Qualis Audio
We're Listening
Qualis Audio, Inc. 15110 SW Boones Ferry Rd., Suite 350, Lake Oswego, Oregon 97035
503-635-9376 Fax: 503-635-3851 www.qualisaudio.com

July 5, 2011

Revised Response to Notice of Proposed Rulemaking

In the Matter of

Implementation of the Commercial Advertisement
Loudness Mitigation (CALM) Act

The Commission is to be commended on the thoroughness of the Notice of Proposed Rulemaking for implementation of the CALM Act. We have had numerous conversations with people across a wide range of the television broadcast industry since the CALM Act was introduced in Congress. All of the questions and concerns we have encountered during this period are cited in the NPRM.

We have comments relevant to many of the technical and practical aspects of the NPRM. Our submission begins with comments as they relate to each paragraph or footnote, in the same order as the material they reference. This is followed by a technical discussion, which presents the material in more detail and in a manner which highlights its relevance to the issues raised. We then consider the practical application of technology in the larger television broadcast ecosystem in the operational issues section. We propose a compliance approach not specifically described in the NPRM for consideration by the Commission. We then summarize our perspective. An appendix describes this authors technical and industry qualifications and describes our firm's position in the broadcast industry.

We sincerely hope these comments, discussion and recommendations help create rules that all stakeholders; the Commission, the broadcast industry and consumers, will appreciate. As the rulemaking process continues we welcome the opportunity to provide further input.

Richard C. Cabot, Ph.D.
Chief Technology Officer
richc@qualisaudio.com

Comments, in sequence order

Paragraph 4 describes the original intent of the AC-3 dialnorm system. However, as we describe in the technical discussion, the basic premise that maintaining a constant dialog loudness will result in consumers perceiving a consistent loudness is flawed. This premise is true for well mixed dramatic productions but does not hold when intentionally manipulated commercial advertisements are considered.

Footnote 21 cites ITU work in process for revising ITU-R BS.1770 and references a January 2010 press release. The ITU approved a revision to BS1770 in late 2010 which was published in March of 2011. It includes changes specifically developed to deal with deficiencies in the original dialnorm concept. Although the research behind these changes is not documented in the revision, it is well documented by the EBU PLOUD working group which developed them. This research, and the need for it, is presented in the technical discussion.

Paragraph 5 raises concerns of industry representatives about broadcast systems which do not use the AC-3 audio system. As we explain in the technical discussion, for a fixed value of dialnorm (and with dynamic range control disabled in the consumers decoder) the AC-3 system becomes a fixed gain multichannel transport system. Proper loudness management does not require the dialnorm mechanism. It merely requires measurement of loudness using a BS1770 compliant meter and subsequent adjustment of overall program level to present content with a uniform loudness into the transport system. Transport systems other than AC-3 can be employed with no reduction in ability to provide consistent loudness to the consumer.

Paragraph 10 raises concerns by stations/MVPDs as to their responsibility for commercials they did not insert into the broadcast stream. We agree with Commissions interpretation that the statute does not allow disparate responsibility based on the source of the commercials. However, we believe that implementation of reasonable *systems and processes* by stations/MVPDs will result in loudness consistency which meets the intent of the Act. The systems involve specifying standards for acceptable content, monitoring the content at ingest (or at least as broadcast) and creating and consistently applying feedback processes which notify content providers of discrepancies in received content. This is described more fully in the discussion and in our proposal.

Paragraph 11 discusses applicability to broadcast systems which do not use the AC-3 audio system. As mentioned earlier and discussed below, proper loudness management does not require the dialnorm mechanism. Other transport systems can be employed with no reduction in ability to provide consistent loudness to the home viewer.

Paragraph 15 relates to practical compliance options for stations/MVPDs. We agree with the idea that regulated entities should have flexibility in implementing solutions o the CALM Act requirements. We hope that the ATSC A/85 RP will be viewed as specifying underlying technology and providing implementation guidelines but not as a limitation on specific implementation architectures. This distinction is critical as there are a wide variety of systems and workflows currently in use and a small number of workflows or solutions described by the ATSC may not fit the practical requirements of individual regulated entities. We discuss this point at length in our operational issues section and our proposal.

