DSPXtreme-FMHD

Broadcast Audio Processor

Operational Manual
Version 1.30

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 two (2) years 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 unit has a manufacturers fault within twenty eight (28) days then BW Broadcast will pay the freight at their discretion.

If the warranty claim proves to be justified, the product will be returned to the user freight prepaid. Warranty claims other than those indicated above are expressly excluded.

Return authorisation number
To obtain warranty service, the buyer (or his authorized dealer) must call BW Broadcast during normal business hours BEFORE returning the product. All inquiries must be accompanied by a description of the problem. BW Broadcast will then issue a return authorization number.
Subsequently, the product must be returned in its original shipping carton, together with the return authorization number to the address indicated by BW Broadcast. Shipments without freight prepaid will not be accepted.

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

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

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

Damages/defects caused by the following conditions are not covered by this warranty:
Misuse, neglect or failure to operate the unit in compliance with the instructions given in BW Broadcast user or service manuals. Connection or operation of the unit in any way that does not comply with the technical or safety regulations applicable in the country where the product is used. Damages/defects caused by force majeure or any other condition that is beyond the control of BW Broadcast. Any repair or opening of the unit carried out by unauthorized personnel (user included) will void the warranty.

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

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

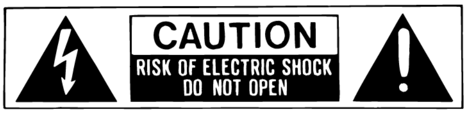
Warranty transferability
This warranty is extended exclusively to the original buyer (customer of retail dealer) and is not transferable to anyone who may subsequently purchase this product. No other person (retail dealer, etc.) shall be entitled to give any warranty promise on behalf of BW Broadcast.

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

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

SAFETY INSTRUCTIONS

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



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

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

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

DETAILED SAFETY INSTRUCTIONS:
All the safety and operation instructions should be read before the appliance is operated.

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

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

Follow instructions:
All operation and user instructions should be followed.

Water and Moisture:
The appliance should not be used near water (e.g. near a bathtub, washbowl, kitchen sink, laundry tub, in a wet basement, or near a swimming pool etc.).
The appliance should not be exposed to dripping or splashing and objects filled with liquids should not be placed on the appliance.

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

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

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

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

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

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

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

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

Damage Requiring Service:
The appliance should be serviced by qualified service personnel when:
- The power supply cord or the plug has been damaged; or
- Objects have fallen, or liquid has been spilled into the appliance; or
- The appliance has been exposed to rain; or
- The appliance does not appear to operate normally or exhibits a marked change in performance; or
- The appliance has been dropped, or the enclosure damaged.

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

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

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

CANADA WARNING: This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications. Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limits applicables aux brouillage radioelectrique edicte par le ministere des Communications de Canada.

Introduction To The DSPXtreme

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

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

The DSPXtreme 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 DSPXtreme to produce equvivalent results to other processors in the market but in a 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 DSPXtreme is driven by fast ARM7 and ARM9 micro-controllers which control an array of specialised analogue and digital circuits. These include 24-bit A/D and D/A converters, analogue level control circuitry, 24 x 24 bit DSP's, an ethernet port, an RS232 port, a trigger port, two 3" TFT LCD screens (one of which is touch sensitive), three sample rate converters, a headphone jack and memory devices to hold the software and firmware.

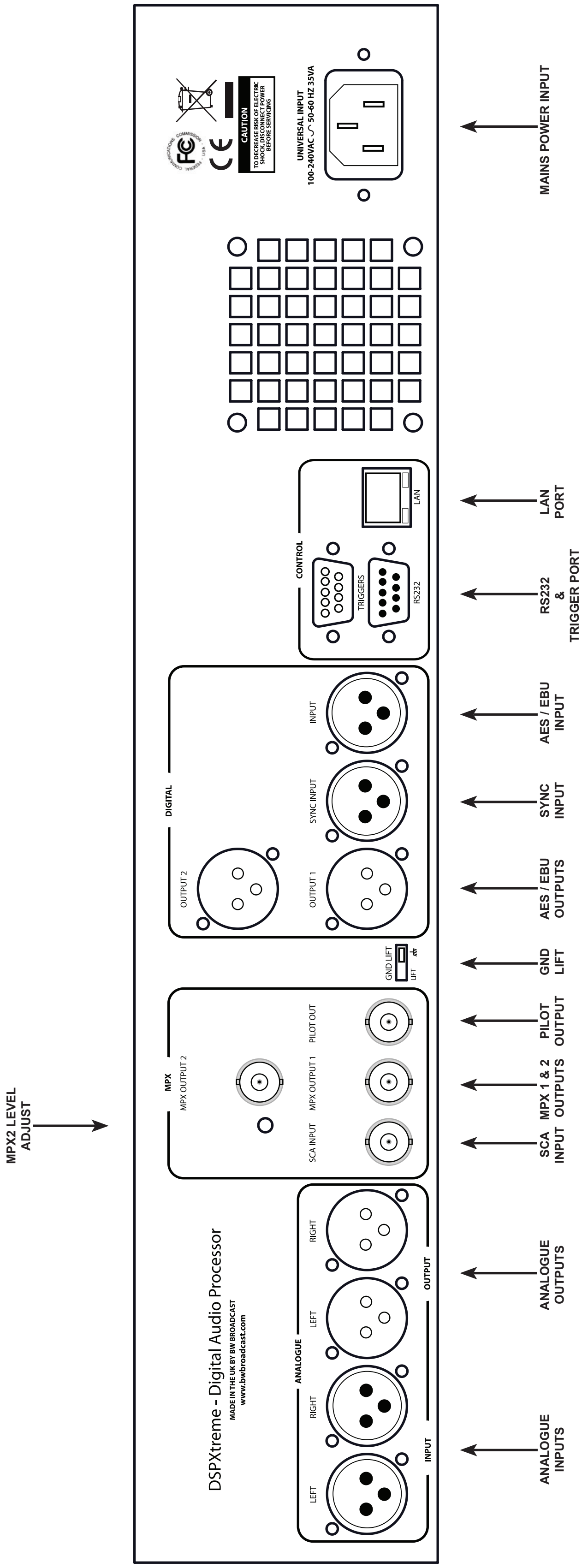
The Processing Architecture!
After input selection the 24-bit digital audio signal is passed through conditioning circuitry before being presented to the RMS levelling block. The four band windowed RMS leveller corrects for input level variations and also improves consistency. The output of the AGC feeds 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 touch-screen control system and a separate metering screen, 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.

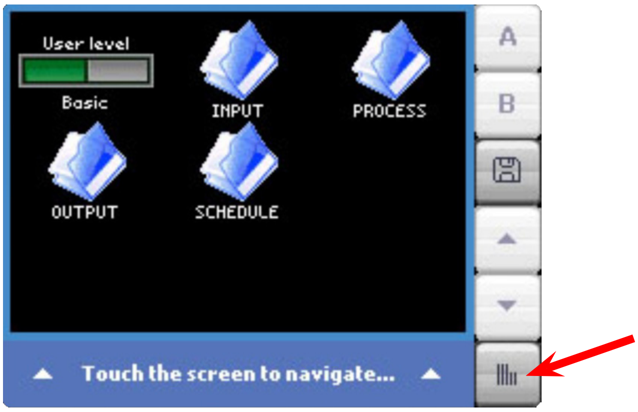
With superb specifications and big and loud on-air presence, the DSPXtreme is a clear winner.

Dynamic, fresh & innovative...
... The DSPXtreme


MAINS POWER INPUT
LAN PORT
RS232 & TRIGGER PORT
AES / EBU INPUT
SYNC INPUT
AES / EBU OUTPUTS
GND LIFT
PILOT OUTPUT
MPX 1 & 2 OUTPUTS
SCA INPUT
ANALOGUE OUTPUTS
ANALOGUE INPUTS
MPX2 LEVEL ADJUST
UNIVERSAL INPUT
CAUTION
CONTROL
TRIGGERS
RS232
LAN
DIGITAL
OUTPUT 2
OUTPUT 1
SYNC INPUT
INPUT
GND LIFT
LIFT
MPX
MPX OUTPUT 2
SCA INPUT
MPX OUTPUT 1
PILOT OUT
DSPXtreme - Digital Audio Processor
MADE IN THE UK BY BW BROADCAST
www.bwbroadcast.com
ANALOGUE
INPUT
OUTPUT
LEFT
RIGHT

DSPXtreme Meters

The DSPXtreme has a dedicated LCD screen on the left displaying instant IO and processing metering. You can switch from processing metering to IO metering (and back) with the button located on the bottom-right side of the touch-screen.



