# **Omnia Forza**

# Omnia Forza User Manual (V1.1.0)



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#### Introduction

#### What is Forza?

Omnia Forza is a new-from-the-ground-up multiband audio processing algorithm. In its initial release, it is being offered as a streaming audio processor delivered either as a Docker container (the version covered here) or as an optional processing style in the Telos Z/IPStream X/2.

Although brand-new algorithmically, Forza maintains the traditional Omnia processing structure of a wideband AGC, five bands of multiband AGC, five bands of multiband limiting, a bass clipper, and a final limiter, plus an ITU-R BS.1770-based loudness controller for streaming applications.

## A New Approach to Adjustment and Control

Forza takes into consideration the evolving definition of "broadcasting," recognizing that in some instances, there may be an experienced broadcast engineer with a solid understanding of audio processing at the helm. In other cases, it may be up to users with IT and networking skills but with minimal background or experience in traditional radio or audio processing to deploy and adjust Forza.

Accordingly, Forza's interface is designed so that less-experienced users can easily and intuitively produce a high-quality, professional-sounding stream with little or no adjustment by simply choosing one of the expertly-designed Omnia Presets.

More experienced users will find a suite of deceptively simple but very powerful controls that perform multiple adjustments in the background in order to customize Forza's sound to suit their particular tastes and needs, which can then be saved as user presets for easy recall and sharing.

### **Purpose-Built for Streaming**

As streaming audio listenership continues to grow, broadcasters and content providers are increasingly aware of the importance of delivering an enjoyable, high-quality experience for their audience, whether on a built-in player on the station's website or through a dedicated commercial streaming platform.

Though its processing structure is similar to that of other Omnia processors, Forza was created on a fresh sheet of paper to deal with the unique demands of streaming audio.

With no need to maximize RF coverage (a goal for AM), create dial-dominating loudness (applicable to both AM and FM), or deal with the FM pre-emphasis curve, it was possible (not to mention liberating) to completely re-imagine the way Forza's various processing algorithms behave.

At the same time, streaming introduces its own unique set of challenges and requirements, which Forza meets head-on.

It features tried-and-true Sensus® codec conditioning to dynamically optimize the audio for lossy codecs and minimize the audible artifacts that processing can sometimes reveal, especially at lower bitrates.

And, critically, it introduces a brand-new ITU-R BS.1770-based loudness controller which allows users to set an LKFS/LUFS loudness target to comply with <u>AES TD1008 guidelines</u> or meet the specific requirements of any streaming platform.

#### **Processing Architecture**

The diagram below illustrates the signal flow through the Forza processing algorithm from input to output.

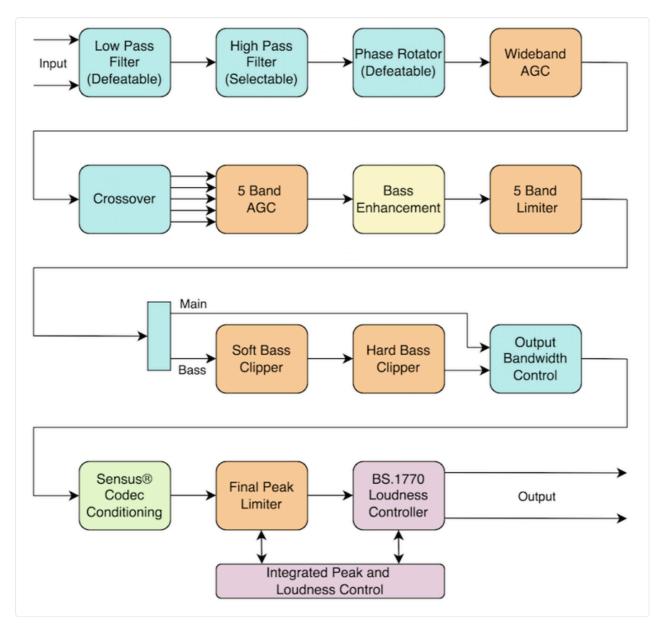


Figure 1 - Forza processing architecture

# Understanding Loudness for Streaming Audio

The concept of "loudness" as it applies to streaming audio is relatively new for many broadcasters. To an even greater degree, it is a foreign concept for content creators and IT specialists charged with managing audio streams who didn't come to the industry through a career in radio and television. To understand what loudness means in the context of streaming audio, a bit of a history lesson is in order.

#### **FM Processing and Loudness**

Audio processors have been an integral part of the transmission chain of FM radio stations for many decades. Initially, a simple compressor was used to ensure the audio stayed above the noise floor and a limiter was employed to prevent over-modulation and signal deviation beyond the legal limits.

It is well beyond the scope of this document to explore the admittedly fascinating and often entertaining history of FM processing. Suffice it to say that for broadcast engineers who have been in "the business" for a few years (decades?), the term "loudness" equates to one of the fundamental goals of FM processing and calls to mind the "loudness wars" of the late 1980s through... well, today, in some markets.

# Television Audio: Where Loudness Control Began

For those who have worked in the television broadcast world over the last ten years, TV audio went from being something that was just "there" ("Do we have sound? Good.") to paying very close attention to it when listeners rang the switchboard to complain about loud commercials and the FCC got involved.

By June 2009, all full-power analog television in the U.S. had made the transition to ATSC digital (DTV). This change introduced an interesting challenge for the audio portion of the

program.

In the NTSC analog format, audio was delivered by an FM carrier alongside the AM visual carrier. While some audio processing was typically used to help manage levels and prevent over-deviation, there were no inherent "loudness" issues with analog TV.

This was not the case with DTV. Despite the fact that the ATSC created a very comprehensive set of recommended practices for digital television audio (known as <u>ATSC A/85</u> and employing the <u>ITU-R BS.1770 metering recommendations</u>) television broadcasters still struggled to keep levels under control, creating the well-known issue of blaring commercials that were significantly louder than the normal program audio.

This became so problematic and annoyed television viewers so much that in late 2010, the U.S. Congress passed the bill that would become known as the Commercial Advertisement Loudness Mitigation Act, commonly referred to as the "CALM Act." It was adopted by the FCC in late 2011 and officially took effect in late 2012. In short, the CALM Act mandated that U.S. television stations adhere to the recommendations of ATSC A/85, or face repercussions from the FCC if enough viewers complained and a pattern of non-compliance could be established.

Meanwhile, in Europe, the European Broadcasting Union adopted its own recommendation known as <u>EBU R 128</u>. While there are important technical differences between ATSC A/85 and EBU R 128, the underlying goal of both is effective loudness management.

### **Loudness in Streaming Audio**

In the early days of streaming audio, many radio stations streamed their content to the internet, through a built-in player on their own website. Listenership was low, and streaming was treated as a "value-added" proposition by the sales department. Audio levels - so long as they didn't clip the input of the streaming encoder - weren't all that important. A listener would open a stream, adjust the volume, and be done. Many stations didn't bother to process their streaming audio at all, and if they did, they simply dusted off an old analog processor from the back room to provide some basic leveling.

In the past few years, streaming listenership and the number of available streams have increased significantly, elevating it from its previous humble status to one of significance both in terms of the number of listeners and revenue. In short, streaming is now a serious business and is being treated as such by broadcasters and individual content creators.

While most commercial stations still stream to their own websites, they also vie for listeners on dedicated streaming platforms and apps. With listeners now instantaneously switching between streams, variations in loudness have become apparent and annoying.

In an effort to deliver a more enjoyable listening experience for their subscribers, most of the major services delivering on-demand or streaming audio have their own individual requirements regarding loudness. In early 2022, the AES (Audio Engineering Society) released a technical document known as TD1008, "Recommendations for Loudness of Internet Audio Streaming and On-Demand Distribution," to address the subject and provide specific recommendations of its own.

TD1008 is specifically intended for distributors of on-demand audio files and internet audio streams, and we encourage you to read, study, and understand it. It is a relatively brief yet very thorough and thoughtfully-crafted document filled with good information about how loudness is measured, perceived, and managed.

In short - and with the understanding that many streaming audio platforms will have their own unique individual requirements - those wishing to adopt the recommendations outlined in TD1008 should deliver music content at -16 dB LUFS.

Because the spoken word is generally perceived as being comparatively louder when normalized to the same measured LUFS value, speech content should be delivered at -18 dB.

Because it is impractical to separately process music and speech in a real-time environment, TD1008 recommends splitting the difference at -17 LUFS, which falls within the +2 dB LU upper tolerance for the speech recommendation.

#### **Loudness Control in Omnia Forza**

Nearly any audio processor - from a simple wideband audio leveler to a full-featured product like Forza - is capable of effectively reducing the dynamic range of incoming audio sources to the extent that listeners won't be annoyed by major differences in levels or sudden loudness shifts.

However, even processed audio can vary in loudness by several dB depending upon the processing algorithms themselves, how they are set up and tuned, and the dynamic characteristics of the source material.

All things being equal, music that was recorded and mastered with heavy compression and limiting and is "dense" by nature will have a higher measurable RMS level and will sound louder than a recording that was more open and dynamic to begin with.

Sometimes these variations are noticeable but not annoying, and sometimes they are quite jarring. In either case, when passed through a LUFS loudness meter using the ITU-R BS.1770 measurement standard, they will usually indicate a measurable difference, sometimes by 3 - 4 dB.

In order to provide the type of longer-term loudness consistency required by most streaming platforms and recommended by AES TD1008, Forza includes an <a href="integrated loudness controller">integrated loudness controller</a> that allows users to set both a specific LUFS loudness target and a maximum peak level. It uses an ITU-R BS.1770 meter to measure the loudness of the processed audio and, when necessary, transparently make adjustments to ensure the average loudness levels match the desired target. The loudness controller can be disabled for applications where precise loudness control is not required.

# Licensing

### Licensing

Begin by clicking on the **Settings icon (B)**, then selecting the **AoIP tab (A)**.

