Ayla Favati
Professor: Paul Geluso
Thesis 1: MPATE-GE 2601
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Thesis Draft:
Loudness Metering Techniques in Mastering Immersive Music

Ayla Favati
Thesis Advisor: Dr. Agnieszka Roginska
New York University
Thesis 1: MPATE-GE 2602
December 4, 2020

Ayla Favati
Professor: Paul Geluso
Thesis 1: MPATE-GE 2601
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1. Introduction

Listeners often have a limited understanding of the music production process and, even within the industry, mastering engineering is often regarded as a mysterious art. As it is the final step in music production, mastering engineering is the final quality check and outfitting for distribution and consumption. Mastering engineers not only choose subtle effects to enhance the entire teams' intention but also review audio carefully for errors and adjust levels accurately. Finally, they must ensure that each file format produced for the product's various destinations will uphold the integrity of each decision made throughout the production process.

These requirements prove increasingly challenging with the addition of various streaming services, loudspeaker reproduction formats, and loudness measurement techniques. With the increasing popularity of immersive audio formats, mastering engineers must account for changes in loudness across each potential reproduction format. Especially with the introduction of object-based audio, reproduction formats can range from headphone to 22.2 loudspeaker reproduction. To account for each format that may be employed, engineers must utilize an accurate loudness metering technique and have the ability to measure exactly how each format will affect the characteristics and loudness of the audio. As in the broadcasting industry, it is essential to have accurate loudness measurement techniques to ensure reliable playback and this must include all parameters associated with loudness.

While loudness metering softwares and techniques exist, metering software has only recently included the characteristics of perceived loudness of audio. The ITU-R BS. 1770 algorithm recommendation takes perceived loudness into account and has widely been utilized within the broadcast and cinema industry. More recently, the recommendation has also been adjusted to include multichannel formats. As immersive audio becomes more popular within the music industry and loudness normalization continues to affect audio streaming, there must be a way for mastering engineers to anticipate changes indirectly caused by changes in loudspeaker format and through the use of loudness normalization. ITU-R BS.1770 is the standard for the broadcast industry and is widely used in the music industry, but its advantages and limitations must be examined more thoroughly.

This thesis aims to address the unpredictability of loudness perception and measurement across immersive audio reproduction formats. By using this metering technique already available and in use in the audio industry and taking into account dynamic range, this work will provide an insight into using loudness units full scale or LUFS to predict perceived loudness in mastering music for immersive audio formats. To determine its success, the ITU-R BS. 1770 and LUFS measurement characteristics must be covered in detail and its practical use within various formats tested objectively and subjectively. That means a comprehensive study of its history, study, and evolutionary capabilities must be presented, an experiment designed and executed, and results carefully considered. The results of this work could prove useful for mastering engineers and within the entire music industry, applied in all parts of the production process, and used to understand production within any multichannel audio reproduction format.

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2. Literature Review

Within this literature review, considerations in mastering engineering and the importance of loudness and loudness perception will be discussed in detail. The history behind the loudness wars and the effects of loudness normalization will be reviewed. Loudness metering methods will be compared by considering the advantages and limitations of each. The introduction of immersive loudspeaker formats and its influence on mastering engineering will also be considered. Lastly, an overview of immersive audio theory and implementation in music technology and production will be introduced.

The Last Stop in Music Production

Mastering Engineering

Mastering is the final part of the music production process. Other than subtle elements such as preserving artist intent and polishing the mix to create a cohesive and binding atmosphere (Hepworth-Sawyer 32), mastering also consists of more overt tasks, such as formatting for distribution and loudness or level adjustment. The mastering engineer's job is largely to prepare the project, preserving its unique characteristics, for enjoyment through distribution across diverse systems (Hepworth-Sawyer 121).

In order to prepare projects for diverse systems, mastering engineers listen to reference material and compare with projects in several different formats including headphones, stereo, and even multiple loudspeaker arrays. This is partially to compare stylistic and creative choices of their peers, but also to compare and choose accurate loudness in the context of the genre (Gebre).

The importance of loudness in mastering engineering is exemplified beginning with vinyl by the 'loudness wars.' Because greater loudness is often perceived as higher quality, masters became louder and louder to capitalize on the listener's perception and cause their music to stand out among the rest. Consequently, the average loudness of music climbed higher (Gebre) causing a dramatic difference in loudness and dynamic range between music produced in the 1950's until today. Therefore, in the age of digital streaming, sharp spikes in loudness due to the average difference between tracks are audible, as listeners can play music on demand from a wide range of decades and genres. Loudness normalization was developed for digital streaming services to combat this issue, using an average loudness detection which consequently and automatically alters the loudness of the song (Gebre) effectively overwriting the mastering engineer's loudness choices.