Change from I/O metering and GR metering with this button

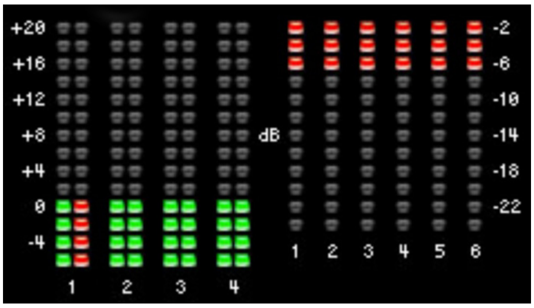
I/O metering
The input meters show the level of the input audio. The meters are 'hooked in' to the DSP code after the input level selection and mode options.

The 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 DSPXtreme. 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



GR meters

G/R metering - four band AGC
The gain reduction/boost meters show the gain reduction and gating status of the multi-band AGC stage. The range shown is -6dB to +20dB in 2dB steps with GATE or HOLD indicated by red color of the meters.

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 DSPX to process two seperate mono feeds.

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 DSPXtreme to process two separate mono feeds.

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 processir. 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.

Source Material Quality

The DSPXtreme has the ability to substantially improve the quality of your ON-AIR broadcast. However the DSPXtreme can only work with what you provide it. The best performance will be obtained when the DSPXtreme 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 DSPXtreme.

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 DSPXtreme can violate the frequency masking characteristics of the bit reduction process, creating distortion that was inaudible prior to the DSPXtreme processing.

Pre-emphasis

If you are using the DSPXtreme to process for FM broadcast you will need to enable the pre-emphasis filter in the DSPXtreme. Even though your STL or transmitter may contain pre-emphasis we recommend disabling it, letting the DSPXtreme handle the pre-emphasis for the transmission system. The DSPXtreme 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 DSPXtreme 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 DSPXtreme 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.

Final Lowpass Filtering

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

Output Level Setting

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

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

The DSPXtreme And Its Processing Structure

The DSPXtreme 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme. The DSPXtreme employs dual output paths for peak control. Your processing application may need you to configure the DSPXtreme 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme 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.

AGC xover

The DSPXtreme employs an adjustable gentle-slope digital IIR filtering to split the audio spectrum into 4 bands while maintaining sonic transparency.

Multi-band RMS leveller

For a transparent input levelling function, the DSPXtreme employs an RMS detected multi-band leveller. With a windowing gating concept, this four band AGC improves consistency and presents a uniform audio signal to the following stages.

Bass enhancement

The DSPXtreme 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.

Limiter xover

The DSPXtreme employs fixed 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme is used to process FM signals at the same time.

Delay

The DSPXtreme has an adjustable time delay for the MPX output that can be optionally selected for the analog and digital outputs as well. Delay should only be used if you're simulcasting analog FM and HD Radio (IBOC) and want synchronisation between these two transmission for perfect blends in the receiver. Separate menu allows adjustment of the delay up to 11 seconds, with coarse, medium and sample-accurate fine control. If you're not using HD Radio and transmitting only analog FM, leave the delay controls at 0 settings.

Output selection, processing and routing

The DSPXtreme allows the user to select where each processing path is routed to and provides output level controls. There are four choices for digital outputs to choose from - FM path, delayed FM path, DR path and monitor. For analog outputs the options are FM path, delayed FM path, same as AES2 and monitor.

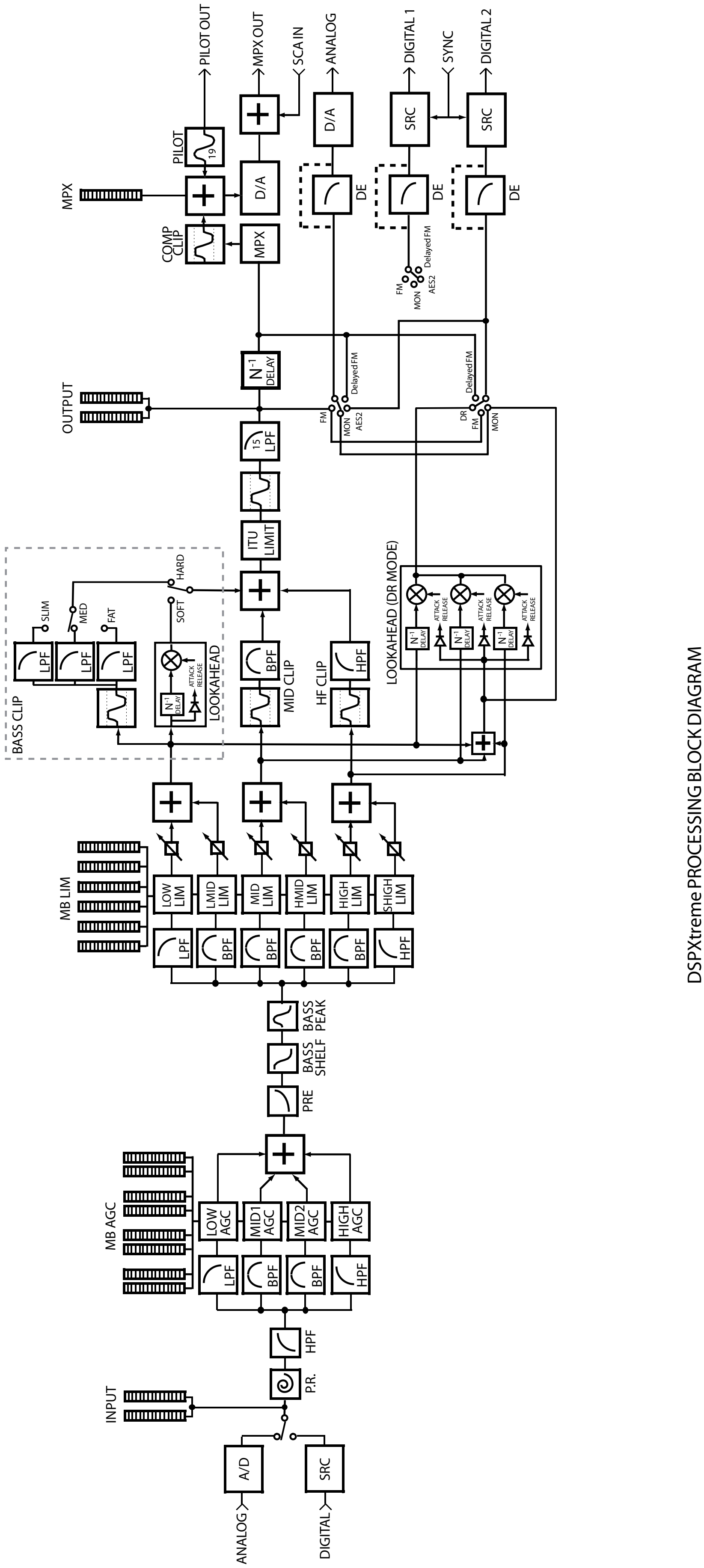
The monitor output is a lower latency (delay) path that bypasses the final clipper stages and reduces delay by more than 4 ms. This can be used to help with talent (DJ's) who can not get to grips with the delay of the main processing path. This output if selected should only be used as a studio monitor 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.

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 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 DSPXtreme'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 and is always delayed by the amount set in the Delay menu. 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.


