Algorithm manual

English

DB4/DB8 MKII

tc electronic®

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INTRODUCTION

This Algorithm Manual contains information about the processing performed by DB4 and DB8. For information about set-up, general use, routing and presets, please consult the Operations Manual.

DB4 and DB8 MKII are capable of running multiple, independent processors simultaneously. One such processor is called an “Engine”. Engines may be routed to deal with independent audio streams, or combined, for instance, to condition one input stream to different outputs, so-called trickle-down processing. Use the Routing page to define how engines are routed, and to assign physical inputs and outputs.

For each Engine, you may recall a different “algorithm”. An algorithm is a specific processor, for instance upconversion or 5.1 loudness correction. Most of this manual describes in detail the different algorithms you may recall into an engine of DB4 and DB8.

Engine presets are compatible between DB4 MKII and DB8 MKII. MKII units also read Engine presets from original DB4 and DB8. Finally, presets based on the algorithm “DTX” are compatible with the stereo processor, DB2.

Channel Distribution In

Surround Algorithms

To best comply with the channel allocation used by most digital AES-format equipment, the Input/Output channels on surround algorithms are allocated as follows:

1 Left
2 Right
3 Center
4 LFE
5 Left Surround
6 Right Surround

These channel allocations comply with the following standards:

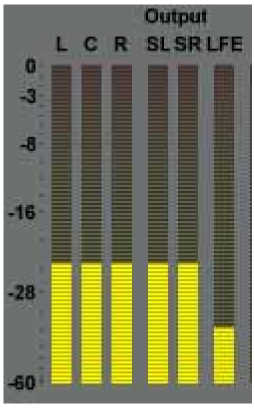
- ITU-Recommendation ITU-R BR.1384, Parameters for International Exchange of Multichannel Sound Recordings, 1998
- SMPTE 320M-1999, for Television - Channel Assignments and Levels on Multichannel Audio Media
- Surround Sound Forum Recommended Practice SSF- 02/1-E-2 (3-5-99), Multichannel Recording Format, Parameters for Programme Interchange and Archiving, Alignment of Reproduction Equipment

Grouping the Inputs/Outputs this way ensures optimal flexibility for further external processing and archiving, when working on setups following the above mentioned standards.
It is, however, worth noticing that total routing-flexibility of physical Inputs/Outputs to Engine Inputs/Outputs is available on DB8/DB4 via the Routing page.

Metering In the Engine Edit Pages

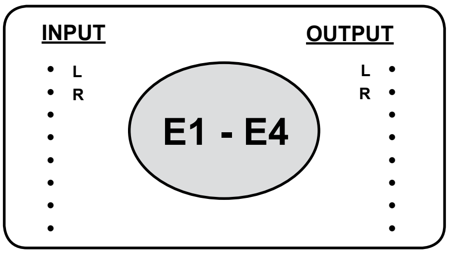
For logical channel metering in the various surround algorithms, the meters on the Engine Edit pages are displayed in the following order.

Left - Center - Right - Left Surr. - Right Surr. - LFE



We believe that by displaying the meters on the Engine Edit pages in the same order as your speakers are physically placed, the most intuitive metering of channel-levels is achieved.

Algorithm Inputs/Outputs are distributed as follows:



Introduction

ATX and DTX algorithms combine BS.1770 based loudness correction, 5-band processing, width control and true-peak limiting into comprehensive, low latency processors for stereo use. Should AV sync be needed, these algorithms also include 24 bit delay capable of being adjusted without noise being generated, while audio is passed. The DTX algorithm is ideal for digital stereo transmission, and for trickle-down processing to low loudness range platforms such as “pod” and mobile. Note: Presets from TC’s DB2 processor are based on the DTX algorithm and may also be directly transferred to DB4 or DB8.

The ATX/DTX algorithms can be operated in three distinctively different modes:

- Stereo. In this mode the Loudness, EQ and Multiband sections operate in tandem: Whatever gain change is applied to one channel, is applied to the other. Also, many parameters have mutual left and right controls.

- Dual Mono. In this mode the Loudness, EQ and Multiband sections treat the two Input signals completely independently.

- Stereo Wide. In this mode the apparent width and image of stereo signal can be altered simultaneously with controlling loudness and peak level. The left and right signal is internally de-composed into an M(Mono) and S(Stereo) component, and reverted to left and right signals before peak limiting on the Output.

DTX/ATX

Two different loudness control algorithms for stereo signals are available: DTX, which is targeted to digital broadcast and distribution, and ATX for analog broadcast or distribution. The ATX is high res, low latency loudness control algorithm with adaptive emphasis limiting for feeding analog transmission. The variations between ATX and DTX is found only on the Limiter page. All other pages are therefore described as the same in the manual section.

Reference Level

Reference Level defines the standard operating level, and scales the Threshold and Target Level parameters of the Loudness control and Multiband section. The Threshold of the Limiter is not influenced by this setting, but is always relative to 0dBFS.

Typical Reference Level settings would be -20 dBFS in USA and some parts of Asia, and -18 dBFS in Europe, Japan and some parts of Asia. With new loudness-based standards being adopted worldwide, Reference Level should be set to the Target Level of a given station, or 1 dB higher. This would typically be in the -24 to -21 dBFS range.

If you wish to relate all levels to 0 dBFS, leave the Reference Level setting at 0 dBFS.

Main page

In Gain

Range: 0dB to Off

Separate level controls for Left and Right Input (A and B).

Phase Inv

Range: Normal/Inverted

Press to phase invert channels A, B or both.

Delay

Range: 0-4000ms

Delay alignment of the Input channels. Depending on selected Configuration type, either one common Delay setting or individual delay settings are available.

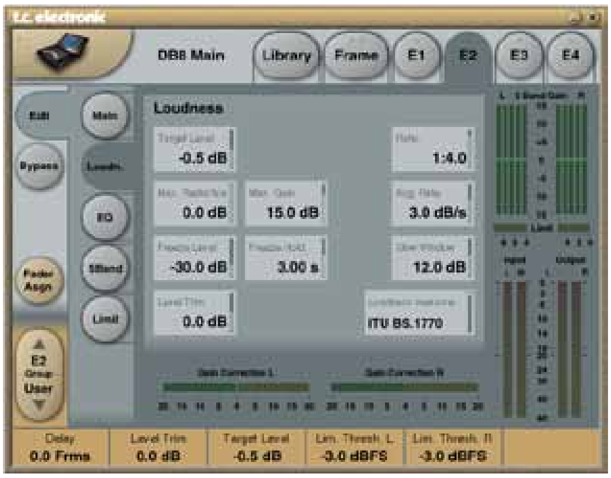
Delay Unit
Range: ms, 24fps, 25fps, 30fps
With this parameter it is possible to select which unit the Delay parameter should be shown in. Changing this parameter does not affect the actual delay value.

Lo Cut
Range: Off to 200Hz
Second order LoCut filter on both Inputs.

Hi Cut
Range: Off to 3 kHz
8th order HiCut filter on both Inputs.

Look ahead Dly
Range: 0-15ms
If the 5 band Compression sections is set to use a very short Attack times (up to approx 10-15ms) overshoots may occur. The Look Ahead function allows the DB8/DB4 to evaluate the material just before processing and artifacts can thereby be prevented.
Be aware that the Look Ahead delay function actually delays the output signal.

Loudness page



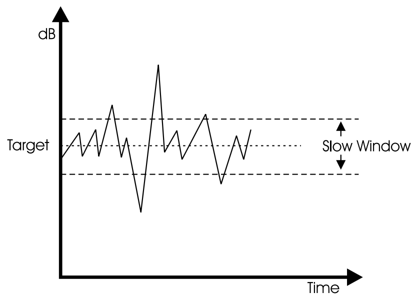
Target Level
Range: +10dB to -10dB
This is the level the Loudness controller will aim at on its output. Target Level is relative to Reference Level on the Main Page.

Max Reduction
Range: -20dB to 0dB
This is the maximum attenuation the Loudness Control is allowed to perform. If set to 0.0dB, the Loudness Control cannot attenuate the signal at all.

Max Gain
Range: 0 to +20dB
This is the maximum gain the Loudness Control is allowed to perform. If set to 0.0 dB, the Loudness Control cannot add gain to the signal at all.

Freeze Level
Range: -10dB to-40dB
Sets the minimum level required before the Loudness Control will start adding more gain. It would typically be set to avoid boosting signals considered noise. The Freeze Level parameter is relative to the Reference Level setting on the Main page.

Freeze Hold
Range: 0 to 5 seconds
When the Input signal drops below the Lo Level, the Gain Correction of the Loudness Section is frozen for the duration of the Hold time. When the Hold period expires, the Gain Correction falls back to 0dB gain.



Level Trim
Range: -18dB to + 18dB
When using the Multiband algorithm, DB8/DB4 operates with 48 bit precision on all audio internally and it is possible to correct loudness manually without the risk of overloads. The Level Trim can be used for permanent offsets or live loudness adjustments.

Ratio
Range: 1:1.25 to 1:6
Ratio is the steering factor used when the Loudness Control applies boost or attenuation to reach the Target Level. The higher ratio, the more rigid steering towards the Target Level.

Average Rate (Avg Rate)
Time constants in the Loudness Control are changed dynamically with the Input signal based on computations by multi-level detectors. When the Output level is close to the Target Level, gain changes are relatively slow.

The Average Rate offsets all time constants to be faster or slower. Values below 1dB/Sec produces a gain change gating effect when the Output level is already in the target zone, while values above 4dB/Sec will add density to sound.

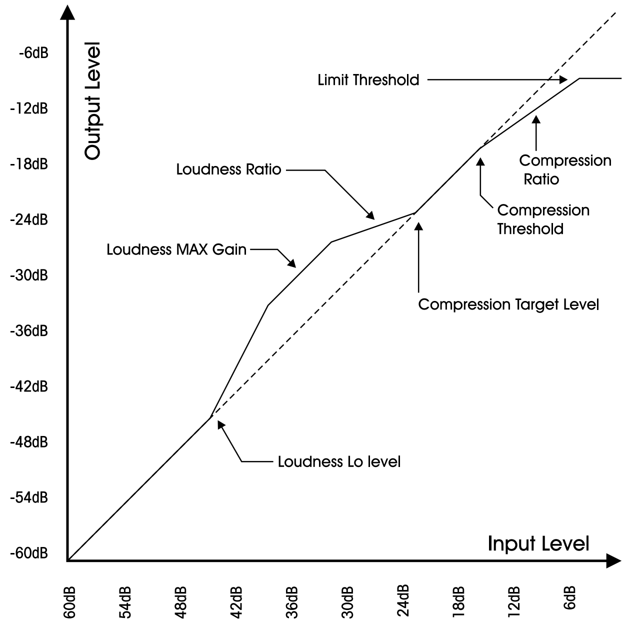
Slow Window

Range: 0 to 20 dB
The slow window is the area around the set Target Level. Within the slow window the Loudness is only gently controlled. When the signal exceeds the limits of the Slow Window the Loudness is treated more radically. Depending on the set Average Rate and Ratio.

Loudness Meassure

Select between TC GRID or standard ITU BS.1770.

Multiband parameter illustration



EQ page

Introduction

This digital EQ features a four-band parametric EQ with high- and low-pass filters switchable between Notch, Parametric, Shelving and Cut filters. The needle sharp notch filter has a range down to 0.01 octave and the shelving filters has a variable slope, ranging from gentle 3 dB/oct over 6 and 9 to 12dB/oct. Cut filters are switchable between 12dB/oct maximum flat amplitude (Butterworth) or flat group delay (Bessel) types. The parametric equalizer features a natural and well defined bandwidth behavior at all gain and width settings:

Basic operation

- Select Freq, Gain, Type or Lo/Hi to access all four parameters on individual bands.
- Press **Bypass EQ** to bypass all four bands.

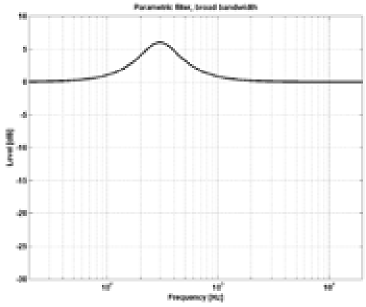
Type Selector

- Press **Type** and use faders 1-3 to select filter types.

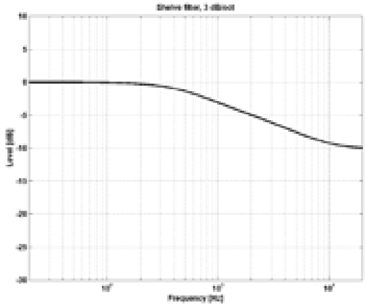
For Lo and Hi filters select between filter types: Parametric, Notch, Shelve and Cut.

For the Mid filter select between filter types:
Parametric and Notch.

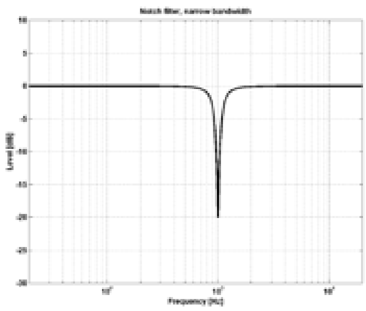
Parametric Filter - Broad type



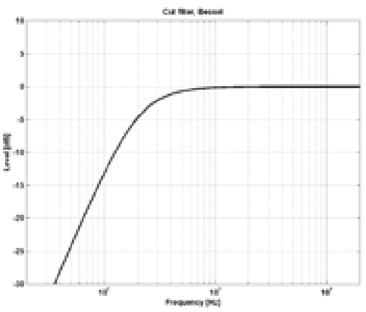
Shelving Filter



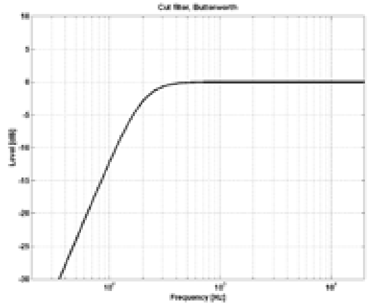
Notch Filter - Narrow Type



Cut Filter - Bessel type



Cut Filter - Butterworth type



Freq

Press **Freq** and use Faders 1 to 3 to adjust frequence for each of the four bands.

Range - Lo band	: 20Hz to 20kHz
Range - Mid band	: 20Hz to 20kHz
Range - Hi band	: 20Hz to 40kHz

Gain

Press **Gain** and use Faders 1 - 3 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:

Lo Gain	: -12dB to +12dB
Mid Gain	: -12dB to +12dB
Hi Gain	: -12dB to +12dB

Range for the Notch filter:

Lo Gain	: -100dB to 0dB
Mid Gain	: -100dB to 0dB
Hi Gain	: -100dB to 0dB

Type

Press and use Faders 1-3 to set BW value for each of the 4 EQ bands.

Range for the Notch filter:

Lo BW	: 0.02oct to 1oct
Mid BW	: 0.02oct to 1oct
Hi BW	: 0.02oct to 1oct

Range for the Parametric filter:

Lo BW	: 0.1oct to 4oct
Mid BW	: 0.1oct to 4oct
Hi BW	: 0.1oct to 4oct

Range for the Shelve filter:

Lo BW	: 3dB/oct to 12dB/oct
Hi BW	: 3dB/oct to 12dB/oct

Range for the Cut filter:

Lo BW	: Bessel or Butterworth
Hi BW	: Bessel or Butterworth

Bandwidth/Q - Key-Values:

BW		**Q**
0.5	-	2.87
0.7	-	2.04
1.0	-	1.41

5 Band Page

Xovers

Press this button to access the four cross-over points between the five-bands. The parameters are Automatically assigned to faders 1-4.
Parameter range:
Xover 1: Off to 1,6kHz
Xover 2: Off to 4kHz
Xover 3: 100Hz to Of,
Xover 4: 250Hz to Off

Defeat Thresh

Range: -3 to -30dB
This is a unique control which holds the gain from the multiband compressor below a certain threshold. No matter the spectral shaping applied from multiband system, below the Defeat Threshold, the frequency response is flat and gain is unity.
Defeat Threshold is relative to Compressor Threshold, which is relative to Reference Level.

Defeat Ratio

Range: Off to Infinity
Controls how close to the Defeat Threshold the make-up gain of the compressor is counteracted. At high ratios, the signal only has to be slightly below the Defeat Threshold before the compressor gain is fully defeated.

Thresholds A & B

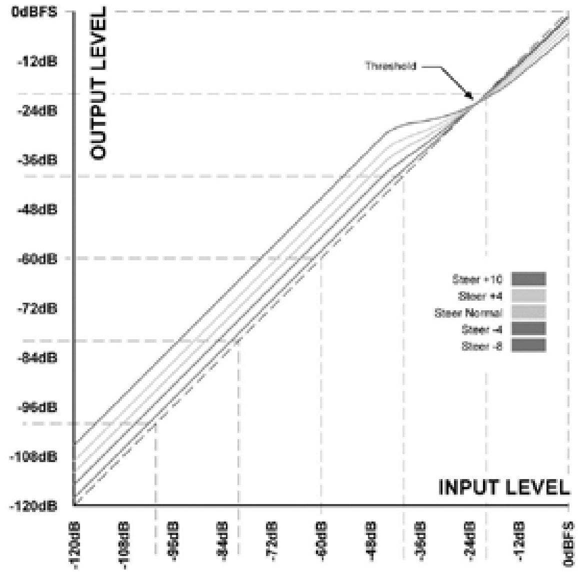
Parameter range: -25 to 20dB
Thresholds and the overall All Threshold. Press this button to access the five individual band Threshold is relative to Reference Level set at the Main page.

Gain

Parameter range: 0 to 18dB
Press this button to access the five individual band Gains and the overall All Gain.

DXP Mode - introduction

The 5-band section is either in normal compression mode, or DXP mode. Instead of attenuating signals above a certain threshold, DXP mode (Detail Expansion) lifts up signals below the Threshold; thereby bringing out details rather than squashing the loud parts. DXP mode therefore is capable of adding intelligibility and air to speech, lifting harmonics, or emphasizing ambience without increasing overall peak level.



As shown on the illustration, gain is positive below threshold, unity at Threshold, and the effect decreases above Threshold. In DXP mode, Ratio becomes Steer. Steer can be regarded as an adaptive Ratio that gradually approaches 1:1 above the threshold.

<u>Multiband DXP</u>
DXP mode can be used with any number of bands up to 5. When used multiband it is particularly effective in bringing out air and clarity.
The processor can act as an automatic Eq that removes a boost when it’s not needed: At very low levels, where noise is dominant, and at loud levels where sibilance would become a problem. Besides from being effective on speech, DXP mode can be used in mastering to bring up low levels, e.g. when preparing film or concerts for domestic or noisy environment listening.

Try setting the Steer and/or Threshold parameters differently in the bands to hear the effect. High Steer values add more detail gain than low values, but remember that Threshold has to be negative to add detail gain at all. DXP Threshold relates to the Reference Level set on the Main page.

NOTE: To disable DXP detail gain at very low levels, use the Defeat Threshold and Defeat Ratio controls. Defeat threshold relates to the DXP threshold, and allows for a certain level-window, inside which detail gain is applied. Defeat Ratio determines the slope at which DXP detail gain is defeated.

Ratio - DXP mode OFF

Parameter range: Off to Infinity:1
Press this button to access the five individual band Ratios and the overall All Ratio.
The parameters are automatically assigned to fader 1-6.

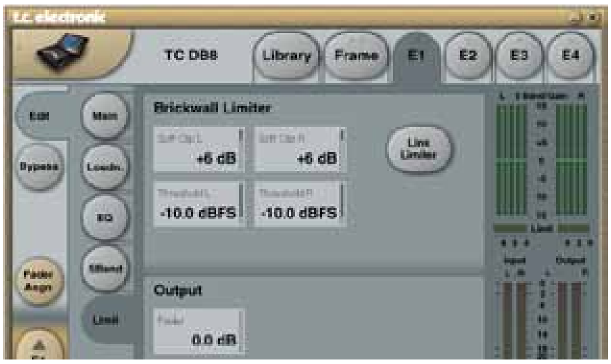
Attack

Parameter range: 0.3 to 250ms
Press this button to access the five individual band Attacks and the overall All Attack.
The parameters are automatically assigned to fader 1-6.

Release

Parameter range: 20ms to 7s
Press this button to access the five individual band Release and the overall All Release.
The parameters are automatically assigned to fader 1-6.

DTX Limit page



Link Limiter

When Link is active, the same amount of peak limiting is always applied to both channels.

Some broadcasters like the sound of operating left and right limiting without stereo coupling because they feel that it maximizes loudness and widens the stereo image. On dual mono sources, of course you should always choose un-linked Limiter operation.

NOTE The Configuration control on the Main page does not affect the Link Limiter setting. This link is running individually from the selected configuration.

Softclip A/L and B/R

Parameter range: - 3dB to Off
When active, Soft Clip applies a saturation effect on signals close to maximum Output level. The threshold is relative to the Threshold of the Brickwall Limiter.
This controlled distortion of transients works well for adding loudness, but is not a desirable effect with some data compression codecs. While the Brickwall Limiter is extremely low distortion, Soft Clip is not. Use your own judgement if you want it or not.

Threshold A/L & Threshold B/R

Parameter range: -12 to 0.0dBFS
Sets the Threshold of the Brickwall Limiter.
The Threshold is relative to 0 dBFS, not to the Reference Level set on the Main page.

The output limiter detects and protects against true-peak signals as defined in ITU-R BS.1770 and in EBU R128. This precision limiter is based on 48 bit processing and utilizes adaptive time constant for low distortion operation.

Fader A & Fader B

Parameter range: Off to 0dB
Fader function on the Output. When Dual Mono configuration is selected, individual Output faders are available.

ATX Limit page

Parameters that are not described under DTX Limit page:

Emphasis

Range: Off, 50µs, 75µs, J17
To pre-condition signal better for analog transmission, the limiter in ATX can take downstream emphasis into account. Note that the output signal of DB4 or DB8 does not contain pre-emphasis, but is linear, so STL data reduction isn't compromised. When the Emphasis parameter is set to Off, linear limiting (like in DTX) is available.

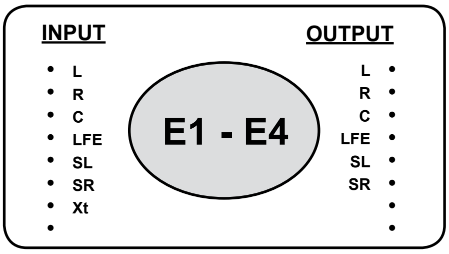
HF Offset

Range: -12dB to 0dB
When set to 0 dB, emphasis limiting precisely follows the selected pre-emphasis curve. However, lack of peak conservation in the downstream signalpath (DA converters, sample rate converters, filters, data reduction etc.) may necessitate a more conservative HF Offset, targeting, for instance, 1 or 2 dB below the theoretical roll-off. When the Emphasis parameter is set to Off, HF Offset has no effect.

Output

Off, -100dB to 0dB
Output level control.

Algorithm Inputs/Outputs are distributed as follows:



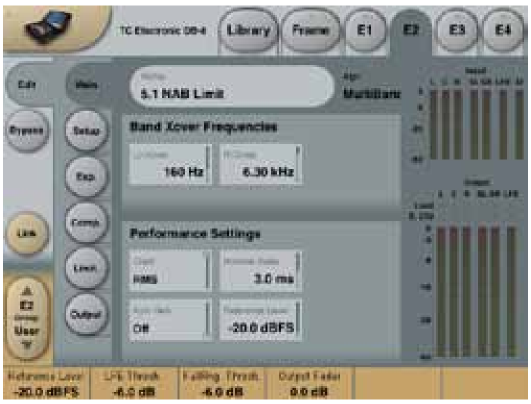
Introduction

The **Multiband-5.1** algorithm is a multi-channel, multi-band optimizer, with Limiters and extensive possibilities to assign channels to multiple Sidechains.
Four-band dynamics are available for 5.1-processing.
With the Multiband-5.1 it is possible to integrate dynamics processing for 5.1 applications offering features, which are not possible if using multiple stereo dynamic processors.

Multiband-5.1 processor contains:

- 5 channels of three band expansion and compression
- Full-range brickwall limiter on all Outputs
- 1 channel of full range expansion, compression and limiting for the LFE (Sub) channel
- 3 Sidechains for the five main channels, that can be assigned in flexible ways
- 1 extra Input channel that can be used for external Side Chain Input.

Main



At the Main page you have access to the general set-up parameters for the Expander and Compressor sections. Meters are shown for all seven Inputs and six Outputs at the right of the display.

Band Xover Frequencies

Lo Xover

Range: Off to16kHz
Sets the Cross-over frequency between the Lo- and the Mid- Expander and Compressor bands for the five main channels (LFr, RFr, Cnt, LSr, RSr).

NOTE: The two Cross-over points are not allowed to cross each other. Therefore the parameter range can be less than 16kHz if the Hi Xover parameter is set below 16kHz.

Hi Xover

Range: Off to16kHz
Sets the Cross-over frequency between the Mid- and the Hi- Expander and Compressor bands for the five main channels (LFr, RFr, Cnt, LSr, RSr).

NOTE: The two cross-over points are not allowed to cross each other. Therefore the parameter range can be less than going down to Off, if the Lo Xover parameter is set above the Off position.

Performance Settings

Crest

Range:
Peak, 6dB, 10dB, 12dB, 14dB, 16dB, 20dB, 24dB, RMS
Select compression method between RMS and PEAK.
The dB steps between RMS and Peak are the dBs needed for a peak-value to override RMS measurement.

Nominal Delay

Range: 0 to 15ms
(<2ms in 0.1ms steps. >2ms in 0.5ms steps)
Sets the nominal Delay of the signal compared to the Sidechain signal. This is also known as "Look ahead Delay", enabling the Compressor section to become more responsive to the incoming signal.

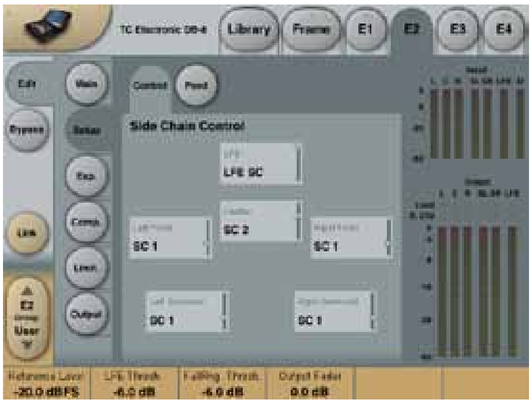
Automatic Make Up Gain

Range: Off/On
Switches the Automatic Make-up gain On or Off. As using compression is a reduction of dynamic range in the signal a compensation for this loss of gain on the Output side is possible. Use the Auto Make Up gain to achieve this.

Reference Level

Range: -24dBFS to 0dBFS in 0.5dB steps
This parameter sets the reference level in the algorithm. The reference level is the level at which the Threshold parameters will start operating when set to 0dB. E.g. if the Reference Level is set to -18dBFS (often referred to as 0dBu), a Threshold setting at -4dB, will cause the Compressor to start operating at -22dBFS.

Side Chain - Control page



The Sidechain assignment possibilities in the Multiband-5.1 are very comprehensive. Carefully selecting which channels should be controlled by which Sidechains, is just as essential as dialing in the correct Threshold and Ratio values.

It is possible to freely select any or none of three Sidechains to control each of the main-channels. This also gives you the option of grouping the channels. In addition to this, the LFE channel has its own Sidechain control. This enables e.g. setting up two Multiband-5.1 algorithms in serial setup, while having six individual Sidechains available, enabling fully individual Sidechain controls of all channels.

