Audio Processing for Radio and Digital Media: Good Practices and Some Pitfalls

Why Process?

- For **Digital Radio** and **Netcasts**:
  - Create a **consistent**, “polished” sound.
  - Create a “**signature sound**” that is part of a transmission’s overall **branding**.
  - Help **compensate** for **imperfect production** and **careless operators**.
  - Create an appropriate **balance** between **speech** and **music**.
  - **Except for increasing coverage**, processing for digital radio and the Internet have the **same goals as processing for analog radio**.

Why Process?

- For **Digital Radio** and **Netcasts**:
  - The main difference between optimum processing for analog and digital radio is the **peak limiter technology**.
  - To **optimize lossy codec performance**, the processor should **add as little additional spectrum to the input signal as possible**.
  - This implies use of **non-clipping limiters**, usually using **look-ahead techniques**.

Processing for Low Bitrate

- Today's MPEG4-standard low-bitrate codecs (**HE-AAC, HE-AACv2,** and the new **xHE-AAC**) use **Spectral Band Replication** (SBR), a **bandwidth-extension** technique.
- The codec uses **straight AAC encoding below a specified crossover frequenc**y, which **depends on bitrate**.
- Above the crossover frequency, the receiver generates "plausible" high frequencies by **frequency-doubling lower frequency ene**rgy.
- The encoder sends a multiband **gain control signal** to match the HF energy generated by the receiver to the amount of HF energy in the original audio as much as possible.

Processing for Low Bitrate

- Processing guidelines:
  - Because most of the problems occur at frequencies above 8 kHz…
    - The more neutral the HF balance, the less SBR is stressed.
    - Don’t use the audio processor to **exaggerate high frequencies unnaturally**.
    - Perform **high frequency limiting** on overcooked source material.
    - Do not let the processor **increase high frequency density beyond that present in the source material**.
  - Other than that, **process freely to achieve your artistic goals**.
    - **AAC**, which handles everything except the very highest frequencies, is a **very good codec**.

Processing for Low Bitrate

- Setting Peak Output Levels
  - Be sure that your peak limiter is "**0 dBFS+ aware**." This means that it **anticipates and compensates for the effect of the analog reconstruction filter after a player's DAC**. This filter can cause levels that are up to **3 dB higher** than the **highest digital sample**.
  - A 0 dBFS+ aware peak limiter uses an **oversampled sidechain** so that it will produce additional gain reduction if needed to quash analog-domain overshoots.
  - Because it removes program energy, the HE-AAC codec can cause **peak overshoots of 2 dB or more. This is above and beyond the 0 dBFS+ effect.**
  - Some player devices will **hard-clip overshoots**, either in their digital or analog circuitry. Others will activate a **questionable peak limiter** to protect against hard clipping.
  - Recommendation: **Use a 0 dBFS+ aware limiter**, and s**et its peak output level to -1.5 dBFS**. Any clipping that remains will have a low duty cycle and will not be heard.

What to do About Bass?

- Bass is important, but it must **never be allowed to damage the midrange** – many listeners **don’t hear bass**, either because their **radios are played quietly** or because their radios **cannot effectively reproduce bass at any volume level**.
- Even **upmarket table radios** like those from Bose, Cambridge, Polk, Boston Acoustics, and others **have little response below 70 Hz**, so for the mass audience, the **midbass is more important than the bottom two octaves**.

What to do About Bass?

- Choose whether to **emphasize mid-bass** or **low bass** performance based on your **target audience** and the **radios they are likely to be using**. **Speaker** or **earbuds**?

- Use processing techniques, like **bass pre-limiting**, that ensure that **bass is not allowed to damage midrange**.

- Add a **tasteful amount of soft clipping to the low bass** so that it produces **extra midbass energy** for small speakers.

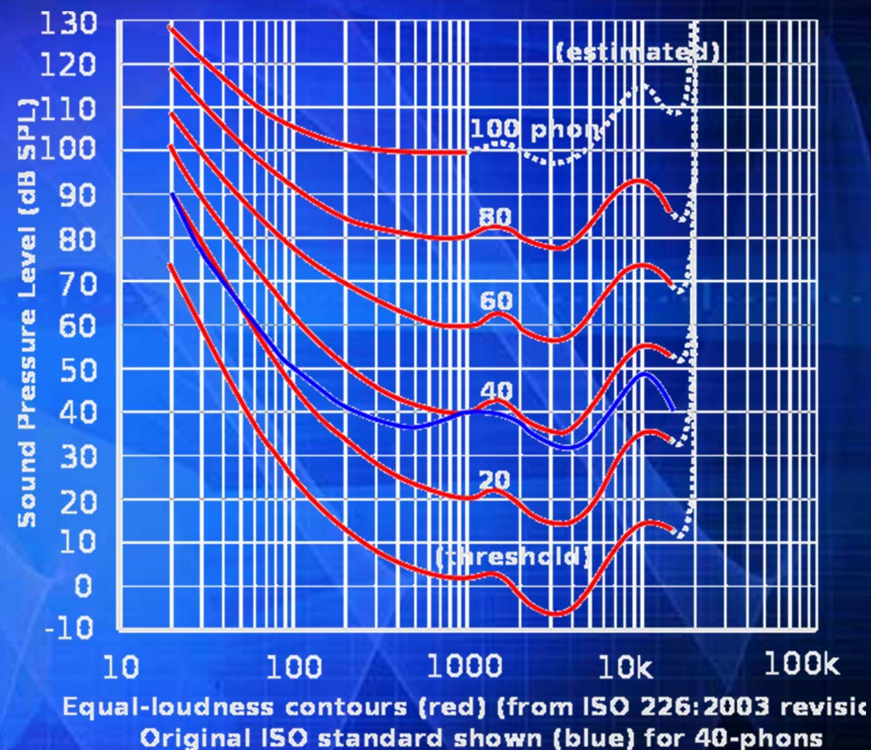
Subharmonic Synthesis

- An **old bass enhancement technique** that has attracted some recent interest from broadcasters.
- Has many **pitfalls and snares** of its own.
- It can help some **older material** with little energy below 70-80 Hz, but never forget the **equal-loudness curves**:

Subharmonic Synthesis

- Material below 50 Hz takes up **lots of peak level to produce significant loudness**.

- Close spacing of curves means **small changes in amplitude lead to large change in subharmonic loudness**.



Subharmonic Synthesis

- Because of the amount of peak level they use up, subharmonics will always make a broadcast sound **quieter for a given amount of processing artifacts/distortion.**
- They **use up peak level that otherwise could be dedicated to audio to which the ear is more sensitive**. (This is an inevitable effect of the equal-loudness curves.)
- Safest when created in the **production studio** where the effect can be **monitored by humans** before it's let loose on the air!

Subharmonic Synthesis

- **Should be defeated automatically** when the program material **already has sufficient bass**.
- Use it in conjunction with **high-compression-ratio multiband compression** to ensure **consistent loudness of the subharmonics**.
- Use bass **intermodulation distortion reduction techniques** in the on-air compressor and peak limiter.
- Subharmonics should **track the level of their generating frequencies in a frequency-dependent way** to keep the amount of LF enhancement **subjectively constant**.
- Subharmonics make **male voice sound weird**, so a subharmonic synthesizer should be used in conjunction with **automatic speech/music detection** and should not be applied indiscriminately to material above 90 Hz.

De-Clipping

- Information is **100% lost** in flat-topped areas and cannot be recovered. A flat-topped waveform is a “singularity.”
- De-clippers must make **educated guesses** about what’s missing based on **interpolation from material surrounding the clipped samples**.

De-Clipping

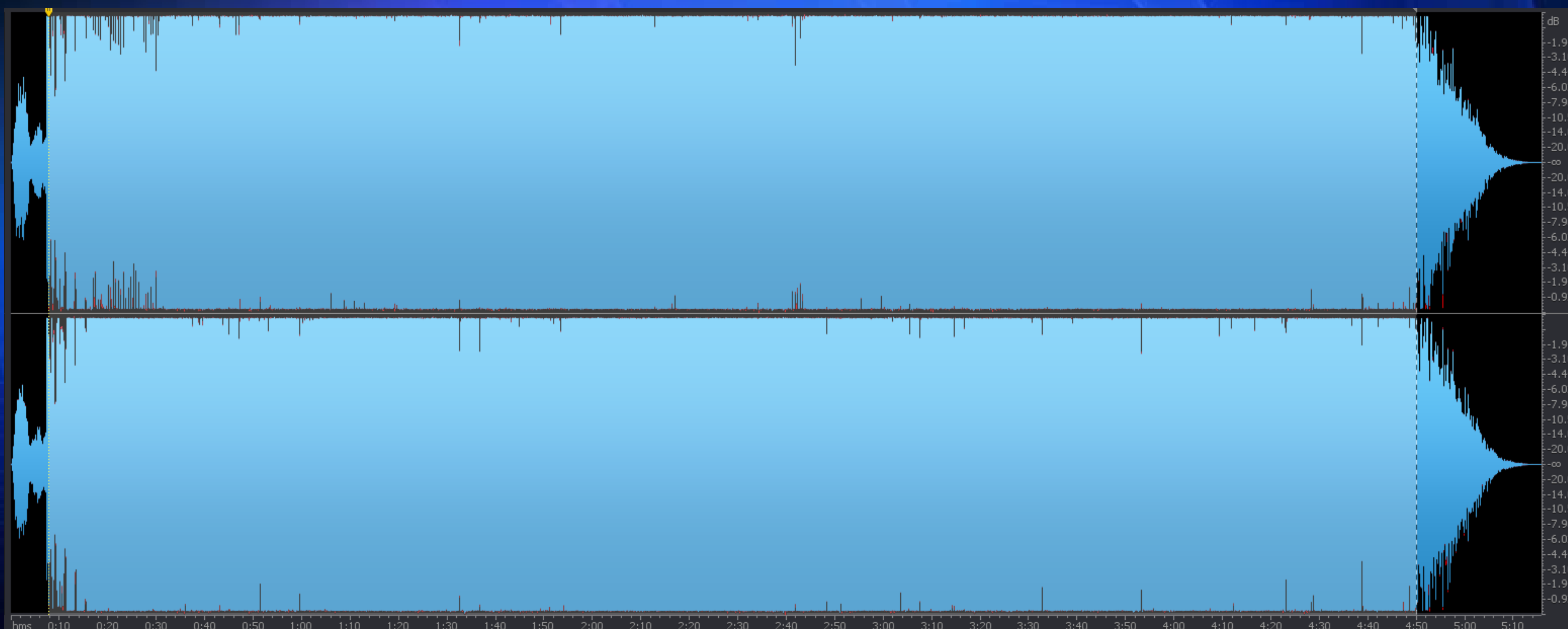
- The interpolation must use a **model of the clipping process**.
- But many waveforms that **look they have been hard-clipped** have, in fact, been **peak-limited by more complex limiting processes** with **sidechains and memory**.
- Each limiter manufacturer has a **proprietary way of computing the sidechain**. For competitive reasons, these are **seldom made public**.
- Even if the sidechain is public knowledge, if the compression ratio is **infinite**, it is still **impossible to deduce what the limiter’s input was**.

De-Clipping

- The following slide shows a 5-minute waveform that looks like it is **ridiculously clipped**.
- Not counting the intro and fade-out, its BS.1770 integrated loudness is about –6.7 LUFS), and **any de-clipper would work very hard to try to "de-clip" it**.
- Based on the waveform, one would think that it must sound awful. Yet in fact, it sounds fine and **does not sound distorted**.

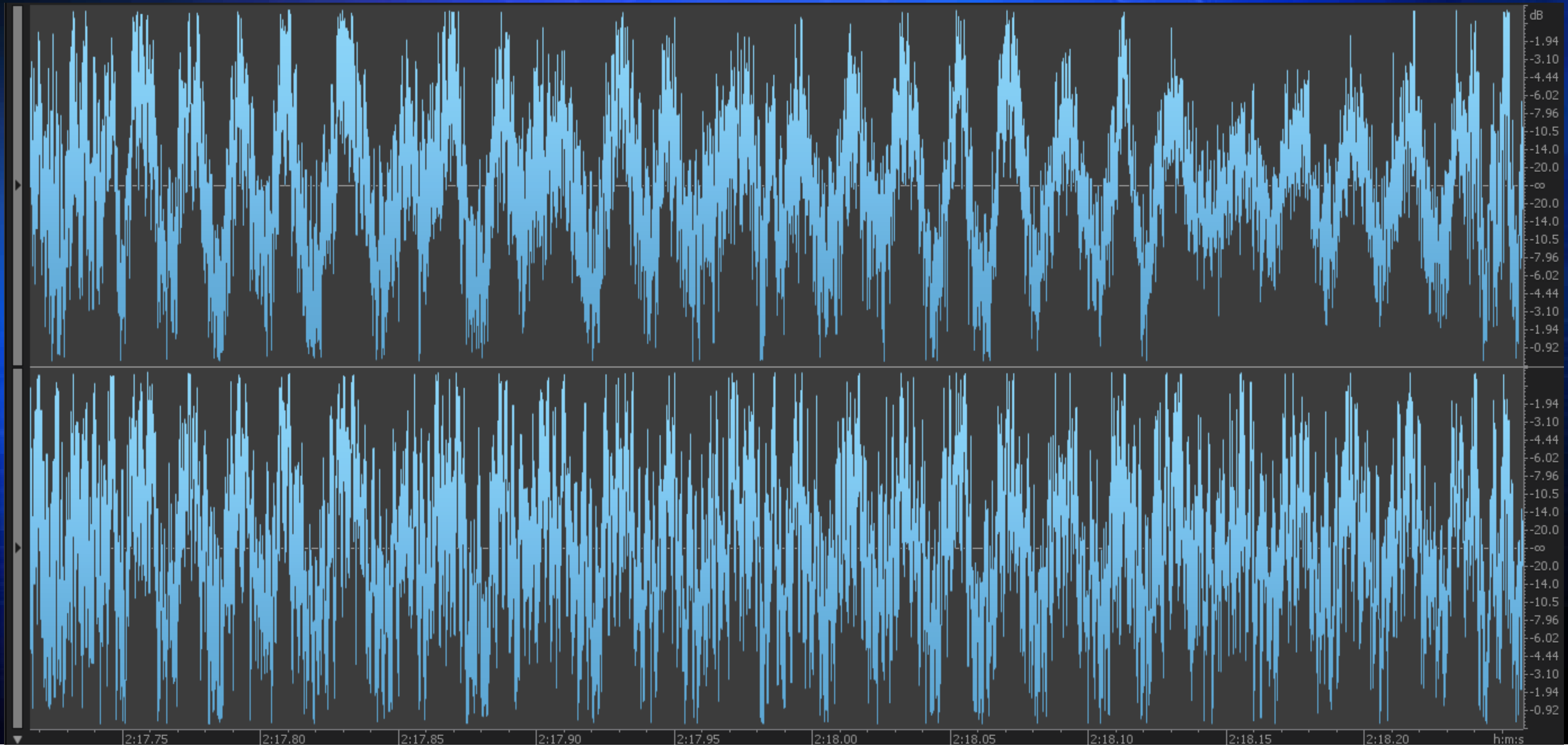
De-Clipping

- This 5-minute track is almost completely clipped, right?
- -6.7 LUFS with momentary loudness to -4.4 LUFS?!? Must sound horrible…