Paragraph 16 questions the breadth of the safe-harbor provisions cited in the Act. Although we lack the legal background to authoritatively comment, we believe that the Commissions interpretation of the station/MVPDs responsibility is essentially correct. (We note that a third party could provide monitoring of a broadcast signal on a contract basis with a third party but the responsibility for acting on this information would remain with the station/MVPD.)

There is a practical aspect to the network/affiliate relationship that is not mentioned. All affiliates insert locally contracted or implemented advertising into their broadcast stream. Although they may receive conformant material from the network, local insertion will create a requirement that they be capable of maintaining conformance of the ultimate broadcast stream.

There is a practical distinction between affiliates and MVPDs for channels on which they perform no local insertion. However, once the MVPD inserts commercials into a broadcast stream, or modifies a broadcast stream, they assume the responsibility for insuring the conformance of the material.

Paragraph 17 addresses application of the safe harbor rule by MVPDs for locally inserted commercials vs. those embedded in content. Practical considerations argue against arrangements where applicability of the safe harbor rule change dynamically with the content source. In our proposal we approach the issue from a general perspective and avoid such problems.

Paragraph 18 requests interpretation of the phrase “commercially reasonable”. We note that the statute uses this phrase in reference to the *application* of the equipment, <u>not</u> the equipment itself. In this context we believe it requires that the equipment be installed, used and maintained in a manner compatible with designer or manufacturer intent. The alternative, which we believe this phrase is designed to prevent, is an installation, connection or operation of the equipment which might give results more favorable to the user but would be inconsistent with equipments intended use. We do not see any relationship to individual circumstances with regard to the regulated entities. However, there may be differences in the meaning of commercially reasonable that occur due to design differences between equipment. .

Paragraph 19 describes an overly restrictive definition. We take installation to mean that the regulated entity has purchased, rented or contracted for equipment and/or software which meets the ITU-R BS1770 standard and completed the steps required to make it functional. We specifically take issue with the statement that equipment must "support the use of dialnorm metadata". Implementations which use a fixed dialnorm value, which the ATSC specifies as -24, would not need to handle metadata in any way. In our opinion, the simplicity and reduced chance for error offered by fixed dialnorm approaches will make them dominant in practice.

Paragraph 20 presents an unusual interpretation. We define utilization consistent with normal English usage, that the installed (in accordance with comments on paragraph 19) equipment is actually being used. This would preclude a regulated entity from installing equipment and ignoring the results it produces or not actually routing audio through it.

Paragraph 21 specifies an overly restrictive interpretation of maintenance. We believe the intent is to insure that equipment is always capable of performing its intended purpose. The reality of modern digital audio equipment is that if it functions it will return correct results. The measurement algorithm is coded in software and it will not change without explicit action on the user's part. Examples of practical failure situations are when audio is no longer routed to the equipment, the alarming/reporting mechanism is broken, personnel disable alarm messages or no longer review logs, etc. We believe that the Commissions proposed interpretation would create an undue, and unnecessary, hardship on regulated entities.

Paragraph 22 introduces the concept of demonstration of compliance. The NPRM presents this as an alternative to the safe harbor provision. For reasons explained in the operational issues section and our proposal, we believe it will be a part of any practical solution. It is a mammoth undertaking to handle the number of disparate programs required to fill a broadcast channel with content 24/7/365 and the number of commercials thus required to finance such an enterprise. Maintaining consistent loudness throughout this process is unlikely without monitoring the emitted signal and using the results to detect and correct errors. Once monitoring is in place, it is available to demonstrate compliance. A common concern among broadcasters is that their own documentation failures will be used against them and result in penalties. Our belief is that a documented corrective response to the occasional failure should protect the entity from penalty.

Paragraph 24 discusses implications of contractual arrangements to shift liability for failures "upstream". This is a logical and practical step for any entity in the broadcast chain from content creation to delivery. We believe such arrangements are beneficial not only in relieving MVPDs of financial liability for errors largely outside their control but also in sensitizing content providers to their responsibility for providing properly created content. However, we believe the optimal situation is one in which the delivery entity can identify non-conformant content and relay that finding to the provider even in the absence of a complaint. Such action on an ongoing basis will reduce errors and decrease the likelihood that consumers will complaint to begin with. If an entity consistently demonstrates attention to such quality improvement its actions should reduce or preferably eliminate its liability for loudness errors in delivery.