At the Feed page it is possible to make additional Sidechain link Inputs, for e.g. having the Center-channel contributing to the Sidechain Inputs of the two Front channels, to create a more coherent sound from the front-channels.

The illustration above reflects the Processing parameter set to Multiband-5.1 in Normal mode.

Basic operation
At the Setup/Control page it is possible to decide which Sidechains should control which channels. Select any of three Sidechains to be assigned to any of the five Main-channels. You can also chose to pass the channels unprocessed through the algorithm. The LFE channel can be assigned to its own separate Sidechain, or left unprocessed.

NOTE Setting a channel to unprocessed will preserve the processing delay through the algorithm, keeping the channel time-aligned to the other (processed) channels.

Side Chain Control

<u>Range - for the five main channels:</u>
- Unprocessed
- Side Chain 1
- Side Chain 2
- Side Chain 3

<u>Range - for the LFE channel:</u>
- Unprocessed
- LFE

Side Chain - Feed page

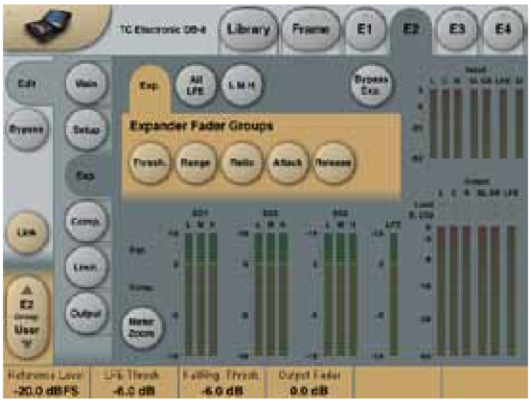
The Setup/SC Feed page holds parameters specifying which Input channels should feed the three Sidechains.

Normal
Range: Off, On
When this parameter is set to “On” the Input channels selected to be controlled by the respective sidechain will also Input to the sidechain.

Add 1, Add 2 and Add 3
Range: Off, LFr Max, RFr Max, Cnt Max, LSr Max, RSr Max, Xt Max, LFr Sum, RFr Sum, Cnt Sum, LSr Sum, RSr Sum, Xt Sum.
These parameters enable extra channels to be assigned to the respective Sidechain Input. The extra Sidechain Input channels will not be processed by the sidechain.
The Sum settings will add the Input to the sidechain, whereas the Max settings only will contribute to the sidechain if the level exceeds the other Input channel levels.

Expander

Exp. page



Pressing Threshold, Range, Ratio, Attack and Release keys will immediately assign Lo, Mid, Hi, All and LFE values for these parameters to Faders 1-4.
Be aware that the range of the All parameter is relative to the settings of the same parameters in the Compressor section.

Threshold
Range: -50dB to 0dB (in 0.5dB steps)
When the signal drops below the set Threshold point the Expander starts to generate downward expansion.

Range

Range: -40dB to 0dB in 0.5dB steps
Sets the maximum range of the expansion.

Ratio

Range: Off to Infinity
Sets the Expansion Ratio below the Threshold point.

Release

Range: 20ms to 7sec.
Sets the time it takes for the Expander to release its attenuation of the signal when the signal exceeds the Threshold again.

Attack

Range: 0.3 to 100ms
Sets the time it takes for the Expander to reach the attenuation specified by the Ratio parameter when the signal drops below the Threshold point.

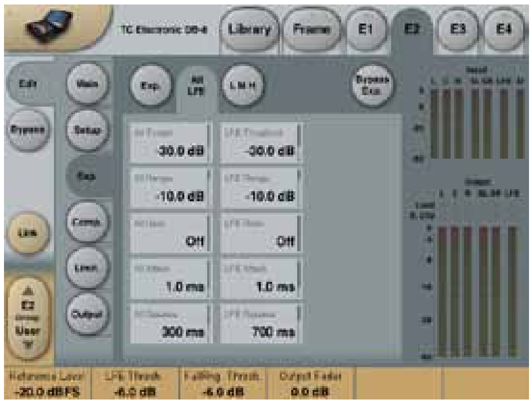
Meter Zoom

Press **Zoom** to decrease meter range and have a more accurate metering.

Bypass Exp.

Press to bypass the Expander section of the MD 5.1 algorithm.

All LFE page



Pressing any parameter will assign this to Fader 6.

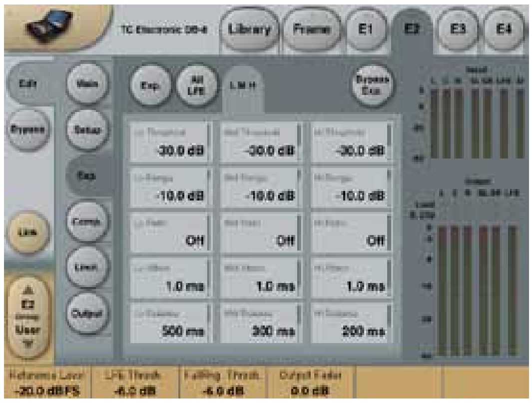
All - parameters

These parameters are equivalent to the “All” - Threshold, Range, Ratio, Attack and Release parameters.

LFE - parameters

These parameters are equivalent to the “LFE” - Threshold, Range, Ratio, Attack and Release parameters.

All L M H page

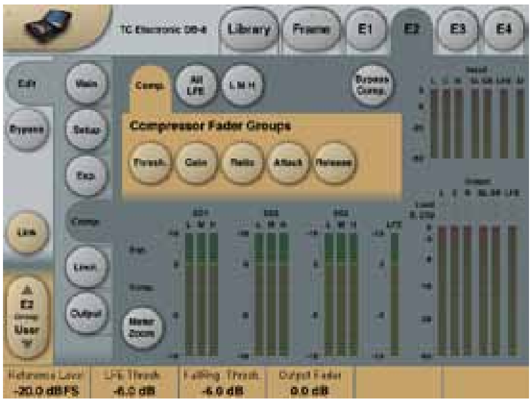


Pressing any parameter will assign this to Fader 6.

This page holds all Expander Threshold, Range, Ratio, Attack and Release parameters for the Lo, Mid and Hi bands.

Compressor

Comp. page



Pressing Threshold, Range, Ratio, Attack and Release keys will immediately assign Lo, Mid, Hi, All and LFE values for these parameters to Faders 1-4. Be aware that the range of the All parameter is relative to the settings of the same parameters in the Expander section.

Threshold

Range: -25dB to 20dB (in 0.5dB steps)
Sets the Threshold level at which the Compressor starts to operate. The Threshold parameter relates to the Reference Level setting.
Example: If the Reference Level is set to -18dBFS, a Threshold setting of -4dB, will cause the compressor to start operating at -22dBFS.

Gain

Range: Off, -18dB to 12dB in 0.5dB steps.
Adjusts the gain after the Compressor.

NOTE If the Auto Make-up gain parameter is set to On in the Main page, these gains will already have been adjusted according to the Threshold and Ratio parameters.

Ratio

Range: Off to Infinity
Sets the Compression Ratio that must be performed above the Threshold point.

Attack parameters

Range: 0.3 to 100ms
Sets the time the Compressor takes to reach the attenuation specified by the Ratio parameter when the level exceeds the Threshold point.

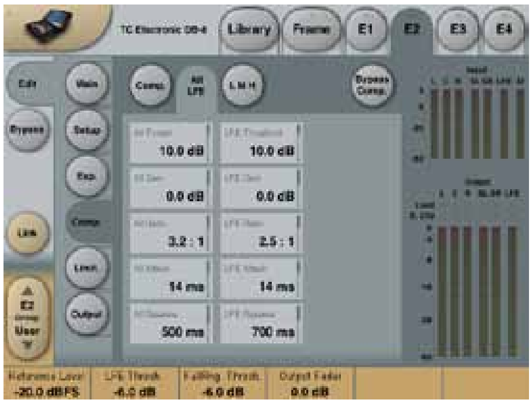
Release parameters

Range: 20ms to 7sec.
Sets the time the Compressor takes to release the attenuation of the signal when the signal level drops below the Threshold point.

Meter Zoom

Press **Meter Zoom** to decrease meter range and have a more accurate metering.

All LFE



Pressing any parameter will assign this to Fader 6.

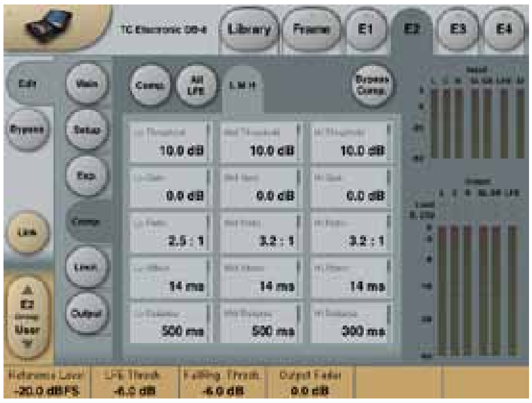
All - parameters

These parameters are equivalent to the “All” - Threshold, Range, Ratio, Attack and Release parameters.

LFE - parameters

These parameters are equivalent to the “LFE” - Threshold, Range, Ratio, Attack and Release parameters.

All L M H page



Pressing any parameter will assign this to Fader 6.

This page holds all Compressor Threshold, Range, Ratio, Attack and Release parameters for the Lo, Mid and Hi bands.

Limiter

The Limiter page is divided into three Sub-pages. One covering the Softclip section, one for the Full Range Limiter and one for the LFE Limiter.

Generic parameters in this algorithm:

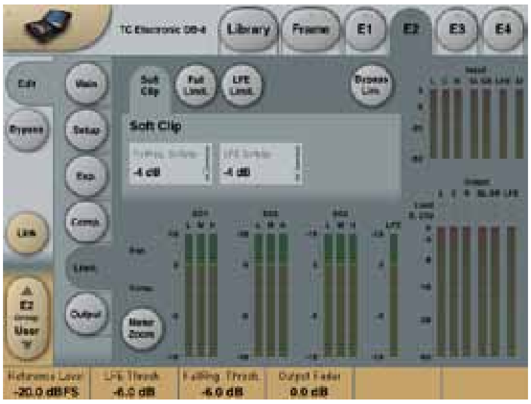
Meter Zoom

Press **Meter Zoom** to decrease meter range and have a more accurate metering.

Bypass Limiter

Press to Bypass the Limiter section of the 5.1 algorithm.

Soft Clip page



Softclip

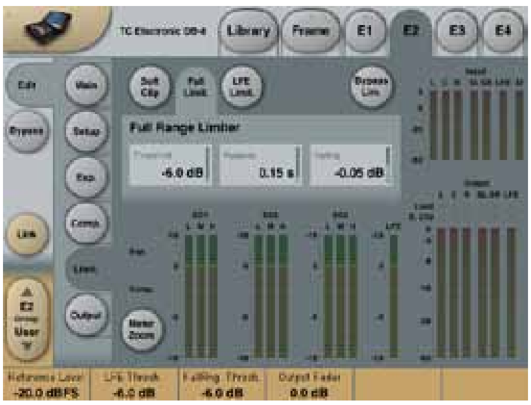
Full Range Softclip

Range: -6dB to Off
Softclipper Threshold setting after the Compressor for the five multiband channels. Threshold is always relative to 0dBFS (Not the Reference Level.

LFE Softclip

Range: -6dB to Off
Softclipper Threshold setting for the LFE channel only.

Full Limit. page



Threshold

Range: -12dB to Off
-6 to OdB in 0.1dB increments
-12 to -6 in 0.5dB increments
Brickwall limiter for the five multiband channels. Threshold is always relative to 0dBFS. LED on each Output meter indicates when Limiter is active.

Release

Range: 0.01 to 1.00 seconds
Release time for the Limiter.

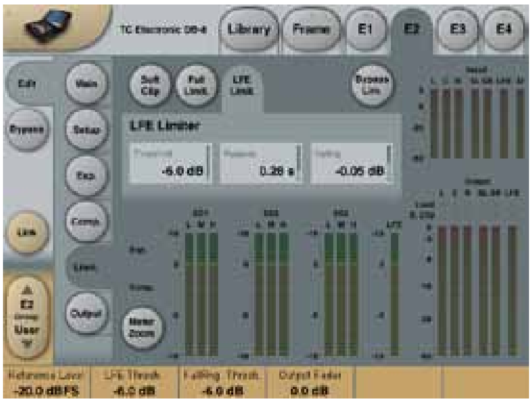
Ceiling

Range: -0.10dB to 0dB
Fine-tuning parameter setting the Ceiling for the Limiter.

WARNING

The Ceiling parameter prevents the Output signal from exceeding the adjusted Limiter Threshold. It can be used to “hide” overloads to downstream equipment, but it does not remove the distortion associated with an overload.

LFE Limiter page



LFE Limiter

Threshold

Range: -12 to +3dB
-6 to + 3 in 0.1dB increments
-12 to -6 in 0.5dB increments
Brickwall limiter for the LFE channel. Threshold is always relative to 0dBFS. LED on each Output meter indicates when limiter is active.

Release

Range: 0.01 to 1.00 seconds
Release time for the Limiter.

Ceiling

Range: 0 to -0.10dB in 0.01dB steps.
Fine-tuning parameter setting the Ceiling for the Limiter.

WARNING

The Ceiling parameter prevents the Output signal from exceeding the adjusted Limiter Threshold. It can be used to “hide” overloads to downstream equipment, but it does not remove the distortion associated with an over.

Output

Trim Levels

Output trims

Range: 0dB to -12dB in 0.1dB steps
Level trim of the Output channels. Only the fader is placed after these trims. These parameters can be used to trim the levels of the monitoring system, but please note that it also affects the recorded material.

Mute
Allows muting of each Output-channel.

Output Fader

Range: Off to 0dB
(<-40dB: in 3dB steps, >-40 in 0.5dB steps)
Output fader for all 6 Outputs. Can be controlled with the optional TC Master Fader connected to the GPI Input.

Compare

Easy switchable On/Off compare function for the entire MD 5.1 algorithm. This is not a bypass function as you are able to set a Compare Level (see below).

Compare Level

Range: -20 to 0dB
This function allows you to set a Compare level of the processed signal to match the unprocessed signal for better A/B listening.

Introduction

MDX5.1 is a high resolution dynamic range processor for multichannel signals. It may also be used to process for mono or stereo, thereby making changes or adjustments unnecessary.
Its combination of low level lift, multi-band structure, output limiting and extensive controls offers the most sophisticated dynamic range translation capabilities in the professional audio industry today. Not surprisingly, MDX5.1 has become the standard for dynamic range control in film and music mastering.

Dynamic Range Tolerance, DRT, at the consumer

The Dynamic Range Tolerance map, Fig 1, illustrates the dynamic range targets for various listening environments. It is therefore a practical tool for optimizing listener pleasure in digital broadcast.
According to recent studies, listeners typically object against too wide dynamic range much more than when the range is too restricted. Lack of speech intelligibility is the second worst offender, and often the cause for requesting more dynamic range limitation. Against the hopes of audio aficionados, as more people are listening through headphones (iPods and other personal entertainment systems), the DRT trend is therefore currently moving towards more dynamic range restriction in broadcast.

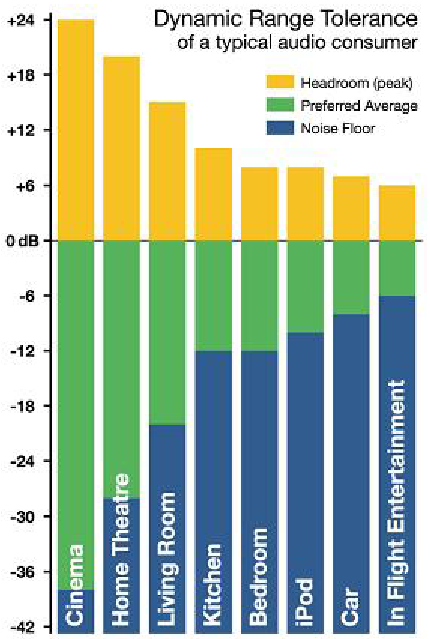


Fig 1. DRT map for consumers under different listening situations.

Dynamic Range of Broadcast Material

Today, program material for TV broadcast is generally aimed at a listener in the Living Room or Kitchen region, see Fig 1. This kind of material should be thought of as having a normal broadcast dynamic range signature.
Commercials, promos and consumer CDs typically have a more restricted dynamic range, and therefore appear loud on TV, where normalization is based only on peak content. This kind of material should be thought of as having a hot dynamic range signature.
On the opposite side we have film production, aimed at a completely different listening scenario, where much softer and much louder level than the average can be reproduced and heard. Production for wide dynamic range listening can also include classical or acoustic music. All material of such nature should be thought of as having a soft dynamic range signature.
Music and entertainment radio is typically aimed at Car listening, so the dynamic range signature is generally hot. The only type of radio with a wider dynamic range typically carries classical music, drama and low key, talk based programming.
To summarize, broadcast material is produced in a way that fits the listening conditions of a wide majority of consumers in the best possible way. The most dramatic difference between program material and consumer requirements concerns feature film. To have a feature film align with domestic listening conditions without loosing too much detail, or distorting the loud parts, low level may need to be brought up by 12-20 dB, and the headroom restricted by 12-16 dB.

Processing for Digital Broadcast

Digital broadcast has the potential to carry more formats at a wider dynamic range than analog. For example, feature films can be presented more like they were mixed and edited, with fewer compromises on the picture as well as on the audio side. However, even for HDTV, audio still needs optimization for a presentation environment different than a cinema, like the picture still needs color space, rate and resolution corrections.
The jumping level problem from analog TV will become bigger if stations transmit feature films with a less suitable dynamic range than today, because film fall way outside the Dynamic Range Tolerance of the average consumer under her domestic listening conditions.
Consequently, dynamic range restriction must take place either at the station, or inside the consumer's receiving device. Dynamic range translation should deal with both overly soft and overly loud parts. Ideally, the perfect re-mapping should happen at the receiving end to accommodate a wide range of listening conditions. Metadata in conjunction with, for instance, Dolby AC3, provides some of these capabilities. However, even if the consumer knows how to adjust the dynamic range of a film to her current listening conditions, the optimum dynamics treatment unfortunately far exceeds the capabilities of an AC3 decoder. The dynamic range control in the codec is acceptable for cut and boost ranges of 4-6 dB, but preparing a feature film for broadcast needs considerably more than this.
If such a large correction is left only to the AC3 decoder, the wide-band gain changes can be quite audible. Film and music dynamic range correction requires a multiband structure so listeners don't sacrifice speech intelligibility, or get subjected to the spectral intermodulation of a crude, wideband range controller.

MDX5.1

The MDX5.1 processor available in DB4 and DB8 is capable of bringing up low level detail, rather than boosting everything, and then having to limit the transients afterwards, see Fig 2. Low level lift can even be applied to specific channels selectively in one, two or three frequency bands.

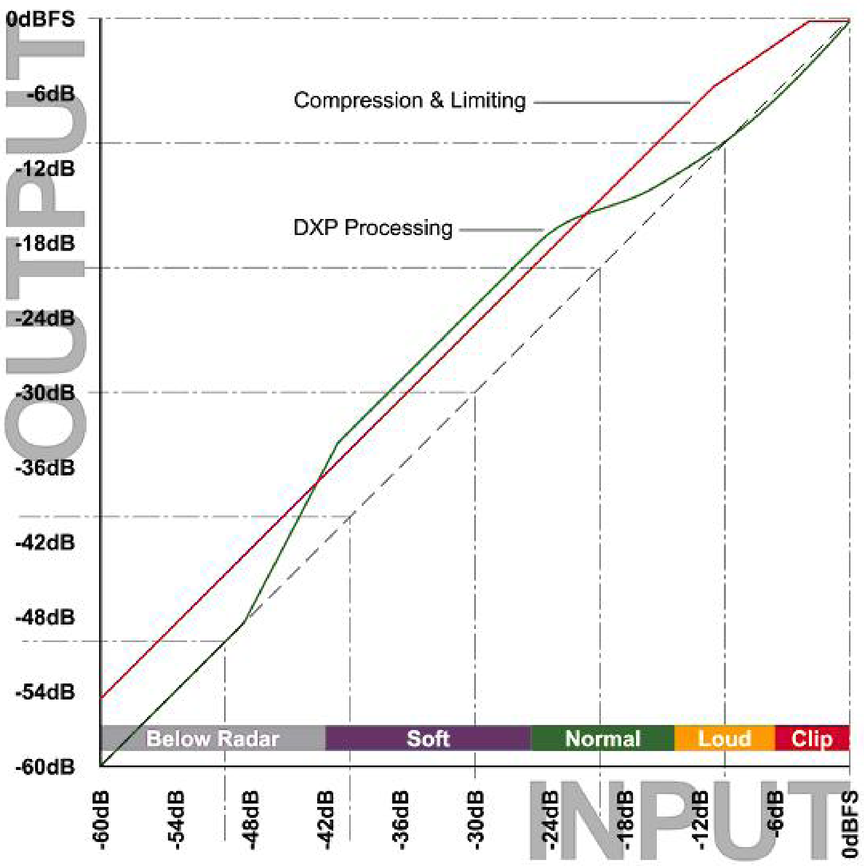


Fig 2. DXP processing vs. traditional Compression and Limiting.
Note how already loud signals are unnecessarily affected when relying on limiting and clipping.

Applications

MDX5.1 is well suited for dynamic range control of any kind of broadcast material. Film, sports, music or game shows. It may be applied during ingest, transmission - or both places.
With suitable parameter settings, high resolution audio can pass through more than one hundred MDX5.1 processors without perceivable degradation of quality The ingenious topology of DB4 and DB8 allows for the processing to be performed instantly (the latency is below 0.5 ms, equivalent to moving a microphone approximately 16 cm or 6 inches), making re-alignment of audio and picture a non-issue.

Processing strategies

The major part of dynamic range translation should be done at the station, leaving only smaller corrections to be performed at the consumer. This ensures competitive audio with regards to consistency, quality and speech intelligibility, and prevents asking more from the AC3 decoder than it can deliver in a civilized manner.

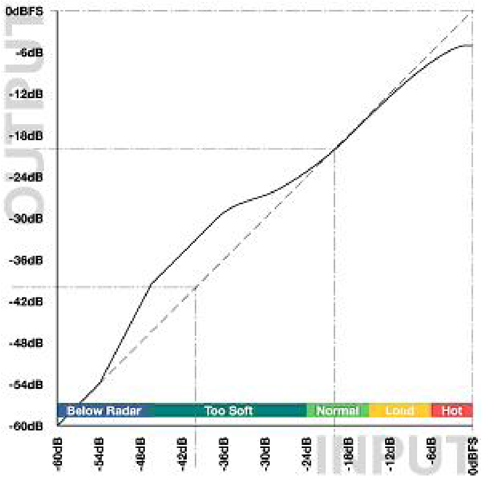


Fig 3. Example of dynamic range re-mapping: From Home Theatre/DVD to Living Room listening conditions (Fig 1).
Fig 3 and Fig 4 show rational transfer characteristics complying with the DRT of the consumer, without affecting levels when they are already on target.

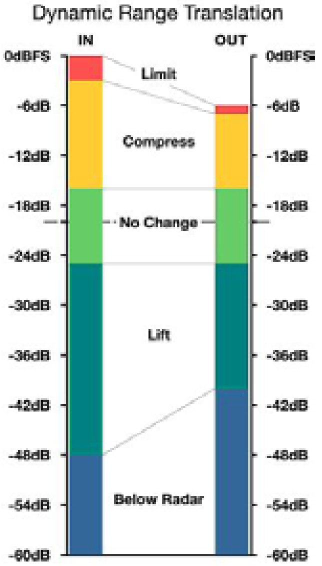


Fig 4. Example of dynamic range re-mapping: From Home Theatre/DVD to Living Room listening conditions (Fig 1).

Basic Operation

On the Main page, MDX5.1 offer Input Gain controls for the Main Channels and for the LFE Channel. This enables positive and negative gain normalization to be performed in the 48 bit domain prior to low level processing and output limiting. These gain controls therefore operate in a safe location, well protected from generating output overloads.

Tip: Use the Input Gains as overload protected level trims in a critical realtime system, such as broadcast, OB or live music.

On the Link pages, the 5 Main channels (L, C, R, SL and SR) can be linked in numerous ways. The concept is to assign a channel to a Sidechain. If all channels are assigned to the same Sidechain, processing is identical on all of them. If a channel is assigned to a different Sidechain, processing on that channel may be different from processing on the other channels.

The DXP pages reveal separate controls for Sidechain 1-3 plus LFE. This enables, for instance, different settings for the Center or Surround channels, where speech intelligibility or low level ambience tend to get lost. Like when a feature film is re-purposed for broadcast or DVD under domestic listening conditions.
If it is required to process more audio channels than 5.1, Engines can be run in parallel to cater for 6.1, 7.1, 10.2, 12.2 or even higher number formats. Parallel Engines attain perfect phase conservation and resolution, and do not compromise audio in any way.

MDX5.1 features 48 bit fixed point processing throughout. Split and reconstruction filters are phase linear when the algorithm is used in multiband modes.

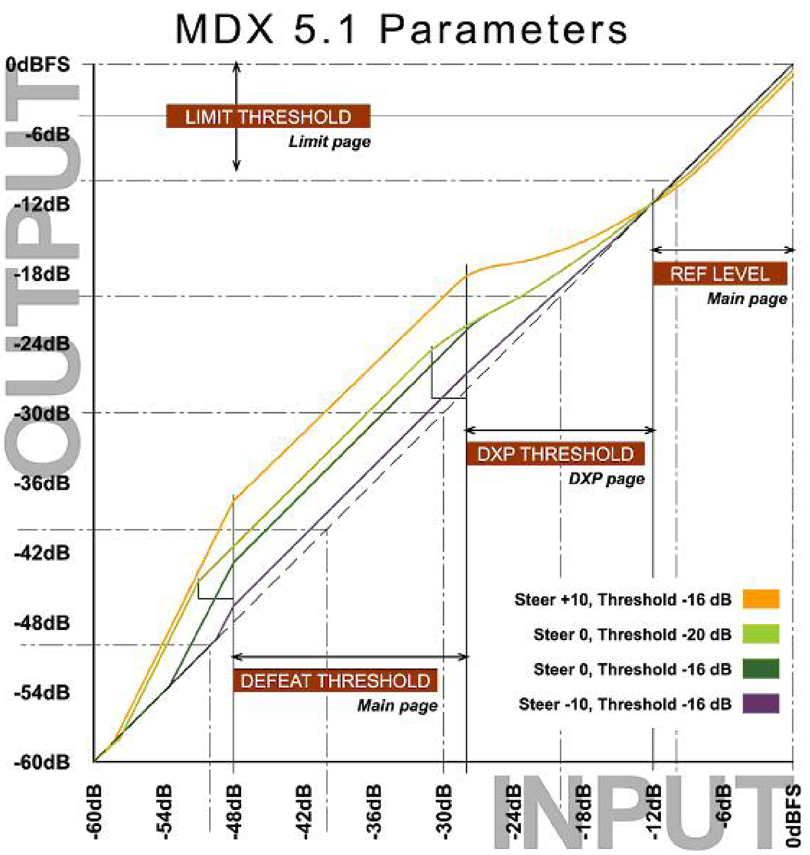


Fig 5. MDX5.1 Level Diagram for different Steer and Threshold settings..
Defeat Threshold relates to DXP Threshold which relates to Ref Level.
Limit Threshold only relates to Digital Full Scale output level.
The Ref Level parameter on the Main page sets the unity gain point for all channels (unless gain offsets are applied), see Fig 5.

