MDT™

Multiband Dynamics Tool™

for use with
Digidesign™ TDM compatible systems

User's Manual

Version 2.2t

ANTARES™

Creating the Future of Music Technology

MDT – Multiband Dynamics Tool™

AnTares Systems
P.O. Box 697
Applegate, CA 95703-0697
(916) 878-4400

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Welcome!

I would like to extend my congratulations to you on purchasing the most powerful dynamics processing tool available. With your purchase of MDT, you have created a relationship with my company which I hope will be long and gratifying.

As a registered user of MDT, you are entitled to notification of software upgrades, technical support, and to special introductory offers on upcoming products. We will be in contact with you to announce new opportunities and to solicit your feedback.

At AnTares Systems, we are committed to excellence in service, quality, and technology innovation. You can count on us to listen to you and to keep our promises to you.

Andy Hildebrand, Ph.D.

andy@antares-systems.com

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CHAPTER 1

Getting Started

The Multiband Dynamics Tool is a unique DSP plug-in for Digidesign's™ TDM system that provides unprecedented control over the dynamics of your recordings. Unlike existing dynamics processors, MDT allows the sound to be shaped interactively, giving you creative possibilities never before available.

MDT can create compressors, limiters, expanders, downward expander/gates, "soft knee" compressor/limiters, "tube" companders, or any combination of these. In multiband mode, MDT can create de-essers, spectral enhancers, and dynamic EQs, as well as compressors, and expanders that suppress the effects of artifacts like "breathing" and "pumping".

MDT replaces expensive digital dynamics processors in applications like mastering, track sweetening, sound effects, sample editing, or any other application where high end digital dynamics processing is needed.

How To Use This Manual

MDT has a transparent user interface and is extraordinarily easy to use. However, the operation of MDT and some functions of the user interface will not be immediately obvious because they do things which have never been done before. We strongly recommend that you read chapters 2 and 3 of the manual to take full advantage of the quality and control that MDT makes possible.

We assume that you already know how to operate ProTools and TDM. If you have questions about this, refer to your ProTools manual or call Digidesign for technical support.

For Those Who Hate To Read Manuals...

We strongly recommend that you work your way through the MDT Tutorial in Chapter 3. It doesn't take very long and the work will pay for itself many times over. If you can't bring yourself to go through the tutorial, we recommend that you keep the manual nearby as you work with MDT. When something puzzling comes up, you can look it up in the Index.

The Contents Of The Manual

Chapter 1: Getting Started
This chapter explains everything about installing MDT, communicating with AnTares Systems, and using the manual.

Chapter 2: Introducing MDT
This chapter explains the scope of MDT's functions. Basic concepts about dynamics processing are also covered. The user interface is explained.

Chapter 3: MDT Tutorial
The MDT Tutorial guides you through a step by step process which shows you how to use MDT in various single band and multiband applications. We recommend this chapter as "must" reading for everyone.

Chapter 4: MDT Reference
This chapter explains every object and function in MDT. Items are organized in alphabetical order by name.

Chapter 5: Theory of Operation
This chapter explains, in technical terms, how MDT works.

Software Notes

MDT 2.0t is a Digidesign TDM compatible plug-in. It runs with ProTools version 2.5 or greater with TDM installed or any other system capable of running TDM plug-ins.

Owner Registration

Your purchase of MDT entitles you to technical support, special introductory offers on new products from AnTares Systems, and notification of software updates. Software updates will be published as the program evolves.

Please fill out and return the Owner Registration Card. The information on the card will allow us to communicate more effectively with you and will enable us to serve you better in the future.

Installing MDT

Instructions for installing MDT are located in the Read-Me file. Please refer to this file for the most up to date information on installing and de-installing MDT.

Technical Support

If you have some problem using MDT that can't be solved by reading the manual, call Richarde & Co. (the AnTares Systems distributor) at (800) 446-2356, or (408) 688-8593 Monday through Friday between 9 AM to 5 PM PST.

Also, you might find what you want at our web page:

www.antares-sytems.com

You can also e-mail:

techsupport@antares-systems.com

CHAPTER 2

Introducing MDT

MDT is a breakthrough among with dynamics processors. It puts your hands on the dials and levers of the DSP process itself, letting you shape the results in ways never available before. Access to this new level of flexibility and control is achieved by shifting the way you think about how compression and expansion work. This chapter introduces MDT's operating paradigm and gives the information needed to use it effectively.

Understanding Compression

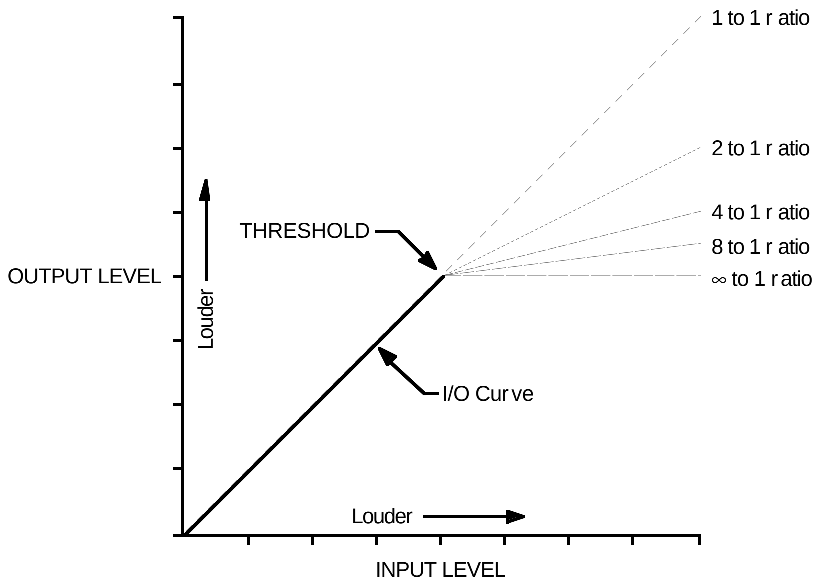
Next to reverb, compression is probably the most important signal process used in today's studios. Simply put, compression reduces the *dynamic range* of a signal. That is, it reduces the difference in loudness between the loudest and quietest parts of a piece of music. Another way to think about this is that the compressor is acting as an automatic fader which fades down when the music gets loud and fades back up when the music gets soft.

Why reduce the dynamic range? Consider mixing a vocal into a pop music bed. Typically, pop music has a relatively consistent level of loudness. If an uncompressed vocal track is added to a typical pop mix, certain loudly sung words or syllables would be very obtrusive, while quieter phrases would be buried underneath the instrumental texture. This is because the difference between the loudest and softest sounds in the vocal, its dynamic range, is very large. This same problem occurs for any instrument which had a dynamic range larger than the music bed into which it is being mixed.

By using a compressor to decrease the dynamic range of the vocal, the softer sounds are increased in loudness and the loudest sounds are reduced in loudness, tending to even out the overall level of the track. This makes the vocal track sound generally louder and more distinct, and therefore, easier to hear in the mix.

Ratio And Threshold

How is compression measured? What is a little compression and what is a lot of compression? The concept called *compression ratio* is the measure of how much the dynamic range is compressed. Look at the illustration below.



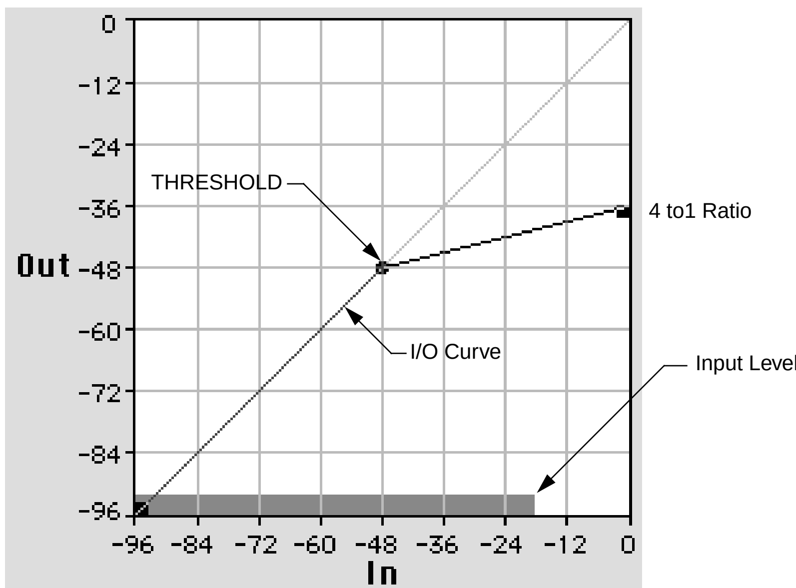
This graph represents the relationship between the input level of the signal and the output level of the signal after compression. Notice that the curve has a breakpoint called a *threshold*. All standard compressors use a threshold. Signals that are louder than the threshold are processed (reduced in level) while those softer than the threshold are unchanged.

As the input signal exceeds the threshold, *gain reduction* (reduction in loudness) is applied. The amount of gain reduction that is applied depends on the compression ratio. The higher the compression ratio, the more gain reduction is applied to the signal.

The graph shows the relationship between compression ratio and gain reduction. Examine the 2 to 1 ratio curve. For signals above the threshold, this curve transforms a range of loudness 2 units large into a range of loudness one unit large. Examine the ∞ to 1 curve. This curve transforms all sounds above the threshold to the same loudness. Dynamics processors which have this sort of curve are called limiters.

MDT As A Compressor

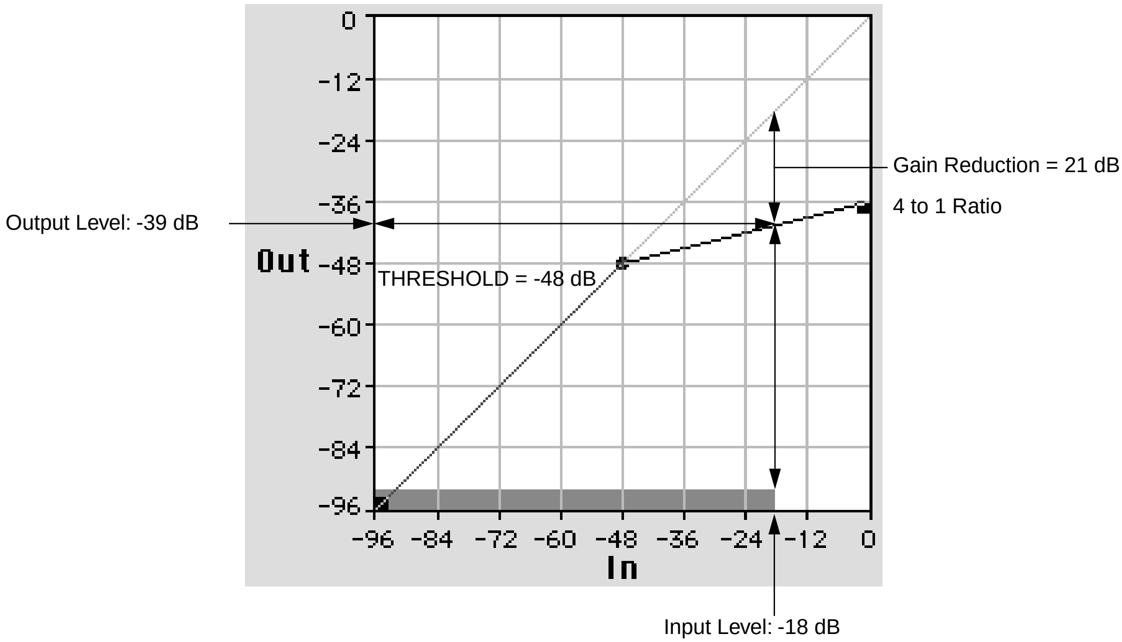
The graph described above appears in MDT's user interface. The following illustration is a screen shot of MDT's In/Out Grid.



Like the first graph, this is a graph of the Input Level versus the Output Level. The curve on the graph has a threshold and the curve segment above the threshold has a 4 to 1 compression ratio.

The gray bar at the bottom of the graph is an input level meter. It shows how loud the input signal is so that you can see where on the curve it falls. In the illustration below, the input level is at -18 dB. The threshold is at

-48 dB. The input is above the threshold so it is getting a gain reduction of -21 dB. This puts the output at -39 dB.