There is nothing preventing advertisers from delivering conformant content to broadcasters. Large, more technically competent post-production houses are already handling requirements that they deliver advertisements mixed to specific loudness targets. Smaller post-production houses or one-stop-shop operations which produce much lower budget commercials are largely unaware of these requirements and will take time to become familiar with the requirements. However, many low cost loudness measurement solutions are already available for PC based mixing platforms. The limiting factors currently are knowledge and experience of the mix engineers at such firms. Industry demand will correct this situation in due course.

Paragraph 25 Although MVPDs may find contractual approaches adequate to eliminate their liability this is unlikely to be a complete solution for small stations because of the need for local insertion. Again, we believe a monitoring system with consistent attention to error identification and recurrence prevention is the best solution. The cost of a basic version of such a system is already within reach of small stations and market pressures are likely to drive this cost lower over the time frame specified by the NPRM.

Paragraph 27 The differences between AC-3 and other systems in use for delivering digital audio should not create any issues for implementation of proper loudness management. For fixed dialnorm implementations the AC-3 system becomes a simple multichannel audio path from MVPD to the consumer. The loudness meters produced by our firm, as well as those produced by our competitors, will measure loudness on PCM streams completely outside an AC-3 ecosystem. Most (if not all) systems intended for consumer use convert surround sound format digital outputs to AC-3 as they leave the set-top box. It is merely necessary to know the loudness in their broadcast chain which corresponds to a -24 LKFS loudness delivered to the consumer. When a 2 channel signal is delivered in PCM (as might occur in a set-top-box configured for 2 channel output) it is a simple matter to establish the appropriate gain scaling factor.

Paragraph 28 It does seem reasonable to grant an exemption to MVPDs when they retransmit channels to consumers on a real-time basis and do not modify the signal in any way.

Paragraph 30 In the case of cable providers carrying local OTA signals it seems counter-intuitive to hold the cable provider responsible for errors which would otherwise be the local stations responsibility had the consumer merely connected an antenna. Indeed, most consumers who encounter a loudness problem in such a situation would likely complain to the Commission about the local station, not the cable provider. Is it then reasonable for the Commission to shift the complaint to the cable company once they become aware that the complainant is receiving the signal via cable?

Paragraph 36 The process described for tracking complaints and deciding, based on trends, whether to initiate enforcement action is exactly the approach we advocate. We believe it is helpful to forward copies of all complaints to the regulated entity with a notation that they are merely for information. These can serve as independent feedback into the entities quality improvement system and should match data it derives internally. Should the entities quality control system fail to detect a problem this additional feedback mechanism warns them that a discrepancy exists. If they determine that the issue is a shortcoming in their systems it allows them to take appropriate corrective steps.

Paragraph 37 We believe that rules and forfeiture provisions should encourage regulated entities to implement loudness management approaches that maximize delivered audio quality. We believe a focus on delivered quality can be encouraged by eliminating forfeiture liability for good faith mistakes and by phasing-in sanctions over a year or two period to allow for inevitable learning curve issues which will arise. The interaction of loudness management and audio quality is discussed in more detail in the operational issues section.

Technical discussion

Dialog normalization

The dialnorm mechanism in AC-3 was developed based on experiments which showed that viewers will set playback volume to make dialog a realistic level. This result is reasonably consistent across different programs and viewers. The programs used were mixed by professional mixers whose intent was to create a realistic, high quality result. If all content creators behaved this way, things would be much simpler.

Unfortunately the motivations of engineers mixing commercials are often at odds with this approach. Their goal is frequently to get their audio noticed, whether or not the consumer wants to pay attention. As such, the dynamic range may be greatly reduced, non-dialog sounds are added and spectral balance may be changed, all in an effort to garner attention. Since they can check what value a meter assigns for the contents loudness they may adjust these variations to maximize the noticeability while minimizing the measured loudness. When loudness measurement is used as a dialnorm setting the result is an audibly louder commercial. The ITU BS.1770-2 standard changes (discussed below) were developed to reduce the ability of mix engineers to finesse the content in this way.

