Digital Entertainment

Perceptual Loudness Management for Broadcast Applications

White Paper

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Executive Summary

Broadcasters and consumers alike need a solution to abrupt changes in loudness that send viewers reaching for remote controls during commercials and action scenes. Since most devices take a heavy handed approach to loudness adjustment that renders audio lifeless and unexciting, broadcasters typically rely on metadata and consumer technology to solve the problem and allow audio to pass-through without adjustment. Perceptual Loudness Management (PLM) is a new approach that addresses the unique way human ears perceive loudness. It leverages best-in-class measurement tools and single-band correction to help broadcasters and their engineers effectively manage loudness across the workflow.

The Need for Perceptual Loudness Management

The consumer search for relief from annoying and inconsistent audio levels is finally forcing action. Task groups are working to develop new loudness standards (ATSC, 2009). In 2009, the “Commercial Advertisement Loudness Mitigation Act” was introduced to address and regulate problems associated with loud TV commercials in the US. This effort is not limited to the US. Consumers and broadcasters are looking for solutions worldwide. Clearly, broadcasters need to respond to demands for a better loudness solution, but many still lack a clear understanding of how consumers perceive loudness and the technology available to consistently regulate it.

Last century’s loudness solution was a fix-all audio box at the end of the broadcast chain that created the undesirable side effect of removing essential audio dynamics. A new solution has emerged that enables engineers to measure loudness correctly and define appropriate loudness rules on the fly, imitating methods used by audio mix engineers actively riding their faders. The solution utilizes technology that is easily integrated with existing broadcasting workflows and metadata, delivering the flexibility engineers need to adjust loudness levels in real-time without removing signal dynamics.

This paper provides insights into the critical components of audio loudness correction, describing where existing methods fall short and how Perceptual Loudness Management delivers on the promise of more transparent, real-time loudness management. It presents relevant changes in broadcasting technology with a view toward understanding the gap between industry standard audio measurement and broadcast delivery systems and why these do not adequately address how the human ear perceives loudness. Finally, the concept of PLM is fully presented as a way to satisfy both listeners and regulators with a fully dynamic audio experience designed for the consumer’s acoustic environment.

Digital TV and the Metadata System

Broadcasters currently employ the digital TV (DTV) and metadata systems introduced in the early 1990s. Many thought the DTV standard (paired with a digital audio standard) would usher in an era of greater fidelity that would facilitate better loudness control. Unfortunately, the DTV audio signal processing approach allows advertisers to transmit audio at levels that are higher than the rest of the signal.

Before DTV, viewers with big home theaters and those with small portable TVs experienced the same audio dynamics since broadcasters used real-time audio leveling to maintain target audio levels before the audio was transmitted (Figure 1). The drawback to this system was that audio had to be processed for the average viewer, which reduced the audio signal dynamics below the optimal performance of the home theater while failing to offer enough dynamics control required for portable TV viewing where intelligibility was more important than fidelity.

DTV introduced metadata, an approach designed to satisfy both the original content creator and the consumer. Metadata describes the audio it accompanies without processing it, so broadcasters do not need to use real-time signal processing to permanently modify signal dynamics and consumers can choose how they want to process audio in their homes. Consumers desiring lots of dynamic range in a home theater can turn off Dynamic Range Control (DRC) while consumers with small audio systems or those listening in noisy environments can enable DRC settings.

The metadata system uses a Dialog Normalization level (Dial Norm) inserted by content creators or broadcasters to describe the loudness level of the average anchor element (often dialog) within content. Consumer devices can read the Dial Norm value and add or decreases gain to help control loudness. Assuming the dialog in each clip of content is described with the correct Dial Norm value, consumers shouldn't need to manually adjust volume.

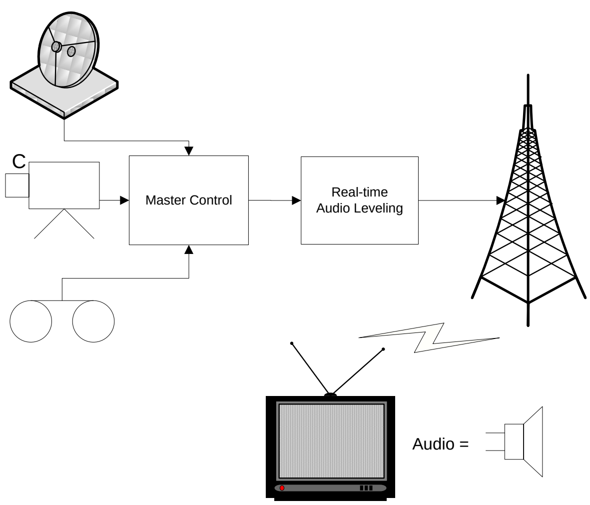

Figure 1 - Traditional broadcast audio processing

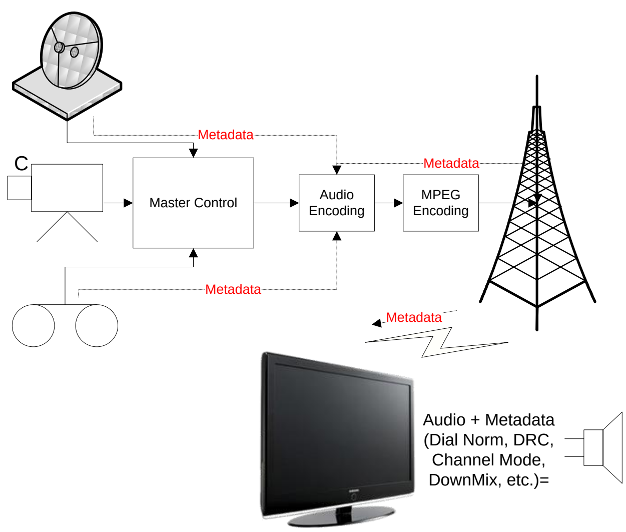

Figure 2 - DTV broadcast signal chain with metadata

Unfortunately, advertisers discovered that broadcasting a loud commercial was as easy as transmitting audio at a higher level while describing the Dial Norm at a lower level. The resulting commercials were much louder than the surrounding content. Also, if content is correctly authored with metadata yet the metadata is lost in the transmission chain or the audio levels are changed without considering metadata, new metadata must be re-authored pre-broadcast. Loss of metadata is fairly common since it travels separate from audio in most broadcast facilities. Broadcasters unaware of the details of this system often allow the final audio encoder's default settings to dictate the newly assigned metadata.   The result is: Content + Wrong Metadata = Inconsistent loudness levels. Lastly, most consumers are not aware of audio controls in equipment setup sub-menus like DRC. Even if content is prepared correctly, incorrect home settings can create wildly varying loudness results.