#### Configuration

The **Configuration section (C)** contains the Node name field and the controls to release and refresh a license.

- **Node name:** Enter a "friendly" name for this Forza instance in the "Node name" field, then click the "Change" button to apply the new name.
- Activation: Though each Forza license can only be used in one place at any given time, it
  can be installed on multiple machines. Clicking the "Release" button temporarily disables
  the license without removing it so that it can be activated elsewhere. To re-enable the
  license, click on the "Refresh" button.
- License: To add a new license key, enter the license number in the License Field (D) then click the "Add License" button. Installed licenses along with their quantity and expiration dates are shown. To completely remove (un-install) a license, click the "Remove" button.

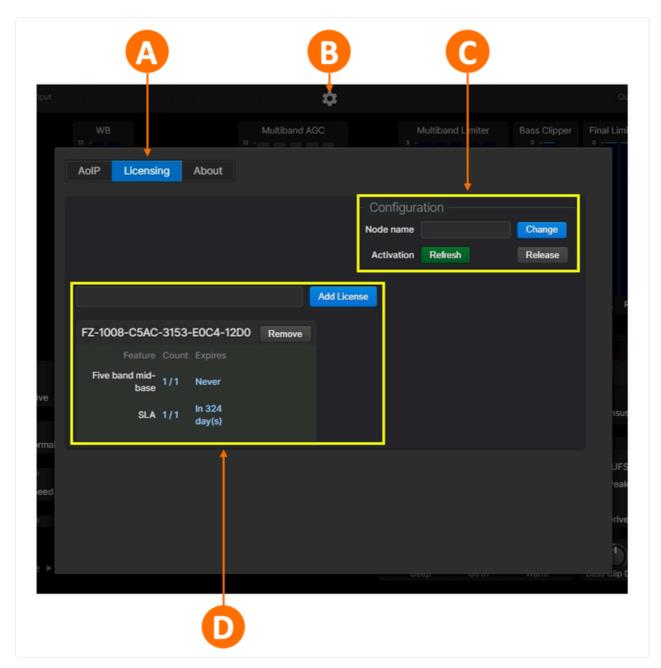


Figure 1 - Licensing menu

## **AoIP Setup**

## **AoIP Setup**

Begin by clicking on the **Settings icon (C)**, then selecting the **AoIP tab (A)**.

In the Basic AoIP section (B):

- Enter the Livewire channel of the audio source to be processed in the "Receiver" field.
- Enter the Livewire channel carrying the processed audio output in the "Sender" field.
- Choose the packet time in the "P-Time" dropdown menu (5ms or 1ms).

Information in the **Advanced section (D)** will automatically populate to match the settings of the AoIP network to which the host computer is connected and/or to the standard default Livewire values when used on a Livewire network. Unless you are using a non-Livewire compatible AES67 stream or are instructed to change these settings by Telos support, we recommend leaving them as-is, with two exceptions as outlined below.

#### **Link Offsets**

It may be necessary to adjust the Standard Link Offset (for Livewire standard streams) and AES Link Offset (for AES67 streams) settings, which specify the amount of available jitter buffering (in samples) available to absorb the sum of:

- Packet jitter caused by the network.
- Transmission jitter from the upstream source machine.
- Clock/execution jitter on the local machine running Forza.

On a robust network, packet jitter may be less than 0.5ms. Transmission jitter can range from less than 0.5ms (for a hardware node) up to 20-30ms (for a Windows machine with a sub-optimal configuration). Clock jitter on the local machine should be less than a few ms on a machine configured with a low-latency kernel and the Forza container operating properly

with real-time privileges. Accordingly, 10-20ms is a safe starting point, which translates to 480-960 samples.

Our recommendation is to set both link offsets for 960 samples and use the override offset to increase the buffer size as needed.

#### **Buffer Mode**

The buffer mode setting needs to match the type of clock configured with the EX\_AOIP\_CLOCK environmental variable (0 for PTP clock, 1 for Livewire clock).

- If the master clock is set to Livewire, the buffer mode **must** be set to Syntonus (Livewire).
- If the master clock is set to PTP, either Syntonus (Livewire) or Synchronous (AES67) may be used.
- If strict conformance to AES67 is required, Synchronous (AES67) should be used.

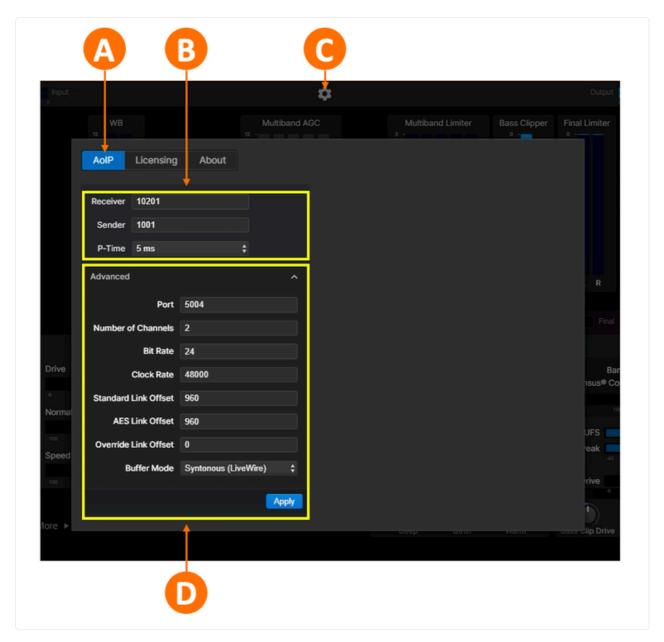


Figure 1 - AoIP setup menu

## **Forza UI and Controls**

#### **Connecting to Forza**

Forza's user interface is web-based, and any mainstream browser (including Google Chrome, Firefox, Safari, or Microsoft Edge) will work. As the UI is based on HTML5, a larger monitor will provide a better experience. It may be necessary to adjust the zoom level within the browser window to make the entire UI visible without the need to scroll on smaller screens.



Figure 1 - Forza user interface

To access Forza, enter the IP address of the host computer followed by a colon and the port number (3000) as follows, where "x" represents the IP address: xxx.xxx.xxx.xxx.3000.

## **Using the Dials and Sliders**

Forza's processing controls are adjusted either by dials or by vertical or horizontal sliders.

Use a mouse or touchpad to position the cursor directly over the control you wish to adjust. Left-click and hold, then drag up and down to make adjustments to dials and vertical sliders, or drag left and right to adjust horizontal sliders.

You may also place your cursor over the control, then scroll or use the up/down arrows on the keyboard. Or, simply left-click anywhere in the control's range to affect an immediate change.

A box displaying a specific value will appear briefly while you are making the adjustment.

Controls whose values are always displayed can be adjusted by placing the cursor over the box, left-clicking, then scrolling, using the up/down arrows on the keyboard, or typing in the desired value followed by the Enter key.

#### **Presets**

#### **Omnia Presets**

Omnia Presets are combinations of settings that have been carefully designed by our processing architects with a unique sonic outcome in mind. If you find one that meets your needs, it can be used as-is without modification.

Or, they can be used as foundational starting points to create your own signature sound by adjusting various controls in the user interface, then saving the changes as a custom user preset.

Each Omnia Preset offers a different combination of density, texture, perceived loudness, and spectral balance.

The first ten are general-purpose presets for music and are arranged with the most neutral, least processed-sounding options at the top of the list.

The last three are format-specific and designed especially for Classical, Talk, and Sports as these programs typically need a more specialized approach to processing.

- Natural Natural is an open-sounding preset that preserves transients and overall dynamic range. It provides a neutral but consistent frequency balance without noticeably boosting any part of the audio spectrum, making it a good starting point for customization.
- **Bright** Bright is similar to Natural in texture, but with more clarity in the presence region of the spectrum.
- **Clear** Clear is similar to Bright but with more prominent highs to provide extra sparkle and brilliance, and a bit more bass for balance.
- Smooth Compared to Natural, Smooth offers stronger bass and more spectral consistency.
- **Full** Compared to Smooth, Full has a bit more density with more prominent bass, smoother highs, and a warmer sound overall.

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**Forward** - Forward has a similar texture to Full but with more bass, more brightness, and less overall warmth.

- Slam Slam is similar to Forward in the mid-range but with a strong, hard-hitting bass texture and a brighter top end.
- **Punchy** Punchy offers the warmth of Full but with crisper highs. It delivers a more polished sound without sounding overly processed.
- Dense Dense has a brighter, more compressed "radio-like" sound and renders a very consistent spectral output regardless of the frequency balance of the source material.
- **Hot** Like Dense, Hot has a processed sound overall but with stronger bass, less warmth in the mid-bass, and a noticeable presence boost.
- **Bold** Bold is similar to Dense but with noticeably more bass and more prominent highs. It offers the most processed sound of all the Omnia Presets.
- Classical Classical is specifically designed to transparently yet effectively deal with source material that has very wide dynamic range. It provides gentle re-equalization but stays relatively true to the spectral balance of the source material. It also employs a longer integration time in the BS.1770 loudness leveler than other Omnia presets to allow more overall deviation from the LUFS target.
- **Talk** Talk is optimized for voice and makes use of the phase rotator to help smooth out asymmetrical peaks. It uses fast multiband attack and release rates to help control the widely varying levels typical of multi-mic studios and call-in programs.
- **Sports** Like Talk, Sports is optimized for voice but with slower attack and release rates and more active wideband and multiband gating to prevent crowd and background noise from increasing during announcer pauses.

## **Choosing an Omnia Preset**

Click on one of the available Omnia Presets in the list (A) to highlight it, which will immediately apply your selection. The Omnia Preset name will also be displayed in the Current window (B).

Note - Regardless of the Omnia Preset you select, most of the controls in the Forza UI will be at their center position to allow adjustment in either direction (the exceptions being the Bass controls). Note that the actual "behind the scenes" values associated with the controls may be different from one Omnia Preset to another.

