
Rename preset allows the user preset name to be edited. You are restricted to 15 characters.

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-FM. 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.

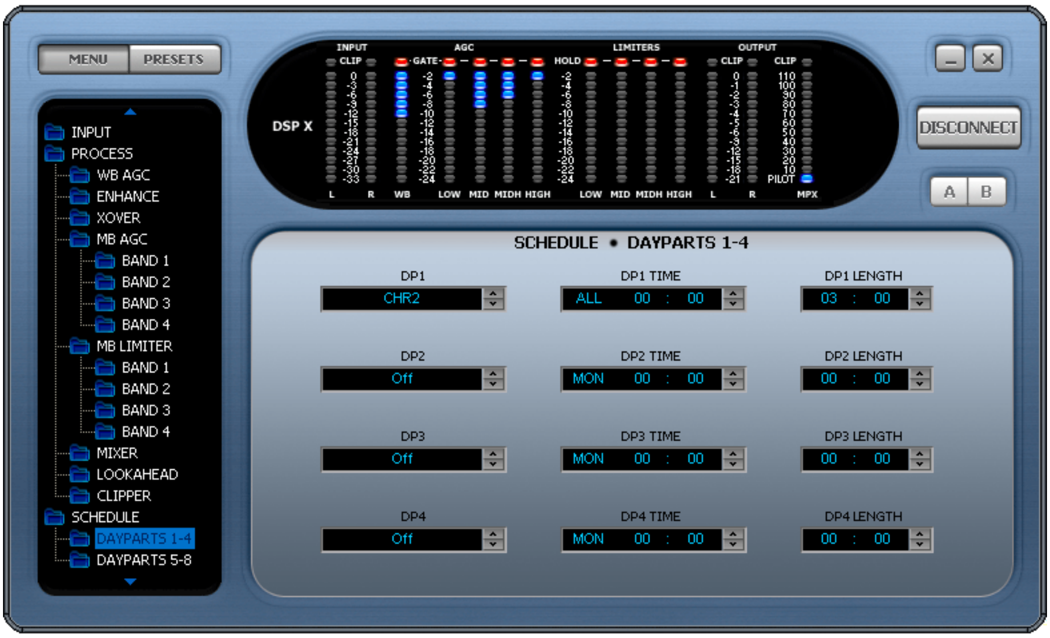
SCHEDULING WITH THE REMOTE APPLICATION

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-FM.



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

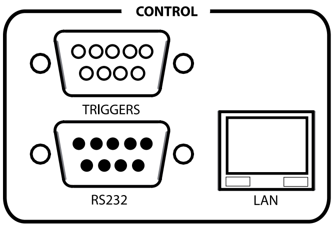
The Remote application has two buttons labelled A and B that are located just below the connection button. These buttons allow you to compare changes you have made to the processing against the saved preset. When you load a preset the buttons should be greyed out but as soon as you make any processing 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 adjustement just reload the preset from the preset selection window.

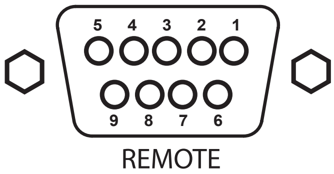
The A/B feature makes it easy to build up your own presets by being able to easily compare before and after processing adjustements. We hope you find it useful.

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

PIN 1	USER PRESET 4
PIN 2	USER PRESET 3
PIN 3	USER PRESET 2
PIN 4	USER PRESET 1
PIN 5	EARTH RETURN
PIN 6	USER PRESET 8
PIN 7	USER PRESET 7
PIN 8	USER PRESET 6
PIN 9	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-FM has two code locks, the main full lock and the output lock. When enabled the full code lock prevents editing of the DSPXtra-FM parameters. When enabled the output lock restricts access to the output menu section of theDSPXtra-FM. 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.

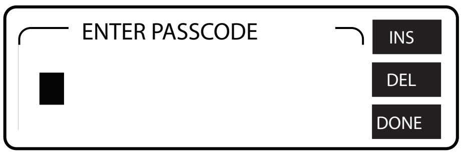
The security code locks if enabled will engage after 4 minutes of no activity on the front panel control system. This time has been chosen as a compromise between having the system lock you out when adjusting processing parameters and a short enough period of time to lock the unit after you walk away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

The DSPXtra-FM 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.