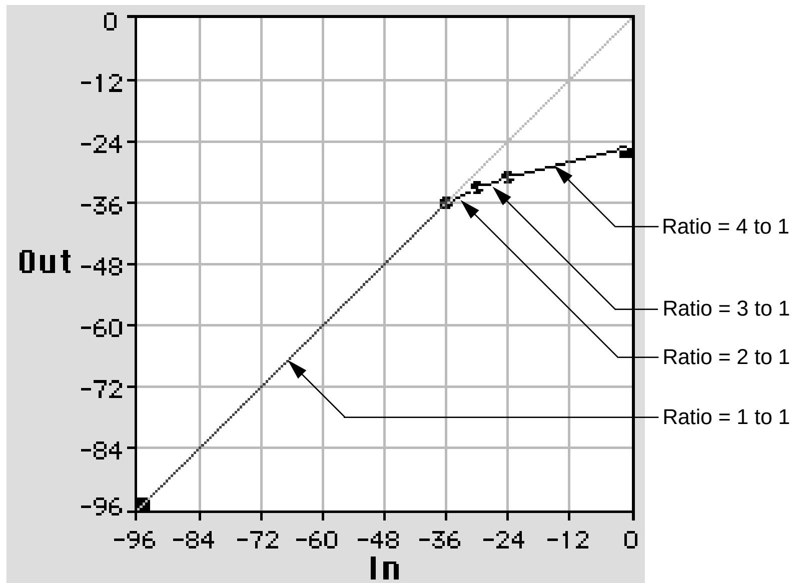
The In/Out Grid

In conventional dynamics processors, there are knobs or sliders that control the threshold and ratio parameters. MDT uses a graphic interface called the In/Out Grid for these adjustments. The In/Out Grid provides visual feedback of the effect the tool is having on the sound. The complexity of the processor's configuration is easily controlled using multiple thresholds and their associated compression ratios.

The details of how to use MDT's graphic interface are explained in Lesson 1 of the MDT Tutorial.

MDT's I/O Curve can be arranged to create many different kinds of dynamic processing devices in addition to compressors. Many esoteric and expensive outboard dynamics processors can also be emulated. The Setting menu contains many settings which you will find useful in creating "sweet" sounding digital expanders, spectral enhancers, companders, and other tools useful in mastering, tracking, and sound design. The examples below explain some of the configurations that are possible using MDT's unique graphic interface.

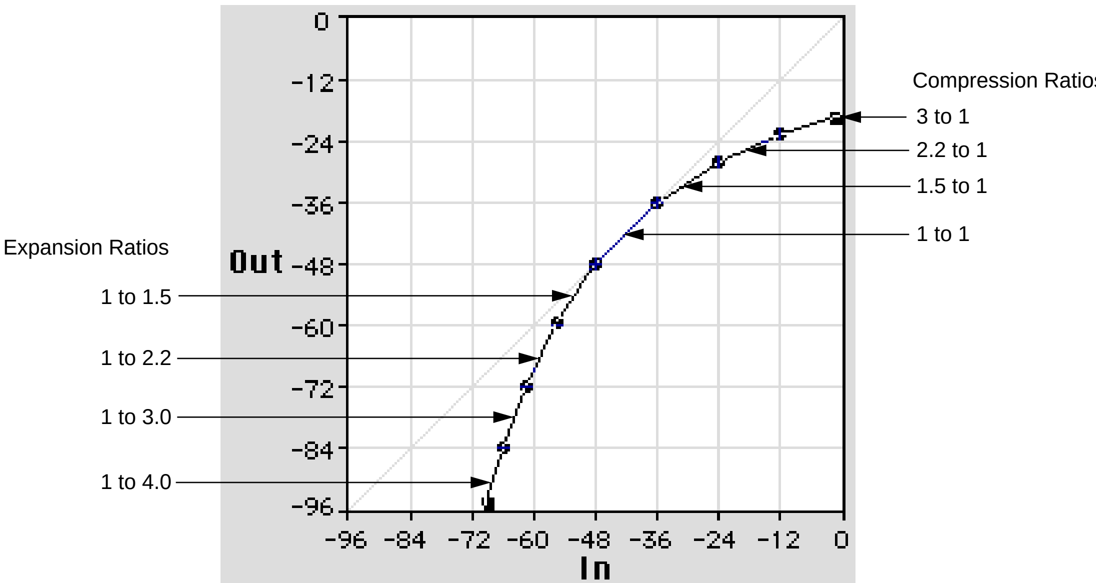
"Soft Knee" Compressor



"Soft knee" compressors sound less obvious than regular "hard knee" compressors because the compression ratio near the threshold changes gradually. In the example above, as the signal gets louder than the threshold, it is first compressed at 2 to1. As it gets louder still, the ratio goes to 3 to1, until, finally, it reaches its maximum ratio of 4 to 1. This setting sounds less "squashed" than a plain 4 to 1 hard knee compressor because only the peaks in the signal get the full 4 to 1 compression.

The "softest knee" dynamics processors available are the tube-type compressors, especially the vintage variety. The following example shows how MDT can be configured to emulate a tube compander.

"Tube" Compander



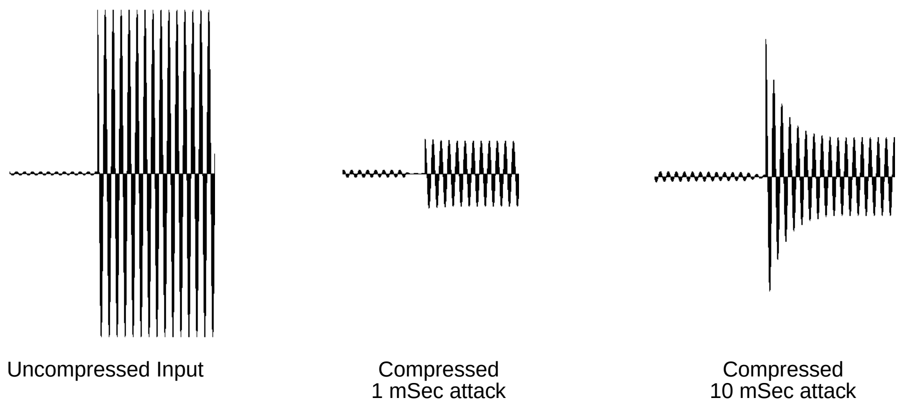
The curve shown above has no clear threshold point where the signal starts to get processed. Instead, the compression or expansion ratios change gradually. It is called a compander because it both compresses and expands the dynamic range, depending on the level of the input. The top half of the curve compresses the dynamic range. The bottom half of the curve expands the dynamic range.

The sonic effect of processing through this kind of setting is very natural. Only the peaks of the sound get compressed heavily, while most of the signal passes through at a 1 to 1 ratio. This assumes that the average input level is around -40 dB. (See "The Input Offset Arrows" on page 27 for an explanation of how to "place" an input on the curve without actually changing its level.)

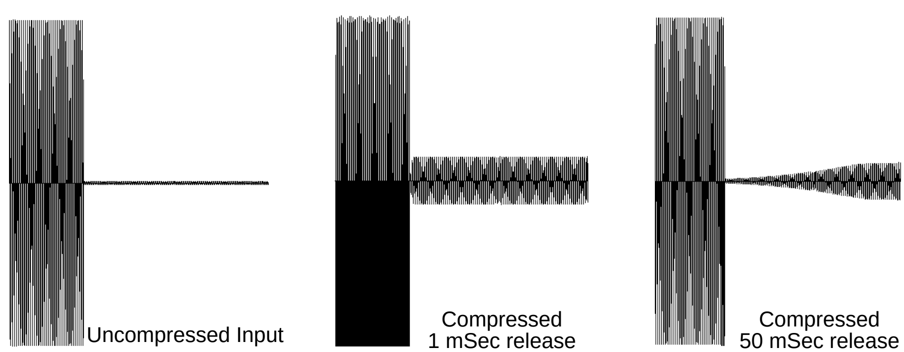
Sounds quieter than -48 dB get expanded downward, that is, they are faded down as they get softer. Because the curve increases in slope gradually, this subtle gating effect sounds very natural. It is useful for eliminating noise and maintaining a sense of wide dynamic range while still compressing the hot peaks in the signal.

Attack And Release Times

The attack time of a compressor is simply how long it takes for the compressor to react once the input level has met or exceeded the threshold level. With a fast attack time, the signal is brought under control almost immediately, whereas a slower attack time will allow the start of a transient or a percussive sound to pass through uncompressed before the processor has time to react. Creating a deliberate overshoot by setting an attack time of several milliseconds is an effective way to emphasize the percussive nature of instruments. The illustration below shows the effect of changing the attack time.



The release time of a compressor is the time it takes for the gain to return to normal after the input level drops below the threshold. Setting too quick a release time can cause a pumping effect in the output. If the release time is too long, the compressor will not accurately track level changes in the input. The illustration below shows the effect of changing the release time.



MDT In The TDM Environment

The TDM software environment is supported by NuBus hardware from Digidesign called the DSP Farm. Each DSP Farm card contains DSP chips on which MDT and other DSP plug-ins run. The TDM system requires that one chip on the DSP Farm be used for the Mixer plug-in which comes with ProTools. Assigning a large number of ProTools voices will sometimes require the use of two DSP chips on the DSP Farm.

MDT processes two audio channels per DSP chip. The table below shows how many DSP chips are used as more MDT channels are allocated.

Number of MDTs used	Number of DSP chips allocated
2 mono to mono MDTs	1
1 stereo to stereo MDT	1
1 mono to mono plus 1 stereo to stereo MDT	2

NOTE: Sometimes TDM will post a warning dialog saying that there are an insufficient number of DSP chips available when you have added a normally legal number of MDTs. This is because of the order in which TDM allocates the DSP chips as you add more inserts. If this occurs, de-assign a MDT or two and then re-assign them. This allows TDM to sort out its DSP allocation, allowing the greatest use of the available DSP chips.

CHAPTER 3

MDT Tutorial

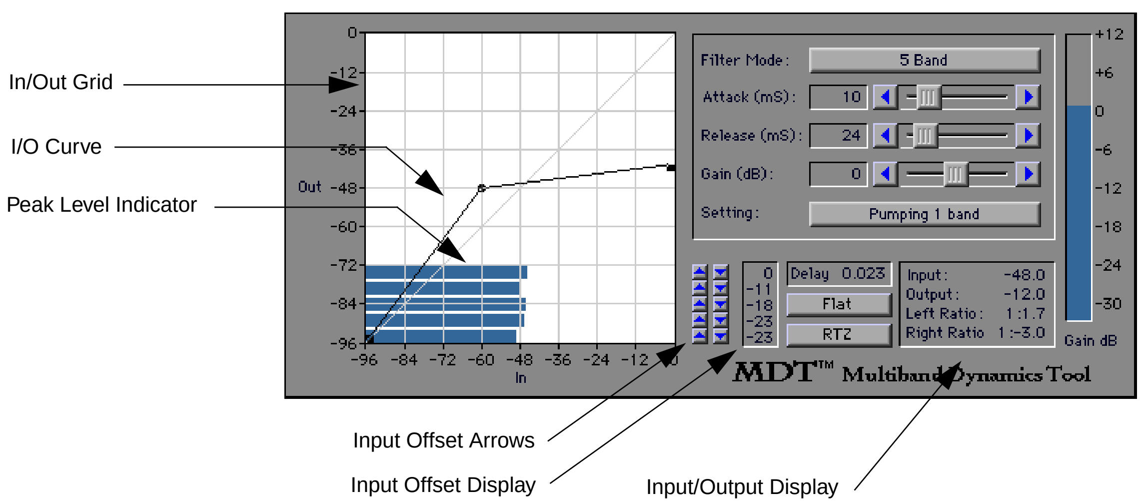
"I don't want to know how it works –
I want to know how to work it!"

– Keith Emerson

This chapter is a step-by-step explanation of how to use MDT. MDT is a breakthrough in technology and is therefore unique. If you follow these lessons one step at a time, you will master MDT and fully benefit from its capabilities.

The whole tutorial takes about 90 minutes to complete.

Lesson 1: MDT Basics



This lesson presents the elements of MDT's user interface.

About DSP Plug-ins

DSP Plug-ins are software programs that run inside ProTools, and other programs to add functionality and provide a variety of DSP tools. All TDM compatible plug-ins have a few basic features in common. This section describes those features.

