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Sound Level Monitoring at Live Events, Part 3–Improved Tools and Procedures

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This is the final installment in a series of three papers looking into the subject of sound level monitoring at live events. The first two papers revealed how practical shortcomings and audience and neighbor considerations (in the form of sound level limits) can impact the overall live experience. This paper focuses on an improved set of tools for sound engineers to ensure a high-quality and safe live event experience while maintaining compliance with local sound level limits. This includes data processing tools to predict future limit violations and guidelines for improved user interface design. Practical procedures, including effective sound level monitoring practice, alongside resourceful mixing techniques are presented to provide a robust toolset that can allow sound engineers to perform their best without compromising the listening experience in response to local sound level limits.

0 INTRODUCTION

Sound level monitoring and management at live events have become increasingly important in recent years because of more events taking place in densely populated areas using progressively powerful sound reinforcement systems. The use of such sound systems also has the potential to put the hearing health of the audience at risk. Despite early warnings voiced via the Audio Engineering Society [1, 2], the industry has only recently taken on a more proactive role regarding sound level regulations [3]. This delayed response has resulted in numerous problematic sound level regulations in terms of sound engineering practice [4].

The second entry in this trio of papers revealed the experience of working live sound engineers across the world, where there is a strong preference for sound level regulations with a 15-min equivalent continuous sound pressure level (Leq) time frame [5]. If there is too long a time frame, engineers will not be informed of sound level violations until after the fact, while if there is too short a time frame, engineers will be unable to explore a wide dynamic range [6]. A 15-min integration time is thought to be a suitable compromise between the immediacy of sound level compliance information and ability to deliver short-term amplitude peaks to the audience.

While the long-term focus of the industry should be to constructively influence future policy, there remains a need to provide tools and procedures for engineers who must work within the existing regulations. The solution cannot be to simply turn down the sound system. This approach is unacceptable to most key stakeholders at a live event (musicians, engineers, and audience members) since it often results in a degradation of the live event experience. Instead a more comprehensive approach should be adopted whereby well-informed system design, intelligent use of dynamic range, and effective communication of sound level monitoring data must be made available to sound engineers, system technicians, and other key stakeholders [7].

This paper presents an expanded set of tools and procedures for sound engineers to ensure symbiosis between live events and regulating/enforcing bodies. These techniques, if embraced within commercially available sound level monitoring software, will provide engineers with the ability to work with most existing sound level regulations while ensuring that the delivered audience experience maintains a high standard, independent of specific sound level regulations.

The paper begins with a comprehensive overview of practical sound level management techniques for live sound engineers, highlighting the challenges commonly encountered in the field. This is followed in Sec. 2 by a detailed inspection of various data processing options, highlighting methods to achieve a consistent responsiveness of sound monitoring data with different time frames from venue to venue. Sec. 3 applies the data processing techniques discussed in Sec. 2 to predict sound level–limit violations sufficiently in advance. The paper is concluded in Sec. 4. Recommendations for further work are highlighted throughout the paper.

1 PRACTICAL CONSIDERATIONS

While the research detailed in this paper focuses primarily on data analysis techniques for sound level management purposes, there are several user-focused aspects that should be considered when designing and implementing a robust sound level management plan. This work specifically explores user interface, augmented L_{eq} monitoring, and perceptual loudness enhancement.

1.1 User Interface

The user interface of a piece of sound level monitoring software is of paramount importance to the software's adoption and acceptance as a useful tool. Presentation of sound level information should be free of visual clutter, given that the display will often be used in an "at a glance" capacity; however there should also be clear indication of what metric is being displayed (for example, a large "95" displayed with no additional context is prone to being misinterpreted). Of course, since sound engineers generally prefer data to be displayed in different ways—some prefer a large, full-screen meter, while others want something clear but compact displayed in the corner of the screen—flexibility is an important consideration.

The use of color to indicate level thresholds, for example, is useful for quick visual parsing of traffic light–style configurations (green: compliance, yellow: warning, red: violation); however this color scheme could potentially exclude colorblind users if the data cannot be portrayed by alternative means. User interfaces that visually present the L_{eq} limit as a "target" or "goal" can lead to sound engineers "mixing to the limit" [8], or constantly running the mix to the maximum allowable level, which is undesirable.

Some engineers have suggested displaying sound level monitoring data in terms of a target live dynamic range. Such an approach would present engineers with a histogram of sound level across the performance, updated in real-time, where a target mask can be overlaid to allow the engineer to achieve a dynamic mix at the intended sound level (and save some so-called "sound capital" for the encore).