DSPXtreme PROCESSING BLOCK DIAGRAM
ANALOG
DIGITAL
A/D
SRC
INPUT
P.R.
HPF
MB AGC
LPF
BPF
BPF
HPF
LOW AGC
MID1 AGC
MID2 AGC
HIGH AGC
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
SOFT
HARD
LOOKAHEAD
ATTACK RELEASE
DELAY
MID CLIP
BPF
HF CLIP
HPF
LOOKAHEAD (DR MODE)
ITU LIMIT
15 LPF
OUTPUT
DELAY
FM
MON
AES2
Delayed FM
DR
FM
MON
Delayed FM
MPX
COMP CLIP
MPX
PILOT
19
PILOT OUT
D/A
MPX OUT
SCA IN
D/A
ANALOG
DE
SRC
DIGITAL 1
SYNC
SRC
DIGITAL 2
DE
DE
FM
MON
AES2
Delayed FM

Menu Structure

- MENU: INPUT
  - INPUT SOURCE
  - STEREO MODE
  - ANALOG INPUT LEVEL
  - RIGHT TRIM
  - INPUT FAIL SWITCH
  - HIGH PASS FILTER
  - PHASE ROTATOR
  - PRE-EMPHASIS
- MENU: PROCESS
  - USER LEVEL
  - MENU: MULTI-BAND AGC
    - AGC ON/OFF
    - WINDOW GATING
    - GATE
    - B1<2 COUPLE
    - B3>4 COUPLE
    - CHANNEL COUPLE
    - MENUS: BAND 1-4
      - ATTACK
      - RELEASE
    - MENU: XOVER
      - B1-2 XOVER
      - B2-3 XOVER
      - B3-4 XOVER
  - MENU: ENHANCE
    - DEEP BASS
    - BASS TUNE
    - BASS PEAK GAIN
    - BASS PEAK FREQ
    - BASS PEAK Q
  - MENU: MULTI-BAND LIMITER
    - MASTER LIMITER DRIVE
    - MENUS: BANDS 1-6
      - DRIVE
      - THRESHOLD
      - LIMITER ATTACK
      - LIMITER DECAY
      - COMPRESSOR ATTACK
      - COMPRESSOR DECAY
      - HOLD
      - DELAY
  - MENU: MIXER
    - B1 MIX LEVEL
    - B2 MIX LEVEL
    - B3 MIX LEVEL
    - B4 MIX LEVEL
    - B5 MIX LEVEL
    - B6 MIX LEVEL
  - MENU: LOOKAHEAD LIMITER
    - DRIVE
    - SHELF EQ
    - LOW ATTACK
    - LOW DECAY
    - MID ATTACK
    - MID DECAY
    - HIGH ATTACK
    - HIGH DECAY
  - MENU: CLIPPER
    - MULTI-BAND CLIP DRIVE
    - BASS CLIP LEVEL
    - BASS CLIP TYPE
    - BASS CLIP SHAPE
    - MID CLIP LEVEL
    - HF CLIP LEVEL
    - HF CLIPPING
    - MAIN CLIP DRIVE
    - COMPOSITE CLIP
    - ADVNACED:
      - MAIN CLIP DISTORTION CONTROL
      - MAIN CLIP FINESSE
      - OVERSHOOT CONTROL
      - ITU LIMITER
- MENU: OUTPUT
  - MENU: ANALOG
    - OUTPUT LEVEL
    - MODE
    - DE-EMPHASIS
    - HEADPHONE LEVEL
  - MENU: DIGITAL
    - AES1 MODE
    - AES2 MODE
    - AES1 OUTPUT LEVEL
    - AES2 OUTPUT LEVEL
    - AES1 RATE
    - AES2 RATE
    - AES1 DE-EMPHASIS
    - AES2 DE-EMPHASIS
  - MENU: STEREO
    - LEVEL
    - PILOT LEVEL
    - PILOT PROTECTION
    - ITU LIMITER
    - 19K REFERENCE OUT
  - MENU : DELAY
    - DELAY COURSE (s)
    - DELAY MEDIUM (ms)
    - DELAY FINE (samples)
- MENU: SCHEDULE (X) REPRESENTS 0-7
  - TIME
  - DAYPARTING ON/OFF
  - TIME CALIBRATION
  - 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
  - TRIGGER PORT
  - REMOTE SOURCE
  - MENU: LAN CONFIG
    - IP ADDRESS
    - DEFAULT GATEWAY
    - SUBNET MASK
    - MAC ADDRESS
    - PORT NUMBER
  - 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 DSPXtreme's A/D converter clip celeing. This would normally be set to +24dBu if driving the DSPXtreme from professional audio equipment. This would translate to the wide-band AGC being driven to 12dB gain reduction by a 0VU (+4dBu) audio signal. If the input level is low, you may decrease this parameter to bring the level up and reduce signal-to-noise ratio. Make sure that the input audio meters NEVER show clipping under any conditions.

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

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

High pass filter: This parameter allows you to select from a variety of high pass filters. You can select from 20Hz, 30Hz, 40Hz, 50Hz and 60Hz. You also have the ability to bypass the high pass filter with the 'OFF' option. The high pass filter can be used to reduce rumble on vinyl or can be effective in removing low frequency energy that 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 DSPXtreme 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 DSPXtreme. There are only sub-menus inside the 'PROCESS' menu. The submenus are laid out in the same configuration as the signal path through the DSPXtreme.

The multi-band AGC in the DSPXtreme like any automatic gain control device is designed to correct input level fluctuations and provide a consistent level to the other DSPXtreme processing blocks that follow it. The AGC in the DSPXtreme is designed to operate in an undetectable, unobtrusive manner similar to a trained operator controlling a mixing desk.

'AGC BYPASS' If you will be using an outboard AGC or if a processor is placed on the other side of an studio-to-transmitter link with the AGC unit being used in front of the link, AGC in DSPXtreme can be bypass in order to avoid "figthing" with the preceeding AGC.

'WINDOW GATING' A separate gating control from the silence gate, the window gate slows down the attack and decay rates when the audio waveforms level has not moved outside of a pre-defined window, set by the window gating control. The window can be set between 1 and 6dB or the gate function can be disabled by setting the control to OFF. The window gating prevents unnecessary gain control when the audio waveform is already well defined or having a low peak to average ratio. The window gating also allows faster rates to be used without the audible effects normally associated with the faster rates.

'GATE THRESHOLD' 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 -20dB to -40dB. The gate can also be switched off or forced on. The gate when turned on will cause the gain reduction to move towards the level set by the RTR LEVEL control and to move towards that level at the rate set by the RTR speed control.

'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).

'B1-4'

'ATTACK' Controls the attack rate of the AGC, the time the AGC takes to respond to an increase of input level. The attack time can be varied between 1 and 10 which corresponds to 100 ms to 30 s on a semi-exponential scale.

'DECAY' Controls the release/decay rate 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 which corresponds to 100 ms to 30 s on a semi-exponential scale.

'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 **'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.

'THRESHOLD' Sets the limiter threshold. If audio is below threshold, there will be no gain reduction. Audio signals that go above will be attenuated. Threshold level can be varied between -6dB to +6dB in 0.5dB steps.

'LIMITER 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 1 ms to 200 ms on a semi-exponential scale.

'LIMITER 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 10 ms to 1000 ms on a semi-exponential scale.

'COMPRESSOR 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.

'COMPRESSOR 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 200 ms to 5000 ms 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 1 ms - 500 ms 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.

BAND 1 MIX: -3dB to +3dB of level adjustment is available.

BAND 2 MIX: -3dB to +3dB of level adjustment is available.

BAND 3 MIX: -3dB to +3dB of level adjustment is available.

BAND 4 MIX: -3dB to +3dB of level adjustment is available.

BAND 5 MIX: -3dB to +3dB of level adjustment is available.

BAND 6 MIX: -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 DSPXtreme 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.

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

'LOW 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 10 ms to 2000 ms on a semi-exponential scale.

'LOW 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 10 ms to 2000 ms on a semi-exponential scale.

'MID 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 10 ms to 2000 ms on a semi-exponential scale.

'MID 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 10 ms to 2000 ms on a semi-exponential scale.

'HIGH 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 10 ms to 2000 ms on a semi-exponential scale.

'HIGH 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 10 ms to 2000 ms on a semi-exponential scale.

The **'CLIPPER'** menu contains the clipping controls that form the final peak limiting stages of the DSPXtreme'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 5 ms 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 DSPXtreme'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, delayed FM, same as AES2 and MON representing the distortion controlled clipping path, delayed distortion controlled clipping path, same as selected on AES2, the look-ahead limiting path and the lower latency 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. DSPXtreme features two AES/EBU outputs, each of which is separately adjustable with the following controls:

'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, delayed FM, DR and MON representing the distortion controlled clipping path, delayed distortion controlled clipping path, the look-ahead limiting path and the lower latency talent (DJ) monitor path.

'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 main composite MPX output. Range is 0dBu to +12dBu. Secondary composite MPX output has an adjustable potentiometer available on the back of the unit.

'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.

'19k REFERENCE OUTPUT' This controls the 19 kHz pilot output reference signal which is available on the DSPXtreme back panel. Options are enabled and disabled**.**

The **'DELAY'** menu contains controls for setting the MPX output delay in order to align it with digital part of HD Radio (IBOC) transmission.

'DELAY COARSE' Adjusts the delay in seconds, from 0 s to 12 s in 0.1 s steps.

'DELAY MEDIUM' Adjusts the delay in miliseconds, from 0 ms to 99 ms in 1 ms steps.

'DELAY FINE' Adjusts the delay in samples, from 0 smps to 47 smps in 1 smps steps.