Loudness Perception

Why is loudness a concern?

Loudness and ear sensitivity have long been a subject of study in the effort to understand psychoacoustics and its importance in music. What is "correct" loudness and how does it change according to taste or media or genre? Most mastering engineers level by ear, as it is regarded as the more accurate tool (Gebre). However, ideal loudness guidelines by genre have been recorded by Bob Katz at RMS (room mean square) level: "-12dBFS (decibels relative to full scale) RMS for broadcast/radio, -14dBFS RMS for pop/rock/country, and -20dBFS RMS for film/classical/ hi-fidelity, with the peaks of each at -0.3 dBFS (Gebre)." Because streaming companies must use loudness normalization to avoiding undesired jumps in audio loudness, mastering engineers must account for automatic audio reduction as well as genre-specific level standardizations in an effort to prevent the audio content from being "squashed" and losing the creative intent (Rumsey & McCormick 302).

Loudness is also characterized as "the primary distance cue used by a listener," and is, consequently, a major consideration discussed in the immersive or spatial audio field (Begault 2000). Equal loudness contours, like the Fletcher-Munson curves or Robinson and Dadson curves, addressed ear sensitivity across the audible frequency range (Rumsey & McCormick 31) and demonstrated that human hearing is more sensitive in the middle range, meaning that changes in this frequency range will affect perceived loudness (Gebre). However, the shape of the curves depends on various test data considerations. For example, the Fletcher-Munson curves were obtained in an anechoic space using pure tones whereas Robinson and Dadson curves were developed using pure tones in a free field listening environment (Holman & Kapmann 1). Yet, both of these studies are not applicable to bands of noise or other methods of reproduction (Holman & Kapmann 2). This is an important distinction, as characteristics of sound greatly affect the perceived loudness. Broadband sounds are perceived as louder than narrow-band sounds, for example, and distorted sounds are perceived as louder in comparison to less distorted sounds (Rumsey & McCormick 32-33).

Loudness descriptors allow for specific characteristics of a signal to be categorized and utilized to then understand and measure perceptual loudness. True peak-level has been used to loudness normalize tracks in music streaming, sometimes compromising or influencing dynamic range (Skovenborg & Lund 2). Though there are other factors that affect loudness perception, however, and these must also be observed and measured. A center-of-gravity must be considered, as there may be dramatically low levels in a signal which also has a high peak-level, so the overall loudness of a track should be measured (Skovenborg & Lund 2). Consistency and density should also be considered which measure loudness over time macroscopically and microscopically, respectively (Skovenborg & Lund 3). These comprise methods by which to measure perceived level across genres and production periods, as genre-based and period-based level standardizations vary greatly, providing challenges in the task of implementing loudness normalization to a vast library of music.

Loudness normalization has been implemented most stringently within the broadcast audio sector, so that standard algorithms have been developed to accurately measure loudness and standards are published and enforced (Norcross, Nanda, & Cohen, 1). There are several methods for measuring loudness and each have their advantages and disadvantages. Peak loudness measurement has been used in loudness normalization by digital streaming companies though it is extremely sensitive to unnoticeable peaks and it does not account for changes over time and frequency range (De Man 2). RMS or root mean square and moving average is a more comprehensive loudness measurement by including perceived loudness as a measurement over time with a weighted average, though it still does not account for frequency changes affecting perceived loudness (De Man 2). Stevens' loudness also lacks frequency sensitivity then Vickers' loudness begins to develop weighting strategies additionally addressing frequency sensitivity. Finally, ITU-R BS.1770 or EBU R-128 using loudness units full scale (LUFS) allow frequency-weighting, gated signal measurement while preserving computing efficiency, as opposed to some more advanced techniques that are less accurate and more computationally exhausting (De Man 2). This algorithm has, indeed, become the standard by which to measure loudness for broadcast and music programs alike (Norcross, Nanda, & Cohen, 1).