Also note that the Bandwidth, Sensus® Conditioning, LUFS target level, and Peak level are higher-level I/O settings that do not change when a different Omnia Preset is selected.

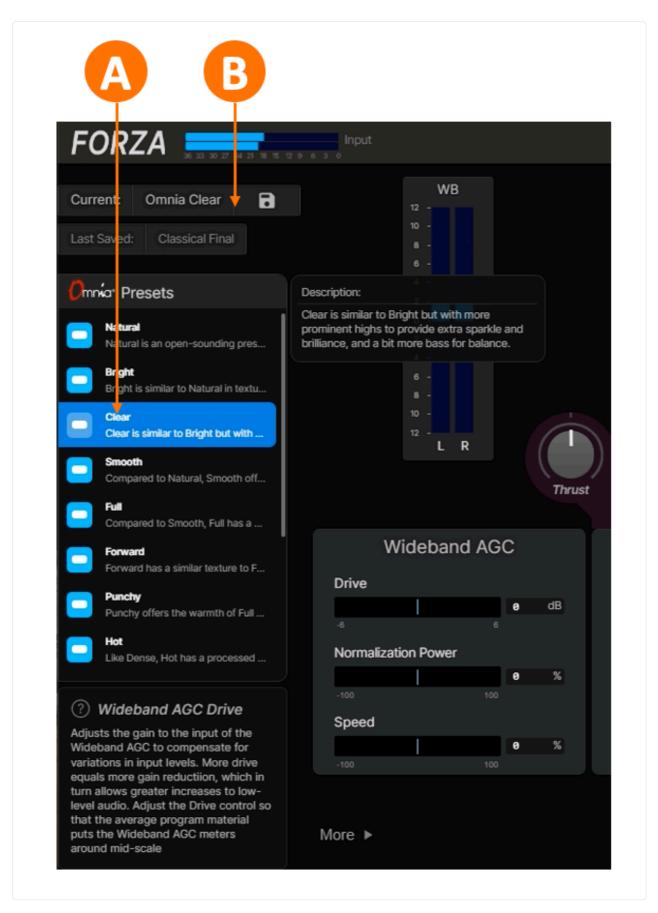


Figure 1- Choosing an Omnia Preset

## **Making Adjustments**

As soon as you adjust one of the processing controls, the Current window will change to a "MODIFIED" status (C) indicating that some change has been made to the original Omnia Preset but not yet saved.

Note - Please see the sections on <u>Processing Controls</u> and <u>Getting the Sound You Want</u> for detailed information about adjusting the processing.

Any **altered controls (D)** (which include Thrust along with Wideband AGC Drive, Normalization Power, and Speed in Figure 2 below) will visibly indicate that they have been changed and to what degree.

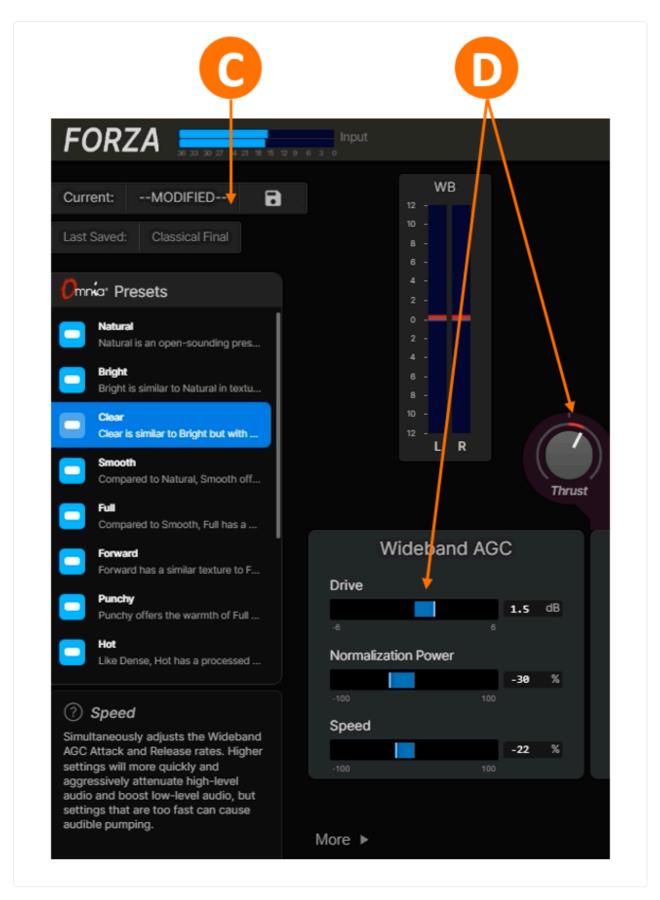


Figure 2 - Modified Omnia Preset

Clicking on the same Omnia Preset from the list will immediately discard any unsaved changes and return you to the original Omnia Preset. Clicking on any other Omnia Preset will discard unsaved changes and switch to the new preset.

## Saving Changes as a User Preset

To save your changes and create a new user preset, click in the box with the **diskette icon** (E) and choose **Save As...** (F) from the dropdown list. Give your new preset a name, then click "OK."

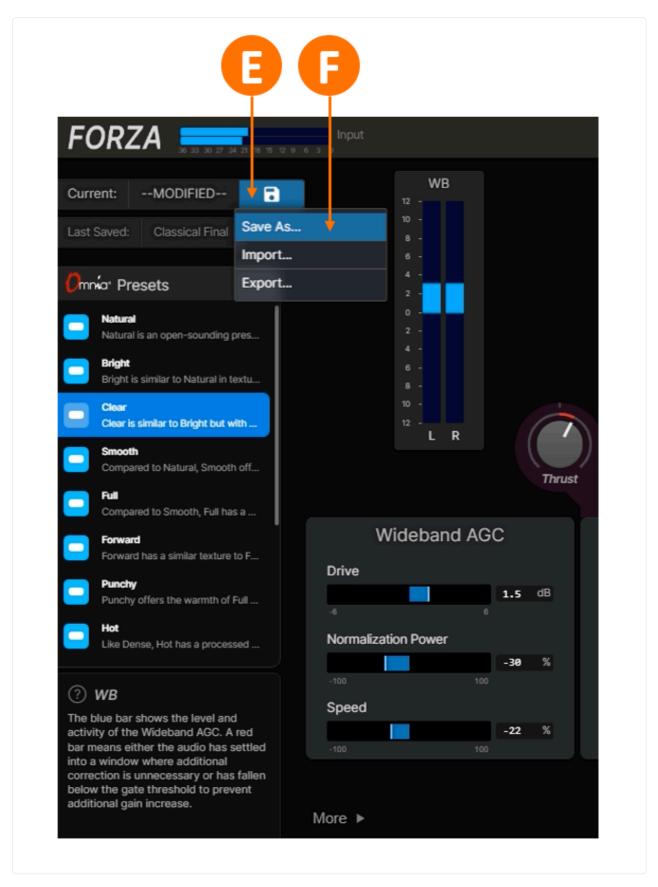


Figure 3 - Saving a user preset

The **Current window (G)** will now display the name of your new preset, which is duplicated in the **Last Saved window (H)**.

Note - When recalling a user preset, the Omnia Preset upon which it was based will be highlighted in the Omnia Presets list, providing a useful reminder of your starting point. Be sure to look at the Current window to determine if the highlight is referencing an unmodified Omnia Preset or the basis for a saved user preset.

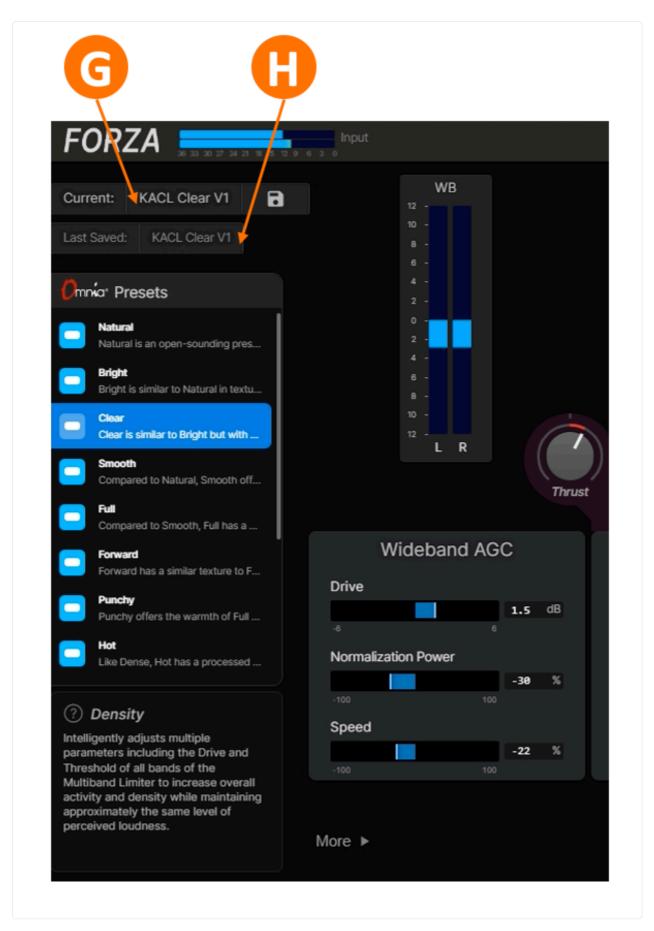


Figure 4 - Current and last saved presets

Once one or more user presets have been saved, clicking in the **Current window (I)** will reveal the user preset list. Clicking on any of the presets will immediately take it to air. To delete a user preset, click on the **trash can icon (J)** next to the preset name.

Note - Presets loaded in the Current or Last Saved position cannot be deleted and will not have a trash can icon next to their name.