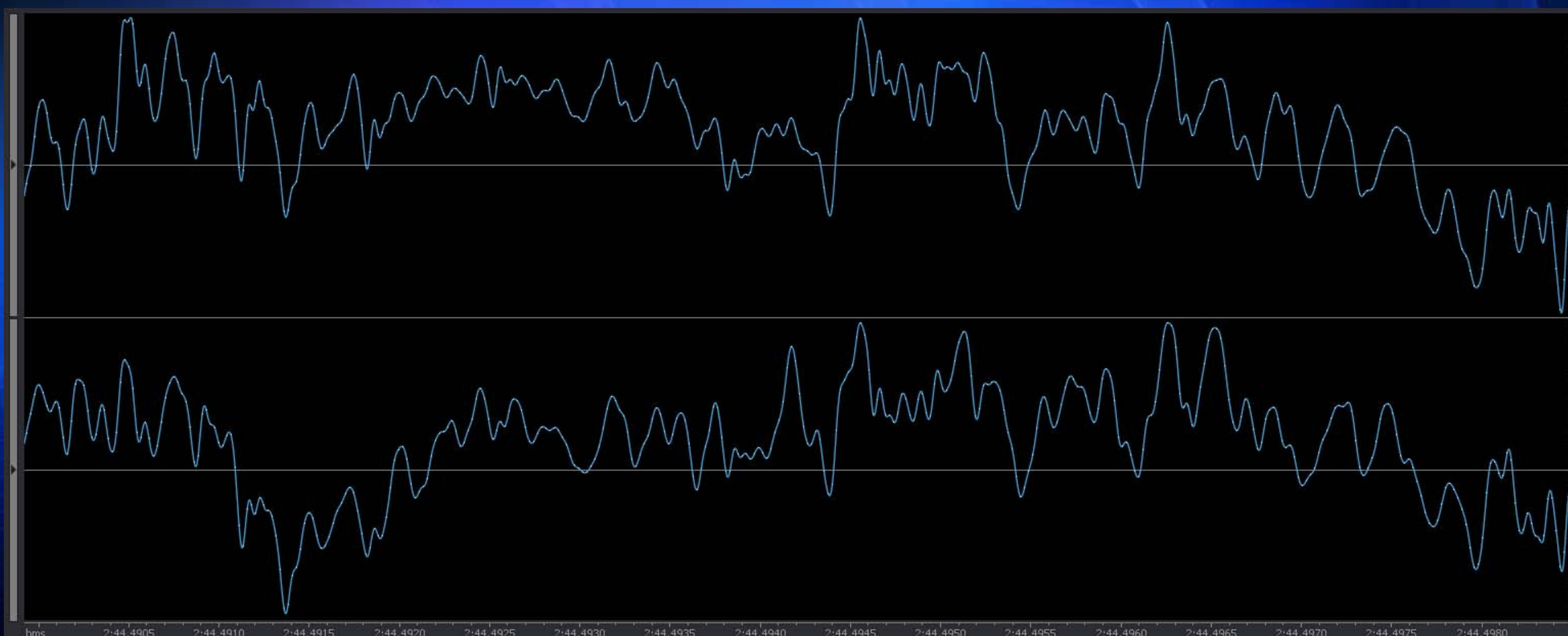
De-Clipping

- OK…let's zoom in to a 2-second segment.
- Still looks ugly and clipped…



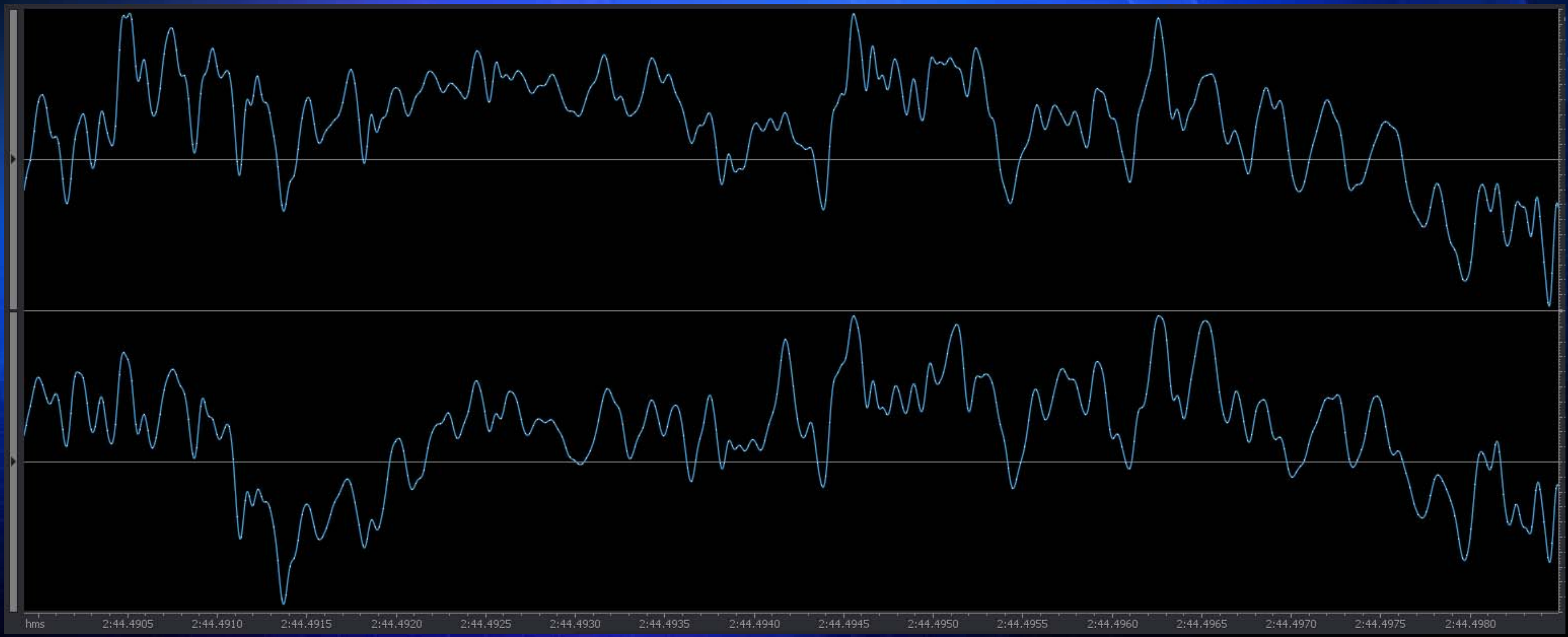
De-Clipping

- Not so fast! In fact, this waveform was made with a very complex **distortion-cancelling peak limiter** (the MX limiter in our Optimod-PCn processor).
- It uses a **psychoacoustic model to minimize audible IM distortion**. There is almost **no actual clipping**, and what there has a **very low duty cycle**.



De-Clipping

- Yet the de-clippers I have tested **misidentified hundreds of peaks as being clipped and tried to “fix” them**.



De-Clipping

- De-clippers can **increase punch on transients** by increasing peak levels by **guessing what the missing waveform is**.
- But this is **not the same as cancelling IM distortion**. Distortion cancellation depends on having a **precise, invertible model of the peak limiting process**. **This is usually impossible**.
- Because de-clipping is a nonlinear process, it can **make its own IM distortion that adds to any IM distortion present in the original track**.
- **The better the original peak limiting algorithm, the more likely it is that de-clipping will add IM distortion, not cancel it**.

De-Clipping

- **Conclusion**:

  If **simple peak clipping** was used on a given track, then **de-clippers can help**. But sometimes they make things worse.

- Therefore, the **proper place for a de-clipper is in the production studio**, so that human ears can determine if the de-clipper is helping or adding another layer of distortion.

- Moreover, in the broadcast processing chain, de-**clipped waveforms force the on-air processor's peak limiter to work harder**. So **use de-clippers with care**!

Phase Skew Correction

- Time delay between identical audio components in the left and right channels causes **comb filtering in the mono sum**:
  - **High frequency rolloff**
  - “**Flanging**” sound in extreme cases
- This is particularly important in **FM broadcast**, where **auto radios often blend** partially or completely to mono.
- In FM, phase skew errors cause the **energy in the stereo subchannel to increase**, which **increases susceptibility to multipath distortion**.

Phase Skew Correction

- Phase skew correction is also important in **low bitrate streaming**.
- "**Parametric stereo**" works by encoding the **mono sum** along with a **low bitrate "steering signal**." This process is most efficient when there are **no phase skew errors**.
- **HE-AAC v2** and **xHE-AAC** codec both use parametric stereo.

Phase Skew Correction

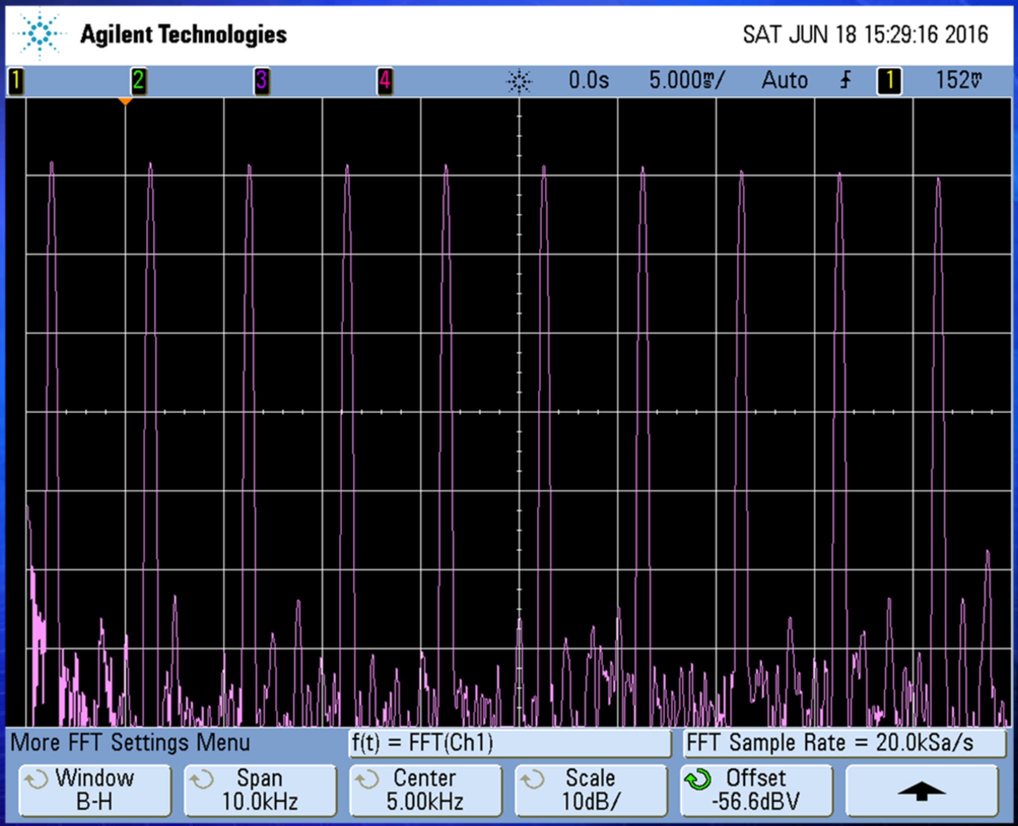
- Various phase skew correctors can have **dramatically different performance and capabilities**.
- Many correctors work by adding a **simple time delay** to the “earlier” channel.
- This models **analog tape gap skew**, but cannot correct multiple delay errors
- For example, a recording could have both **analog tape gap skew** and **comb filtering caused by two microphones picking up the same instrument in the studio**. A simple time delay-based phase skew corrector **cannot fix this problem**.

Phase Skew Correction

- **Orban's "Multipath Mitigator**" phase corrector is **multidimensional: it can fix multiple, unequal delay errors in a recording.**
- It transforms the source to "**intensity stereo**," maintaining **full stereo separation** while **removing phase shifts between elements common to the left and right channels**.
- To illustrate this ability, I created a **10-tone test stereo waveform** with a **90 degree phase difference between each tone** in the left and right channels. Power (RMS) in the two channels is the same.
- The 90 degree phase shift produces a **different differential time delay for each tone**: Each time the frequency is halved, **the delay doubles**.