1. **Start ProTools and open the session named "MDT Tutorial".**
2. **Select "Show Inserts View" from the Display Menu.**
3. **Click on the MDT insert in the track labeled "Aux 1".**
4. **Choose Reset Settings from MDT's Setting menu.**
5. **In the Transport Window, click on the locator point labeled "Voice".**

The Peak Level Indicator

1. **Start playback.**

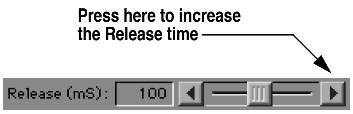
   A Peak Level Indicator appears at the bottom of the In/Out Grid of MDT. It displays the current input level that MDT uses to make gain adjustments.

   As the soundfile plays, notice the dynamic range that the Peak Level Indicator covers.

MDT uses look-ahead peak averaging to determine the loudness of a sound. Look-ahead peak averaging means MDT scans the data about to be processed for upcoming peaks while calculating the average peak level of the data it's currently working on. It uses both numbers to derive the final input level. MDT can scan up to 1024 samples ahead, depending on the DSP card being used.

2. **Use the mouse to press on the right arrow of the Release slider. Increase the release time to 4,999 mS (milliseconds).**

   Notice that, as the release time increases, the Peak Level Indicator's movement slows down and the displayed signal level gets higher on the IN scale. This is because, as the release time increases, it takes longer for hot peaks to be averaged out of the level calculation. The actual signal



may be fluctuating wildly, but the displayed level will always be the average value of the peaks inside the time window determined by the Attack and Release settings.

3. **Drag the Release slider to the far left position. Notice the change in the Peak Level Indicator.**

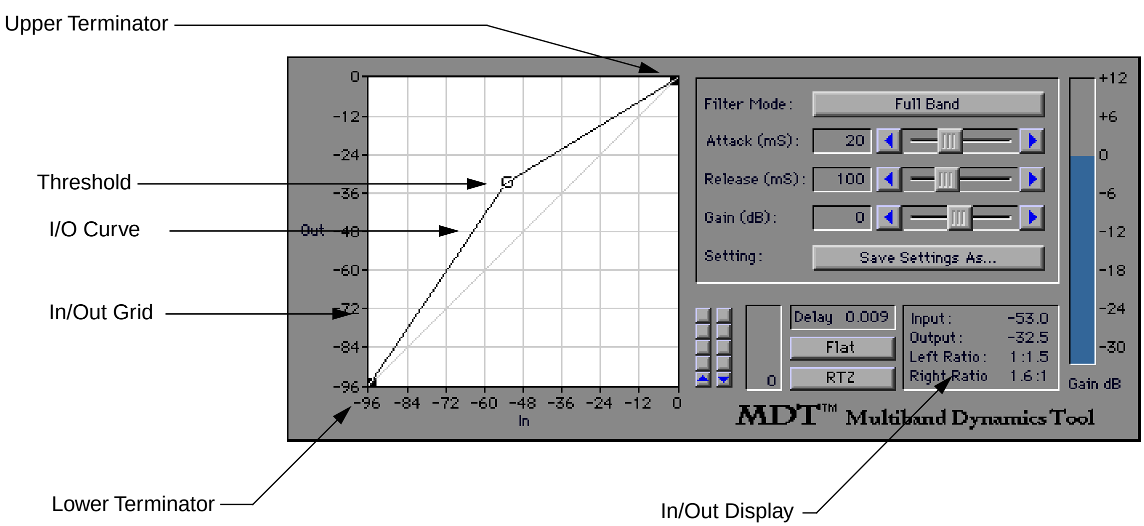
   With a release time of 0 milliseconds, the Peak Level Indicator more closely reflects the actual instantaneous sound level.

4. **Stop playback.**

5. **Reset MDT by choosing Reset Settings from the Setting menu.**

The I/O Curve

MDT uses a unique graphic interface to describe the relationship between Input and Output levels. This relationship is called the *transfer characteristic* or, in MDT parlance, the I/O Curve. This section demonstrates the manipulation of the I/O Curve.



NOTE: Points on the I/O Curve or the In/Out Grid will always be written as a pair of negative numbers, e. g. -60:-60. The first number is the location of the point on the In coordinate of the In/Out Grid. The second is equal to its Out value on the Grid.

1. **Move your mouse cursor into the In/Out Grid.**

   The cursor changes into the cross cursor shape.

2. **Click and drag the mouse inside the Grid.**

   A threshold appears in the I/O Curve. As you move the threshold around the Grid, the I/O Curve "rubber bands" to follow the movements of the cursor. Notice that the threshold location is displayed in the In/Out Display. Also notice that the In/Out Display shows the angle of the two moving line segments expressed as the ratio of the Input to the Output.

3. **Release the mouse.**

   The threshold remains at the release point. It is now highlighted as shown in the illustration above.

4. **Move the threshold to In/Out coordinates -60:-60 by placing the cursor over the threshold and dragging it to the new location.**

5. **Place the cursor over the lower terminator (the small square object in the lower left corner of the In/Out Grid) and drag it to the right to -60.5:-96.**

   Notice that the lower terminator is "glued" to the In axis. It will always have an Out value of -96 dB. The In/Out Display shows the ratio of the moving curve segment. As you drag the lower terminator to the right, notice that it cannot go past the threshold at -60:-60. This is to prevent the I/O Curve from having more than one output value for any given input value.

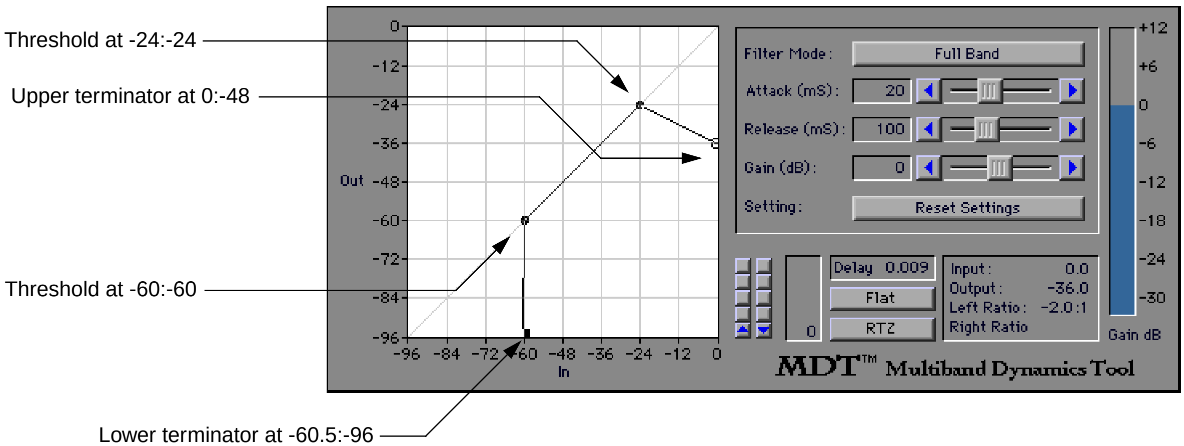
6. **Drag the upper terminator (in the upper right corner of the In/Out Grid) to 0:-48.**

   The upper terminator is glued to the Out axis. It will always have an In value of 0dB. During the drag, the In/Out Display shows the ratio of the moving curve segment.

7. **Add a second threshold by clicking the cursor at -24:-24.**

   While positioning this threshold, notice that the In/Out Display is displaying the positive going curve on the left as an N to 1 ratio, and the negative going curve on the right as a -N to 1 ratio. It is possible to use MDT to create inverse gain functions.

At this point in the exercise, MDT should look like this:



8. **Delete the thresholds by pressing <option> and clicking on the thresholds.**

   The thresholds disappear and the curve snaps to its new position automatically.

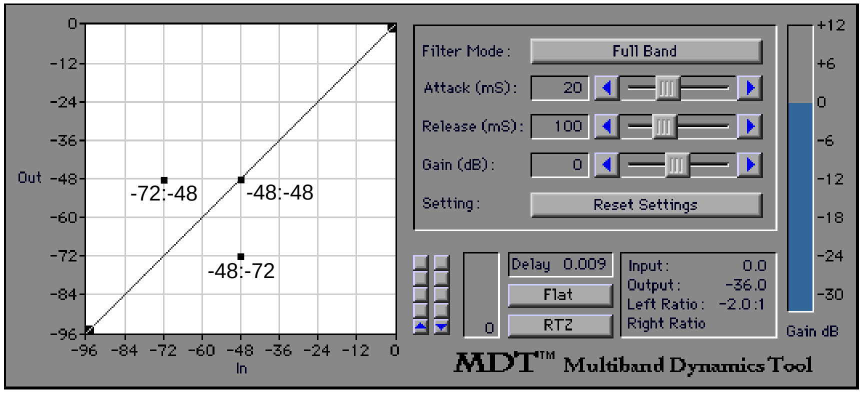
9. **Reset MDT by choosing Reset Settings from the Setting: menu.**

The In/Out Grid

MDT works by detecting the level of the input signal, looking for the point on the I/O Curve with that input value, and then adjusting the gain so that the output level equals that of the I/O Curve at that point.

All the possible points where the I/O Curve can exist are represented by the In/Out Grid. It is divided by a diagonal line that represents the points on the Grid that have input levels equal to output levels (e.g. -48:-48). Placing the I/O Curve on this diagonal will cause MDT to output the signal at the same level as it was input.

The points on the Grid which have output levels lower than their input levels (e.g. -48:-72) are located below the diagonal. Placing the I/O Curve below the diagonal will cause MDT to output the signal at a lower level than it was input (gain reduction).



Points on the Grid which have output levels higher than their input levels (e.g. -72:-48) are located above the diagonal. Placing the I/O Curve above the diagonal will cause MDT to output the signal at a higher level than it was input (gain amplification).

The following section demonstrates the relationship between I/O Curve position and gain.

1. **Choose 1:1, -12 dB gain from the Settings menu.**

2. **Start playback.**

   The voice is heard 12 dB lower in volume because the I/O Curve shifts every point on the In axis -12dB on the Out axis.

3. **Adjust the Gain slider to read 12 dB.** 

4. **Click the Bypass button on and off to verify the level match.**

   The processed level is now precisely the same as the original level because of the additional gain supplied by the Gain slider. The gain factor set into the Gain slider is applied after the signal is processed by the I/O Curve and is the last stage of control in MDT.

5. **Choose 1:1, -24 dB gain from the Settings menu.**

   The voice is heard 24 dB lower in volume because the I/O Curve has shifted everything an additional -12 dB lower on the Out axis.

6. **Adjust the Gain slider to read -12 dB.** 

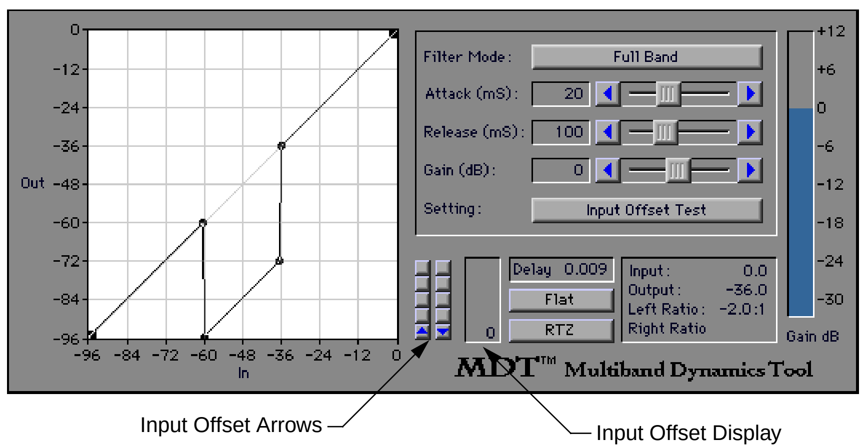
7. **Choose 1:1, +12 dB gain from the Settings menu.**

The voice is now heard at the original volume level. The I/O Curve is amplifying it by 12 dB and the Gain slider is reducing the gain by -12 dB. Use the Bypass button to verify this.

8. **Stop playback.**
9. **Choose Reset Settings from the Settings menu.**

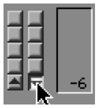
The Input Offset Arrows

Input Offset is another unique feature of MDT. The Input Offset Arrows (see illustration) allow you to change the relationship of the input signal to the I/O Curve without actually changing the signal level of the input data.



In a single band configuration, the placement of the thresholds can be tweaked using the Input Offset without disturbing the I/O Curve. In multi-band configurations, spectral bands can be independently adjusted. By adjusting the different bands to the same level, coloration is reduced. Adjusting the different bands to completely separate sections of the I/O Curve allows independent compression ratios and levels for each band. These multiband cases will be demonstrated later.