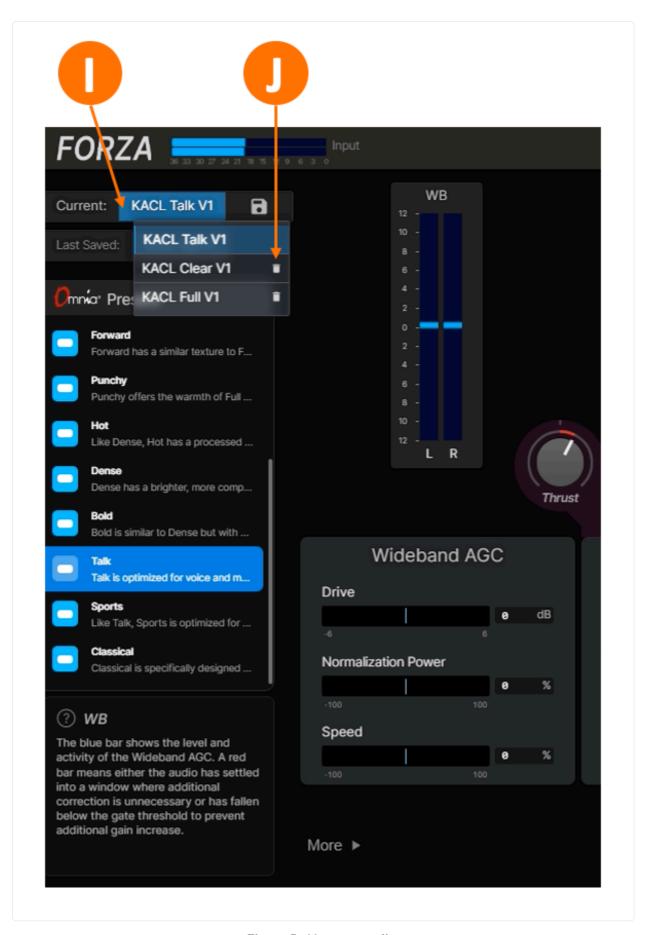


Figure 5 - User preset list

## **Importing Presets**

The import feature allows Forza user presets that were created on a different machine to be saved to your local host machine and uploaded to your Forza instance.

Click on the **diskette icon (K)**, then click on **Import Preset (L)**. A dialog box will open; navigate to the location of the desired preset, highlight it, then click "Open" (or simply double-click the preset file). A confirmation dialog box will appear. Once confirmed, the file will be uploaded and available from your user preset list.

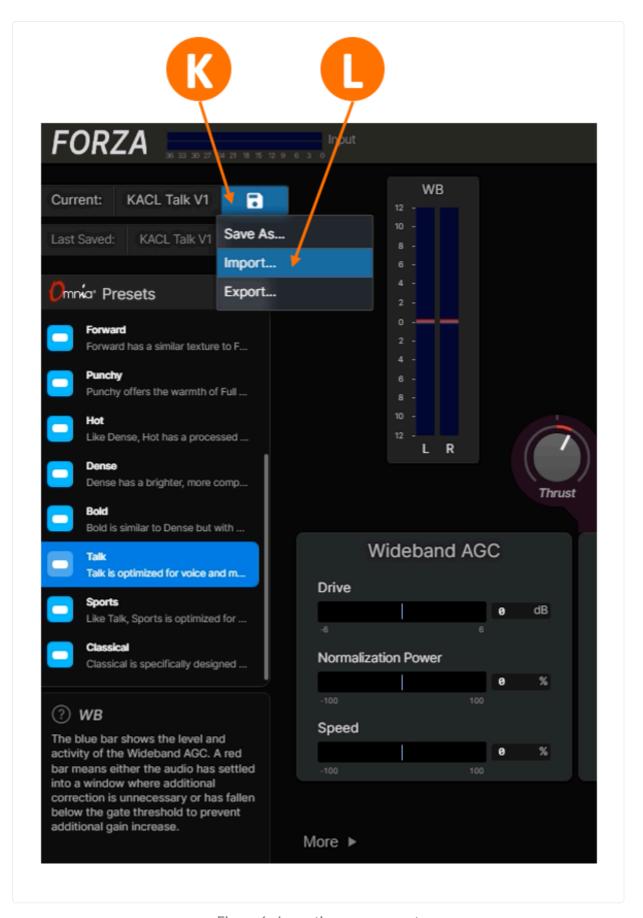


Figure 6 - Importing a user preset

#### **Exporting Presets**

The export feature allows you to save a Forza user preset you have created on your local machine either as a means of saving and safeguarding a copy for your own use later or for sharing and uploading to a different Forza instance.

Click on the **diskette icon (M)**, then click on **Import Preset (L)**. A dropdown list of available user presets will appear. Select the preset you wish to export, then click the "Export" button. The file will be downloaded to your Downloads folder (or whatever folder on your computer has been designated as the default for downloads).

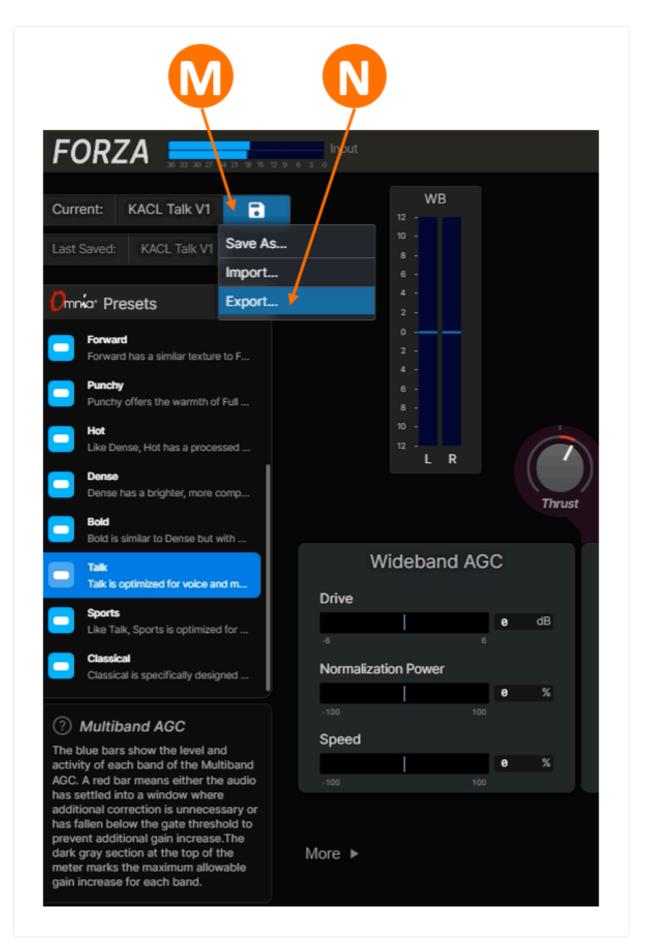


Figure 7 - Exporting presets

# **Processing Controls**

Forza's controls can be roughly divided into six sections:

- **Texture:** High-level "smart controls" that alter overall density and texture by adjusting multiple background parameters with a single knob or slider.
- Wideband AGC: Adjusts the drive, density, and speed of the Wideband AGC.
- Multiband AGC: Adjusts the density and spectral balance of the Multiband AGCs.
- Bass: Contains the controls to adjust the amount of texture of the bass, including bass enhancement tools and bass clipper drive.
- Multiband Limiter: Adjusts the density and spectral balance of the Multiband Limiters.
- **Final:** Contains the controls pertaining to the output lowpass filter, Sensus® codec conditioning, final drive level, and ITU-R BS.1770 loudness leveler.

A full explanation of the controls in each section follows.

#### **Texture**

Forza features three powerful high-level controls that offer a quick way to alter the overall processing texture and density by changing multiple parameters in the background with a single adjustment: Thrust, Activity Balance, and Limiting Balance.

- Thrust: Adjusting the Thrust control is the easiest way to achieve more or less overall
  processing without having to adjust individual settings.
  - Thrust affects Multiband AGC drive, Multiband Limiter drive, the Bass Clipper, and the Final Limiter.
  - Small adjustments to the Thrust control will have more of an effect on the Multiband AGCs, while larger adjustments yield more changes in the Multiband Limiters.
- Activity Balance: The Activity Balance slider controls the amount of processing taking place in the Multiband AGC section compared to the Multiband Limiters.
  - Sliding the control toward "Compression" works the Multiband AGCs harder in comparison to the Multiband Limiters. Sonically, this tends to provide a more homogenized and compressed sound overall while allowing peaks and transients (such as the attack of a snare drum) to pass through. This can help create a more dynamic and "punchy" sound, especially with "peaky" material (think of well-recorded music such as Steely Dan).
  - Sliding the control toward "Limiting" relaxes the multiband AGCs and transfers more
    of the heavy lifting to the Multiband Limiters. This approach can make music sound
    less "squashed" and compressed overall but at the expense of attenuating transients.
- **Limiting Balance:** Though the Multiband Limiters and Final Limiter are designed to work together, adjusting the balance between them can have a very audible effect on the texture of the processing.
  - Sliding the control toward "MB" relies more on the Multiband Limiters. This approach
    ensures that high peaks in one band won't affect the audio outside of that particular
    frequency range, but can cause unwanted density or spectral compromise in the
    Multiband Limiter section.
  - Sliding the control more toward "Final" relaxes the Multiband Limiters and drives the
    Final Limiter harder. This avoids building up too much multiband density and the
    "busy" sound that can result, but controlling peaks in one frequency range can
    sometimes cause audible artifacts elsewhere in the spectrum. Driving the final limiter
    too hard can also flatten transients to the point of making music with a high peak-toaverage ratio sound lifeless, so adjust with care.