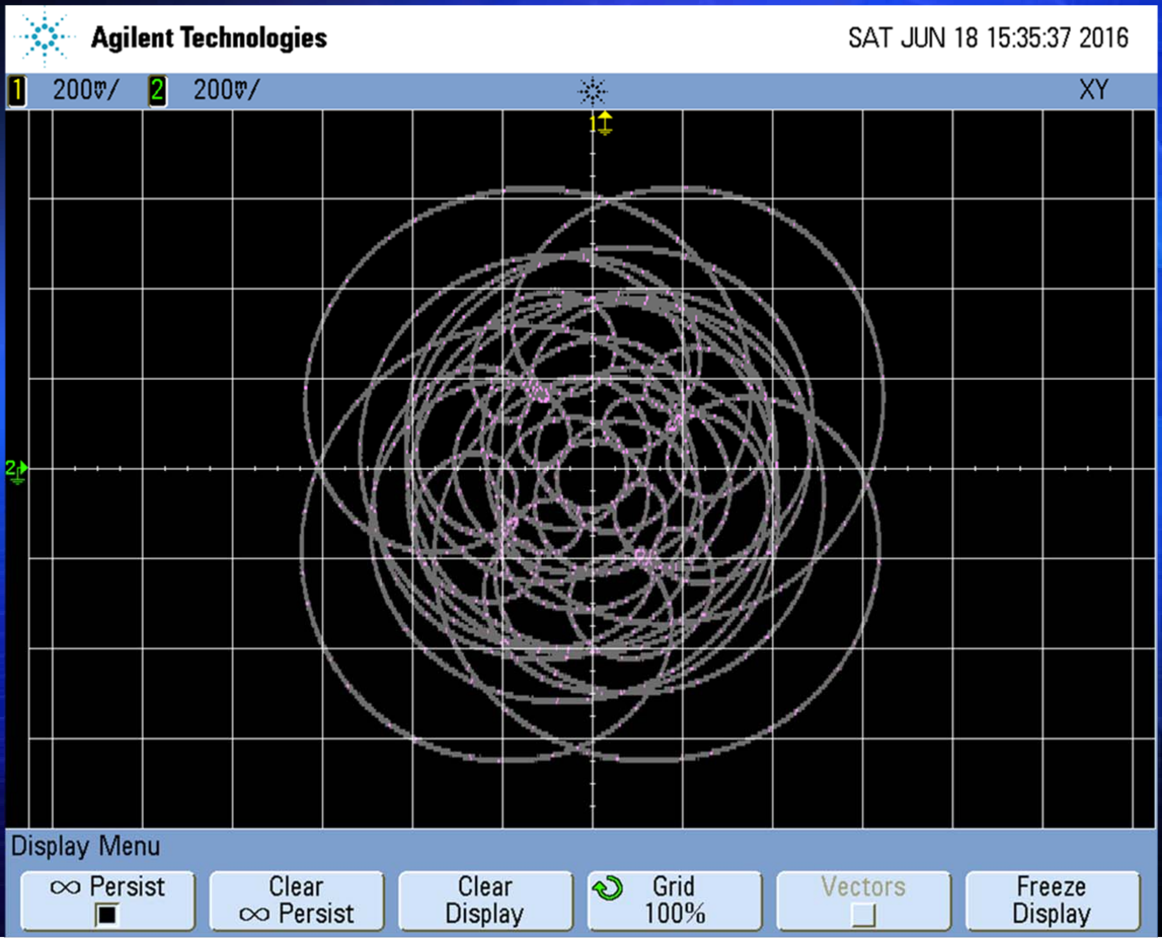
Phase Skew Correction

- **FFT** of the left channel of the test signal:
- Frequencies are 250, 1250, 2250, 3250, 4250, 5250, 6250, 7250, 8250, and 9250 Hz.



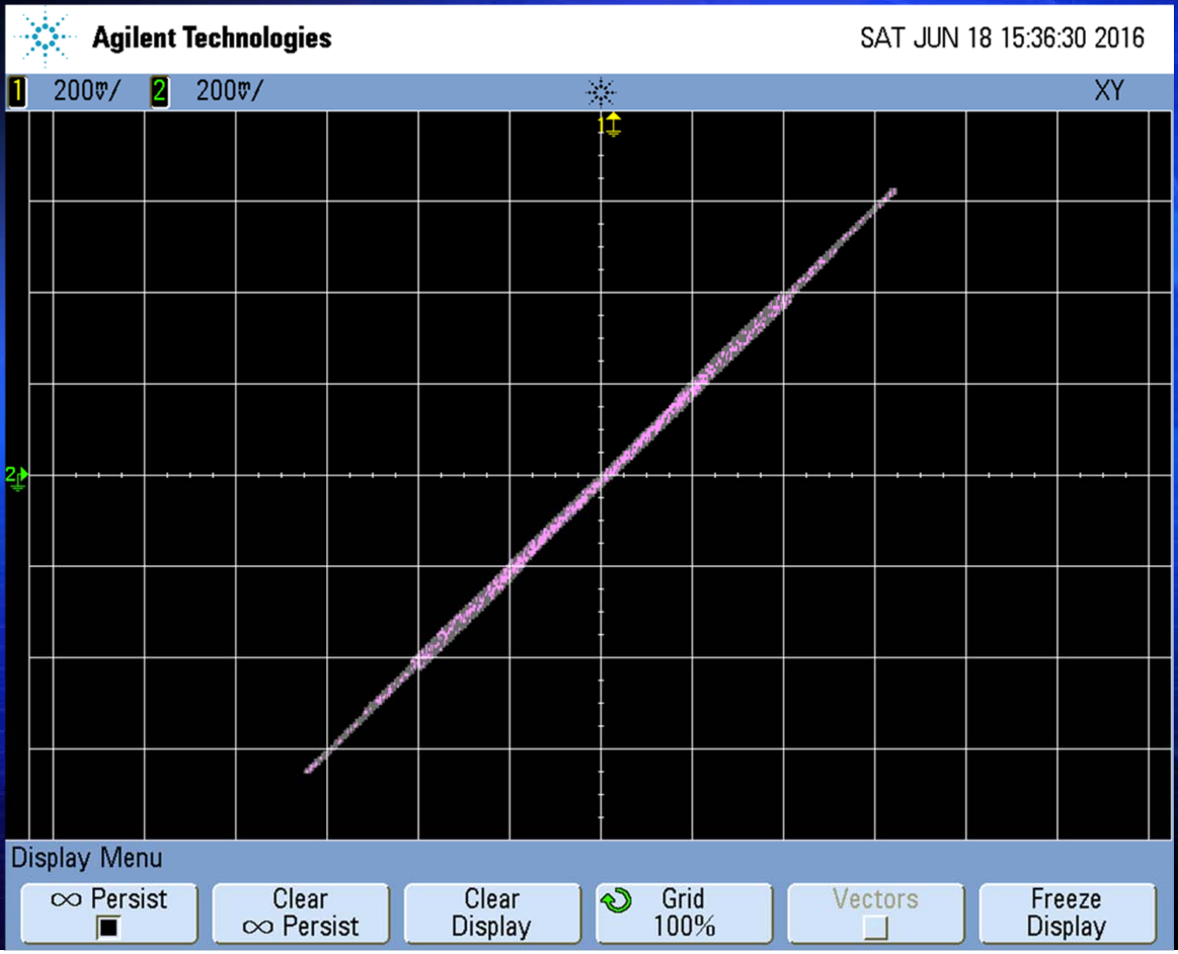
Phase Skew Correction

- The **uncorrected left and right channels** make an interesting **Lissajous pattern**:



Phase Skew Correction

- **Activating the Orban phase corrector puts all of the tones in-phase**:



Phase Skew Correction

- The Orban phase corrector has a **crossover frequency control** so that users can choose the **frequency above which the phase corrector is active**.
- Using an **800 Hz** crossover frequency preserves subjective “**envelopment**” caused by phase differences in the bass and lower midrange, while **still correcting problems in the range where comb filtering is likely to occur**.
- Using a **lower crossover frequency** may be **better in FM broadcast** if **reducing multipath distortion is more important than envelopment**. This depends on the terrain in the station’s market area.

Trouble in Paradise?

A Heretical Look at Loudness Control:

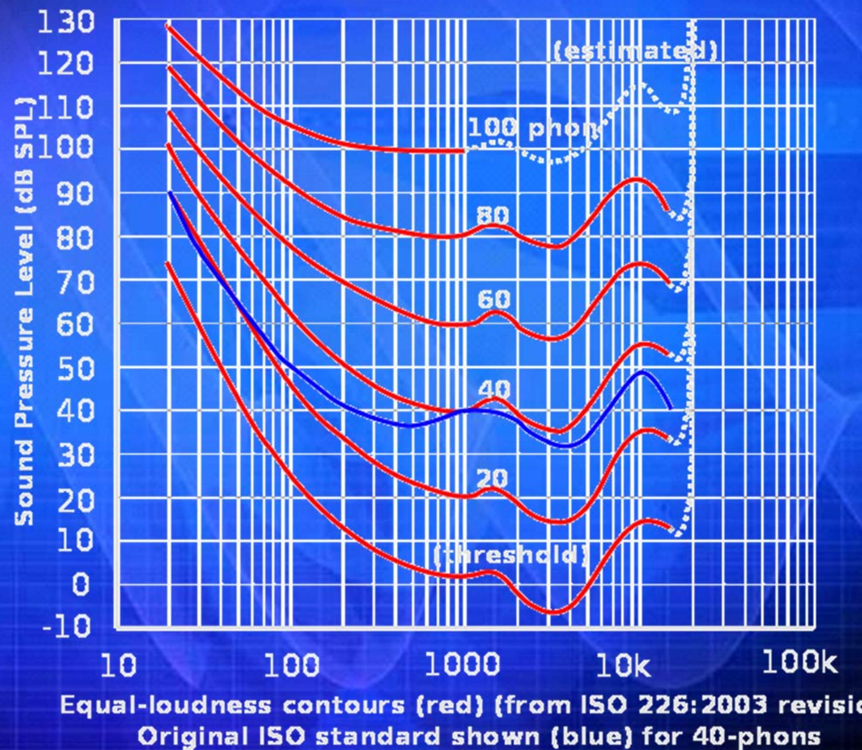
-BS.1770
-Jones and Torick

Subjective loudness metering and automatic on-line loudness control have a long history.

- **The first on-line automatic loudness control technology was developed by CBS Laboratories in the mid-1960s** in response to a Federal Communications Commission study regarding audience complaints about objectionably loud commercials.
- **Bronwyn Jones and Emil Torick** at CBS Technology Center revisited this work in **1981** to **improve loudness meter accuracy**. This work was published in the SMPTE Journal.
- In **1983**, the FCC Office of Science and Technology tested the J&T loudness controller, concluding that it was **likely to reduce complaints caused by loud commercials**.
- In **2005**, Orban made substantial **improvements** to the J&T loudness controller **gain computer sidechain** to:
  - **improve smoothness**
  - make operation more **audibly subtle**
  - **produce more consistent dialog loudness,** even when the dialog is **mixed with music and/or effects**

Ideally, a loudness meter should take into account:

- **Frequency Dependence**: The ear's perception of loudness is strongly dependent on frequency.



Ideally, a loudness meter should take into account:

- **Loudness Summation**: For a given total sound power, the sound becomes louder as the power is spread over a larger number of *psychoacoustic critical bands* (about 1/3-octave).
- **Loudness Integration over Time**: A given amount of acoustic power sounds progressively louder until its duration exceeds about 200 milliseconds, at which point no further loudness increase is heard.

The BS.1770 meter takes only frequency dependence into account:

•The **BS.1770 meter** is a wideband time-integrated power meter preceded by a frequency weighting filter. It uses **gating** to **ignore low amplitude parts of the audio** that do not contribute significantly to loudness perception.

•The BS.1770 meter **does not model loudness summation** or the **short-term loudness integration time constants** (~200 ms) of human hearing.

BS.1770-3 disagrees with human listeners by up to 6 dB.

To maintain a +2/-5 dB "comfort zone," the **straight line must be shifted downwards**, so the meter **under-reads a significant amount of program material**.

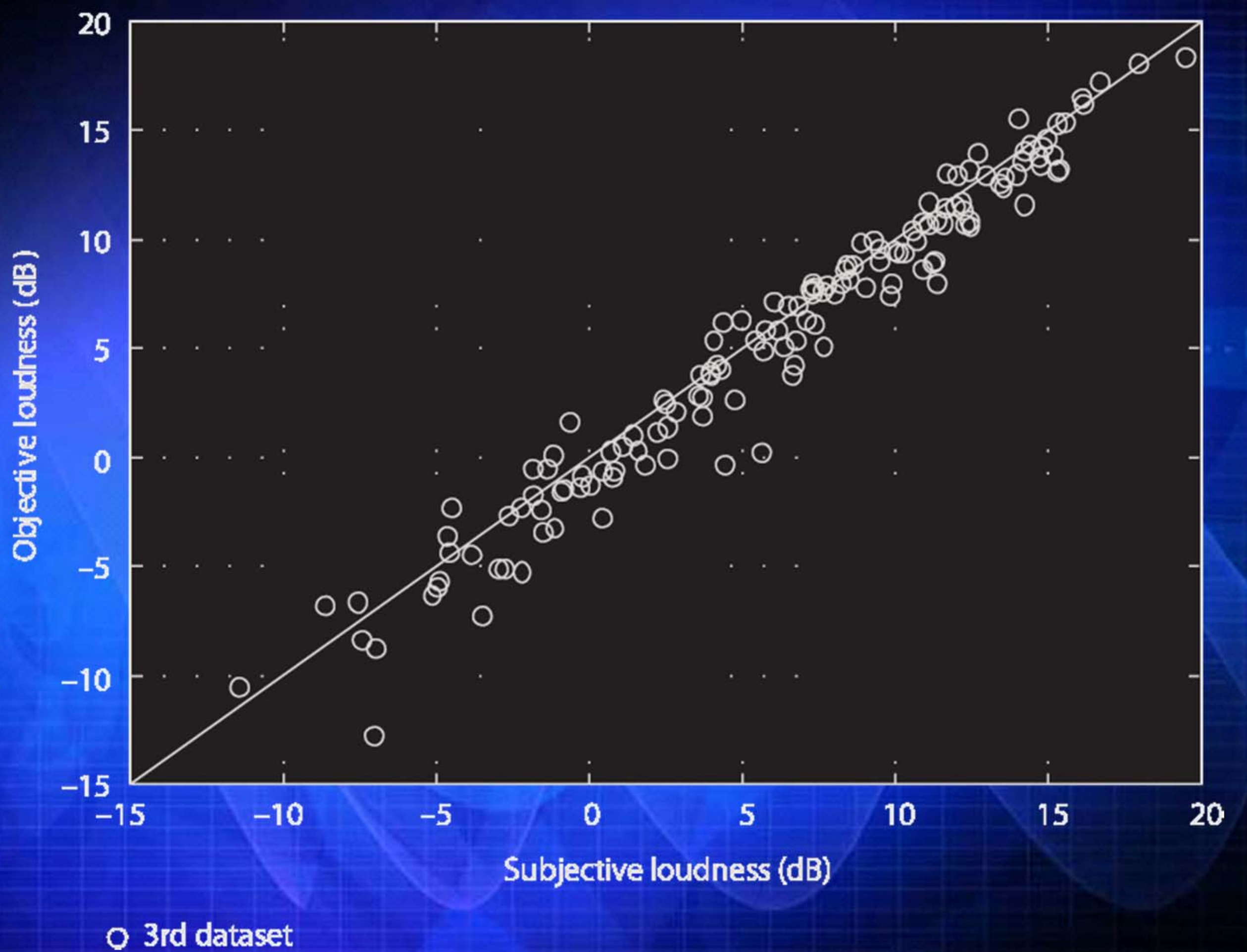


Figure 13, BS.1770-3 standard

Mechanically relying on the BS.1770 meter has caused complaints in the Hollywood production community

For example…

"I did a –24 piece for Fox that was wall to wall singing and music for two minutes. Because of the overall loudness and continued full audio signal **I had to bring it down and when it aired it was 3 db too quiet even though it matched the magic LKFS number**. I have no problem using these meters or meeting specs but **they are faulty**."

