Study of Audio Loudness Range for Consumers in Various Listening Modes and Ambient Noise Levels

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Abstract - *Various systems have been developed for consumer audio playback to control loudness range or limit maximum loudness; these systems have generally been ad hoc developments that vary in operational controls and performance. Many audio control systems are designed to focus on the signals, such as electrical signal peaks, rather than effect of loudness variation on listeners. This study, funded by the Consumer Electronics Association, investigates the range of audio program loudness that is desired by listeners in typical noise environments. Listening conditions include the ambient noise environments typically encountered by listeners with fixed and portable audio playback equipment, such as average homes, business offices, automobiles and public transit. The results may inform the design of audio loudness management control systems for future consumer audio playback systems.*

Introduction

In the production and transmission of audio content, automatic systems have been available for more than 70 years to control audio signals. For good reason, these systems are focused on the technical needs of the *channel*, in order to avoid overload and to optimize the dynamic range of the medium. Their designs are seldom explored in an empirical way, and have generally been estimates of listener's needs for loudness range, without awareness of the end listening conditions [1]. However, it is likely that the loudness range desired by listeners is dependent on the amount of ambient (background) noise in the listening environment. Ambient noise conditions typically encountered by listeners may include average home noises, business office machines and background chatter, automobile cabin noise and, particular to portable audio equipment, public transit. To investigate the relation between ambient noise and loudness range, this study measured the sound levels desired by listeners in specific noise environments at known sound pressure levels. The audio content was derived from popular music, fine arts performances, newscasts and talk shows currently distributed by Internet audio streams, radio, etc. Both loudspeaker and earbuds listening modes were tested.

The Effects of Excessive Loudness Range

Figure 1 illustrates a condition in which the audio level (variation in loudness) drops below the ambient noise level in some sections. This 30-minute sample of audio was taken from an Internet music streaming service, using a "movie music" genre selection. Both loudness, based on the ITU-R BS.1770 meter, and signal peaks are shown [2]. For the sake of illustration, the change in loudness in this sample is assumed to be greater than listeners prefer. (This is not intended to suggest that listeners should dislike a larger loudness range, which can impart naturalness and even excitement to the sound, in a suitable listening environment. As discussed further below, however, listeners may dislike unnatural or unexpected variations in loudness and alternatively, reaching for the volume control to manage undesired variations in loudness.)

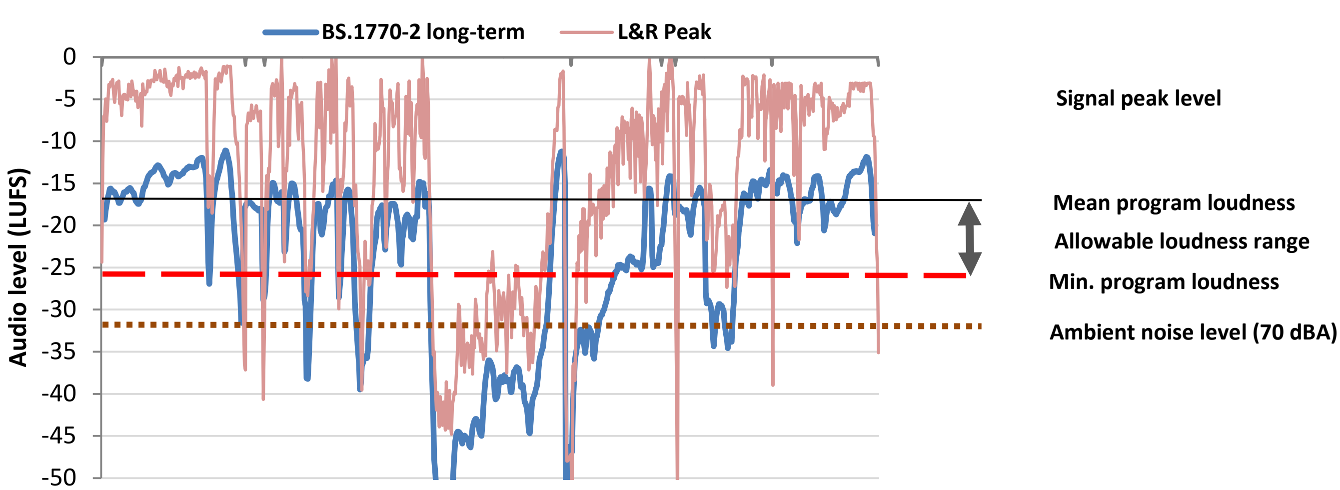

Figure 1 - Measurement over a half-hour period of musical content from an online music service. This sample includes commercial announcement breaks, indicated by tick marks on the top line.