Loudness in multichannel formats has also become a topic of consideration. ITU-R BS.1770/EBU R-128, for example, was revised to include an "arbitrary number of channels, enabling the measurement of new channel-based immersive audio formats (Norcross, Pribadi, and Nanda 1)." Though this has been used to address other audio formats, this metering technique was developed specifically for immersive broadcast audio programs (Norcross, Nanda, & Cohen 1) and is still a relatively new metering technique for immersive, object-based music. Norcross, Pribadi, and Nanda, indeed, address loudness in object-based audio, stating that measured loudness of the 5.1 surround render "enables consistent loudness for a range of configurations…" including "7.1.3 channel layout down to a stereo layout (Norcross, Nanda, & Cohen 3). However, as Rumsey & McCormick point out, judgements theoretically proven correct may not be equally as correct subjectively (Rumsey & McCormick 490).

Music streaming companies such as iTunes have included a feature to notate loudness as suggested by ITU-R BS.1770/EBU R-128 standards, measured in LUFS (loudness units related to full scale), using a song's metadata. Using metadata instead of actual signal data allows the preservation of the audio while allowing for various measurement factors within a single audio stream (Grimm 2) and, due to these advantages, loudness metadata is increasingly becoming standard practice in mastering engineering (Rumsey &McCormick).

Additionally, because the broadcasting recommendation ITU-R BS.1770/EBU R-128 was developed for the broadcasting field, its experiment controls were designed for loudness in the broadcasting sector. Preferred listening levels, however, vary between program type. ITU-R BS.1770/EBU R-128 was tested and developed using a listening level control of 60dBA SPL (Ronan, Ward, & Sazdov 2). However, 60db SPL is the lowest listening level reported in the Audio Engineering Society's analytical listening tests from 2006-2016 and it is suggested to use the mean listening level of 72.67db SPL in designing perceptual experiments (Ronan, Ward, & Sazdov 3).

Immersive/Spatial Audio

While stereo reproduction is still the industry standard, wider loudspeaker arrays can provide a more natural listening experience. Because listeners are entrenched in an immersive soundscape in everyday life, it could be an advantage to utilize immersive audio formats in music (Kendall, 23). This effort will provide subjectively pleasing qualities like spaciousness and depth. Some argue that this is the main objective in creating an aesthetically pleasing mix: to give “the impression of spaciousness in the sound field” and, therefore, “create believable illusions” (Rumsey & McCormick 491). Through this effort, rendering software has been developed which allow the freedom to transfer a mix to various loudspeaker formats from binaural reproduction over headphones to 22.2 loudspeaker formats and more.

Object-based Audio

The object-based audio format allows rendering of sound objects, which are audio elements that may be assigned to specific locations in a 3D space around the listener through metadata and manipulated individually from the group in a digital audio workstation (Roginska & Geluso 244). Once processed and organized, object-based audio has the advantage of preserving artistic intent and upholding location information across various loudspeaker formats such as the 5.1 and 7.1 surround sound or a 22.2 loudspeaker system (Roginska & Geluso 244). This differs from popular channel-based audio systems that are specific to the loudspeaker format in which they were produced and, therefore, must be played back in that format (Rumsey 987).

A major consideration in mastering audio in an object-based environment is how to develop new workflows since object-based format requires different techniques and strategies than those developed for channel-based audio (Hestermann 1). Mastering object-based audio might involved a technique similar to mastering stems, which had been widely unpopular in the past (Hepworth-Sawyer). Mastering for multiple loudspeaker systems at once, as Atmos is able to decode to multiple loudspeaker systems, is a particular challenge in mastering for object-based audio. Newer multichannel formats, such as the Hamasaki 22.2 format, are difficult and expensive to install (Hestermann 2). However, mastering engineering techniques summarized previously often reiterate that the best test across formats is to listen within each reproduction format (Hepworth-Sawyer 32), which would require installation of intended loudspeaker arrays.

Nicolas Tsingos illustrates difficulties in rendering object-based audio, namely in loudness metering: “1) extending current standards to meter and correct loudness of object-based content, and 2) ensuring that playback of such content has consistent loudness regardless of the rendering configuration (Roginska & Geluso 269).” In the field of cinema, for example, it is standard practice to increase the levels of surround channels by 3dB to account for acoustic environments in certain movie theaters (Rumsey & McCormick 537).