If such a dynamics-based approach were to be implemented, the mask could be pre-generated based on the local sound level regulation so that a level violation would be impossible provided the engineer mixes within the limits indicated by the mask. At the time of writing, no such user interface exists. This should be explored within further research.

1.2 Secondary L_{eq} Monitoring

Timely information and feedback are essential to permit effective decision-making regarding sound levels. In the experience of the authors, a professional sound engineer's prime interest is not maximum sound level at an event but rather a sound level that meets the expectations of the various stakeholders [9]: the audience, promoter, musicians, and sound engineer, themselves [10]. This can be referred to as preferred listening level (PLL) and is inextricably linked to being able to provide a positive experience that delivers a so-called "democracy of sound," the same high-quality listening experience to all audience members, permitting an engaging and enjoyable listening experience for all.

To achieve the PLL and an overall democracy of sound, accurate, timely sound level information is required, presented in a manageable form (as discussed in Sec. 1.1), along with a clear indication of what the long-term implications are regarding any imposed sound level regulations. As is explained in Sec. 2.1, because of the dynamic nature of music, shorter L_{eq} time frames can provide too erratic a readout to provide helpful guidance to an engineer. Long L_{eq} time frames (30–60 min) are difficult for engineers to use on their own because of the severe lag in useful information, so it is essential that short-term information is also provided. Some engineers use a sound pressure level (SPL) slow setting for this purpose, typically a 1-s average [11], to track the real-time sound level.

To provide a crude analogy, an engineer is expected to mix with the dexterity of a racecar driver but with an imposed speed limit. It is impossible to comply with a speed limit without a speedometer (sound level monitoring). Similarly, it is impossible to comply with a speed limit when using a speedometer that only displays the speed averaged over a long period of time (sound level monitoring with a long L_{eq} time frame).

To put it another way, long L_{eq} time frames are akin to braking distances, $L_{eq,60min}$ is like stopping an oil tanker on a dime [12]. Once the infringing sound level data has been recorded, there is little an engineer can do to prevent the (eventual) sound level–limit violation.

It must be noted that extremely short time frames such as $L_{eq,1min}$ or SPL slow could inadvertently cause engineers to overreact to measurement data. For example, at the start of a performance the crowd typically is screaming and shouting (especially at pop music events), which from the authors' experience has the potential to register extreme sound pressure levels (up to 112 dBA SPL over a few seconds, in some cases). Should an engineer notice such levels on the monitoring system, they could panic and reduce the level of the mix. What typically happens, though, is that this sound level peak dissipates a few seconds into a song. Engineers must keep this in mind and understand that they have control over the sound system output but not over the audience's behavior. As such, engineers should exhibit caution and use critical analysis when using such short time frames for secondary sound level monitoring.

One solution to the problem of overreacting to sound level monitoring data is to use a real-time two-channel measurement system, where inputs are taken from a measurement microphone and direct output from the mixing desk. When no significant input is detected from the mixing desk, an alert can be displayed in the sound level monitoring software so that the engineer knows that the current sound levels are not controllable. Alternatively, 1/3 octave band monitoring could be used to identify non-musical data (as used with live dynamic range [6]). When no significant low-frequency content is detected, the sound level monitoring software can flag this as uncontrollable non-musical content.

If such a system is not available, placing the measurement microphone so that it is not near audience members is a straightforward solution, although this will require a correction factor to be applied to measurements to account for the distance between the microphone and audience. Further research should inspect the effectiveness of these potential solutions, since all existing commentary on such approaches is largely anecdotal.

It must be stressed that sound level regulations do not distinguish between musical and non-musical content. All sound exposure contributes to the official measurements. Distinguishing non-program content, however, can prevent an engineer from overreacting to sound level readings that are uncontrollable.

A common technique to avoid some of the above data reliability issues is to use two L_{eq} monitors simultaneously: one set to the official L_{eq} limit and the other set to provide more timely sound level monitoring information. All entries in datasets A (2019 music festival [13]) and B (5 years of data from an international touring act [14]) that were analyzed in this paper series adopted this approach.

Dataset C (official monitoring data from multiple music festivals and one-off events in Europe, 2019) had approximately two-thirds of entries with secondary L_{eq} monitoring. In this case, the secondary L_{eq} monitoring was an official requirement.

In some countries there is a second L_{eq} limit stated in the official regulation [4]. For example in Belgium the sound level limit for music events is 100 dB A-weighted, equivalent continuous sound pressure level (L_{Aeq}) for a 60-min time frame, measured at the mix position (front of house). This is based on World Health Organization advice from 1999 [15] and an older Swiss law [16]. In practice, the time frame of 60 min results in a slowly changing sound level; therefore the use of a secondary L_{eq} time frame was added to the Belgium law as 102 dB $L_{Aeq,15min}$.

