Delivering Effective Loudness Control
By Guy Marquis — Infrastructure Product Manager, Grass Valley, a Belden Brand

November 2011
Loudness control is probably the source of more complaints for broadcasters than any other technical issue. Viewers have grown weary of continuously adjusting the television volume, especially between programs and commercials.

This lack of audio consistency between segments has arisen due to the technical difficulties inherent with stabilizing loudness across multiple sources of programming, along with a desire to make commercials more noticeable to viewers. For a long time, the audio for commercials has been compressed to generate a more even but louder soundtrack.

However, this landscape is changing quickly, and there is new impetus behind measures to control loudness. This is being driven by multiple factors, including upcoming legislation around the world, like the CALM Act in the US, and new audio recommended practices, such as ATSC A/85. There’s also a desire to offer viewers a better “quality of experience” and, perhaps most importantly, new loudness control technology that’s creating more options for broadcasters.

Traditional loudness control during production

There are now multiple different approaches to loudness control, including new recommendations for content origination, as well as new devices providing a loudness control safety net during playout. Let’s consider the content origination option first.

Dolby has long been active in loudness measurement and control through the use of Dialog Normalization (DialNorm). During content production, audio engineers can set the DialNorm value and profile metadata using their Dolby E encoders. This DialNorm value can be preserved in the VANC all the way through the playout process, and ultimately to the viewer’s set-top box. Whenever the set-top box detects a change in the DialNorm value, a different attenuation is applied to bring the loudness back to the desired range, that is an average dialog level at -31dBFS.

This is an elegant solution which preserves audio characteristics, while minimizing both channel-to-channel and intra-channel loudness differences. However, many broadcasters have found it difficult to preserve the DialNorm metadata throughout their playout chains, which typically comprise a large number of devices from multiple vendors. Another issue is that broadcasters do not typically have full production control over all the content they broadcast. A typical facility will use feeds from other broadcasters, live news feeds, and even web content. Live news feeds are especially problematic because the DialNorm value is usually set after the program is produced, because it represents the average dialog loudness of the entire program. However, this process is just not conducive to fast-turnaround news environments. Naturally, all these different factors can undermine this loudness control process.

To help address these problems associated with broadcasting content from multiple third party sources, the ATSC A/85 Recommended Practice proposes using a standardized, dialog average loudness level of -24 dBFS for content creation. In essence, the aim here is to offer more control to broadcasters by limiting the variation in the loudness level from different content producers. However, this proposal has not been widely adopted at this point.

New safety net loudness control in playout

In view of the difficulties associated with using third party content, it makes sense to reinforce any loudness control processes used during content production with a “safety net” that can be activated during playout. This can be provided by a new generation of loudness control processors, such as Automatic Loudness Control (ALC) from Grass Valley. In essence, this represents an advanced automatic gain control processor, with a range of processing options to best suit individual broadcasters. Choices include “set-and-forget” type loudness processing, and active management of the processing according to the content type. Other options include wideband or multiband processing, which each offer different strengths to different types of facilities.

When a loudness control processor is used in a ‘set-and-forget’ mode, a target average loudness is set per channel program, and this will be maintained by the processor during commercial breaks, as well as when sources from different origins are played out.

A more sophisticated loudness control method involves adapting the loudness processing according the type of content. Different profiles can be used, which offer optimized processing for music, speech and other forms of content. With this method, a facility’s traffic team will tag the content according to its type so the playout automation can select different profiles for the loudness processor, using GPI or serial commands. Another variant of this method involves using the automation to trigger a by-pass of the processing for certain types of content. For instance, some engineers may like to only use the loudness processing for content that has been generated by third parties in order to preserve original dynamics as much as possible for their own originated and reliable material.

A good loudness controller must be able to react swiftly to fast loudness changes, such as going to a commercial, as well as over the longer term to manage an entire program that might be too low or too high in relation to the DialNorm value in use. So, a good solution will always be composed of a long term automatic gain control and a short term compressor/expander/limiter. For equipment vendors, the challenge has been to come up with good controlling algorithms and, more importantly, the right parameter combination. Unless these key criteria are addressed, the dynamic range of programmes can be adversely affected by loudness control, and artifacts like “pumping” can become apparent to viewers.
There are two core approaches to high performance, real-time loudness control: wideband and multiband processing. Both methods measure the audio, and adjust the gain in real-time. Wideband algorithms continuously control a full audio bandwidth gain stage in real-time.

In contrast, multiband algorithms apply their gain compensation on different frequency bands independently, reflecting the fact that the perception of loudness by the human brain varies with the different frequencies of the audio spectrum.
Multiband and wideband processing techniques both have fierce advocates, and each has different strengths. In general, the multiband approach has gained more traction in the Americas, whereas wideband processing has proved very popular in Europe.

The multiband approach provides more flexibility, and possibly offers the best results with vocal tracks. However, it can be more complex to configure, and it's generally more expensive. In contrast, wideband processing offers strong performance, with simpler configuration and adjustment, along with lower costs. The bottom line is that both of these approaches can usually be used effectively to address the same loudness problems across multiple applications, such as broadcast playout, server ingest, production and post-production. The best solution will depend on an individual broadcasters’ particular channel mix and program dynamics, as well as the associated playout workflow and equipment.

In view of the fact that there is no single best-fit solution for all types of broadcasters, Grass Valley has chosen to provide a range of loudness control solutions as part of its Automatic Loudness Control (ALC) option for its interfacing modules. It offers wideband processing based on its new advanced audio processing algorithms, as well as proven multiband and wideband solutions through partnerships with audio industry leaders. These include Junger Audio’s Level Magic wideband processing, and Linear Acoustic’s Aeromax multiband processing. These three different Automatic Loudness Control solutions can provide up to eight channels of loudness processing per program and can process up to four programs simultaneously. Operation can be based on either ‘set-and-forget’ or actively managed processing modes, with playout automation driving a bypass mode or different profiles.

This loudness processing technology is typically operated in master control, just ahead of the transmission encoder, to provide an effective safety net. However, it can also be used effectively to normalize audio levels at a station’s ingest point, so that all signals within the plant are considered controlled, and thus meet a specific target loudness. Naturally, loudness control is equally applicable to cable, IPTV and satellite headend environments.
Conclusion

It’s now fully evident that new audio processing technology offers a level of sophistication that allows broadcasters to overcome traditional loudness control issues. As with many new broadcast technologies, it does require careful selection, commissioning and management to ensure optimum results. However, this is well within the grasp of station engineers, and it’s highly likely that automatic loudness control will become as prevalent as other core broadcast technologies. For the long suffering television viewer, it offers the chance to put down the remote and relax at long last. It’s what television should be all about!