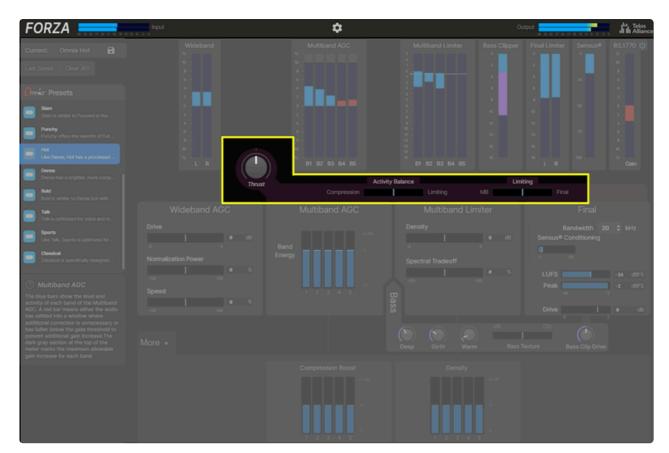


Figure 1- Texture controls

## Wideband AGC

Source input audio levels are often very inconsistent. Songs may have been imported into an automation system at different levels over the years. Modern music is mastered at much higher levels than material from 20 or 30 years ago. Live sports and talk formats are a challenge to mix well due to multiple microphones and phone calls.

The Wideband AGC automatically compensates for those variations and also keeps the levels in downstream processing stages consistent. It is often described as a "slow hand on the fader."

- **Drive:** Adjusts the gain to the input of the Wideband AGC. More drive results in more gain reduction which in turn allows low-level audio to be increased more. Adjust the Drive control so that the average program material puts the Wideband AGC meters around 0 (mid-scale).
- Normalization Power: Simultaneously adjusts the Ratio and Threshold of the Wideband AGC. Higher settings provide more source-to-source consistency and more overall Wideband AGC activity.
- **Speed:** Simultaneously adjusts the Wideband AGC Attack and Release rates. Higher settings will more quickly and aggressively attenuate high-level audio and boost low-level audio, but settings that are too fast can cause audible "pumping."

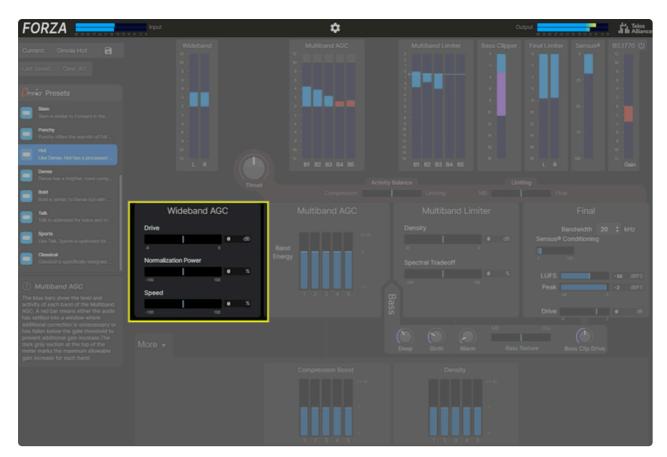


Figure 1 - Wideband AGC controls

## **Multiband AGC**

The Multiband AGC section has two primary functions.

The first is to build RMS energy which increases perceived loudness and delivers a more "processed" sound when compared to the original source material.

The second is to provide spectral consistency to the processed audio. This is often called the "sonic signature" of a stream or station. Songs that are bass-shy (think 70's rock) receive a boost in the lower frequencies. Songs that are overly bright (think synth-heavy 80's pop) get their top-end smoothed out.

- Band Energy (B1 B5): The primary method of achieving a particular frequency balance of the Multiband AGCs. Higher settings result in higher output levels from that band.
- **Per-Band Compression Boost:** Available by expanding the "More" menu, these controls adjust the drive and activity of the Multiband AGCs on a per-band basis.

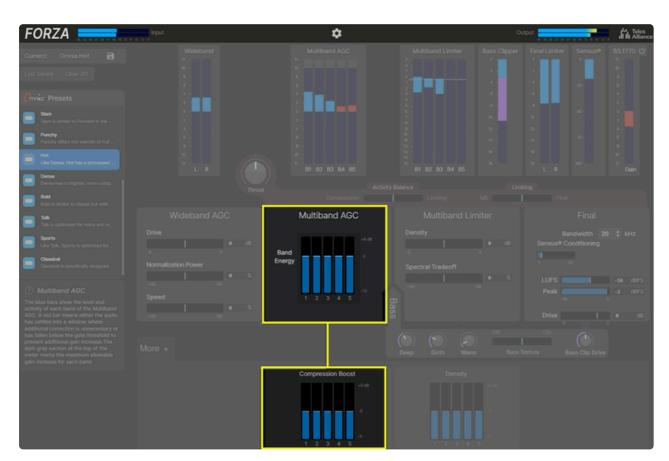


Figure 1 - Multiband AGC controls

## **Bass**

The controls in this section are used to tune the level and texture of the bass.

The rule of "a little goes a long way" applies here, so make small adjustments and listen carefully over time using a variety of source material and a variety of speakers. Consider how your target audience will listen to your content. Boosting very low frequencies may sound good on higher-end home and car stereos, but tablets, mobile phones, and smart speakers are often incapable of reproducing these frequencies.

Bear in mind that changes to these controls can affect downstream processing stages and introduce audible artifacts in higher frequency ranges if not adjusted properly.

- Deep Bass: Applies a static boost to the lowest bass frequencies.
- **Girth Bass:** Adds subtle low-mid to upper-bass detail. The effect is program-dependent and is similar to the "Phat Bass" control found in other Omnia processors.
- Warm Bass: Applies a static boost to mid and upper bass frequencies.
- Bass Texture: Determines how much bass peak control is performed by the first band of the multiband limiters vs. the bass clipper. Settings toward "MB" will result in more bass limiting and a softer texture, while settings toward "Clip" will allow more peaks through to the bass clipper for more "slam" on low-frequency transients.
- Bass Clipper Drive: Adjusts the drive to the Bass Clipper. Higher settings can add "punch" and "slam" to the bass, but too-high settings can introduce audible distortion on some material.

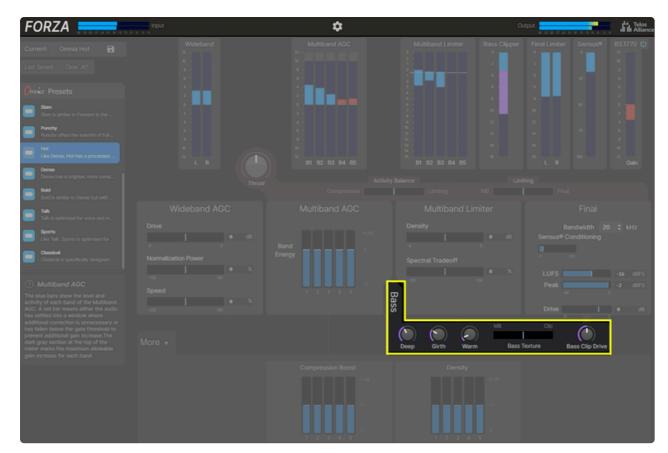


Figure 1 - Bass controls

## **Multiband Limiter**

The Multiband Limiter section has two primary functions.

The first is to control audio peaks and transients in individual parts of the audio spectrum.

The second is to provide additional density and define the texture of the processed audio.

- Density: Intelligently adjusts multiple parameters including the Drive and Threshold of all bands of the Multiband Limiter.
  - Higher settings increase the overall activity and density while maintaining approximately the same level of perceived loudness.
- Spectral Tradeoff: Depending upon the nature of the program material (and of course various settings) traditional multiband limiters can significantly alter the "EQ curve" created by the multiband AGCs that precede them in an effort to produce additional density or better control peaks and transients to keep from over-working any downstream processing stages.

Forza's multiband limiters can operate in this fashion, but also offer the ability to apply a varying degree of spectral correction to bring the balance more in line with what has been created in the multiband AGCs.

The Spectral Tradeoff control adjusts how much spectral re-alignment occurs.

- Moving the control to the right mimics the behavior of a traditional multiband limiter, allowing more overall spectral variation to achieve increased density and a cleaner sound but with more potential to change the frequency balance set by the Multiband AGCs.
- Moving the control to the left reduces the amount of allowable spectral variation and better preserves the balance set by the Multiband AGCs but at the expense of density or cleanliness with some program material.
- **Per-Band Density:** Available by expanding the "More" menu, these controls adjust the Drive and Threshold of the Multiband Limiters on a per-band basis.

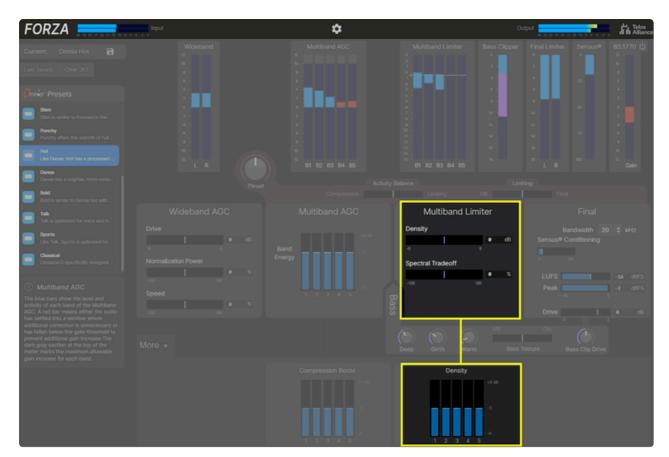


Figure 1 - Multiband Limiter controls

## **Final**

The Final menu contains the controls that manage Forza's final processing stages and output level.