The Thresholds on the DXP pages are relative to Ref Level, so in this particular drawing, Ref Level is set at -12 dBFS, while most DXP Thresholds are set at -16 dB. If you invoke the Defeat Threshold, gain reverts to unity for “below radar” input levels. Defeat Threshold is relative to DXP Threshold. In the drawing, the Defeat Threshold is set at -20 dB.
Note, that the lower the DXP Threshold, or the higher a Steer setting, the more low level boost is applied. The low level boost can be different in different channels, and even in different frequency bands.
Also observe that the Limiter threshold setting is not relative to Ref Level, but always referenced to output full scale.

Reading the Gain Meters

Gain meters in indicate absolute gain. The upper segments of a meter gives an indication of the boost and frequency response applied to low level signals, while the lower segments of a meter gives an indication of the current (dynamic) gain and frequency response, *see Fig 6*.
In this example, low level signals are subject to a 5 dB boost in the Low and Hi band. The Low frequency band is currently attenuated by 2 dB, while the Mid and Hi bands are at 0 dB gain.

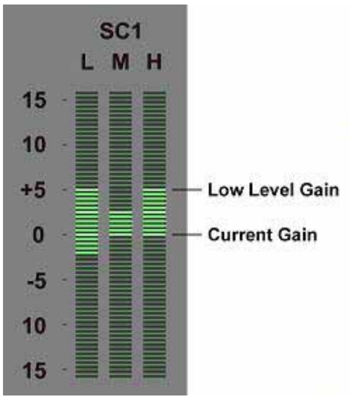


Fig 6. Example of MDX5.1

Gain Meter.The meter shows max low level gain and spectral response, plus current gain and spectral response.In the example, the Low band is currently attenuated by 2 dB, while Mid and Hi bands are at unity gain (0 dB).

Adjustment Tips

The easiest way to specify the yellow area of Fig 1 is to set an appropriate difference between the Ref Level parameter and the Limit Threshold.

Wide dynamic range material for a high resolution delivery might be broadcast with a substantial difference between the two, for instance 15 dB or more.
If the audio bandwidth is low, and the listener environment presumably noisy, the difference between Reference and Limit Thresholds should smaller. For heavily data reduced multi-channel broadcast, best results are typically obtained with a 6-10 dB difference.

When significant data reduction is to be used, also be careful not to allow peaks going all the way to 0 dBFS. Consider bringing down the Limit Threshold between 1 and 4 dB. Judge the quality of loud, spacious material passing through MDX5.1 plus data reduction plus decoding, while listening to the output of the data reduction decoder. Pay special attention to transient distortion, and if the sound image collapses at high levels.

In general, and especially for feature film re-mapping in ingest, start by processing all channels by the same amount. This can be achieved by assigning all channels to Sidechain 1, or by using different sidechains with identical settings. Then conclude if speech in the center channel, ambience in the surrounds or activity in the LFE channel etc. needs special attention and processing.

When it is indicated to bring up dialog level and speech intelligibility, you may end up with something like the level diagram presented in Fig 5. This particular transfer curve has been used successfully at stations with special attention to speech intelligibility.

Compare against the DRT chart, fig 1, and note how the Center channel is given an extra low level advantage compared to the four lateral channels, without the basic mix balance being generally changed. This curve ensures that dialog can still be heard when the words could otherwise be lost to listening room noise. The lateral channels are linked two and two, or all in one group. Presets of this nature is located in Engine Factory Bank F2 (“Loudness, Multichannel”), decade 3, preset 0-3 (“Film Curve C3 - C12”).

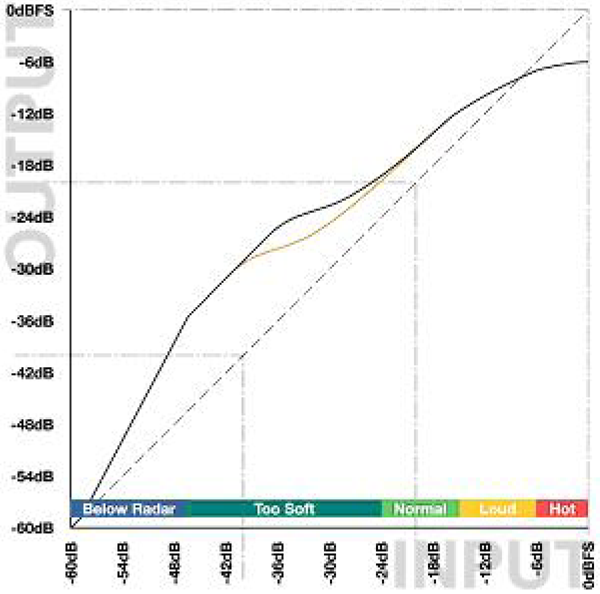


Fig 7. Example of multiband dynamic range re-mapping of a 5.1 feature film to domestic listening conditions. Preset names: “Film Curve C3-C12”.
Black curve: Center channel. Orange curve: L, R, Ls, Rs.

Tip: To produce multiple ingest versions from the same source material, start doing the one for the highest resolution. Lower resolution versions can be achieved by adjusting the Limit Threshold to comply with the alternative delivery format, then adjusting the Ref Level to optimize results under the new, restricted dynamic range conditions. In many cases, no further tweaking will be needed.

Please be advised that some reproduction systems distort when downmixing hot multichannel signals to stereo. Therefore, don’t abuse multichannel formats by bringing all channels close to 0 dBFS at the same time, except for short duration, loud incidents.

Tip: When making the final transmission adjustments, try changing the Ref Level parameter up and down a few dB. This is an efficient way of trimming hundreds of parameters in MDX5.1 at the same time. Listen to the result, while deciding what is the optimum setting for that particular broadcast platform.

MDX5.1 Factory Preset Nomenclature

Engine presets based on the MDX5.1 algorithm is located in Factory Bank F2 (“Loudness, Multichannel”), decade 2 and 3. Presets are labelled Film Curve A-D plus a number.

Film Curve A presets add the same amount of boost to all 5.1 channels. At Reference Level, the gain is unity (0 dB). At low level (- 35 dBFS and below), the number after the “A” in the preset title indicates the amount of low level boost. For example, the preset “Film Curve A6” adds 6 dB of low level gain to all 5.1 channels.

Film Curve C presets add the same amount of boost to all 5.1 channels, but the max gain is achieved earlier for the Center channel than for the rest (like in Fig 5). At Reference Level, the gain is unity (0 dB). At low level (- 35 dBFS and below), the number after the “C” in the preset title indicates the amount of low level boost. For example, the preset “Film Curve C6” adds 6 dB of low level gain to all 5.1 channels.

Film Curve D presets add 3 dB more gain to the Center channel than to the other channels. Max gain is also achieved earlier for the Center channel than for the rest (like in Fig 5). At Reference Level, the gain is +3 dB for the Center channel, but unity (0 dB) for the others. At low level (- 35 dBFS and below), the number after the “D” in the preset title indicates the amount of low level boost. For example, the preset “Film Curve D6” adds 9 dB of low level gain to the Center channel, but 6 dB of low level gain to the rest of the channels.

Main page

Input Gain Normalizer for

Main and LFE channels

Range: -18dB to +18dB
As we process in a 48 bit domain both positive and negative gain normalization can be performed prior to low level processing and output limiting. These gain controls therefore operate in a safe location, well protected from generating output overloads.

Reference Level

Range: -24dBFS to 0dBFS in 0.5dB steps
This parameter sets the reference level in the algorithm. The reference level is the level at which the Threshold parameters will start operating when set to 0dB. E.g. if the Reference Level is set to -18dBFS (often referred to as 0dBu), a Threshold setting at -4dB, will cause the Compressor to start operating at -22dBFS.

Crest

Range: Peak, 6dB, 10dB, 12dB, 14dB, 16dB, 20dB, 24dB, RMS
Select compression method between RMS and PEAK.
The dB steps between RMS and Peak are the dBs needed for a peak-value to override RMS measurement.

DXP Defeat Level

Range: Off, -30dB to -3dB
MDX5.1 may remove low level gain below the threshold set with this parameter to avoid having irrelevant sources (e.g. background noise) become audible. Low level gain is not revoked if the DXP Defeat Level parameter is set to Off.

The Defeat threshold is relative to DXP Band Thresholds, which are relative to Reference Level.
Example: If Reference Level is set at -20 dBFS, Band Thresholds at -15 dB, and DXP Defeat at -22 dB, low level boost starts rolling off at -47 dBFS. See example at page 18.

Nominal Delay

Range: 0 to 15ms
(<2ms in 0.1ms steps. >2ms in 0.5ms steps)
Adds a delay to the passing audio in order to have regulation start "ahead of time". Using this control can reduce the need for peak limiting, and prevent dynamic distortion from being added to sensitive material.

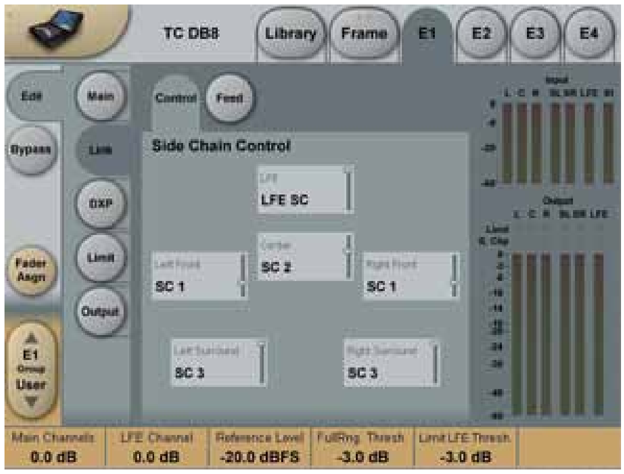
Note that look-ahead is scaled with Attack per band.

Example: If a 5 ms Nominal Delay has been set, and Attack is 10 ms on the low band and 1 ms on the high band, audio is delayed 5 ms on all bands (phase linear topology). However, to prevent pre-transient holes from being generated, Attack regulation starts 5 ms "ahead of time" on the low band, but only a little more than 1 ms "ahead of time" on the high band.

Hi/Lo Crossovers

MDX5.1 uses a phase linear, 48 bit split and re-combination filter structure in order to enable different low level detail boost at different frequencies. This counteracts spectral inter-modulation, and is useful in order to preserve speech intelligibility. Two-band or wide-band DXP processing can be accomplished by setting one or both crossover points to Off.

Link Control page



The Sidechain assignment possibilities in the MDX5.1 are very comprehensive. Carefully selecting which channels should be controlled by which Sidechains, is just as essential as dialing in the correct Threshold and Ratio values.

It is possible to freely select any or none of three Sidechains to control each of the main-channels. This also gives you the option of grouping the channels. In addition to this, the LFE channel has its own Sidechain control. This enables e.g. setting up two Multiband-5.1 algorithms in serial setup, while having six individual Sidechains available, enabling fully individual Sidechain controls of all channels.

At the Feed page it is possible to make additional Sidechain link Inputs, for e.g. having the Center-channel contributing to the Sidechain Inputs of the two Front channels, to create a more coherent sound from the front-channels.

The illustration above reflects the Processing parameter set to MDX5.1 in Normal mode.

Basic operation

At the Setup/Control page it is possible to decide which Sidechains should control which channels. Select any of three Sidechains to be assigned to any of the five Main-channels. You can also chose to pass the channels unprocessed through the algorithm. The LFE channel can be assigned to its own separate Sidechain, or be left unprocessed.

NOTE Setting a channel to unprocessed will preserve the processing delay through the algorithm, keeping the channel time-aligned to the other (processed) channels.

Sidechain Control

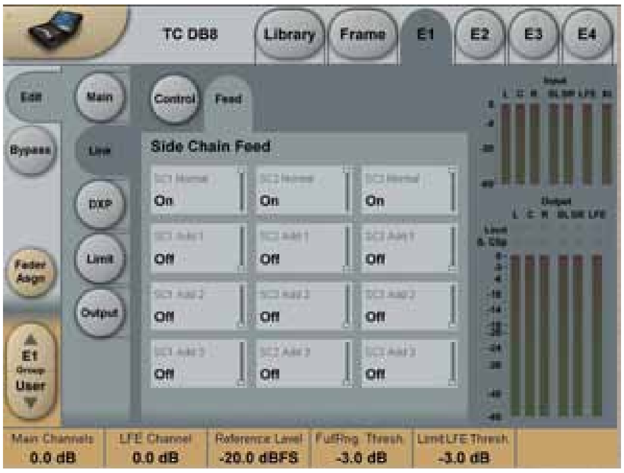
Range - for the five main channels:

- Unprocessed
- Side Chain 1
- Side Chain 2
- Side Chain 3

Range - for the LFE channel:

- Unprocessed
- LFE

Link Feed page



The Setup/SC Feed page holds parameters specifying which Input channels should feed the three Sidechains.

Normal

Range: Off, On

When this parameter is set to "On" the Input channels selected to be controlled by the respective sidechain will also input to the sidechain.

Add 1, Add 2 and Add 3

Range: Off, LFr Max, RFr Max, Cnt Max, LSr Max, RSr Max, Xt Max, LFr Sum, RFr Sum, Cnt Sum, LSr Sum, RSr Sum, Xt Sum.

These parameters enable extra channels to be assigned to the respective Sidechain Input. The extra sidechain Input channels will not be processed by the sidechain.

The Sum settings will add the Input to the sidechain, whereas the Max settings only will contribute to the sidechain if the level exceeds the other Input channel levels.

DXP page

Sidechain Fader Groups

The DXP pages reveal separate controls for Sidechain 1-3 plus LFE. This allows for different settings for the Center or Surround channels, where speech intelligibility or low level ambience tend to get lost, like when a feature film is re-purposed for broadcast or DVD under domestic listening conditions.

If it is required to process more audio channels than 5.1, Engines can be run in parallel to cater for 6.1, 7.1, 10.2, 12.2 or even higher number formats. Parallel Engines attain perfect phase conservation and resolution, and do not compromise audio in any way.

Limit page - Soft Clip

The Limiter page is divided into three Sub-pages. One covering the Softclip section, one Main Limiter and one for the LFE Limiter.

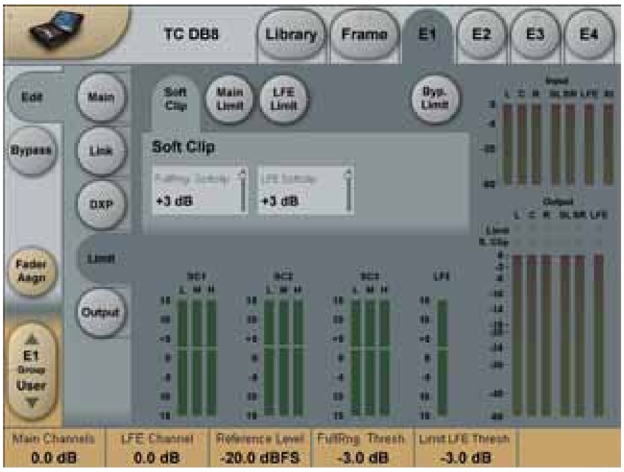
Generic parameters in this algorithm:

Meter Zoom

Press **Meter Zoom** to decrease meter range and have a more accurate metering.

Bypass Limiter

Press to Bypass the Limiter section.



Softclip

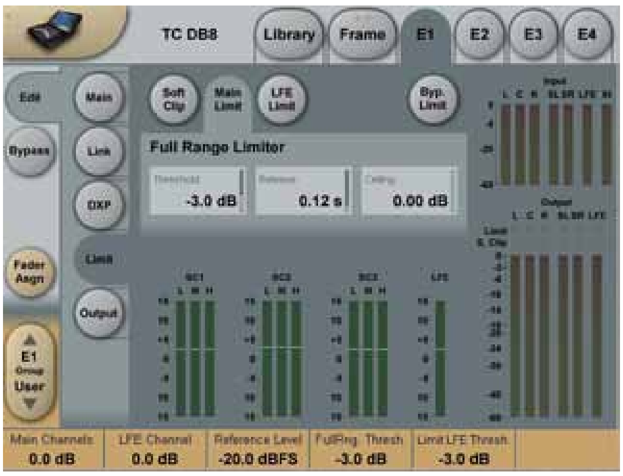
Full Range Softclip

Range: -6dB to Off
Softclipper Threshold setting after the Compressor for the five multiband channels. Threshold is always relative to 0dBFS (Not the Reference Level).

LFE Softclip

Range: -6dB to Off
Softclipper Threshold setting for the LFE channel only.

Limit page - Main



Threshold

Range: -12dB to Off
-6 to OdB in 0.1dB increments
-12 to -6 in 0.5dB increments
Brickwall limiter for the five channels. Threshold is always relative to 0dBFS. LED on each Output meter indicates when Limiter is active.

Release

Range: 0.01 to 1.00 seconds
Release time for the Limiter.

Ceiling

Range: -0.10dB to 0dB
Fine-tuning parameter setting the Ceiling for the Limiter.

WARNING

The Ceiling parameter prevents the Output signal from exceeding the adjusted Limiter Threshold. It can be used to "hide" overloads to downstream equipment, but it does not remove the distortion associated with an overload.

LFE Limiter

Threshold

Range: -12 to +3dB
-6 to + 3 in 0.1dB increments
-12 to -6 in 0.5dB increments
Brickwall limiter for the LFE channel. Threshold is always relative to 0dBFS. LED on each Output meter indicates when the Limiter is active.

Release

Range: 0.01 to 1.00 seconds
Release time for the Limiter.

Ceiling

Range: 0 to -0.10dB in 0.01dB steps.
Fine-tuning parameter setting the Ceiling for the Limiter.

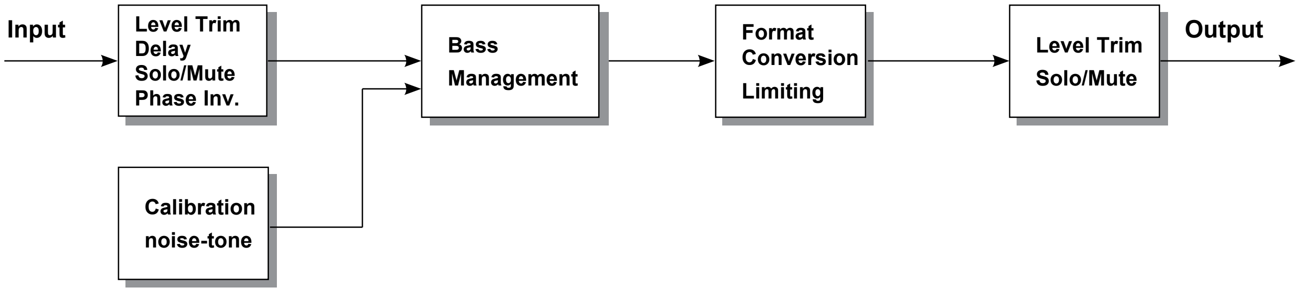
WARNING

The Ceiling parameter prevents the Output signal from exceeding the adjusted Limiter Threshold. It can be used to "hide" overloads to downstream equipment, but it does not remove the distortion associated with an overload.

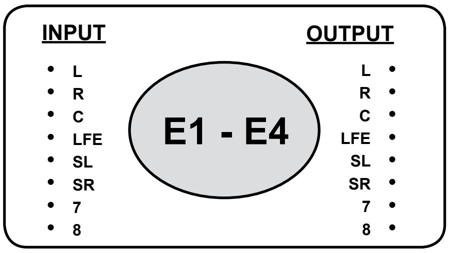
DOWNCONVERT 5.1

Introduction

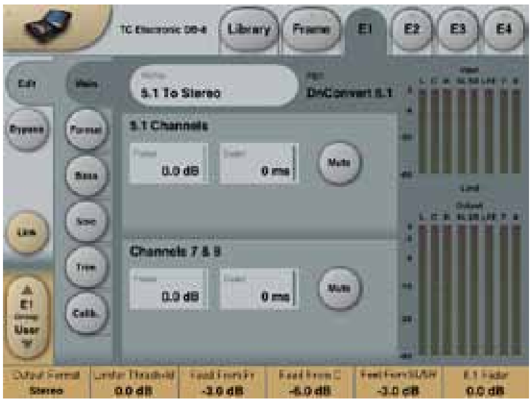
DownConvert-5.1 is an algorithm offering mix-down functionality of different multi-channel formats to LCRS, Stereo or Mono mixes. LFE(Sub) channels can also be Extracted or Distributed to and from the 5.1 main Input channels (Bass-management). Also 5.1 calibration tools with different noise and sine outputs are available. On top of the 5.1 capabilities, DownConvert-5.1 contains two thru channels at I/O 7 and 8, with adjustable level and delay.



Algorithm Inputs/Outputs are distributed as follows:



Main page



Fader 5.1

Parameter range: Off, -120 to 0dB

For the 5.1 I/O channels (L, C, R, SL, SR and LFE), this fader performs Output level control.

Delay 5.1

Parameter range: 0 to 1200ms

For the 5.1 I/O channels (L, C, R, SL, SR and LFE), this parameter Delays all channels simultaneously. The Delay can be changed seamlessly on the fly.

NOTE The individual Sample Delay parameters at the Trim page are additional delay to the setting of this parameter.

Mute 5.1

Parameter range: On/Off

Toggle this switch to Mute all 5.1 output channels.

Fader ch. 7-8

Parameter range: Off, -120 to 0dB

For the I/O channels 7 and 8, this fader performs Output level control.

Delay ch. 7-8

Parameter range: 0 to 1200ms

For I/O channels 7 and 8, this parameter Delays the channels simultaneously. The Delay can be changed seamlessly on the fly.

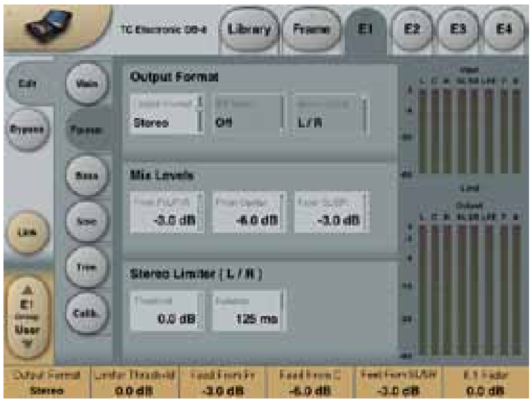
NOTE The individual Sample Delay parameters at the Trim page are additional delay to the setting of this parameter.

Mute ch. 7-8

Toggle this switch to Mute the Output of channels 7 and 8.

Parameter range: On/Off

Format page



The format conversion block enables you to down-mix 5.1 signals to LCRS, Stereo or Mono mix's including Limiter function.

Output Format

The Output Format section is basically used to convert Multi-channel signals to other formats. E.g. when going from a 5.0 mix to a Stereo or mono signal.
Note that the Bass management is placed before this format conversion in the signal chain. Use the distribute part of the Bass-Management to convert from 5.1 to 5.0 mix.

Output Format

Range: 5.1 (=Off or Thru), LCRS, Stereo or Mono
Selects the Output format in which your five main channels Input material will be mixed down to.

90º Mono

90 degrees mono Insert. This option is placed just before the two Limiters, meaning at LFr + RFr when Output format is set to Mono, and LSr + RSr channels when LCRS is selected as Output format.

Mono Output

Range: Center, LFr+RFr
Selects the Output channel when Mono is selected as Output format.

Mix Levels

From L/R

Range: -100dB to 0dB
Sets the Input level from the Left and Right front channels.

This parameter is only available when Output is set to Mono or Stereo.

From Center

Range: -100dB to 0dB
Sets the Input level from the Center channel.

This parameter is only available when Output is set to Mono or Stereo.

From SL/SR

Range: -100dB to 0dB
Sets the Input level from the Left and Right surround channels.

Limiter

Two channels of broadband Output brickwall limiter, that are placed differently according to the selected Output format.

Output format: 5.1 Thru

The Limiter is inactive.

Output format: LCRS

The Limiter operates on the SL and SR channels.

Output format: Stereo

The Limiter operates as a Stereo Limiter on Left and Right front channels.

Output format: Mono

The Limiter operates on the Mono sum Output.

Threshold

Range: -12 to 0dB
Limiter Threshold level for the two limiters available. The Limiters will be placed at LFr + RFr Outputs when Stereo or Mono mode is selected as Output formats, and at LSr + RSr when LCRS is selected as Output format.

Release

Range: 10 to 1000 ms
Sets the Release time for the selected Limiter.

Bass management page

Bass Management

LFE Mode

Range: Extract, Distribute, Inactive
When the LFE Mode parameter is set to Distribute, the Bass Management enables you to add LFE information to the six Output channels in the system. This can normally be compared to a 5.1 -> 5.0 process, but it can also be a 5.1 -> 5.1 process, leaving the LFE channel unprocessed, while adding LFE information to the five Main-channels.
The Bass Management is placed just before the Output Format conversion.

Main Channels

Lo Cut

Range: 10 - 200Hz
Sets the frequency for the Lo Cut filter, on the five main Output channels (LFr, RFr, Cen, LSr, RSr)

Order

Range: Off, 2nd, 4th order
Sets the slope of the Main channels Lo Cut filter.

LFE Channel

Hi Cut

Range: 10 - 200Hz
Sets the frequency for the Hi Cut filter on the LFE channel.

Order

Range: Off, 2nd, 4th order
Sets the slope of the LFE Hi Cut filter.

Main Channels To LFE/ LFE To Main Channels

Depending on the selected Bass Management Mode, Distribute or Extract, the Last section on the Bass page will appear as: "Main Channels to LFE" or "LFE to Main Channels".

Via the parameters: L Front, Center, R Front, L Surround, LFE and R Surround, - it is possible to either:

- feed the main channels with signal from the LFE channel.
- feed the LFE channel with signal from the Main Channels.

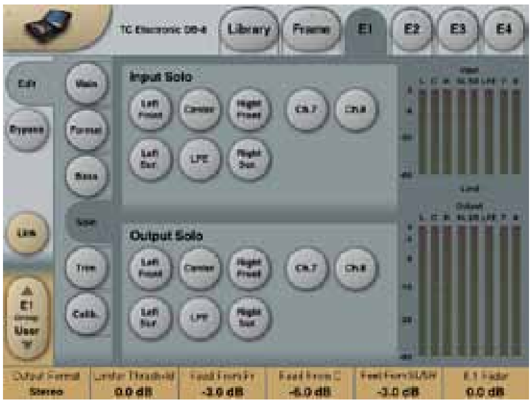
L Front, Center, R Front,L Surround, LFE, R Surround

Range: -100 - 0dBFS
-100 -> -40dB in 3dB steps,
-40 -> 0dB in 0.5dB steps

Main Channels To LFE - Extract mode
In this mode the Level controls are used to extract signal from the Main Channels and feed them to the LFE channel.
Use this mode when converting a 5.0 format to 5.1.

LFE To Main Channels - Distribute mode
In this mode the Level controls are used to distribute the LFE signal to the five Main Channels.
Use this mode when converting a 5.1 format to 5.0.

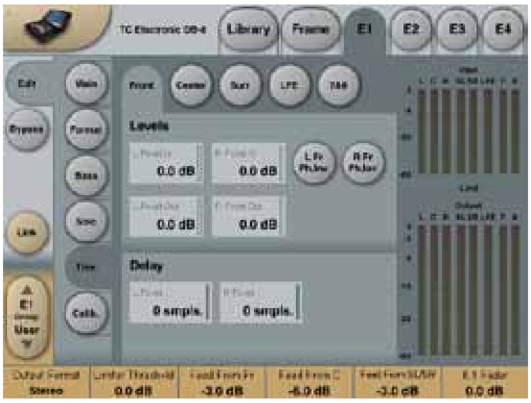
Solo page



Solo buttons

This page contains individual Solo buttons for all Inputs and Outputs. Several channels can be soloed simultaneously.