— "wheresmyfroggy," AVID board, 3-28-2011

2015: The AES to the Rescue!

- **Fortunately,** the **AES TD1004.1.15-10 “Recommendation** for Loudness of Audio Streaming and Network File Playback” (2015) **takes “genre” into account**:
  - “Within a given program, the largest perceived difference to be noted is **speech versus music**. Speech normalized to the same Integrated Loudness as a music stream **inevitably sounds too loud**. It is recommended to normalize speech (dialog) segments within other segments 2 to 4 LU (or more) below the loudness of the other segments.”

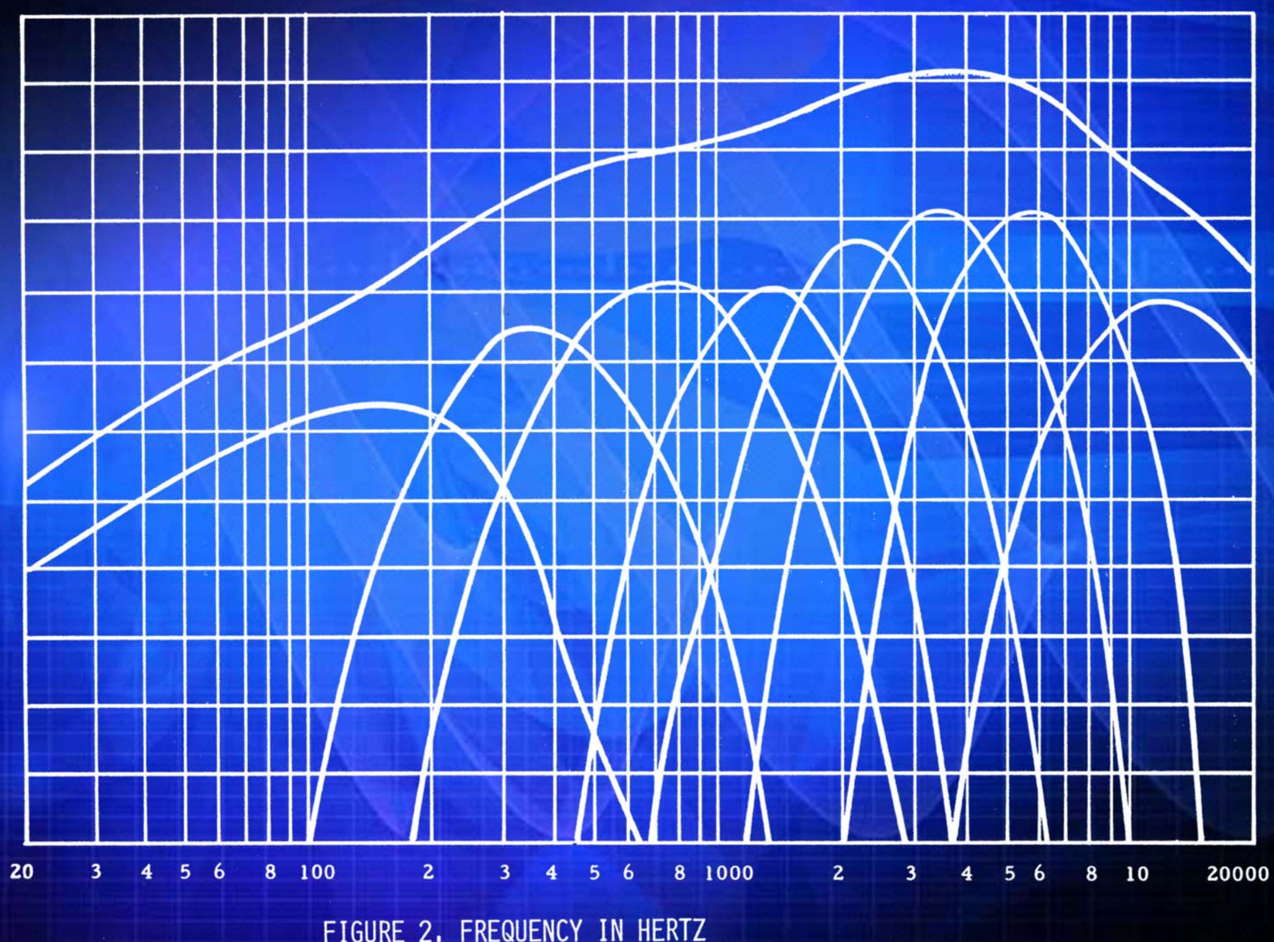
J&T Loudness Meter Technology: Psychoacoustic model

- **Loudness Summation**: The meter first divides the signal into **eight frequency bands** and applies each band to a rectifier followed by a fast averaging, which mimics the **"instantaneous" loudness integration time of human hearing**.

- **Frequency Dependence**: The averaged outputs of the bands are summed with unequal gains that mimic the **frequency dependence of the ear**, as determined by experiments with listeners using **octave-band noise**, heard on loudspeakers in a room **typical of a home listening environment**.

- **Loudness Integration in Time**: The sum of the smoothed filter outputs is applied to a filter with an **integration time of approximately 200 ms**. This makes the J&T a "**short-term**" or "momentary" loudness meter.

J&T Filterbank Curves & Summation



Loudness Meter Accuracy Limitations 1

- Loudness meter accuracy is inherently limited by the fact that human listeners **disagree by several dB** when asked to match the loudness of test program material with a reference tone or wideband noise. **Different people perceive loudness differently**.
- A loudness meter can only be calibrated for a **fixed acoustic listening level** because the equal-loudness curves show the **ear's sensitivity as a function of frequency to be level-dependent**.

Loudness Meter Accuracy Limitations 2

- The **room acoustics** and **frequency response** at the receiver are **unpredictable,** particularly at **bass frequencies**.
- These issues mean that **automatic loudness measurement and control for broadcast will always be approximate**.
- However, it is still important to **minimize the average error** by choosing a loudness meter that exhibits good correlation to the average loudness as perceived by **many listeners in aggregate**.

Automatic Loudness Control

- Automatic on-line loudness control must start with an **objective reference**:

  A **loudness meter** whose indications closely match **subjective loudness as perceived by listeners**.

- The listeners being tested should **match the typical demographics of television viewers**, in age and gender.

- Both the BS.1770 and J&T meters have been tested in this manner.
  - The **BS.1770** meter exhibited a **worst-case disagreement of more than 5 dB** with listeners.
  - The **J&T** meter exhibited a **worst-case disagreement of 3 dB** with listeners, although a smaller set of program items was tested.

Automatic Loudness Control

- To make an **automatic loudness controller**, one can **insert an loudness meter into a gain computer sidechain,** where the sidechain produces gain reduction that is the **inverse of the loudness meter's output above a preset threshold**.
- This topology is **similar to a compressor** except that the **loudness meter is used** instead of a simple RMS or weighted peak detector.

Automatic Loudness Controller Program Context Limitations

- An automatic loudness controller operates with reference to an **absolute subjective loudness threshold** that does not adapt to program context as well as a human mixer.
- For example, if there is a **transition between very quiet program material** (like footfalls through rustling leaves or quiet underscoring) and a **commercial**, the commercial may still **seem offensively loud** even though the loudness controller is controlling its loudness correctly with reference to other sounds that reach full-scale loudness. For this reason, **mixers have learned to begin and end program elements with "bumpers"** that are intended to be at the **same loudness as previous or succeeding commercials** and other non-program material.
- While **automatic speech/non-speech discrimination** can help a loudness controller understand context, it cannot deal with all situations (like the examples above, where adjacent elements are both "non-speech").

Comparing On-Line Processing Algorithms

- **2-Band compression**: Does not control loudness well enough to avoid viewer annoyance in TV audio.

- **2-Band compression + Loudness Control**: Loudness control that mostly preserves the spectral balance of the input.

- **AGC + 5-Band compression + Loudness Control**:
  - Most effective loudness control
  - prevents audible gain pumping caused by spectral gain intermodulation.

AGC+5-Band Compression + J&T and BS.1770 Loudness Control

- The Jones & Torick Loudness controller can be combined with a **"BS.1770 Safety Limiter."**
- Located after the J&T Loudness Controller, the BS.1770 Safety Limiter **constrains the reading of the BS.1770 meter to a preset threshold (0 to +6 LU) with respect to the Target Loudness (dialnorm)**.
- The limiter's **10-second attack time** minimizes (but cannot eliminate) "loudness ducking" on material with low peak-to-RMS ratio. Loudness ducking is an **inevitable side effect of relying on the BS.1770 algorithm** to estimate the loudness of such material.
- The limiter's **3-second release time** prevents dialog that follows a loud commercial from being too quiet for an annoying length of time.
- The limiter's asymmetrical attack and release times can sometimes **cause the BS.1770 meter indication to overshoot**. However, using symmetrical attack and release times would be **perceptually inferior**.

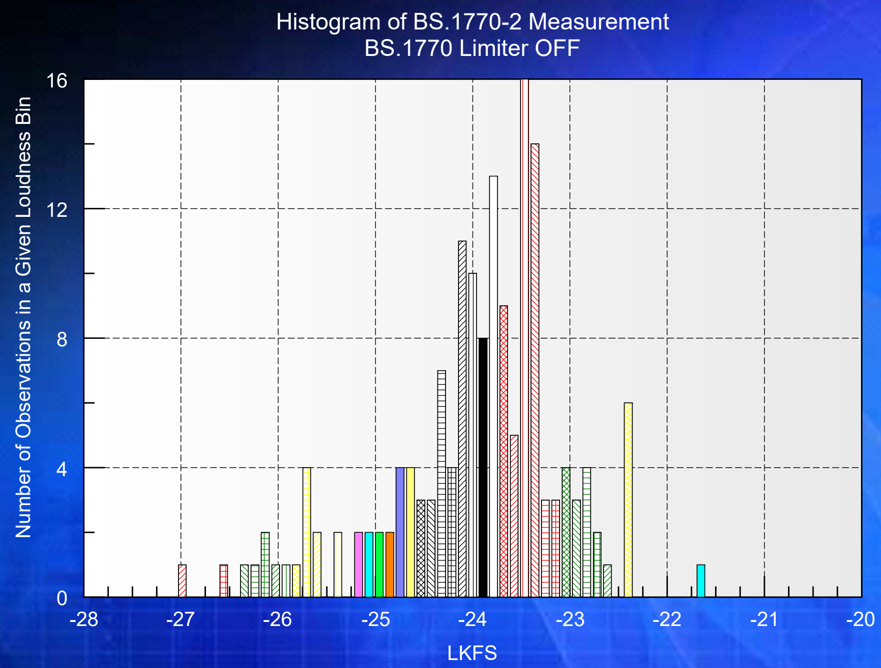
AGC+5-Band Compression + J&T and BS.1770 Loudness Control

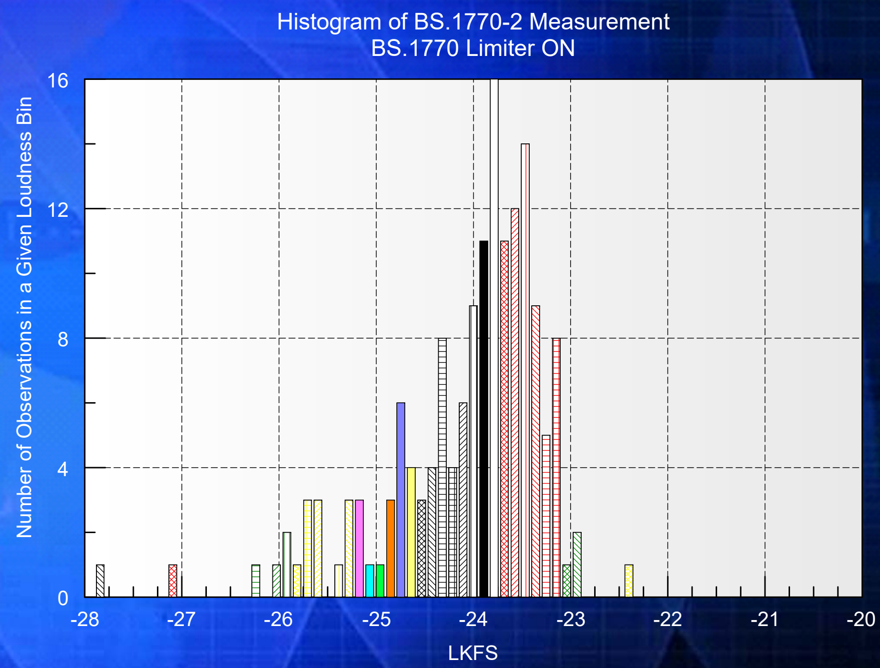
- When the J&T Loudness Controller is placed before the BS.1770 Safety Limiter, the **J&T controller prevents the BS.1770 controller from unnaturally increasing the level of unadorned dialog**. This is because **the J&T controller locks onto dialog better than BS.1770**, particularly when the dialog is mixed with music and/or effects.

- **"Inverse BS.1770" gain reduction sounds unnatural** when used by itself:
  - It will **subtly modulate dialog levels when underscoring or effects appear behind the dialog.**
  - Highly produced material with **low peak-to-RMS** ratio will be **quieter than dialog**.

- **Recommendation**: Use a BS.1770 Safety Limiter if **controlling overall loudness is more important than achieving the best subjective source-to-source consistency of the "anchor element"** (usually dialog). Otherwise, use the J&T Loudness Controller alone.