- Bandwidth: Sets the cutoff of the output lowpass filter. Streaming audio typically relies upon the use of lossy codecs that discard bits of data in order to reduce the required bandwidth. The effects of such data compression, especially when using less capable codecs and lower bitrates, can become audible in the higher frequencies. Adjusting the Bandwidth control to filter out frequencies above a certain threshold can help minimize such artifacts, though at the expense of dulling the sound.
- **Sensus Conditioning:** Sets the frequency range of the dynamic conditioning applied to the audio to help mitigate audible coding artifacts, much like the "Sensus Frequency" control in the Omnia.11 processor.
  - Higher settings increase the available range to include lower frequencies, allowing the algorithm to react to more of the audio spectrum.
  - Lower settings decrease the range, allowing the algorithm to work on a narrower portion of the audio spectrum more limited to higher frequencies.
  - Setting this control to "0" disables codec conditioning altogether.
- LUFS Target: Sets the desired loudness target for the ITU-R BS.1770 LUFS/LKFS loudness controller. The main Forza algorithm provides very consistent output levels, but variations in the energy and texture of the source material can still cause variations in excess of what is tolerable on a particular streaming platform or service. The loudness leveler measures and adjusts the output level to ensure a consistent "set and forget" long-term loudness target.
  - Note The loudness controller can be disabled entirely where strict long-term loudness management is not required by clicking on the "Power" button at the top of the BS.1770 meter.
- **Peak Target:** Sets the highest allowable peak value to prevent overshoots which can cause distortion at the output of Forza or at the input to downstream encoders.
- **Final Drive:** Adjusts the drive to the Bass Clipper and the Final Limiter.
  - Higher settings will result in increased loudness and density, but care must be taken
    when making adjustments here since the affected parameters are at the very end of
    the processing path.

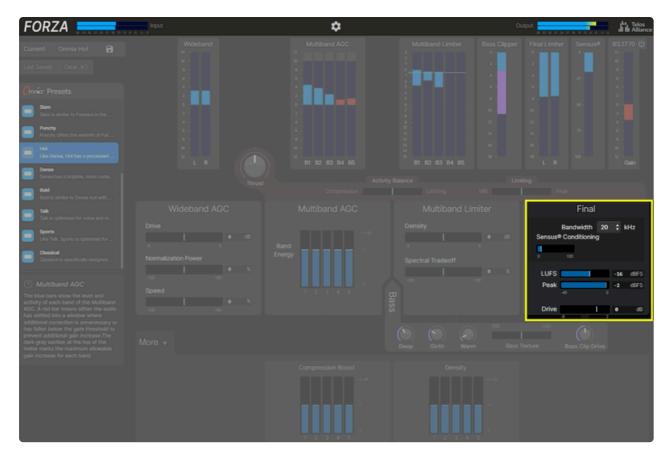


Figure 1 - Final controls

## **Forza Meters**

### **Input and Output Meters**

The level of the unprocessed source audio is displayed on the Input Meters. The Output Meters show the level of the processed audio.

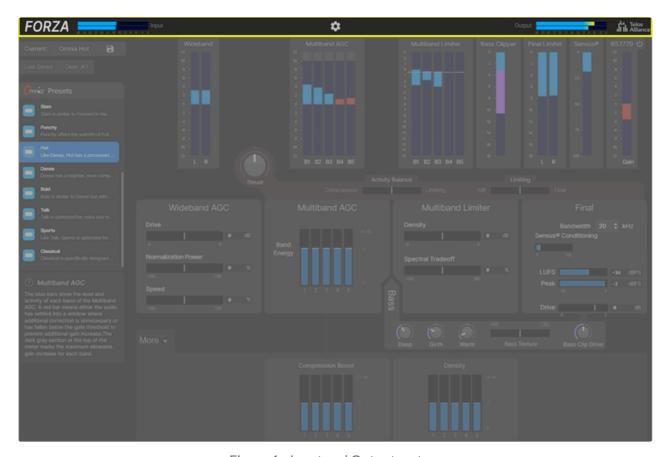


Figure 1 - Input and Output meters

#### Wideband AGC Meter

The blue bars show the individual level and activity of the left and right channels of the Wideband AGC.

If the audio has settled into a window where no additional correction is necessary, the meters will turn red and stop moving. When the audio falls below the gate threshold - the point at which any additional increase in gain is undesirable - the meters will remain red and begin moving toward zero.

Note - The range of the "do nothing" window, the gate threshold, and the speed at which the AGC returns to zero once gated are determined by inherent settings in each individual Omnia Preset.

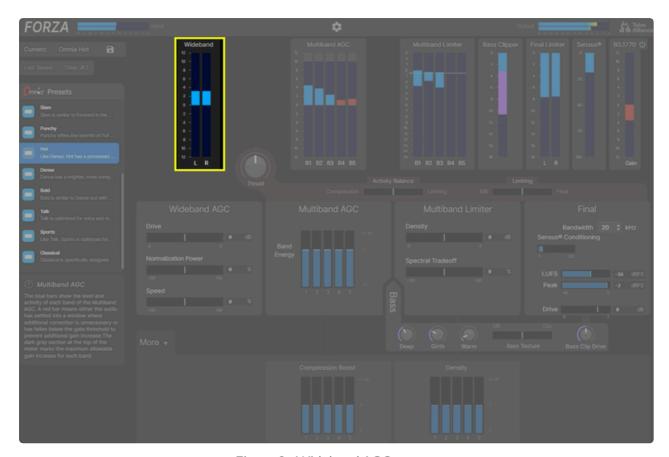


Figure 2 - Wideband AGC meters

#### **Multiband AGC Meters**

The blue bars show the level and activity of each of the five bands in the Multiband AGC. When the audio falls below the gate threshold, the meters will turn red and begin moving toward 0.

The dark gray section at the of the meter marks the maximum allowable gain increase for each band.

Note - The gate threshold, the speed at which the AGC returns to zero once gated, and the maximum allowable gain for each band are determined by inherent settings in each individual Omnia Preset.



Figure 3 - Multiband AGC meters

### **Multiband Limiter Meters**

The blue bars show the level and activity of each band of the Multiband Limiter.

The floating white bar across all bands indicates the amount of spectral compensation being applied as set by the <a href="Spectral Tradeoff">Spectral Tradeoff</a> control. More variation in the blue bars above this line indicates the limiters have more latitude to alter the spectral balance to provide more consistent loudness and fewer artifacts.

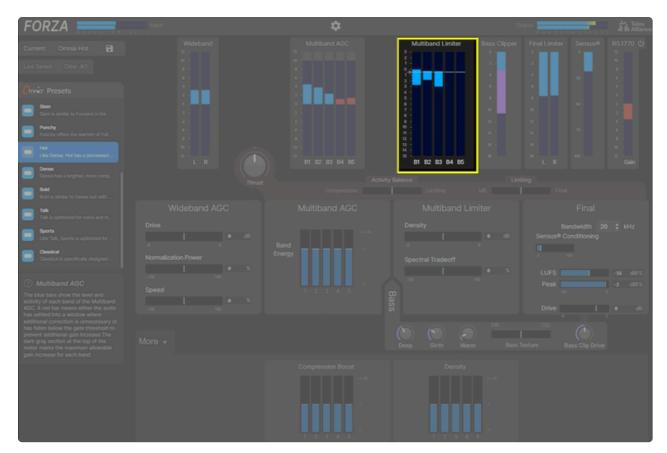


Figure 4 - Mulitband Limiter meters

## **Bass Clipper Meters**

The blue bar shows the level and activity of the soft bass clipper. The magenta bar shows the level and activity of the hard bass clipper.

i Note - Some Omnia Presets may use only the soft or hard clipper, while others may use both or neither depending on their sonic goals.

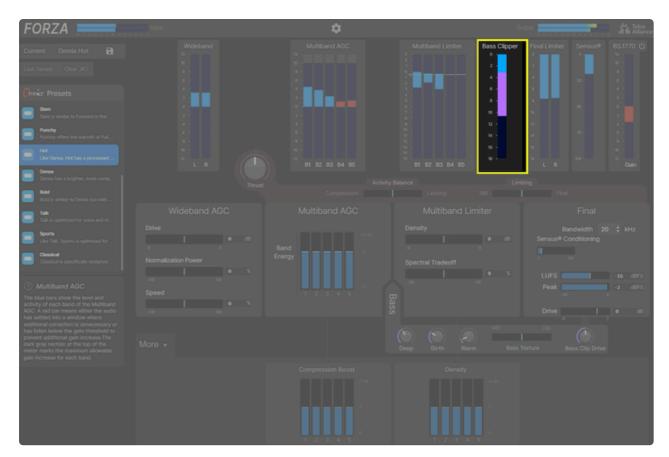


Figure 5 - Bass Clipper meter

### **Final Limiter Meters**

The blue bars show the individual level and activity of the left and right channels of the Final Limiter. The meter is an indicator of actual peak levels, not necessarily of how hard the limiter is being driven. Therefore, its activity may not directly correlate to the audible effects of the limiter.

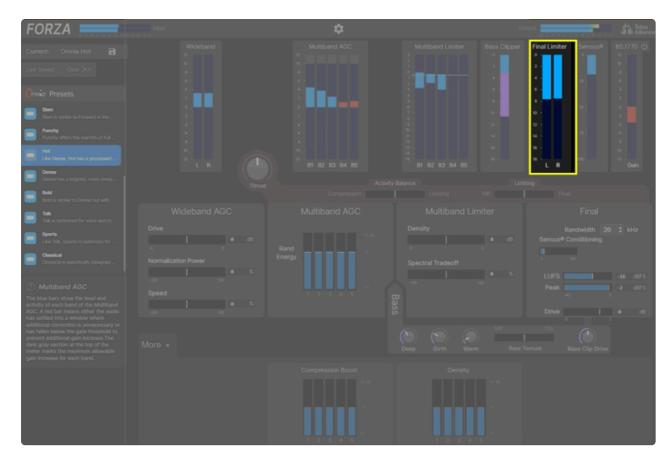


Figure 6 - Final Limiter meter

## Sensus® Codec Conditioning Meter

The blue bar shows the amount of Sensus conditioning being applied to dynamically optimize the audio for lossy codecs. When Sensus is disabled by setting the Sensus Conditioning slider to 0, the meter will be grayed out but will still deflect indicating how much activity would be taking place if Sensus were enabled and set to the default level as set by the selected Omnia Preset.