Binaural Audio

Binaural refers to two-channel audio heard through ears that is encoded with naturally or mathematically derived spatial cues (Roginska & Geluso 88). Through the psychoacoustic study of Head-Related-Transfer-Functions binaural cues may now be calculated through signal processing (Roginska & Geluso 88), allowing filtering of pre-recorded tracks to create binaural, or virtual spatial audio (Begault 4). However, HRTFs are slightly different for each individual (Roginska & Geluso 90) and are consequently difficult to calculate for a wide range of listeners, therefore also difficult, but not impossible, to initiate over headphones (Rumsey &McCormick

500). Additionally, while the ITU-R BS.1770/EBU R-128 algorithm addresses the effects of head shadowing in loudspeakers through a high frequency shelf, issues were found when combining this filter with binaural room impulse response (BRIR) processing in binaural reproduction over headphones (Ronan, Ward, & Sazdov 4).

5.1 Loudspeaker Array

Though this loudspeaker format has obtained wide success, it ought to be discussed as it may not be ideal for all of its applications (Roginska & Geluso 188). Also known as 3-2 Stereo, this array allows for stereo effects, called left and right, as well as an additional front image speaker, called center, and two surround speakers, called left and right surround (Rumsey & McCormick 532). 5.1 allows for use of the low-frequency enhancement (LFE) component but was not intended for 3D audio reproduction and the left and right surround channels could create a large sound hole behind the listener (Rumsey & McCormick 537).

7.1.4 Loudspeaker Array

The 5.1 loudspeaker array also does not provide height channels, a significant addition in the production of more accurate 3D audio. Interestingly, it also provides a basis for many loudspeaker reproduction formats like 7.1 or 7.1.4 in which additional surround and height loudspeakers are added to account for the sound hole behind the listener. In the 7.1.4 configuration, for example, the 5.1 format is embellished with two additional surround channels as well as 4 height channels to aid in more accurate reproduction of 3D audio. This could provide a listener with increased envelopment, dimension, and spaciousness (Roginska & Geluso 216).

Dolby Atmos: Advantages and Strategies

There are some advantages to using Dolby Atmos to render object-based audio. First, Dolby Atmos is compatible with ProTools Ultimate, an industry standard for mixing and mastering software. Some of the channel-based audio mastering strategies curated in the past can also transfer to immersive audio, as Dolby Atmos allows for a hybrid channel- and object-based audio format through the integrate use of channel-based audio signal beds (Rumsey & McCormick 552). Dolby Atmos also has the option to automatically render to binaural audio, allowing for easy downmix to two-channel audio and therefore greater accessibility to immersive audio through stereo headphone distribution, as well as the ability to render to many loudspeaker systems due to the use of object-based audio (Dolby Laboratories).

3. General Method

3.1 Completed Work

3.1.1 Approaches and Techniques

The approaches and techniques used in this experiment include both creative and technical consideration. The objective of the experiment aims to understand whether or not LUFS metering may be used to accurately master binaural audio and whether or not the method of binaural mixing contributes to the success of the metering.

Creatively, the goal is to gain experience in mastering samples with various dynamic ranges and for binaural playback. This also includes employing the techniques for binaural mixing in the "objects-in-space" style, utilizing the entire 3D sound field, and in a more traditional, stereo-field style, focusing on the stereo field but with the 3D sound field being utilized for reverberation as if in an orchestral-style concert hall.

The techniques employed in this experiment include, in a broad sense, mastering in a digital audio workstation, using a binaural rendering plugin, and with the intention of playback over headphones. This also requires the ability to measure samples and clips with a LUFS metering plugin and meticulously documenting and organizing each exported sample. I have previously used a binaural sample to level with the obtained LUFS metering plugin. The meter on the sample track seems to respond to changes in level, supporting the continuation of this experiment.

Materials required are as follows: computer, digital audio workstation, binaural rendering plugin, reliable headphones, and computer and headphones at minimum for each of the participants.

3.1.2 Experimental Design

With the assistance of Professor Schuyler Quackenbush, the design and survey being used in the experiment has gone through some iteration. That which will be used in the experiment is an A/B comparison survey described in section 4.2 and 4.3 of this methods document.

The questionnaire has been designed and it collects relevant information to be collected voluntarily from each participant. This includes information that may contribute to the results and data of the experiment. The consent form in which contact information and privacy information is provided will be included in this section as well (see Appendix B). The IRB application is submitted and is in the process of approval.

3.2 Planned Work

3.2.1 Stimuli

The stimuli, at the moment of this document's submission, is partially chosen and will be mixed in the binaural sound field and mastered at various comfortable LUFS measurements in the range typical of its genre. Each sample of 10-15 seconds in length will be rigorously documented for input in the survey and experiment.