Comparison: BS.1770 Safety Limiter Off and On

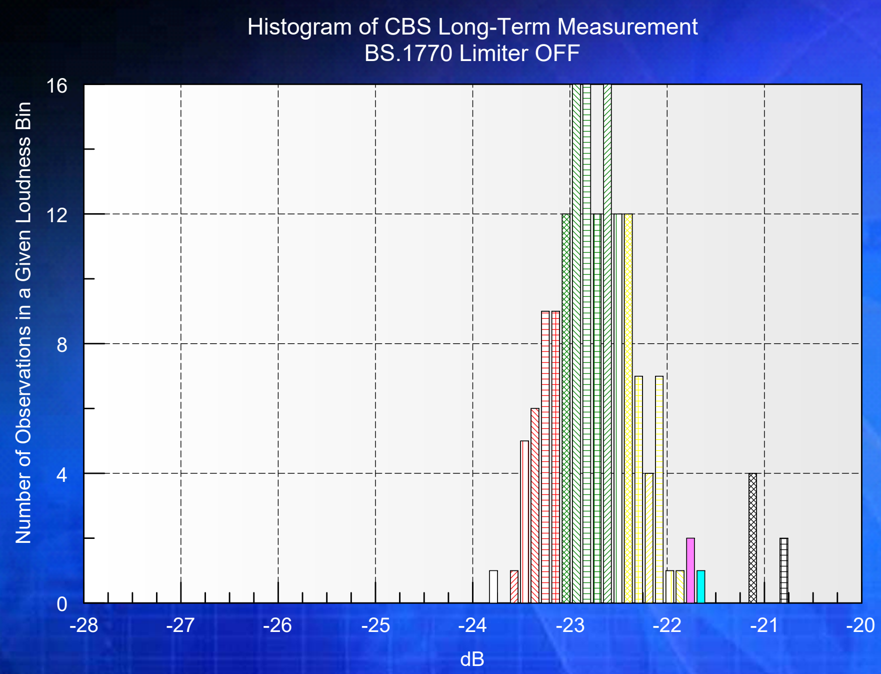


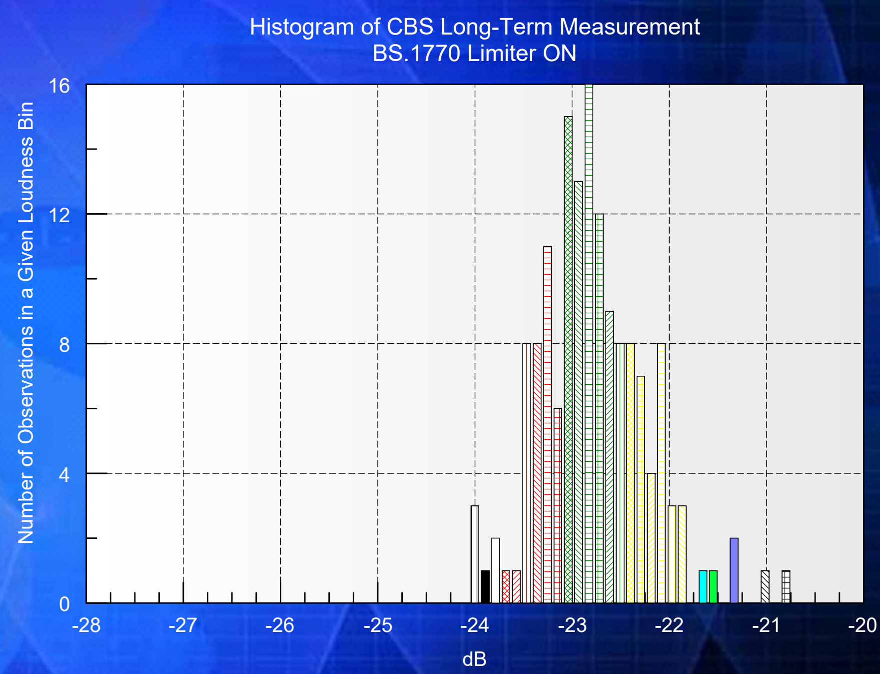


BS.1770-2 Meter with 10-second Integration Time

Loudness Histogram

Comparison: BS.1770 Safety Limiter Off and On





J&T Long-Term in 10-second period

Potential Pitfalls

- If not **optimally designed**, on-line loudness controllers can introduce **objectionable audible artifacts**:
  - **"Spitty" dialog** with hollowed-out midrange caused by **inappropriately designed multiband compression**.
  - **Ambience pumping and breathing** caused by **poor or no silence gating** in the compressor. (Ironically, a problem **first solved in 1959** by the CBS Labs Audimax!)
  - **Stereo image shifts** caused by **unsophisticated gain coupling** between audio channels.
  - **Slow pumping of loudness** caused by using **loudness meter time constants** in an **loudness controller sidechain**.
- Mechanical reliance on the BS.1770 meter can cause **inconsistent loudness between program segments**, although **inconsistency usually does not exceed 3 LU**. While this is within the +2/-5 LU "comfort zone" defined in ATSC A/85, it can nevertheless give the impression that the **broadcast is sloppily produced**.

Loudness Normalization: Streaming & Digital Radio

- There is **no noise penalty** for lowering average modulation.
- Compared to analog FM, **a much stronger argument can be made in favor of loudness normalization.**

Loudness Normalization: Streaming & Digital Radio

- However…loudness normalization must be done in a **subjectively benign way**.
- The BS.1770 meter **penalizes formats specializing in music (like Dance/Techno/etc.) that was highly dynamically compressed in production**. It can easily over-indicate the loudness of such material by 3 dB.

Loudness Normalization: Streaming & Digital Radio

- If the BS.1770 meter is relied upon to balance the music and announcers/presenters, this can cause the **announcers to be substantially louder than the music**.

- For some formats, short-term "Inverse BS.1770" loudness control simply sounds wrong, **sucking the impact out of music and causing weird-sounding shifts in loudness between sources**.

Loudness Normalization: Streaming & Digital Radio

- Modern radio-style audio processors have been crafted to achieve **subjectively pleasing loudness balances between sources**. They do not require extra BS.1770-based loudness control, which I believe does more harm than good.

Loudness Normalization: Streaming & Digital Radio

- Setting a **static, very long-term BS.1770 target loudness for a given transmission can often be useful** and is **much better than doing nothing at all**.

- But at most, BS.1770 should be used to set the static output level of the audio processor, **not to adjust loudness between sources without considering genre**.

Loudness Normalization: Streaming & Digital Radio

- Another potential problem is **setting the target loudness too low.**
- iDevices **cannot achieve satisfying levels** into typical earbuds when the target loudness is **– 23 LUFS. There’s not enough gain available.**
- Player devices often have **built-in peak limiters of unpredictable quality**, so it’s unwise to stream material having a extremely high peak-to-average ratio that can trigger these limiters.

Loudness Normalization: Streaming & Digital Radio

- A **good compromise is –16 LUFS**, which allows satisfying listening levels and is still capable of very high subjective quality.
- The AES streaming loudness recommendation specifies a **target loudness of –16 to –20 LUFS.**

Loudness Normalization: Streaming & Digital Radio

- **Let the transmission processor, not the player device, do the peak limiting!**
- The AES recommendation advises setting the **peak limiter threshold to –1.0 dB TP.**
- **–1.5 dB TP** is more appropriate for netcasters using the **HE-AACv2 codec** at **very low bitrates** like **32 kbps.**

Loudness Normalization: Some Radio Measurements

- Multiband radio-style audio processing achieves **satisfactory source to source consistency without need for extra loudness control**.
- Orban processors **take genre into account**, and have **automatic speech/music discriminators**.
- In our experience, **the Jones and Torick Loudness Meter takes genre into account better than BS.1770**, so it can be used **guide loudness balances between genres** without needing "correction offsets."

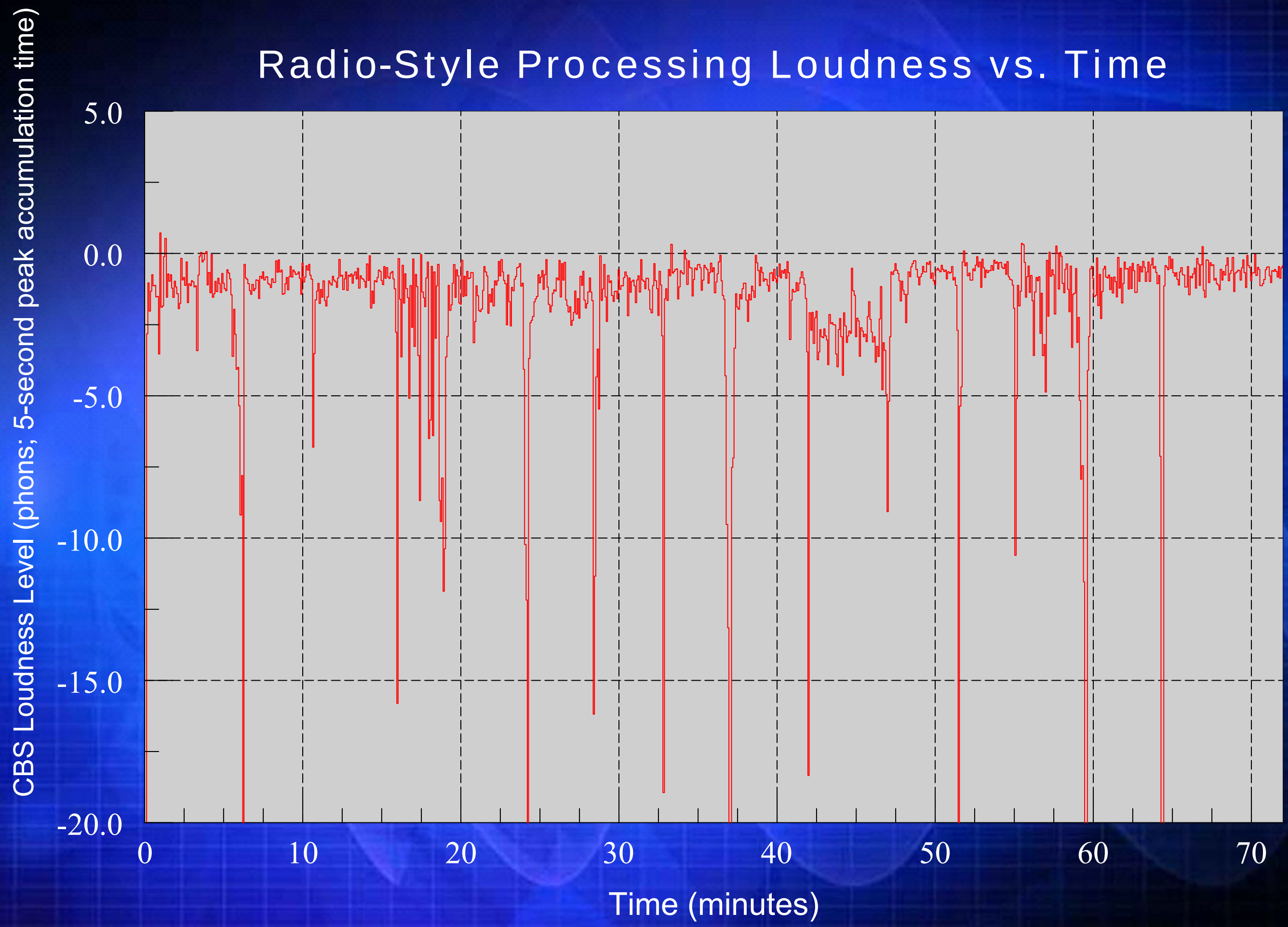
Loudness Normalization: Some Radio Measurements

- The source of the measurements was a 17 minute recording from the program line of KBIG, Los Angeles with most of the music edited out.
- The recording consists of a wide variety of material, including male and female voice, spots, promos, a traffic report from an aircraft, and some music.

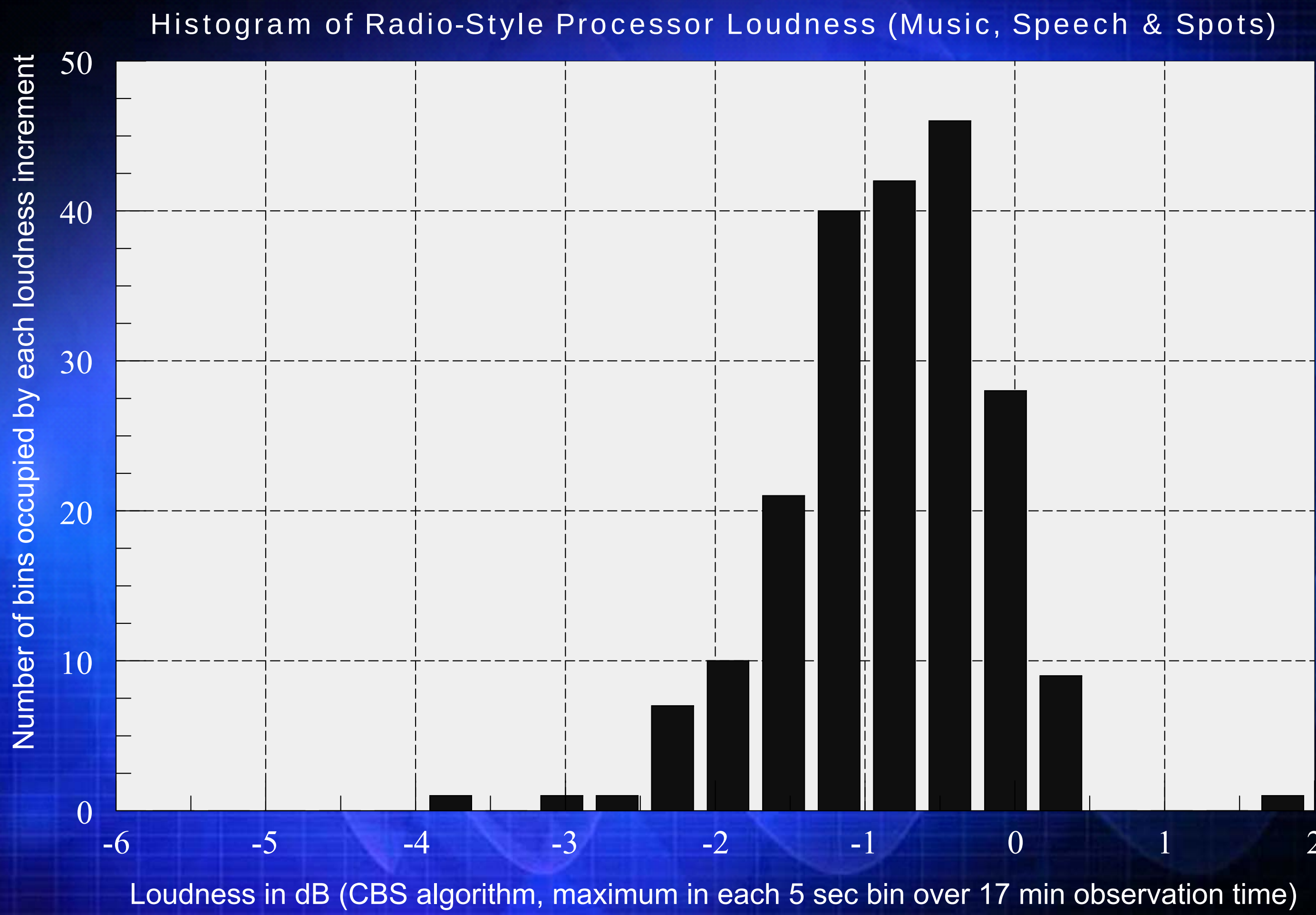
Loudness Normalization: Some Radio Measurements

- Audio processor was an Optimod-FM 8500 running the LOUD-HOT preset.
- The first plot is **loudness vs. time**, using the second-generation Jones and Torick loudness meter.