Figure 7 - Sensus meter

#### ITU-R BS.1770 Loudness Meter

The blue bar indicates the level and activity of the ITU-R BS.1770-based loudness leveler. The meter turns red when the audio falls below the threshold where additional level correction is necessary to maintain the target output level set by the LUFS control.

The loudness leveler can be enabled and disabled by clicking on the Power button at the top of the meter.

• Note - Depending upon how much correction is being performed by the leveler in order to achieve the desired target level, enabling and disabling it may result in sudden output gain changes.



Figure 8 - ITU-R BS.1770 meter

## **Getting the Sound You Want**

Forza's user interface was designed with simplicity and ease of use in mind, but with enough controls to allow more skilled users to customize the processing to their liking.

The information in this section will walk you through the initial processing setup and provide answers to common "how do I?" questions.

#### Start By Choosing an Omnia Preset

Each Omnia Preset offers a different combination of density, texture, perceived loudness, and spectral balance. A complete list of Omnia Presets with descriptions of each can be found in the Processing Controls section.

If you like the sound of one of the Omnia Presets as-is, your work here may very well be done! If you find that you are generally happy but things aren't quite perfect yet, read on.

#### How do I make the sound more (or less) processed overall?

The first control to reach for is <u>Thrust</u>, which makes multiple changes in the background to the behavior of the processing stages responsible for how processed (or unprocessed) the sound is overall.

## How do I get certain parts of the spectrum to be more (or less) prominent?

Adjusting the <u>Band Energy</u> controls of the Multiband AGC is the primary means of tailoring the spectral balance of the processed audio.

#### Roughly speaking:

- Band 1 affects the low bass.
- Band 2 affects the mid-bass and the "warmth" of the sound.

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- Band 3 affects vocals and mid-range frequencies.
- Band 4 affects the "presence" portion of the spectrum.
- Band 5 affects the highest frequencies and the "sparkle" of the top end.

## How do I get better control over inconsistent source material?

The Wideband AGC is responsible for smoothing out variations in the incoming audio. Increasing the Normalization Power is the first step to improving consistency. If the AGC takes too long to respond to the changes (or, in contrast, seems to overreact to sudden changes) adjust the Speed control accordingly.

#### How can I get better spectral consistency?

Spectral consistency - that is, the degree to which the processed audio has a consistent "EQ curve" irrespective of the frequency balance of the source material - is primarily the work of the Multiband AGCs, and to some degree, the multiband limiters as well.

Advancing each of the <u>Band Energy</u> sliders by an equal amount will adjust the respective band's drive and threshold. Advancing each of the <u>Compression Boost</u> sliders by an equal amount will alter the threshold, speed, and ratio of each band.

The degree to which the Multiband Limiters are permitted to "undo" and re-equalize the EQ curve set by the Multiband AGCs is determined by the Spectral Tradeoff control. Sliding this control to the left will keep things more consistent but at the possible expense of compromising density and loudness with some source material.

## Can I get more/less action in the Multiband AGC section and more/less action in the Multiband Limiters?

The <u>Activity Balance</u> control adjusts how hard each of these processing sections works relative to the other.

# How do I change the behavior of certain bands in the Multiband AGC or Multiband Limiter without affecting others?

Open the "More" dropdown menu (located below the Wideband AGC controls) to reveal the per-band Compression Boost sliders. Moving the sliders up and down will increase or reduce the activity in the respective AGC band. Adjusting the Density sliders will do the same for the Multiband Limiters.

### How can I get more/less bass?

To adjust the amount of very low bass frequencies, use the <u>Deep Bass</u> control. For the midand upper-bass range, adjust the <u>Warm Bass</u> control. The <u>Girth Bass</u> control can add detail to the upper bass frequencies on some program material.

### How can I get more bass impact and punch?

Carefully advancing the <u>Bass Clipper Drive</u> control can add punch to the bass by driving the bass clipper harder. Sliding the <u>Bass Texture</u> control toward "Clip" allows more bass peaks and transients to pass through the multiband limiter to the bass clipper, which also adds more low-end "slam."

With either control, it is important to remember the rule of "a little goes a long way" as driving the bass clipper too hard can cause distortion on some music. Be sure to listen on speakers that can faithfully reproduce very low frequencies as well as on devices that are frequency limited (such as smartphones, tablets, and smart speakers) as you make your adjustments.

## How do I set a specific LKFS/LUFS loudness target?

The <u>LUFS</u> slider in the Final section sets the output loudness target. It is worth noting that since the loudness controller works on average levels over a fairly long-term integration time (usually between 10 and 30 seconds depending upon the Omnia Preset), it is quite

normal for the loudness controller to remain in a gated state as indicated by a red bar on the BS.1770 meter.

Note - Both the LUFS target and peak level are "global" output settings and are not affected by which Omnia Preset or user preset is selected.

## **Glossary of Terms**

The world of audio - and more specifically the realm of audio processing - has its own lexicon of terms that may not be familiar to all users. Many of these words are used on a recurring basis throughout this user manual. To untangle the slang, we've provided a short glossary of terms below.

- Note A quick sidebar for our fellow audio processing gurus: We realize that some of these explanations are rather broad and that we are taking a bit of liberty here and there. Our goal is to strike a balance between "too simple to be helpful" and "my head hurts from all this technobabble," not provide a deep and nuanced technical explanation (as much as blathering on about such things does give us a certain guilty pleasure).
- AGC An acronym for Automatic Gain Control. In processing, an AGC is a type of compression that evens out audio levels by increasing the gain of quieter material and decreasing the gain of louder material. An AGC attempts to keep the output audio at or near a pre-determined target level no matter how much the levels vary at its input.
- Attack Time A setting on a compressor or limiter that determines how quickly it reacts in order to lower the level of an incoming signal when it exceeds a certain threshold.
- **Brilliance** The highest frequencies of the audio spectrum (roughly those above 8 kHz) that add a sense of "sparkle" or "airiness" to the sound. If over-boosted, it can make the audio irritatingly bright and exaggerate unwanted noise and codec-induced artifacts.
- Clipping A form of waveform distortion that is generally to be avoided as it "clips" the
  tops of audio peaks and results in unpleasant audible distortion, particularly in digital
  audio. When properly managed, however, the "distortion" can be used creatively to
  create a desirable effect (such as in a bass clipper). In AM and FM broadcast applications,
  clipping of some kind is routinely used to build loudness and as a means of absolute peak
  control.
- Codec In audio terms, a codec encodes a data stream at its source (either as an audio file or an audio stream) and decodes it at its destination (in a media player or web browser). The most common codes used for audio storage and transmission are MP3 and AAC. Both are "lossy" codecs, meaning they use a data compression algorithm to remove bits of data to reduce the size of a source file or the bandwidth of the audio stream. Higher bitrates throw away less data to try and preserve as much audio quality as possible, while lower bitrates throw away more data at the expense of audio quality in order to minimize file size or bandwidth requirements.

Compression (Audio) - The process of reducing dynamic range by raising the gain of low-level audio and lowering the gain of high level-audio. In the example below, the top waveform is the original unprocessed audio. The bottom waveform has been passed through a compressor.



Figure 1 - Unprocessed audio

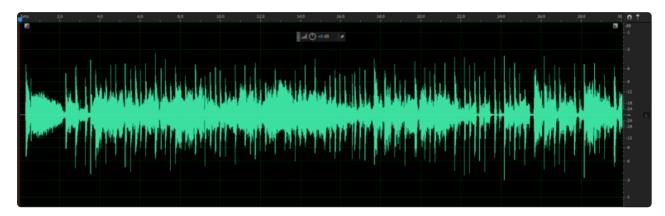


Figure 2 - Audio after compression

- Compression (Data) The process of reducing the number of bits in a digital file in order
  to reduce the amount of required storage capacity or the amount of bandwidth needed
  to deliver a data stream (including an audio stream). See Codec.
- Drive The amount of gain introduced at the input of an AGC, compressor, or limiter.
   Depending on the design of a particular circuit, more drive generally results in more overall processing and an increase in output level.
- Density A term used to describe how "processed" audio sounds, specifically in terms of how much the dynamic range has been reduced. Audio with less dynamic range is often described as sounding "compressed" or "dense," as compared to audio with more dynamic range which is often described as sounding "open." On the plus side, more density can make an audio stream more intelligible and easier to listen to in noisy environments or on small speakers or earbuds. Too much density, however, can result in

a flat sound that robs music of the natural variations in levels that add to its enjoyment and ultimately results in listener fatigue and tune-out.

- Dynamic Range The ratio between the softest and loudest sounds on a recording. Technically, and in ideal circumstances, the dynamic range of human hearing is 140 dB effectively the difference between a murmur in a soundproofed room to operating noisy machinery or sitting close to the stage at a rock concert. Digital audio is generally said to have around 120 dB of potential dynamic range. In the real world, however, such a wide dynamic range is impractical as music and other produced content are consumed in very noisy environments such as moving cars and busses, crowded bars and restaurants, or busy homes. This is one of the primary reasons broadcasters and streaming content providers process their audio to reduce the dynamic range and make it more consistent and listenable.
- **Gain** Technically speaking, a term used to express the ratio between the level of a signal at the input of some circuit or processing stage compared to the level at the output, expressed in decibels (dB). The terms "gain" and "volume" are often incorrectly used interchangeably, but the former usually refers to signal level while the latter refers to a measure of perceived loudness.
- Gate For our purposes in a processor such as Forza, the gate is the control that slows
  or freezes the action of an AGC when the audio falls below a pre-determined threshold
  to prevent noise or other unwanted audio from being increased to the same level as the
  program audio.
- **High Pass Filter** A filter that allows audio with a frequency higher than its cutoff point to pass through, while preventing audio with a lower frequency from doing so. High pass filters are typically found at the input of an audio processor where they are used to prevent very low frequencies (which have no musical value but whose presence could cause undesirable downstream artifacts and mayhem) from passing through.
- ITU-R BS.1770 The recommended algorithm to measure loudness as presented by the International Telecommunication Union. Their full document is available from the <a href="ITU">ITU</a> website.
- LKFS The method of measuring loudness in the ITU-R BS.1770 recommendation. It indicates loudness units ("L") referenced to digital full scale ("FS") using the K-weighted measurement curve ("K") which approximates how the human ear perceives loudness. One LKFS loudness unit (commonly abbreviated as "LU") is equal to 1 dB. Though the acronyms are different, LKFS is identical to LUFS, and the terms are used interchangeably in different parts of the world.
- LUFS An abbreviation of "Loudness Units Full Scale," and used interchangeably with LKFS.

