

## **Sound Solution**

Studio and FM broadcasting oriented multiband processor unit

### **Features:**

- Multiband Compressor-Expander-Limiter
- Center frequencies B1=170Hz, B2=1000Hz, B3=3200,B4=7200,B5=12000
- Gated AGC
- Stereo expander
- Dual band output limiter / compressor / clipper
- Bass Equalizer
- Built in preemphasis 50-75uS generator and lowpass antialias filter

### **What it does:**

Recording or broadcasting an audio signal requires that it should be limited in its dynamic range. In broadcasting the band must be limited in order to not interfere with the adjacent channels while in recording studios dynamics must be limited to avoid the final media saturation (BTW, if you analyze most CD contains severe clipping due to this effect).

This plugin realizes a complete multiband processor to “pump up” your music to make it ready to be broadcasted or recorded into a CD.

All controls are accessible to the user for maximum flexibility, even if this could be a bit nasty for the unexperienced people.

**Functional blocks:**

the functional blocks are explained in the same order that the audio signal encounters them.

**STEREO EXPANDER:**

it is a basic but effective stereo expander and it has two controls:

Level: the effectiveness of the stereo expander expressed in percent.

Depth: in a certain sense, is the “size” of the stereo image; large amounts can make the high frequencies (cymbals) to loose definition.

A “stereo image monitor” allows a visual indication of the stereo image content.

**AGC:**

the AGC is a slow acting leveling block that keeps the input signal of the multiband processor almost at the same level.

The “GATING” function increments the release time of the multiband processor to prevent noise rush up during low level passages. Both AGC and the multiband compressor will not remain “stuck” forever, but they will recover the maximum gain in about 1 minute.

The GATE THRESHOLD should be set around -30, -40dB for most music.

All compressor bands are connected to the same *gating* bus, except if you operate the processor in “INDEPENDENT GATE”: in such way every band samples *its input level* instead of the general input. Sincerely i cannot say to hear so much difference between the “independent” and “non-independent” mode, but i presented this control to increase flexibility.

Finally, the “OVERDRIVE” control sets the compressor driving level; high settings results in a louder and denser sound.

**BASS EQ:**

to strengthen the bass of your music you can use this small parametric equalizer.

Frequency: adjusts the frequency of the resonator. For deep bass set it around 60-65Hz, to enhance “drum punch” it could be set around 100-150Hz.

Gain db: adjusts the effectiveness of the resonator filter. The higher the setting, the higher the enhancing.

Remember that it *cannot create* “bass energy” if it is not present at all.

## **MULTIBAND PROCESSOR:**

it is the real “heart” of this plugin and many controls and functions are available to the user.

The three processes (compressor, expander, limiter) and the five bands are arranged as a 5x3 matrix; using the “CFG” for the *row* and “COMPRESSOR”, “LIMITER” or “EXPANDER” for the *column* you can tweak any parameter of any process.

For a more comfortable set up, the “GLOBAL” function make the parameter that is under editing to be copied to the other bands related to the same process.

That is, if, for example, you want to set all the compressor thresholds to –6dB, simply push the “COMPRESSOR” button and any of the “CFG” buttons of the multiband limiter.

Press the “GLOBAL” button and set the “Threshold” slider to –6dB.

Depress the “GLOBAL” button and check, pushing the other “CFG” buttons, that all bands have the threshold set to –6dB.

The “PROCESS ACTIVE” checkbox indicates if that process (compressor, expander or limiter) is active or not.

### **Compressor:**

a compressor is used to decrease the dynamic range of an audio signal so that a “x” dB increase of the input signal will result in a “x/ratio” change of the output.

For example, if the compressor ratio is set to “4.0” and the input signal suddenly rises of 10dB, the output signal will rise only of “ $10/4.0=2.5\text{dB}$ ”.

Below the threshold the signal is unaffected.

