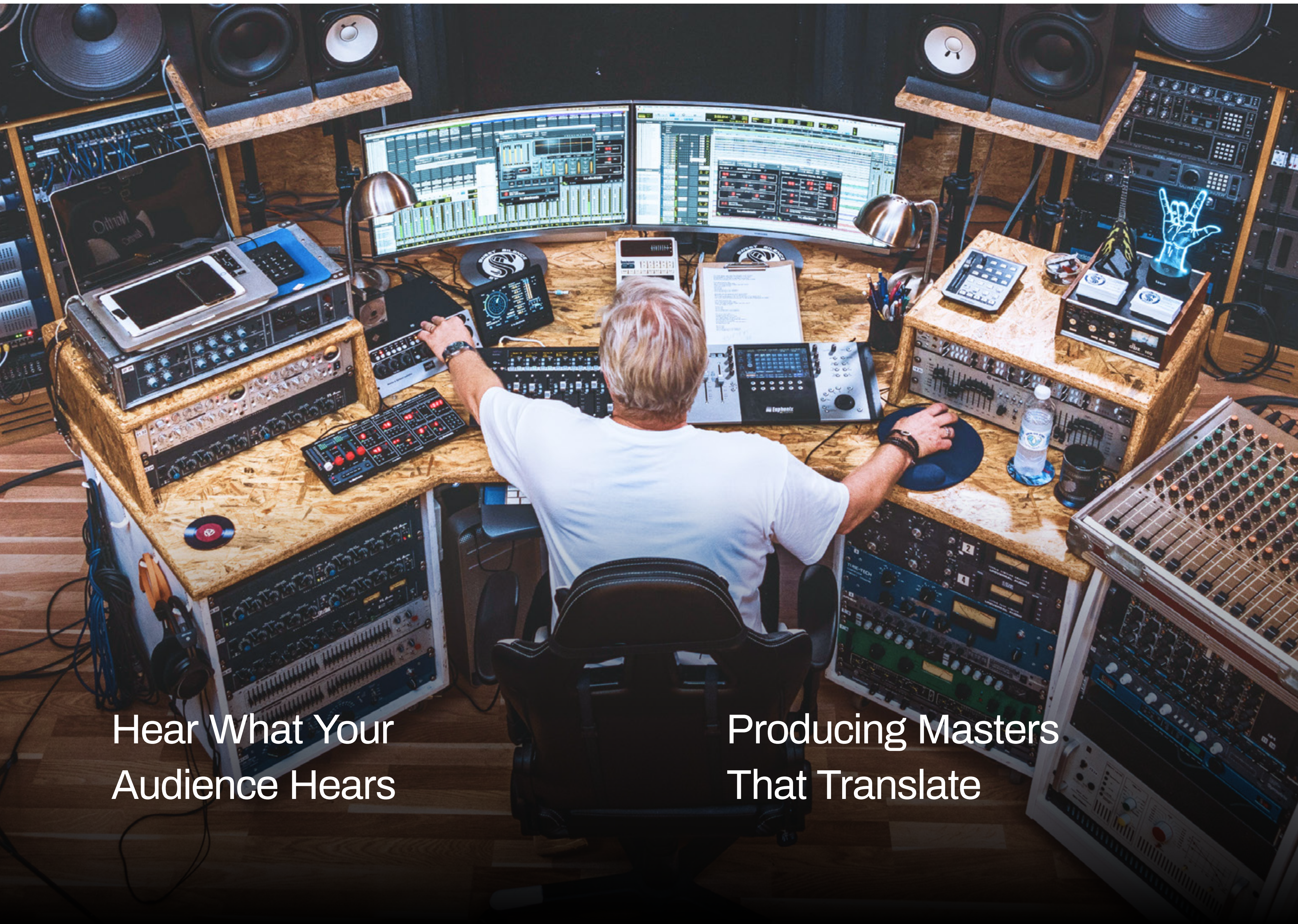
sonarworks

Creation and Translation



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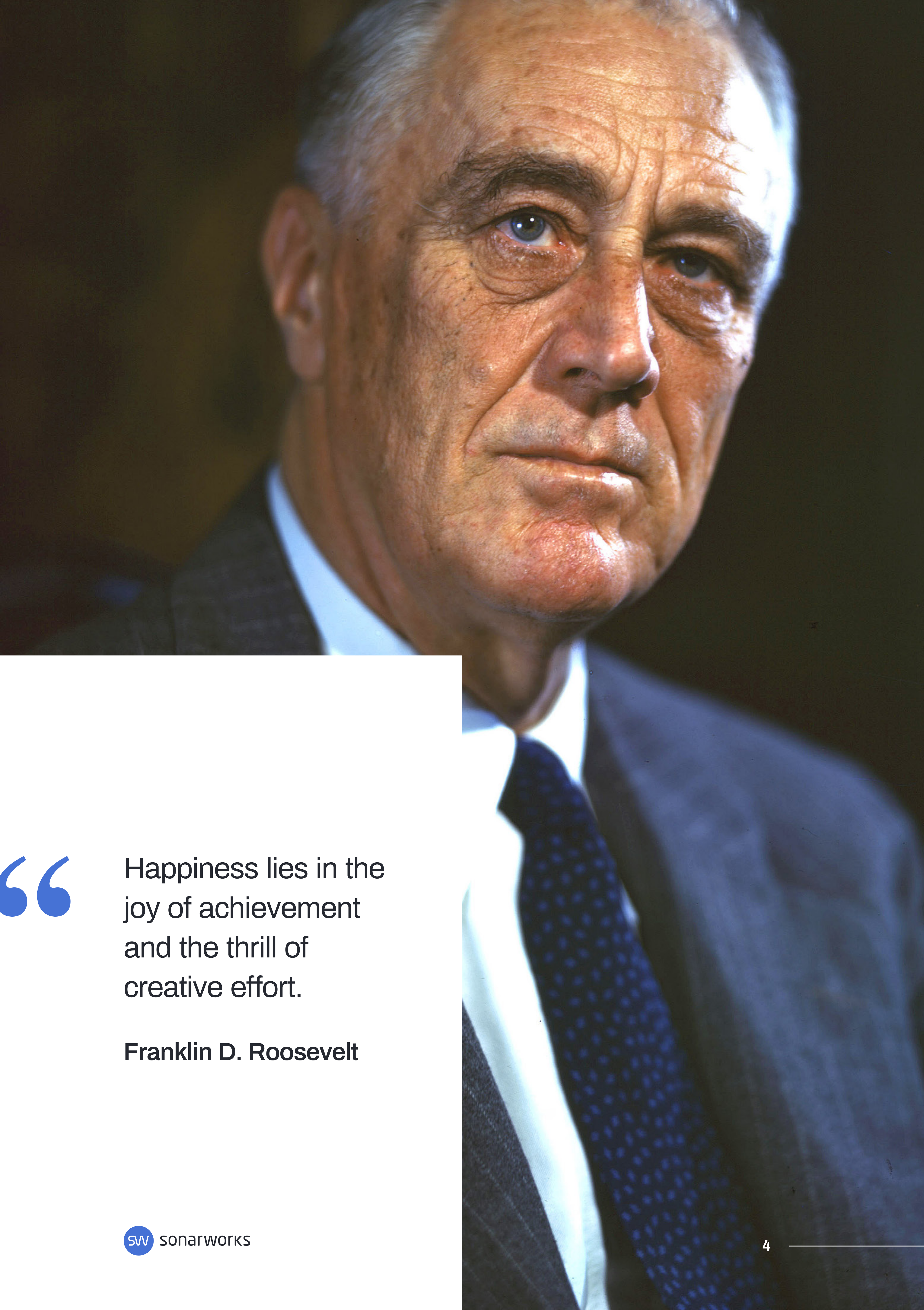
Introduction

Creating music, sound, images or any kind of art nourishes our soul and makes us human. Creating art is a gift that we should never take for granted and by sharing our vision with the world, we benefit both ourselves and those around us. Creating can also be a curse. We chase elusive feelings and even question our own self-worth when we stumble while expressing our vision.

Music and sound, unlike other artforms, transform to fit the environment that we listen in, so sound uniquely creates an experience that fills time and space. Because sound is an experience, the sounds we produce influence the space around us as the space around us interacts with the sounds we create. When we better understand our listening space, we can create more freely-with the confidence that the rest of the world will hear our artistic visions just as we mean them to be heard.

The topics in this eBook will bring into focus an understanding of how our listening conditions can be controlled to provide a trustworthy and translatable representation of what our audience will hear. How we as creators listen ultimately affects what our audience hears. How loud should we listen? How pure must our monitor system be? What kind of files need to be distributed? These questions will be answered in this eBook.



“ Happiness lies in the joy of achievement and the thrill of creative effort.

Franklin D. Roosevelt

Chapter 1

Hear What Your Audience Hears

Listening and creating sound is what we do. We are experts. Well, we are inspired creators and we work hard at becoming expert at effectively recording, producing, mixing and mastering sound. Our instincts and our soul inform us about what is good, musical, powerful, sad, or humorous. How do we know what we hear as “loud” or “powerful” will be interpreted by others as loud and powerful? Or bright? Too bright? Now we get into a bit of the science behind the art.

It turns out that humans tend to perceive qualities of sound, like loudness and frequency response, in measurable ways. If we are aware of the basic principles of audio perception, our own sound will translate better to our audience. For example, we need to understand what loudness really means and we need to know how accurate the frequency and dynamic response of our monitor system must be in order to deliver consistent results to our listeners.