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

'TIME' Sets the time and day of the DSPXtreme'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

'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.

'REMOTE SOURCE' This selects the serial 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' DSPXtreme version number and design credits.

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

Setting up the Processing on the DSPXtreme

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

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 DSPXtreme 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 Multiband AGC

The multiband AGC in the DSPXtreme employs an RMS based level detector for superior performance. This enables the DSPXtreme to control input level variations based on the true loudness of the input waveform unlike other simpler average responding peak detectors used in other digital audio processors. Because of the RMS based level detectors the multi-band AGC can re-equalise the sound in a more natural manner than the peak limiter stages which use peak detectors. When you couple the advanced detector with the user adjustable and hidden intelligent controls you really do have a powerful levelling tool.

The multiband AGC stage of the DSPXtreme has two main functions:
1. To re-equalise the program material to provide a consistent tonal balance and sonic signature.
2. To prevent excessive limiting by the following peak limiter stages.

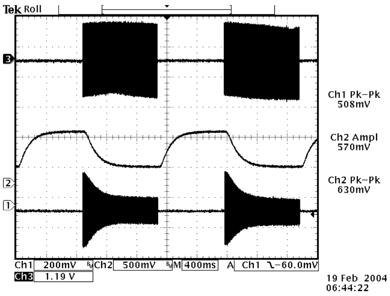
Over the course of the next few pages we have included several scope shots clearly illustrating the input and output of the AGC together with the AGC control signal. The effect of the control signal is clearly evident on the output audio waveform. These scope shots help to visually illustrate the concepts under discussion.

The first control that most people reach for is the drive control, stop! As the wideband AGC is designed to respond in a slow manner increasing the drive level won't help.

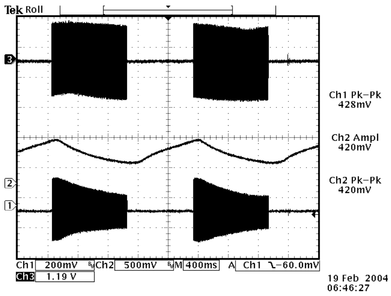
The attack and decay times of the AGC have a range of 1-10 and this corresponds to time constants of 100mS to 30S. We suggest an attack somewhere in the region of 3-4 and a decay setting of 1 or 2 positions higher than that.

Like most competent audio processors the AGC stages in the DSPXtreme are gated. This slows down the release time of the AGC when the program material drops below a certain level. This prevents noise suck up and gain hunting from occurring during quiet periods or lulls in the audio. The gate level control is the level at which the program material must fall below for the gate to become active, and this can be adjusted over a range of -20dB to -40dB. The gate level control has two more options, OFF and ON. OFF is self explanatory and prevents the gate from having any effect. ON is often referred to in this manual as 'forced gating' as it has the effect of switching the gate on at all times with any level of program material. This option is used to bypass the AGC and provide a fixed gain through it. The fixed gain (in dB) being that of the (input drive plus 12) minus the level

The effects of time constant speeds



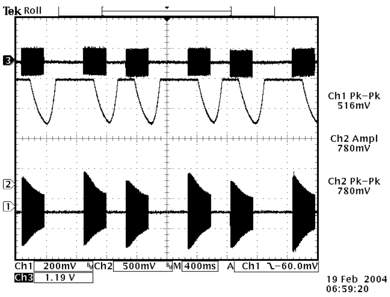
AGC attacking and decaying with faster time constants



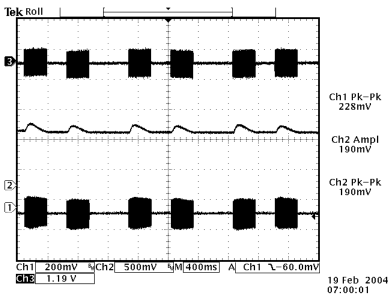
AGC attacking and decaying with slower time constants

in dB that the return to rest level is set at.
Window gating is separate from the silence gating that we have been discussing in that it does not work on the amplitude of the audio, rather the peak to average ratio of the waveform and how the waveform is changing over time. The window gating feature if enabled freezes the gain over a pre-defined range and will only let gain control commence once the waveform has fallen outside of this pre-determined range. This is useful because it does not apply gain correction to audio waveforms that are only changing by small amounts. The images below clearly

The effects of gating



AGC attacking and decaying with faster time constants



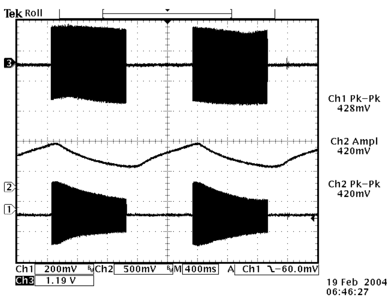
AGC attacking and decaying with faster time constants but gating enabled

illustrate the effect of window gating on the control signal that controls the output waveform. The size of the window can be set to 1dB, 2dB or 3dB with an 'OFF' option to turn the window gating off. Window gating has the extra advantage in that it enables us to use faster time constants than what would have been possible without it. Faster time constants have the disadvantage in that the constant re-adjustment of the waveform can become audible. With a higher window gating setting we are able to reduce the audibility of the faster time constants by freezing out these changes when the waveform is within our window. Once the waveforms falls outside our window the faster time constant will track it quickly and effortlessly and then the window gating will kick back in.

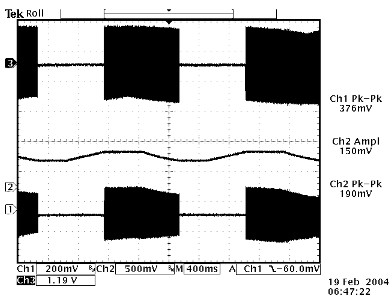
Band-couplings

The DSPXtreme band-coupling controls allow us to reduce the effects of the multi-band processing by coupling the bands to a certain extent. This may be desirable if we want to limit the re-equalisation effects of the multi-band AGC and limiters. By carefully selecting the coupling ratios we can also reduce any possible spectral skewing when the processing stages are driven into heavy gain reduction. The multi-band AGC and limiters each have there own band coupling controls

The effect of window gating



AGC attacking and decaying with faster time constants



AGC attacking and decaying with faster time constants but with a 2dB window gating setting

The coupling control controls the amount of audio that is fed from one bands detector into the neighbouring bands detector. The audio is fed from one bands detector through the coupling ratio control and then the highest level wins.

For example, if we coupled band 2 to band 1 with a coupling of 50% we would ensure that the band 1 gain reduction could never decrease (less gain reduction) more than 6dB past the band 2 gain reduction. If the coupling was set to 100% then the band 1 gain reduction would follow the band 2 gain reduction when the band 1 gain reduction would have been less than the band 2 gain reduction. Setting the coupling ratio to 0% would let both bands operate independently of each other.

We can limit the amount of low and high frequency re-equalisation by carefully setting the B1<2 and B4<3 coupling controls. If lighter processing is desired it is common to link the bands to a certain amount, where pop and CHR formats usually desire low or uncoupled ratios. Coupling ratios around 30% are usually a good compromise to maintain cut-to-cut consistency through multi-band re-equalisation while maintaining most of the original spectral balance of the source material.

Band-to-Band Coupling
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 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’).

Channel coupling
The control allows you to gang the left and right channel gain controls by a percentage set by the control. If set to 0dB, the coupling works on a highest level wins basis where the channel with the maximum gain reduction controls the gain reduction of both channels. This preserves the stereo balance of the original source material. The control can be adjusted all the way down to OFF at which point the two DSPXtreme multiband AGCs operate fully independently. This is often known as a dual-mono architecture.

While we would prefer you to buy two processors you could use a single DSPXtreme as two processors, processing two mono audio feeds. You would need to bear in mind if running the DSPXtreme this way that the front panel gain reduction metering may not behave as expected because the displayed gain reduction value for that processing block is the highest gain reduction of the left and right channels.

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 DSPXtreme 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.

Xover

The Xover in the DSPXtreme has five selectable frequencies per band split.. The band 1/2 split can be set to 80, 100, 125, 160 or 200 Hz. The band 2/3 split can be set to 400, 500, 650, 800 or 1000 Hz and the band 3/4 split can be set to 1100, 1400, 1800, 2250 or 2800 Hz.