### **Expander:**

an expander is the complementary process of the compressor so that the output signal is decreased of “x\*ratio” dB for an “x” dB fall of the input signal below the threshold; it is used for noise reduction.

For example, if the expander ratio is set to “2.0” and the input signal falls 4dB below the threshold, the output signal will fall “ $4*2.0=8\text{dB}$ ”.

Above the threshold the signal is unaffected.

### **Limiter:**

a limiter is an *infinite ratio* compressor, so for any increase of the input signal above the limiter threshold, there is no change in the output signal.

As for the compressor, below the threshold the signal level is unaffected.

### **Ratio:**

As said, it is the effectiveness of the compressor or expander. For the limiter the control is still present but has no effect.

### **Threshold:**

The level above or below which the compressor/expander becomes active.

**Attack-release:**

All these level changes are applied to the signal within some time.

The time the processor takes to adapt to an increase of the signal is called the “Attack time”, the time it takes to recover its state after a transient is called the “Release time”.

Both affects the sound, a small “attack” time makes the processor to react very quickly to the changes of the signal thus preventing clipping, but sounding quite unnaturally if very short times are used. Short release times tends to create a “denser” sound, but exaggerating this effect leads to unpleasant and fatiguing listening.

It is a wise solution to adjust the attack time according the operating band, that is using a slow attack time (40-60mS) for the bass frequency band (band1), a medium-slow (20-30mS) time for the low-medium band (band2) and shorter times for the high frequency bands.

**Band Gain:**

After the block of bandpass filters that split the audio signal into separate bands, after the expansion/compression/limiting processes, the signal is summed again to form the broadband signal. The “band gain” control sets the level of mixing of the various bands in order to give the signal a sort of equalization. Adjust it according your taste.

**Band Link:**

It links, in percent, two bands so preventing, for 1-2 band link, an abnormal bass expansion during a voice solo or, for the higher bands, it helps in reducing high frequency clipping.

**Dual Band Compressor Limiter:**

Well, now the signal has been stereo expanded, leveled, splitted into five bands, dynamically expanded, compressed and limited then summed again.

It is now spectrally balanced, it might sound louder, but it still presents some peaking and, for this reason, it is not still suited to “fit” perfectly a transmission medium.

So, a final two bands compressor/limiter smoothes the processed sound splitting it again into two bands, a high-frequency band above 6KHz and a broadband band with all below 6KHz.

The controls are the same of the multiband processor so no further explaining is necessary, but i may suggest to use high ratios for both the LF and HF bands in order to prevent high clipping induced distortion.

Adjust the “Density Overdrive” control to fit your needs of loudness.

Last control, not present into the multiband limiter, is the clipping level; since some peak always escapes the multiband compressor and the final two bands limiter, a soft clipper removes the remaining peaks obtaining the output peak limited signal.

## **OTHER CONTROLS:**

**M (Mute):** as the name says, pressing it results in the muting of the related band. It is useful to have an idea on how the bands are divided into.

**Test tone:** a 324Hz tone is generated to allow the alignment of the recording equipment to the maximum output level.

**Input conditioning:** when pushed, it turns on and off the input conditioning filters. Even if it is suggested to keep this feature on, you may leave it according your taste.

**Bypass all:** the output is directly connected to the input, and it is useful for comparison.

**No Clipper:** self explaining. It switches off the clipper.

**No Expander:** it turns off the stereo expander. Useful for comparison.

**Preemphasis on:** turns on the preemphasis generator. If you use this processor for FM broadcasting, you can feed directly a stereo exciter. In this case you have to turn off the preemphasis generator of your exciter and turn on the one into your processor. The time constant should be selected according to the rules of your country.

**Levels Monitor:** it simply shows the input and output levels thus giving an idea of the processor efficiency. The "OUT LEV" slider sets the output volume.

**Load-Save Preset:** the first time the plugin is loaded, it creates an ss0.dat file "DEFAULT" preset. You can edit and save (choosing a number) up to 21 presets (0-20). When a loaded preset is edited, an "E" (edit) will flash near the "ON AIR PROCESS" indication. If the process is saved or another is loaded, the flashing "E" will disappear.