Trim page



General operation

The tabs in the top of the page (Front, Center, Surr, LFE, Ch.7/8) is used to select parameters for the respective I/O channels. Following parameters are available for each I/O channel:

Input Level

For each of the eight Inputs, separate Input level controls are available.
Parameter range: Off, -120 to 0dB

Output Level

For each of the eight Outputs, separate Output level controls are available.
Parameter range: Off, -120 to 0dB

Phase Invert

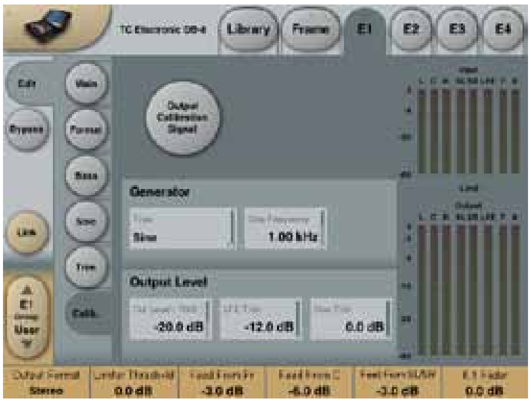
For each of the eight Inputs, the ability to phase-invert the signal 180-degrees is available
Parameter range: On, Off

Delay in samples

For each of the eight channels, fine-adjustable Delay measured in samples can be added.
The Sample Delay is additional to the delay parameter in milliseconds.

NOTE: The corresponding value in milliseconds depends whether the DB8/DB4 is running at 44.1 or 48kHz sample rate. E.g. 48 samples is equal to 1ms at 48kHz and 1.088ms at 44.1kHz.

Calibration page



Test signal generator (Oscillator)

Downconvert-5.1 integrates a comprehensive test-signal generator meant for aligning the monitor system.

When a Test signal is selected, the Input source will not be present on the Outputs.
The Calibration tone is delivered on the very Input of the Downconvert.

Generator

Type

Range: Sine, PinkNoise
WhiteNoise
LPF Pink Noise (Low Pass Filtered Pink noise),
HPF Pink Noise (Hi Pass filtered pink noise)
This parameter selects the Signal generator type.
Default: Sine

Sine Frequency

Range: 20Hz to 20kHz
Selects the frequency when Osc. Type is set to Sine.
Default: 1kHz

Output Level

Output Level (RMS)

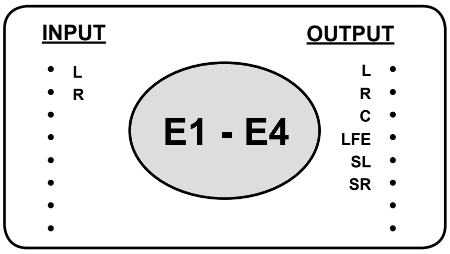
Range: -60 - 0dBFS
-60 -> -6dB in 1dB steps
-6 -> 0dB in 0.1dB steps
Sets the level of the selected generator to all six Output channels.
Default: -20dBFS

LFE Trim

Range: -12 - 0dB, in 0.1dB steps
Attenuates the LFE Output channel relative to the main test-generator level.

Thru - Thru channels are “hardwired” without any adjustment options.

Algorithm Inputs/Outputs are distributed as follows:



Introduction

Unwrap HD in use

Unwrap HD measures phase, delay and spectral differences between a pair of stereo channels to create a 5.1 result. For different program material there will be different optimum settings that best represent the qualities put into the original mix.

Please familiarize yourself with the controls and parameter-ranges on known material before you attempt Unwrap HD.

Setting up

We suggest that you try Unwrap HD with the Output being monitored through the Downconvert 5.1 (e.g. by loading preset "5.1 Monitor Matrix"). This way you can collapse the 5.1 signal to stereo or mono, and make sure the result is still pleasent to listen to.

Try loading some of the Unwrap HD presets. You can A/B the process by pressing Bypass on the Unwrap HD Engine, or collapse the signal to stereo again by selecting Stereo format on the Downconvert Engine, if it is inserted downstream as suggested above.

Time alignment

When all Delays are set at "0", all Outputs from Unwrap HD are aligned with sample precision. The basic Delay through the algorithm in this case is 3.6 ms at 44.1 and 48kHz. Try offsetting the Delays in samples and ms, and note the shift in image.

Delays may be used...
- on the Surround channels to ensure that sounds appear to originate from the front speakers.
- on the Center channel to compensate for its position.
- on the LFE channel to compensate for speaker position or to advance/delay it for artistic reasons.

When the front channels are not assigned the same Delay, please note that a subsequent stereo down-mix may not work so well.

Bit Transparency

When 0% L/R Processing is selected, Input Trims and Output Levels are at 0dB, the Inputs are bit transparently cloned to the L Front and R Front Outputs.

Main Page

Input trims are provided to carefully match the L/R balance. If working from analog tape, adjust balance with a 1kHz calibration tone. If working from a digital master with stereo levels at full scale, it may be necessary to adjust down Input levels a little bit to avoid Unwrap HD overloads.
The L/R Processing parameter determines how much the L and R front channels are processed. At 0% Unwrap HD only adds sound to the 4 other channels preserving the original L and R as they were. Somewhere between 60 and 70% the width of the original mix is typically preserved even though a Center channel is added. Tip: A/B the width soloing the three front channels and toggle by-pass.

Unwrap HD may derive an LFE signal from the Input. It is recommended to lowpass it between 40 and 120Hz using a 2nd or 4th order filter.

Center Page

To better separate and optimize the Center Output, EQ and contour controls are provided.

First set the Ref. Level control at the approximate reference level of the Input signal. For a typical level, set Ref. Level at -10 to -18dB. With a full scale digital Input, Ref. Level would be set high, typically 0 to -12dB. With a quiet or highly dynamic Input, set it between -15 and -25dB.

Then choose between the Contour Styles, and finally apply EQ to the center channel if desired.

Unwrap HD's 48 bit EQ can work wonders on most signals and be used to selectively suppress spectral ranges where the L/R width could otherwise get compromised, or to boost selected frequencies to strengthen the center anchor function.

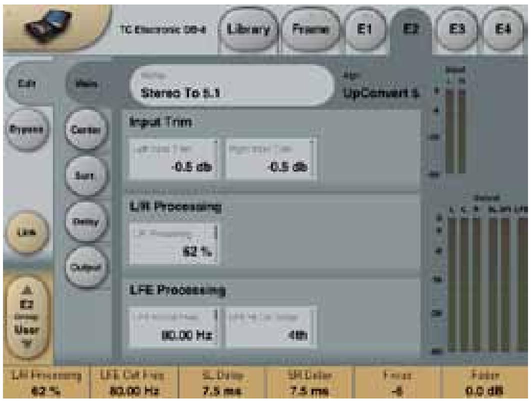
Surround Page

To control the surround channels, decorrelation, EQ and contour controls are provided.

First set the Ref. Level control at the approximate reference level of the Input signal. For a typical level, set Ref. Level at -10 to -18dB. With a full scale digital Input, Ref. Level would be set high, typically 0 to -12dB. With a quiet or highly dynamic Input, set it between -15 and -25dB. Then choose between the Contour Styles, and select a Decorrelation style complementing your program material.

The different decorrelation styles should always be tried. They are highly subjective and best evaluated with the Focus control set at "0". When a style is found, try changing the Focus control to check if further optimization is possible. It may prove convenient to solo the surround channels while doing so.
Now adjust the Decorrelation Tone and EQ parameters. Tuning of the surround parameters is an iterative process and should include the Delay settings as well.

Main



Left/Right Input trim

Range: -100 to 0dB
Input level trim parameters. You may use these parameters to attenuate a too hot input signal.

L/R processing

Range: 0 to 100%
This parameter controls the amount of left/right content of the signal. E.g. if the Center channel level has been increased the perceived stereo image may seam considerably reduced collapsed. Increase the L/R processing to compensate. To find the best suitable setting you may bypass the entire algorithm and compare while focusing on the stereo image.

LFE Processing

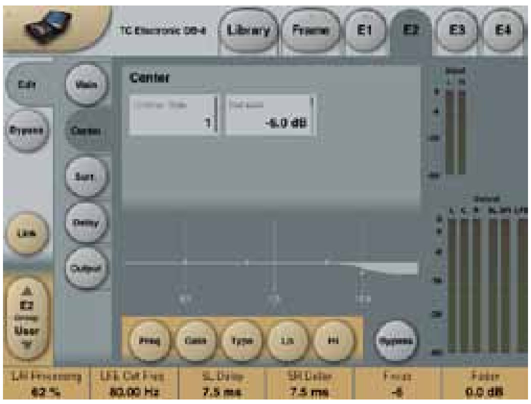
LFE Hi Cut Frequency

Range: 10 to 200Hz
Sets the Hi Cut frequency for the Output from the LFE channel.

LFE Hi Cut Slope

Range: Off, 2nd, 4th
Sets how steep the LFE hi cut filter should operate.

Center



Center Contour Style

Select between different styles as processing for the Center channel Output.
Parameter range: Off and a selection of styles.

Center Contour Threshold

Sets the Threshold point for the Contour Style to be operating.
Parameter range: -25 to 0dB

EQ

The EQ for the Center channel features four-band parametric EQ with high- and low-pass filters switchable between Notch, Parametric, Shelving and Cut filters.

Basic operation
- Select Freq, Gain or Type to access the same parameter for the four EQ bands.
- Select Lo or Hi to access the three parameters for the individual EQ band.
- Press Bypass EQ to bypass the entire EQ.

NOTE Bypass does not affect the selected Contour Style.

Type Selector

Press Type and use faders 1-4 to select filter types.
For Lo and Hi filters select between filter types:

Parametric, Notch, Shelve and Cut.
For Mid 1 and Mid 2 filters select between filter types: Parametric and Notch.

Freq

Press Freq and use Faders 1 to 4 to adjust frequency for each of the four bands.

Range - Lo band	: 20Hz to 5kHz
Range - Mid1 band	: 20Hz to 20kHz
Range - Mid2 band	: 20Hz to 20kHz
Range - Hi band	: 500Hz to 20kHz

Gain

Press Gain and use Faders 1 - 4 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:

Lo Gain	: -12dB to +12dB
Mid1 Gain	: -12dB to +12dB
Mid2 Gain	: -12dB to +12dB
Hi Gain	: -12dB to +12dB

Range for the Notch filter:

Lo Gain	: -100dB to 0dB
Mid1 Gain	: -100dB to 0dB
Mid2 Gain	: -100dB to 0dB
Hi Gain	: -100dB to 0dB

Type

Press and use Faders 1-4 to set BW value for each of the 4 EQ bands.

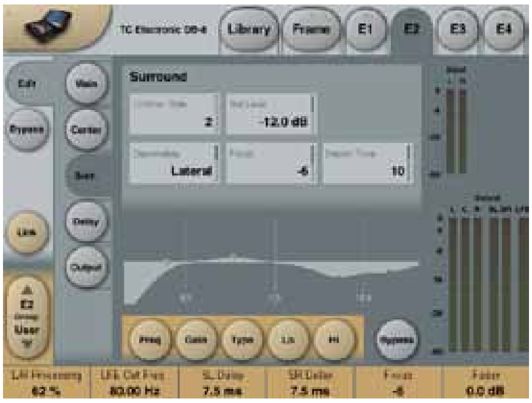
Range for the Notch filter:
Lo BW : 0.02oct to 1oct
Mid1 BW : 0.02oct to 1oct
Mid2 BW : 0.02oct to 1oct
Hi BW : 0.02oct to 1oct

Range for the Parametric filter:
Lo BW : 0.1oct to 4oct
Mid1 BW : 0.1oct to 4oct
Mid2 BW : 0.1oct to 4oct
Hi BW : 0.1oct to 4oct

Range for the Shelve filter:
Lo BW : 3dB/oct to 12dB/oct
Hi BW : 3dB/oct to 12dB/oct

Range for the Cut filter:
Lo BW : Bessel or Butterworth
Hi BW : Bessel or Butterworth

Surround



Contour Style

Select between different styles as processing for the surround channels Output.
Parameter range: Off and a selection of styles.

Contour Threshold

Range: -25 to 0dB
Sets the Threshold point for the Contour Style to be operating.

Decorrelate Style

Range: A selection of styles
Select between different styles of decorrelating the sound in the two surround Output channels.

Decorrelate Amount

Range: 0 - 100%
Set how much you want to decorrelate the sound in the surround Outputs.

Decorrelate Tone

Range: +/- 40 steps.
Adjust the tone (color) of the decorrelated part of the sound on the surround Outputs.

EQ

The EQ for the Center channel features four-band parametric EQ with high- and low-pass filters switchable between Notch, Parametric, Shelving and Cut filters.

Basic operation
- Select Freq, Gain or Type to access the same parameter for the four EQ bands.
- Select Lo or Hi to access the three parameters for the individual EQ band.
- Press Bypass EQ to bypass the entire EQ.

NOTE Bypass does not affect the selected Contour Style.

Type Selector

Press Type and use faders 1-4 to select filter types.
For Lo and Hi filters select between filter types:

Parametric, Notch, Shelve and Cut.
For Mid 1 and Mid 2 filters select between filter types: Parametric and Notch.

Freq

Press Freq and use Faders 1 to 4 to adjust frequency for each of the four bands.
Range - Lo band : 20Hz to 5kHz
Range - Mid1 band : 20Hz to 20kHz
Range - Mid2 band : 20Hz to 20kHz
Range - Hi band : 500Hz to 20kHz

Gain

Press Gain and use Faders 1 - 4 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:
Lo Gain : -12dB to +12dB
Mid1 Gain : -12dB to +12dB
Mid2 Gain : -12dB to +12dB
Hi Gain : -12dB to +12dB

Range for the Notch filter:
Lo Gain : -100dB to 0dB
Mid1 Gain : -100dB to 0dB
Mid2 Gain : -100dB to 0dB
Hi Gain : -100dB to 0dB
Type
Press and use Faders 1-4 to set BW value for each of the 4 EQ bands.

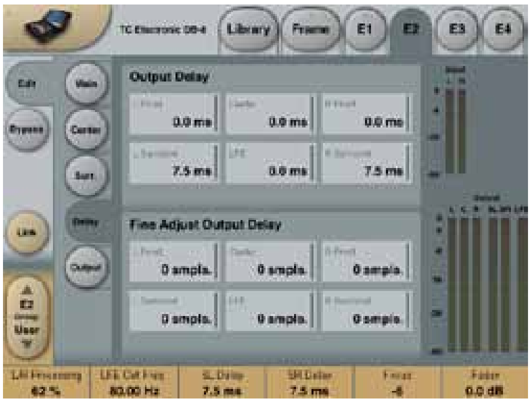
Range for the Notch filter:
Lo BW : 0.02oct to 1oct
Mid1 BW : 0.02oct to 1oct
Mid2 BW : 0.02oct to 1oct
Hi BW : 0.02oct to 1oct

Range for the Parametric filter:
Lo BW : 0.1oct to 4oct
Mid1 BW : 0.1oct to 4oct
Mid2 BW : 0.1oct to 4oct
Hi BW : 0.1oct to 4oct

Range for the Shelve filter:
Lo BW : 3dB/oct to 12dB/oct
Hi BW : 3dB/oct to 12dB/oct

Range for the Cut filter:
Lo BW : Bessel or Butterworth
Hi BW : Bessel or Butterworth

Delay



Output Delay

Range: 0 - 200ms
For each of the six Outputs it's possible to adjust the Delay time in Milliseconds.

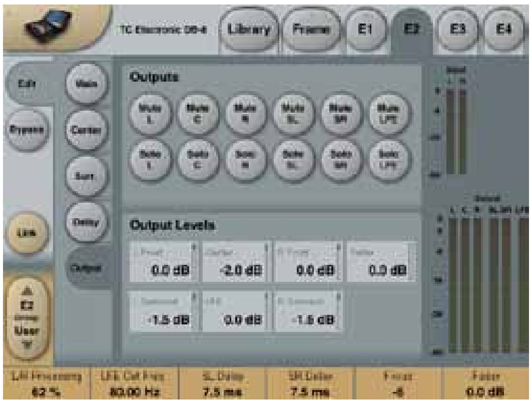
Fine Adjust Output Delay

Range: 0 - 100 samples
In addition to the Output Delay in milliseconds, it's possible to adjust each of the six Output Delays in samples resolution.

NOTE The total Delay on an Output channel is the normal ms Delay setting, PLUS the Sample Delay setting.

NOTE The actual time a Delay set in Samples varies depending on running Sample Rrate. E.g. if you are running 48kHz, a 48 samples delay equals 1ms, and at 96kHz it equals 0.5ms.

Output



Outputs

Mute

Range: Muted/Unmuted
Sets the Mute-status on the Output for each of the 6 channels.

Solo

When a Solo button is selected, the Outputs of all the five remaining channels will be set to Off, but they can be selected as additional solo channels.

Output Levels

Range: -120 to +12dB
Individual Output levels for the six Output channels.

Fader

Range: -120 to 0dB
Fades all six Outputs simultaneously.
Preserves the individual Output levels until either the max. or min. value is reached.

Introduction

UpCon HD is an automatic, realtime 5.1 up-conversion audio processor for DB8 and DB4. It continuously monitors the format of the incoming audio, and if the signal falls back from a true 5.1 to stereo, UpCon HD seamlessly cross-fades into a convincing 5.1 surround up-conversion without adding any interruptions or artifacts. Detection does not require metadata or GPIs to function correctly, and the processing delay is only 2.8 ms (less than 1/10th frame). Therefore, no extra delays are required to maintain A/V sync.

UpCon is used in Transmission or Ingest to ensure the availability of an uninterrupted 5.1 signal, or to extend the production capabilities of an audio studio from stereo to 5.1 using the UpCon+ functionality described.

Note that this algorithm may be operated in different modes. Make sure to select the one which fits your station environment best possibly. In all modes, the 5.1 input is always fed to channel 1-6, while a stereo signal may either be fed to inputs 1-2 (i.e. the same channels also used for 5.1), or a stereo signal may be input through separate physical channels 7-8. Please find more details in the UpCon Applications section of this manual section.

When deciding on a generic station setting, a recommended starting point may be found in the Engine preset bank, F4-0-0, under the preset name "UpCon HD BS1770". This preset is typically loudness neutral when using the ITU-R BS1770 loudness measure, i.e. the 5.1 output will typically have close to the same Loudness and Loudness Range as the stereo input.

The first part of this manual section is a description of all parameters. Be sure also to read the following section giving in-depth information and operational tips. Also refer to the Unwrap HD introduction.

Main page

Left/Right Input trim

Range: -100 to 0dB
Input level trim parameters. You may use these parameters to attenuate a too hot input signal.

L/R processing

Range: 0 to 100%
This parameter controls the amount of left/right content of the signal. E.g. if the Center channel level has been increased the perceived stereo image may seam considerably reduced or collapsed. Increase the L/R processing to compensate. To find the best suitable setting you may bypass the entire algorithm and compare while focusing on the stereo image.

LFE Hi Cut freq and Hi Cut Slope

Correct settings of these parameters depend on the quality of the satellite speakers on your system. Best result from the LFE channel is achieved if the HiCut Freq is set relatively low (e.g. around 80Hz) with a 4th order filter. However, these settings require that the satellite speakers perform well to as low as 100-120Hz. Good results with smaller satellite speakers however, can be achieved with a higher set LFE frequency and a 2'nd order filter. - The main object is to cover the entire frequency range yet having the LFE HiCut set as low as possible.

Center page

Contour Style

Range: 1-4
Contour styles emphasize different properties of the source material. Experiment with the setting for an optimum fit to typical material.

Ref Level

Range: -25dB to 0dB
Set reference level according to your system settings.

EQ

The EQ for the Center channel features a four-band parametric EQ with high- and low-pass filters switchable between Notch, Parametric, Shelving and Cut filters.

Basic operation
- Select Freq, Gain or Type to access the same parameter for the four EQ bands.
- Select Lo or Hi to access the three parameters for the individual EQ band.
- Press Bypass EQ to bypass the entire EQ.

NOTE: Bypass does not affect the selected Contour Style.

Type Selector

Press Type and use faders 1-4 to select filter types.
For Lo and Hi filters select between filter types:

Parametric, Notch, Shelve and Cut.
For Mid 1 and Mid 2 filters select between filter types: Parametric and Notch.

Freq

Press Freq and use Faders 1 to 4 to adjust frequency for each of the four bands.

Range - Lo band : 20Hz to 5kHz
Range - Mid1 band : 20Hz to 20kHz
Range - Mid2 band : 20Hz to 20kHz
Range - Hi band : 500Hz to 20kHz

Gain

Press Gain and use Faders 1 - 4 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:
Lo Gain : -12dB to +12dB
Mid1 Gain : -12dB to +12dB
Mid2 Gain : -12dB to +12dB
Hi Gain : -12dB to +12dB

Range for the Notch filter:
Lo Gain : -100dB to 0dB
Mid1 Gain : -100dB to 0dB
Mid2 Gain : -100dB to 0dB
Hi Gain : -100dB to 0dB
Type
Press and use Faders 1-4 to set BW value for each of the 4 EQ bands.

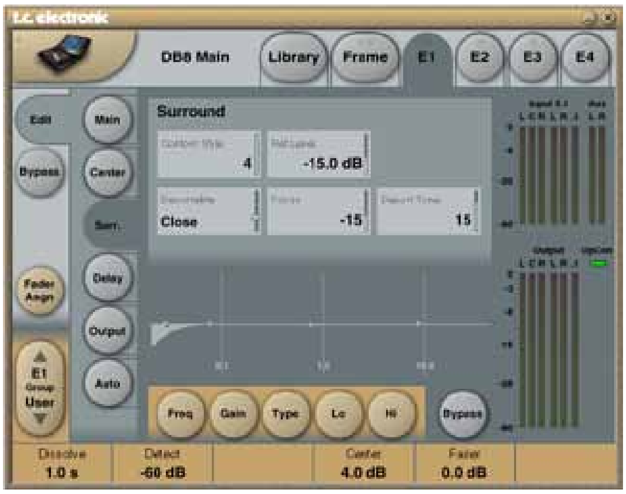
Range for the Notch filter:
Lo BW : 0.02oct to 1oct
Mid1 BW : 0.02oct to 1oct
Mid2 BW : 0.02oct to 1oct
Hi BW : 0.02oct to 1oct

Range for the Parametric filter:
Lo BW : 0.1oct to 4oct
Mid1 BW : 0.1oct to 4oct
Mid2 BW : 0.1oct to 4oct
Hi BW : 0.1oct to 4oct

Range for the Shelve filter:
Lo BW : 3dB/oct to 12dB/oct
Hi BW : 3dB/oct to 12dB/oct

Range for the Cut filter:
Lo BW : Bessel or Butterworth
Hi BW : Bessel or Butterworth

Surround page



The parameters on the Surround page are difficult to describe precisely as they have slightly different impact depending on the source material. Experiment !

Contour Style

Range: 1 - 4
The Contour Style parameter decides which type of the signal to focus on. E.g. speach, music etc.
Depending on the source material the styles may emphasize certain sources or timbre. Experiment with the setting for an optimum fit to typical material.

Ref. Level

Range: -100 to 0dB
Ref. level should be set at the approximate reference level of the Input signal. For a typical level, set Ref. Level at -10 to -18dB. With a full scale digital Input, Ref. Level would be set high, typically 0 to -12dB. With a quiet or highly dynamic Input, set it between -15 and -25dB.

Decorrelate

Options: Dry, Close, Dorsal, Lateral, Diffuse or Wet
Select between different styles of decorrelation in the surround output channels. These styles in combination with the Focus and Tone parameters positions the source material.

Focus

Where the Decorrelate parameter positions the source material, the Focus parameter will enhance or attenuate the perceived position.

Tone

Once Decorrelation type and Focus is set the Tone may further enhance or smoothen the surround information.

Delay page

Output Delay

0 to 100ms output delay for each of the six channels. The Delay may be used to align or compensate according to the listening position.

Output page

Outputs

Mute and Solo functions for all channels.

Output Levels

Individual output levels or all channels.
The “Fader” level allows for simultaneous attenuation of all channels using a single fader.

From software version 2.00 upwards, UpCon can be used with three distinctively different input routing and automatic switching configurations. Make sure to choose the input configuration and Automation Mode that fits your station infrastructure and requirements the best. Note that the basic routing is set on the Frame/Routing page.

Same Inputs for Stereo and 5.1

In this configuration, audio is always fed to the 5.1 inputs of UpCon, regardless if the incoming format is Stereo, LtRt or 5.1. A Stereo or LtRt signal uses only two of the six input channels (green inputs on Fig 1), while a 5.1 signal makes use of all six. When the input falls back to Stereo or LtRt, UpCon cross-fades into up-conversion mode. If the input becomes 5.1,

Using this mode, UpCon only looks at the Main inputs (1-6), while Aux inputs are always kept separate (e.g. for Dolby E). This is equivalent to the “Main Only” mode in previous versions of UpCon, but now with an important Aux Thru addition suitable for e.g. handling of codecs, see below.

To select this mode of operation, adjust the Auto Processing parameter to “Main Only” and route incoming stereo as well as 5.1 signal to inputs 1-6. Data reduced audio may be kept separately on I/O 7-8.

Two Alternating Inputs with 5.1 Input Priority

This configuration requires audio to be fed to different inputs depending on its format. 5.1 is fed to the Main Inputs (channels 1-6), while Stereo or LtRt is fed to the Aux Inputs (channels 7-8). Aux inputs are only enabled when a 5.1 signal is not present. In this situation the Aux inputs are automatically upconverted to 5.1.

If both inputs become active, priority is given to the 5.1 input, while the Aux input is muted. The stereo input may be used as fallback/local insert/redundancy input. All changes are applied doing smooth crossfades.

To select this mode of operation, adjust the Auto Processing parameter to “Main 5.1 Priority” and route incoming 5.1 to inputs 1-6, incoming stereo or LtRt to inputs 7-8.

Two Alternating Inputs with Stereo Input Priority

This configuration requires audio to be fed to different inputs depending on its format. 5.1 is fed to the Main Inputs (channels 1-6), while Stereo or LtRt is fed to the Aux Inputs (channels 7-8). UpCon only enables 5.1 inputs when the Aux stereo/LtRt signal is not present. When an Aux input is available, UpCon simultaneously crossfades into upconversion.

If both inputs become active, priority is given to the Stereo input, while the 5.1 input is muted.

To select this mode of operation, adjust the Auto Processing parameter to “Aux Priority” and route incoming 5.1 to inputs 1-6, incoming stereo or LtRt to inputs 7-8. You may also feed stereo to both groups of inputs. In case both stereo inputs become active at the same time, priority is given to inputs 7-8.

Note: Aux Priority mode may also be used to crossfade between two stereo signals, and for UpCon+ functionality.

UpCon and MPEG, AAC, AC3, Dolby E

With software 2.00 upwards, a bit transparent pass-through from input 7-8 to output 7-8 has been established. Whatever Auto Processing mode you have selected, inputs 7-8 are available on outputs 7-8. This functionality was requested by broadcasters using linear audio on some channels and data-reduced signals on others (e.g. MPEG, AAC, AC3, Dolby E etc.). The most suitable automation mode when handling both linear audio and a codec is normally “Main Only”, see above.