The xover employs linear phase FIR filtering for audio transparency. Extra delay lines time align the audio bands ensuring a flat response across the whole audio spectrum regardless of gain reduction levels. Unlike most other audio processors the DSPXtreme maintains this linear phase time aligned property throughout all of its processing stages. This can be verified by the application of a low frequency square wave to the DSPXtreme's inputs and monitoring the flat top response on the DSPXtreme's outputs. You must make sure that all of the bass enhancement and input conditioning filters such as the phase rotator are switched to off before conducting this test. The bypass preset will do this for you (pre-emphasis and de-emphasis is not switched by a preset change).

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 DSPXtreme 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.

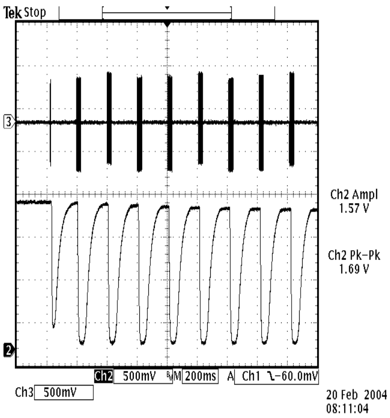
The average attack time is perhaps the most important control in the dual time constant detector as it sets the balance between peak and average energy in the detector. With smaller numbers more energy is transferred into the average circuit and a higher platform level is created so more time will be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

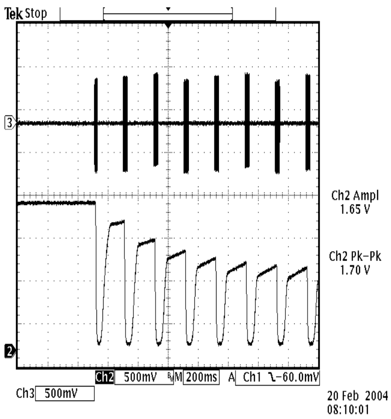
Hold

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'.

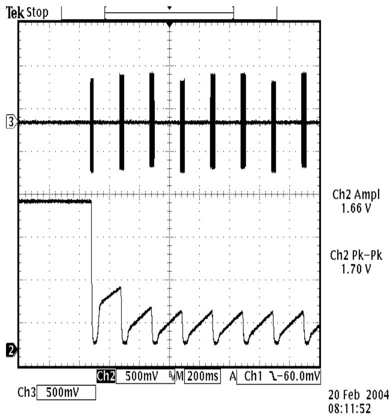
Limiter control signals response to tone bursts



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

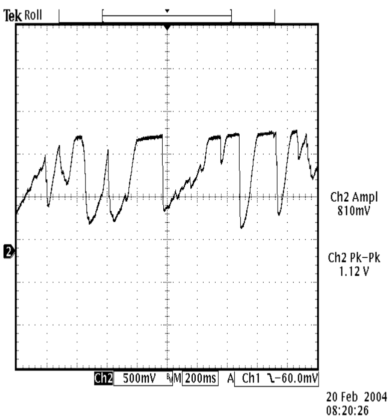


Peak time constants dominating to a lesser degree due to a high setting of average attack

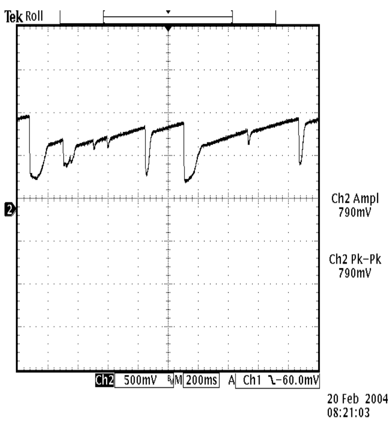


Peak time constants dominating to a much lesser degree due to a lower setting of average attack

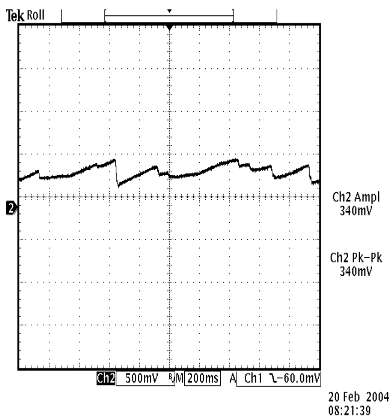
Limiter control signals response to program material



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

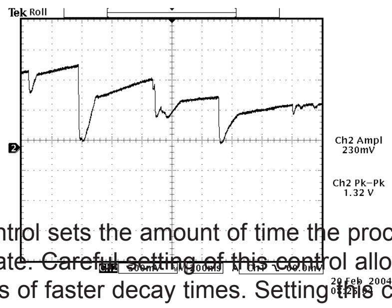


Peak time constants dominating to a lesser degree due to a high setting of average attack

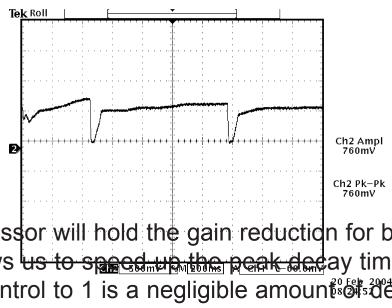


Peak time constants dominating to a much lesser degree due to a lower setting of average attack

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 DSPXtreme 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme.

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 DSPXtreme 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 DSPXtreme 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 DSPXtreme to provide peak control for the DR path of processing. This is not just a simple single band peak limiter. The DSPXtreme 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 DSPXtreme 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.

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

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 DSPXtreme 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 DSPXtreme , 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 DSPXtreme 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.

Stereo generator
The DSPXterme-FM digital stereo generator creates the multiplexed output with superb stereo separation, excellent pilot stability, low distortion and low crosstalk between main and sub-channel. MPX level is adjustable via front panel and has a wide range to produce proper modulation level with any exciter. Pilot injection level is adjustable as well, with the option to turn it off for mono operation. If you are already broadcasting mono audio, turning pilot off is a good idea as it will increase your signal to noise ratio by about 23 dB. There is also a switchable 19 kHz pilot reference output for RDS encoders.

Composite clipper
The composite clipper in the DSPXtreme'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.

If you are using high composite clipping drive (lots of composite clipping), an optional pilot protection notch filter will protect the pilot area from being contaminated by clipping harmonics. Be aware that the filter will remove some of the harmonics which might result in slight increase of overshoots, depending on how much composite clipping you are using. Therefore you might need to re-adjust the output level to keep modulation peaks below 75 kHz (100% modulation) when the filter is activated.

Auditioning individual stages of the DSPXtreme processing

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

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

Getting The Sound You Want

While the DSPXtreme 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.

Multi-band AGC:
Speed up the release times, making them faster.
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.

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.

Multi-band AGC:
Slow down the release times.
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: (DR USE)
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.

Multi-band AGC:
Speed up the band 1 release.
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: (DR USE)
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.

Multi-band AGC:
Speed up the band 4 AGC release time, making it faster.
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: (DR USE)
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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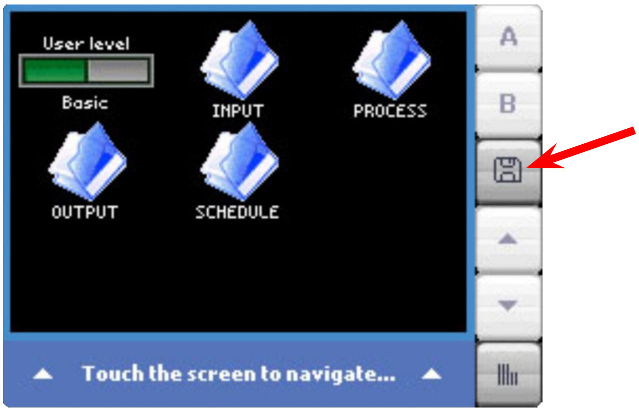
Managing Presets (front panel control)

The DSPXtreme has an assortment of factory presets and provision for 16 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 DSPXtreme are accessed from the main screen by pressing the disc icon on the left side of the touch-screen.



To enter preset menu, press the disc icon

Selecting a preset.
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 on the bottom of the screen. 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. Now pressing the 'B 'button will reload the saved preset allowing you to compare before and after your changes. The 'A' button will return you to the adjusted preset. The other option you have if is the 'LOAD' button, reloading the saved preset and discarding your changes. This way 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 select the position you want to save the preset and then press the 'SAVE' button on the bottom of the touch-screen. The new preset will be saved to the DSPXtreme's memor. You can then name the preset by pressing the 'RENAME' button and enetering the name of your preset on the on-screen keyborad.

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 DSPXtreme 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.

Beta 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 DSPXtreme 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 DSPXtreme matches the peak output of the DSPXtreme.