1. **Choose Input Offset Test from the Settings menu.**
2. **Start playback.**



3. **Press and hold down the downward Input Offset Arrow.**

   As the Peak Level Indicator moves to the left, notice that the loudness of the playback remains the same. This is because the input data is not being changed. Only its relationship to the I/O Curve is being changed.

   When the Peak Level Indicator moves through the region between -36 dB and -60 dB on the Input scale, notice that the loudness of the sound decreases by 36 dB. This is because the I/O Curve is set for a -36 dB gain reduction in that region. As the Peak Level Indicator moves below -60 dB, the output goes back to normal because the I/O Curve is back at unity gain (0 dB gain).

   The Input Offset Arrows always increment in 1 dB steps. Press <option> then click or press on the arrows slows the rate of change.

4. **Press on the upward Input Offset Arrow until the reading returns to 0.**

   The value displayed in the Input Offset Display (see illustration) is the difference in dB between the actual input level and the displayed level.

5. **Stop playback.**

6. **Choose Reset Settings from the Settings menu.**

The Setting Menu

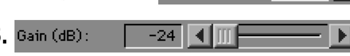
MDT has a Setting menu which allows you to store your most used MDT set-ups and recall them almost instantaneously. The Setting menu allows an unlimited number of "snapshots" of MDT's settings to be saved. This data is saved in the "AnTares Preferences" file in the System's "Preferences" folder and are compatible with settings from the Sound Designer version of MDT. In this section you will learn how to save, recall, and delete settings from the Setting menu.

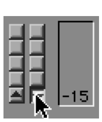
1. **Put four thresholds on the I/O Grid at In:Out = -12:-48, -24:-12, -36:-48, and -48:-12.**

2. **Choose 5 band AP from the Filter Mode menu.**

3. **Increase the Attack time to 50 mS.** 

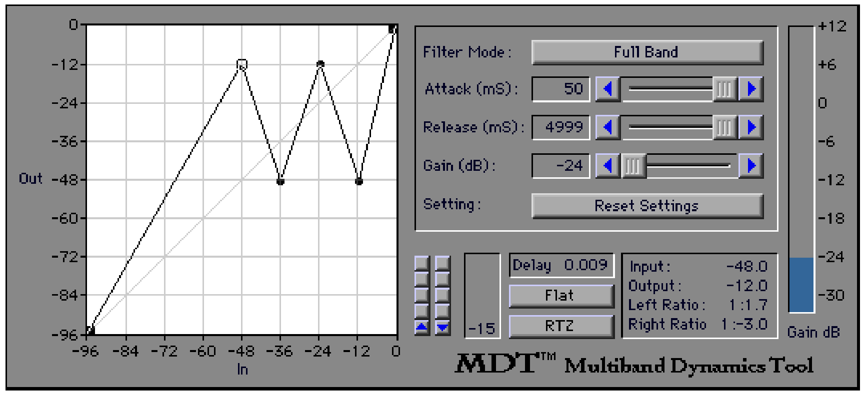
4. **Increase the Release time to 4,999 mS.** 

5. **Decrease Gain to -24 dB.** 



6. **Decrease the Low Band Input Offset to -15.**

At this point MDT should look like this:



To save these settings,

1. Choose Save Settings As... from the Setting menu.

A dialog box appears requesting a setting name.

2. Type Test **in the text box and click on Save.**

The settings are now saved under the name "Test". These settings appear at the bottom of the Setting menu.

3. Choose Reset Settings from the Setting menu to revert to the default values.

To recall the setting;

1. Choose Test from the Setting menu.

The settings are recalled as you saved them.

To delete the setting;

1. Choose Delete Settings... from the Setting menu.

A dialog appears with the list of the settings in the menu.

2. Scroll down to Test and select it by clicking on it.

3. Click on the Delete button.

The setting is deleted from the menu. You may select multiple settings for deletion by shift-clicking or click dragging over the items to be deleted. To make a non-contiguous selection, press <command> and click on the items to select them. Note that the settings themselves remain on MDT until you move them yourself or choose another setting.

Congratulations! You now know everything there is to know about MDT's user interface. If anything is unclear at this point, go over the section in question before moving on.

Lesson 2: Single Band Applications

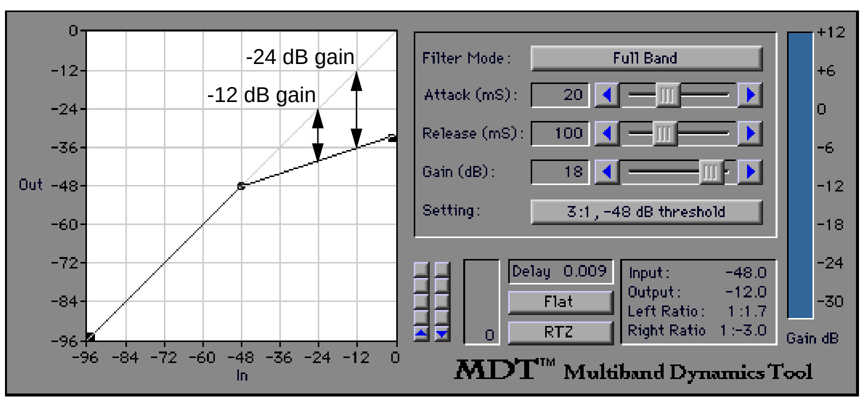
The following section demonstrates how to configure MDT as a single band compressor, limiter, gate, expander, "soft knee" compressor/gate, and "tube" compander.

Compressor

1. **In the Transport Window, click on the locator point named "Fetes 1".**
2. **Click on the MDT insert in the track labeled "Aux 1".**
3. **Choose Reset Settings from the Setting menu.**
4. **Start playback.**

   The selection is an excerpt from Fetes (rhymes with "pet"), the second of Debussy's Nocturnes for orchestra. Notice the relative loudness of the brass notes compared to the quiet strings that follow. The Peak Level Indicator shows an approximate dynamic range of >24 dB.

5. **Choose 3:1, -48 dB threshold from the Settings menu.**



Notice the shape of the I/O Curve. Sounds that have input levels lower than -48 dB are in a 1 to 1 relationship with the output. Above -48 dB, the signal will get a different gain reduction depending on its input level.

For example, a sound input at -12 dB will get a -24 dB gain reduction and be output at -36 dB. A sound at -24 dB input will only get a -16 dB gain reduction and be output at -40 dB, and so on. A piece of music with a dynamic range from -12 dB to -36 dB (a 24 dB dynamic range) would be output from -36 dB to -44 dB (an 8 dB dynamic range). Shrinking 24 dB down to 8 dB is a 3 to 1 compression ratio.

The Gain slider is set at 18 dB to compensate for the gain reduction of the compressor. The loudest sounds will come out at approximately the same level as before compression.

6. **Start playback.**

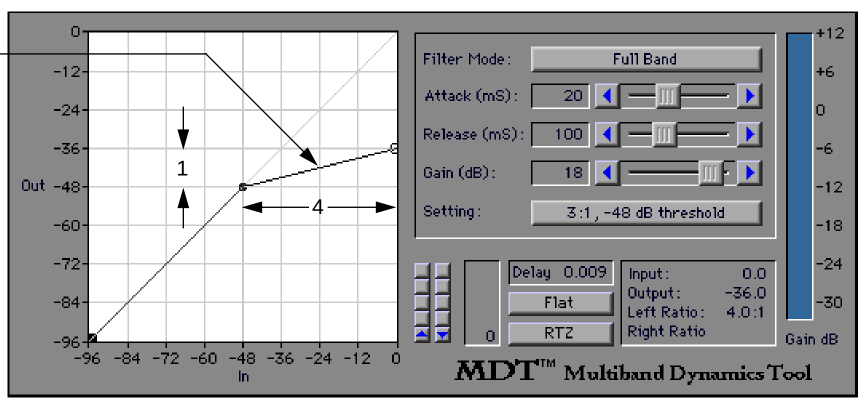
   Notice how the quiet string passage is now louder than before. The brass remains at its previous level.

7. **Use the Bypass button to compare the original with the compressed signal. Deactivate Bypass when you're done.**

8. **Change the compression ratio to 4 to 1 by dragging the upper terminator down until the Left Ratio field in the I/O Display equals 4:1.**

   The compression ratio approaches ∞ to 1 as the I/O Curve approaches the horizontal. At a ratio of ∞ to 1, all sounds above the threshold would be output at the same level.

A four unit change in the input gives a one unit change on the output.



9. **Stop playback.**

Attack and Release Times

The time it takes for MDT to react to an incoming peak is controlled by the Attack Time. The Release Time determines how fast the input level will return to its quiescent state after the peak comes through. The following section is a demonstration of the effect that Attack and Release times have on the sound.

MDT should now be adjusted to a 4 to 1 compression ratio from the last section. If it isn't, please do this now.

1. **Click on the locator point named "Pluck" in the Transport Window.**
2. **Reduce the Gain setting to 14.** 

3. **Start playback.**
4. **Use the Bypass button to compare the original with the compressed signal. Deactivate Bypass when you're done.**

   The region, "Pluck", is a loud violin section pizzicato with a reverb tail. Notice how the compression exaggerates the reverb and the noise in the sample. Notice the "shape" of the reverb tail and its decay time. Also notice the body of the attack portion of the sample.
5. **Reduce the Attack time to 0 mS.** 

   The initial attack is now compressed. MDT reacts immediately to change the gain so that the attack is output at a much lower level.
6. **Press on the right arrow of the Attack slider. Notice the change in the body of the sound as the Attack time increases.**

   At 50 mS, the attack is much louder than the sustained portion of the "Pluck". Changing the Attack time lets you tailor the aggressiveness of the attack to your needs.
7. **Return the Attack setting to 20 mS.** 

8. **Increase the Release setting to 4,999 mS.** 

   Notice how the reverb tail sounds almost exactly like the uncompressed version. This is because the Release time is much longer than the reverb time. The shape of the reverb tail is not effected because the detected input level (as shown by the Peak Level Indicator) decays very slowly when the Release time is this long.

This slow release time has an additional effect. Since the peak input level is high, the gain reduction of the output signal is also high, causing it to play back more softly than before. The Gain slider can be adjusted to compensate for this effect.

9. Decrease the Release time to about 300 mS. 

At this setting, the rate of change in output gain is approximately equal to the reverb decay rate. The result sounds like a smooth decay with the reverb compressed.

To increment the Release time in 1 mS steps, hold down the <option> key while pressing on the Release slider arrows.

10. Stop playback.

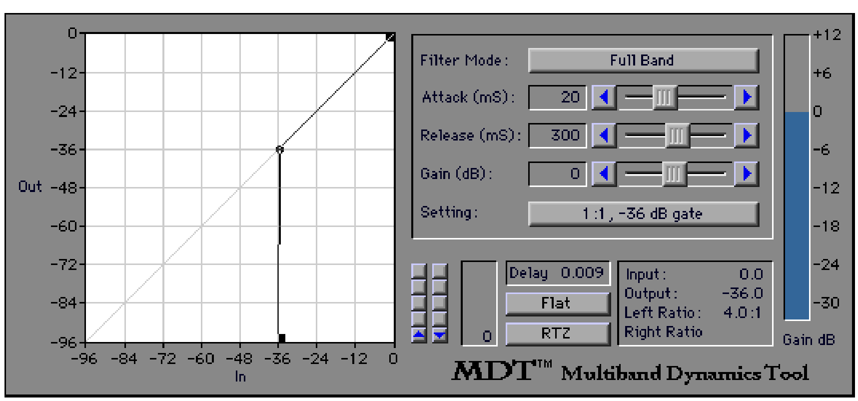
Expander/Gate

MDT can be used as an expander/gate to eliminate noise and for creating gated effects. This section demonstrates how to use the I/O Curve to configure a downward expanding gate.