Additional complications with normalizing dialog occur when there is infrequent or no dialog in content as would occur when broadcasting a concert. The solution to this dilemma given in A/85 is to measure the “anchor element” in the program. Unfortunately, without a human being in the loop this is virtually impossible to do.

Informal experience with a wide range of broadcast content shows many other difficult cases, including content with large amounts of engine noise and game shows with loud music or highly repetitive gongs which punctuate events in the program. Dialog loudness does not accurately represent consumers loudness impressions of this content.

Consumers judge a program in the home as excessively loud based on a relatively short listening period. Grandma walking into a room where the television is on will generally yell "turn it down" within a few seconds if she finds the loudness oppressive. She won't wait for several minutes and certainly not for the end of the program. As such, the goal becomes to assess short term maximum loudness. It is the responsibility of content creators to set the dynamic range according to their artistic desires.

Consequently it is useful to think of dialnorm as a “loudness norm” parameter as that is the function it serves in practical broadcast use.

Loudness measurement

There is a fundamental problem is trying to assign a single number to the loudness of a program. This stems from the desire to reduce loudness differences between programs without using the dynamic range compression common in earlier times.

Listeners assessment of loudness is formed in a relatively short time, opinions of this time range from 3 - 5 seconds. When loudness is rated over longer periods it requires the listener to perform an unnatural judgment. They don't perceive the loudness as constant so assigning a single number requires a cognitive process that will vary with the individual, the program dynamics, and many other factors. A mathematician would describe it as trying to assign a single scalar value to a multidimensional, time-varying data set. The mapping function will be highly nonlinear and not receptive to either interpolation or extrapolation in any dimension. The result will be a compromise, and the assessment of what mapping is "best" will vary greatly with how each person plans to use the output and what constraints they expect to be placed on the inputs.

The experiments described in ATSC A/85 focused on assessing the loudness of modest length program segments. As such, their results take into account some of this cognitive processing. However, they do not scale in a predictable way to longer programs. Longer programs greatly increase the degrees of freedom in the underlying multidimensional data set. Consider the program loudness vs. time as an ordered sequence of short term loudness measures which (we will assume for now) can be reliably assessed. The number of permutations of this sequence increases geometrically with the program length. The result for even modest length programs is an untestably high number of permutations.

The 2011 revision of BS.1770 was based on experiments performed under the auspices of the EBU. These focused on properly assessing the loudness of audio segments substantially longer than the previous experiments. Because of the high number of possible permutations mentioned above, the EBU experiments were necessarily limited. Extrapolating from these results to other situations is difficult.

Narrow dynamic range programs, regardless of length, are more easily quantified by any given algorithm because the rating for a subsection of a long program is likely to be consistent with that of another subsection. Wide dynamic range programs are not quantified well as their length increases because the subsections aren't likely to be consistent.

Things are complicated further by the very real prospect that advertisers will "game the system" and manipulate the program dynamics, even to the extent of inserting background noises, to make their material "louder". The noble desires of the ATSC committee members and most broadcast industry audio engineers to improve audio quality and improve the listener experience are not shared by everyone who will be creating material to be broadcast. As with writing laws or rules in any other aspect of society we must be wary of unintended consequences.

The new ITU BS.1770-2 loudness measurement does not represent the loudness of a program or commercial. This assertion is not meant to denigrate the ITU standard but rather reflects the reality that the loudness of a typical audio segment lasting more than a few seconds cannot be characterized by a single number.

The BS.1770-2 measurement represents “the loudness of the loud parts” of a program. This concept results from the recognition that consumers are sensitive to being blasted out of their sofas. Basing loudness assessment on this makes disparate audio content coexist well. Its intended purpose isn't to be the most accurate loudness measurement possible but to be the best compromise dialnorm setting for broadcast content.