Measuring Human Hearing and Loudness

Loudness solutions should measure loudness the way human ears perceive it. Human perception of audio signals (the sensation level) is properly called "loudness." Humans do not hear using standard measures of intensity. Therefore, correcting audio using volume management tied directly to these measures will not agree with human perception. As early as 1933, research by Fletcher and Munson showed that human hearing sensitivity varies based on frequency and sound pressure level (Fletcher & Munson, 1933).

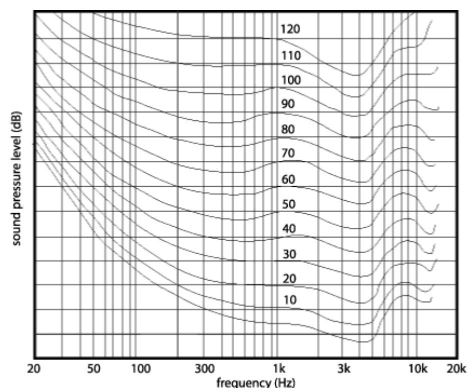

Figure 3 - Fletcher/Munson Equal Loudness Curves

To create a measure that more accurately represents the human experience of loudness, frequency weightings (or equalization curves) are often placed ahead of power measures. These frequency weighted power measures have been devised including: Leq(A),(B),(C),(M). The term "Leq" attempts to relate "L"oudness to an "eq"uivalent amount of energy in a standard signal, typically a 1 kHz sine wave. The most current (R2LB) weighting is known as the standard ITU-R BS.1770 measure seen in Figure 4. The proper notation for audio measured with BS.1770 is LU, or LKFS when referenced to the dB FS scale. While this is a valid way to relate loudness to intensity, decibels (dB) are not a natural unit of loudness. The natural unit is "sones" or "phons" (Stevens, 1936).

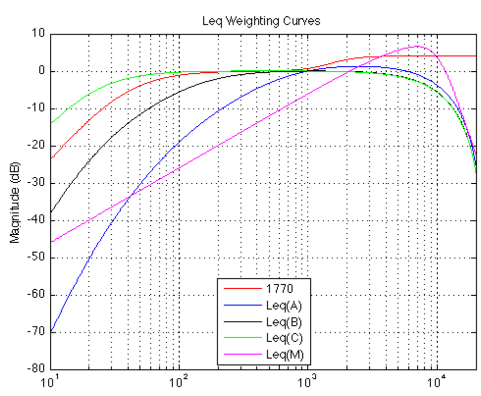

Figure 4 - Leq weighting curves

As shown in Figure 5, each Leq revision results in a different level of correlation between the objective measure and the human listener's assessment of loudness. Notice that the lowest mean error score was achieved by the Neural Loudness Measure (NLM) which will be discussed in the next section.

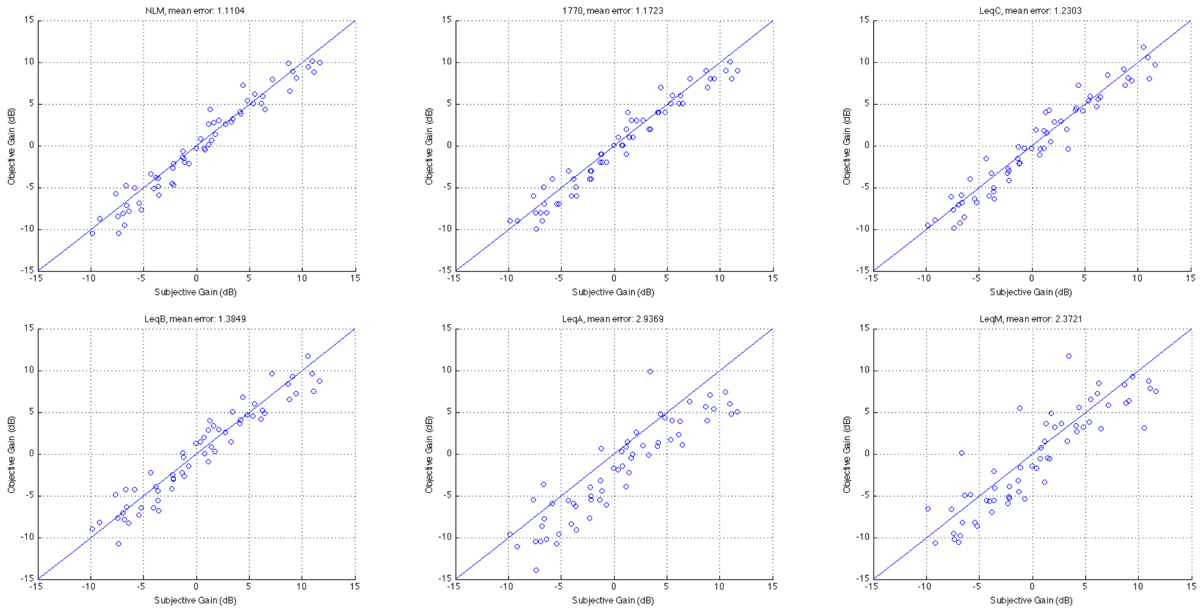

Figure 5 - Subjective (human) vs. objective (measured) correlation

While certain Leq measures show a higher correlation to human perception, there are still many broadcast content types that fail to strongly correlate, and these must be addressed by an effective loudness solution.