Example 1: Downward Expander

1. Choose 1:1, -36 dB gate from the Settings menu.

2. Start playback.

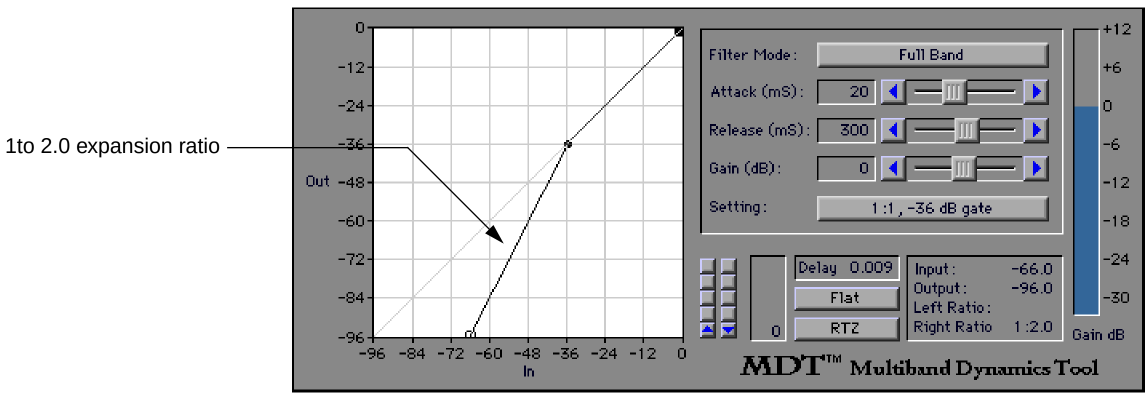


As the reverb decays below the -36 dB threshold, the output gain changes to -60 dB, gating the sound off.

3. **Press <option> and then the downward Input Offset Arrow until the display reads -15.**

   The reverb tail is now gated off sooner because the input level reaches the -36 dB threshold sooner. The Input Offset Adjust Arrows can be used in conjunction with the Release time slider to get exactly the gate time you wish.

4. **Move the lower terminator to the left until the value displayed in the I/O Display equals 1:2.0.**



The reverb now decays evenly because the output gain tapers gradually below the -36 dB threshold.

5. **Stop playback.**

Example 2: Compressor with Noise Gate

1. **Click on the locator point named "Voice" in the Transport Window.**
2. **Choose 3:1, -48 dB threshold from MDT's Settings menu.**
3. **Start playback.**

   Notice that the region has a high noise floor. The noise level is just below the -48 dB threshold.

4. **Drag the lower terminator to the right as far as it will go.**

You will hear that the noise is gated out for the most part. There is still a small noise just after the words "…Sound Designer II™…(noise)".

5. **Decrease the Gain setting to 9 dB.** 

6. **Press <option> and then the downward Input Offset Arrow until the display reads -15.**

   The noise is virtually eliminated because the input signal has been shifted down in relationship to the gating threshold. The noise now gates off immediately.

7. **Stop playback.**

"Tube" Compander

Many complex signals like full mixes and vocals require a high degree of dynamic control, but suffer from the effects of heavy compression or limiting. The use of a "soft knee" compressor or "tube-like" compander can make the effects of strong compression less obvious. A "soft knee" compressor is characterized by having a gradual change in compression ratio at its threshold. A "tube" compander has a continuously varying compression ratio over the range of the input.

Example 1: "Soft Knee" comp/gate

1. **Choose "soft knee" 4:1 -36 dB from the Settings menu.**

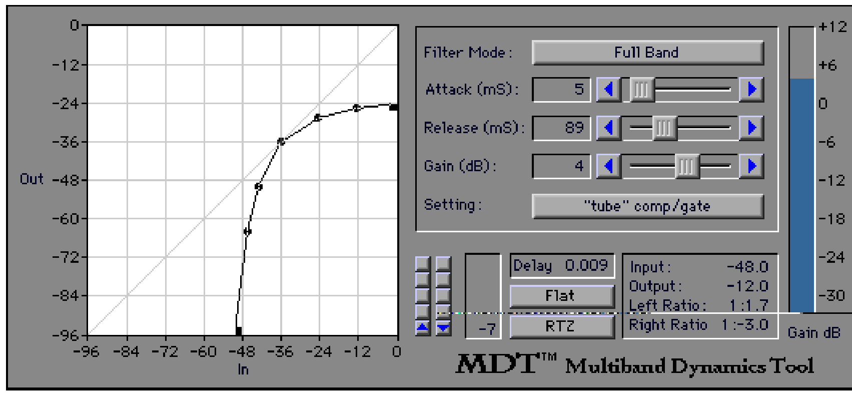
2. **Start playback.**

   Compare this sound with the previous example. The voice sounds less "squashed" even though the compression ratio is higher because the change in ratio at the threshold is less abrupt.

3. **Stop playback.**

Example 2: “tube” comp/gate

1. **Choose “tube” comp/gate from the Settings menu.**



2. **Start playback.**

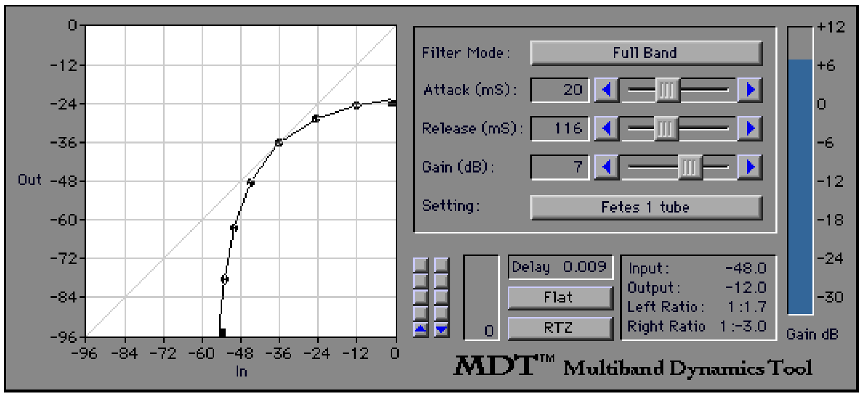
   Contrast the sound of the voice with the previous two settings. The vocal sound is even more natural because, most of the time, it is sitting on an area of the I/O Curve which has relatively small compression/expansion ratios. Only the peaks or valleys of the signal are effected by the high ratio segments of the I/O Curve.

   The reason “tube” companders work so well for complex signals is because the compression/expansion ratio changes gradually from some central operating point – in this case, -36 dB, making the effects of compression less audible. Normal compressors maintain the same ratio over a wide dynamic range.

3. **Stop playback.**

Example 3: Full mix

1. **In the Transport Window, click on the locator point named “Fetes 1”.**
2. **Start playback.**



The ratios of the downward expander part of the curve have been relaxed in this setting to accommodate the lower level and dynamic range of this excerpt. Compare the sound of this setting with the sound of the 3:1, -48 threshold setting that you used above.

3. **Use the Bypass button to compare the original with the compressed signal. Deactivate Bypass when you’re done.**
4. **Stop playback.**

End of Lesson 2

Lesson 3: Multiband Applications

In this lesson, you will learn how to use MDT in multiband mode. This capability of MDT makes it the most powerful and flexible dynamics processing tool available.

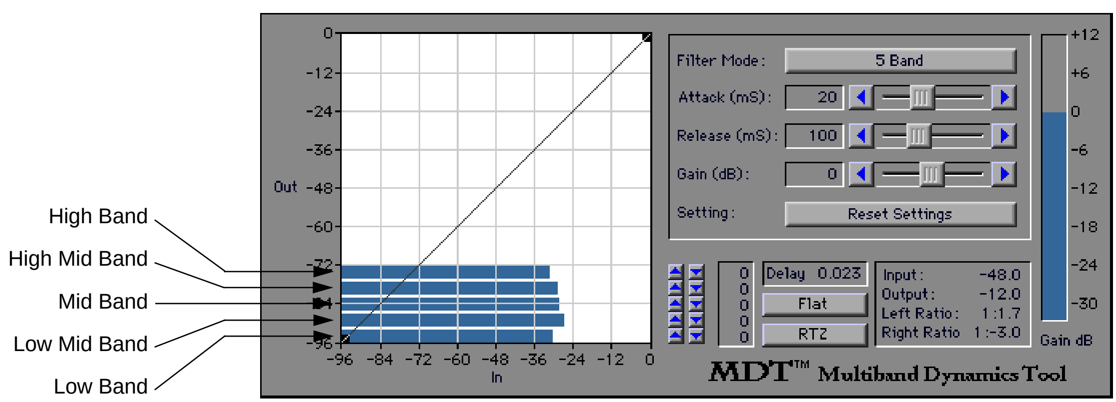
Multiband Peak Level Indicators

In multiband mode, MDT divides the energy of the signal into three or five equally spaced bands. The peak level of each band is displayed by its own Peak Level Indicator at the bottom of the In/Out Grid.

1. **In the Transport Window, click on the locator point labeled "Voice".**
2. **Select 3 band from the Filter Mode menu.**
3. **Start playback.**

   The Peak Level Indicators display the level of each band. The three bands (High, Mid, Low) are shown in that order, the Low band being at the bottom of the In/Out Grid. Each band is 3.3 octaves wide and has its own Input Offset Arrows.

4. **Select 5 band from the Filter Mode menu.**



In 5 band mode, five Peak Level Indicators are shown - High, Hi-Mid, Mid, Low-Mid, and Low. Each band is 2 octaves wide.

The Filter Modes

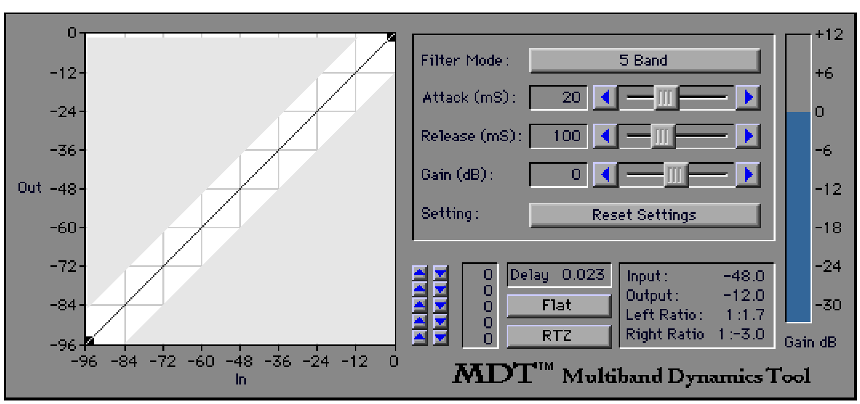
MDT uses digital filters to separate the energy of the input signal into multiple bands. These filters have particular characteristics which need to be understood to use MDT most effectively.

5. **Select 3 band from the Filter Mode menu.**
6. **Listen to the voice for tonal balance.**
7. **Use the Bypass button to compare the original with the multiband signal.**

   Notice a subtle change in the tonal balance of the voice when listening to the 3 band version. You are hearing the characteristics of the filters.
8. **Select 3 band AP from the Filter Mode menu.**
9. **Use the Bypass button to compare the original with the multiband signal.**

   Notice that the change in tonal balance is gone. The AP (all pass) versions of the multiband filters are optimized for minimum coloration at gains around 0 dB.
10. **Stop playback.**

Why not use the AP mode all the time? The AP mode filters have a limited gain reduction range of -12 dB. This means that the input signal can't be attenuated more than 12 dB without having its tonal balance seriously altered. In cases where more gain reduction is needed, the normal non-AP version of the filter will give the best result.



Placing the I/O Curve in the shaded parts of the Grid will cause unpredictable changes in the tonal balance when using AP mode filters.

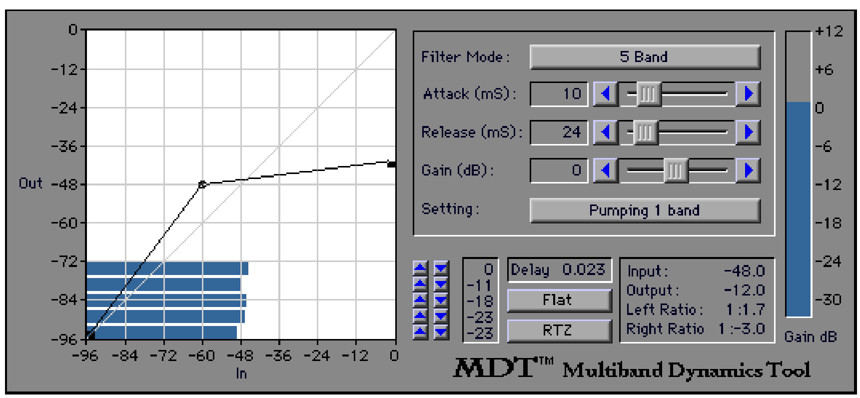
Multiband Compressor

There are many times when it is necessary to compress a complete mix. Often, especially when compressing at a high ratio, “pumping” can occur. This is when one sound or group of sounds in the mix is modulating the loudness of the other sounds. The following example demonstrates the use of a multiband compressor to alleviate the pumping problem.