Loudness Normalization: Some Radio Measurements



Loudness Normalization: Some Measurements



Summing UP 1

- **Take ATSC A/85's prime directive seriously:**

  "Because loudness is a subjective phenomenon, ***human hearing is the best judge of loudness."***

- Relying solely on BS.1770 without listening is a **recipe for substandard source-to-source consistency**:
  - **Dense material** will often be more than **3 LU quieter than unadorned dialog**.
  - Dialog levels will **vary** depending on the **amount of underscoring and/or effects in the track**.

Summing UP 2

- The **J&T** loudness meter (and loudness controllers based on it) **tend to lock onto dialog**.
- The BS.1770 meter indicates the **approximate overall loudness of the program**, although it tends to **over-read material with a low peak-to-RMS ratio, so it is necessary to take genre into account.**
- **If dialog levels are held constant**, the BS.1770 meter will indicate that **dialog mixed with underscoring or effects is louder than unadorned dialog**, even though the **dialog levels have not changed**.
- The BS.1770 Short-Term measurement (3-second integration time; ungated) is particularly prone to this behavior and **should not be used as the sole reference for an automatic loudness controller**.

Summing UP 3

- Automatic loudness control is **unlikely to ever be as good as a human mixer** when the most esthetically pleasing results are desired. Only humans can understand the **subtleties of context**.

- **Cascading a J&T loudness controller and a BS.1770 "overshoot limiter" is often a good compromise**. The J&T controller **prevents unadorned dialog from being unnaturally pumped up in loudness**, while the BS.1770 controller catches material whose **overall loudness might be considered excessive,** depending on the loudness control philosophy of the broadcaster.

- File-based loudness control is more likely than on-line loudness control to **create loudness inconsistencies at the boundaries between program elements**.

Summing UP 4

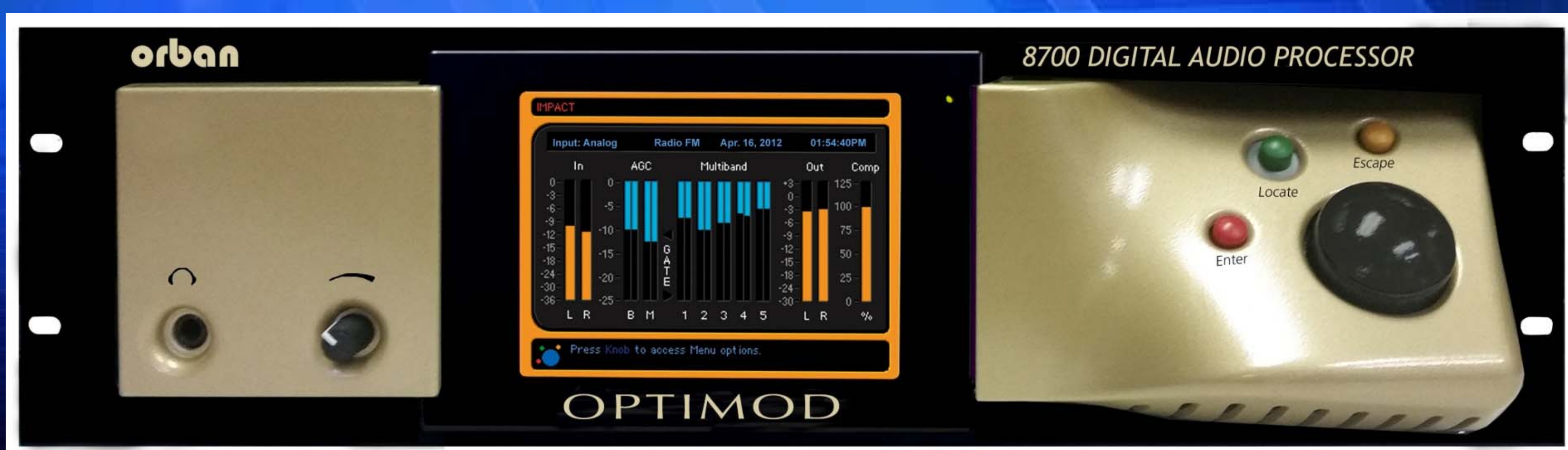
- In digital transmission channels, **static loudness normalization using BS.1770 is much better than doing nothing at all**.
- -16 LUFS matches current popular players better than -23 LUFS. AES TD1004.1.15-10 recommends target loudness of -16 to -20 LUFS.
- Modern radio-style processors provide **satisfactory source-to-source consistency without needing additional dynamic loudness control**.
- **All receivers have volume controls, and all listeners know how to use them**.

Summing UP 5

- Providers of program streams via digital media should be strongly encouraged to **statically normalize the loudness of their transmissions** to make then consistent with others.
- If providers are not using online audio processing, then it becomes necessary to **normalize each program element before playout**.
- But strict normalization to BS.1770 will **penalize some program elements, formats, and styles of processing by making them subjectively quieter than the media's target loudness**.
- **Take genre into account** per AES TD1004.1.15-10, and **trust your ears if they disagree with the meter**.

Optimod 8700i:

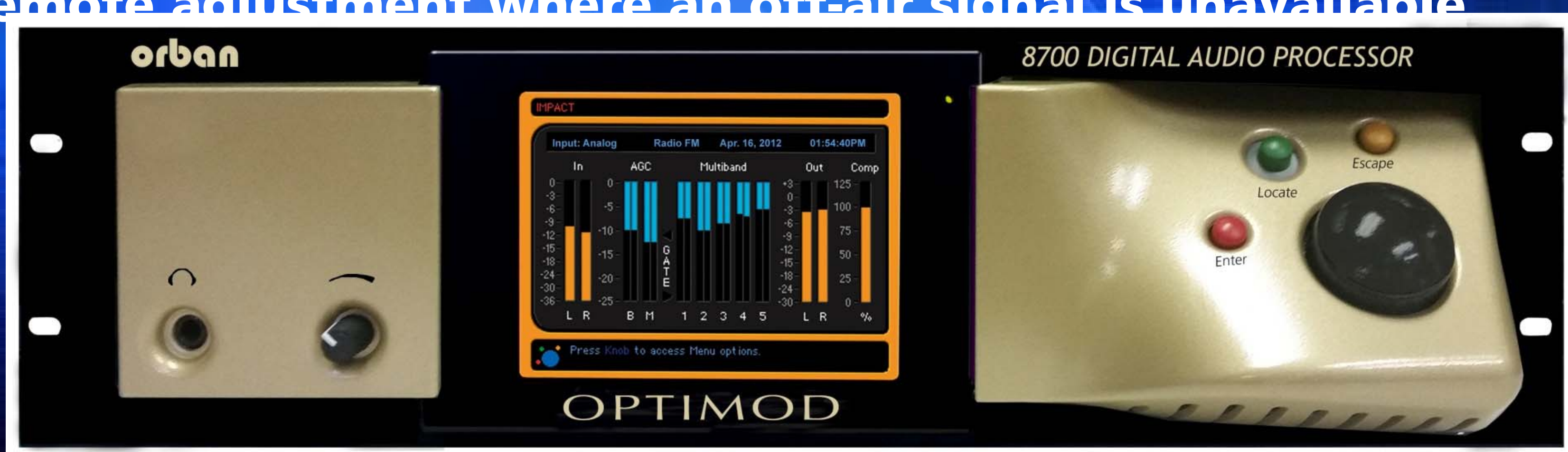
Orban's flagship FM processor evolves.



Optimod 8700i

All the features of Optimod 8600 and more:

•Built-in **Dante** (100% **AES67**-compatible) **Audio-Over-IP networking**.
•**Dual power supplies** with separate line cords.
•**Standard digital MPX connection:** 192 kHz AES3 output
•**Two digitized SCA inputs** allow SCAs to be included in the digital MPX signal.
•Built-in **MP3/OPUS streaming of the processed output** to allow **remote adjustment where an off-air signal is unavailable.**



Optimod 8700i

All the features of Optimod 8600 and more:

•Program-adaptive **subharmonic synthesizer**:
 creates **punchy bass without over-enhancement**.

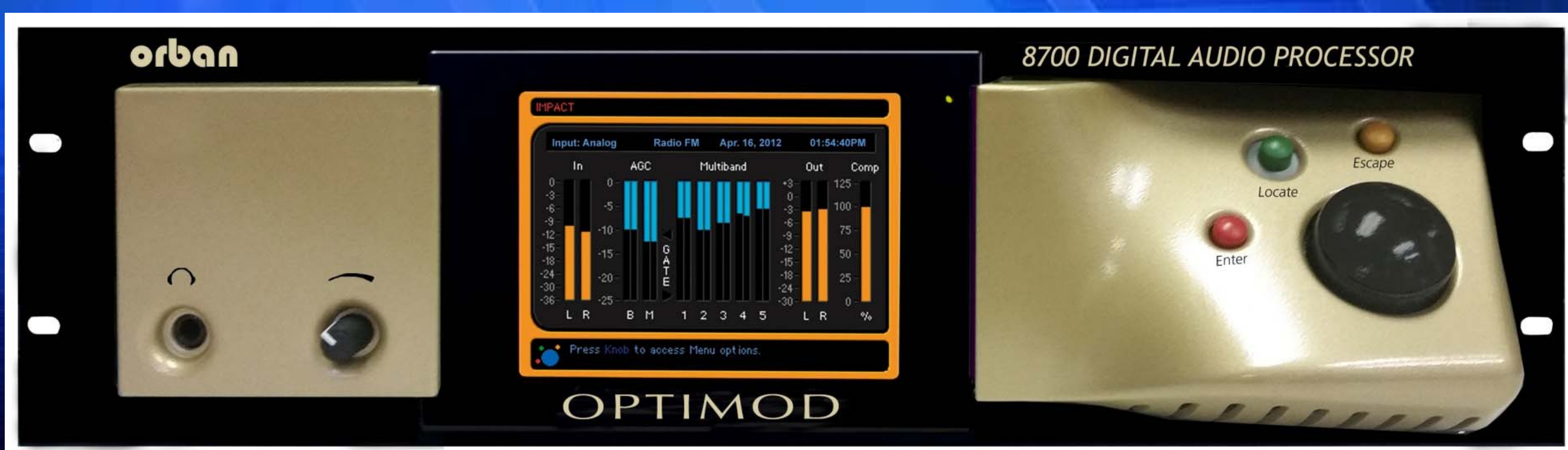
•**Xponential Loudness:** Psychoacoustic processing that **reduces listening fatigue**, particularly from "hypercompressed" source material.

•Low-delay monitoring now includes **peak limiting** to better **simulate the final on-air sound** through talent headphones.

Optimod 8700i

Features Inherited from Optimod 8600:

- Orban’s exclusive **Multipath Mitigator** phase corrector.
- Built-in **RDS generator** supports **dynamic PS**.
- Full **remote control**, including **telnet** connections to accept **simple text strings** from an **automation system** or other sources.
- **SNMP support**: Monitor the 8700i over your network.
- **Reliable architecture** using **dedicated DSP chips** for audio processing.



Optimod 8700i

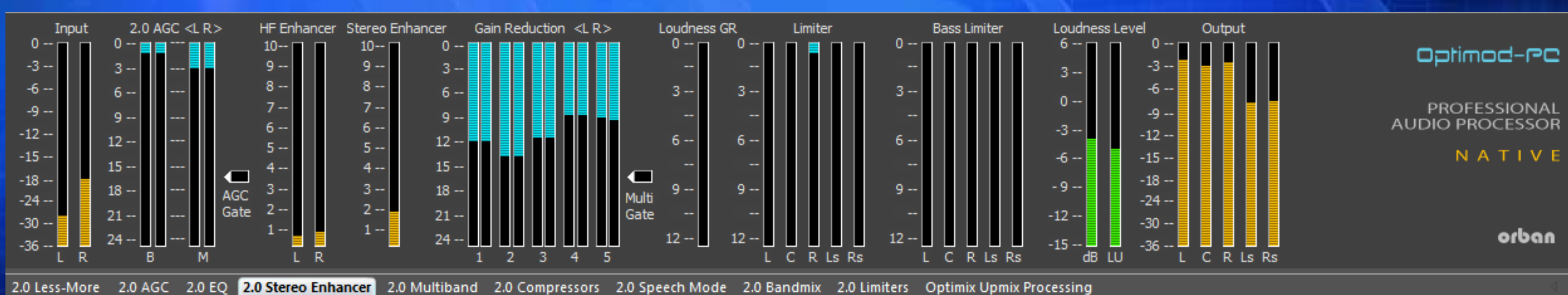
Shared With Optimod 8600:

•Orban's **MX-technology peak limiter** uses a **psychoacoustic model** to **control distortion**, **increase transient punch**, and improve **high frequency power handling**.
•**Two-band** and **Five-Band** processing.
•Advanced two-band **window-gated AGC**.
•**The "Optimod Sound": proven to help attract and hold audiences**.