For example, a narrowband signal may sound less loud than an equally intense wideband sound, or vice-versa, depending on the relationship between the rendering level, absolute threshold, and signal content. Also, content consisting of large amounts of low frequency energy is often inaccurately measured by BS.1770 due to heavy low-frequency roll-off (see Figure 4). There are several reasons for lower correlation in this type of content. The equal loudness curves of Fletcher and Munson show that the perception of low frequency loudness increases rapidly after reaching an audible level. Simply suppressing energy at low frequencies (as in BS.1770) without accounting for excess gain in loudness versus intensity just above threshold will underestimate the loudness of a low frequency signal. Additionally, small amounts of bass distortion can greatly increase the perceived loudness of a signal as the bandwidth is spread, long before the actual distortion is noticeable. There is an ongoing debate in the broadcast field regarding the relevance of low frequencies, be it in one of the main audio channels or isolated specifically in the low frequency energy (LFE) channel. Some TVs and radios lack the ability to produce low frequency energy, while it is exaggerated in other audio environments like automotive.

A meter running even the newest BS.1770 standard will often vary from a subjective measure, especially in the short term. This variance is sometimes acceptable in a long-term metering application. Loudness meters typically display a measurement value that has been smoothed over many seconds, minutes or hours. Small deviations between the instantaneous perception of loudness and the long-term measured average are acceptable in metering applications. Loudness meters typically offer an audio visualization method that human operators can use (along with their ears) to make decisions about any corrective action required.

Inaccuracies noted in BS.1770 are acceptable when smoothed over long-term averages but become apparent when implemented in instantaneous devices such as real-time loudness control. Loudness control devices must make decisions in the absence of a human operator. They instantaneously measure and correct audio levels in order to bring consistency to the audio output. Any discrepancy between the human perception of loudness and the loudness measure will translate into inconsistent loudness and audible signal level changes. This occurs because human hearing does not have a long-term averaging mechanism for loudness perception (as is often implemented in loudness meters). Instead, the averaging mechanism of the ear is understood to last about 200 milliseconds – not seconds, minutes, or hours. For real-time loudness correction, a more complex loudness model is required in order to react to signal changes like a human operator.

Critical Band Measurement

Another key consideration in designing an effective loudness solution is whether or not to include critical bands in the measurement. Critical bands describe the auditory filters within the human cochlea as seen in Figure 7. Single-band measurements cannot account for complexities in the human auditory system. For example, frequencies within the same critical (Bark) band have different perceived loudness levels compared to equal energy frequencies spread across multiple critical bands.

An example of loudness versus frequency bandwidth is shown in Figure 6. One tone is presented to the human test subject and consecutive tones are added one by one until 25 is reached. Each tone is presented with equal energy in 1 to 25 critical bands. The total signal energy is kept constant regardless of the number of tones. For example, going from one to two tones, the energy of each of the two tones is exactly half that of the single tone. While the amount of audio energy never changes, the perceived loudness increases as the energy spectrum spreads out across more tones (Johnston, 2006).

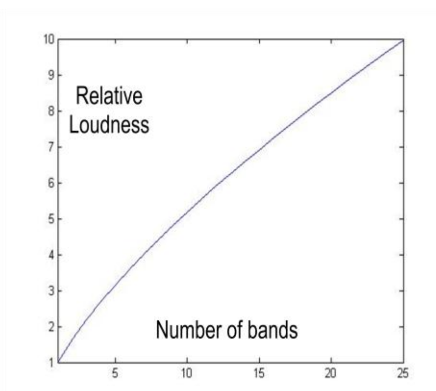

Figure 6 - Loudness and critical band relationship

In this sense, “close” refers to sounds inside an individual cochlear filter where compression occurs, and “farther removed” refers to sounds outside the same cochlear filter bandwidth. Critical band filters are very closely spaced in the ear, with extensive overlap, and a subset of the filters are shown in Figure 7 (Allen, 2010).

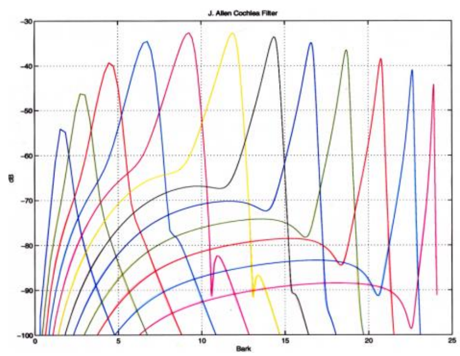

Figure 7 - J. Allen Cochlea Filters

As a step closer toward modeling this behavior, Bronwyn Jones and Emil Torick created a revised CBS Loudness Meter in the early 1980s that consisted of eight filters, each covering three critical bands due to hardware limitations and DSP costs at that time. This approach did prove to deliver better subjective modeling than its predecessors (Jones & Torick, 1982).

In 2004, Skovenborg and Nielsen published an AES paper titled “Evaluation of Different Loudness Models with Music and Speech” that expanded on the idea of critical band loudness measures with a method called HEIMDAL (Nielsen & Skovenborg, 2004). The HEIMDAL method separated spectra into nine bands via an octave band filter bank, avoiding the FFT time / frequency tradeoff. While the HEIMDAL multi-band model did not achieve the complexity of cochlear modeling as seen in the J. Allen example (Figure 7), it was a step in the right direction. The HEIMDAL method had the lowest error compared to any other loudness model tested, including Leq(RLB), which later became a part of the BS.1770 standard. Tests conducted by DTS, Inc., have achieved the same results, as illustrated in the top left graph of Figure 5.