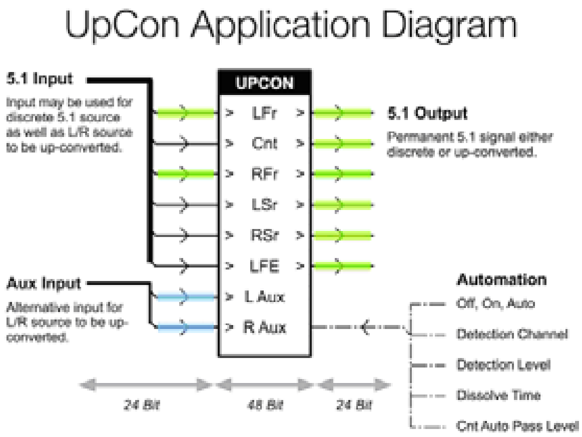


Fig 1. UpCon Input Routing.

Green inputs are used for Stereo and LtRt with Constant Input routing.*

Blue inputs are used for Stereo and LtRt signals with Alternating Input routing.*

In both modes, 5.1 input signals are fed to the 5.1 inputs.

** To see colors - download PDF from our website at www. tcelectronic.com*

Station Routing

DTV stations handle Loudness and Format control differently. How much processing is done at the station, and how much is left to the consumer, varies from station to station, as does the generation and reliance on metadata.
UpCon does not need metadata to function correctly, but it can easily be integrated even where stations take metadata usage to the extreme (see example 2 in Fig 2). More typical scenarios are shown in example 1 and 3, where the station doesn't spend time and money on more metadata handling equipment than necessary. The advanced detection circuitry in Upcon ensures consistent operation without the need for metadata.

UpCon automatically switches between 24 bit-transparent bypass and Up-conversion based on the settings in the Auto page (Fig 3). The algorithm may also switch between two incoming stereo signals.

Processing selects between three different up-conversion and switching modes. The Automation Processing parameter in combination with how you route signal to UpCon, defines how the algorithm operates. Please refer to the first page of this section for details.

When UpCon is up-converting, the green UpCon indicator next to the output meters is lit.

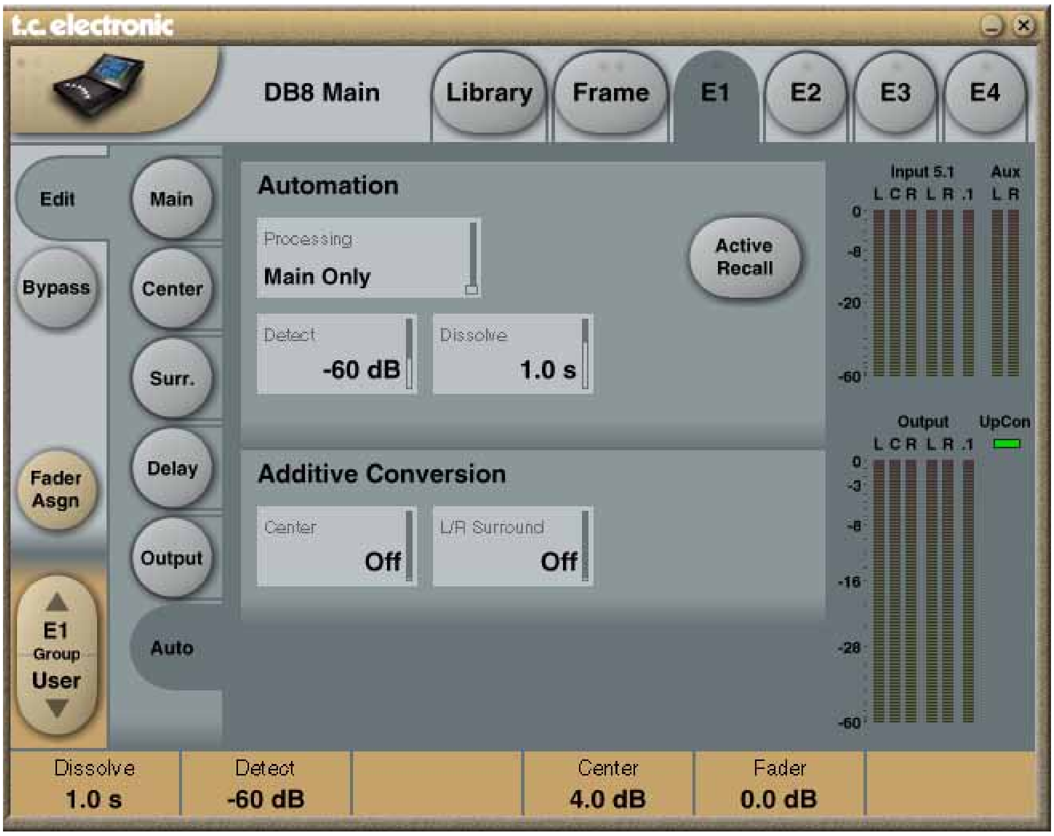


Fig 3. The Auto page parameters.
Note green UpCon indicator to show up-conversion currently active.

Detection Modes

To avoid the need for metadata to control the switching between formats, Upcon's detector makes use of advanced sensing with appropriate hysteresis and timing computations. The Detect parameter sets the conditions for engaging or disengaging up-conversion. The 24 bit, 20 bit and 16 bit settings enable detection based on the presence of dither. The -60, -50, -40, -30, and -20 dB settings enable detection based on audio level.

When the Main Only mode is selected, the automation system measures the Center, L and R Surround inputs. For instance, if Detect is set at “16 bit”, UpCon reads dither on the C, LSr and RSr inputs. If dither is available on any of them, UpCon assumes that a 5.1 signal is available, and cross-fades into 5.1 bypass.
Note that this automation mode gives priority to a 5.1 signal, and that outputs are never muted. When no 5.1 signal is present, up-conversion is engaged.
When the Aux Priority mode is selected, the automation system measures the L and R Aux inputs. For instance, if Detect is set at “-60 dB”, UpCon reads the audio signal on the Aux inputs. If audio is available on any of them, UpCon assumes that a 5.1 signal is not available, and cross-fades into up-conversion based on the Aux inputs.
Note that this automation mode gives priority to the Aux input, though the 5.1 inputs can be used simultaneously with the Aux inputs to add to the up-conversion (“UpCon+” functions). When no signal is present on the Aux inputs, up-conversion is bypassed.

Dissolve

Sets the cross-fade time between 5.1 and up-conversion. The green UpCon indicator reads out the up-conversion status before the Dissolve time is applied.

NOTE The outputs of UpCon are never muted. Dissolve only sets the duration of the cross-fade.

Active Recall

Sets the basic state of UpCon when the preset is recalled. If Active Recall is active, the preset will recall with up-conversion engaged. This function enables recall of different up-conversion presets without disengaging up-conversion even shortly. (The difference between Active Recall or not may be noticeable when long Dissolve times are used).
Note: Presets that should recall engaged have to be saved with Active Recall enabled.
UpCon preset examples are found in Factory Preset Bank F4-6.

UpCon+

UpCon offers the ability to transform a stereo broadcast studio into a 5.1 production environment. Besides normal stereo production tools, only a DB4 or DB8 plus extra speakers are needed.

UpCon Plus preset examples are found in Factory Preset Bank F4-8. In these presets, note that the PLUS controls (Center and Surround) are instantly accessible on fader 3 and 4.

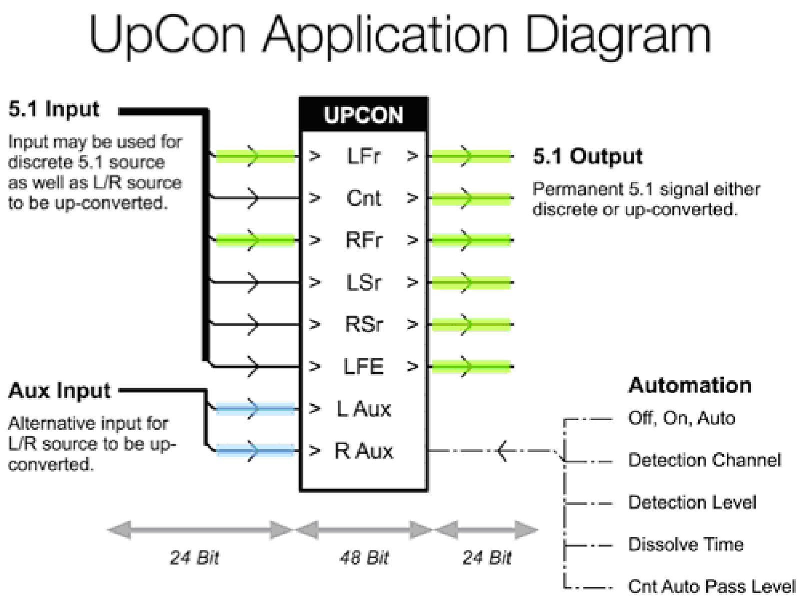


Fig 4. UpCon Plus application example.
UpCon together with a Monitor Matrix Engine provides a 5.1 simulcast upgrade solution for a stereo studio or OB truck - including monitor format control and confidence check.

Though the Monitor Matrix preset loaded to another engine inside DB4 or DB8 is not strictly needed to achieve stereo and 5.1 simulcast, it is recommended for compatibility check in the production suite. The Monitor Matrix provides easy access to both the stereo signal, the 5.1 up-mix, as well as a subsequent down-mix of the 5.1.

PLUS parameters

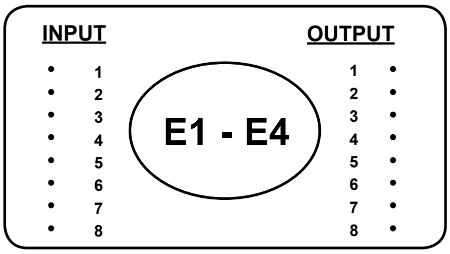
These parameters offer additional features when a stereo signal is input to the Aux channels (Aux Priority configuration). Several broadcasters have asked for tools to add true extra audio features to a 5.1 signal, even though the basic production is done in mono or stereo.

<u>*Example 1*</u>
A Sports event or Music concert transmission gets its basic 5.1 audio from up-converted stereo, but an audience/ambience signal is additionally fed to the L and R Surrounds. The basic production sound is fed to UpCon's Aux inputs, while the add on material is fed to the L and R Surround 5.1 inputs. Adjust the L/R Surround parameter to get the desired amount of additional ambience sound in the rear channels.

<u>*Example 2*</u>
A News transmission gets its basic 5.1 audio from up-converted stereo, but additional studio reader audio is required in the Center channel. The basic production sound is fed to UpCon's Aux inputs, while the add on mono reader is fed to the Center 5.1 input. Adjust the Center parameter to get the desired amount of additional studio sound to the Center channel.

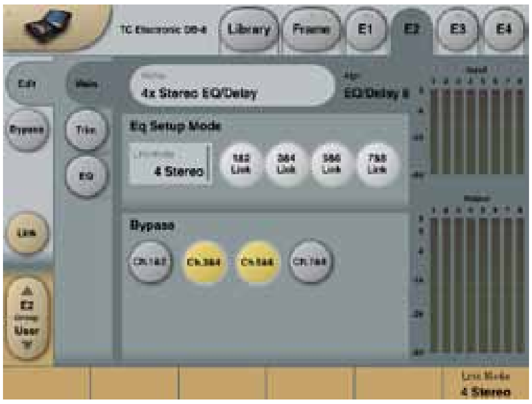
EQ-DELAY 8

Algorithm Inputs/Outputs are distributed as follows:



EQ/Delay-8 is a multi-channel EQ and Delay algorithm, with built-in flexibility to cover several different applications and I/O-format setups.

Main page



Link Mode

Select between two basically different channel setups:
1) Four stereo/dual-mono
2) 5.1 plus one stereo/dual-mono

When switching between the two modes, I/O-labels and linking functionality changes to fit the different applications in the best possible way.
The number of available EQ-filters and Delay-time is unchanged when switching between the two modes.

Link butons

When “4 Stereo” is selected, four individual link buttons is available for linking in stereo-pairs or leave the channels for individual operation (dual-mono).

NOTE: When “5.1 & ch.7/8” is selected, the choice of linking all five main-channels or just the front and surround set of channels are available. On top of this, channels 7 and 8 can be linked or left unlinked for individual operation.

When linking a stereo pair the lowest channel number settings will be copied into the higher number. When linking all Main-channels, the Center settings will be copied to the four remaining channels.

Bypass buttons

Depending on the selected channel setup and activated links, corresponding Bypass buttons are available.

Trim page

Press **Front/Center/Surr. or LFE** (side tab) to access parameters for each of the channel groups.

Following parameters are available for each I/O channel:

Input Level

Parameter range: Off, -120 to 0dB
For each of the 8 Inputs, separate Input level controls are available.

Output Level

Parameter range: Off, -120 to 0dB
For each of the eight Outputs, separate Output level controls are available.

Delay in milliseconds

Parameter range: 0 to 1000ms.
For each of the eight channels, a Delay measured in milliseconds can be added for aligning purposes. The Delay can be changed seamlessly on the fly.

Delay in samples

For each of the eight channels, fine-adjustable Delay measured in samples can be added.
The Sample Delay is additional to the delay parameter in milliseconds.

NOTE: The corresponding value in milliseconds depends whether the DB8/DB4 is running at 44,1 or 48kHz sample rate. E.g. 48 samples is equal to 1ms at 48kHz and 1,088ms at 44,1kHz.

EQ page

Basic operation

The available buttons will be labeled depending on the selected Link Mode at the Main page.

Introduction

This digital EQ features a four-band parametric EQ with high- and low-pass filters switchable between Notch, Parametric, Shelving and Cut filters. The needle sharp notch filter has a range down to 0.01 octave and the shelving filters has a variable slope, ranging from gentle 3 dB/oct over 6 and 9 to 12dB/oct. Cut filters are switchable between 12dB/oct maximum flat amplitude (Butterworth) or flat group delay (Bessel) types. The parametric equalizer features a natural and well defined bandwidth behavior at all gain and width settings:

Basic operation

- Press keys **Lo, Mid1, Mid2 and Hi** to activate/deactivate the EQ bands.
- Select Freq, Gain, Type or Lo/Hi to access all four parameters on individual bands.
- Press **Bypass EQ** to bypass all four bands.

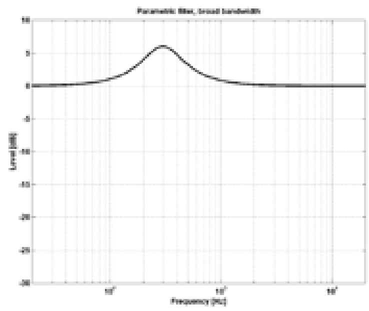
Type Selector

- Press **Type** and use faders 1-4 to select filter types.

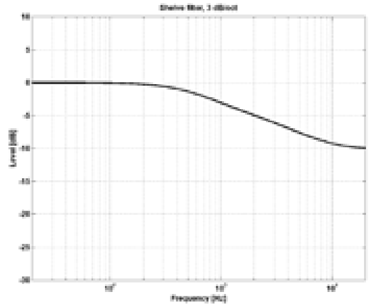
For Lo and Hi filters select between filter types:
Parametric, Notch, Shelve and Cut.

For Mid 1 and Mid 2 filters select between filter types:
Parametric and Notch.

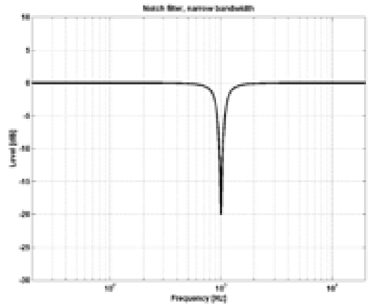
Parametric Filter - Broad type



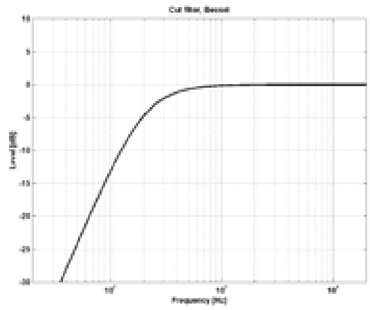
Shelving Filter



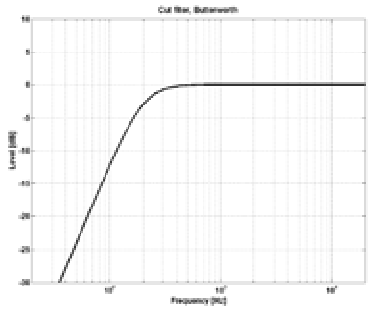
Notch Filter - Narrow Type



Cut Filter - Bessel type



Cut Filter - Butterworth type



Press **Freq** and use Faders 1 to 4 to adjust frequence for each of the four bands.

Range - Lo band : 20Hz to 20kHz
Range - Mid1 band : 20Hz to 20kHz
Range - Mid2 band : 20Hz to 20kHz
Range - Hi band : 20Hz to 40kHz

Gain

Press **Gain** and use Faders 1 - 4 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:
Lo Gain : -12dB to +12dB
Mid1 Gain : -12dB to +12dB
Mid2 Gain : -12dB to +12dB
Hi Gain : -12dB to +12dB

Range for the Notch filter:
Lo Gain : -100dB to 0dB
Mid1 Gain : -100dB to 0dB
Mid2 Gain : -100dB to 0dB
Hi Gain : -100dB to 0dB

Type

Press and use Faders 1-4 to set BW value for each of the 4 EQ bands.

Range for the Notch filter:
Lo BW : 0.02oct to 1oct
Mid1 BW : 0.02oct to 1oct
Mid2 BW : 0.02oct to 1oct
Hi BW : 0.02oct to 1oct

Range for the Parametric filter:
Lo BW : 0.1oct to 4oct
Mid1 BW : 0.1oct to 4oct
Mid2 BW : 0.1oct to 4oct
Hi BW : 0.1oct to 4oct

Range for the Shelve filter:
Lo BW : 3dB/oct to 12dB/oct
Hi BW : 3dB/oct to 12dB/oct

Range for the Cut filter:
Lo BW : Bessel or Butterworth
Hi BW : Bessel or Butterworth

Bandwidth/Q - Key-Values:

BW		**Q**
0.5	-	2.87
0.7	-	2.04
1.0	-	1.41

ITU-R BS.1770 Loudness Correction for TC DB4 and DB8

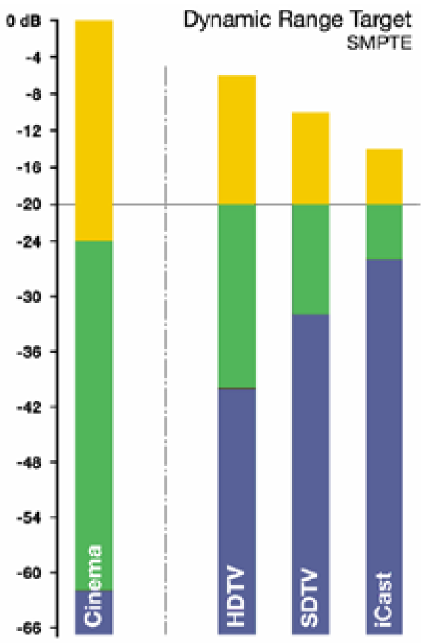
Introduction

Years of research and standardization work on loudness and true-peak level has enabled TC to design high resolution, low latency loudness measurement and control equipment such as this new Automatic Loudness Correction processor, ALC5.1.

In broadcast, digitization is driving the number of AV channels and platforms up, while the total number of viewers remains roughly the same. Using only a dialog-based level control principle has led to ambiguous level management, more “level jumps between programs”, and extra time spent on audio production and management in general. Non-dialog based level jumps are currently creating havoc in digital TV; and ALC5.1 helps correct that situation.

Fig 1

Target loudness for selected broadcast platforms based on a consumer’s Dynamic Range Tolerance, DRT.
When processing is centered around average loudness, the –20 dB line, transparent platform “trickle-down”, where the dynamic range can be restricted step by step, is automatically enabled.
Note how different the broadcast requirements are from those of Cinema. Several TC papers are available about the subject. Visit the Tech Library at the TC website for more details.



ALC5.1 is part of a universal approach to loudness control, starting at the production or live engineer with an easy-to-read loudness meter and universal delivery specifications. When downstream dynamic range is a known quantity it can be adjusted during the production or ingest phases, requiring less processing at later stages of a distribution chain. The chain ends with the capability of quality controlling previous stages by applying the same loudness measure for logging purposes: A closed loop based on the open standard ITU-R BS.1770.

The full leveling process needs not be put in place all at once. Production engineers may keep using VU, PPM or Dorrough meters with which they are comfortable, as long as the average loudness normalization process and platform ranging is known, and can be taken into account.

Welcome to a new world of leveling, where distorted and overly loud audio is unacceptable. where program content with different dynamic range may be broadcast back to back, without abrupt level changes.

Automatic Loudness Correction for Stereo and 5.1

ALC5.1 offers processing complementary to ITU-R BS.1770, EBU R128 and ATSC A/85 based normalization for use in broadcast ingest, linking and transmission. ALC5.1 may fully or partly correct level jumps within broadcast programs and at transitions between them. The resolution of ALC5.1 is sufficiently high that more than one hundred processors may be cascaded without degradation of sound quality.

ALC5.1 can be used to control level and improve sound, not only in Dolby® AC3 based transmission and linking, but also on other broadcast platforms, such as analog TV, mobile TV and IPTV. The Engine uses the new ITU-R BS.1770 standard, which measures speech, music and effects equally well, and can deal with mono, stereo and 5.1 signals.

ALC5.1 makes life with Dolby AC3 easier for the broadcaster by 1) limiting the amount of work which has to be put into generating metadata, 2) making the end-listener experience more predictable, 3) reducing the amount of level jumps between programming, and 4) improving the overall DTV sound quality.

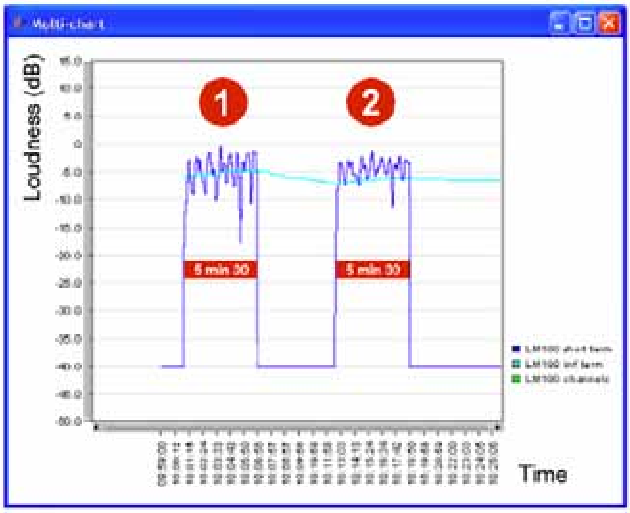


Fig 2
The example shows transition jumps between programs
1) without ALC5.1 and
2) including ALC5.1 in the signal path.

In the illustration, 11 broadcast programs were put back over a period of 5 minutes and measured with Dolby LM100.

The goal in multi-platform broadcast should be to use the same loudness measure for

- Production
- Ingest
- Linking
- Master Control Processing
- Logging

- thereby ensuring better audio quality not only in DTV audio, but across all broadcast platforms. ALC5.1 is ideally used with ITU-R BS.1770 based loudness meters, such as TC Electronic LM6, but can also smoothen out level jumps when normalization is based on Dorrough, PPM, VU or Dolby's LM100 meter. ALC5.1 greatly increases the usability of LM100 because it compensates for its blind angle: Non-dialog material at unexpected mix-levels.

Features

Low latency (1ms), high resolution loudness processor for mono, stereo and 5.1 signals.
Loudness control adhering to ITU-R BS.1770, EBU R128 and ATSC A/85
True-peak limiting adhering to ITU-R BS.1770, EBU R128 and ATSC A/85

Presets

ALC5.1 presets are found in the “Loudness, Multichannel” Engine Factory Bank.

ALC5.1 presets with “Limit” in the title, perform only negative loudness and peak level correction. These presets cannot add gain.

ALC5.1 presets with “Correction” in the title, may perform both positive and negative gain correction depending on the loudness of the signal.

ALC5.1 Basic Use

Two ALC5.1 processors may be loaded in DB4 (additional I/O may be required for two 5.1 streams), while DB8 accommodates two ALC5.1 processors plus room for 2 Stereo ALCs with an additional I/O card. If the same audio route is used at the station for changing format between mono, stereo and 5.1, it may be of advantage to use ALC5.1 universally rather than switching between different processor types.

The basic latency of ALC5.1 AES/EBU I/O is 1 ms, and processing is performed at 48 bit resolution. ALC5.1 is primarily designed for use in broadcast Ingest, Linking and Transmission.

To control ALC5.1 from a PC or a Mac, TC Icon is used. Screen shots from TC Icon is shown on the next pages.

Main Page

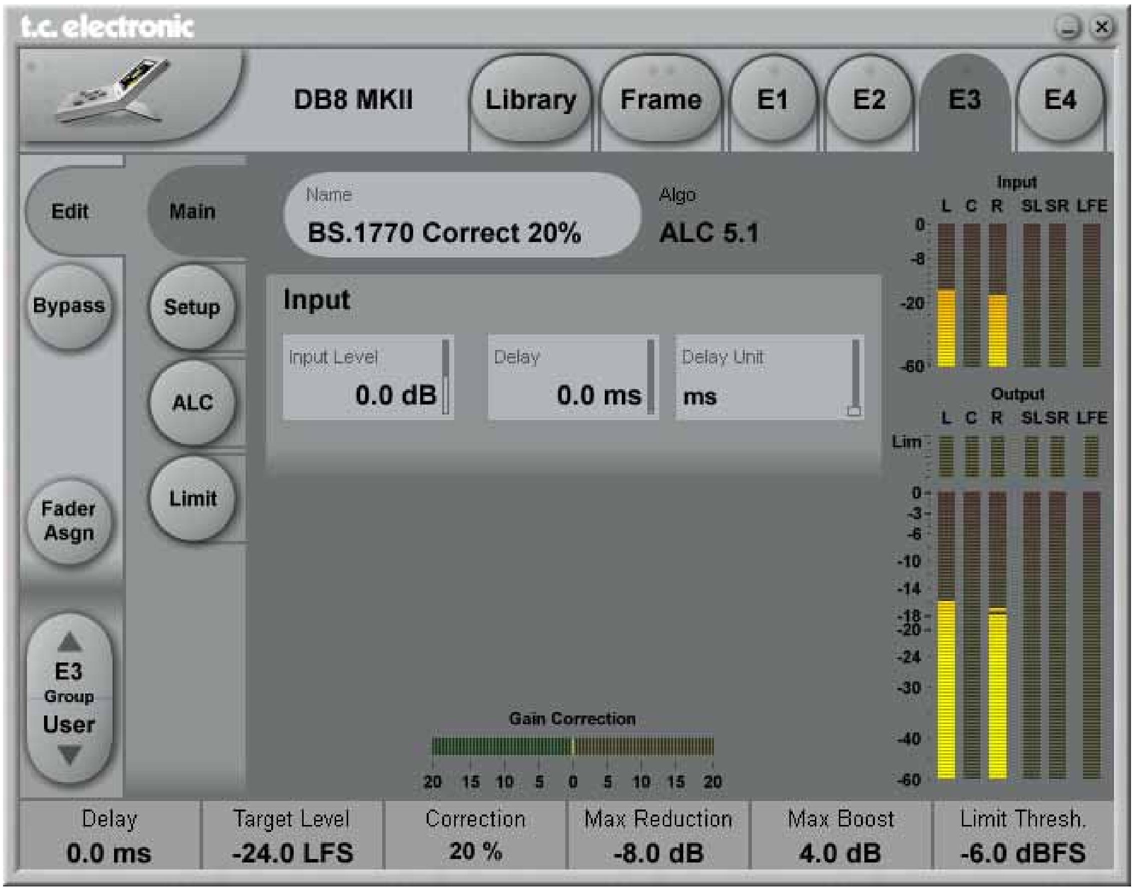


Fig 3
TC Icon view of ALC5.1 Main page parameters.
Be sure to use Icon version 3.82 or higher when controlling ALC5.1

Preset Title
The Main page of any algorithm in DB4 and DB8 displays the title of the current preset. Click on the Name field to edit a preset title, and Store the changes if you wish to keep them.