F2 80's
Intended for those ever-popular 80's tunes that were recorded back when record loudness was not the ultimate goal and records still had some dynamic range. Smooth multi-band equalization and tasteful final limiting for a consistent yet unfatuating sound.

F3 AC
A nice balance for Adult Contemporary formats where ultimate loudness is not everything, but quality sound is.

F4 CHR
This preset provides a good starting point for current and popular music formats.

F5 CHR HOT
Same as above, but for more competitive markets.

F6 LOUD!
If you need to be big and loud, this is the preset for you.

F7 EASY LISTENIN'
Smooth and light processing for easy listening...

F8 LOW BASS
Urban, dance and all other formats can benefit from the nice and warm low-end provided here.

F9 POP
Leaning a bit towards bright and agressive, this preset provides a good starting point for pop music.

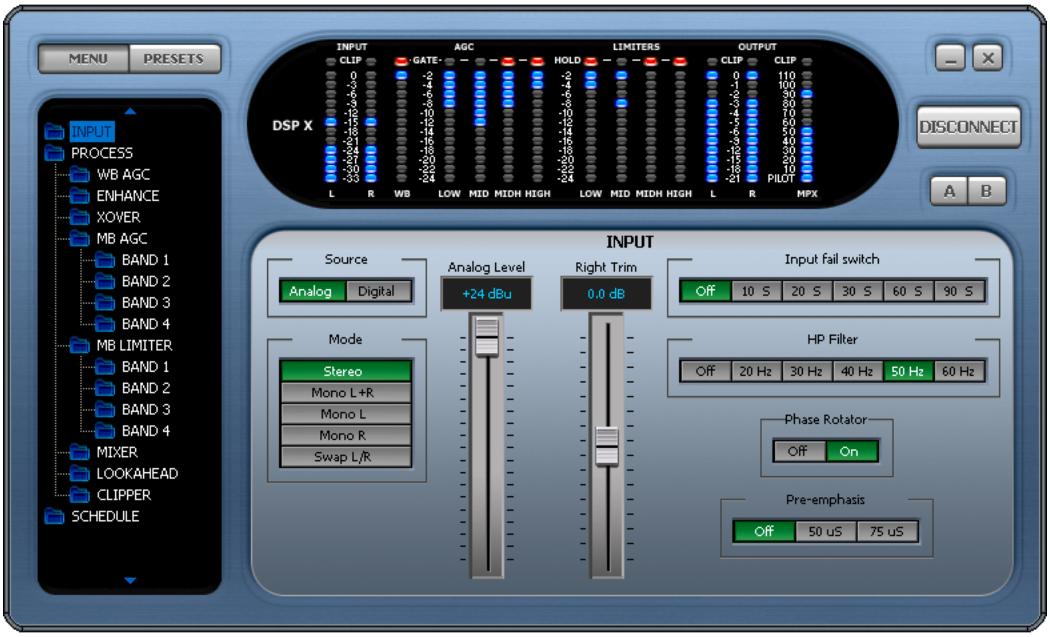
F10 R'N'R
Nice EQ of bass and HF for solid Rock'n'Roll sound!

Remote Control Of The DSPXtreme

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

The DSPXtreme can only to talk to the serial system (RS232) 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 DSPXtreme's front panel.

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

Serial port system
The serial port system consists of two serial ports, one on the front and the other on the rear panel. Only one of the ports can be used at any one time and the active port can be selected from the front panel LCD control system. The serial port selection parameter is located in the 'SYSTEM' menu.

Firmware (new versions of DSPXtreme code) will also need to be uploaded into the DSPXtreme via the serial port system. The BW Broadcast development team have built the serial communications XMODEM protocol. Any standard terminal program will allow you to send an update file into the DSPXtreme via XMODEM protocol.

CONTROL OF THE DSPXtreme BY RS232 (SERIAL)
If you wish to use the RS232 port to control the DSPXtreme follow the steps below
1. Connect the supplied serial cable to the rear RS232 port
2. Navigate to the 'REMOTE SOURCE' parameter (also in system) and select the serial option
3. Run the DSPXtreme 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
4. A password needs to be entered, regardless of password settings on the DSPXtreme itself. A password still needs to be entered even if the passwords on the DSPXtreme have been disabled.
5. Click connect on the application and you should receive a 'please wait' box while the information is retrieved from the DSPXtreme. Once connected you are then free to control the DSPXtreme with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtreme 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 DSPXtreme is 3779. You are free to change these on the DSPXtreme itself (see information on password control elsewhere in this manual)

Remote application connection screen

NET/LAN PORT

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

CONTROL OF THE DSPXtreme BY THE NET/LAN PORT

If you wish to use the NET/LAN port to control the DSPXtreme follow the steps below

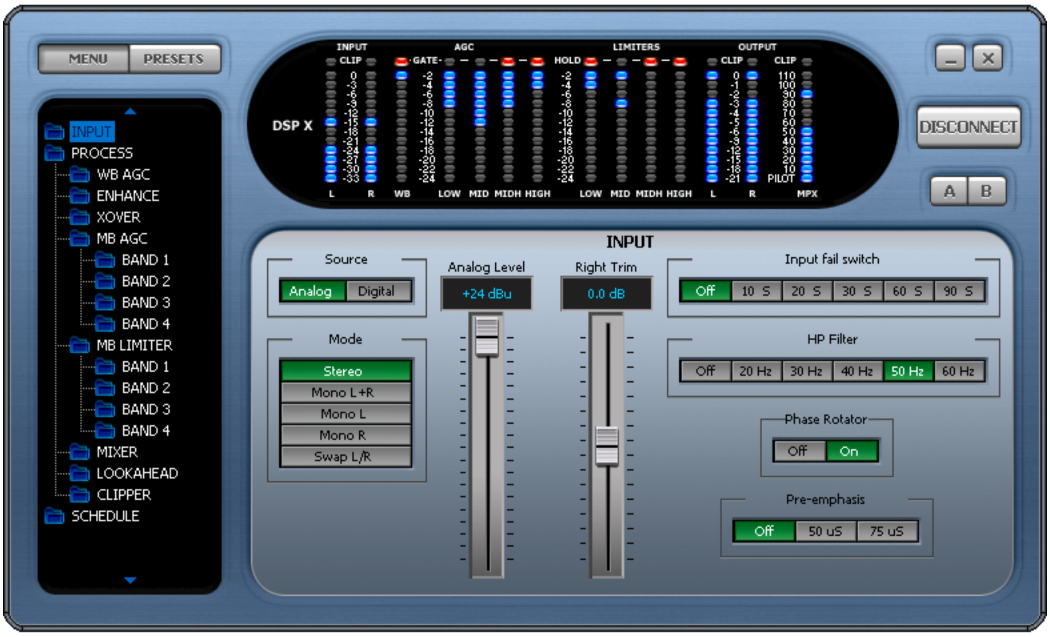
1. Connect a cat 5 cable to the RJ45 port on the DSPXtreme and plug this into your network hub/switch. A Xover cable can be used to connect directly to a PC if you don't have a switch or hub.
2. . Navigate to the 'REMOTE SOURCE' parameter contained in the system menu on the DSPXtreme and select the Network option.
3. Run the DSPXtreme 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 DSPXtreme itself. A password still needs to be entered even if the passwords on the DSPXtreme have been disabled.
5. Click connect on the application and you should receive a 'please wait' box while the information is retrieved from the DSPXtreme. Once connected you are then free to control the DSPXtreme with the remote application. Further information on the remote application is contained on the following pages. If the DSPXtreme 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 DSPXtreme is 3779. You are free to change these on the DSPXtreme itself (see information on password control elsewhere in this manual)

PASSWORD ACCESS

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

These passwords can only be set from the DSPXtreme 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.



Remote application connected

When the DSPXtreme 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 DSPXtreme 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 DSPXtreme 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 DSPXtreme 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.