This shorter time frame allows for more immediate data presented to sound engineers, permitting greater dynamic mixing. Although the official L_{eq} time frame is 60 min, a secondary time frame of 15 min is typically the focus. In practice, sound engineers concentrate on the $L_{eq,15min}$ data and generally lose sight of the $L_{eq,60min}$ data.

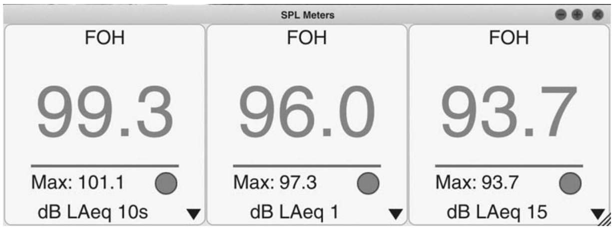

Fig. 1. Example of triple equivalent continuous sound pressure level (L_{eq}) time frame monitoring during a recent live event.

1.3 Multiple Time Frame L_{eq} Monitoring

An expansion of the secondary L_{eq} monitoring technique is to employ multiple L_{eq} time frames in parallel. For example, viewing $L_{eq,1s}$ and $L_{eq,1min}$ alongside an official $L_{eq,15min}$ monitor can offer a sound engineer more context not only on compliance status but also on sound level trends. The short L_{eq} time frames provide rapid feedback for level consistency and dynamic range, while viewing multiple time frame monitors simultaneously allows a sound engineer to see the effect of short-term fluctuations on long-term L_{eq}.

In the example shown in Fig. 1, $L_{Aeq,10s}$ is used for near-immediate feedback but with more stability than the traditional SPL slow setting. $L_{Aeq,1min}$ indicates the level trend of the last song section and can be compared against $L_{Aeq,15min}$, the enforced limit. The use of multiple L_{eq} time frames gives a best-of-both-worlds approach by which the sound engineer can derive more context to confidently deliver short-term dynamics while maintaining long-term L_{eq} compliance.

A logical extension to this approach is to devise a light emitting diode (LED) ladder whereby each individual LED represents a different time frame. The bottom green components of the LED ladder would represent the shortest time frames, middle-range time frames would be yellow, and the longest (and regulated) time frame would be in red. This would give an engineer a clear visual indicator of sound level trends in relation to the imposed limit. Since the official sound level monitoring is likely to be slower moving than the actual musical dynamics, any peaks in this metric would be like a peak hold function on a conventional meter, while the real-time level may indeed be below the limit during this time. As stressed in Sec. 1.1, care must be taken to avoid such a tool causing sound level maximization.

1.4 Perceptual Loudness Enhancement

Engineers actively seek methods of lowering sound levels at concerts while still maintaining the subjective PLL. There are several practical options utilizing techniques from broadcast and record mastering to decrease dynamic range with multiband compression, for example [17].

One of the authors has explored enhancing low-frequency content in terms of bandwidth and sound level, in which recent research indicates that there is a perceptual trade-off between bandwidth and subwoofer system level, where an increase in system bandwidth results in a lower PLL [18]. By producing enhanced mechanosensation at a lower overall sound level, audiences may experience a greater feeling of immersion, making the decrease in sound level inconsequential. However this extension of subwoofer system bandwidth could potentially impact off-site noise pollution because of the increase in very low frequency content. More research is required in this area.

Another method for perceptual loudness enhancement utilizes the phenomenon of the missing fundamental (a.k.a. virtual bass), where a low-frequency auditory sensation is stimulated through the addition of a carefully weighted series of higher harmonics [19, 20]. This technique, if used sparingly, can give the impression of enhanced low-frequency content; however its effectiveness in lowering PLL has not been explored in formal research.

A common technique loosely related to virtual bass is the use of subtle amounts of harmonic distortion to increase perceived loudness without materially inflating the objective sound level. Many engineers are familiar with this effect in the form of tube saturation, tape emulation, and the like. It is also the unfortunate generalized association between distortion and perceived loudness that contributes to this effect, whereby it has been shown in previous research that listeners often (but not always) equate an increase in system distortion with an increase in perceived loudness [21].

While the authors oppose the intentional addition of distortion into a sound reinforcement system as a matter of course, the introduction of subtle harmonic distortion should be retained for scenarios where the sound level limit proves especially difficult to adhere to.