1. **Click on the locator point labeled “Bass Solo” in the Transport Window.**
2. **Choose Pumping 1 band in the Setting menu.**

   The compression ratio above the threshold is 10 to 1.
3. **Start playback.**

   Notice how the loudness of the cymbal and sustained pad is “pumped” by the bass solo.
4. **Choose Better 5 band in the Setting menu.**



   This setting is the same as before except for the use of multiband mode. Notice how the pumping effect is greatly reduced.

Extreme settings are used in the example so as to create pumping in the output for the purposes of illustration. In a real world application, with a more moderate compression ratio and release time, the multiband compressor would provide even better results.

Managing Tonal Balance In Multiband Mode

You may have noticed in the previous exercises that the multiband Peak Level Indicators almost never have the same simultaneous level. This is because the energy distribution across the spectrum changes from moment to moment. Since all the bands are using the same I/O Curve, each band may receive a different gain depending on its current level and placement on the I/O Curve. The following example demonstrates this behavior.

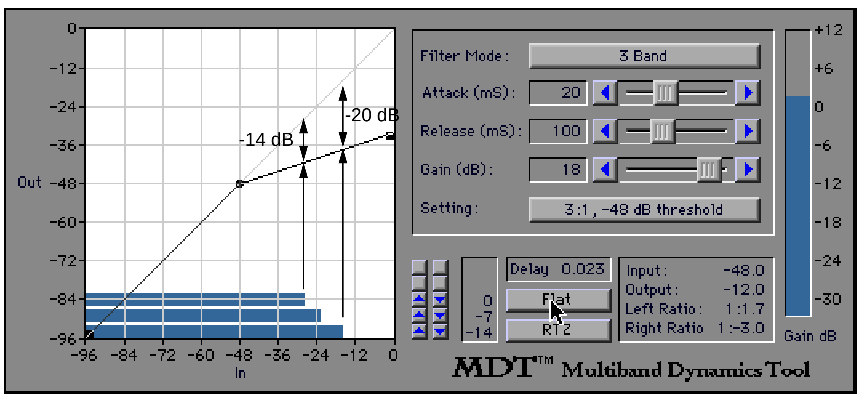
5. Click on the locator point labeled "Voice" in the Transport Window.

6. Choose 3:1, -48 dB threshold in the Setting menu.

7. Start playback.

Listen for the tonal balance of the voice.

8. Choose 3 Band in the Filter Mode menu.



Notice how the tonal balance is brighter. The highs are accentuated because the average level of the High band is about 5 dB lower than the other bands. This means that it is getting less gain reduction than the other bands and is therefore "hotter" in relation to them.

Since all three bands are occupying the same segment of the I/O Curve, they are all getting the same degree of compression. The only difference is the relative level of gain reduction each band is receiving.

9. Stop playback.

How To re-Gain Tonal Balance

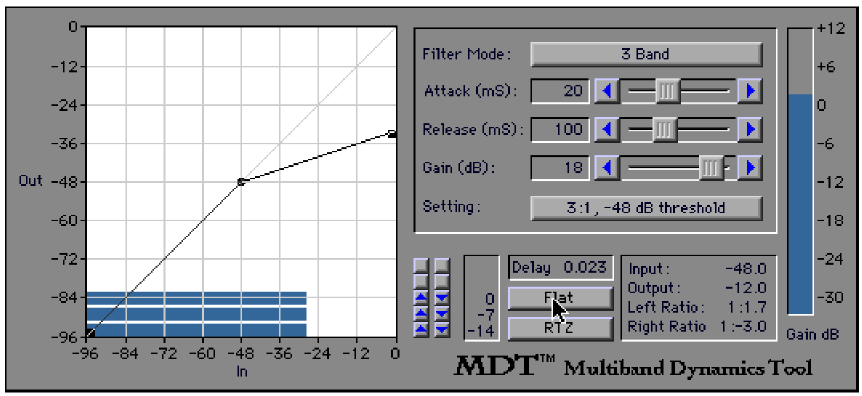
It is often desirable to preserve the soundfile's original tonal balance while benefiting from using multiband mode. The key element in preserving tonal balance is having each band get the same amount of gain at the same time. You can do this by offsetting the input level of the "hotter" bands to equal the lowest level band. The following exercise demonstrates this.

1. Start playback.

2. Increase the Release Time to approximately 1,000 mS.

3. Press the "Flat" button.

Offsetting the inputs so that they are about equal insures that all the bands receive the same gain.



The Input Offset values for each band are automatically adjusted so that they have equal amounts of gain reduction. Clicking on the "RTZ" (return to zero) button will zero the Input Offset values.

4. Return the Release time to about 100 mS.

Notice how the tonal balance changes as you make these adjustments. As the Peak Level Indicators are adjusted down the scale, the bands receive less gain reduction, and therefore get louder.

5. Choose Full Band in the Filter Mode menu.

Notice how the 1 band tonal balance almost matches the tonal balance of the adjusted 3 band sound.

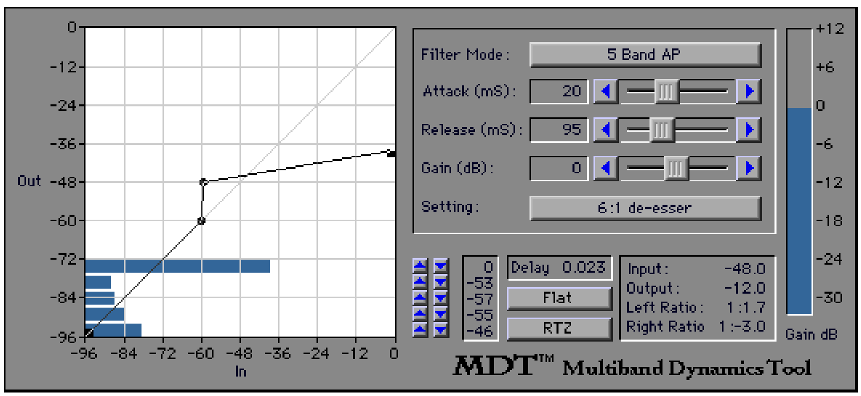
6. Stop playback.

MDT As A Spectral Enhancer

The tonal flexibility demonstrated above can be used to modify the existing tonal balance dynamically. De-essers, spectral enhancers, and dynamic EQs all modify the tonal balance dynamically. This example below shows how to use MDT to perform these functions.

De-esser

1. **Click on the locator point labeled "Voice" in the Transport Window.**
2. **Choose 6:1 de-esser in the Setting menu.**



The compression ratio above the threshold is 6 to 1. Notice how the I/O Curve straddles the diagonal. Sounds above the threshold and to the left of the diagonal get amplified. Sounds to the right of the diagonal receive gain reduction.

3. **Start playback.**

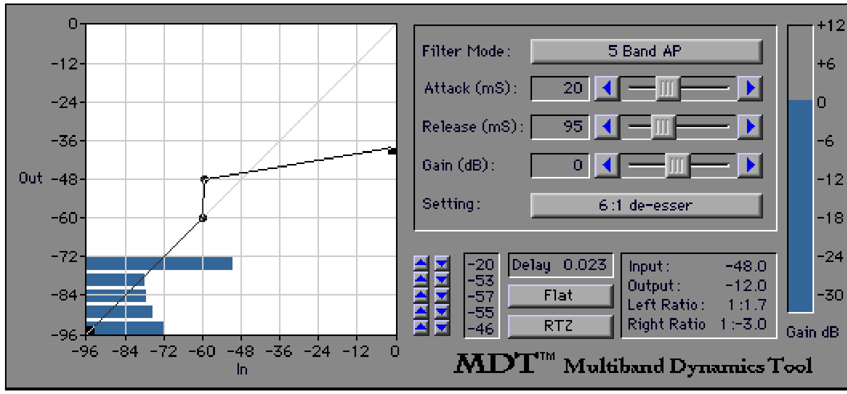
Notice how the High band is effected by compression and the other bands are offset so they get no compression.

4. **Use the Bypass button to compare the original with the compressed signal.**
5. **Press the High band's downward Input Offset Arrow until the display reads -14.**

The High band peaks will occur at approximately the 0 dB gain line (the diagonal). This setting should sound very similar in tonal balance to the original. Any high frequency peaks will be compressed at a 6 to 1 ratio.

Spectral Enhancer

1. **Press the High band's downward Input Offset Arrow until the display reads -20.**



The High band peaks are now occurring in the area of the I/O Curve to the left of the diagonal. The sound is brighter because the highs are getting amplified while still being compressed at a 6 to 1 ratio.

2. **Experiment with changing the tonal balance by moving the other bands into the compression segment of the I/O Curve.**
3. **Use the Bypass button to compare the original with the enhanced signal.**
4. **Stop playback.**

With the current setting of the I/O Curve, the louder a band gets, the more gain reduction it receives. The next example shows how to configure MDT as a dynamic EQ

Dynamic EQ

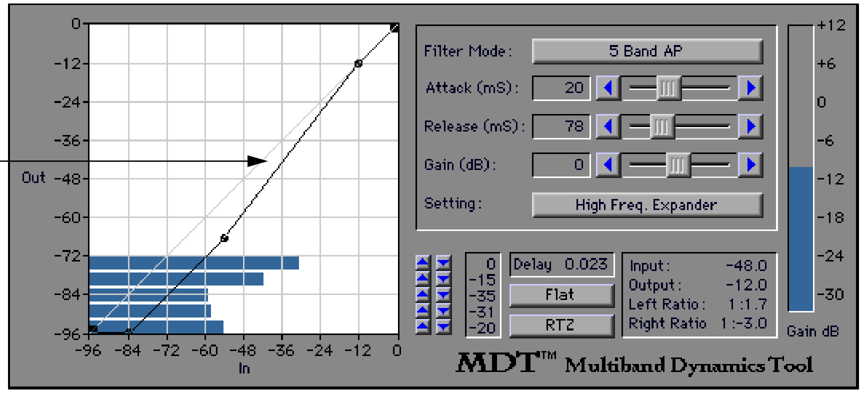
A dynamic EQ is a device that changes the amount of equalization it applies as a function of input level or some other changing parameter of the signal. MDT can be configured as a dynamic EQ as shown below.

1. **Click on the locator point labeled "Fetes 1" in the Transport Window.**
2. **Choose High Freq. Expander in the Setting menu.**

   The expansion ratio above the threshold is 1 to 1.5.

3. **Start playback.**

1 to 1.5 expansion ratio



Notice how the High and Hi Mid bands are effected by expansion and the other bands are offset so they get no expansion. The louder the input signal, the greater the amplification of the highs.

4. **Use the Bypass button to compare the original with the enhanced signal.**

5. **Experiment with changing the brightness of the sound by changing the offset of the High and Hi Mid bands.**

   The effect can range from subtle to gross depending on the ratio of the I/O Curve and the position of the band on the curve.

6. **Stop playback.**

End of Lesson 3

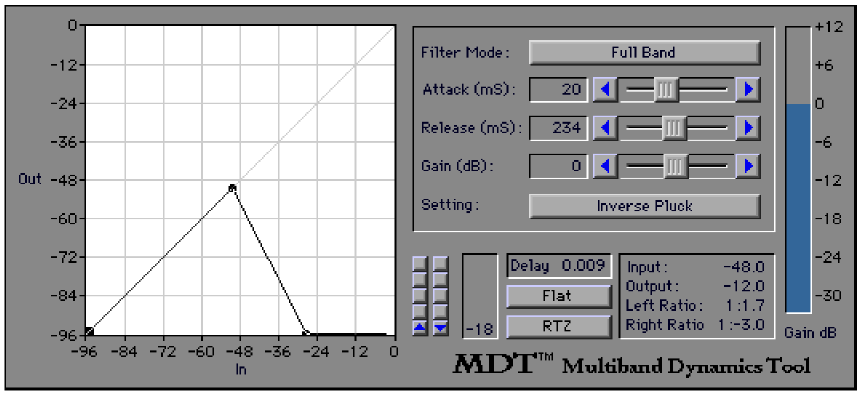
New Possibilities

You have probably thought of numerous applications for MDT that have not been mentioned in the Tutorial. We hope that you have learned enough about what's possible with MDT that you feel free to experiment and create ways to use it that are far beyond anything we expected.

Here is something to play with as a starting point that could produce unique results in a sound design application.

Inverted Gain Curves

In the region effected by the inversion, increasing input level decreases the output. Try out the Inverse Pluck setting with the region “pluck”.