<u>Dialnorm and metadata management</u>

The A/85 RP describes two usage models for dialnorm. One employs a variable dialnorm value which must track the measured loudness of content. The other employs a fixed dialnorm value and all content is adjusted so that its measured loudness matches the fixed dialnorm value. We believe that the fixed dialnorm approach will ultimately be used throughout the industry as practical issues arise in using the first method.

The fixed dialnorm approach requires all content to be mixed to a target loudness equal to the fixed dialnorm value. If the target is not achieved the content must be fixed by adjusting the overall level by the difference between the measured loudness and the target value. Content providers must deliver conformant material or the user must adjust it upon receipt. Consistency of incoming content must be checked at least on a sample basis to insure compliance.

The variable dialnorm approach allows content to be mixed to whatever loudness the content creator desires. Since this loudness must be known the practical result is that all content must be measured. The stated loudness must be checked, at least on a sample basis to insure accuracy. The resulting loudness value must travel with the content throughout the broadcast chain and be used to set the dialnorm value during the transmission of that content.

The two methods require similar work when content is created and at ingest. The creator must measure loudness and the recipient will likely verify it at ingest. However, the variable dialnorm method requirement that the metadata follow the content throughout the transmission chain all the way to emission is onerous. Facilities which route audio throughout their plant using Dolby E have a ready place to store dialnorm metadata. Facilities which route audio as multichannel PCM, whether via AES, MADI, SDI or IP based systems have no such way to marry metadata to the audio. However, even in facilities maintain the metadata when routing it is difficult to insure that every piece of equipment in the chain, particularly during editing, will maintain its integrity. Operations which do more than simple “razor blade” edits can affect the program loudness and destroy the relationship between content and metadata. However, edits in a fixed metdata implementation may require that overall content level be adjusted to insure that average loudness still meets target.

The metadata integrity issue becomes a immense problem in practice since any metadata disruption will result in loudness error. Problems will also occur if the metadata is not transferred properly to the AC-3 encoder. The dialnorm field must contain something. If it isn’t set by the associated metadata value it has a 1 in 30 chance of being correct. As regulatory forfeiture is a possible outcome of such errors most organizations will be unwilling to risk using the variable metadata scheme.

The likely outcome of this situation is that broadcasters will view variable dialnorm as impractical and dangerous to implement. They will operate with the recommended fixed dialnorm of -24 and normalize all content to a loudness of -24. When live material is broadcast the mix engineer will target a -24 value for loudness. Using this approach there is no requirement to pass metadata from the file to the encoder. Consequently there is no risk of this path being broken.

Most organizations with which we have discussed this issue state that they will not use the variable metadata implementation. The ones which have been most adamant in this position are the ones that have previously used variable metadata.

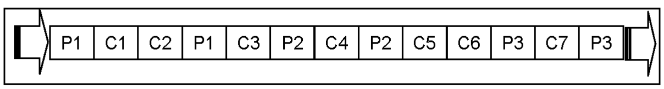
It is useful to keep this in mind when discussing systems and methods likely to be implemented in the field. When considering regulated entities that do not use the AC-3 system it is best to compare their likely operation with that of an AC-3 based implementation using fixed metadata. As mentioned previously, when fixed dialnorm is employed the AC-3 system becomes a simple multichannel delivery mechanism.

Real Time Monitoring and Logging

The audio portion of a typical television broadcast consists of sequential programs which are separated and/or interrupted by commercial advertisements. The ITU standard specifies, and commercial equipment implement, the ability to start, stop, pause and reset loudness measurements. This allows the user to begin measurement of a selected program, pause it during measurements of commercial breaks, resume measurement when the program resumes, stop the measurement when the program concludes and reset the meter to allow another measurement. This control can be effected manually or through hardware connections to the measurement equipment. This suits measurement of pre-recorded and file based content but has limitations when applied to real-time measurement.

Real-time monitoring and logging requires independent assessment of program material and of the commercials which separate or interrupt it. Performing independent assessment of both the program and the commercials requires multiple measurement meters or convoluted and error prone manipulation of the data provided by a single meter.