2 DATA PROCESSING

2.1 Simple Moving Average

Virtually all official sound level regulations are based on L_{eq} measurements, as governed by Eq. (1) for instances where discrete samples are used [22].

$$L_{eq,T} = 10\log_{10}\left(\frac{1}{T}\sum T_i 10^{0.1 L_{eq,T_i}}\right), \tag{1}$$

where L_{eq} is calculated based on the L_{eq} time frame, T; sampling period of the discrete L_{eq} calculation, T_i; and L_{eq} for each individual sample, $L_{eq,Ti}$.

In Eq. (1), each sample point in time is equally weighted to arrive at the overall L_{eq} value. If L_{eq} is analyzed in sound level monitoring software with a sliding analysis window, then this approach is termed a linear-weighted simple moving average (SMA).

The chosen time frame for an SMA can have a profound effect on the role of a sound engineer. For illustrative purposes, data was synthesized with a live dynamic range (LDR) of 0 dB [6] so that $L_{eq,1s}$ remained at a constant

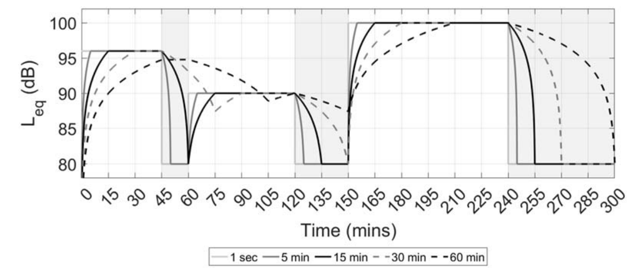

Fig. 2. Illustration of varying equivalent continuous sound pressure level (L_{eq}) time frames when using a simple moving average (SMA) on synthesized data (gray shading indicates that the stage was inactive).

value for each of the performances contained within the data. The data simulates a three-act event, where each act was fixed to a different SPL. The data was processed to show the sound level monitoring response for four common L_{eq} time frames: 5, 15, 30, and 60 min (Fig. 2). In all cases the L_{eq} analysis buffers were pre-filled (from –60 min) with synthesized audience data (set to 80 dB SPL) to allow for the L_{eq} calculation to use the intended time frame. This is akin to starting the sound level monitoring well in advance of the start of a performance (which is relatively standard practice, based on the authors' experience in the field).

There are three important observations to be made from the data in Fig. 2.. First, with an LDR of 0 dB for each act in the synthesized data, the time it takes for the chosen $L_{eq,T}$ to reach the true performance level is precisely the L_{eq} time frame, T. This means that for the first act, $L_{eq,60min}$ never presents the engineer with useful data, since the act only performed for 45 min. The lack of timely data can be avoided by starting the sound level monitoring at the beginning of the performance, since most available monitoring software calculates L_{eq} based on the available data rather than using pre-filled analysis buffers. This, of course, does not comply with the official L_{eq} definition [22].

Second, the changeover time between the first and second act was 15 min. The changeover allowed for a so-called reset of $L_{eq,5min}$ and $L_{eq,15min}$. However $L_{eq,30min}$ and $L_{eq,60min}$ were still influenced by the first act during the second act's set. In both cases, L_{eq} begins high since the first act performed at a higher level than the second. There is a characteristic dip below the second act's level (as seen at 105 min in Fig. 2), though, due to the influence of the changeover L_{eq} data, before L_{eq} settles on the true performance level. The high initial L_{eq} would likely be recognized by the engineer as inaccurate, but it is probable that the dip below the actual level (at 105 min in this example) would give the engineer a false impression of available headroom in relation to the sound level limit.

Third, since the final act is higher in level than the second act, there is no lingering effect from the previous act when the final act begins. In all cases, the L_{eq} reaches the actual performance level after the full L_{eq} time frame has elapsed.

Considering the analysis example in Fig. 2., the shorter the L_{eq} time frame is, the closer the SMA follows the instan-

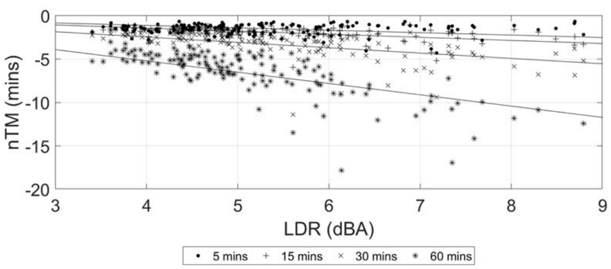

Fig. 3. Normalized time to reach actual performance level (nTM) versus live dynamic range (LDR) for 164 sets of real-world performance data. Solid lines represent best-fit for each $L_{eq,T}$ using a linear regression model.

taneous sound level. This is good in the sense that it gives an engineer near real-time information but bad in that the sound level monitoring will be more sensitive to short-term amplitude peaks, thus limiting the ability of an engineer to fully utilize a sound system's dynamic range.