The audio content is shown above with a horizontal line, labeled Mean Program Loudness, which was measured with an ITU loudness meter using the LUFS (loudness units, where 1 LU = 1 dB, relative to full scale) to the left. The mean (integrated) loudness of the entire audio sample is 16 LUFS, shown by the horizontal black line. Using the sound pressure level scale in Figure 2, "Stereo music" is indicated at a sound pressure range of 60 to 80 dBA (an acoustic level measurement that attempts to match the response of the human ear to noise). A background noise environment of 45 to 67 dBA represents a typical office (contributed by air handling systems, distant conversations, computer equipment, etc.), and so on.

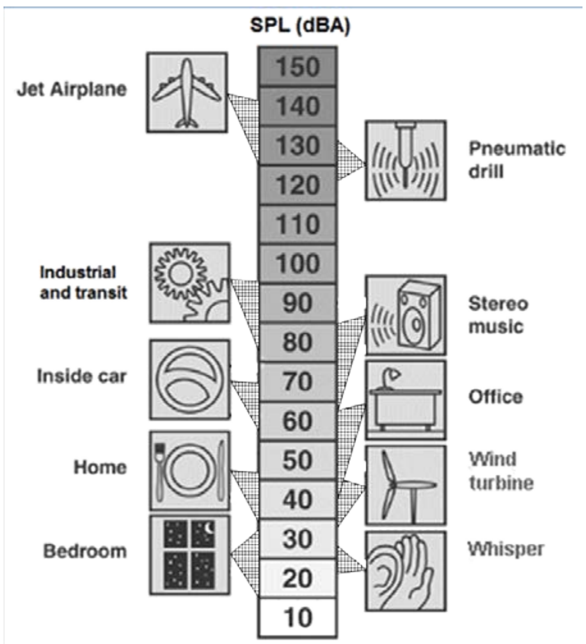

Figure 2 - Some examples of noise environments and ranges of sound pressure level, in dBA.

If noise in a car averaged 70 dBA, it is apparent that some sections of the audio would be equal to, or less than the ambient noise. From a review of the SPLs in Figure 2, we can see that some common environments produce sound levels that may prevent some audio content of wide loudness range from being heard.

In addition to hearing the content, a related issue with wide loudness range is variation that exceeds the listener's preference, such as program audio with low-level dialog is interspersed with loud dialog or sound effects. This can affect the listener's enjoyment, and possibly even render sections of the audio inaudible. Sudden increases also may disturb a sleeping partner or neighbor, or simply exceed the tastes of the listener. Thus, it is important to limit variations that exceed the requirements of the listener.

Listener Control of Playback

Listeners are, of course, at liberty to adjust playback volume, but the necessity to readjust the control during a program has a cost in terms of annoyance. The authors conducted an earlier test of listener reaction to volume changes during audio streams [3]. The study found the reactions to adjusting the volume are indicative of how listeners may feel about "gain riding" a program with too much variation in loudness. (Volume changes are not the same as loudness range, as loudness range is usually a natural part of the audio content, while loudness shifts are inadvertent changes in level during a program.)

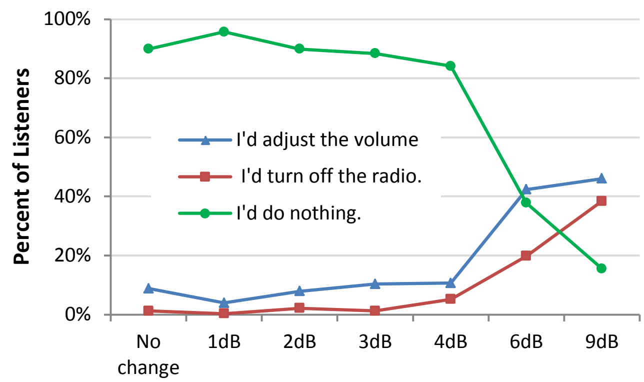

Figure 3 - Listener's behavior to change in loudness IF the change was frequent. Loudness units (LU) with the ITU loudness meter are equivalent to dB

Figure 3 shows that listeners were inclined to "do nothing" until the level shifts reached 4 dB, beyond which listeners decided they would adjust the volume knob. At a level shift of 6 dB, another 20 percent of listeners would "turn off the radio". At a change of 9 dB, the percentage of listeners who would adjust the volume did not increase much, but nearly 40 percent of listeners said they would stop listening. This result suggests that listeners tolerate a degree of loudness range variability, beyond which they may give up on the program.