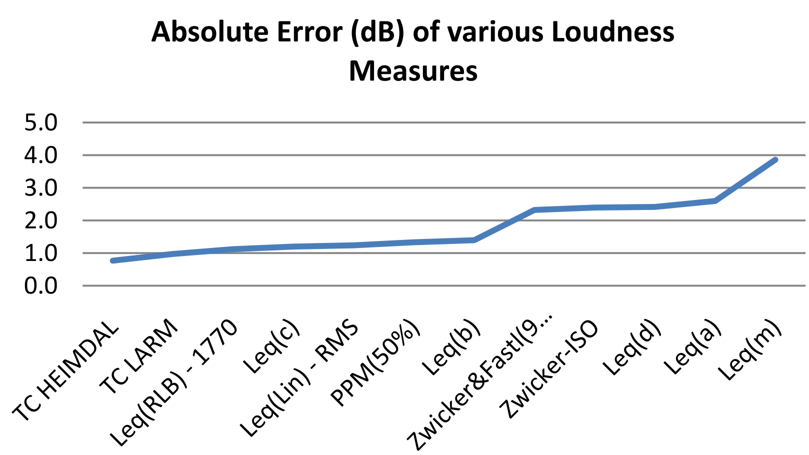

Figure 8 - Loudness measures compared

As DSP resources increase and research advances, cochlear modeling methods remain a cornerstone of all modern-day perceptual audio codecs (including MP3, AAC, and AC3), by removing inaudible components from a signal without perceived loss. Similarly, research in cochlear modeling can be applied to loudness perception.

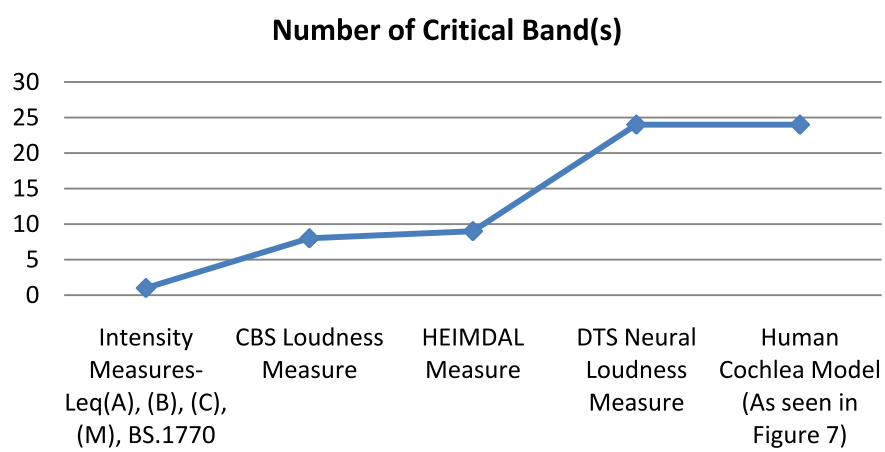

Figure 9 - Comparison of bands within loudness measures

Despite the clear benefits of critical band analysis in loudness measurement, there are very few companies currently able to offer it for commercial applications. The extra processing resources required to accomplish it to date have prevented widespread adoption. DTS Neural Loudness Control is the first real-time perceptual loudness control product available that embraces this advanced approach to loudness measurement.

Traditional Approaches to Real-Time Audio Level Processing

Single-Band Approach to Level Correction

Automatic Gain Control (AGC) is a single-band, energy-based loudness solution that has existed since the beginning of broadcasting. With early products like the Gates “Level Devil” or CBS “Audimax,” volume leveling was as simple as taking an RMS measure (a measure of energy, not a perceptual measure) of the audio signal and then turning the entire signal up or down to reach a desired level as in Figure 10. Devices like these that modify the signal spectrum in its entirety are called wide-band or signal-band approaches to signal processing.

The key advantage of single-band loudness processing is that gain change is the *only* modification made to the original signal. Early versions of these devices offered audio leveling, but they introduced audible side effects like “pumping” and “breathing” as energy ramped up and down and the entire signal was modulated. Eventually, single-band loudness solutions were replaced in many radio and TV applications with multi-band processing.

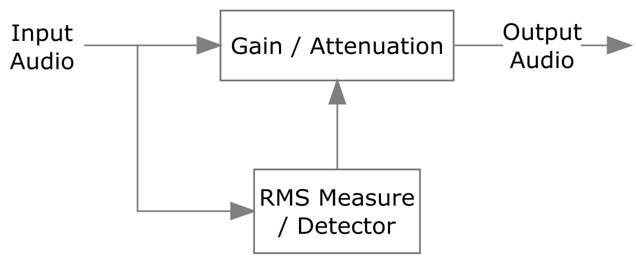

Figure 10 - Single-band block diagram

Multi-Band Approach to Level Correction

Multi-band processing became popular in the 1970s with products like the Dorrough Electronics "Discriminate Audio Processor" and the Orban "Optimod-FM." These devices add complexity to the processing chain by following the slow, wide-band AGC with three to five compressors, each adjusting the energy in different frequency bands (Figure 11). The compressors are followed by fast-limiting on each of the bands before the signal is recombined. Multi-band processing offers the advantage of energy control in each audio signal sub-band without affecting other parts of the signal.