Input Level
Input gain applied to all 5.1 channels before loudness detection or processing is applied. The range of the Input Level parameter is -18 to +18 dB. Because DB4 and DB8 use 48 bit processing, a positive Input gain does not create overload, even if the input signal is already at full scale.

Delay
Time alignment of all 5.1 channels at 24 bit resolution. The delay function makes use of silent update technology so adjustments may be performed live on air. Minimum latency through ALC5.1 is 1 ms. Additional delay of up to 1 sec may be added using this parameter.

Delay Unit
Sets the unit used to display delay time, frames or milliseconds (30fr, 25fr, 24fr, ms).

Setup Page

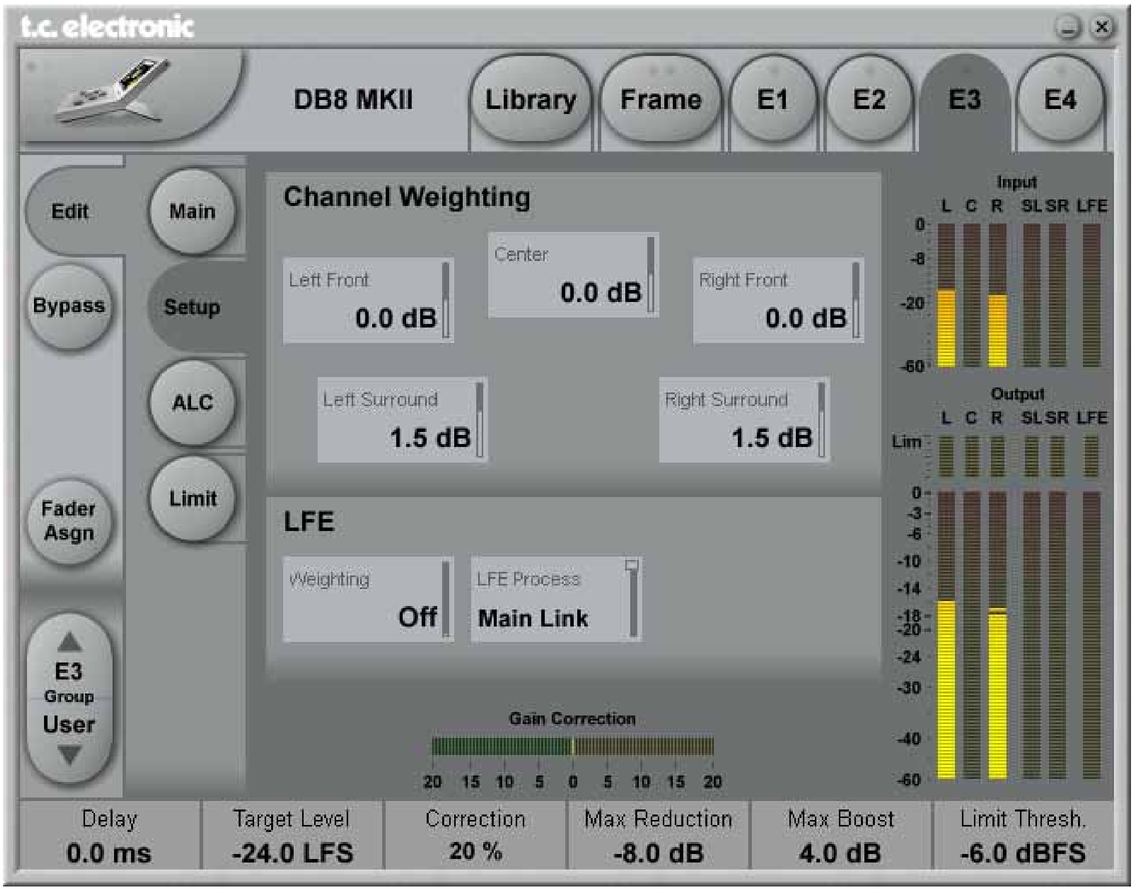


Fig 4

The Setup page of ALC5.1 showing settings according to ITU-R BS.1770.

Channel Weighting

Sets the weighting of each Main channel to the loudness measurement. BS.1770 specifies the front channels to be set at 0.0 dB, and the surrounds at +1.5 dB. However, it’s possible that more ideal compromises may be found. To have the combined result stay the same, all channels should sum at +3.0 dB. (For example, all channels except for Center at 0.0 dB, and Center at +3.0 dB. Or L/R at 0.0 dB, and all others at +1.0 dB).

LFE Weighting

Determines whether the LFE channel should contribute to the loudness measurement or not. According to original BS.1770, the LFE should not contribute. However, the debate is on, and the recommendation might change. If you find that commercials start using unexpectedly high LFE level, you may wish to bring LFE into the equation. The ALC5.1 algorithm enables you to keep flexible on this issue..

LFE Process

Determines if LFE gain follows the Main channels or not.

ALC Page. Automatic Loudness Correction

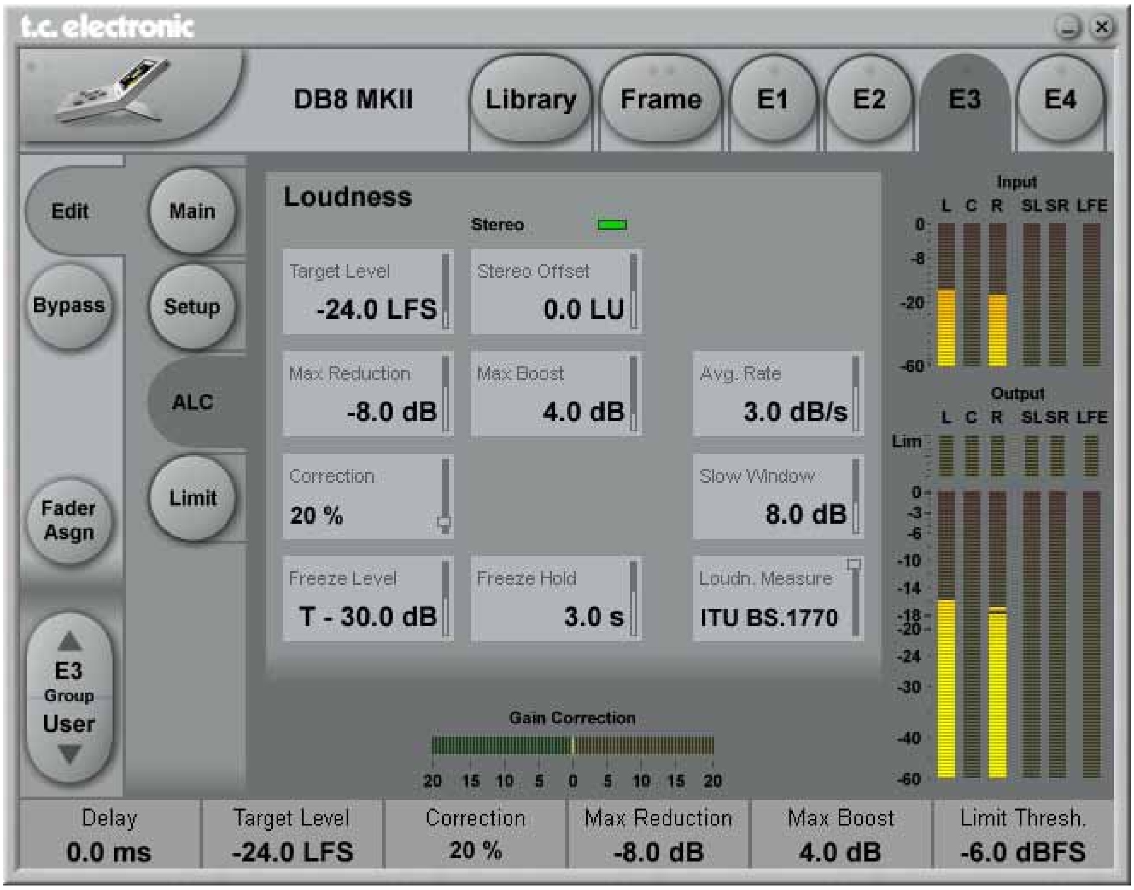


Fig 5

The Automatic Loudness Correct page set for using the BS.1770 loudness measure.
Settings shown are suitable for a static Dialnorm value of between -24 and –26 in AC3 transmission.
For SDTV and Mobile TV feeds, a higher Target Level should normally be chosen.

Target Level

Sets the Loudness Target, aimed for by ALC5.1. The unit is “LFS”. When “ITU-R BS.1770” is selected as loudness measure, LFS denotes “LKFS”, which also is the same as “LUFS”. See Fig 7, parameter no 1. For normal broadcast, the value should typically be between -18 and -24 LFS. Note that the distance between this value and Limit Threshold is a quality defining factor. If the difference is too small, wide dynamic range material may be hampered. See Limit Threshold details in the next section and *Fig 6, 9, 10 and 11.*

In broadcast environments working against a fixed dialnorm value, Target Level should typically be set 2-4 dB higher than the permanent Dialnorm value. This will ensure the best listening result if a consumer engages reproduction processing.

Stereo Offset

While the BS.1770 measurement works for stereo as well as for 5.1 signals, a different Target Level may be better in some distribution scenarios: When end-listener down-mix is relied on, having the same Target Level for stereo and for 5.1 can create systematic level-jumps at consumers listening to stereo. Therefore, ALC5.1 includes a novel automatic discrimination function, allowing for slightly different Target Levels to apply when the signal is strereo compared to when it’s 5.1.

The Stereo Offset parameter allows a smooth and automatic Target Level change when the input is stereo. For instance, if Target Level is set to -21 LFS and Stereo offset is set to -3 LU, ALC5.1 uses a Target Level of -21 LFS for 5.1 programs, but a Target Level of -24 LFS for stereo.

Max Reduction

Sets the maximum number of dBs the processor is allowed to attenuate the signal. If this parameter is set to 0.0 dB, level reduction is disabled regardless of other settings such as Correction.

Max Boost

Sets the maximum number of dBs the processor is allowed to boost the signal. If this parameter is set to 0.0 dB, level boost is disabled regardless of other settings such as Correction.

Correction

Sets how much correction is applied when the actual loudness is different from the Target Level. For instance, if Correction is set at 40%, and loudness is 6 dB away from the Target Level, the processor will apply a correction of 2.4 dB. Be careful when setting this parameter, as it may take a little "time testing" to arrive at the best value, especially if you wish to cover within program level jumps and inter-program level jumps using one preset. See Fig 6.

Freeze Level

Sets the level below which a Gain Boost is gradually revoked. Use Freeze to avoid boosting signals meant to remain below the noise floor of a certain broadcast platform. Freeze relates to Target Level. For instance, if Target Level is set at –21 LFS, and Freeze Level is set at –15 dB, positive gain (if enabled) will be gradually nulled when level falls below –36 LFS. See Fig 7, parameter no 3.

Freeze Hold

Sets the time in seconds before the processor resets to 0dB gain change, when the level falls below Freeze Level. See Fig 7, parameter no 4.

Fig 6

The Correction parameter.
With a setting of 30%, program which is 10 dB off target will be corrected by 3 dB.

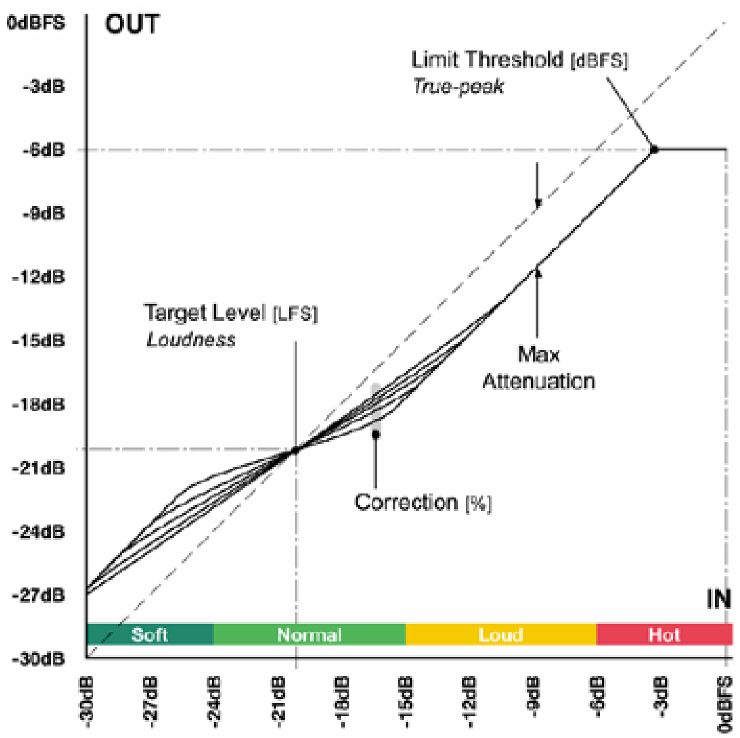
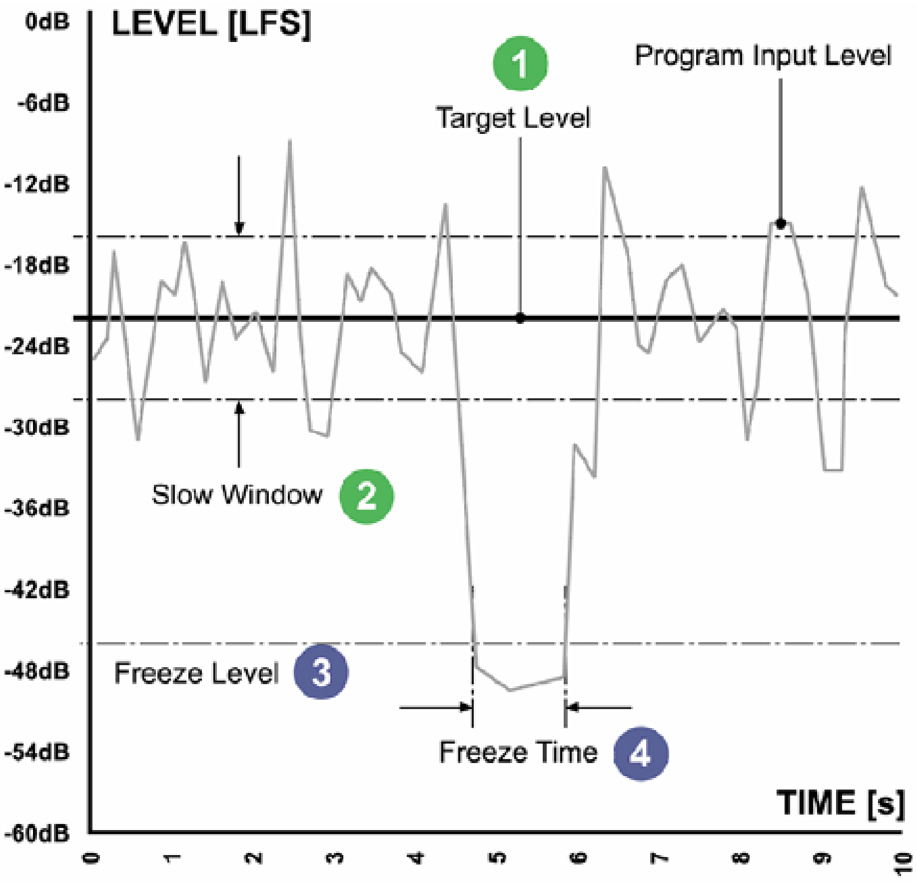


Fig 7
Slow Window and Freeze parameters.

Gain corrections happen more slowly when program level is already within the Slow Window.

The loudness has to drop below the Freeze Level for the duration of the Freeze Hold setting before unity gain is gradually reinstated.

In the illustration, parameters are set like this:
Target Level = -22 LFS = -22 LKFS = -22 LUFS
Slow Window = 12 dB
Freeze Level = -24 dB (relative to Target Level)



Average Rate
Sets the speed by which gain changes as a result of loudness variations. The rate adapts to the signal, and takes the Slow Window into account, so this parameter shows an average number.

Note how a fast Average Rate is more asymmetrical than a slow rate: The DB becomes faster at turning down than turning up because listeners typically object more to obtrusively loud sounds (promos, commercials) than to audio becoming soft.

Slow Window
Sets a window around the Target Level inside which gain changes happen more slowly. Use this parameter in combination with Average Rate. See Fig 7, parameter no 2. (6dB = ±3dB from target)

Loudness Measure
Controls which loudness model is used for the measurement. Select between TC Grid and the ITU-R BS.1770 standard model.

Limit

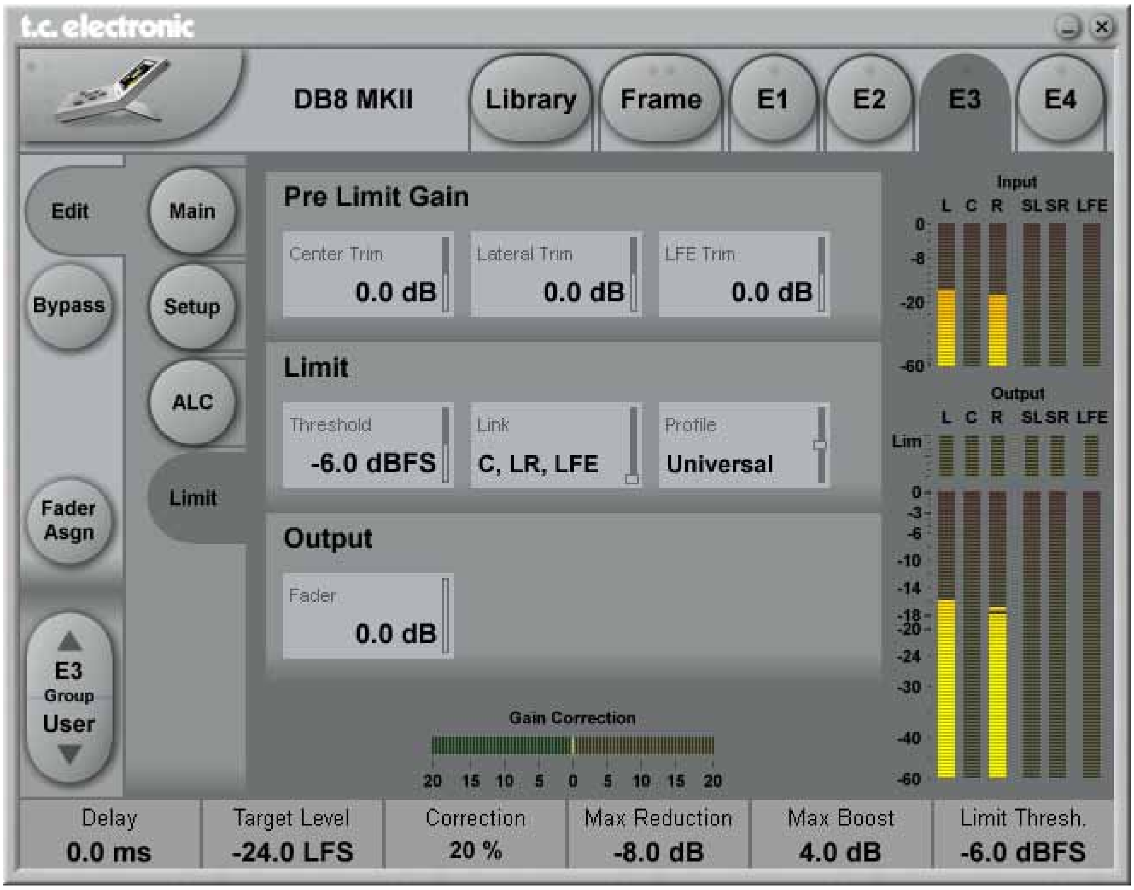


Fig 8

The Limit page. The Limiter in ALC5.1 uses true-peak detection as specified in BS.1770.
In the example, the Limit Threshold has been set at –10 dBFS. Note limit indication above output meters.

Center Trim

Static gain control for the Center channel after the ALC section, but before the output limiter. The range of the Trim parameter is -18 to +18 dB. Because DB4 and DB8 use 48 bit processing, a positive setting does not create overload, even if the signal is already at full scale.

Lateral Trim

Static gain control for all the Main channels, except for Center, after the ALC section, but before the output limiter. The range of the Trim parameter is -18 to +18 dB. Because DB4 and DB8 use 48 bit processing, a positive setting does not create overload, even if the signal is already at full scale.

LFE Trim

Static gain control for the LFE channel after the ALC section, but before the output limiter. The range of the Trim parameter is -18 to +18 dB. Because DB4 and DB8 use 48 bit processing, a positive setting does not create overload, even if the signal is already at full scale.

NOTE: Trim parameters are the perfect place to change the Target Level for broadcast platforms that require a higher average than in the -24 to -22 LFS range.

Limit Threshold

Sets the Limit Threshold for all limiters. The limiters in ALC5.1 use true-peak detection as per ITU-R BS.1770. True-peak detection makes overload of downstream devices, such as data reduction codecs, sample rate converters and DA converters, less likely.

Though digital samples may go to full scale, it is recommended to always use a conservative Limit Threshold, even in digital transmission. Reserve the top of the digital scale for occasional peaks in wide dynamic range material (feature films, wide dynamic range music), so don't go above -6 dBFS in HDTV for normal broadcast programming. This way, down-mixing or bass management at the consumer will also not generate unexpected distortion. See Fig 6, 9, 10 and 11.

The distance between the Target Level of the ALC section and the Limit Threshold is an important audio quality defining factor. Though you may be typically working with a distance of 10 dB in analog TV, consider widening this to maybe 14-16 dB in DTV, see Fig 1. Widening can be accomplished by moving down the Target Level and/or raising the Limit Threshold. For instance, a Target Level of -20 LFS or -22 LFS with a Limit Threshold of -6 dBFS would widen the dynamic range of DTV, while a Limit Threshold of -9 or -10 dBFS could be kept on the analog feed.

Limiter Link

The Limit Link settings define which limiters work together.
ALL: If a threshold is exceeded in any channel, all channels are limited.

LCR, LFE: If a threshold is exceeded in one of the Main channels, all Main channels are limited. If the threshold is exceeded on the LFE channel, LFE is limited independently.

C, LR, LFE: If the threshold is exceeded in the Center channel, only that channel is limited. If the threshold is exceeded in one of the other Main channels, all Main channels excluding Center are limited. If the threshold is exceeded on the LFE channel, only that channel is limited.

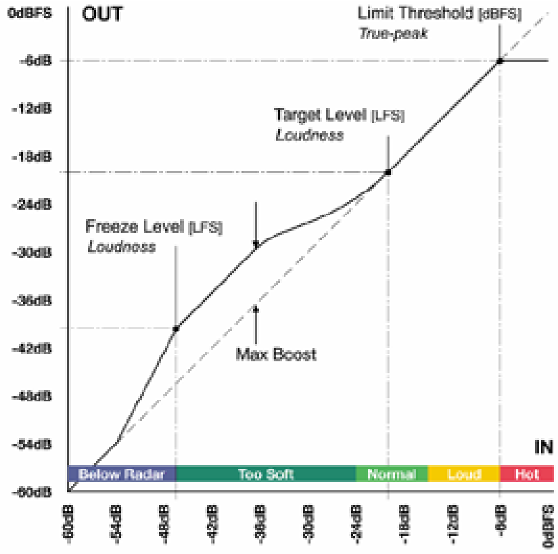


Fig 11

Loudness control allowing Boost.
In this illustration,
Target Level = -20 LFS,
Limit Threshold = -6 dBFS,
Max Boost = 6 dB
Freeze Level = -46 LFS (Target -26 dB)

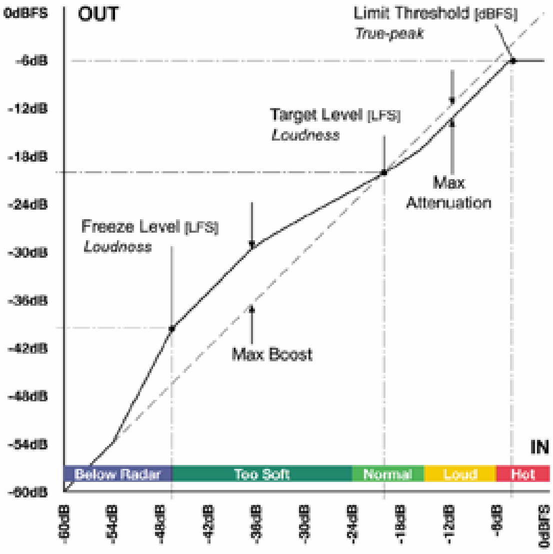


Fig 10

Loudness control allowing both
Boost and Attenuation.
In this illustration,
Target Level = -20 LFS,
Limit Threshold = -6 dBFS,
Max Boost = 6 dB
Max Attenuation = 2 dB
Freeze Level = -46 LFS (Target -26 dB)

LM6 represents a quantum leap away from simply measuring audio level to measuring perceived loudness. The old level method is responsible for unacceptable level jumps in television, for music CDs getting increasingly distorted, and for different audio formats and program genres becoming incompatible: Pristine music tracks from the past don't co-exist with new recordings, TV commercials don't fit drama, classical music or film and broadcast doesn't match. The most fundamental audio issue of all – control of loudness – every day makes millions of people adjust the volume control over and over again.
LM6 is part of a universal and ITU standardized loudness control concept, whereby audio may easily and consistently be measured and controlled at various stages of production and distribution. LM6 works coherently together with other TC equipment, or with equipment of other brands adhering to the same global standard. Follow the guidelines given to allow audio produced for different purposes to be mixed, without low dynamic range material such as commercials or pop CD's always emerging the loudest.

- Loudness meter fully compliant with EBU R128
- Loudness meter fully compliant with ATSC A/85
- Loudness meter fully compliant with ITU-R BS.1770
- Loudness meter fully compliant with ITU-R BS.1770-2
- Radar meter showing Momentary and Short-term loudness
- True-peak bargraph meters
- Advanced Logging functionality

Introduction

Since 1998, TC has performed listening tests and evaluation of loudness models; and therefore holds an extensive, Universal Database of loudness, based on ten thousands of assessments. The database covers all sorts of broadcast material, music, commercials, feature film and experimental sounds, and is verfied against other independent studies.