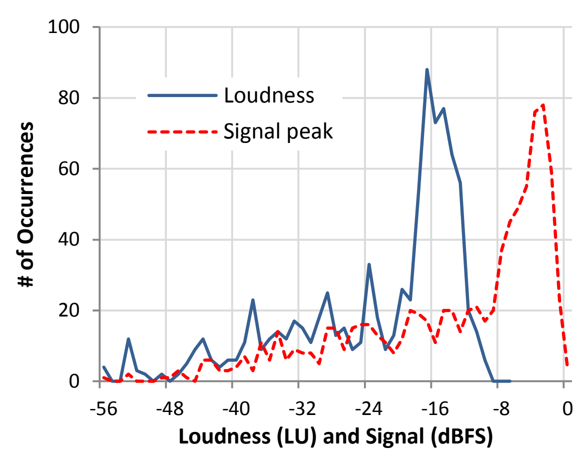

Figure 4 - Histogram of audio loudness from audio measured in Figure 1

Figure 4 is a histogram based on the audio levels measured in Figure 1 showing ITU loudness and signal peaks. Despite the relatively large loudness range of this audio, the instantaneous signal peaks show a common effect with program audio that has been pre-processed: the distribution of audio peaks end abruptly at some level (near 0 dBFS, due to peak-limiting in this case) with the majority of occurrences of the signal bunched close to the maximum. The auditory system hears a much more time-averaged signal, with emphasis on frequencies above 2 kHz, as measured by psychoacoustic studies [4]. The ITU loudness measurement uses a frequency weighting filter and time integration to represent the perceived loudness of program audio. In this study, audio loudness was measured and displayed using a Short Term integration time, having a 3-second averaging interval.

Because audio loudness peaks are often distributed around a maximum, the test planning recognized that listeners usually hear drops below an average level. Realistic audio samples, then, would have decreases below the average, rather than increases above the average.

Test Layout

The current test presented listeners with a series of one-half to two minute audio samples. Each sample was played back at program levels that vary according to the original production. Listeners were surrounded by noise fields, such as an office environment, roadway noise within a car, etc. These environmental noises may interfere with the subject's hearing of the audio samples. The test subjects were able to readjust playback gain during each sample as necessary to overcome the environmental noise and to suit their taste. Different audio samples were played with different levels of environmental noise during the course of the test. A MATLAB program managed the test playback and continuously recorded the listener's playback settings during each audio sample.

The test included 37 participants under the age of 65, recruited from the general public and NPR staff. (Those recruited from NPR staff could not be involved with audio engineering, as audio engineers may have predispositions to loudness range that may not represent the general public.) The test required less than two hours of participants' time, including instructions by the experimenter and separate listening sessions with loudspeakers and earbuds.

Testing was conducted in NPR Labs' Audio Laboratory, an acoustically isolated and treated room approximately 18 feet by 23 feet in size. The layout of the room, shown in Figure 5, placed the test subject near the room center, sitting in a swivel armchair. The experimenter was positioned to the participant's far left with an equipment cart containing the computer that ran the audio testing and recorded the test subject's responses.

The test subject was centered between four Mackie HR824 bi-amplified loudspeakers, labeled "LF", "RF", "LR" and "RR", mounted on stands approximately 3 feet above the floor. The speakers reproduced the environmental noise that surrounded the test subject. A low table with a small flat-screen monitor was positioned in front of the listener to view the test questions and control the playback volume.

Figure 5 - Audio test room showing mid-field monitors (yellow cones) and playback volume control (inset)

Half of the audio material during the listening test was heard over loudspeaker. For these tests, the left and right main speakers were KRK Rokit PR10-3 mid-field monitors, a 3-way design with a rated frequency response of 31 Hz to 20 kHz.

Figure 6 - Apple EarPods®

The other half of the listening test was conducted with earbuds, which are popular with users of smart phones, tablets and music players. The device, shown in Figure 6, was the Apple "EarPod®", which was selected because it is one of the earbud products most widely used by consumers, and because it is an open-air type, that is, having minimal isolation (acoustic attenuation) of outside sounds. One study reports the isolation (attenuation) of outside sounds to be less than 2 dB [5]. Noise-blocking earbuds and super-aural (over-the-ear) headphones were unsuitable because they would attenuate the ambient noise environment heard by listeners. This would have varied the results according to the isolation performance of each model of earbuds.

Listener Controls and Data Collection

A small flat-screen monitor was positioned on a small table in front of the listener. On the screen, listeners were able to view on-going test instructions and track their progress through the overall test. Each audio sample was presented in sequence and played by pointing and clicking on a box next to the track label on the screen.

During audio sample playback, listeners had the option to change playback gain of the loudspeaker or earbuds in response to changes they heard in the audio loudness. Their changes were stored by the computer and later compared to the loudness of the sample.

Listeners' playback gain could be controlled by a USB control "knob" on the table in front of them, seen in the inset of Figure 5. Listeners could raise or lower their volume by spinning the knob with their fingers: clockwise to raise loudness and counter-clockwise to reduce loudness, at any time during the sample playbacks. The experimenter instructed each listener on the operation of the screen and playback gain control before their test, and asked them to adjust volume for the most pleasing playback effect. They were informed that they could change their playback level at any time during each sample, but they could leave the level in place if they were satisfied with the sound.