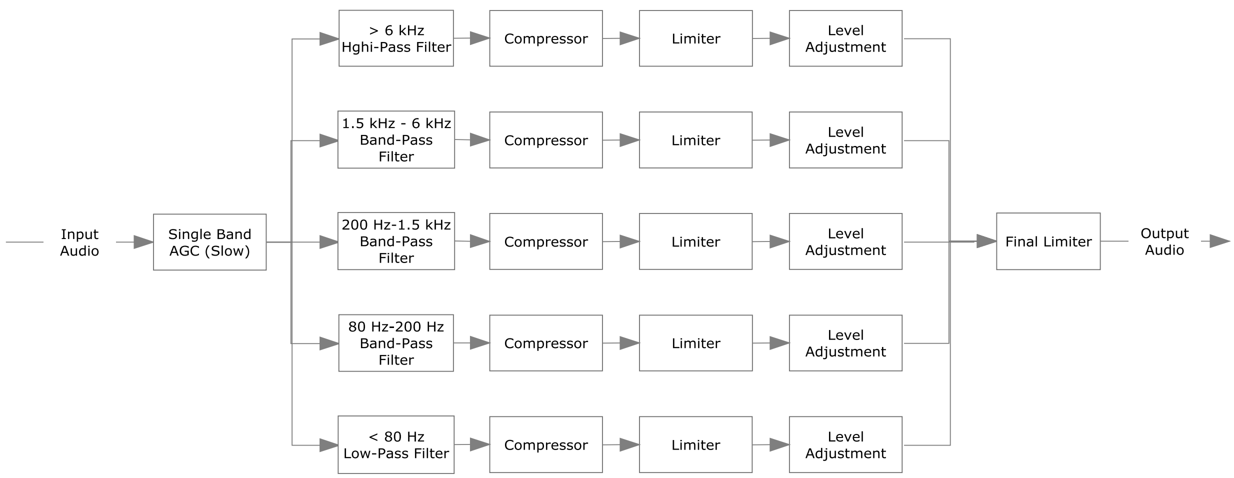

Figure 11 - Multi-band block diagram

If a signal lacks high or low frequency energy, multi-band processing adds gain to these bands while attenuating mid-band frequencies (Figure 12). This enables audio to be presented at a consistent level. Multi-band processing quickly became the signature sound of FM radio worldwide, enabling broadcasters to deliver maximum audio loudness (yes, loudness). Remember that loudness is a function of the amount of energy across multiple critical bands. In an extreme case, if every critical band contains the maximum amount of energy (white noise) presented at the limits of the transmission path, maximum loudness will be achieved.

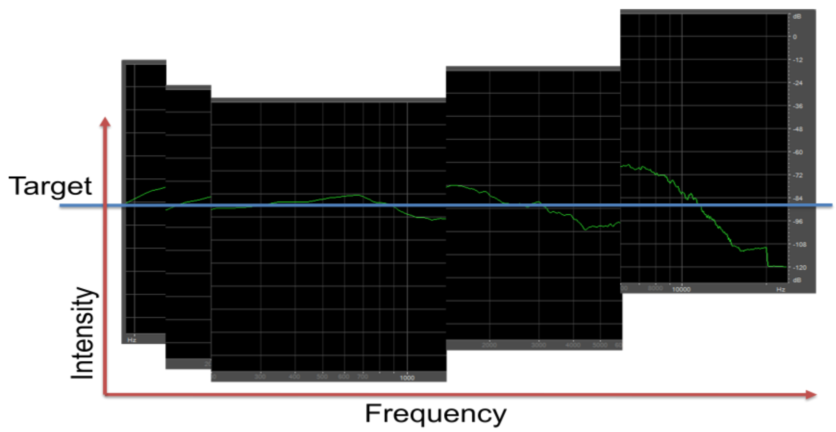

Figure 12 - Multi-band correction

This same approach has been adopted by broadcast TV over the past two decades. Instead of targeting the limits of the transmission path, TV broadcasters target a specified Dial Norm setting. This approach effectively regulates loudness levels to the Dial Norm target.

While multi-band processing does a good job modifying all content to achieve one homogenized sound, the drawback is that artistic intent is heavily modified. In fact, many sound engineers are required to audition their content through a multi-band compressor to reveal what sort of flavoring is added and how it changes the original audio. Dramatic audio that is desirable – like a heavy bass beat, a rumbling car engine, or crashing cymbals – should be over-emphasized in the original content to punch through multi-band processing. As a result, many artistic decisions are overridden. Multi-band processing has been a good tool for achieving competitive loudness for mastering rock albums, but it is a heavy-handed approach.

Comparing Multi-Band Measurement and Correction

It might be easy to confuse the benefits of measuring audio loudness using a multi-band (critical band) based measure with the correction that occurs in a multi-band processor. It is important to differentiate between the measurement process and the correction process.

Critical band based measures asses the audio signal and can provide an accurate single loudness measure that can be used to manually or automatically offset the entire audio signal with gain or attenuation to achieve consistent loudness. This approach does not change the relationship of frequencies within the audio signal. The only difference between the original and processed audio signals is the perceived loudness.

Multi-band correction separately measures each band of the audio signal and changes the amount of energy in that band in relationship to the other bands. By changing the relationship of frequencies within each band, the original and processed audio will have dramatically different artistic qualities. Critical band measurement coupled with single-band correction minimizes the change to artistic content. Multi-band correction modifies the spectral balance of content.

The Objective of Real-Time Loudness Control

When correcting loudness, the ideal solution should create consistency in perceived loudness without modifying any other audio signal characteristic, leaving equalization and spectral density changes to separate processes with different objectives. The solution should only make modifications when content strays beyond the user-defined target loudness level ranges, leaving content unchanged if it is already within range. Any audio adjustments should be transparent and imperceptible to the listener.