Long L_{eq} time frames such as 60 min, and to a lesser extent 30 min, are unusable for sound engineers. While the long time frames allow for strong dynamics to be incorporated into the performance, the engineer will have no meaningful way to comply with the sound level limit, since any adjustments will be made in response to sound levels up to 1 h in the past. Such relationships were confirmed though the analysis of data from over 300 performances in [5].

The synthesized data imposed an LDR of 0 dB, which would never be the case in reality. To inspect the responsiveness of L_{eq} with an SMA, datasets A (23 performances at a 2019 music festival [13]) and B (141 performances from an international touring act [14]) were analyzed. The analysis was based on the relationship between LDR and time it took for $L_{eq,T}$ to reach the mean performance level (using a cumulative rolling average of the $L_{eq,1s}$ or $L_{eq,1min}$ data, depending on what was available within each dataset).

The time lag of the $L_{eq,T}$ data was normalized based on the L_{eq} time frame, so that 0 min indicates a similar time response as seen for the synthetic data (e.g., 5 min to reach the true performance level for $L_{eq,5min}$), with negative times indicating that the $L_{eq,T}$ reached the actual performance level faster than the L_{eq} time frame. As before, L_{eq} time frames of 5, 15, 30, and 60 min were analyzed (Fig. 3). Only A-weighted data was considered since, at present, most sound level regulations contain primary limits using L_{Aeq} [4].

The data in Fig. 3. was further analyzed using a linear regression model to arrive at a generalized equation that can be used to relate LDR and L_{eq} time frame (T_{Leq}) to the normalized time required to reach true performance level (*nTM*), as expressed in Eq. (2).

$$nTM = -0.0218 T_{Leq} LDR. \tag{2}$$

What emerges is a direct relationship between LDR (musical dynamics of a performance) and the timeliness of the sound level monitoring information. A low LDR results in accurate L_{eq} information becoming available only after the full L_{eq} time frame has elapsed. A higher LDR, however, causes L_{eq} to reach the true performance level several minutes faster than expected.

Alternative comparisons were inspected aside from LDR, including difference in level between the first 5 min and remainder of a performance, difference in level between the first and last 5 min of a performance, and absolute level over the first 5 min of a performance. LDR was the only measure to show any meaningful relationship to the nTM.

The relationship between LDR and sound level monitoring timeliness should not be confused with greater responsiveness of the L_{eq} monitoring, since the full time frame of data will still equally influence L_{eq}. This demonstrates that a performance exhibiting a high level of musical dynamics will allow a sound engineer to more quickly ascertain the act's sound level in relation to the imposed L_{eq} limit. The authors do not recommend that sound engineers compromise their mixing techniques to allow for somewhat timelier L_{eq} information. The relationship is a point of intertest but may be of limited practical use.

2.2 Exponential Moving Average

All conventional sound level monitoring platforms utilize an SMA. This could result in a non-ideal operating environment for sound engineers, where too short an L_{eq} time frame (below 5 min) will limit the engineer's ability to explore a wide dynamic range, while too long a time frame (above 15 min) will prevent the engineer from receiving timely feedback on sound level–limit infractions.

Many engineers circumvent this issue (at least for long L_{eq} time frames) by using a secondary L_{eq} time frame (as discussed in Sec. 1.2), which is typically unrelated to the official L_{eq} limit but provides more timely information. To date, all instances of this practice use SMAs.

Use of a secondary L_{eq} monitor with a shorter time window (and possibly a higher limit) is often (but not always) outside official regulations, where the longer time window is what matters. Instead of complicating the monitoring process, it would be ideal to have a metric that was useful for the engineer, which takes into account the full time range of data that the official L_{eq} limit specifies.

An exponential weighted moving average (EMA) is one possible solution [23, 24]. EMAs are commonly used when analyzing the stock market, where they give more weighting to the most recent time data. EMAs are typically defined using a recursive equation [Eq. (3)].

$$L_{eq}(n) = 10\log_{10}\left[10^{\frac{SPL(n)}{10}}\left(\frac{SF}{1+N}\right) + 10^{\frac{L_{eq}(n-1)}{10}}\left(1-\frac{SF}{1+N}\right)\right], \tag{3}$$

where the EMA L_{eq} at the n^{th} discrete time step is defined by the current sound pressure level, *SPL(n)*; the L_{eq} at the previous time step, $L_{eq}(n-1)$; the smoothing factor, *SF*; and length of the L_{eq} time frame, *N*, in units corresponding to the discrete time step, *n*.