Noise Environment and Playback SPL

The A-weighted sound pressure level of the environmental noise, at the position of the listener's head, was measured and set in advance of the experiment. During the test, the environmental noise started at least 20 seconds before the playback of any test material to allow the test subject to become accustomed to the conditions. Measurements of the loudspeaker playback system were taken with a Tenma Digital Sound Level Meter, model 72-947, at the same position of the listeners' head. The SPL of the noise environments, using slow averaging, are listed in Table 1.

Table 1 – Listening modes, sound pressure levels associated with each environment and test genres

Listening Mode		Noise Environment (dBA)		Genre Series 1 & 2			
				Classical	Speech	Pop	Jazz
Loudspeaker	Earbuds	Quiet office	42	C1	SP1	P1	J1
	Earbuds	Restaurant chatter	53	C2	SP2	P2	J2
	-	Outdoor roadway	65	C3	SP3	P3	J3
	-	Vehicle cabin	67	C4	SP4	P4	J4

The loudspeaker tests included an automotive environment simulating the in-cabin noise of a vehicle at highway speeds.[1] The quiet office and restaurant recordings were selected from sound effect tracks with care to avoid any distracting or sudden noises that might disrupt the listener's hearing of the program audio. To minimize phase effects from the four loudspeakers producing the environmental noise, the stereo recordings were played back with a small amount of DSP reverberation. The slight reverberation avoided "phasiness" in the sound field when one moved their head around the middle of the sound field.

In lieu of the in-vehicle noise, the ear bud tests included a public transit environment, similar to riding on a modern light rail system. After the tests, however, examination of the data revealed a number of cases where listeners turned up the playback gain to maximum. A smaller number of such cases occurred with the outdoor roadway environment. This did not occur with the loudspeaker tests because higher potential SPL was available. Because it could not be determined if listeners desired more playback gain, a decision was made to exclude all of the public transit and outdoor roadway noise test data from earbuds.

[1] The stereo recording was taken from an actual passenger car on a concrete roadway at 60 MPH. The noise level at the center of the cabin was 67 dBA; the same level was used for playback in the audio environment.

Measurement of the sound pressure level from the listener's loudspeaker playback system was made with one-octave pink noise centered at 1 kHz. The listener's playback SPL was determined with the test noise recorded at a level of -23 LUFS, the same as the reference level of recorded program material. With the playback gain control at full, the SPL was 81 dBA. For the Apple EarPods, the electrical voltage at the terminals was measured using a 1 kHz tone. With the playback gain control at full, the voltage corresponded to a measured SPL of 71 dB [6].

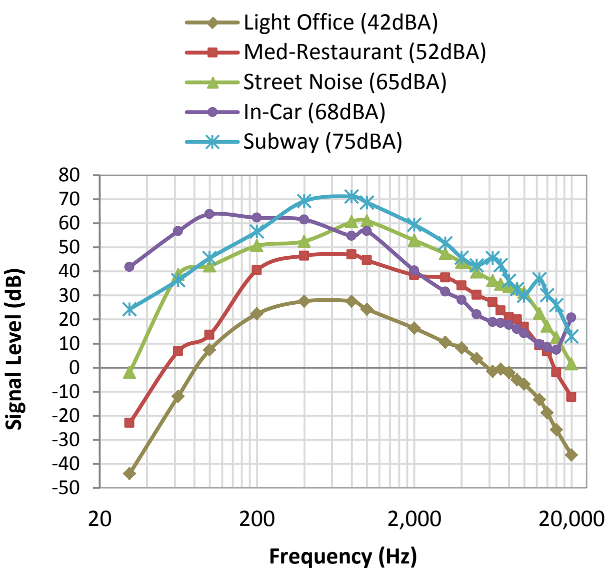

Figure 7 - Spectral distributions of noise environments

The spectral distribution of the noise environments can affect the listener's playback volume preferences, in addition to the individual dBA measurements for sound pressure level. To help illustrate the differences in noise environments, Figure 7 shows the spectral distribution for each of the noise environments.[2] The distribution of the medium-level restaurant environment is similar to the quiet office from 200 to 2,000 Hz, neglecting the 10 dB overall difference in SPL, with a slight rise above 2,000 Hz. The in-car environment, while 16 dB louder than the restaurant environment in A-weighted SPL terms, has the same or less power at frequencies from 2,200 Hz to 7,500 Hz, but nearly 50 dB more power at 40 and 60 Hz. The distributions of street noise and subway (public transit) are relatively stronger in the mid-frequencies from 300 or 400 Hz to 2000 Hz than the light office and restaurant noise.