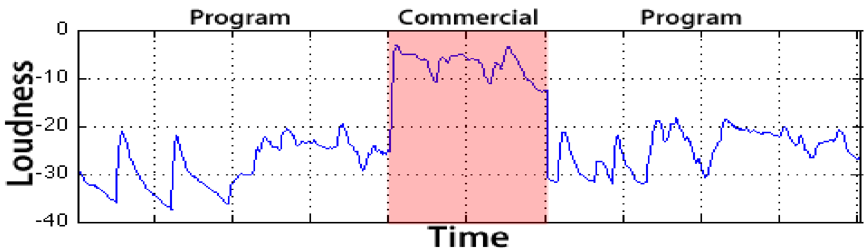

Figure 13 - Original content requiring loudness control

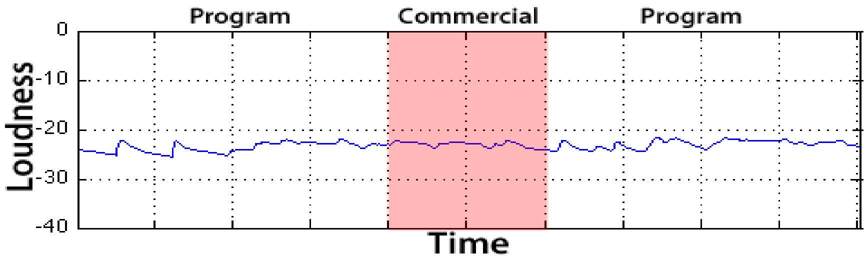

Figure 14 - Overly aggressive loudness control

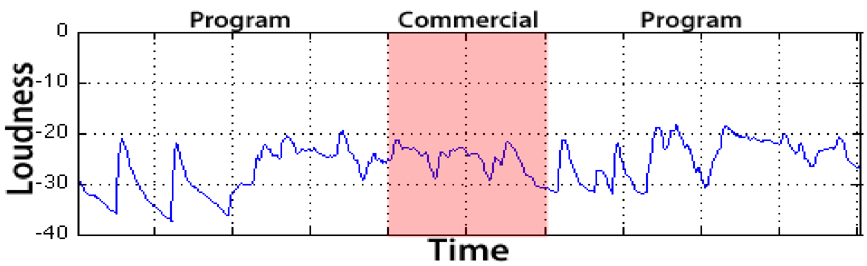

Figure 15 - Proper loudness control

DTS Neural Loudness Control

DTS Neural Loudness Control (NLC) was designed as a real-time perceptual loudness control solution with the above goals in mind. As a single-purpose loudness device, NLC is based on a wideband loudness correction approach that does not modify other aspects of the original audio signal.

The heart of NLC is the Neural Loudness Measure (NLM); a complex critical band based loudness measurement previously applied only in the academic world. NLM is the basis for NLC's transparent performance and it redefines the perceived limitations of real-time signal processing. For comparison, users can toggle between NLM and the ITU-R BS.1770 loudness measure as the basis of correction. Both loudness measures are calibrated to produce the same long-term loudness measures, with NLC favoring more natural short-term correction.

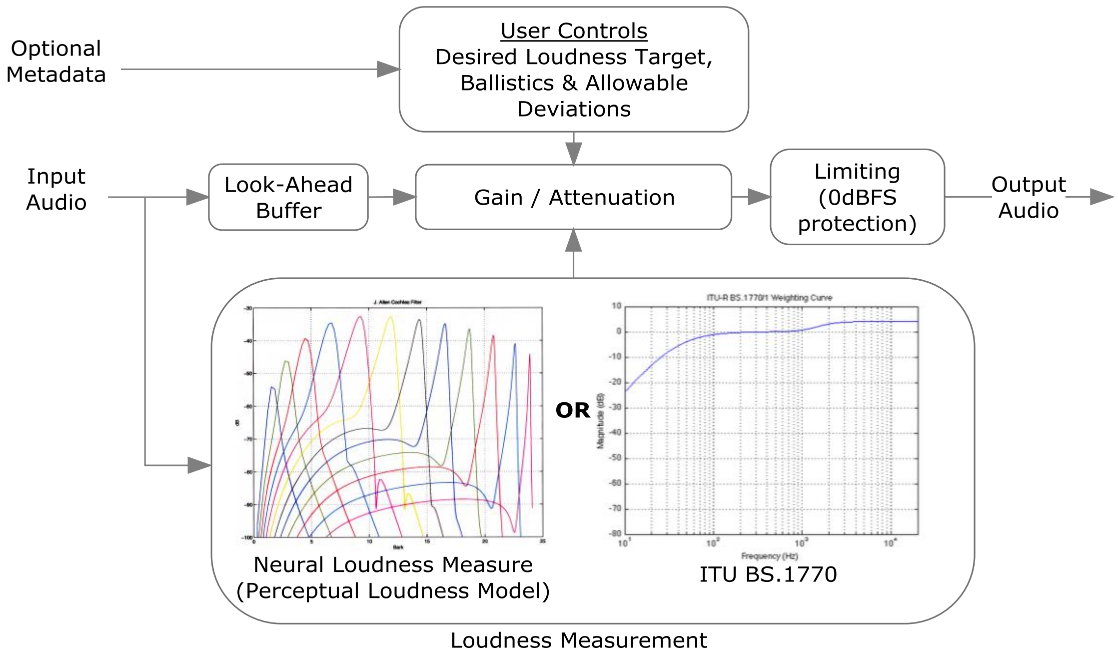

Figure 16 - DTS Neural Loudness Control block diagram

After loudness measurement, NLC applies a user-based rule set to determine the difference between the desired target loudness (or Dial Norm, if audio is paired with metadata) and the perceived measured loudness that is allowed before correction is applied. Additional user controls allow management of correction ballistics and the threshold where low level noise is ignored. These controls even allow NLC to be placed in Protection Only mode, enabling the user to define an acceptable loudness range so only audio that is out of tolerance is adjusted.

NLC uses a look-ahead approach when making signal processing decisions. A delay of 48 ms allows NLC to make adaptive corrections based on audio events that will occur in the near future. Corrections occur before inconsistencies in the audio are heard and any gain changes are nearly imperceptible, due to masking by the corrected elements.

After the user rules and ballistic rules have been applied, NLC summarizes this information into gain or attenuation applied to the signal in a manner similar to an audio mix engineer riding a fader. The combination of Neural Loudness Measure, look-ahead correction and single-band correction create the most transparent approach to loudness control that can be achieved while preserving the artistic decisions in the original content. As a final stage in NLC, a soft-knee limiter subtly removes any content overshoots that exceed the capacity of the digital full scale audio transmission path.

Loudness Tools in the Broadcast Workflow

Broadcasters who employ loudness solutions ride a fine line between a) annoying consumers and motivating legislation with not enough level control, and b) squashing audio dynamics with too much or inappropriate level control, which can alter the artistic intent of the content.

The challenge of allowing dynamics while maintaining consistent loudness is best handled with a hybrid approach that utilizes multiple loudness tools and techniques. Instead of an overcompensating one-size-fits-all product placed at the end of the broadcast audio chain, different types of loudness management components can be employed for different types of content throughout the broadcast workflow.

The top four tools required to manage loudness in a broadcast workflow include:

1. A Listening Ear – No amount of metering can replace the value of active listening. Broadcasters should use trained technicians who work in listening environments that accurately represent the consumer experience as a final line of defense against inappropriate loudness. The listening environment should be set up to apply the accompanying metadata to simulate different consumer listening scenarios.

2. A Loudness Meter – The broadcast industry as a whole has made strides toward standardization with the BS.1770 loudness measure. Operators should use this tool at various places within the broadcast workflow to objectively verify that audio loudness levels are within tolerance. Limited training is required to learn the basics of loudness measurement. Operators can use short-term rolling loudness measures live to confirm that the mix is adhering to broadcaster-established metadata Dial Norm targets.