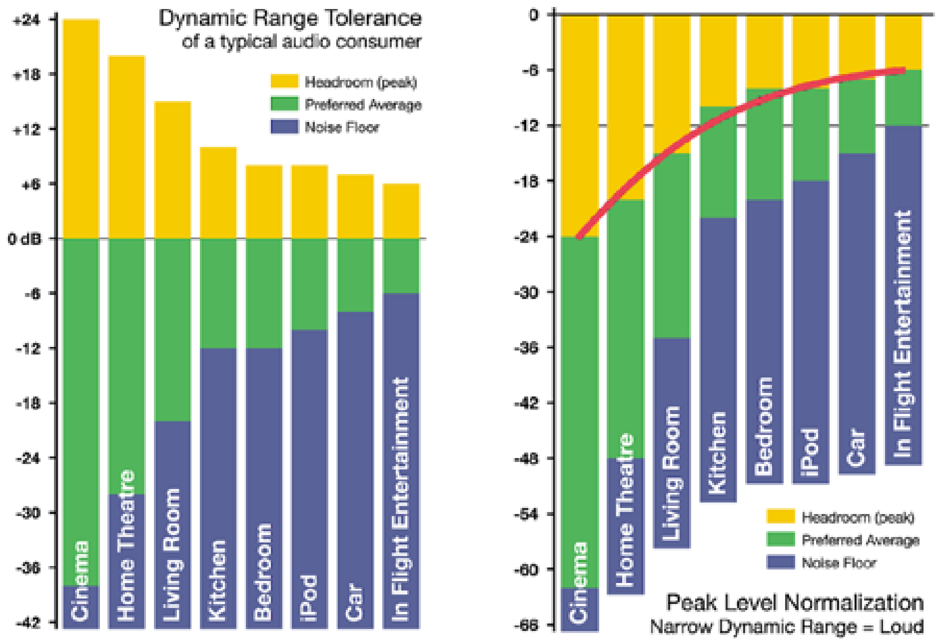


Fig 1.
Left: DRT for consumers under different listening situations
Right: Peak level normalization means that material targeted low dynamic range platforms gets loud.

The Universal Database is authoritative from an academic as well as a practical point of view. It has been indispensable when designing the LM6 meter, because it provided the missing link between short-term and long-term loudness, and enabled the statistically founded Universal Descriptors of LM6 .

The chart of Dynamic Range Tolerance in Fig 1 is a side-effect of the studies mentioned: Consumers were found to have a distinct Dynamic Range Tolerance (DRT) specific to their listening environment. The DRT is defined as a Preferred Average window with a certain peak level Headroom above it. The average sound pressure level, which obviously is different from one listening condition to another, has to be kept within certain boundaries in order to maintain speech intelligibility, and to avoid music or effects from getting annoyingly loud or soft.

Audio engineers instinctively target a certain DRT profile when mixing, but because level normalization in broadcast and music production is based on peak level measures, low dynamic range signatures end up the loudest as shown by the red line in Fig 1, right. Audio production is therefore trapped in a downwards spiral, going for ever decreasing dynamic range. By now, the pop music industry is “right of” In Flight Entertainment in the illustration.

LM6 offers a standardized option: The visualization of loudness history and DRT in combination with long-term descriptors from production onwards, is a transparent and well sounding alternative to our current peak level obsession. Not only for music, but also in production for broadcast or film. The engineer, who may not be an audio expert, should be able to identify and consciously work with loudness developments within the limits of a target distribution platform, and with predictable results when the program is transcoded to another platform.

LM6 therefore color codes loudness so it’s easy to identify target level (green), below the noisefloor level (blue), or loud events (yellow), see Fig 2.

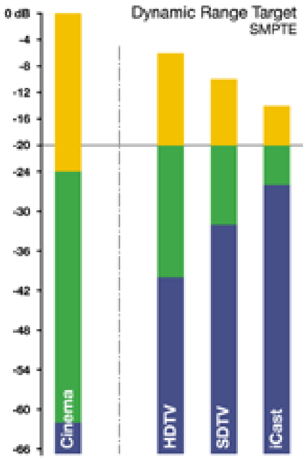


Fig 2
Color coding and target loudness for selected broadcast platforms based on a consumer's Dynamic Range Tolerance, DRT. The aim is to center dynamic range restriction around average loudness, in this case the –20 dB line, thereby automatically avoiding to wash out differences between foreground and background elements of a mix.

Note how different the broadcast requirements are from those of Cinema.

When production engineers realize the boundaries they should generally stay within, less dynamics processing is automatically needed during distribution, and the requirement for maintaining time-consuming metadata at a broadcast station is minimized.

In broadcast, the goal is to use the same loudness measure for

- Production,
- Ingest,
- Linking
- Master Control Processing
- Logging

thereby ensuring better audio quality not only in DTV audio, but across all broadcast platforms. LM6 and TC processing can co-exist with PPM meters, VU meters or Dolby's LM100 meter. LM6 greatly increases the usability of LM100 in production environments because it provides running status, and gives a standardized and intuitive indication of both dialog and non-dialog program.

Basic Use

LM6 makes use of a unique way of visualizing short-term loudness, loudness history, and long-term statistical descriptors. It may be used with mono, stereo and 5.1 material for any type of program material.
Press the Radar key to bring up the Radar page. This page will be used most of the time. The basic functionality of the Radar page is shown in Fig 3.

Fig 3 - Radar page features of LM6 in DB4 and DB8. Target Loudness is displayed at 12 o'clock of the outer ring, and at the bold circle of the radar indicated also by the transition from green to yellow. The descriptors Loudness Range and Program Loudness, are the yellow numbers in the lower part of the display. Press the Reset key to reset Radar and Descriptors.

The "Transport Controls", Pause and Reset, are used to make the radar and descriptor measurements run, pause and reset. Press the "Main" key to change preset name and for adjusting more parameters. Press the "Setup" key to change setup parameters. Presets can be stored specifying target loudness, noise floor, overload conditions etc using normal DB4 and DB8 preset handling procedures.

Radar Page

Current Loudness: Outer Ring

The outer ring of the Radar page displays Momentary loudness. The 0 LU point (i.e. Target Loudness) is at 12 o'clock, and marked by the border between green and yellow, while the Low Level point is marked by the border between green and blue. The "0 LU Equals" and "Low Level Below" parameters are found on the Setup page. For instance, if 0 LU is set at -22 LUFS, and Low Level is set at -20 LU, the color coding of Fig 3 applies.
The user should be instructed to keep the outer ring in the green area, and around 12 o'clock on the average. Excursions into the blue or the yellow area should be balanced, and not only go in one direction.

The numbers associated with the outer ring may be referenced at either maximum loudness, or have a zero point set set at Target Level. Choose "LUFS" or "LU" at the Loudness Scale selection on the Main page depending on your preference. Either way of looking at loudness is valid. LUFS reading is in line with how peak level is typically measured in a digital system, and compatible with Dolby AC3 and E metadata, while the LU approach calls for a certain Target Loudness to have been predetermined, like e.g. a VU meter.

Long-term measurements

Universal descriptors may be used to make program-duration measurements, or you may “spot-check” regular dialog or individual scenes as required. It is recommended not to measure programs of a shorter duration than appoximately 10 seconds, while the maximum duration may be 24 hours or longer.

Reset Key

Before a new measurement, press the Reset key. This resets the descriptors, the radar and the true-peak meters. Run the audio, and watch the radar and descriptor fields update accordingly. It is normal that the descriptors wait five seconds into the program before showing the first readings, while the radar updates instantly. The first five seconds of a program are included in the descriptor calculations, even though they are not shown instantly.

LM6 incorporates an intelligent gate, which discriminates between foreground and background material of a program. Consequently, a measure doesn’t start before audio has been identfied. It also pauses the measurement during periods of only background noise, and in the fade-out of a music track.

Universal Descriptors and Dolby LM100

Unlike methods that measure dialog only, LM6 may be used with any type of audio – which includes dialog, of course. If you wish to measure dialog, it’s recommended to do a manual spot check of a program or a film. Find 10-30 secs of regular dialog and measure it with LM6 . Where dialog may be soft, regular or loud, and shift by more than 15 dB inside a film, regular dialog tends to be less ambiguous and more consistent across a program.

NOTE

For compatibility with a proprietary measure such as Dolby LM100, only some of these meters are updated to use ITU-R BS.1770 and Leq(K) while others are locked at Leq(A). The software version of LM100 should be 1.3.1.5 or higher in order for it to comply with BS.1770, and to have its average loudness reading be compatible with Center of Gravity in LM5 or Program Loudness in LM6. Even used just on speech, Leq(A) is not a precise approximation to perceived loudness, so please update the unit to BS.1770 to obtain similar readings and predictable results.

To measure dialog with LM6 the same way Dolby LM100 is sometimes used, solo the Center channel during a spot check to momentarily disable the channel weighting specified in BS.1770, if you’re working on a 5.1 stem.

Universal Descriptors and AC3 Metadata

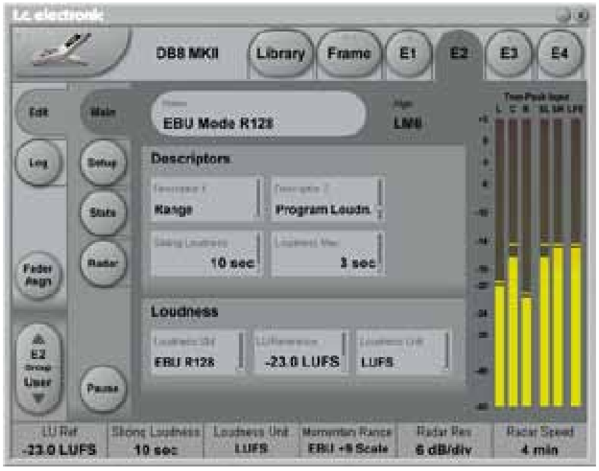
The “Dialnorm” parameter in AC3 metadata should indicate the average loudness of a program. Basic dynamic range and level control that rely on this parameter may take place in the consumer’s receiver. Therefore, its value should not be far off target, or the consumer results become highly unpredictable.

Program Loudness in LM6 is directly compatible with Dialnorm in AC3. Most broadcast stations work with a fixed dialnorm setting, for instance –23 LUFS. This would be the Program Loudness target level for any program.
If your station is more music than speech, better inter- channel leveling may be obtained with dialnorm permanently set 1 or 2 LU lower than the Program Loudness target level.

True-peak meters

The peak meters of LM6 display true-peak as specified in ITU-R BS.1770. True-peak meters give a better indication of headroom and risk of distortion in dowstream equipment such as sample rate converters, data reduction systems and consumer electronics than digital sample meters used e.g. in CD mastering. Note that the standard level meters in most digital workstations and mixers are only sample peak (Final Cut, Avid, ProTools, Yamaha etc.), and should only be used as a rough guideline of the headroom.
Note that the meter scale is extended above 0 dBFS. Most consumer equipment distorts if you see readings above 0. It’s not a problem to have true-peak level going to -1 dBFS in production, but legacy platforms (analog, NICAM etc.) and some data-reduction codecs may distort unless true-peak level is kept lower. With Dolby AC3 and with low bitrate codecs, -3 dBFS should be considered the limit, while legacy platforms requiring emphasis may need even further restriction. Like described in EBU R128, it’s recommended to make full use of the headroom with true-peaks going to -1 dBFS in production, and to only restrict peak level further during distribution/transmission.

Main page



Descriptor 1-2

Loudn. Range

Loudness Range, standardized in EBU R128 and abbreviated “LRA”, displays the loudness range of a program, a film or a music track. The unit is LU, which can be thought of as “dB on the average”.

The Loudness Range descriptor quantifies the variation of the loudness measurement of a program. It is based on the statistical distribution of loudness within a program, thereby excluding the extremes. Thus, for example, a single gunshot is not able to bias the LRA number.

EBU R128 does not specify a maximum permitted LRA. R128 does, however, strongly encourage the use of LRA to determine if dynamic treatment of an audio signal is needed and to match the signal with the requirements of a particular transmission channel or platform.

Consequently, if a program has LRA measured at 10 LU, you would need to move the master fader +- 5 dB to make loudness stay generally the same over the duration of the program. (Not that you would want that).

In production, Loudness Range may serve as a guide to how well balancing has been performed, and if too much or too little compression has been applied. If a journalist or video editor isn’t capable of arriving at a suitable LRA, he could be instructed to call an audio expert for help.

This may be regarded as initial production guidelines:
HDTV and digital radio: Stay below LRA of 20 LU.
SDTV: Stay below LRA of 12 LU.
Mobile TV and car radio: Stay below LRA of 8 LU.

Remember to use LRA the other way around too: If there is an ideal for a certain genre, check its LRA measure, and don’t try go below it. LRA should not be used for Limbo. Allow programs or music tracks the loudness range they need, but not more than they need.

Loudness Range may also be measured on a broadcast server to predict if a program is suitable for broadcast without further processing. LRA is even a fingerprint of a program and stays the same downstream of production if no dynamics processing has been applied. You may even check the number out of a consumer’s set-top box to verify that distribution processing and Dolby DRC has been disabled.

Like with Program Loudness and Loudness Max, the meter should be reset before measuring LRA.

Prog. Loudn.

Program Loudness returns one loudness number for an entire program, film or music track. Its unit is LUFS. Some vendors and countries use the unit “LKFS” or “LUFS”, but all three are the same: An absolute measure of loudness in the digital domain, where the region around “0” is overly loud and not relevant for measuring anything but test signals. Expect readings of broadcast programs in the range between -28 and -20 LUFS.

Program Loudness is used as a production guideline, for transparent normalizing of programs and commercials, and to set loudness metadata in delivery if so required. For delivery or transmission of AC3 format, the metadata parameter “dialnorm” should reflect Program Loudness. The easiest way to handle multiple broadcast platforms is to normalize programs at the station to a certain value, thereby being able to take advantage of the normalization benefits across platforms, at the same time enabling static metadata.

Loudness measurements in LM6 are all rooted in ITU-R BS.1770. However, subtle differences exist between different regions of the world. Therefore LM6 also includes the “Loudness Standard” parameter. Be sure to set this parameter correctly for compliance in your region.

The Program Loudness target is more or less the same for broadcasters around the world, especially when taking the measurement differences into account. Target numbers range between -24 and -22 LUFS.

Like with Loudness Range and Loudness Max, the meter should be reset before measuring Program Loudness.

Sliding Loudn.

Sliding Loudness, unlike Program Loudness, Loudness Range and Loudness Max, is a continuously updated measure that doesn’t need to be reset. This type of descriptor is especially useful when “mixing by numbers”, i.e. when there is no access to the extremely informative radar display. When mixing by numbers, having Program Loudness as one descriptor and Sliding Loudness as the other displays simultaneous information about the full program side by side with the most recent loudness history.

Note 1: Because the Sliding Loudness measurement is completely un-gated, it may also be used to spot check sections of a program complying to “raw” ITU-R BS.1770 and the first revision of ATSC A/85.

Note 2: LM6 makes use of optimized statistics processing in order to display a sliding loudness value (a prognosis) as quickly as possible after a reset.

Loudness Max
Loudness Max displays the maximum loudness registered since the meter was last reset. Loudness Max is an especially useful parameter when checking and normalizing short duration programs such as promos and commercials. BCAP rules from the UK is an example of using Loudness Max as an efficient instrument to reduce listener complaints regarding loud commercials. While Program Loudness is adequate to normalize a consistent mix, Loudness Max may be used as a second line of transparent defense against overly short and loud event.

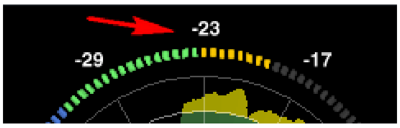
Target

Range: -36 LUFS to -6 LUFS
The parameter specifies the loudness level to generally aim at. It affects a number of functions and displays in LM6, and must be set according to the standard you need to comply with. Current broadcast standards require Target to be in the range between -26 and -20 LUFS. For instance, EBU R128 calls for -23 LUFS while ATSC A/85 specifies -24 LUFS.

The Target parameter affects these LM6 functions and displays:

1. Target sets the reference point for loudness measurements in LU. If the Loudness Unit parameter is set to LU, Program Loudness, Sliding Loudness and Loudness Max will be shown in LU relative to Target. On Target measurements will consequently read “0.0 LU”.

2. Target defines the “12 o’clock” value of the Radar meter.



Loudness Unit

LUFS
All measurements of program loudness and sliding loudness are shown in units of LUFS, that is, in Loudness Units on the absolute scale. This is the normal setting for the Loudness Unit parameter, that we recommend for most applications.
Loudness Range is always shown in units of LU, because it is basically a measurement of ‘range’ or of the distance between a high and a low loudness level.

LUFS/LU
This setting is similar to the ‘LUFS’ setting, except that the Radar display uses an LU scale rather than an LUFS scale, on the Icon. There is no difference between the LUFS and LU/LUFS settings, when the LM6 is used in stand-alone mode.

LU
In this setting, measurements of program loudness and sliding loudness are shown in units of LU, that is, in Loudness Units on a relative scale. The 0 LU is by definition the target loudness level, such as -23.0 LUFS. So by selecting ‘LU’, one can immediately see if a loudness level is above the target level (e.g. +1.2 LU) or below (e.g. -3.4 LU).

Loudness Std.

Options: BS.1770-2, Leq(K) or Cnt of Grav.
The Program Loudness measure is always rooted in the ITU-R BS.1770 loudness model. This parameter sets measurement gating. Note that the parameter only influences Program Loudness, and not Sliding Loudness or Loudness Max.

BS.1770-2
This setting reflects the latest revision of ITU-R BS.1770. Relative gate at -10 LU, safety gate at -70 LUFS.

Leq(K)
This setting reflects the original version of ITU-R BS.1770. No measurement gate besides from at safety gate at -70 LUFS, so the user doesn’t need to precisely start and stop a measurement in order to avoid bias from complete silence.

Cnt of Grav.
The standard setting from early versions of TC radar meters.
Relative gate at -20 LU, safety gate at -70 LUFS.

International Standards

Note how the three Loudness Standard settings generally return the same Program Loudness result for Narrow Loudness Range (“NLR”) programs, such as commercials and pop music, but can differ significantly with Wide Loudness Range (“WLR”) programs such as film, drama, acoustical music etc.

For an update on international standards, check for new versions of this manual, or download the Loudness Glossary available at www.tcelectronic.com/loudness

This is the situation as of August, 2011:
Japan, Canada, Brazil, China, Europe and most other countries specify the use of BS.1770-2 to make Program Loudness perform well across genres. BS.1770-2 enables the meter reliably to focus on foreground sound, and to transparently control loud commercials. ARIB (Japan) specifies BS.1770-2 in TR-B32. EBU (Europe) specifies BS.1770-2 in EBU R128 and in associated Tech Doc 3341. Target Level in these countries is -23 LUFS or -24 LUFS, measurement gating at -10 LU.

United States
Page 11 of ATSC A/85 (May 25, 2011) references ITU-R BS.1770-1, even though BS.1770-2 was in effect at that time. The same page also says that “All referenced documents are subject to revision”. The wording is ambiguous and it’s up to the reader to decide whether or not a relative gate (the difference between BS.1770-1 and BS.1770-2) is applied when measuring Program Loudness. The “Leq(K)” setting in LM2 disables the relative gate, while the setting “BS.1770-2” includes a relative gate at -10 LU. The BS.1770-2 setting is better across genres and for controlling loud commercials. Check in at www.atsc.org to see if the CALM act has forced ATSC to make up their mind.

Target Level in United States is -24 LUFS, measurement gating not clearly defined.

Measure Scale

This parameter can be set to either “Loudness Units, LU” or “Loudness Full Scale, LUFS”. Note that “LKFS” is the same as “LUFS”.

When “LUFS” is selected, the numbers in the outer ring of the Radar page apply. When “LU” is selected, numbers are shown around a “0” denoting LU Reference.

LU Reference

0 LU Equals sets the loudness required to obtain a 12 o’clock reading on the outer ring, which is the same as the border between green and yellow on the Radar page. 0 LU is the reference to aim at.

Peak Indicator

This parameter sets at which level the peak indicator will be invoked.

Setup

Momentary Range

EBU +9 or EBU +18
Set range on the radar-meter
EBU mode meters are able to display to show two different momentary displays: One with a narrow loudness range intended for normal broadcast and denoted “EBU +9”, and one with a wide loudness range intended for film, drama and wide range music denoted “EBU +18”.

The “EBU +9” setting gives a momentary meter range from -18 to +9 LU, while the “EBU +18” settings gives a momentary range from -36 to 18 LU.

Radar Speed

Radar Speed controls how long time each radar revolution takes. Select from 1 minute to 24 hours. You may “zoom” between the settings, as long as the history isn’t reset. Pressing the Reset key resets the meter and descriptor history.

Radar Resolution

Radar Resolution sets the difference in loudness between each concentric circle in the Radar between 3 and 12 dB. Choose low numbers when targeting a platform with a low dynamic range tolerance. You may “zoom” between the settings, as long as the history isn’t reset.

Low Level Below

Low Level Below determines where the shift between green and blue happens in the outer ring. It indicates to the engineer that level is now at risk of being below the noise floor.

Alert Indicator

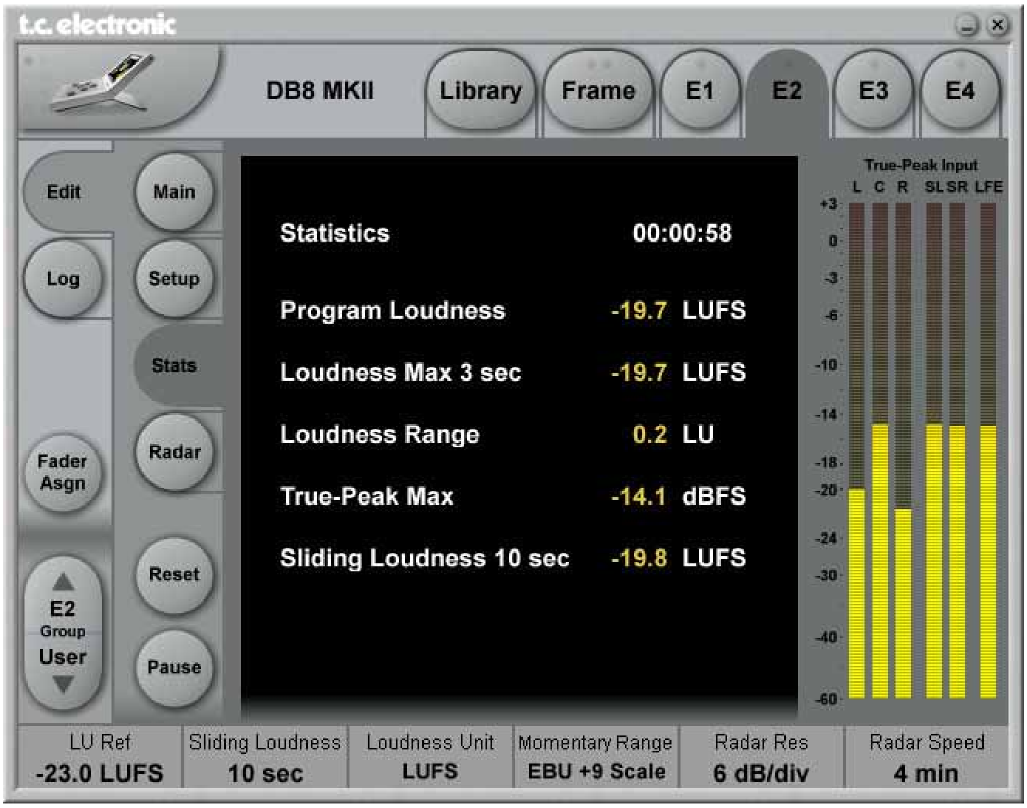
Stereo Integrity
The indicator indicates a lack of stereo integrity based on measuring the difference of left/right inputs. If there is a consistent difference between left and right over a prolonged time, the LED is lit.

5.1 Integrity
In this mode, Integrity is based on the signal levels on L,R,C,LS and RS channels. If one or more of the channels drop out over a prolonged time, the LED is lit.

Stereo or 5.1 Integrity
In this mode, Integrity is given when either Stereo or 5.1 Integrity are detected. This means that the LED is lit when neither valid Stereo nor 5.1 signals are detected.

Off
The Alert indicator is disabled.

Stats Page



The Stats page gives an overview of essential descriptors.

! *Note! The Reset button resets the meters and the log file.*

Level versus Loudness

When level normalization in audio distribution is based on a peak level measure, it favors low dynamic range signatures as shown in Fig 1. This is what has happened to CD.

Quasi-peak level meters have this effect. They tell little about loudness, and also require a headroom in order to stay clear of distortion. Using IEC 268-18 meters, the headroom needed is typically 8-9 dB.

Sample based meters are also widely used, but tell even less about loudness. Max sample detection is the general rule in digital mixers and DAWs. The side effect of using such a simplistic measure has become clear over the last decade, and CD music production stands as a monument over its deficiency. In numerous TC papers, it has been demonstrated how sample based peak meters require a headroom of at least 3 dB in order to prevent distortion and listener fatigue.

The only type of standard level instrument that does not display some sort of peak level is the VU meter. Though developed for another era, this kind of meter is arguably better at presenting an audio segment’s center of gravity. However, a VU meter is not perceptually optimized, or ideal for looking at audio with markedly different dynamic range signatures.

Unlike electrical level, loudness is subjective, and listeners weigh its most important factors - SPL, Frequency contents and Duration - differently. In search of an “objective” loudness measure, a certain Between Listener Variability (BLV) and Within Listener Variability (WLV) must be accepted, meaning that even loudness assessments by the same person are only consistent to some extent, and depends on the time of day, her mood etc. BLV adds further to the blur, when sex, culture, age etc. are introduced as variables.

Because of the variations, a generic loudness measure is only meaningful when it is based on large subjective reference tests and solid statistics. Together with McGill University in Montreal, TC Electronic has undertaken extensive loudness model investigation and evaluation.

The results denounce a couple of Leq measures, namely A and M weighted, as generic loudness measures. In fact, a quasi-peak meter showed better judgement of loudness than Leq(A) or Leq(M). Even used just for speech, Leq(A) is a poor pick, and it performs worse on music and effects. An appropriate choice for a low complexity, generic measurement algorithm, which works for listening levels used domestically, has been known as Leq(RLB). Combined loudness and peak level meters exist already, for instance the ones from Dorroughs, but BS.1770 now offers a standardized way of measuring these parameters. In 2006, ITU-R Working Party 6J drafted a new loudness and peak level measure, BS.1770, and the standard has subsequently come into effect. It has been debated if the loudness part is robust enough, because it will obviously get exploited where possible. However, with a variety of program material, Leq(RLB) has been verified in independent studies to be a relatively accurate measure, and correlate well with human test panels. It therefore seems justified to use Leq(RLB) as a baseline measure for loudness, especially because room for improvement is also built into the standard. The final BS.1770 standard included a multichannel annex with a revised weighting filter, R2LB – now known as “K” weighting - and a channel weighting scheme. These two later additions have been less verified than the basic Leq(RLB) frequency weighting.

The other aspect of BS.1770, the algorithm to measure true-peak, is built on solid ground. Inconsistent peak meter readings, unexpected overloads, distortion in data reduced delivery and conversion etc. has been extensively described, so in liaison with AES SC-02-01, an over-sampled true-peak level measure was included with BS.1770.

In conclusion, BS.1770 is an honorable attempt at specifying loudness and peak level separately, instead of the simplistic (sample peak) and mixed up measures (quasi-peak) in use today. The loudness and peak level measurement engine of LM6 follows the standard precisely. Possible updates to the ITU standard may be released as LM6 updates, provided that processing requirments doesn’t exhaust the system.

Technical papers from AES, SMPTE, NAB and DAFX conferences with more information about loudness measurement, evaluation of loudness models, true-peak detection, consequences of 0 dBFS+ signals etc., are available from the TC website. Visit the Tech Library at www.tcelectronic.com/techlibrary.asp for details.

Meter Calibration

Because of the frequency and channel weighting, and of the way channels sum, only specific tones and input channels should be used for calibration.