3.2.2 Pilot Test

The pilot test must be conducted once all stimuli is chosen, edited, and input into the survey drafted by Professor Quackenbush. The pilot test will require that three participants complete the survey and give feedback. The experiment will go through revision before final distribution.

3.2.3 Experiment and Analysis

Once final distribution has started, the experiment's data will be collected and analyzed according to the number of participants. Number of participants must be between 25 and 40 and there must be a range of intermediate and expert listeners at the minimum. If there are enough students with little music or audio background, there may be a secondary analysis to determine how listening experience affects accuracy. Experiment distribution will be mostly in the Music Technology program at New York University Steinhardt. However, there will be distribution among Steinhardt and Tandon students in other programs.

3.3 Evaluation/Experiment Method

3.3.1 Participants

An as yet undetermined amount of students from different majors at New York University will participate in this experiment. The participants will be remote and kept anonymous and, therefore, will not list identifying characteristics like their ethnicity, grade, or major. However, the experiment will be distributed to a wide student base, making this sample representative of university students of various age and background. Their mean age is as yet undetermined as well. Participants are not offered compensation or school-related benefits of any kind. Participation in the experiment will be completely voluntary and participants may cancel participation at any point in the experiment. Any results from participants that fail to respond to all items will not be included in the analyses.

3.3.2 Materials

The stimuli will be three audio clips of ten to fifteen seconds in length at various dynamic ranges and genres. Each clip will be mixed as "objects in space" at various points around the binaural sound field or in a more "stereo-like" field. Each clip will be mastered at various, comfortable-listening LUFS measurements unknown to the participant. The experiment is split into three trials, one trial for each clip. Each trial will include ten randomized A/B comparisons in which the clip is at two different, but comfortable LUFS measurements.

Responses will be recorded immediately after each A/B comparison. The participant may listen to each clip once all the way through and will then use a checkbox to indicate which is louder.

Participants responded to a questionnaire (see Appendix A) inquiring about hearing ability to account for any hearing discrepancies between participants. They will be asked about music and audio education as well as hearing ability during the time of the experiment. They will also be asked to test their playback system during a test comparison to account for any system errors.

3.4 Procedure

The participants will be provided with a link that will allow for their results to be collected remotely and on the participants' own time. No person will administer the survey nor observe the participant taking the survey in any way. Consent will be obtained remotely and anonymously as a part of the survey. Because the experiment will be remote, participants will self-report listening environment and playback system and they will only receive instruction in writing. The full survey will be split into two major sections: the questionnaire including sample experiment and the experiment itself.

The questionnaire (See Appendix A) will allow for participants to self-report their respective payback system used in the experiment, including audio interface, headphone make and model, amp, listening environment etc. They will also be asked about their hearing, whether or not they are able to hear with both ears and whether or not they had a cold or sinus issues at the time of completion of the experiment. They will also be asked how much experience they have in audio engineering and whether or not they have experience using loudness metering tools. The questionnaire will also include the consent form (See Appendix B) which will remind the participant that the experiment is completely anonymous and that they may exit and remove their data from analysis at any time. They will have an unlimited amount of time to complete the experiment and will have primary contact information included within the consent form so that they may ask questions at any time before, during, or after the experiment.

Included in this portion will be the sample comparison, in which the participant will be instructed to adjust their loudness level to comfort. They will then be instructed to fill out a sample A/B comparison similar to the experiment, in which they will choose which sample is louder and asked if both channels in their headphones work properly. Once the experiment section has started, the participants will be asked to keep their playback level constant through to the end. This will account for any issues with the playback system or chosen playback level which could skew experiment results.

The second section, the experiment itself, will be A/B comparisons asking participants to choose which sample is louder by clicking a corresponding checkbox. Each of the three trials will have ten A/B comparisons, which will amount to about five minutes in length each and fifteen minutes total. Composers and performers of each sample will be credited after the last section of the experiment.

A pilot test will be conducted before distribution of the full experiment. The pilot test will require three participants to complete the questionnaire and experiment. Each of the three participants will provide feedback during and after each section. They will be asked what is clear or unclear and if they had any issues navigating or completing the survey. Then, any changes necessary will be made and the experiment will be distributed.

3.5 Analysis Approach

Once data is collected, it will be determined whether or not there are enough subjects with musical and non-musical backgrounds to analyze data with a repeated-measures ANOVA. If so, the data will be analyzed with students of musical and non-musical background between the trials and conditions. The rejection level for all analyses will be set to $p=0.05$. JASP will be used to analyze data.