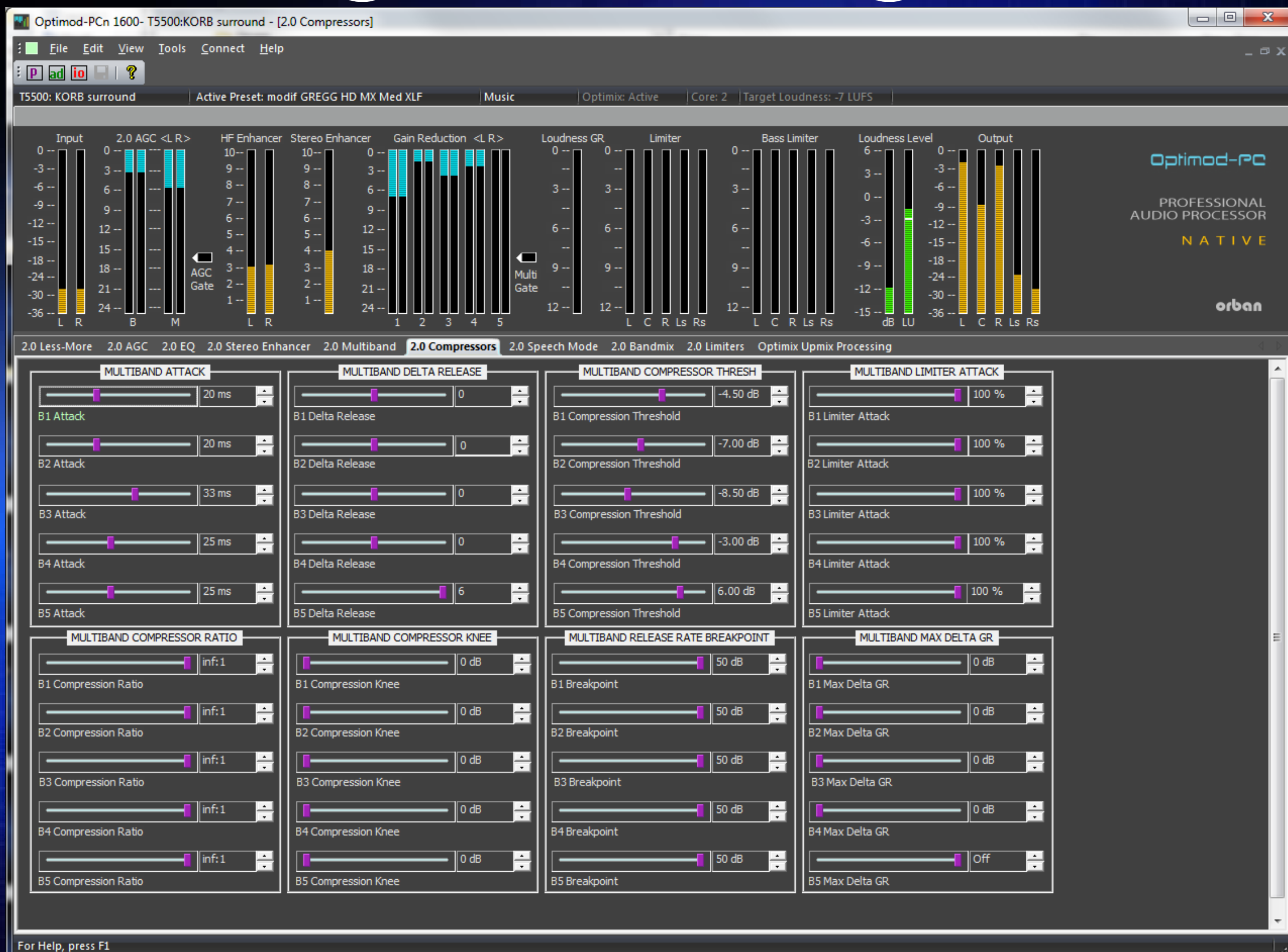
Orban Optimod-PCn 1600:

Advanced Audio Processing

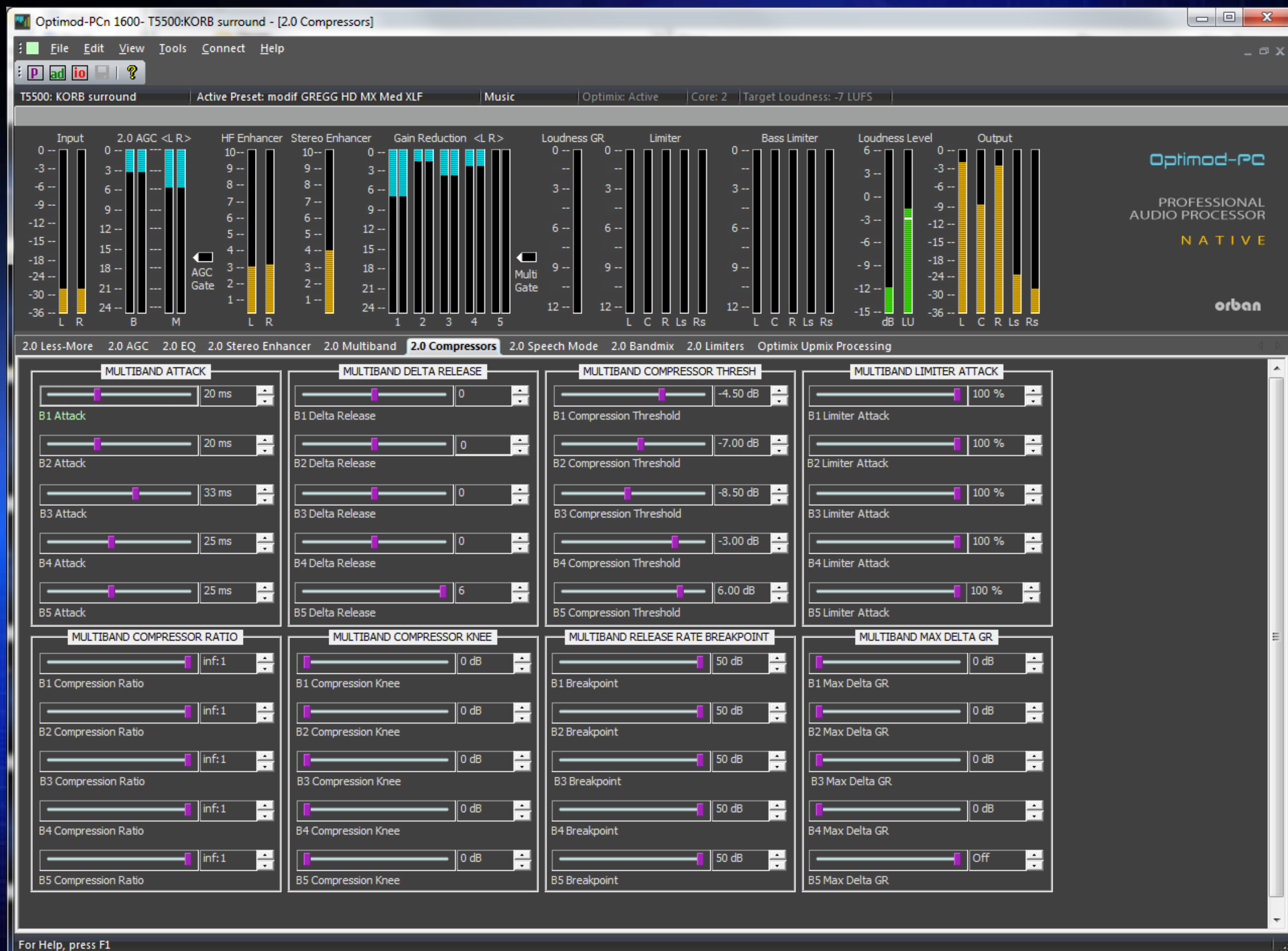
Running Native on Intel/Windows PCs



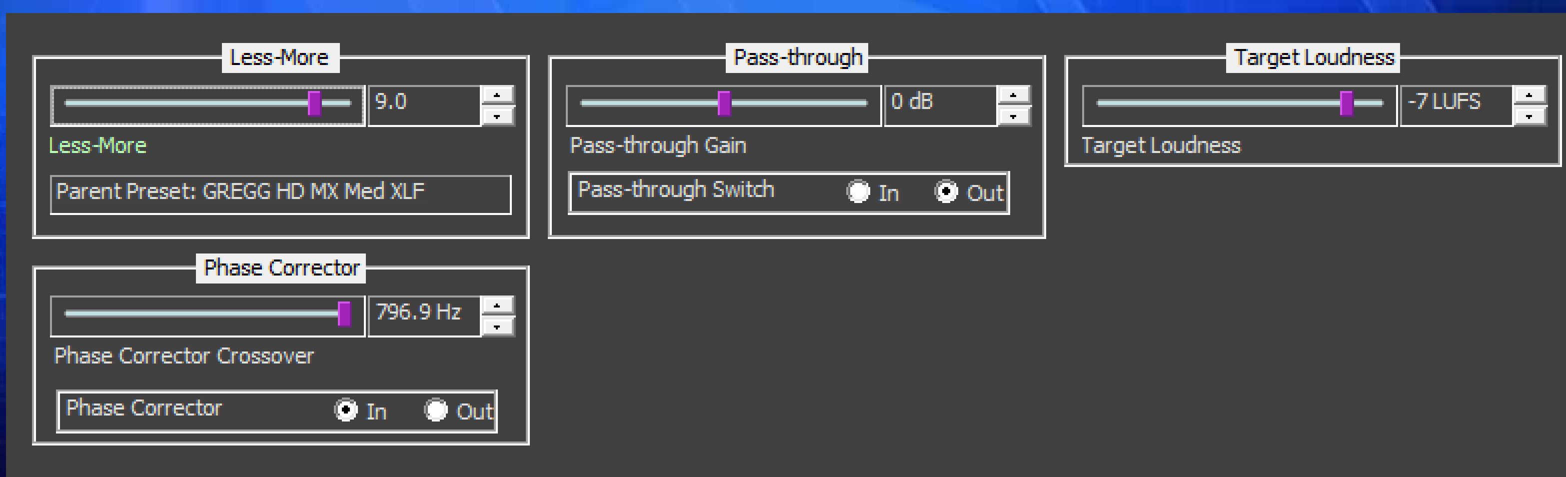
Orban’s Most Advanced Streaming/Mastering Processor



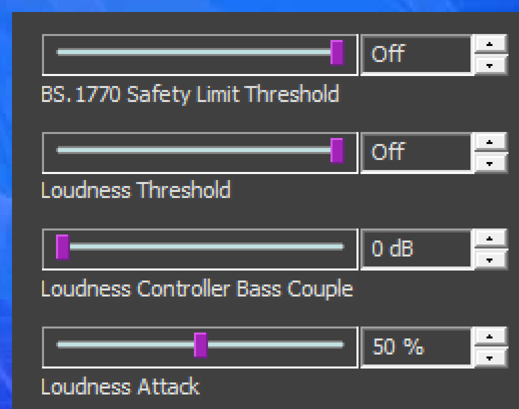
All of the processing features of Orban's Optimod-PC, plus more!



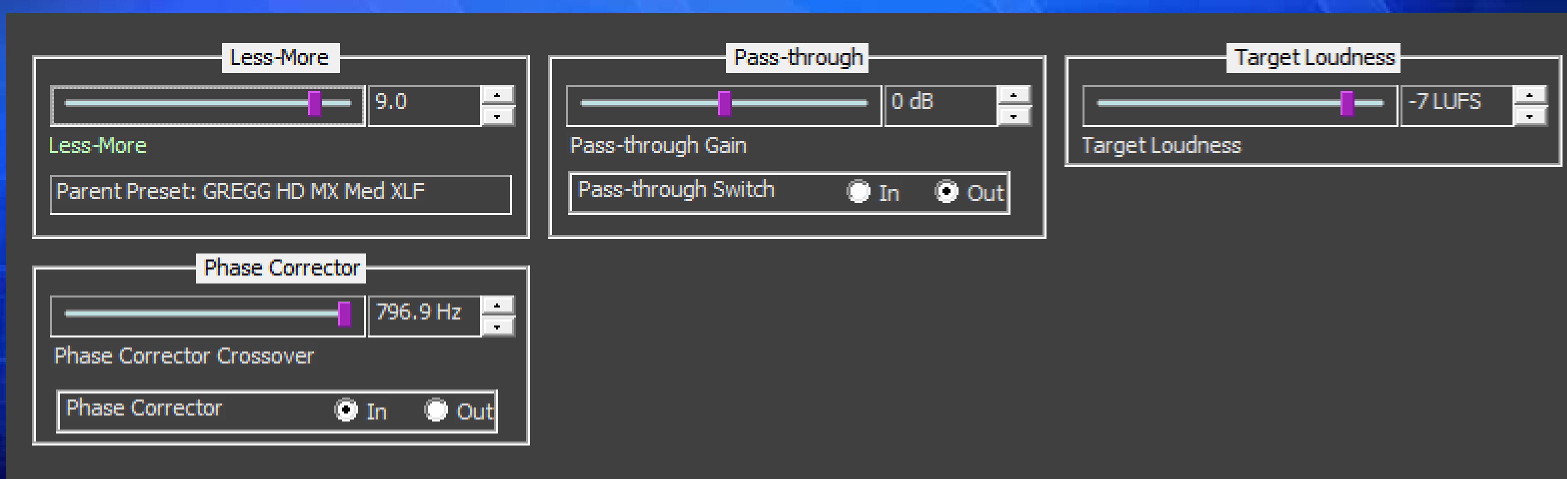
Supports modern “**target loudness**” (BS.1770) concepts: **EBU R 128** and **ATSC A/85**-aware.



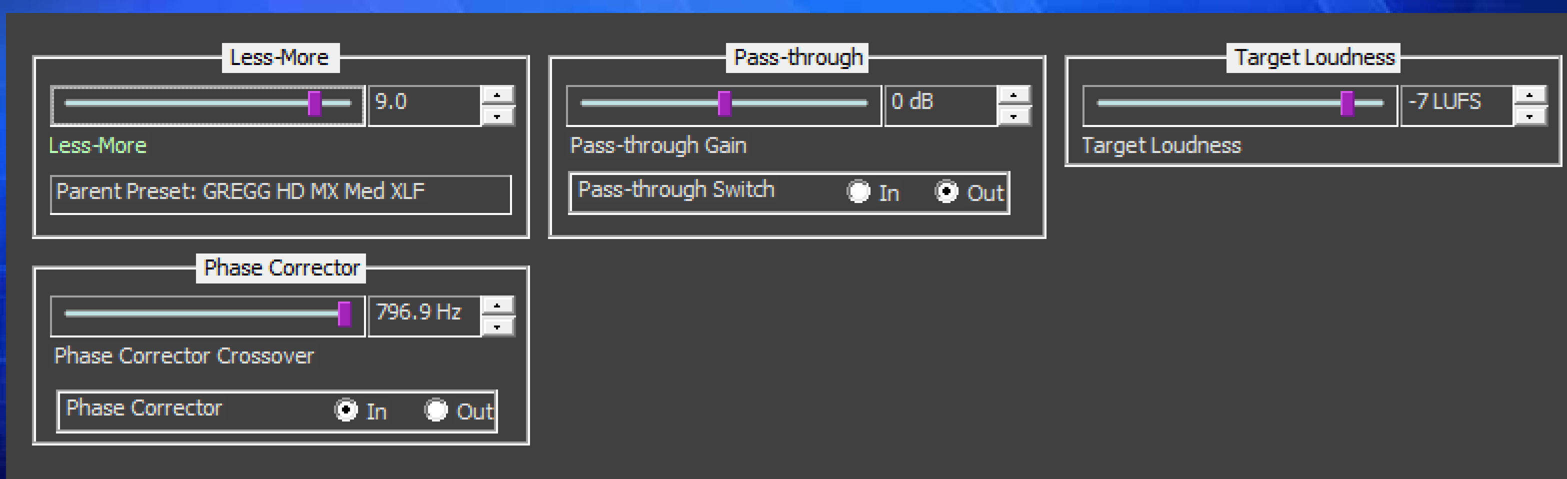
Built-in BS.1770 and J&T **Automatic Loudness Controllers.**