Selection and Measurement of Audio Test Material

The audio material for loudness evaluation was comprised of four genres: classical, popular and jazz music, and dramatic (spoken) performance. The samples ranged in length from approximately 70 to 100 seconds, all with a loudness structure in three distinct sections:

- A beginning section of 12 to 30 seconds, recorded at a target level of -23 LUFS. This gave the listener an opportunity to establish a desired playback loudness; (the system gain structure and speaker or ear bud drive were chosen to make the beginning section softer than most listeners would prefer, which encouraged them to actively set the desired playback gain);
- A second section that dropped in loudness, lasting either approximately 25 or 50 seconds, to test whether listener accepted this loudness or desired a higher level by adjusting their playback gain;
- An ending section that equaled the loudness of the beginning section, to test listener's response to an increase in loudness that may include more playback gain added in the second section.

To reduce listener fatigue and avoid familiarity with audio selections, music and speech needed to be unique for each trial. Within each genre, audio was selected with an almost identical profile to allow data comparisons between listening environments. A total of 64 samples, 16 in each genre, were prepared. Table 1 also lists the four genres associated with each noise environment for the loudspeaker playback. For the earbud tests, only the 42 dBA and 53 dBA noise environments apply.

Finding samples meeting the above conditions by hand (and ear) is difficult. Consequently, a MATLAB program was developed to "crawl" through more than 3,000 music and dramatic speech recordings in NPR's Audio Library, filtering the recordings approximating the "normal-low-normal" sequence. The resulting candidates were checked by ear and by ITU loudness charts. The chosen samples were cut from the full-length recordings at appropriate start and end points. Level of the opening and closing sections were normalized to -23 LUFS. To provide better accuracy in the listener tests, ITU loudness within the middle sections were adjusted by audio editor by up to ±2 LU to match their same-genre samples.

Figure 8 charts the loudness range test procedure using the 102 second classical sample C2. This example includes an actual listener response using loudspeakers. In the figure, the ITU Short Term Loudness of the sample is indicated by the green line in the lower part of the graph. In the opening section of the sample, from the start to point A, the loudness is set to an average of (normalized) to -23 LUFS. The loudness of this opening section is standard for all the samples.

[2] Each audio file was analyzed with an 8192-point Fast Fourier Transform and the values in 20 frequency bands were adjusted to the A-weighting for each band's frequency. The signal power of the 20 frequencies for each environment was summed and offset to the same value as the measured, A-weighted sound pressure levels.

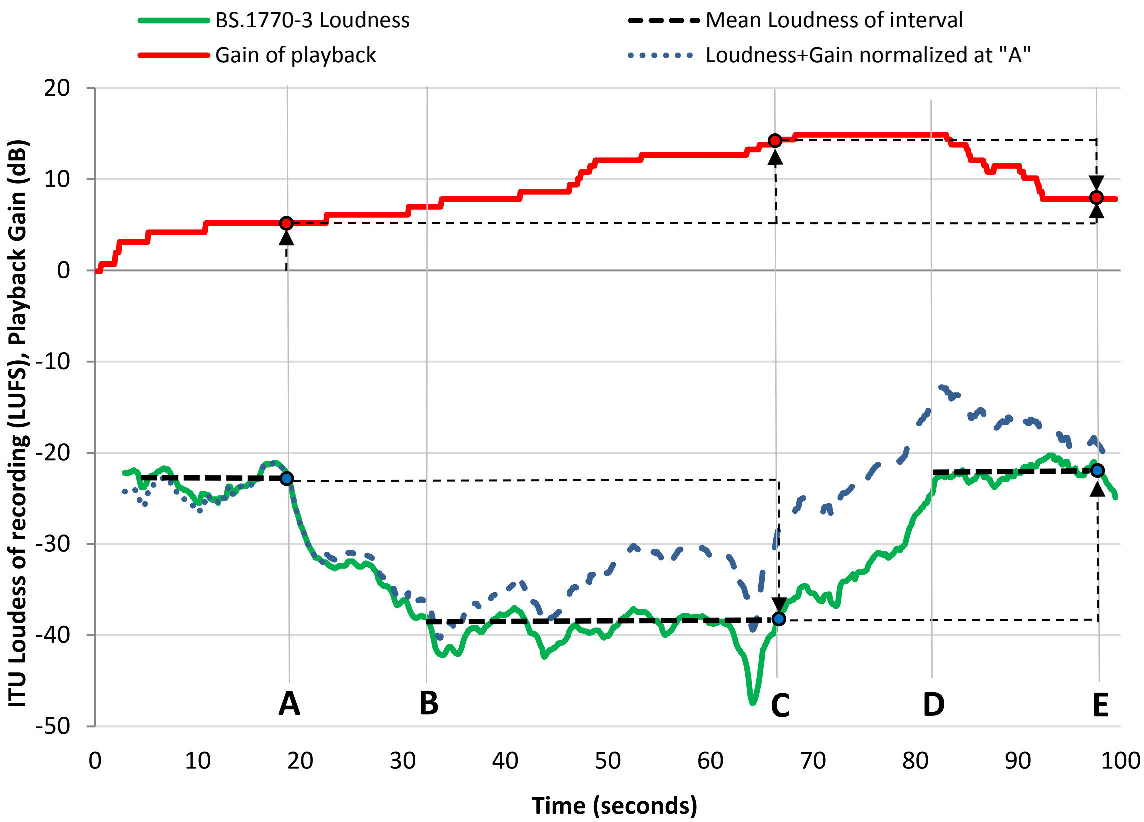

Figure 8 - Chart of an audio sample with actual listener adjustment of playback gain