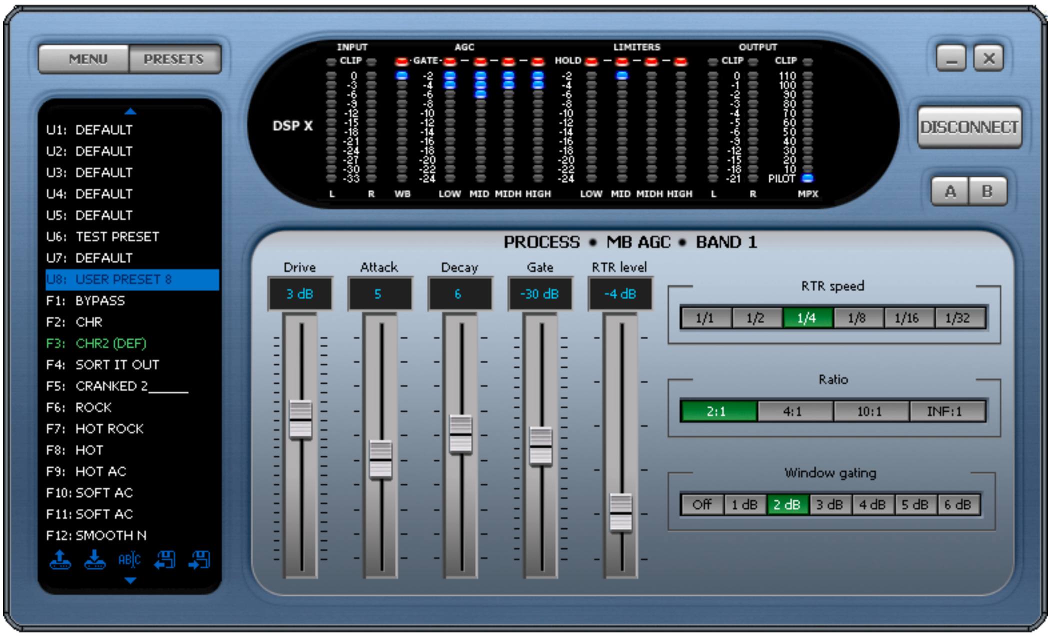
WORKING WITH PRESETS
The DSPXtreme remote application makes it easy to load, save and change presets. Click the preset button (top left) if not already depressed. You should then see the preset list in the left hand window (see image).

Understanding the preset list
The preset list contains all of the presets contained in the DSPXtreme. 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.



Preset window shown

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 DSPXtreme 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.

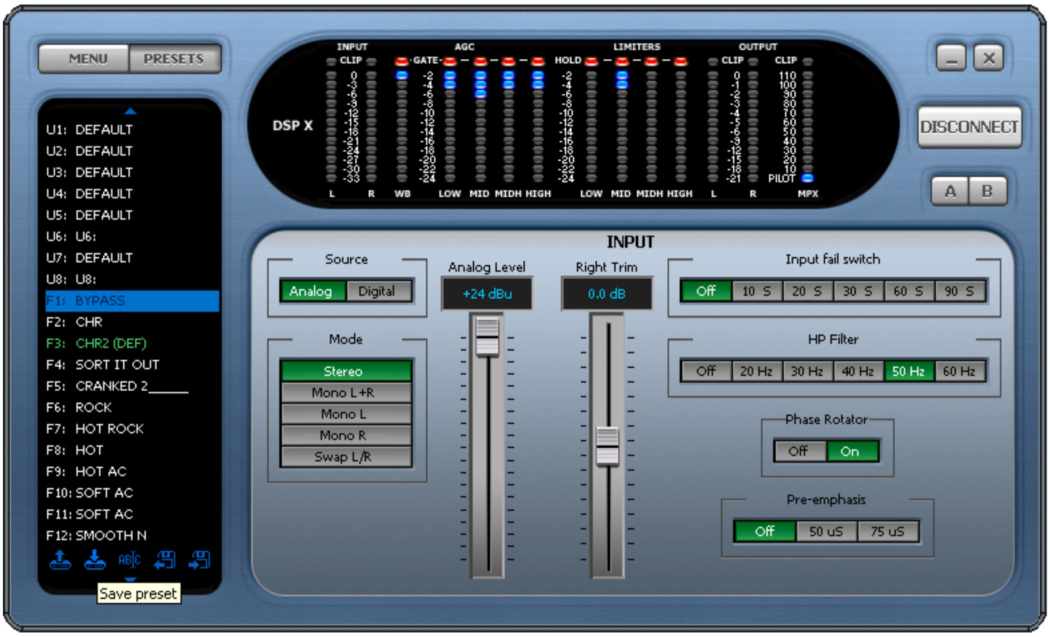
Load preset
Load preset will change the default preset to the one selected. This usually means that this preset will become 'on air'. The exception to this is when the default preset is being overridden by a daypart or remote trigger. In this case the DSPXtreme will 'on air' the selected preset when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

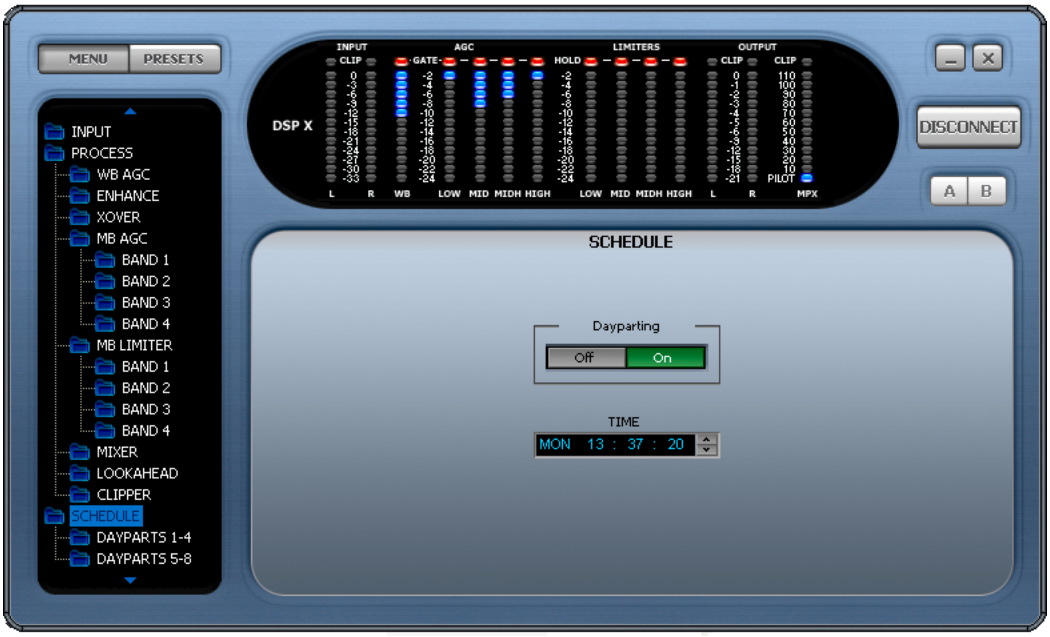
Loading presets from a PC
This option will pop up the standard windows load dialog box. You can browse to and select a preset file to be loaded into the DSPXtreme. The preset location that is loaded is the currently highlighted (in blue) preset, not the one that is currently 'on-air' You can only load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



Preset icons

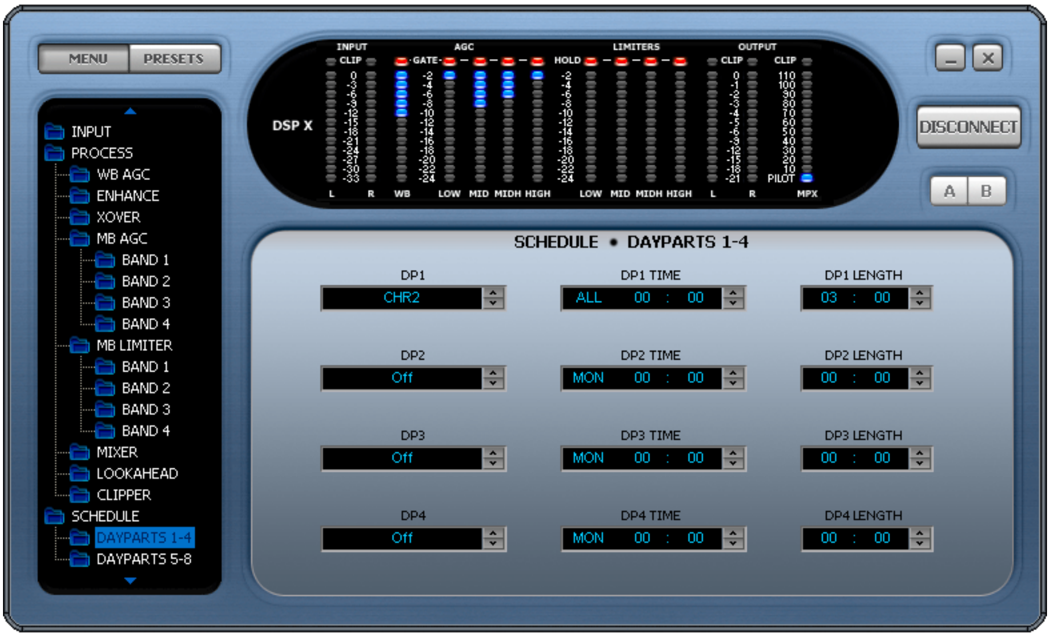
SCHEDULING WITH THE REMOTE APPLICATION
Using the menu tree to change the processing is quite simple and really doesn't need a lot of explanation. The scheduling screens that controls the dayparting may appear daunting so we are going to give you a quick guide to using it.