In the literature [23, 24] a variety of smoothing factors are explored, but in practice a smoothing factor of 2 appears

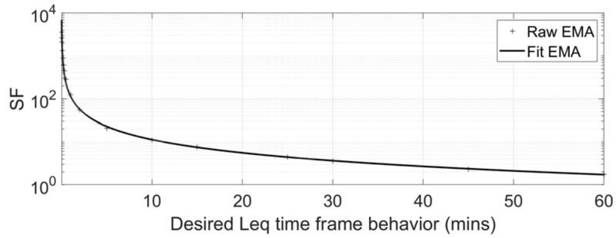

Fig. 4. Example of smoothing factor (SF) for 60-min exponential moving average (EMA) to achieve the desired responsiveness of the sound level monitoring in line with a shorter equivalent continuous sound pressure level (L_{eq}) time frame.

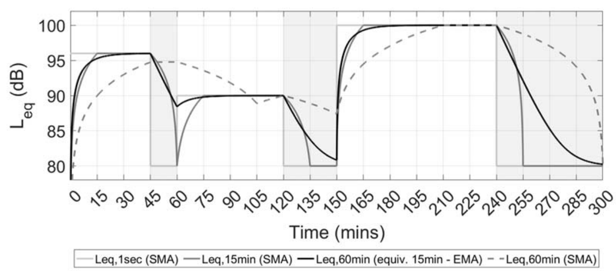

Fig. 5. Illustration of difference between simple moving average (SMA) and exponential moving average (EMA) data analysis using synthesized equivalent continuous sound pressure level (L_{eq}) data (gray shading indicates that the stage was inactive).

to be most common (where nearly all examples of use fall within the financial sector) without any discussion of how this was determined.

Smoothing factor was explored in this work to see whether it can be used to cause a long–time-frame EMA to respond as if a shorter time frame was being used. Since the 60-min time frame is what some organizations (including the World Health Organization) currently recommend (although this may change in the near future) [3, 4, 15, 25, 26], a best-fit function was derived to relate SF to the desired responsiveness [Eq. (4)].

$$SF_{60\,\mathrm{min}} = \frac{5x10^{-3}}{\log_{10}\left(10^{-4}T_i + 1\right)} - 0.2, \tag{4}$$

where the SF for a 60-min EMA is related to the desired equivalent averaging time (T_i, in minutes) (Fig. 4.).

While the correct choice of SF does not lead to the EMA directly following the SMA, it does cause the EMA to respond at the same rate as the corresponding SMA (assuming T_i for the EMA is the same as the SMA). This is illustrated in Fig. 5. by comparing a 60-min–time-frame SMA and EMA, where the EMA is calibrated to exhibit the responsiveness of a 15-min SMA (a 15-min SMA is included for a direct comparison).

In the above example, the 60-min EMA was calibrated to exhibit the responsiveness of a 15-min SMA. The rise time of the EMA can be seen to resemble the targeted SMA, although it takes longer to settle on the true performance level and decay to the background noise level since the EMA includes all data in the longer time frame in its calculation. Compared to the 60-min SMA, however, the EMA can provide an engineer with more timely and useful data.

3 VIOLATION PREDICTION

With sound level regulations quickly becoming commonplace across the globe, the need for tools to enable sound engineers to easily comply with local limits is becoming increasingly important. Without tools that assist with compliance, engineers run the risk of causing live events to incur large fines and even lose the ability to hold future events at a given location [4].

While it is typically the case that sound engineers utilize a secondary short time frame L_{eq} monitor when working with regulations stipulating long L_{eq} time frames, such practice could cause engineers to mix a performance at a lower level than is required by the L_{eq} limit. To overcome this limitation, a form of L_{eq} prediction is required.

Such a technique has been developed over several years by one of the authors while working as a sound level management consultant at large-scale European music events, although this is not the only option for sound level–limit violation prediction. A well-known predictor, the maximum average manager within the software 10EaZy [27], is commonly used by practitioners, and another notable effort [28] was detailed recently.

The prediction technique used within this research provides a forecast of the imposed $L_{eq,T}$ assuming the SPL will fluctuate similarly to what was previously recorded in the data (whereas the maximum average manager provides analysis largely based on the engineer maintaining the current SPL). The procedure examines the existing L_{eq} (SMA) data that will impact the future L_{eq} reading being predicted. If the prediction is targeted for X min into the future and L_{eq} time frame is Y min (assuming Y is always greater than X), an $L_{eq,(Y-X)min}$ (SMA) is calculated. While this constitutes a certain proportion of the predicted L_{eq}, X-min worth of data has yet to be logged, hence the need for a prediction. This is achieved by using an $L_{eq,Xmin}$ (EMA) that is smoothed so that the responsiveness of the X-min EMA resembles that of a Y-min SMA, remembering that this is meant to serve as a prediction for $L_{eq,Ymin}$ X min into the future.