The most transparent results are obtained using a 1 kHz sine tone for calibration. Other frequencies or types of signal may be used (square wave, noise etc.), but don’t expect similar results. The beauty of the system lies in its RMS foundation, so this is a feature, not an error. The same feature enables the loudness measure to identify overly hot CDs or commercials, and to take out of phase signals into account just as much as signals that are in phase.

If we stick to standard methods for measuring peak audio level in a digital system, where a sine wave (asynchronous of the sample rate) with digital peaks at 0 dBFS, is regarded a 0 dBFS tone, BS.1770 and LM6 output these results:

One front channel fed with a –20 dBFS, 1 kHz sine tone => Reading of –23,0 LUFS.
Two front channels fed with a –20 dBFS, 1 kHz sine tone => Reading of –20,0 LUFS.
All 5.1 channels fed with a –20 dBFS, 1 kHz sine tone => Reading of –15,4 LUFS.

Display

LM6 may use either the measurement unit of LU (Loudness Units) or LUFS (Loudness Units Full Scale). LU and LUFS are measurements in dB, reflecting the estimated gain offset to arrive at a certain Reference Loudness (LU) or Maximum Loudness (LUFS) as defined in BS.1770. Since a common reference point for LU has not been agreed on at the time of writing, LUFS (or “LKFS”, pointing specifically to the Leq(R2LB) weighting of BS.1770), might be favored initially to avoid ambiguous use of the term LU.

The effectiveness of any loudness meter depends on both the graphical appearance and dynamic behavior of its display, as well as on its underlying measurement algorithms. A short-term loudness meter also relies on the measurement algorithm’s ability to output pertinent loudness information using different analysis windows, for instance, 200-800 ms for running realtime updates. It should be noted how the optimum size of this window varies from study to study, possibly because the objective of a running display hasn’t been fully agreed upon.

Formal evaluation of a visualization system is challenging: First of all, one or more metrics must be defined by which the display should be evaluated. The correspondence between the sound heard and the picture seen is one aspect to be evaluated. Another metric could characterize the speed of reading the meter reliably.
In TC Electronic LM2, LM5 and LM6, short-term, mid-term and long-term of loudness measurements are tied together coherently, and displayed in novel ways (angular reading and radar) that were preferred in its development and test phases. However, we remain open to suggestions for further improvement of the visualization of loudness.

Post Script

Control of loudness is the only audio issue that has made It to the political agenda. Political regulation is currently being put into effect in Europe to prevent hearing damage and disturbances from PA systems, and to avoid annoying level jumps during commercial breaks in television. In Australia, something similar may happen.

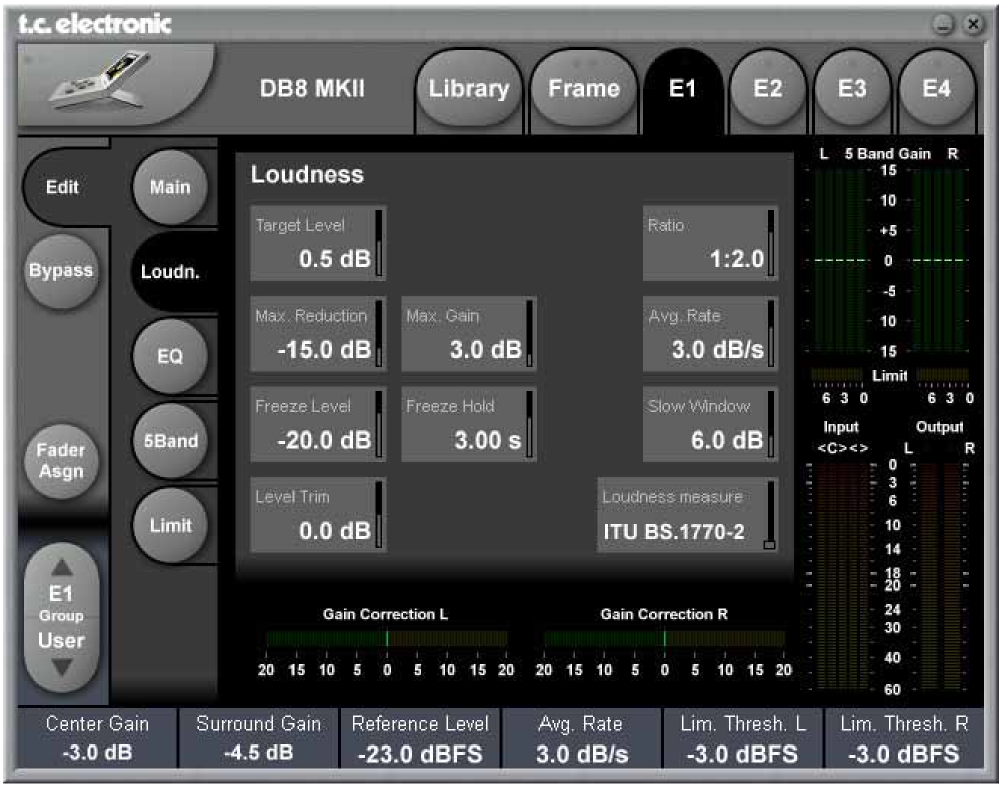
Many years of research into loudness of not only dialog, but also of loudness relating to any type of audio programming, has brought TC to the forefront of companies in the world to perform realtime loudness measurement and control. Therefore, TC has taken active part in loudness standardization efforts in Japan, the United States, Europe and other areas.

In broadcast, digitization is driving the number of AV channels and platforms up, while the total number of viewers remains roughly the same. On the sound production side, it is therefore important that delivery criteria can be easily specified and met, even by people not primarily concerned with audio: Journalists, musicians, video editors, marketing professionals etc.

Using only dialog based audio measurements in digital broadcast, has led to ambiguous level management, more level jumps between programs, and extra time spent on audio production and management in general. Non-dialog based level jumps are currently creating havoc in digital TV, and LM6 helps correct that situation. The LM6 Loudness Meter can be used to control level and improve sound, not only in Dolby AC3 based transmissions, but also on other broadcast platforms, such as analog TV, mobile TV and IPTV.

To summarize: LM6 is part of a holistic and universal approach to loudness control, starting at the production or live engineer. When she realizes the dynamic range at her disposal, less processing is needed at later stages of a distribution chain. The chain ends with the capability of quality controlling everything upstream by applying the same loudness measure for logging purposes: A closed loop.

Welcome to a new, standardized world of audio leveling. Across genres, across formats, across the globe.



DMix: Optimum Mobile Platform Delivery

In just one engine, DMix can downmix, loudness process and true-peak limit any mono, stereo or 5.1 source. Input formats are dealt with automatically without the need for metadata, downmix takes place at overload-proof 48 bit resolution, loudness processing complies with ATSC or EBU standards, and transparent transcoding keeps the output perfectly conditioned for mobile TV, iPod or IPTV. Even a wide loudness range feature film is transcoded automatically on the fly at an impeccable audio quality.

DMix presets for ATSC and EBU standards may be found in the new Down-conversion factory preset bank. Be sure to try the new iX presets featuring image enhancement for an extra enveloping experience when listening in headphones

BS.1770-2 Based Processing

New EBU R128 and ATSC A/85 compliant processing algos and presets for mono, stereo, 5.1 and format conversion.

BS.1770-2 Compliant Metering

New LM6 loudness radar meter compliant with EBU R128, ATSC A/85, TR-B32 and ITU-R BS.1770-2. For legacy purposes, LM6 can also be switched to the ungated, original BS.1770 measure of Program Loudness.

Loudness meters in MKII frames feature 24/7 logging capability without even seeing a computer. Measurement and logging presets are found in the new Metering factory preset bank.

More Improvements

Version 3.20 includes various other enhancements. To name a few: Centralized preset handling, more SNMP functions, anti-aliased meter graphics, new Metering, Down-conversion and Up-conversion preset banks.

Main page

In Gain
Range: 0dB to Off
Separate level controls for Left and Right Input (A and B).

Phase Inv
Range: Normal/Inverted
Press to phase invert channels L (letft), R (right) or both.

Delay Unit
Range: ms, 24fps, 25fps, 30fps
With this parameter it is possible to select which unit the Delay parameter should be shown in. Changing this parameter does not affect the actual delay value.

Delay
Delay alignment of the Input channels. Depending on selected Configuration type, either one common Delay setting or individual delay settings are available.

Delay unit: “ms”	: 0 to 4000 ms
Delay unit: “Frames 24”	: 0 to 96 Frames
Delay unit: “Frames 25”	: 0 to 100 Frames
Delay unit: “Frames 30”	: 0 to 120 Frames

Center Gain

Range: Off, -12.0 to 0.0 dB
Downmix gain for the Center input relative to L and R front. Default Center gain would be -3.0 dB, but DMix employs a high resolution downmix structure with loudness, 5-band and true-peak limiting performed at 48 bit, fixed point precision. This enables the downmix gain to be set freely without worrying about overload or the loss of resolution. For extra emphasis on the Center channel, the gain may be run all the way up to 0.0 dB still without any risk of internal or output overload.

Surround Gain

Range: Off, -12.0 to 0.0 dB
Downmix gain for the Surround inputs relative to L and R front. Default Surround gain would be between -3.0 and -6.0 dB, but DMix employs a high resolution downmix structure with loudness, 5-band and true-peak limiting performed at 48 bit, fixed point precision. This enables the downmix gain to be set freely without worrying about overload or the loss of resolution. For extra emphasis on the Surround channels, the gain may be run all the way up to 0.0 dB still without any risk of internal or output overload.

Configuration

Select between Stereo, Dual Mono, Stereo Wide, Sum Mono, Left Mono, Right Mono.

Look ahead Dly
Range: 0-15ms
If the 5 band Compression sections is set to use a very short Attack times (up to approx 10-15ms) overshoots may occur. The Look Ahead function allows the DB8/DB4 to evaluate the material just before processing and artifacts can thereby be prevented.
Be aware that the Look Ahead delay function actually delays the output signal.

Reference level

Range: -24 to 0 dBFS
This parameter defines the 0 dB point for Target level in the Loudness section as well as the 0 dB point for the Thresholds in the 5-band section. It does not, however, affect the threshold of the output limiter, which is always referenced to 0 dBFS.

<u>Example</u>: To target an output loudness level of -23.0 LUFS (the same as -23.0 LKFS), Reference Level could be set to -23 dBFS and Target Level in the loudness section to 0.0 dB; or you could set Reference at -20 dBFS and Target Level at -3.0 dB. With the latter setting, the Threshold of the 5-band section would be 3 dB higher.

To target an output loudness level of -24.0 LUFS (the same as -24.0 LKFS), Reference Level could be set to -24 dBFS and Target Level in the loudness section to 0.0 dB; or you could set Reference at -20 dBFS and Target Level at -4.0 dB. With the latter setting, the Threshold of the 5-band section would be 4 dB higher.

To target an output loudness level of -27.0 LUFS (the same as -27.0 LKFS), Reference Level could be set to -24 dBFS and Target Level in the loudness section to -3.0 dB; or you could set Reference at -20 dBFS and Target Level at -7.0 dB. With the latter setting, the Threshold of the 5-band section would be 4 dB higher.

Loudness page

Target Level

Range: +10dB to -10dB
This is the level the Loudness adjustment section will aim at. Target Level is relative to Reference Level on the Main Page. See the Set-up Tip at the end of the DMix manual about how to fine-tune this parameter.

Max Reduction

Range: -20dB to 0dB
This is the maximum attenuation the Loudness Control is allowed to perform. If set to 0.0dB, the Loudness Control cannot attenuate the signal at all.

Max Gain

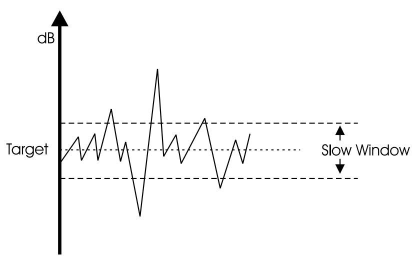
Range: 0 to +20dB
This is the maximum gain the Loudness Control is allowed to perform. If set to 0.0 dB, the Loudness Control cannot add gain to the signal at all.

Freeze Level

Range: -10dB to-40dB
Sets the minimum level required before the Loudness Control will start adding more gain. It would typically be set to avoid boosting signals considered noise. The Freeze Level parameter is relative to the Reference Level setting on the Main page.

Freeze Hold

Range: 0 to 5 seconds
When the Input signal drops below the Lo Level, the Gain Correction of the Loudness Section is frozen for the duration of the Hold time. When the Hold period expires, the Gain Correction falls back to 0dB gain.



Level Trim

Range: -18dB to + 18dB
The processing resolution of DMix is 48 bit, so it's possible to also convert and correct loudness manually without the risk of overloads. The Level Trim can be used for permanent gain offsets or for risk-free live adjustments.

Level Trim is the perfect control for shifting broadcast platform loudness target. While a target loudness of -23.0 or -24.0 LUFS is generally assumed for the HD platform, that's too low for mobile and pod platforms. (Remember how "LUFS" is the same as "LKFS". -24.0 LUFS is the exact same loudness level as -24.0 LKFS). A suitable loudness target for mobile platforms is in the range between -11 and -18 LUFS/LKFS.

Based on investigation of the gain structure in Apple devices, we suggest aiming mobile platforms at -15 LUFS/LKFS. A higher mobile target level is possible, of course, but at the risk of damaging audio integrity more than necessary. (Details in the NAB 2011 BEC paper, "ITU-R BS.1770 Revisited", by Thomas Lund).

If the HD platform is aimed at -24 LUFS/LKFS, and all programs consequently pre-normalized to that level, DMix may in one pass do format change, loudness adjustment, loudness target shifting to -15 LUFS/LKFS, and true-peak limiting. The Target setting in the Loudness section of DMix should stay around -24 LUFS/LKFS, while Level Trim should be set to +9.0 dB (the difference between the HD target and the mobile platform target).

Note: You may need to also move the All Threshold parameter in the 5-band section up in order not to invoke too much 5-band processing.

Ratio

Range: 1:1.25 to 1:6
Ratio is the adjustment factor used when the Loudness section applies boost or attenuation to aim at a certain Target Level. The higher the ratio, the more rigid steering towards the Target Level.

Example: With a setting of 1:2, the Loudness control section adjusts the gain by 1 dB when the input is 2 dB off target (if a gain adjustment is allowed by the Max Attenuation and Max Gain parameters).

With a setting of 1:1.25, the Loudness control section adjusts the gain by 1 dB when the input is 5 dB off target (if a gain adjustment is allowed by the Max Attenuation and Max Gain parameters).

Average Rate (Avg Rate)

Time constants in the Loudness Control are changed dynamically with the Input signal based on computations by multi-level detectors. When the Output level is close to the Target Level, gain changes are relatively slow.

The Average Rate offsets all time constants to be faster or slower. Values below 1dB/Sec produces a gain change gating effect when the Output level is already in the target zone, while values above 4dB/Sec will add density to sound.

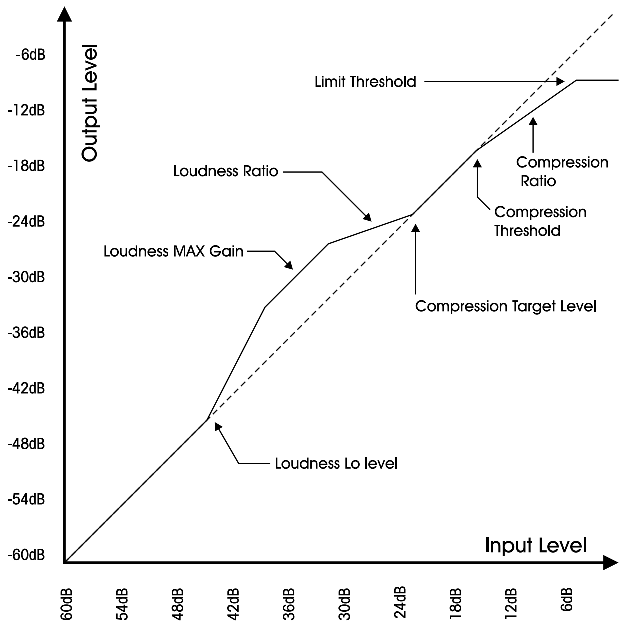
Slow Window

Range: 0 to 20 dB
The slow window is the area around the set Target Level. Within the slow window the Loudness is only gently controlled. When the signal exceeds the limits of the Slow Window the Loudness is treated more radically. Depending on the set Average Rate and Ratio.

Loudness Meassure

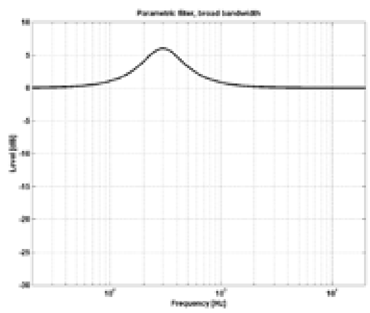
Select between ITU BS.1770 and ITU BS.1770-2
The loudness model employed in the Loudness section is based on Leq(K) weigthing. This parameter selects if programs should generally aim at Target values measured without gating, like in the original ITU standard (BS.1770 setting), or measured with gating, like in the current ITU standard (BS.1770-2 setting).

Multiband parameter illustration

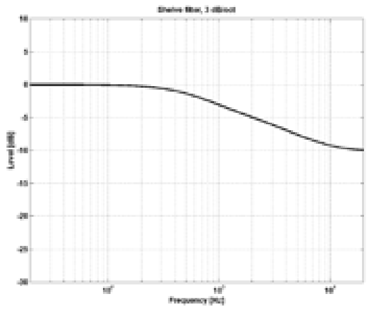


<u>For the Mid filter select between filter types:</u>
Parametric and Notch.

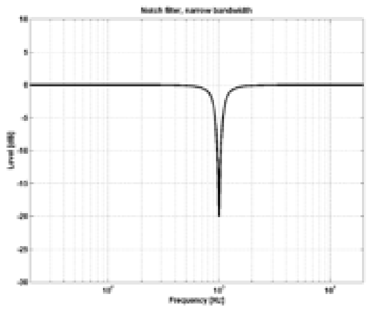
Parametric Filter - Broad type



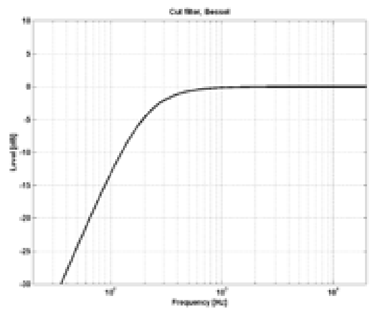
Shelving Filter



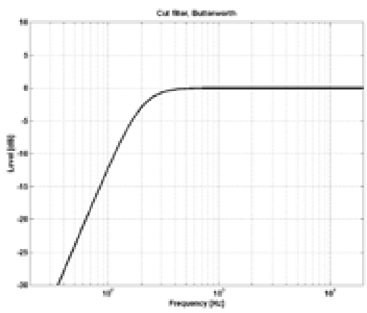
Notch Filter - Narrow Type



Cut Filter - Bessel type



Cut Filter - Butterworth type



Freq

Press **Freq** and use Faders 1 to 3 to adjust frequence for each of the four bands.

Range - Lo band : 20Hz to 20kHz
Range - Mid band : 20Hz to 20kHz
Range - Hi band : 20Hz to 40kHz

Gain

Press **Gain** and use Faders 1 - 3 to adjust gain for each of the four EQ bands.

Range for the Parametric, Shelve and Cut type:

Lo Gain	: -12dB to +12dB
Mid Gain	: -12dB to +12dB
Hi Gain	: -12dB to +12dB

Range for the Notch filter:

Lo Gain	: -100dB to 0dB
Mid Gain	: -100dB to 0dB
Hi Gain	: -100dB to 0dB

Type

Press and use Faders 1-3 to set BW value for each of the 4 EQ bands.

Range for the Notch filter:

Lo BW	: 0.02oct to 1oct
Mid BW	: 0.02oct to 1oct
Hi BW	: 0.02oct to 1oct

Range for the Parametric filter:

Lo BW	: 0.1oct to 4oct
Mid BW	: 0.1oct to 4oct
Hi BW	: 0.1oct to 4oct

Range for the Shelve filter:

Lo BW	: 3dB/oct to 12dB/oct
Hi BW	: 3dB/oct to 12dB/oct

Range for the Cut filter:

Lo BW	: Bessel or Butterworth
Hi BW	: Bessel or Butterworth

Bandwidth/Q - Key-Values:

BW		Q
0.5	-	2.87
0.7	-	2.04
1.0	-	1.41

5 Band Page

Xovers

Press this button to access the four cross-over points between the five-bands. The parameters are Automatically assigned to faders 1-4.

Parameter range:
Xover 1: Off to 1,6kHz
Xover 2: Off to 4kHz
Xover 3: 100Hz to Of,
Xover 4: 250Hz to Off

Defeat Thresh

Range: -3 to -30dB
This is a unique control which holds the gain from the multiband compressor below a certain threshold. No matter the spectral shaping applied from multiband system, below the Defeat Threshold, the frequency response is flat and gain is unity.
Defeat Threshold is relative to Compressor Threshold, which is relative to Reference Level.

Defeat Ratio

Range: Off to Infinity
Controls how close to the Defeat Threshold the make-up gain of the compressor is counteracted. At high ratios, the signal only has to be slightly below the Defeat Threshold before the compressor gain is fully defeated.

Threshold

Parameter range: -25 to 20dB
Press this button to access the five individual band Threshold is relative to Reference Level set at the Main page.

Gain

Parameter range: 0 to 18dB
Press this button to access the five individual band Gains and the overall All Gain.

Ratio - DXP mode OFF

Parameter range: Off to Infinity:1
Press this button to access the five individual band Ratios and the overall All Ratio.
The parameters are automatically assigned to fader 1-6.

Attack

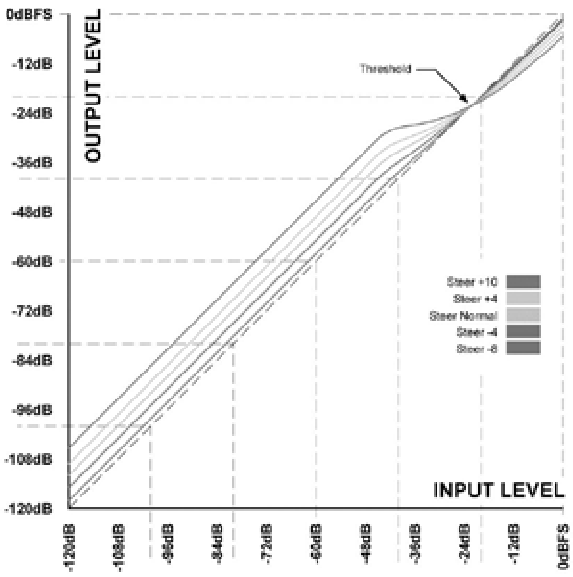
Parameter range: 0.3 to 250ms
Press this button to access the five individual band Attacks and the overall All Attack.
The parameters are automatically assigned to fader 1-6.

Release

Parameter range: 20ms to 7s
Press this button to access the five individual band Release and the overall All Release.
The parameters are automatically assigned to fader 1-6.

DXP Mode - Introduction

The 5-band section is either in normal compression mode, or DXP mode. Instead of attenuating signals above a certain threshold, DXP mode (Detail Expansion) lifts up signals below the Threshold; thereby bringing out details rather than squashing the loud parts. DXP mode therefore is capable of adding intelligibility and air to speech, lifting harmonics, or emphasizing ambience without increasing overall peak level.



As shown on the illustration, gain is positive below threshold, unity at Threshold, and the effect decreases above Threshold. In DXP mode, Ratio becomes Steer. Steer can be regarded as an adaptive Ratio that gradually approaches 1:1 above the threshold.

<u>Multiband DXP</u>
DXP mode can be used with any number of bands up to 5. When used multiband it is particularly effective in bringing out air and clarity.
The processor can act as an automatic Eq that removes a boost when it's not needed: At very low levels, where noise is dominant, and at loud levels where sibilance would become a problem. Besides from being effective on speech, DXP mode can be used in mastering to bring up low levels, e.g. when preparing film or concerts for domestic or noisy environment listening.
Try setting the Steer and/or Threshold parameters differently in the bands to hear the effect. High Steer values add more detail gain than low values, but remember that Threshold has to be negative to add detail gain at all.
DXP Threshold relates to the Reference Level set on the Main page.

NOTE: To disable DXP detail gain at very low levels, use the Defeat Threshold and Defeat Ratio controls. Defeat threshold relates to the DXP threshold, and allows for a certain level-window, inside which detail gain is applied. Defeat Ratio determines the slope at which DXP detail gain is defeated.

Limit page

Link Limiter

When Link is active, the same amount of peak limiting is always applied to both channels.

Some broadcasters like the sound of operating left and right limiting without stereo coupling because they feel that it maximizes loudness and widens the stereo image. On dual mono sources, of course you should always choose un-linked Limiter operation.

NOTE: The Configuration control on the Main page does not affect the Link Limiter setting. This link is running individually from the selected configuration.

Softclip L/R

Parameter range: - 3dB to Off
When active, Soft Clip applies a saturation effect on signals close to maximum Output level. The threshold is relative to the Threshold of the Brickwall Limiter.
This controlled distortion of transients works well for adding loudness, but is not a desirable effect with some data compression codecs. While the Brickwall Limiter is extremely low distortion, Soft Clip is not. Use your own judgement if you want it or not.

Threshold L/R

Parameter range: -12 to 0.0dBFS
Sets the Threshold of the Brickwall Limiter.
The Threshold is relative to 0 dBFS, not to the Reference Level set on the Main page.

The output limiter detects and protects against true-peak signals as defined in ITU-R BS.1770, ITU-R BS.1770-2 and in EBU R128. This precision limiter is based on 48 bit processing and utilizes adaptive time constant for low distortion operation.

Fader

Parameter range: Off to 0dB
Fader function on the Output. When Dual Mono configuration is selected, individual Output faders are available.

Set-up Tips

DB processors feature precise ways to probe the current loudness status of a station. When deciding the amount of processing needed, it's suggested to load an LM6 loudness meter on the input and one on the output of DMix. After a few days, you will have a picture of how much input and output loudness fluctuates. This should trigger advice to production from time to time, and maybe adjustments to delivery specifications or normalization procedures.

Note: When reading LM6, remember that units "LKFS" and "LUFS" are the same (besides from the letter "K" vs. letter "U"). A Program Loudness reading of, for instance, -25.3 LUFS, is precisely the same as -25.3 LKFS.

The goal should be an ever improving and predictable loop, spanning from production to distribution, and not to process more than necessary for a certain broadcast platform. Don't take pride in being the loudest station, but in being the best sounding and most consistent one.

For broadcast stations early in the process of converting production to loudness based criteria, a relatively high Loudness adjustment Ratio may be initially needed, for instance 1:2, in order to avoid too much loudness fluctuation during transmission. Once production adopts loudness metering, and programs are normalized prior to transmission, the Ratio control should be relaxed and/or the Max Attenuation and Max Gain should be moved closer to 0.0 dB.

Based on LM6 output measurements, it may be indicated to raise Target Level over the expected. While the BS.1770-2 Loudness Measure setting already helps on the average, a slightly higher Target may be needed (depending on type of programming) to get close to the station's loudness Target.

All loudness adjustment algorithms in DB processors feature extreme flexibility. Processing may be used to only attenuate or to only boost, and the amount of cut and boost may be restricted. Furthermore, it's easy to switch to limiting only on the fly, or to completely bypass processing, should certain programs have been precisely normalized and controlled already.