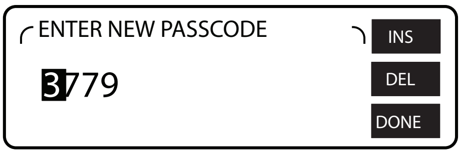
Enabling the security code lock feature: Navigate to the system menu and set the code lock option to enabled. Confirm the current password. The lock will engage after four minutes of no front panel control system activity.

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-FM 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-FM 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-FM should then be unlocked.



Entering the pass code



Confirming or changing the pass code

Forgotten DSPXtra-FM lock code:
Contact DSPXtra-FM support and request the procedure to reset the DSPXtra-FM lock code back to factory default. In resetting the DSPXtra-FM 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.

DATASHEET

bw broadcast

DSPXtra-FM | Audio Processor with Ariane Leveler

DSPXmini-FM	DSPX-FM	DSPXtra-FM
DSPXmini-AM	DSPX-AM	DSPXtra-AM
DSPXmini-HD	DSPX-HD	DSPXtra-HD

OVERVIEW

When you don't want to pay the high price of many "high-end" FM processors, yet can't afford anything less than the best for your digital broadcast services, the DSPXtra-FM is your best choice. No other FM processor, at any price, has both a BW Broadcast six-band look-ahead limiter and an **integrated Ariane RMS leveller**. The DSPXtra-FM harnesses 22 DSPs to keep your station's sound consistently loud and artefact free. In addition to multi-band limiting and the Ariane leveller, the processing core includes parametric and shelf bass enhancement, along with composite clipping with pilot protection. Dual processing paths make the DSPXtra-FM effectively two processors in one, feeding both your analogue transmitter and a digital service. Proven presets, as well as intuitive local and remote interfaces help you get your DSPXtra-FM on air fast.

FEATURES

- Dual processing path enabled for simultaneous FM & Digital processing
- 22 24-bit DSP's provide over 1 GIGA-MIPS of processing
- Comprehensive BLUE LED audio metering and screen
- Digital and Analogue IO
- Ariane intelligent RMS leveller
- Multi-Band program dependent Limiting
- Multi-Band look ahead limiting and distortion cancelled clipping
- DSP stereo encoder with composite clipping control
- USB front panel interface , RS232 back panel interface
- TCP/IP Net interface
- Real time clock for preset scheduling
- Remote trigger port
- Full Range of User Presets with A/B Switching
- Security (password) Control
- Software 'FLASH' Upgradable
- ITU BS.412 power limiter
- Pilot protection filter

SPECIFICATION

ANALOG INPUT MODULE

Operative nominal level	-15 to +24dBu
AD Clipping Point	>20dBu
Connectors	XLR
A/D conversion	24 bit 48 Khz 128x oversampled
Distortion	<0.01% THD

ANALOG OUTPUT MODULE

Output Level	-12 to 24dBu
Connectors	XLR
D/A conversion	24 bit 48 Khz 128x oversampled
Distortion	<0.01% THD
Impedance Source	10 Ohm / 20 Ohm / 200 Ohm

DIGITAL INPUT MODULE

Sampling rate	32-96 KHz
Connector	XLR
Operative Nominal Level	-24.0 to 0.0 dBFs
Distortion	<0.01% THD
Format	AES/EBU

DIGITAL OUTPUT MODULE

Sampling rate	32, 44.1, 48 KHz
Connector	XLR
Operative Nominal Level	-24.0 to 0.0 dBFs
Distortion	<0.01% THD
Format	AES/EBU

DIGITAL SYNC INPUT (AES/EBU)

Sampling rate 32-96 KHz	32-96 KHz
Connector XLR floating,EMI suppressed	XLR floating, EMI suppressed