## **Known bugs:**

As for the "Tomass Limiter", this plugin has the same bug related to the buffers used in the stereo expander. So, using the stereo expander, you might experience clicks and pops while passing from one song to another, this because I still don't know how to catch the "song change" event from the Winamp window.

I focused my attention on DSP routines, but this bug is still alive. Have you some DDT to kill it? This cannot be noticed if you use any of the continuous or gapless play plugins available.

### **Some last words:**

An audio processor is a nonlinear process, that is, even if the output signal might be slightly more attractive respect to the input signal, it has *surely* more distortion respect to the input.

For this reason, always try to use the cleanest possible audio material to get the cleanest output audio and remember that the correct adjustment of a multiband processor is a quite complicated task, so, don't be frustrated if after thirty minutes you still cannot get your sound.

Remember also that "extreme loudness" settings are well suited only for certain kinds of music (pop, disco, rap...) while are awful for other kinds of music; so always adapt the processor to the format of your radio.

General settings often sound well with most music formats even if they have not the same loudness. The output sound is always a tradeoff between loudness distortion and cleanness and you must choose the tradeoff between them according your taste.

Another last tip is that you might avoid the use of high stereo expander levels especially with low bitrate mp3s; mpeg puts a lot of "trash" into the stereo image so, often, enhancing the stereo image you could enhance just "mpeg noise" (a sort of "tingling" you may hear at medium-high frequencies).

Last recommendation is to use a good quality audio board with a "flat" output, that is no further bass or treble equalization must be added and only in this way you may expect to have a good peak control and a satisfactory audio quality.

The design of this processor represents my studies about DSP and C programming and the efforts of many good people who tested this product and gave me precious indication to make it better and better. I'm still looking for people who wants to contribute freely to this project, to make many "one penny radios" to be on air with a satisfactory audio quality without having to spend much money for a professional equipment.

You can contribute with suggestions about audio algorithms and or audio processing techniques (clippers, bass enhancers, stereo expanders...) or simply with your ears to test my "frankenstein's" processors; i'm here to listen your ideas.

### **Thanks to:**

Paul Gallo for ideas and support.

Jean Marc, Tom, Steve for precious suggestions about broadcast processing.

### **Mail bombs and cheers to:**

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Made in Prepo (PG) Italy.

#### Updates in version 1.2 rev1

- Added RMS detector in multiband processor. The two bands limiter still process peak level.
- Added clipping level indication

#### Updates in version 1.2 released

- **15KHz lowpass input filter (15KLP button)**  
Use it when broadcasting/webcasting to avoid above 15KHz material to excessively drive HF compressing/limiting circuitry.
- **Alternate 4 bands filter bank**  
Maybe it can provide a more natural sound, but i still like the five bands configuration because it is a bit more “sparkling”.
- **Intelligent dual timed attack/release circuitry**  
Further improvement to reduce pumping.  
For the moment the second release time is not accessible.
- **Multiband Clipper**  
Still under development. In this version the amount of multiband clipper effect is done with the “HF Clipper” slider. The “LF Clipper” slider controls a standard broadband soft clipper.
- **Look ahead processing**

#### Bugs fixed

- **Crashes with plugin stackers and even alone**  
“Quick and Well cannot stay together”...some wrong initialized parameters caused such crashes, sometimes also a conflict with msvcr7.dll caused the same nasty behaviour.  
Thanks to Ross Delaforce.
- **Processor window now closes and pops up only if asked**  
Another stupid bug into parameter saving-restoring routine.  
By the way, window position restoring was added.

#### Under development

- **Variable knee hard/soft clipper**  
Testing it.
- **LF and HF shelving equalizers**  
Designing.
- **Ver. 2 stereo expander**  
Designing.

#### The future

**Bah!....who knows?**