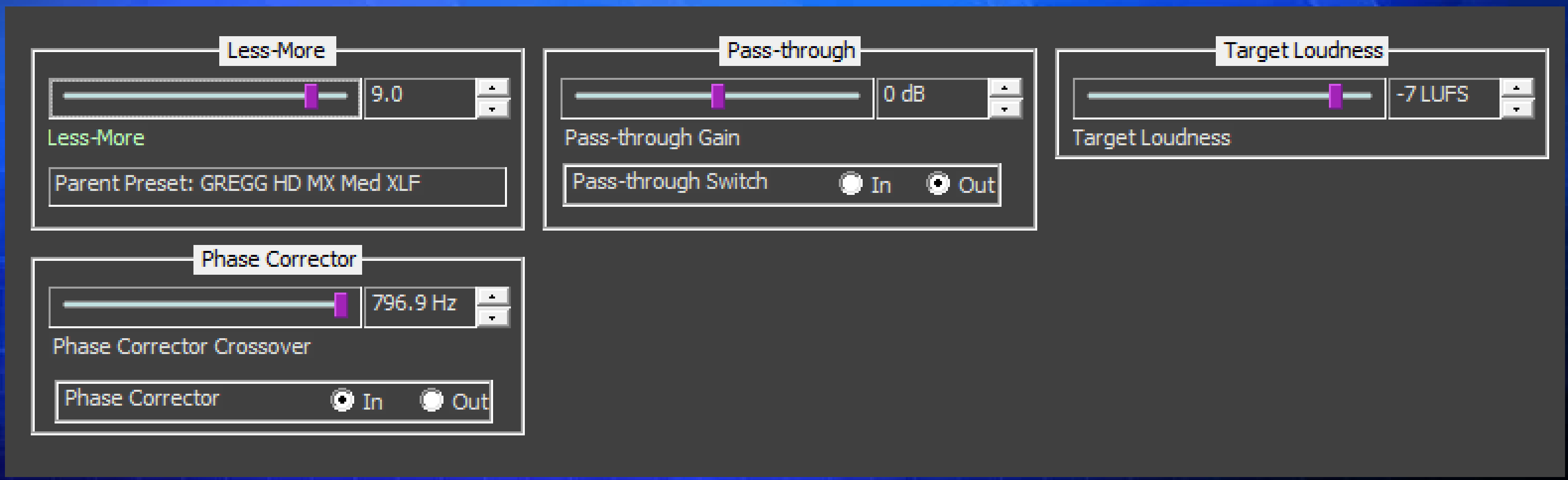
Easy LESS-MORE adjustment of processing presets.



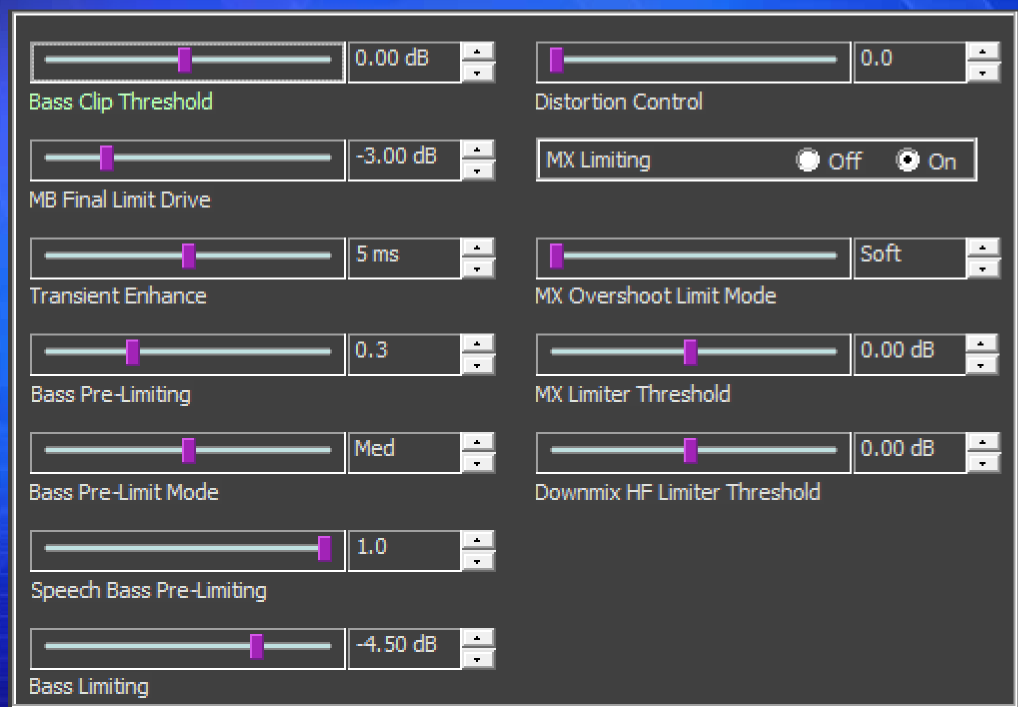
Pass-through Mode allows you to smoothly **bypass the processing** with a **delay-matched crossfade**.



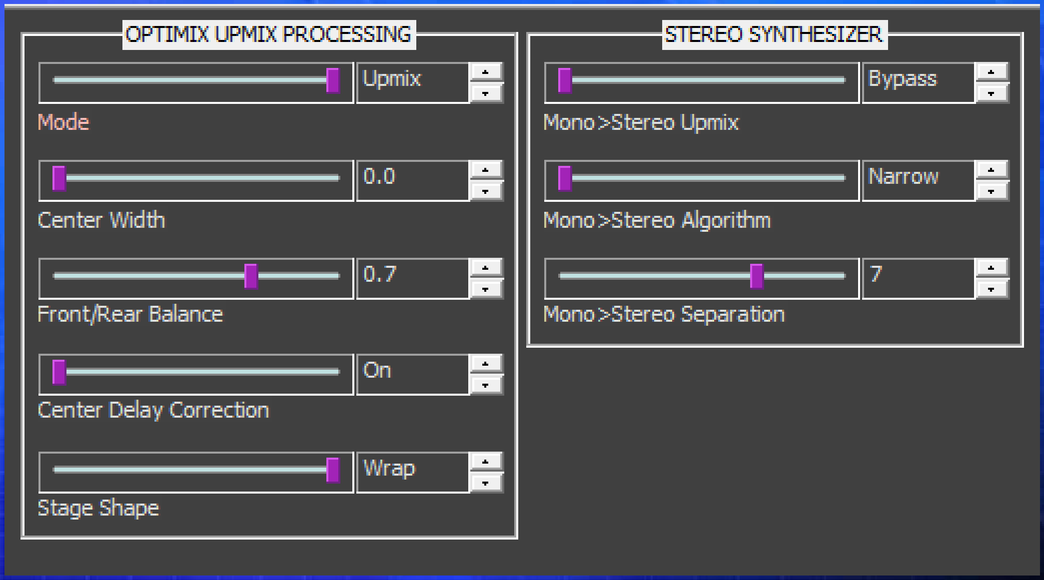
Phase skew corrector corrects phase cancellation in the mono sum.



MX peak limiting adds **crispness** and **punchiness**, even when **processing for loudness**.



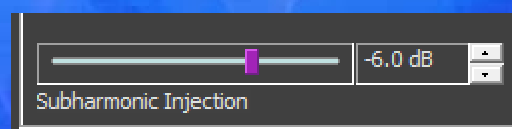
Orban's **Optimix®** 5.1 surround upmixer creates "**Instant Surround**" from stereo or **even mono**!



Mono bass and **steep-slope crossovers** are available to let you **customize bass** like never before.

Program-Adaptive Subharmonic Synthesizer

adds bottom and punch to old recordings.

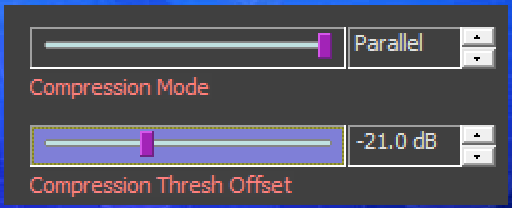


Parallel compression

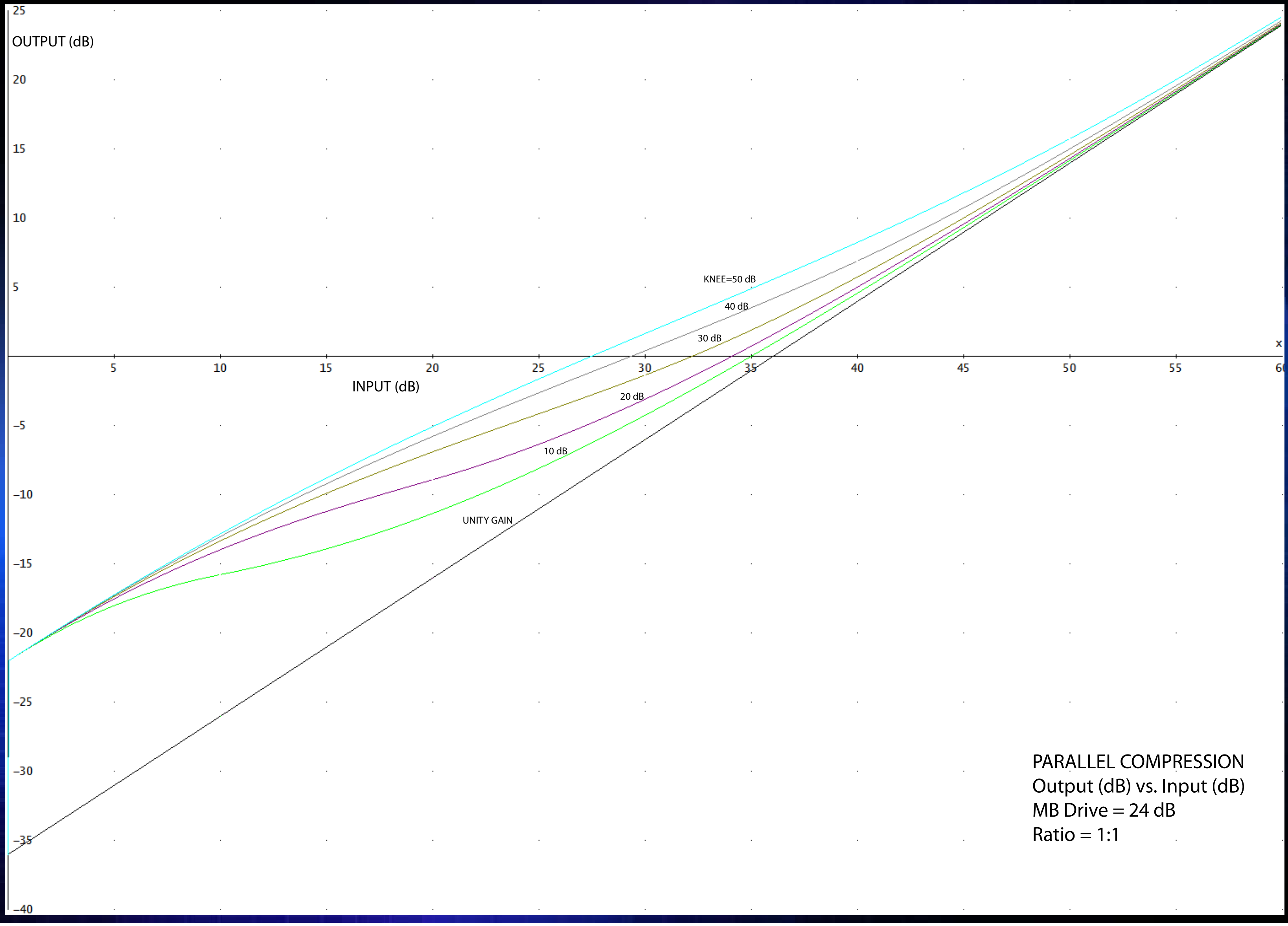
-Subtly **amplifies quiet material without affecting the punch of loud material**.

-Use as a **pre-processor ahead of Optimod-FM** for almost any format.

-Great for **classical music**!

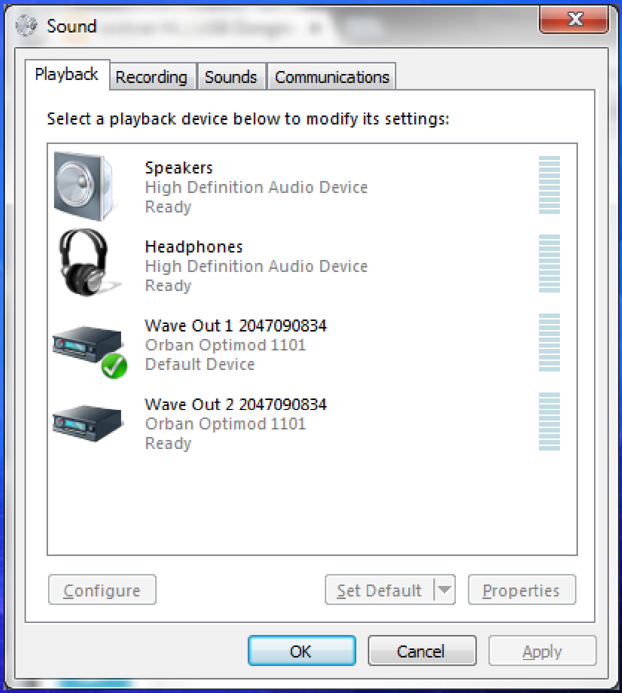


Parallel compression



- Uses **standard Windows MME and WASAPI audio I/O**: compatible with most soundcards.

- With third-party drivers, supports **AES 67-compliant audio-over-IP connections.**



- USB key authorization makes it **easy to move processors between main and backup computers**.
- **The processors move with the key.**
- **No Internet connection or re-authorization is required.**

- Optimod-PCn processing runs as a **Windows Service**. It is controlled by **separate PC Remote software** running on the Service host computer or **elsewhere on the network**, and connected to the Service via **TCP/IP**.

- TCP/IP access to the Service can be protected by **multiple levels of security**. You have **complete access control**.

- **Install the Service on a primary and one or more backup computers**. If the primary computer fails, **move the key to a backup,** re-route the audio I/O, start the Service, and you're **up and running**.

- **PC Remote software is not copy-protected**. Install it on as many computers as you want.

Cost-Effective

- Run up to **16 highly advanced audio processors** on one computer with **dual Xeon processors**. Run up to eight with **inexpensive i7 hardware**.
- Combine with **playout systems** and **streaming software** like Modulation Index's StreamS to create up to 16 integrated "**Internet radio stations in a box**."
- This product is available exclusively through Modulation Index and its dealers. www.indexcom.com

Thanks for your attention!