3. Offline Loudness Analysis and Fixed-Offset Correction – When content is stored before playback, an opportunity is created for non-real-time loudness measurement and correction. Long term loudness of an entire clip, often referred to as an “infinite measure,” can be measured using:

    a. An automated file-based loudness process before server ingest
    b. An operator to manually start and stop an infinite loudness meter around certain content

    To achieve the desired loudness level, this value can be used to either update the Dial Norm metadata or apply a fixed-offset to the actual audio content after the infinite loudness of an entire piece of content has been measured.

    This type of fixed-offset correction is the most transparent loudness adjustment that can be made within a broadcast workflow. It preserves most content dynamics while making sure the infinite loudness measure correctly matches the Dial Norm level. While this may sound like the ideal loudness solution, it alone will not completely address viewer loudness complaints. The following scenario illustrates the problem.

    Imagine a movie that starts with a dramatic scene where the first 10 minutes consists of speech and sound effects measured at -31 LKFS. The second half of the scene includes a loud car chase measured at -17 LKFS. The Dial Norm correctly describes the full scene as -24 LKFS. However, the viewer perceives the audio level as very low for the first half of the clip followed by audio that is much louder during the second half. This extreme dynamic range may be acceptable in a movie theater, but it is unacceptable in the typical consumer's room environment. The infinite average and fixed-offset approach to solving loudness problems has no way of addressing such intra-content issues.

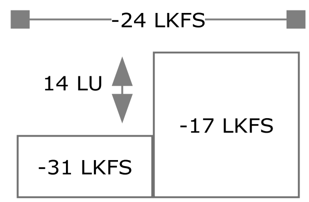

Figure 17 - Short-term loudness changes within content

Some broadcasters might defend the distribution of content as-is, standing behind the infinite measurement as proof they are sending content correctly matched to the Dial Norm. In the best case scenario, if the metadata (including DRC settings) is correct and the consumer device has DRC functionality enabled, dynamics may be limited enough to keep viewers from reaching for the remote control. The more likely scenario, however, is that the component chain is not adjusted correctly and listeners manually adjust the volume level because full audio dynamics are undesirable.

The second challenge that offline analysis and fixed-offset correction cannot solve is how loudness is interpreted in the context of playback. Consider the scenario where a TV program ends quietly followed by a high-energy commercial. Even if both pieces of content are properly prepared and played back at the correct infinite loudness levels, an abruptly louder "intro" will create a jarring effect when played back immediately after a quiet ending as seen in Figure 18.

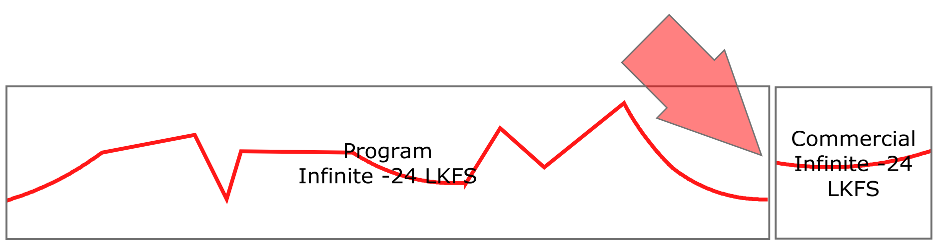

Figure 18 - Loudness changes between content sources

This demonstrates the need for online, real-time signal correction – even *with* the assistance of new tools for loudness measurement and offline analysis.

4. Real-time Loudness Correction – Real-time loudness correction is the process of measuring and correcting audio loudness as live content passes through devices. This is the most traditional approach to volume control. As discussed in the next section, it also has an important role in newer loudness-aware work flows.

Unlike offline fixed-offset loudness correction, real-time correction monitors the audio signal across the short window of time as audio passes through the device. All correction decisions are made based on these short-term (sometimes instantaneous) measures. This might at first seem like a disadvantage. As learned before, however, the human ear perceives loudness in a very similar short-term fashion. Therefore, real-time loudness correction offers loudness regulation that surpasses other management approaches. Because this process can be so effective, broadcasters should be cautious about how aggressively they apply it and allow some dynamics to be preserved within the signal. When signal processing is used instead of a human operator, care should be taken to choose the appropriate device and settings to achieve the most transparent, natural results.

Real-time loudness correction permanently re-authors audio within the stream to achieve broadcast level consistency. This is a good tool for broadcasters who want to do so without depending solely on the DRC system components for audio dynamics compression in the consumer's home. By managing dynamics at their facilities, broadcasters make the final decision about how audio will be heard in the consumer environment.

Figure 19 and Figure 20 show a comparison of off-line versus real-time loudness correction. Notice that while some signal dynamics have been reduced in real-time correction, the transition from program to commercial has been smoothed.