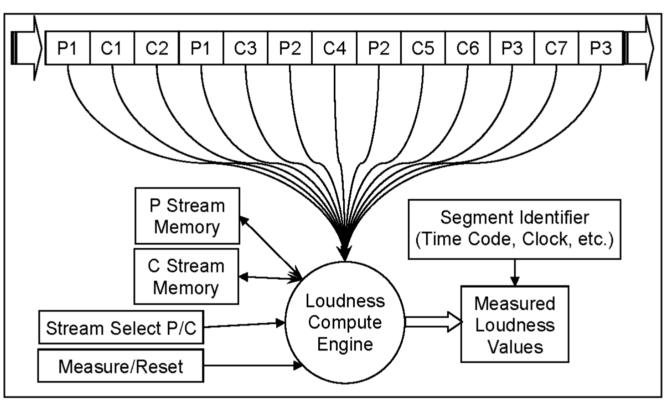
The problem posed by a typical broadcast audio stream is illustrated below. The signal consists of multiple programs broadcast sequentially, identified in the figure as P1, P2, etc. These programs are interrupted by commercials, identified in the figure as C1, C2, etc.



A typical broadcast audio stream

The interruptions may consist of a single commercial or several sequential commercials as illustrated in the figure. When one program concludes it is typically followed by one or more commercials, identified in the figure by the letter C. When the inter-program commercial break concludes the next program is transmitted. It is interrupted and followed by commercials in the same manner as the preceding program.

The figure below illustrates our solution to this problem. The measurement is controlled by two signals; a stream select and a measure/reset. These can be logic level inputs from a playout server, dedicated buttons or soft buttons on the instruments GUI. Similar control commands received over the instruments Ethernet port are possible.



The Qualis Audio dual loudness measurement architecture

The stream select signal causes the loudness measurement engine to measure the program stream (Pn) when in one state, for example when high, and the commercial stream (Cn) when in the other state, in this example when low. The engine always measures one stream or the other. However, there is additional functionality built into the way the measurement engine uses this control signal. When the signal transitions from the P state to the C state the program stream measurement is paused. The program stream measurement memory is not cleared or affected in any way. When the signal transitions from the C state to the P state the current loudness measurement is saved and logged as a final result and written to the results memory. The commercial stream memory is then cleared to prepare for another measurement and the loudness measurement engine now continues measurement of the ongoing program stream.

The measure/reset input causes the loudness measurement engine to save and log the current measurement value as a final result and clear the stream memory to begin a new measurement. This is done regardless of which stream is being measured. If the engine is measuring the program stream when reset is asserted the current reading is written to the result memory and the program stream memory is cleared. If it is measuring the commercial stream when reset is asserted the current reading is written to the result memory and the commercial stream memory is cleared.

The loudness measurement engine also keeps a running Short Term measurement which is not affected by the control lines. This continuous measurement tracks listener perception of the stream in real time and provides assessment of trends in the loudness of the current stream.

The complete data stored in the log files allows customized report generation for quality control reporting, monitoring, query response or general management use. A time period or section of interest may be selected via GUI cursors and saved. This data may be loaded in Excel, analyzed and displayed in whatever combination desired.

Alternately, the entire process may be run from a batch file eliminating the need for any human intervention. This approach may be used, for example, by broadcasters who want an automatically generated report each morning of the previous day’s broadcasts. The script tool may load as-run logs, creating a report that lists each program and commercial along with their loudness or other desired parameters.

Operational Issues

Interaction of loudness management with audio quality

For many years analog television managed its loudness with automatic gain control devices. These sense the deviation of audio level from a desired target and automatically increase or decrease their gain to bring the signal with the desired range. These devices seriously altered the program dynamics and often introduced nonlinear distortion. The limited performance of the analog television audio system made the limitations of such devices less noticeable than they otherwise would have been.

The last several years have witnessed the development of similar devices targeted at digital television. They sense program amplitude using techniques similar to that employed in BS.1770. However, they can't be equivalent as they do not have the entire program available in advance to perform a measurement. As such, they must reduce program dynamics to effect a result within the target range. If the dynamic range compressor uses weighting filters in its loudness measurement (as specified in BS.1770) it will impact the program's frequency response. Although modern digital signal processing enables gain adjustments while introducing much less nonlinear distortion than older analog techniques they are not distortionless.