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 DSPXtreme.



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

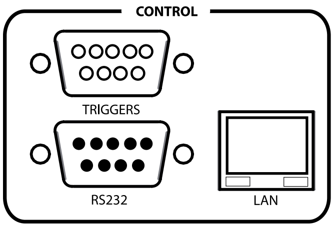
The Remote application has two buttons labelled A and B that are located just below the connection button. These buttons allow you to compare changes you have made to the processing against the saved preset. When you load a preset the buttons should be greyed out but as soon as you make any processing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

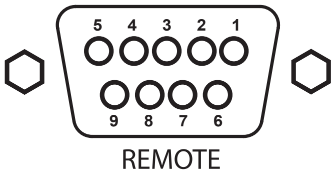
The A/B feature makes it easy to build up your own presets by being able to easily compare before and after processing adjustements. We hope you find it useful.

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.

Clock Based Control (Dayparting)

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

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

The schedule menu contains the following options:

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

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

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

Daypart ON/OFF enables or disables the dayparting.

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

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

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

Rather than setup the dayparts as

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

We could instead setup the dayparts as

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

which saves a daypart position.

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

SPECIFICATIONS

Specifications apply for measurements from analog left/right input to stereo composite output and to FM analog left/right output. Measurements apply to FM mode of operation.

Frequency Response (Bypass Mode): Follows standard 50µs or 75µs pre-emphasis curve ±0.10 dB, 2.0 Hz–15 kHz. Analog left/right output and digital output can be user configured for flat or pre-emphasised output.

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

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

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

Analog Audio Input

Configuration: Stereo.
Impedance: >10k, load impedance, electronically balanced.
Nominal Input Level: Software adjustable -6dBu to +18dBu (24dB range clip ceiling, adjustable in 1dB steps).
Maximum Input Level: +24 dBu
Connectors: XLR female Pin 1 chassis ground, Pins 2 (+) and 3 electronically balanced, floating and symmetrical.
A/D Conversion: 48KHz 24 bit 128x oversampled delta sigma converter with linear-phase anti-aliasing filter.
Filtering: RFI filtered.

Analog Audio Output

Configuration: Stereo, flat or pre-emphasised (50µs or 75µs), FM delayed or not delayed, AES2 or monitor output, software-selectable.
Source Impedance: 10 Ohm, electronically balanced and floating.
Load Impedance: 600 Ohm or greater, balanced or unbalanced.
Output Level (100% peak modulation): Software adjustable from –12 dBu to +24 dBu peak, into 600 Ohms or greater load.
Signal-to-Noise: >= 90 dB unweighted (Bypass mode, de-emphasised, 20 Hz–15 kHz bandwidth, referenced to 100% modulation).
L / R Crosstalk: <= –70 dB, 20 Hz–15 kHz.
Distortion: <= 0.01% THD (Bypass preset, de-emphasised) 20 Hz–15 kHz bandwidth.
Connectors: XLR male. Pin 1 chassis ground, Pins 2 (+) and electronically balanced, floating and symmetrical.
D/A Conversion: 48KHz 24 bit 128x oversampled.
Filtering: RFI filtered.

Digital Audio Input

Configuration: AES/EBU Stereo, 24 bit resolution, software selection of stereo, mono from left, mono from right or mono from sum.
Sampling Rate: 32, 44.1, 48, 88.2, or 96 kHz, automatically selected.
Connector: XLR female. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110 ohm impedance.
Filtering: RFI filtered.

Digital Audio Outputs (AES1 and AES2)

Configuration: Stereo per AES/EBU standard. Output configured in software as flat or pre-emphasised (50µs or 75µs), FM delayed or not delayed, AES2 (DR path) or monitor output, software selectable.
Sample Rate: Internal free running at 32, 44.1 or 48 KHz selected in software. Can also be synced to the AES/EBU digital input at 32, 44.1, 48, 88.1 or 96 kHz, as configured in software.
Connector: XLR-type. Pin 1 chassis ground, pins 2 and 3 transformer balanced and floating, 110 ohm impedance.
Output Level (100% peak modulation): –20.0 to 0.0 dBfs software controlled.

Filtering: RFI filtered.

MPX output 1

Source Impedance: 10 Ohm Single-ended, floating over chassis ground.
Load Impedance: 600 Ohm or greater.
Maximum Output Level: +12.0 dBu (peak) software-controlled output level control.
Minimum Output Level: 0 dBu (peak)Software-controlled output level control.
Pilot Level: Adjustable from 6.0% to 12.0% and OFF, software controlled.
Pilot Stability: 19 kHz, ±1 Hz (10 degrees to 40 degrees C).
D/A Conversion: 24-bit
Signal-to-Noise Ratio: <= –85 dB (Bypass mode, de-emphasised, 20 Hz – 15 kHz bandwidth.
Distortion: <= 0.02% THD (Bypass mode, de-emphasised, 20 Hz – 15 kHz bandwidth.
Stereo Separation: Typ. > 70 dB 30 Hz - 15 kHz.
Crosstalk-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel.
Crosstalk-Non-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel.
38 kHz Suppression: >= 70 dB.
76 kHz & Sideband Suppression: >= 80 dB.
Connectors: BNC, floating over chassis ground.
Filtering: RFI filtered.

MPX output 2

Source Impedance: 10 Ohm Single-ended, floating over chassis ground.
Load Impedance: 600 Ohm or greater.
Maximum Output Level: +12.0 dBu (peak) trimmer output level control on the back of the unit.
Minimum Output Level: 0 dBu (peak) trimmer output level control on the back of the unit.
Pilot Level: Adjustable from 6.0% to 12.0% and OFF, software controlled.
Pilot Stability: 19 kHz, ±1 Hz (10 degrees to 40 degrees C).
D/A Conversion: 24-bit.
Signal-to-Noise Ratio: <= –85 dB (Bypass mode, de-emphasised, 20 Hz – 15 kHz bandwidth.
Distortion: <= 0.02% THD (Bypass mode, de-emphasised, 20 Hz – 15 kHz bandwidth.
Stereo Separation: Typ. > 70 dB 30 Hz - 15 kHz.
Crosstalk-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel.
Crosstalk-Non-Linear: <= –80 dB, main channel to sub-channel or sub-channel to main channel.
38 kHz Suppression: >= 70 dB.
76 kHz & Sideband Suppression: >= 80 dB.
Connectors: BNC, floating over chassis ground.
Filtering: RFI filtered.

Subcarrier (SCA) Inputs

Configuration: Subcarrier input sums into composite baseband output.
Impedance: >10K
SCA input level Sensitivity: Sums into MPX output at 10% injection.
Connector: BNC, unbalanced and floating over chassis ground.

Pilot Output

19 kHz Pilot Reference: BNC, 5V sine unbalanced and floating over chassis ground. Software selectable

Remote Control

Serial Port: DB9 (rear panel) 19200 kbps.
Ethernet Port: 10 Mbit/sec on RJ45 female connector.
Remote Control (trigger port): DB9 opto-isolated and floating. Eight pull low inputs.

Other

Voltage: 100–240 VAC, 50–60 Hz, 35 VA.
Connector: IEC. Detachable 3-wire power cord supplied.
Grounding: Circuit ground is independent of chassis ground, and can be isolated or connected with a rear panel switch.
Dimensions (W x H x D): 44mm x 482mm x 200mm

DSPXtreme-FMHD v1 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								
MULTI-BAND AGC								
AGC								
WINDOW GATING		dB		dB		dB		dB
GATE		dB		dB		dB		dB
B1<2 COUPLE		%		%		%		%
B3>4 COUPLE		%		%		%		%
CHANNEL COUPLE		%		%		%		%
BAND 1								
ATTACK								
RELEASE								
BAND 2								
ATTACK								
RELEASE								
BAND 3								
ATTACK								
RELEASE								
BAND 4								
ATTACK								
RELEASE								
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		dB		dB		dB		dB
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
LP FILTER		kHz		kHz		kHz		kHz
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								

bw broadcast

www.bwbroadcast.com