The final prediction is made using a weighted combination of the SMA and EMA, where the weighting coefficients are initialized to represent the proportion of the overall time frame for each, (Y–X)/Y and X/Y, respectively. The weightings are updated at each time step by examining the current difference between the SMA and EMA values and defining the SMA weighting according to Eq. (5). The EMA weighting is derived by subtracting the SMA weighting from one.

$$W_{SMA}\left(n\right) = \frac{Y - X}{Y} - \frac{X}{10Y}\left(L_{eq,X\,\mathrm{min}} - L_{eq,(Y-X)\,\mathrm{min}}\right). \tag{5}$$

The multiplier in the denominator of the second term in Eq. (5) is to limit the influence of the data at the current point in time. Without this, the prediction becomes less stable. The greater the proportion of the X-min look-ahead calculation that is based on the EMA, the greater the expected

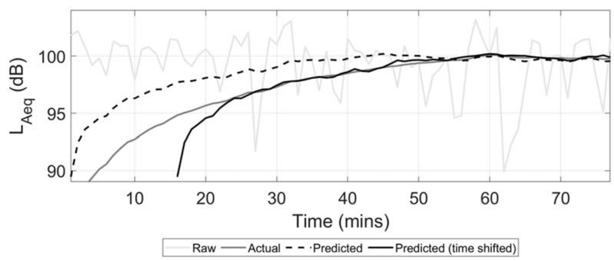

Fig. 6. Example equivalent continuous sound pressure level prediction for A-weighted data (L_{Aeq}), where $L_{Aeq,60min}$ was predicted 15 min into the future.

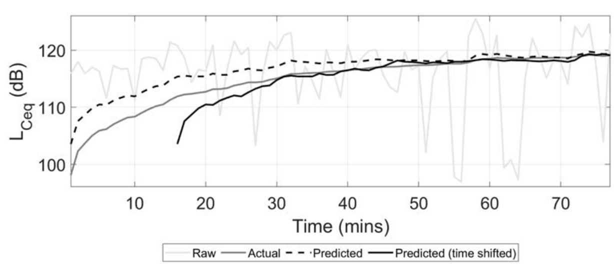

Fig. 7. Example equivalent continuous sound pressure level prediction for C-weighted data (L_{Ceq}), where $L_{Ceq,60min}$ was predicted 15 min into the future.

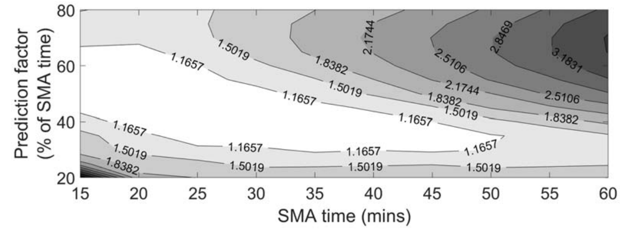

Fig. 8. Mean root mean square error (RMSE) (dBA) for A-weighted $L_{eq,T}$ prediction algorithm examined over data from 141 events.

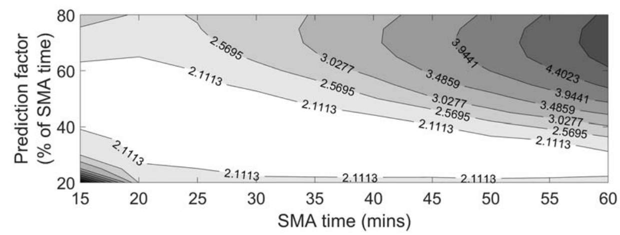

Fig. 9. Mean root mean square error (RMSE) (dBC) for C-weighted $L_{eq,T}$ prediction algorithm examined over data from 141 events.

error in the prediction (since the EMA will be attempting to predict for a greater span of unavailable data).

By way of example, a 15-min L_{eq} violation prediction was implemented for an imposed $L_{eq,60min}$ limit. In this instance, the prediction is made using $L_{eq,45min}$ (SMA) and $L_{eq,15min}$ (EMA) data (Figs. 6. and 7).

For both weightings, the 15-min prediction accurately tracks the true SMA roughly 15 min into the performance. The root mean square error was calculated as 0.84 and 1.6 dB for A and C-weighted data, respectively. While this will fail to give an engineer an accurate warning of a potential limit violation at the beginning of a performance, it will provide useful warnings for most of the performance aside from the so-called initialization period, which is relative to the prediction time frame. Without such a predictive metric, an engineer would need to rely on a second L_{eq} monitor using a shorter time frame (which could have unintended negative consequences on the musical dynamics).