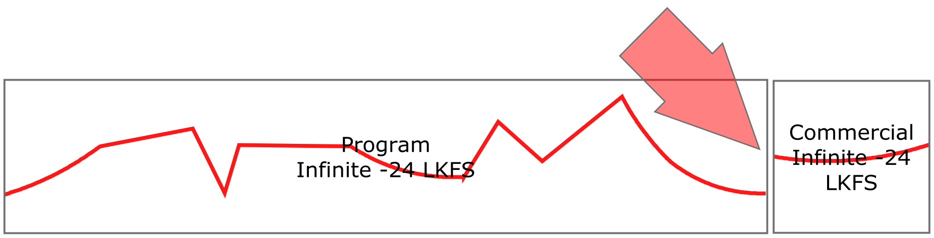

Figure 19 - Content after offline fixed-offset correction

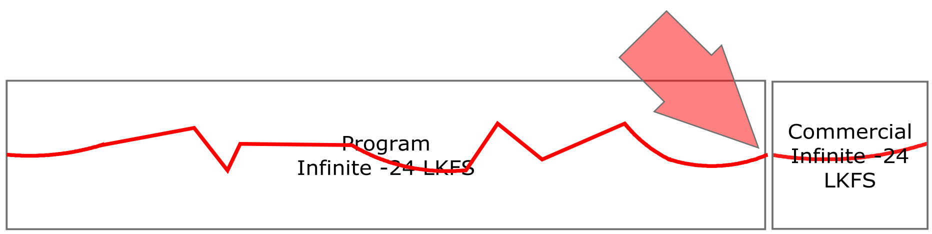

Figure 20 - Content after real-time correction

Real-time loudness correction can be inserted wherever loudness consistency is in question. For example, many newsrooms are turning to automated control for everything from teleprompters to camera movements and the opening and closing of audio faders. In the absence of a human ear to mix and adjust audio levels, wild variances may occur between on-camera talent and the intro / outro music of the newscast. By placing real-time loudness correction in line with the newsroom output, loudness levels can be automatically adjusted to a prescribed loudness target. This can be coordinated to match the metadata paired with the audio signal at the time of broadcast.

Another common application of real-time loudness correction occurs at the final broadcast output, as the last line of defense if the above methods fail to catch out-of-tolerance content. Some real-time loudness correction devices, such as DTS Neural Loudness Control, can be set to Protection Only mode to only modify content outside of the acceptable range. This eliminates rouge content that escapes detection by the other loudness management methods.

It is important to remember the relationship between the audio signal and the metadata. If metadata is re-authored, this should be done with consideration for the loudness, dynamics and channel properties of the audio signal. If the audio signal is modified, the metadata must be updated to reflect the new audio characteristics.

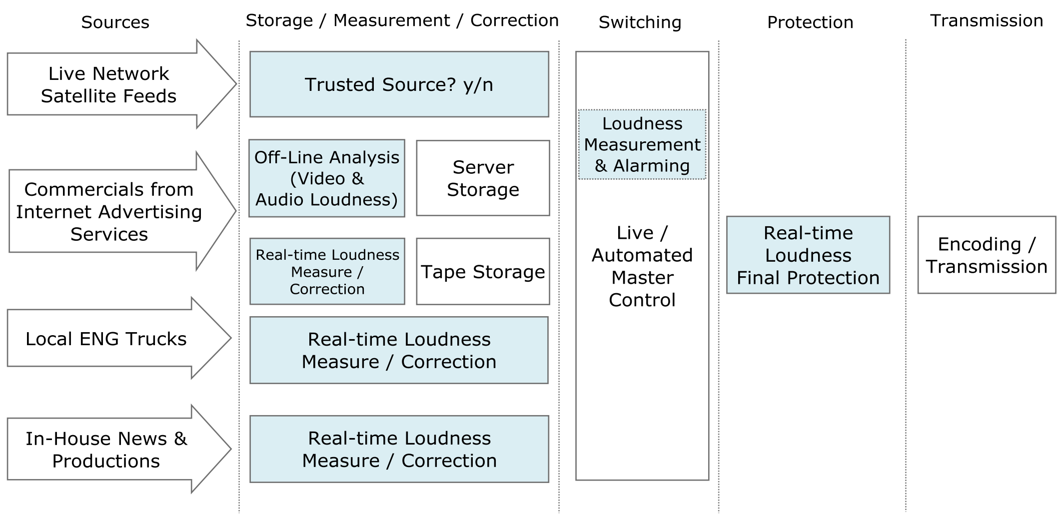

Figure 21 - Loudness measurement and control points (noted in blue)

To select loudness correction devices that are most appropriate for the application design, it is important to know the tools available and where they can best be utilized in the workflow as seen in Figure 21. While audio correction devices have been available for many decades with varying levels of transparency and performance, perceptual loudness control is a relatively new concept and can be substituted into the modern day work-flow where appropriate.

Conclusion

Perceptual loudness measurement and correction is enabling new approaches to audio loudness management that achieve better sound and more consistent loudness levels in broadcasts through an approach that mimics the way the human ear detects and responds to loudness. As a result, broadcasters can now select better tools as they architect new facilities and workflows.

Solving all loudness problems is not as easy as placing a “fix-all” box at the end of a broadcast chain. Automated correction is an important part of the required toolset, but it should be used sparingly. The goal of the correction system should be to transparently adjust loudness without changing any other signal aspect, thereby preserving any artistic decisions reflected in the original audio.

Audio loudness should be measured and managed across the entire workflow along with the metadata that impacts how audio is ultimately rendered. Each facility should monitor, measure, and correct audio loudness problems using a combination of real-time and off-line tools. While the BS.1770 measurement standard provides an objective numerical value for perceived loudness, it is a reasonable approximation only over the long-term. This makes it a useful tool for operator assisted correction or off-line analysis then correction. Optimal real-time loudness correction solutions employ a multi-band loudness measurement followed by a single-band correction.

Broadcasters can now achieve excellent loudness consistency with virtually imperceptible content changes through Perceptual Loudness Management. For applications where real-time correction is required and a human operator is not present, an advanced critical band loudness model enables DTS Neural Loudness Control to manage loudness with natural sounding and consistent results.

Works Cited

Allen, J. (2010, May 20). Nonlinear Cochlear Signal Processing. San Diego, CA, USA.
ATSC. (2009, November). *ATSC Recommended Practice: Techniques for Establishing and Maintaining Audio Loudness for Digital Television.* Retrieved 5 2010, from ATSC: http://www.atsc.org/cms/standards/a_85-2009.pdf
Fletcher, H., & Munson, W. A. (1933). Loudness, its definition, measurement and calculation. *Journal Acoustic Society* , 82-108.
Johnston, J. (2006). *Loudness Tutorial.* www.aes.org/sections/pnw/ppt.htm.
Jones, B. L., & Torick, E. L. (1982). A New Loudness Indicator for Use in Broadcasting. *AES* , 1878.
Nielsen, S. H., & Skovenborg, E. (2004). Evaluation of Different Loudness Models with Music and Speech Material. *AES* , 6234.
Stevens, S. S. (1936). A scale for the measurement of the psychological magnitude: loudness. *Psychological Review* , 405-416 .