As in all samples, after inflection point A, the loudness drops to point B, at which the audio levels off and then continues roughly the same loudness until point C. In this sample, the mean loudness between B and C is approximately -38 LUFS – a drop of 15 LU from point A. In all samples, the loudness increases to a third section, from point D to point E, which has a mean loudness of approximately -23 LUFS. All of the inflection points were pre-determined by visual study of the loudness test samples.

For this example, the environmental noise heard by the listener was continuous, unintelligible conversation from multiple voices in a restaurant environment (but free of loud and distracting 'clanks' from plates and silverware). The average SPL of this environment was approximately 53 dBA.

The chart displays this listener's playback gain adjustments with the red line. (The gain was programmed to always start 23 dB below maximum, but is shown at 0 dB here for simplicity.) Subject #2 raised the gain by approximately 5 dB in the first 10 seconds, and then left it in place for another 10 seconds. This subject's desired playback gain was posted at point A. After point A, the loudness began to decrease and the listener began to raise the playback gain. By the end of the lower-loudness section, at point C, this listener had increased the playback gain by approximately 10 dB, relative to the opening section at point A. Loudness for the middle section had dropped 15 LU from the opening section.

In the third section of the test, loudness of the C2 sample had returned to the nominal -23 LUFS level. This listener can be seen turning down the gain by approximately 7.5 dB, from point C to point E. The loudness level equaled the first section, and this listener had returned the playback gain to within 2.5 dB of its initial setting at point A.

Analysis of Listener Test Data

The gain changes represented by the dashed lines in the upper part of Figure 8 when compared to the loudness changes indicated by the dashed lines in the lower part of the figure, comprise the process of measuring listener preferences for loudness change. Participant's behavior was analyzed separately for loudspeaker listening and earbuds. Preliminary analyses showed that there were both age and gender differences, so these demographic characteristics were considered during analyses. Four analyses were conducted to determine the following behavior:

- How people adjusted their volume at the start of the listening experience to set their nominal listening level (start to point A)

- How people adjusted their volume in response to audio becoming more quiet in the middle (point A to point C)
- How people adjusted their volume in response to audio become louder at the end (point C to point E)
- The difference between their original nominal listening level and the end level after mid-sample adjustments had been made (point A to point E).

The concept of *loudness compensation* is introduced, which represents how much listeners offset the change in loudness they heard. Referring to the mean loudness points in Figure 8, the change in loudness from point A to point C is:

$$(L_C - L_A) = (-38 - (-23)) = -15\ LU.$$

Similarly, the change in listeners' playback gain at the same inflection points is:

$$(G_C - G_A) = (15 - 5) = 10\ dB.$$

Using the actual values from the figure, the loudness compensation between point A and point C combines these values as:

$$(L_C - L_A) + (G_C - G_A) = -15 + 10 = -5\ dB$$

This may be read as: "the listener's loudness compensation was 5 dB less than the decrease in loudness" (of 15 LU). If a listener fully compensated for the change in loudness, as measured by the ITU meter, the result would be 0 dB. The loudness compensation value is used in the following discussion to discuss responses to listening conditions and listener groups.

Model for Loudness Management Based On Listener Tests

The listener tests reveal some complex relationships between loudness change and the desired "make up" in gain to please listeners. However, some generalizations emerge from the data upon which a model can be based, as shown in Figures 9 and 10. In these charts, the listeners' desired changes in playback gain with loudspeakers are plotted in relation to the change in loudness, measured per ITU, from inflection points A to C.

The columns of points indicate listener responses in common with various audio samples. The four noise environments are divided into two charts for clarity: office and roadway representing the lowest and highest noise environments, respectively, while the restaurant and in-vehicle noise are in between.

Figure 11 provides the listener's change in gain versus change in loudness for earbuds. As explained earlier, the two highest noise environments were excluded as a precaution against poor data.