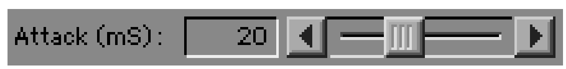


CHAPTER 4

MDT Controls

This chapter explains the detailed operation of every aspect of MDT's user interface. The items are organized in alphabetical order to make them easy to locate.

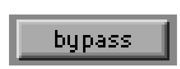
Attack Slider



The Attack slider controls the amount of time it takes for MDT to react to an incoming peak. The range of the control is from 0 milliseconds to 50 milliseconds. The current attack time is displayed in the window to the left of the slider.

Clicking on the arrows decrements/increments the attack time by one millisecond. Pressing on the arrows decrements/increments the attack time continuously. Press <option> to slow the speed at which the value increments. The thumb wheel can be dragged for making gross adjustments.

Bypass Button

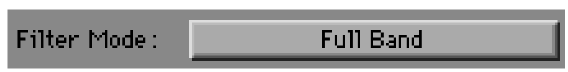


Clicking on the Bypass button bypasses the settings of MDT. This is useful for comparing the original soundfile to the processed version being previewed. Clicking again will revert to the current settings.

Delay Window

The Delay Window shows the amount of time in seconds that MDT will delay a track's sound output. This delay is introduced due to MDT's "look ahead" peak detection feature and processing delay. The amount of the delay introduced gets larger with the number of bands used. Tracks can be slipped to the left by the amount shown in the Delay Window to resynchronize their output with other non-processed tracks.

Filter Mode Menu

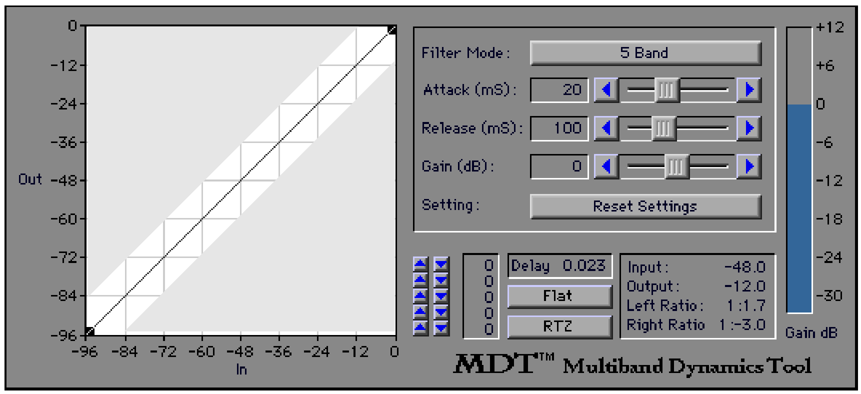


MDT uses digital filters to separate the energy of the input signal into multiple bands. The Filter Mode Menu contains the following choices:

- Full Band: No filters are used. The sound is processed in one band. The frequency response of this mode is 0 Hz to the Nyquist, ±0.0 dB with no passband ripple.
- 3 Band: The signal is divided into 3 equal bands, 3 octaves in width. The frequency response of this mode is 0 Hz to the Nyquist, ±0.0 dB with ±0.7 dB passband ripple.
- 5 Band: The signal is divided into 5 equal bands, 2 octaves in width. The frequency response of this mode is 0 Hz to the Nyquist, ±0.0 dB with ±0.3 dB passband ripple.
- 3 Band AP: The signal is divided into 3 equal bands with a special filter design optimized for the flattest pass band near unity gain. The frequency response of this mode is 0 Hz to the Nyquist, ±0.0 dB with ±0.0 to 0.7 dB passband ripple.
- 5 Band AP: The signal is divided into 5 equal bands with a special filter design optimized for the flattest pass band near unity gain. The frequency response of this mode is 0 Hz to the Nyquist, ±0.0 dB with ±0.0 to 0.3 dB passband ripple.

The AP (all pass) versions of the multiband filters are optimized for minimum coloration at gains around 0 dB. Because of hardware limitations, the AP mode filters have a limited gain reduction range of -12 dB. This means that the input signal can't be attenuated more than 12 dB without having its tonal balance seriously altered. In cases where more gain reduction is needed, the normal non-AP version of the filter will give the best result.

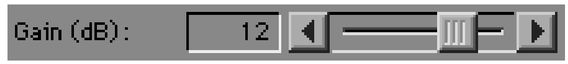
Placing the I/O Curve in the shaded parts of the Grid will cause unpredictable changes in the tonal balance when using AP mode filters.



Flat Button

Pressing the Flat Button equalizes the apparent input level of the bands displayed in multiband modes. MDT adjusts the offset of each band to equal the apparent level of the quietest band at the moment the button is pushed. Increasing the Release Time before clicking on the Flat Button will integrate the spectral content of a sound over time and produce a more uniform and realistic sounding result.

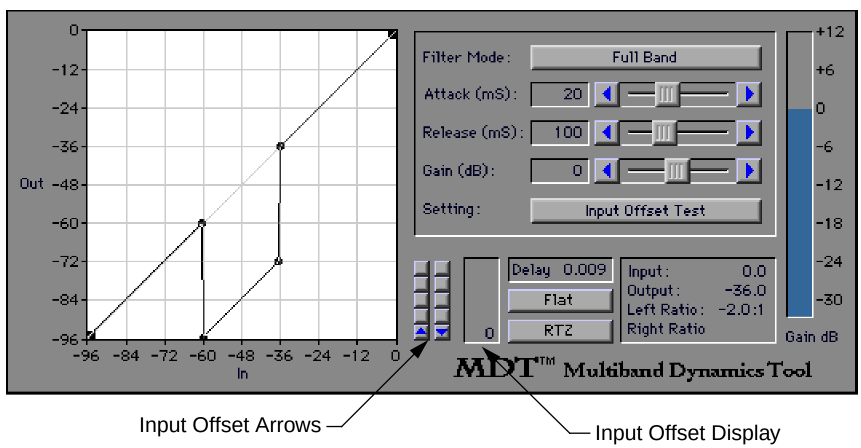
Gain Slider



The Gain slider adjusts the loudness of the signal after processing. The range of the control is from -24 dB to +24 dB. The current gain is displayed in the window to the left of the slider.

Clicking on the arrows decrements/increments the gain by one deciBel. Pressing on the arrows decrements/increments the gain continuously. Press <option> to slow the speed at which the value increments. The thumb wheel can be dragged for making gross adjustments.

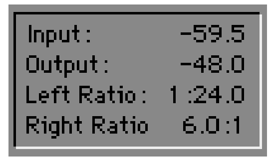
Input Offset Arrows



The Input Offset Arrows allow you to change the relationship of the input signal to the I/O Curve without actually changing the signal level of the input data. The placement of thresholds can be tweaked in this way. In multiband configurations, spectral bands can be independently adjusted. By adjusting the different bands to the same input level, coloration is reduced. Adjusting the different bands to completely separate sections of the I/O Curve allows independent compression ratios and levels for each band. The range of adjustment is from 0 dB to -96 dB.

The Input Offset Arrows normally increment in 2 dB steps. Press <option> then click or press on the arrows to increment is 1 dB steps.

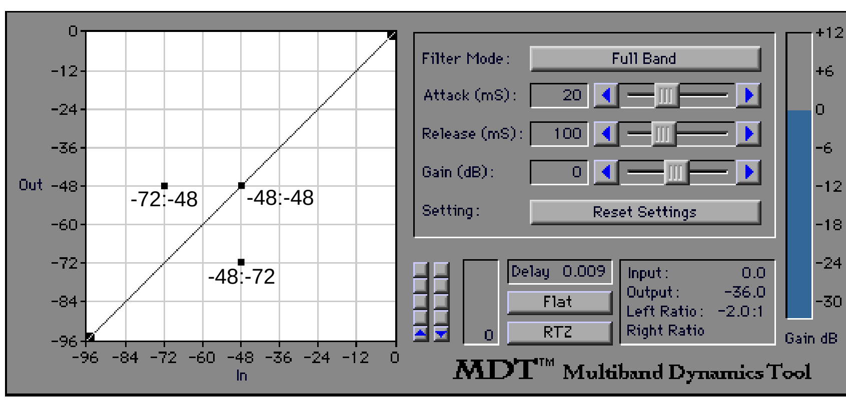
In/Out Display



The In/Out Display shows the position of the cursor when it is located inside the In/Out Grid. The units displayed are in dB below full scale. The In/Out Display shows the location of thresholds and terminators when they are being dragged.

When dragging a threshold, the In/Out Display also shows the compression or expansion ratios of the curve segments on either side of the threshold. Ratios expressed as N to 1 are compression ratios. Ratios expressed as 1 to N are expansion ratios. Ratios expressed as 1 to -N or -N to 1 are inverse gain curves which don't have equivalents in the analog world.

In/Out Grid



The In/Out Grid is the primary interface for controlling MDT.

All the possible points where the I/O Curve can exist are represented by the In/Out Grid. It is divided by a diagonal line that represents the points on the Grid that have input levels equal to output levels (e.g. -48:-48). Placing the I/O Curve on this diagonal will cause MDT to output the signal at the same level as it was input.

The points on the Grid which have output levels lower than their input levels (e.g. -48:-72) are located below the diagonal. Placing the I/O Curve below the diagonal will cause MDT to output the signal at a lower level than it was input.

Points on the Grid which have output levels higher than their input levels (e.g. -72:-48) are located above the diagonal. Placing the I/O Curve above the diagonal will cause MDT to output the signal at a higher level than it was input.

The I/O Curve can be moved during playback. The new setting takes effect when the mouse is released. Press and hold down <shift> before changing the I/O Curve to have MDT incorporate the changes while the adjustment is being made.

Release Slider

The Release slider adjusts the time it takes for MDT to return to its quiescent state after a peak comes through. The range of the control is from 0 milliseconds to 5,000 milliseconds. The current release time is displayed in the window to the left of the slider.

Clicking on the arrows changes the release time in small increments. Pressing on the arrows decrements/increments the release time continuously. Pressing <option> before clicking or pressing on the arrows decrements/increments the release time in 1 millisecond steps. The thumb wheel can be dragged for making gross adjustments.

RTZ Button

Clicking the RTZ (return to zero) Button zeros the Input Offset values shown in the Input Offset Window.

Setting Menu

The Setting menu contains the list of settings that are stored in the AnTares Preferences file in the System Extensions folder. Any number of settings can be saved in the Setting menu.

To save a setting;

1. **Choose Save Settings As… from the Settings menu.**

   A dialog box appears requesting a setting name.

2. **Type the desired name in the text box and click on Save.**

   The settings are now saved. These settings appear at the bottom of the Setting menu.

To recall a setting;

1. **Choose it from the Setting menu.**

   The settings are recalled as you saved them.

To delete the setting;

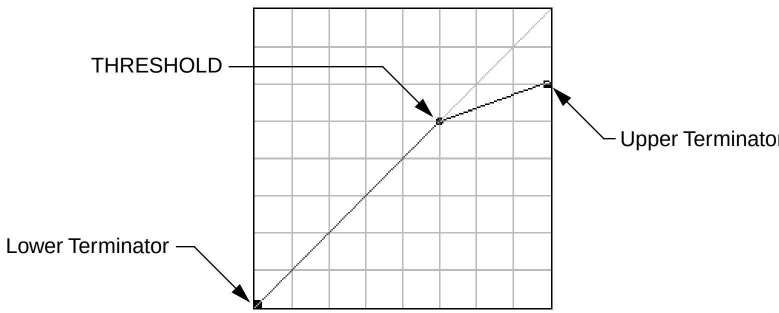
1. **Choose Delete Settings… from the Setting menu.**

   A dialog appears with the list of the settings in the menu.

2. **Scroll down to Test and select it by clicking on it.**
3. **Click on the Delete button.**

The setting is deleted from the menu. You may select multiple settings for deletion by shift-clicking or click dragging over the items to be deleted. To make a non-contiguous selection, press <command> and click on the items to select them. Note that the settings themselves remain on MDT until they are moved or another setting is chosen.

Thresholds And Terminators



Up to 30 thresholds can be placed on the In/Out Grid to "rubber band" the I/O Curve. The rules for using thresholds are as follows.

To:	Do This:
place a threshold on the curve,	click in the In/Out Grid.
display the location of a threshold	click on the threshold
move an existing threshold,	click and drag it to the new location.
delete a threshold,	press <option> and click on the threshold.

The lower terminator is "glued" to the In axis. It will always have an Out value of -96 dB. It can never exist to the right of a threshold, that is, have a higher input value than a threshold. This is to prevent the I/O Curve from having more than one output value for any given input value.