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Appendix A

Please use headphones for this experiment.

1. How many years have you studied music?
2. How many years have you studied audio engineering?
3. How many years have you studied mastering engineering?
4. Have you used loudness metering tools?
5. If so, which tools? e.g. VU Meters, RMS, LUFS
6. Do you have the ability to hear equally with both ears?
7. Do you report having a cold, sinus issues, or ear infection(s) at the time of this listening test?
8. Please describe your listening environment. e.g. quiet, noisy, outside, inside etc
9. What equipment are you using to take this test? Please include headphone make/model, any amps, audio interface, other parts of your playback system etc.

For the test comparison, you will answer a follow up question to determine the accuracy of your playback system. Please adjust your levels during this test comparison only. Once you have adjusted your levels to comfort during the test question, please keep the levels constant for the remainder of the experiment.

10. Are you able to hear sound in both headphone channels?

Appendix B

Consent Form for IRB-FY2021-4916

You have been invited to take part in a research study to learn more about loudness metering techniques in mastering immersive audio. This study will be conducted by Ayla Favati, STEINHARDT - Music & Performing Arts Professions, Steinhardt School of Culture, Education, and Human Development, New York University, as a part of her Master's Thesis. Her faculty sponsor is Professor Dr. Agnieszka Roginska, Department of STEINHARDT - Music & Performing Arts Professions, Steinhardt School of Culture, Education, and Human Development, New York University.

If you agree to be in this study, you will be asked to do the following:

- Complete a short questionnaire about the equipment you're listening on
- Complete a short exercise to demonstrate how to complete the survey
- Listen to short samples of music
- Evaluate the loudness in comparison to other sample(s)

Participation in this study will involve 5 minutes to complete the questionnaire and practice example, 30 minutes to complete the survey. There are no known risks associated with your participation in this research beyond those of everyday life.

Although you will receive no direct benefits, this research may help the investigator understand loudness metering techniques in mastering immersive audio..

Confidentiality of your research records will be strictly maintained by not collecting any identifying information and in submitting forms without identifiers, so that all answers shall remain anonymous. Your information from this study will not be used for future research.

Participation in this study is voluntary. You may refuse to participate or withdraw at any time without penalty. For interviews, questionnaires, or surveys, you have the right to skip or not answer any questions you prefer not to answer. Nonparticipation or withdrawal will not affect your services or academic standing at NYU. You may withdraw at any time, anonymously and without consequence.

If there is anything about the study or your participation that is unclear or that you do not understand, if you have questions or wish to report a research-related problem, you may contact Ayla Favati at af3791@nyu.edu, or the faculty sponsor, Dr. Agnieszka Roginska at (212) 998-5141, roginska@nyu.edu, 35 West 4th Street Room 236, New York, NY 10012 USA.

For questions about your rights as a research participant, you may contact the University Committee on Activities Involving Human Subjects (UCAIHS), New York University, 665 Broadway, Suite 804, New York, New York, 10012, at ask.humansubjects@nyu.edu or (212) 998-4808. Please reference the study # (IRB-FY2021-4916) when contacting the IRB (UCAIHS).

You have received a copy of this consent document to keep.

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4. Categorized Annotated Bibliography

Mastering Engineering

Gebre, W. (2013). *Complete Audio Mastering: Practical Techniques*. McGraw Hill.

This is a comprehensive guide to mastering from gear, to technique, to beginning a mastering business.

Hepworth-Sawyer, R. & Hodgson, J. (Eds.). (2019). *Audio Mastering: The Artists Discussions from Pro-Production to Mastering.* Focal Press.

This is essentially a transcript of conversations with prolific mastering engineers, discussing their career start in the field and tips and tricks they have learned over time including preferred signal chain, digital versus analog preference, editing, and loudness wars.

Hestermann, S., Seideneck, M., & Sladeczek, C. (August 2018). *An Approach for Mastering Audio Objects*. Paper presented at the Audio Engineering Society Conference on Spatial Reproduction. Tokyo, Japan.

This paper presents some techniques for mastering audio objects in an object-based immersive audio format.

Ramsey, F. & McCormick, T. (2014). *Sound and Recording: Applications & Theory* (7th ed.). Focal Press.

This is an essential book for beginners interested in music technology. The book covers all aspects of producing music from recording to mastering as well as innovation in the field as of the book's publishing date.