STEREO ENCODER

Output level	0 to +12dBu
Connector	BNC, floating
D/A	24 bit 768 KHz
Stereo seperation	>70dB (typ. > 75dB)
Sca input	BNC floating
Pilot output	BNC, 5V
USB	USB B connector (front panel)

REMOTE INTERFACE

RS232	DB9 (rear panel)
Remote	DB9 opto isolated and floating
NET/LAN	RJ45 EMI shielded

Other

AC Rate	105-110VAC / 210-255 VAC, 50/60Hz
Size	44mm x 482mm x 200mm
Weight	1.6 Kg
Operating temperature	-5° to 50°C

DSPXtra-FM | Audio Processor

Green Ready Ultra Audio Plan B

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 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”.

DSPXtra-FM v2 preset sheet

	Preset 1		Preset 2		Preset 3		Preset 4	
INPUT								
SOURCE								
MODE								
ANALOG LEVEL		dBu		dBu		dBu		dBu
INPUT FAIL								
HP FILTER		Hz		Hz		Hz		Hz
PHASE ROTATOR								
PRE-EMPHASIS		µs		µs		µs		µs
PROCESS								
ARIANE								
MATRIX MODE								
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						
STEREO MODE								
IDR		dB						
GATE		dB						
ATTACK								
B1 RELEASE								
B2 RELEASE								
B3 RELEASE								
B4 RELEASE								
B1<B2 COUPLE		dB						
B3<B4 COUPLE		dB						
CHANNEL COUPLE		dB						
XOVER								
B1-2		Hz		Hz		Hz		Hz
B2-3		Hz		Hz		Hz		Hz
B3-4		Hz		Hz		Hz		Hz
ENHANCE								
DEEP BASS		dB		dB		dB		dB
BASS TUNE								
BASS PEAK GAIN		dB		dB		dB		dB
BASS PEAK FREQ		Hz		Hz		Hz		Hz
BASS PEAK Q								
MB LIMITER								
MASTER DRIVE								
B1<2 COUPLING		%		%		%		%
B2>3 COUPLING		%		%		%		%
B3>2 COUPLING		%		%		%		%
B3>4 COUPLING		%		%		%		%
L/R COUPLING		%		%		%		%
BAND 1								
DRIVE		dB		dB		dB		dB

	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
BAND 2									
	DRIVE		dB		dB		dB		dB
	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
BAND 3									
	DRIVE		dB		dB		dB		dB
	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
BAND 4									
	DRIVE		dB		dB		dB		dB
	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
BAND 5									
	DRIVE		dB		dB		dB		dB
	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
BAND 6									
	DRIVE		dB		dB		dB		dB
	THRESHOLD		dB		dB		dB		dB
	LIMIT. ATTACK								
	LIMIT. DECAY								
	COMP. ATTACK								
	COMP. DECAY								
	HOLD		dB		dB		dB		dB
	DELAY								
MIXER									
	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
CLIPPER									
	MB CLIP DRIVE		dB		dB		dB		dB
	BASS CLIP LEVEL		dB		dB		dB		dB
	BASS CLIP TYPE								
	BASS CLIP SHAPE								
	MID CLIP LEVEL		dB		dB		dB		dB
	HF CLIP LEVEL		dB		dB		dB		dB
	HF CLIPPING								
	MAIN CLIP DRIVE		dB		dB		dB		dB

COMPOSITE CLIP		dB		dB		dB		dB
ADVANCED								
MAIN CLIP DIST CTRL								
MAIN CLIP FINESSE								
OVERSHOOT CTRL								
ITU LIMITER		dBr		dBr		dBr		dBr
LOOKAHEAD								
DRIVE		dB		dB		dB		dB
SHELF EQ		Hz		Hz		Hz		Hz
LOW ATTACK								
LOW DECAY								
MID ATTACK								
MID DECAY								
HIGH ATTACK								
HIGH DECAY								
OUTPUT								
STEREO								
LEVEL		dBu		dBu		dBu		dBu
PILOT LEVEL		%		%		%		%
PILOT PROTECTION								
ITU LIMITER		dBr		dBr		dBr		dBr
19kHz SYNC								

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www.bwbroadcast.com