To further investigate the effectiveness of the prediction algorithm, dataset B (taken from 5 years of an international touring act's performances) from parts 1 and 2 of this paper series [5, 6] was processed with an imposed L_{eq} limit ranging from 15 to 60 min (SMA) and required a prediction from 20% to 80% of the full SMA. The algorithm's performance was rated based on mean root mean square error across all 141 events in the dataset (Figs. 8 and 9).

The prediction algorithm's behavior demonstrates poor performance for relatively short and long-term predictions (below 25% and above 60% in this example). The short-term error is due to the EMA operating on a limited number of data points (note that the analysis presented in Figs. 8. and 9 was carried out using $L_{eq,1min}$ data); therefore the EMA will be highly sensitive to temporally anomalous fluctuations in sound level. The long-term errors indicate the prediction algorithm will lose accuracy when forecasting beyond half of the L_{eq} time frame under analysis, which is expected because of the dynamic nature of live music.

The best-performing A-weighted predictions indicate accuracy within nearly 1 dBA, while the best C-weighted predictions are accurate around 2 dBC. Note that the error calculation includes the initialization period of the prediction algorithm's output, which exhibits an initial rising edge (as illustrated in Figs. 6. and 7). Such predictions can provide engineers with a warning dozens of minutes (and multiple songs) before a local L_{eq}-limit violation occurs and can therefore permit engineers to fully explore the available dynamic range without constantly adjusting to ensure L_{eq}-limit compliance.

4 CONCLUSIONS AND FURTHER WORK

Live event sound level regulations are fast becoming widespread across the globe. Such limits are typically in place for two reasons: (1) to protect the neighboring community from excessive noise pollution and increasingly (2) to protect audience members from hearing damage.

Critically, these sound level restrictions do not imply the need to simply turn down sound reinforcement systems. With appropriate system design, including a focus on the so-called "democracy of sound," and the use of an informed selection of the tools and procedures outlined in this paper, an engineer can simultaneously achieve the subjective preferred listening level while complying with the local sound level limit.

Some potentially useful tools have been presented here, including an enhanced data processing technique, whereby an SMA and EMA can be implemented within a sound level prediction algorithm to present an engineer with a look-ahead sound level in relation to a long L_{eq} time frame. This has been shown to give accurate predictions (within roughly 1 dBA) over 30 min into the future. Since it is unlikely that official limits will deviate from exclusive use of SMAs, a prediction such as this, along with use of secondary L_{eq} monitoring, will help an engineer ensure sound level–limit compliance without negatively affecting the listening experience.

It is critical for software providers to develop sound level monitoring user interfaces that provide this sort of useful information without encouraging engineers to use the imposed limit as a target, which has been shown in previously published research to sometimes occur [8]. Additionally, future developments in user interface should consider focusing on live dynamic range [6] and the ability to flag sound level monitoring data that is outside the control of the engineer (principally crowd noise). Regarding the introduction of distortion to enhance perceptual loudness at live events, this practice should only be used in extreme situations, where other tools and procedures are inadequate, instead of as standard practice.

Overall, this paper series sets out a comprehensive analysis of sound level monitoring and management at live events. New and improved algorithms (such as live dynamic range [6] and L_{eq} prediction) have been detailed and validated using captured data from hundreds of live events. The current voice of the live sound engineering community has been captured through a detailed survey [5], drawing clear conclusions regarding attitudes and common practices across the globe.

Adding the procedures and tools detailed in this paper to the existing collection of standard sound level management techniques, the authors hope to have provided a clear focus for the industry as a way forward to ensure symbiosis between live event professionals, audience members, regulating/enforcing bodies, and communities potentially impacted by such live events.

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Jon Burton is a live sound engineer with many years of touring experience and is currently a senior lecturer at the University of Derby. Jon has toured with acts as varied as Radiohead, Stereophonics, Lulu, and for the last 15 years The Prodigy. Jon has worked at the highest level for over 30 years. Along with live sound, Jon has worked in radio and TV, on such shows as "Later... with Jools Holland." Jon is also a partner in the Laundry Rooms studio complex in Sheffield, which he built from scratch with fellow engineer Dave Hadley. Jon regularly lectures at universities and colleges in the UK and abroad. Having no prior formal education in sound, Jon completed an M.Sc. in Music Technology from York University in 2017 and became a Fellow of the Higher Education Academy. Jon is a part-time writer for *Sound on Sound* magazine and contributes to online education site *Soul Sound*.

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