Loudness Perception and Management

De Man, B. (October 2018). *Evaluation of implementations of the EBU R128 loudness measurement*. Paper presented at the Audio Engineering Society Convention. New York, NY.

EBU R128 and other loudness measurement techniques are examined and evaluated in terms of accuracy and functionality.

Francombe, J., Brookes, T., Mason, R., & Melchior, F. (October-November 2015). *Loudness Matching Multichannel Audio Programmer Material with Listeners and Predictive Models*. Paper presented at the Audio Engineering Society Convention. New York, NY.

This paper summarizes a listening test conducted using multiple audio reproduction methods and listening materials to determine whether loudness models could be applied across listening formats.

Grimm, E. (October 2019). *Analyzing Loudness Aspects of 4.2 million Music Albums in Search of an Optimal Loudness Target for Music Streaming*. Paper presented at the Audio Engineering Society Convention. New York, NY.

Using LUFS measurements, Tidal's music collection was analyzed for loudness aspects and normalization techniques were evaluated.

Holman, T., & Kapmann, F. (November 1977). *Loudness Compensation: Use and Abuse*. Paper presented at the Audio Engineering Society Convention. New York, NY.

This paper proves well-known, standard loudness measurements and controls of the time inappropriate for realistic spaces and presents reasonable and reliable alternatives in proper loudness control.

Norcross, S., Pribadi, M., & Nanda, S. (May 2017). *Practical Loudness Measurement and Management for Immersive Audio*. E-Brief submitted to the Audio Engineering Society Convention. Berlin, Germany.

This brief discusses loudness within object-based audio formats. Object-based audio delivers to multiple playback formats which presents the issue of loudness variation. The brief discusses measurement and management through the lens of broadcasting.

Norcross, S., Sachin, N., & Cohen, Z. (June 2016). *ITU-R BS.1770 based Loudness for Immersive Audio*. Paper presented at the Audio Engineering Society Convention. Paris, France.

This convention paper, written my Dolby Laboratories' engineers, addresses the coding revisions necessary to adopt ITU-R BS.1770 loudness measuring for rendering dynamic, object-based audio.

Ronan, M., Ward, N., & Sazdov, R. (June 2017). *Considerations when calibrating program material stimuli using LUFS*. E-Brief submitted to the Audio Engineering Society Convention. Paris, France.

This e-brief examines the calibration techniques important in the use of LUFS loudness perception measurement. The conclusions address the importance of considering experimental variables and present an absolute SPL level suggestion.

Skovenborg, E., & Lund, T. (October 2008). *Loudness Descriptors to Characterize Programs and Music Tracks*. Paper presented at the Audio Engineering Society Convention. San Francisco, CA.

Loudness properties of audio are examined and categorized specifically in reference to the ITU-R BS. 1770 loudness level. This was in the effort to measure and describe loudness perception across genres to diagnose problems and aid in streamlining delivery.

Immersive Audio Theory & Related Considerations

Begault, D. (April 2000). 3-D Sound for Virtual Reality and Multimedia. *NASA Center for Aerospace*. NASA Center for AeroSpace Information.

This is a comprehensive overview of spatial hearing, sound, reproduction, and signal processing including practical applications.

Kendall, G. (Winter 1995). A 3-D Sound Primer: Directional Hearing and Stereo Reproduction *Computer Music Journal*, 19 (4), 23-46. p://www.jstor.org/stable/3680989

The author provides a broad summary of 3-D audio technology by breaking down the science of directional hearing and reproduction techniques. This article is particularly useful as an introduction to 3-D audio and for learning important language used within the field.

Roginska, A., & Geluso, P. (2018). *Immersive Sound The Art and Science of Binaural and Multi-Channel Audio*. New York, NY: Routledge Taylor & Francis Group.

The authors and contributors provide the a comprehensive and current guide to immersive audio and its binaural and multi-channel reproduction. Through historical, psychological, physiological, technological, acoustical, and mathematical means, the use of immersive audio and its potential applications are explored in this book.

Rumsey, F. (November 2018). Spatial Audio: Channels, objects, or ambisonics? *Journal of the AES*, 66 (11), 987-992. http://www.aes.org/e-lib/browse.cfm?elib=19873

This article explored the positive attributes and trade offs associated with various immersive audio workflows. It provides more in-depth considerations affecting the evolution of immersive audio reproduction technology.