In contrast, a properly implemented loudness management system following the A/85 RP does not need to alter dynamics in any way. It simply adjusts a gain control which is applied as a fixed value across the entire program. As this is a linear process it introduces no nonlinear distortions.

These devices pose a serious potential consequence to overly aggressive or inflexible loudness regulation or enforcement. The concern is that regulated entities may decide that the effort required to comply with the CALM Act using loudness measurement and gain scaling or dialnorm adjustment techniques is excessive. Or, they may slip-up and be subject to forfeiture once too often. They may then take the easy way out and install a dynamic range compression device that is guaranteed to solve their loudness problem at the expense of audio quality. At that point the quality improvement of DTV audio largely disappears and it becomes merely a surround sound version of analog TV. Such a result would be a travesty for the consumer.

New service opportunities

Regulated entities will demand that commercial advertisers provide content which is pre-adjusted to the -24 LKFS loudness target. If they operate a variable metadata plant they may demand that advertisers provide content with measured loudness values inserted in metadata. A rapidly growing service in the television industry is the storage of commercials for download by broadcasters. When a station accepts a commercial for insertion into its stream it downloads the appropriate file from the storage provider. These commercial repositories are ideally positioned to perform loudness verification services for advertisers and broadcasters. Rather than each broadcaster independently checking, and if necessary correcting, a file it can be done by the central repository. Again, the fixed -24 LKFS target loudness model will likely prevail in practice due to its simplicity.

A Proposal

We propose that the creation of content specifications, proper work flows and procedures, along with the implementation of, and reasonable attention to, a continuous improvement audio quality system would itself constitute compliance with the statutory safe harbor rule for all content passing through the regulated entity.

In other words, a regulated entity which continuously makes a good-faith effort to maintain its loudness within the boundaries specified by ATSC A/85, checks its success at this goal and takes concrete steps to correct, and prevent the recurrence of, occasional failures would, by these actions, always be in compliance with the statute.

The statutory requirement that the regulated entity:
"installs, utilizes, and maintains in a commercially reasonable manner the equipment and associated software"
is satisfied because the equipment necessary to monitor loudness as part of the closed-loop quality improvement system would assess loudness according to the A/85 RP (which references BS.1770).
"in compliance with the regulations issued by the Federal Communications Commission in accordance with subsection (a) shall be deemed to be in compliance with such regulations."
would be satisfied because the regulated entity would be implementing the standards methods and procedures specified in A/85
"insofar as such recommended practice concerns the transmission of commercial advertisements"

This is not an argument attempting to find a loophole in the statutory requirements but rather represents an effort to also comply with the spirit of the Act. It is impossible for a regulated entity which makes a good faith effort to implement and maintain a continuous improvement quality control system not to get effective control over its broadcast loudness and deliver to consumers a satisfactory, indeed very high quality, result.

Though standards might be fairly similar across regulated entities which choose this approach to compliance, the procedures and work flows would be unique to the requirements of each regulated entity. The Commission can assess the entities commitment by using the enforcement mechanism cited in paragraph 36 of the NPRM. A regulated entity making good faith efforts with a quality improvement system could document the steps taken to identify the cause of any individual loudness management failure. The records should likewise demonstrate a clear downward trend over time in the quantity and severity of failures.

In any large or complex operation involving people, equipment, or both, mistakes *will* occur. A regulatory environment which punishes those entities which attempt to meet the regulatory goal merely because they have documentation of their failures will ultimately be less successful than one which encourages entities to document their failures and use that documentation to improve their operation in the future. The former encourages burial of operational problems not their elimination. The proper role for regulatory enforcement is to punish those entities which attempt to evade the process and reward those who embrace it as a path to greater customer satisfaction. Regulatory attention and penalties should be based on the pattern of failures and the actions in response to those failures rather than the existence of failures.

Gradually phasing in forfeiture provisions will allow regulated entities to risk occasional violations in the early years as they refine their procedures and learn the fine points of managing audio loudness. Early implementation of stiff forfeiture amounts for minor infractions or “learning curve mistakes” will encourage simplistic overkill solutions which will eviscerate the DTV promise of improved audio quality. We are not suggesting that the Commission implement rules with no teeth. Rather like a puppy, the teeth grow as the dog matures, learning when to sleep and when to bite.