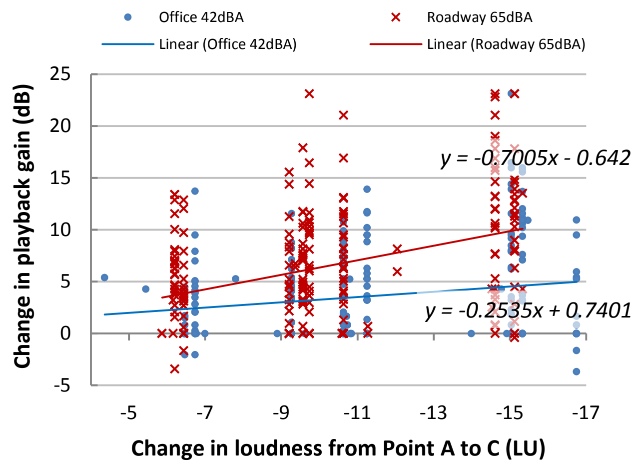

Figure 9 - Change in loudness for all listeners and all genres with office and roadway noise environments using loudspeakers

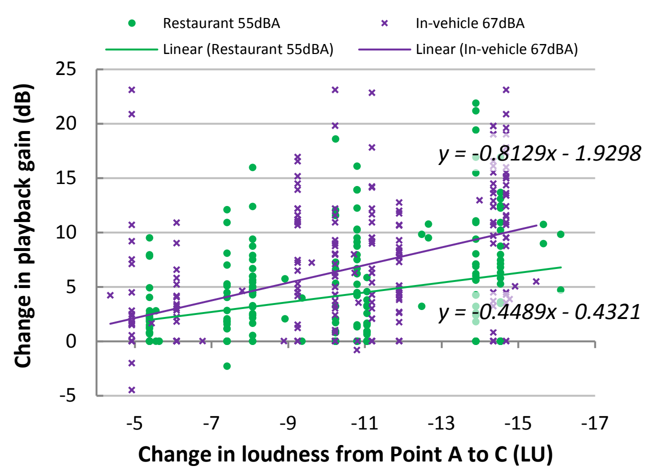

Figure 10 - Change in loudness for all listeners and all genres with restaurant and in-vehicle noise environments using loudspeakers

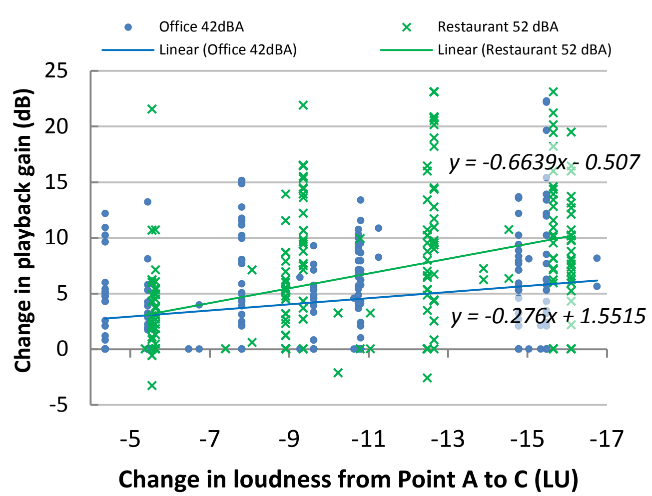

Figure 11 - Change in loudness for all listener and all genres with office and restaurant noise environments using earbuds

For each figure, there are a number of points on the 0 dB line, indicating that some listeners decided that their initial gain setting was fine across the low-loudness trough. While it could be argued that no loudness management is needed for these listeners, their scores were considered in the following analysis. (There are also a small number of gain scores in the negative range, meaning that the listener preferred to turn down their playback gain, in opposition to the reduced loudness. Their data were too infrequent to affect the overall results.)

For each noise environment, a linear regression was performed on the data: the slope and intercept are shown adjacent to each trend line. The upward slope to the right indicates that as the loudness change increases, the listener means for playback gain also increase.

As shown in Figures 9 to 11, as the noise environment level increases, the slopes increase. Thus, loudness management, or the desired compensation of loudness variation, is based on two elements:

(1) Measuring the change in loudness by an appropriate system such as specified by the ITU, and

(2) Active sensing the level of ambient noise in the listening environment by microphone.

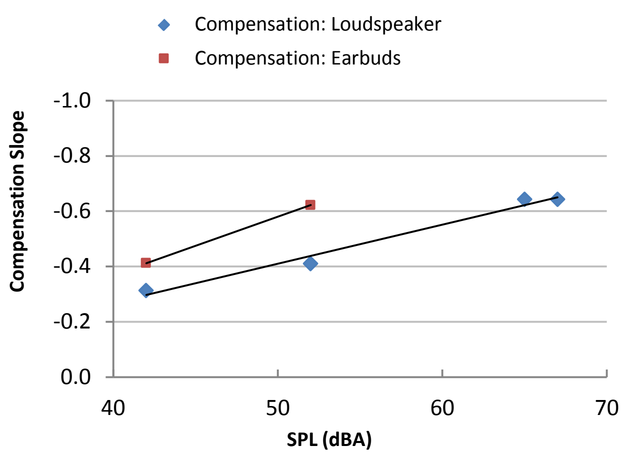

Figure 12 - Chart illustrating the compensation slopes as a function of environmental noise level

To convert the results into a form that is practical for loudness management, the slopes must be considered with the noise environments. The intercepts complicate the reduction, but fortunately are relatively small in magnitude (+1.5 to -0.6 dB). The regression can be set to intercept with 0, a noise-free condition for which no loudness management is assumed necessary. The slopes with zero intercept are plotted as a group in Figure 12.