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Limiter – A type of dynamics processor which typically uses faster attack and release times and higher ratios compared to a compressor or AGC. Limiters with very fast time constants and an Infinate:1 ratio are generally referred to as "peak limiters" as their job is to prevent audio peaks from exceeding a certain level. Limiters can also be used to increase program density ("program limiters") or work on only certain parts of the audio spectrum ("multiband limiters").

• **Limiting** – The process of reducing dynamic range by lowering the levels of audio peaks but without raising low-level audio. In the example below, the top waveform is the original unprocessed audio. The bottom waveform has been passed through a peak limiter.

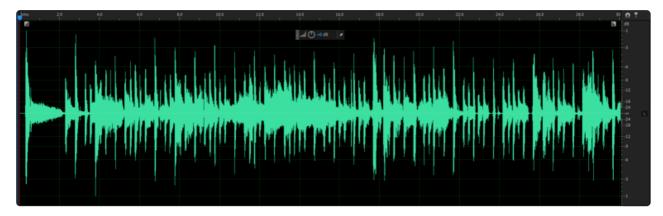


Figure 3- Unprocessed audio



Figure 4 - Audio after limiting

• Loudness – In audio applications, loudness is generally used as a comparative term between two songs, stations, or streams. Loudness levels can be quantified by using a meter employing the LUFS (sometimes referred to as LKFS) loudness measurement unit as standardized in the ITU-R BS.1770 measurement recommendation. Loudness also has a qualitative element, as "perceived" loudness plays a role when comparing two signals which may show the same LUFS value on a meter but be experienced very differently. For example, in the files below, the top waveform shows the original, unprocessed audio which had a measured LUFS of -17.28 dB. The bottom waveform shows a heavily

compressed and limited version which sounds significantly louder (and subjectively much worse) but it too measures -17.28 dB LUFS.



Figure 5- Unprocessed audio measuring -17.28 dB LUFS

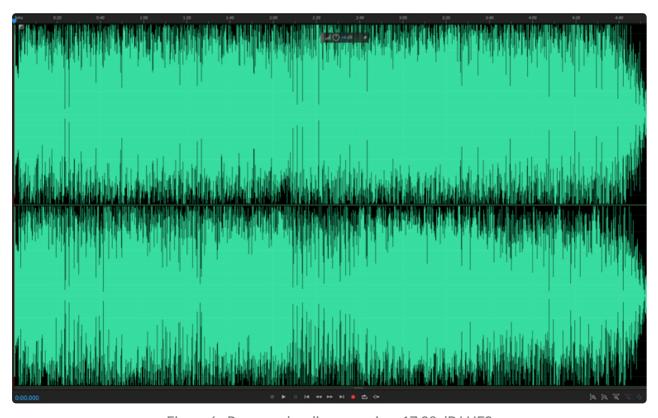


Figure 6 - Processed audio measuring -17.28 dB LUFS

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• Low Pass Filter – A filter that allows audio with a frequency lower than its cutoff point to pass through while preventing audio with a higher frequency from doing so. Low pass filters are typically found at the output of an audio processor where they are used to roll off high frequencies, as this is the range where the artifacts of low bitrate codecs are most often audible.

- Multiband A type of AGC or limiter that divides the audio spectrum into several different bands or frequency ranges. Multiband processing allows the spectral balance of the output audio to be very consistent, as each band works independently of the others. This approach also prevents certain processing artifacts common in wideband processing that can occur when a strong signal in a narrow frequency range (such as the "thump" from a kick drum) causes the compressor or AGC to attenuate all frequencies by an equal amount in response.
- **Presence** The range of the audio spectrum between 2 kHz and 6 kHz. It can add clarity and definition to the sound, but too much of a presence boost can cause an irritating "honky" or "nasal" sound.
- Processing Artifacts By its very nature, processing changes one or more
  characteristics of the audio, and those changes are rife with compromise. There is no
  free ride. Much like a balloon, once it is filled with air and tied off, you can squeeze one
  end to make it smaller, but the opposite end puffs up. Squeeze too much, and the balloon
  breaks. Below is a list of some of the most common audible processing artifacts and a
  word or two on what causes them.
  - Distortion Technically speaking, any change to the original audio is a form of distortion, but in this case, we are talking about objectionable distortion. Most often we hear distortion as a fuzzy or edgy sound to the audio, not unlike the sound when you turn up the volume too far on a stereo. Sometimes it is subtle enough that only someone who knows what to listen for can hear it. Other times, listeners might not be able to identify it overtly, but be subconsciously annoyed to the point of switching off the music. And sometimes such as when digital audio exceeds 0 dB full scale, the sound is catastrophically and unmistakably broken. Distortion has many causes, but in general, it is an indication that some component or circuit is being driven beyond its intended range or some threshold has been exceeded. In a streaming audio processor, over-driving the limiters or the bass clipper can sometimes cause audible distortion. Over-driving the input of the streaming encoder can also cause issues, as can a low-quality decoder on the listener's side of the equation.
  - Ducking This describes a sudden rapid attenuation of the existing audio when a new audio component is introduced. For example, imagine an announcer begins a song, lets it run for a few seconds, and then decides to open the mic and talk over the intro.
     As soon as they begin to speak, the music suddenly disappears to a low level. This is

typically caused by fast attack times that instantaneously drive the level of the music down to make room for the voice.

- Intermod More properly called "intermodulation distortion," this artifact results in some manner of distortion or bad behavior in a part of the audio spectrum outside of the fundamental frequency. For example, if the low band of a multiband limiter is working outside of its ideal range, it may create harmonics that show up in a higher band as distortion.
- Pumping This effect occurs most often in wideband processing when low frequency peaks – such as those from a kick drum – rapidly modulate the rest of the audio spectrum. When a peak occurs, everything else disappears for a split second until it passes, at which point the levels are instantaneously brought back up to their former level. Multiband processing – particularly multiband limiting – can help alleviate this artifact.
- Ratio This is the relationship between a gain change at the input of a compressor or limiter compared to the resulting change at the output. For example, a 1:1 ratio means that a 6 dB change to the input audio level will result in a 6 dB change at the output (which effectively means the processor is being bypassed). A 4:1 ratio means it takes a 4 dB change at the input to change the output by 1 dB (or a 12 dB change at the input to yield a 3 dB change at the output). Higher ratios are described as being "tighter," while lower ratios are referred to as "looser." Sonically, a lower ratio will result in a more open and natural sound with a greater sense of dynamic range but may not provide enough control with inconsistent source levels. A higher ratio will provide a more consistent output level-wise but can result in flat sound with no sense of natural dynamics.
- **Release Time** A setting on a compressor or limiter that determines how quickly it reacts to raise the level of an incoming signal when it falls below a certain threshold.
- **Spectral Balance** This refers to the relationship of audible frequencies to one another. In the example below, the top graph shows a fairly "flat" curve where all frequencies are at around the same level. The bottom graph shows a curve with a boost in the bass and presence frequencies and a cut in the lower mid-range. Spectral balance is sometimes referred to as the "EQ curve."

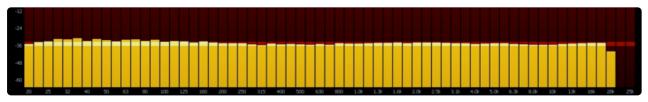


Figure 7 - A relatively flat spectral balance

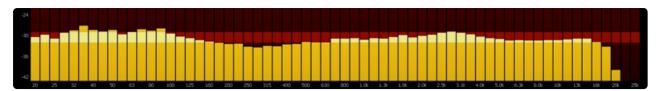


Figure 8 - A spectral balance with a boost in the bass and presence, and a cut in the lower mid-range.

- Spectral Consistency The degree to which the spectral balance of the output audio stays the same regardless of the balance of the input audio. Multiband processing is used to actively "re-equalize" the sound so that a bass-shy song gets a lift in the low end while an overly bright recording gets its high end reduced. Some degree of spectral consistency is generally considered a good thing as it helps smooth out variations between songs and provides a more consistent experience for the listener. With some formats, however such as classical music it is often considered more important to respect the artistic intent of the original recording, and so less spectral consistency may be preferred.
- Threshold In a traditional compressor or limiter, the threshold is the level the audio needs to exceed before compression takes place. In an AGC, it is the level above which the audio will be attenuated and below which it will be increased. In the case of a gate, it is the level below which the audio will be considered "noise" and not increased toward the target gain setting.
- Warmth A term used to describe a fullness in the upper-bass and lower-mid frequencies (approximately in the range of 100 Hz to 200 Hz). The range of male announcers and singers falls roughly into this range, as does the lower end of an acoustic guitar and the middle range of a bass guitar. A small boost in these frequencies can yield a full and pleasing sound but can result in an unpleasant "boomy" or "tubby" sound if done in excess.
- Wideband A type of compressor or AGC that acts on changes to all parts of the audio spectrum, in contrast to a multiband design that divides the audio into smaller frequency ranges. Wideband AGCs are typically the first dynamic gain stage in a processor and "ride gain" over inconsistent input levels. They also help to consistently drive downstream processing stages, typically the multiband AGCs.