Summary

We have provided comments on a wide range of questions raised in the NPRM. We have suggested a method of compliance with both the requirements and the spirit of the CALM Act that can be adapted to the unique requirements of each regulated entity. We believe the proposal reduces the risk that Commission rules will unduly burden any individual entity while achieving the result envisioned by Congress when the CALM Act was passed.

An optimal solution to loudness management requires industry wide co-operation, involving content producers, post production houses, networks, affiliates, station groups, independent broadcasters, MVPDs, advertisers, commercial aggregators, hardware and software vendors. As challenging as such industry wide co-operation would likely be under normal circumstances, it becomes even more so under current economic conditions.

We believe the industry will best be served by enforcement rules which recognize the learning curve faced by regulated entities and by the industry at large. The goal of the legislation is to effect consistency in television loudness as experienced by consumers. It would be tragic if this is achieved at the expense of the audio quality improvement digital television offers consumers. A knee-jerk insertion of level compression devices (as is now happening at numerous DTV stations) will eliminate loudness complaints but does so at the expense of dynamic range and audio quality.

Ultimately the industry needs a reasonably priced, commercially available solution which fulfills the statutory requirements yet may be adapted to the unique circumstances of each regulated entity. Our firm specializes in delivering such solutions. There are other firms in a position to do the same. We believe the market will provide a wide array of solutions at a range of price points.

We have already delivered an automated monitoring and reporting system which is now in daily use at a large network as part of its commitment to quality improvement. The long lead time before any solution was required by law is itself evidence of the managements commitment to the project. We believe this effort will result in delivery of higher quality program material to its stations, a steady reduction in noncompliant content and an improvement in consumer satisfaction.

We hope the result of the NPRM is similar improvements across the entire television industry.

Appendix

Qualis Audio, Inc. manufactures the Sentinel, an automated audio quality monitoring product which includes, in addition to numerous other capabilities, loudness measurement, logging and alarming. Acting as an electronic listener it monitors digital audio surround signals in PCM, AC-3 or Dolby E formats and supplies measurement results, logs and alarms via a local area network or the internet. Fully compliant with the 2011 version of ITU-R BS.1770, it is capable of measuring interleaved programs and commercials in a single stream. It can perform these measurements in real time or after-the-fact based on as-run logs or based on user input via the units GUI. The Sentinel technology is the subject of numerous patent applications on its unique audio assessment capabilities.

Dr. Richard C. Cabot received a BSEE, MEngEE, MS Mechanics (acoustics) and a PhD EE from Rensselaer Polytechnic Institute. His PhD thesis was on the psychoacoustics of quadraphonic sound systems and included additional studies at SUNY Albany. He then served as principal investigator for an NSF funded study on Amplified Sound as a Source of Acoustic Trauma and taught courses in acoustics and electronics, also at RPI. After 6 years in engineering at Tektronix, he co-founded Audio Precision. Dr. Cabot designed the System One analog generator and the digital and DSP sections of all the products until selling the company in 2000. Upon leaving Audio Precision in 2001 and completing an MBA at Pepperdine University, he started XFRM, Inc. a research and consulting firm in digital audio technology. After working on some home theater products he settled on digital audio for broadcast, developing the Qualis Audio Sentinel. He has held several positions on the Audio Engineering Society Board of Governors, including President and was elected Fellow. Dr. Cabot chaired the AES Subcommittee on Digital Audio Measurements for 12 years and was responsible for the development of the AES-17 standard on Digital Audio Measurement Techniques as well as directing its liason effort with the ITU study group on loudness measurement. He is a Senior Member of the IEEE, a member of the ASA and several other scientific and technical societies. He has presented numerous papers on audio technology to AES conventions and conferences as well as to other organizations. Dr. Cabot is a registered professional engineer in the state of Oregon in the fields of Electrical Engineering and Acoustics. He holds numerous patents on electronics, digital signal processing and audio measurement technology.