To apply the result, consider an ambient noise level of 50 dBA. The compensation slope for loudspeakers is approximately -0.40, meaning that if the audio program loudness changed by -10 LU, the compensation would be: -10 x -0.40 = +4 dB, an increase in gain of 4 dB under these circumstances. At an ambient noise level of 65 dBA, the compensation for a change of -10 LU would be: -10 x -0.62 = +6.2 dB. Being a simple algorithm, this automatic process could be added to the DSP volume control in a range of consumer audio playback equipment, including portable devices and automobiles.

Conclusions

While the data represent the central tendency of listeners, there are listeners who desire more or less control than the mean gain. It is recognized that no "one size fits all" solution exists, but there is a consistent and usable trend. Depending on the listener's tastes, an "activity" control could be included to change the slope, so that more or less compensation could be applied to the changes in audio loudness. As the data indicate that speech needs more compensation that music, the compensation could include a switch for either program type.

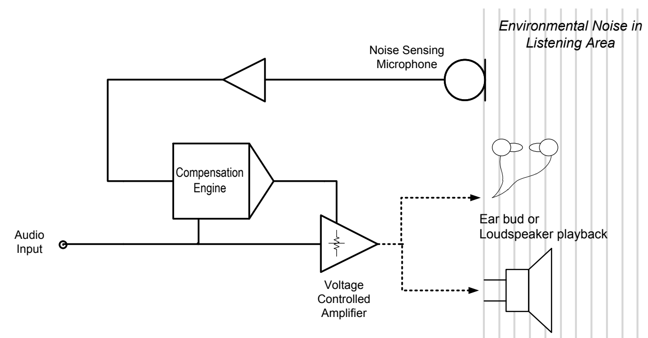

Figure 13 - Diagram showing how the compensation algorithm could be implemented in a playback device

Figure illustrates the basic components of the loudness compensation system within the playback device, such as a smartphone or home stereo. The noise sensing microphone picks up the noise field near the listener and presents the noise signal to the Compensation Engine, which determines the sound pressure level and applies a slope factor to produce a gain control signal for the voltage controlled amplifier. For loudspeaker playback, a linkage from the input audio is shown, which could freeze the environmental noise measurement once program audio begins, which could be misinterpreted as environmental noise. The microphone is optional, however, if a control were available so that the listener could set the average conditions, such as "quiet office" or "noisy home" to effect the appropriate loudness compensation.

The results indicate that the adjustment is not a dB-for-dB compensation to the change in loudness. Listeners do not require a high "compression ratio" (to put it in terms of audio processor design) for control of most program audio. Apparently, they do not wish a constant "signal-to-noise ratio" in listening to program material with various levels of ambient noise. (Another way to look at it may be that they have limits on how loud they wish the program material to be, and they consciously or subconsciously anticipate that program material could become louder at a later time.)

It should be noted that most program material delivered to consumers has undergone dynamic processing, possibly at more than one stage from studio production to mass-distribution. The material selected for listener testing was chosen to have loudness variation, however, much of the audio content sold online, mass-produced on CD or delivered by radio and television has minimal loudness range. In these cases, a loudness management system of the type described herein would likely set playback gain relative to the ambient noise and "sit out" further changes until the program content varied.

This study looked at decreases in loudness, not unexpected increases due to mismatched levels in audio content. As discussed at the outset, this is another issue that deserves attention. The solution for the management of inconsistent levels across different content could easily be integrated into the described system.

This study suggests further study of loudness measures. For example, the ITU loudness meter provides a "Momentary" measurement with an integration time of 0.3 s, in addition to the Short Term measurement of 3 seconds used in this study. The correlation of faster Momentary values to listener data may reveal more about the speed at which listeners respond to changes in loudness – something beyond the scope of this study. The ITU standard's Loudness Range (LRA) is a relatively long-term measure, which could provide interesting results in comparison to listener responses. The CBS Loudness Meter is a different measure of loudness that could be compared in listener response to the ITU meter [7]. This study is an initial investigation into the loudness preferences of average listeners. Further research would add to the approach and optimize it for specific playback systems.

Acknowledgements

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