The upper terminator is glued to the Out axis. It will always have an In value of 0dB.

CHAPTER 5

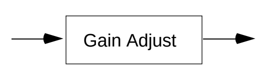
Theory of Operation

Mdt has been designed to provide enormous power and flexibility through the use of a simple user interface. However, since mdt is a new technology, some users may find it interesting or helpful to have a deeper understanding of the conceptual elements of mdt's actual processing algorithms.

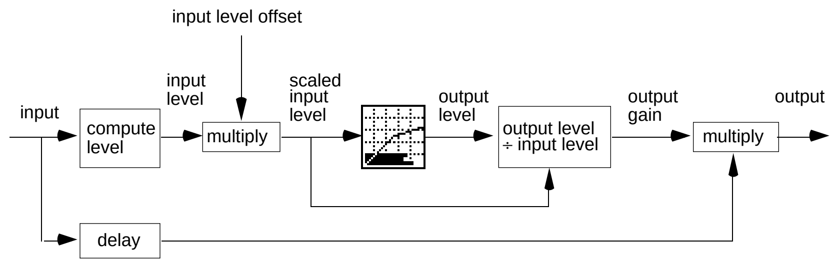
This chapter has been written to satisfy that need. The concepts behind mdt's processing algorithms are presented here in sufficient detail so that the behavior of mdt is precisely explained. To provide as much simplicity as possible, the various sections of this chapter describe the processing that occurs for monaural sound data. The last section of the chapter describes how mdt processing occurs for a stereo sound source.

The Gain Adjust Algorithm

The most basic computation in mdt is the Gain Adjust. In the various diagrams in this chapter, the Gain Adjust computation is represented by the symbol:



The purpose of the Gain Adjust computation is to perform the gain computation on one channel of data. This includes computing the signal level, scaling the level according to the user defined input level offset value, and computing and applying the final gain. The details of the Gain Adjust computation are given by the diagram:



The diagram above shows that as each sound sample is presented, a new input level is computed. The input level is a positive valued number between 0 (-96 dB) and 1 (0 dB). The input level is then multiplied by the user defined input level offset value to give the scaled input level. This value is presented to the user on the In/Out Grid as a horizontal bar. It is then used as the input setting on the I/O Curve to look-up the output level. The output level is divided by the input level to produce the output gain.

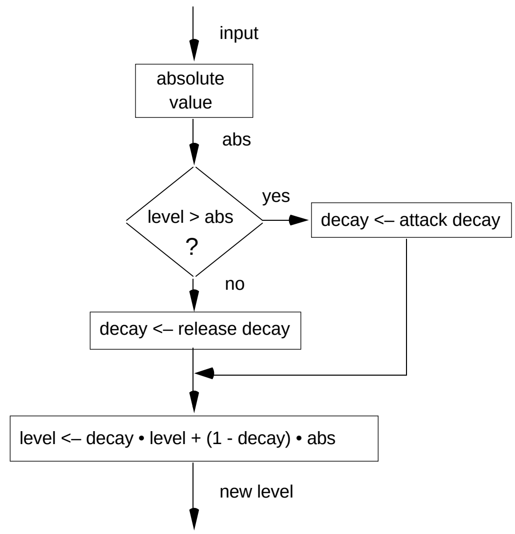
As an example of this processing, consider the case where the I/O Curve is a 45 degree 1:1 diagonal. In this case, the output level will always equal the scaled input level and the resulting output gain will always be unity. This means the sample output will be identically equal to the sample input.

Each incoming sound sample is also stored in a first in/first out (FIFO) buffer. To compute the output sound, samples are taken from the FIFO buffer. This is represented in the diagram above as delay. The amount of delay introduced by mdt is shown in the Delay Window.

For example, suppose the I/O Curve represents a compressor plus gain. Soft sound data would result in a high output gain. With no delay, if the sound were to become loud more quickly than the attack time, the high output gain would be applied to the loud sound resulting in output overflow and clipping. With a delay present, the output gain would be reduced before the delayed sound arrives, and a reduced output gain would be applied to the data, preventing overflow and clipping.

It is important to distinguish the use of the input level offset. This value only changes the displayed level, the corresponding output level look-up from the I/O Curve, and the gain computation. In this way, the input level offset serves only to change the position on the I/O Curve from which the gain is computed.

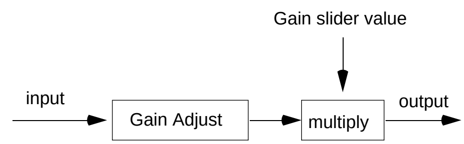
In the Gain Adjust computation, the compute level box depicts the computation of the input level from the sound samples. The compute level computation is depicted in the flowchart:



This flowchart shows that the absolute value of each sound sample is first compared with the previously computed level. If the absolute value is greater than the previously computed level, then a decay value corresponding to the attack time is used, otherwise a decay value corresponding to the release time is used. (The decay values are positive and slightly less than one, which result the proper decay or release times.) The decay value is then used to update the level value using the equation shown in the flowchart. When the attack time is shorter that the release time, the resulting level approximates the largest instantaneous amplitudes in the data. Other settings result in levels that are less than the largest instantaneous amplitudes in the data.

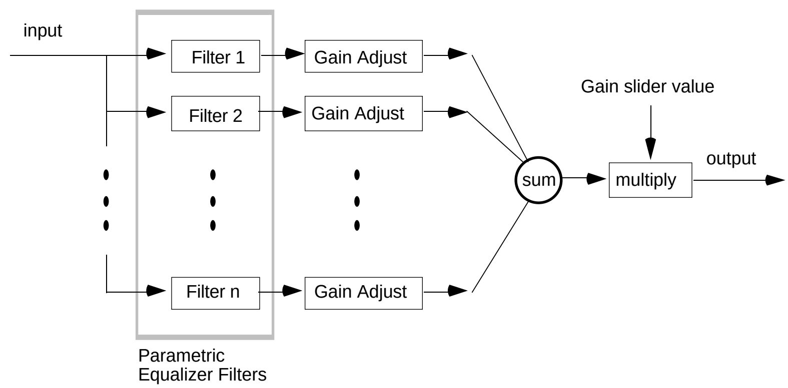
The “Full Band” Filter Mode

This is the simplest computation. In this filter mode, the sound data is input to the Gain Adjust. The output of the Gain Adjust is multiplied by the Gain slider value for the final output:



The “Standard” Filter Modes

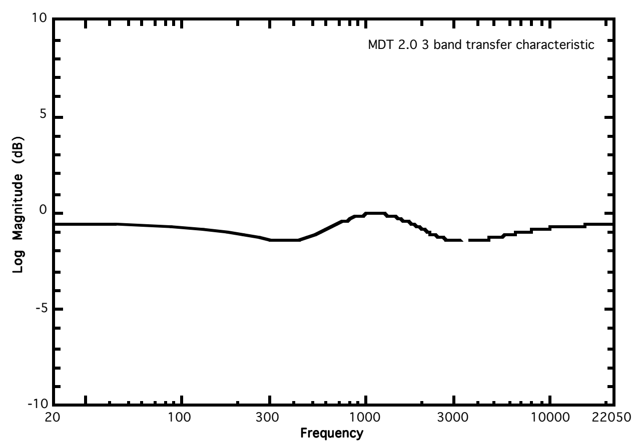
The following diagram depicts the “3 band” and “5 Band” filter modes:

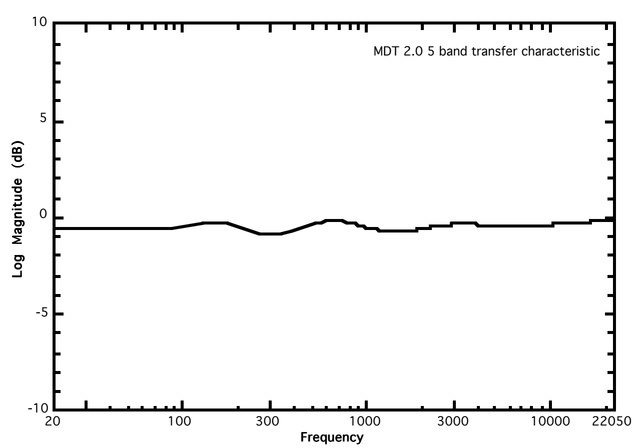


The input signal is first split into separate bands using parametric equalizer filters. Each of these filtered signals is processed by a separate Gain Adjust algorithm. Hence each band embodies separate input level computations and well as separate gain computations. The delays in each of the Gain Adjust computations are the same. Hence no phase effects are present from differing delays. After the Gain Adjust computation, the separate bands are summed together and multiplied by the Gain slider value to give the output.

The center frequencies of the parametric equalizer filter pass bands are 140, 1120, and 8960 Hz (3 octave spacing) for the 3 Band filters and 40, 160, 640, 2560, and 10240 Hz (2 octave spacing) for the 5 Band filters.

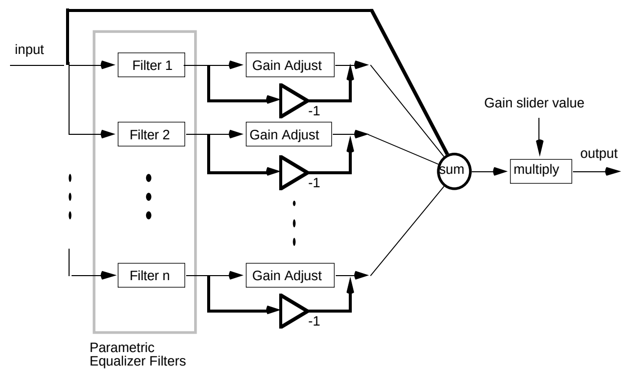
Consider the case where the I/O Curve is a 45 degree 1:1 diagonal. In this case, all output levels will equal the scaled input levels and the resulting output gains will be unity. This means the Gain Adjust output will equal the Gain Adjust inputs. The resulting overall output will then have the transfer function characteristic of the parametric equalizer. The graphs below show the frequency response of the multiband filters with unity output gain.



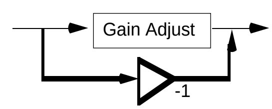


The “AP” Filter Modes

In some cases, the ±0.3 dB of passband ripple in the parametric equalizer filters of the standard filter modes are not desirable. The AP filter modes do not have these artifacts provided certain limitations can be met. The following diagram depicts the “3 band AP” and “5 Band AP” filter modes:



This diagram is the same as the standard filter mode diagram except for the additional data paths show as heavier lines. Consider the following sub-diagram:



The triangle near the -1 shows the signal is multiplied by -1. In the case where the I/O Curve is a 45 degree 1:1 diagonal, the output of the Gain Adjust is identically equal to the input. Therefore the output of the sub-diagram is zero. Referring now back to the larger diagram, the only input to the “sum” is directly from the overall input, in which case, the output of the AP filter mode is simply the Gain slider applied to the input with no parametric equalizer filters artifacts.

When the I/O Curve deviates from a 45 degree 1:1 diagonal, output occurs from the sub-diagram, and the characteristics of the re-combined signal depend on the phase characteristics of the parametric equalizer filters. Specifically, these filters are designed so that there is a zero degree phase shift in the center of each pass band. Consequently, a gain increase indicated by the I/O Curve results in a gain increase in the overall output and a gain decrease indicated by the I/O Curve results in a gain decrease in the overall output.

In this way, the AP filter modes are able to accomplish compression and expansion computations without artifacts from the parametric equalizer filters.

However, these effects are limited to the I/O Curve operating within plus and minus 12 dB of the 45 degree 1:1 diagonal. Operating above +12 dB of the diagonal causes the parametric equalizer artifacts to again dominate the output as they would in the non-AP case. This is not a serious problem. Operating below -12 dB is more of a problem: Because there is a zero degree phase shift in the center of each pass band, these frequencies will be handled properly. However, the phase changes in the crossover and stop band of each filter will cause the sound at these frequencies to not be correctly subtracted, leaving a significantly audible sound in the output.

Stereo Sound Processing

The left and right channels of stereo signals are processed completely separately except for a single exception: they both share the same levels. The signal level computation is described in the first section of this chapter using the compute level flowchart. In the case of stereo data, the data presented to the compute level algorithm alternates between the left and right channels: L, R, L, R, L, ...

As a result of this arrangement, the same gain will be applied to both channels of the stereo data at any instant in time.

Another result is that incorrect compression/expansion results will occur if the two channels contain differing material, or if the channels are not scaled relative to one-another. In these cases, the louder channel (over the period of the time scales involved) will dominate the computation.