A bit of knowledge in those areas will save you a considerable amount of trial and error. This chapter explains some of the principles of hearing and monitoring audio while we work.

What is Flat Sound and Should I Care?

by Adam Kagan

Choosing new studio monitors or headphones seems to be a very straightforward process. At the heart of the issue, we basically need a monitor system with flat sound. How much can I afford? What product has the best specifications for my budget? Easy enough. Is it really that easy? We should all suspect that the answer is not that simple. When choosing the best headphones or monitors many factors come into play that seem to have little to do with specifications or even price.

What we really want in studio monitors or headphones boils down to two key elements. First, we desire a speaker that produces a wide range of frequencies accurately with a wide dynamic range. Second, we desire a monitor system that is enjoyable to listen to for long periods of time. We need a monitor system that is both accurate and fun. Let's first take a look at the accuracy part.

Flat Sound Is…

Let's define flat sound to mean that any sound played through a monitor system sounds exactly like the original source. That is a difficult, if not impossible task for any monitor system. For example, if you listen to a violin in a room, the intensity and timbre of the instrument sounds different at any listening position in the room. So who is to say what does the actual source even sound like? Now let's just assume that we want to accurately reproduce the sound of a stereo pair of microphones that captured a musical performance.

That should be relatively easy. Compare the size of the microphone diaphragms to the size of typical speaker components. Obviously, there is not a simple one-to-one relationship between the device that captured the recording (microphone) and the playback device (speaker). So, we can see how the translation from source to monitor is a difficult journey, but we still desire accurate monitors that can reproduce all the frequencies and timbres we need to hear, so let's define accuracy.

Spec Speak

There are many specifications that manufacturers use to describe the performance of monitors and headphones, some of which describe their accuracy and some of which simply describe their physical or electrical attributes. (See sidebar) For this discussion, the most important specification relating to flat sound is frequency response. We basically need our monitors to reproduce frequencies that humans can hear–from 20 Hz up to 20 kHz. In reality, it is extremely expensive and complicated for a monitor speaker to cover this wide range, so we settle for a reasonable range that covers the music or sound that we mostly need to focus on.

For instance, string instruments from double bass up to violins, cover from a low of about 40 Hz to as high as 17 kHz. For hip-hop or electronic music with synth sub basses or even for pipe organs and concert bass drums, the lowest frequencies may extend slightly below 20 Hz. We can assume that most monitors produce adequate high frequencies because the power requirements and physical manufacturing of high frequency tweeters or horns is relatively simple. Accurate bass reproduction, on the other hand, requires much higher power, large physical devices and sometimes some very sophisticated tricks of physics.

Headphones will almost always beat studio monitors in their ability to affordably reproduce bass frequencies and even modestly priced headphones can boast the ability to reproduce frequencies from below 20 Hz to above 20 kHz. Studio monitors without high-powered subwoofers, or very large woofers, rarely produce much below 40 Hz, with many affordable models rolling off around 70 Hz. Therefore, we simply have to buy the speakers or phones with widest frequency response for the price we can afford. Well, maybe that's not really the whole story…

Monitor manufacturers often boast about their monitor's capabilities in a way that hides the flaws of the true frequency response by leaving out the data that doesn't look so good. The more respectable manufacturers don't simply show a frequency response range, but they also mention the amount of tolerance, or variability, throughout that range. For instance, a spec of 20 Hz to 20 kHz looks good, unless it is more correctly shown as 20 Hz to 20 kHz (± 6dB). That plus or minus number tells you that perhaps 1 kHz is 6 dB above the average response and 100 Hz is 6 dB below the average response. That gives us a 12 dB window of accuracy at any frequency. Not so good. Be wary of any frequency specification that does not mention the range of tolerance.

Basic Monitor Specifications

Sensitivity

Useful for headphones or passive monitors. A measurement of a speaker's efficiency. How much sound is produced for a given input level. Higher sensitivity numbers mean the monitor will play louder with a given input signal. Written as dB SPL output for a given input, like 90dB/1milliwatt. Doubling the power (mW) will increase the loudness by 3 dB.

Impedance

Impedance, a measurement of electrical resistance in ohms (Ω). For headphones, low impedance phones (30 ohms) will play loudly with a portable music player, like a phone. Higher impedance headphones (250 ohm) are more appropriate with professional headphone amplifiers and studio equipment.

Frequency Response

A monitor's ability to produce sounds from low bass to high treble, in hertz. Usually this spec is accompanied by a tolerance in decibels, like 20 Hz to 20 kHz (± 3 dB). Ideal human hearing covers 20 Hz to 20 kHz.

How good is good?

In reality, a tolerance of ± 2dB or even ± 3dB is acceptable, as long as the overall frequency response curve doesn't dip up and down like a picket fence. Small variations in frequency response over wide frequency ranges are easily smoothed over by our brain and even a perfectly flat response from a monitor will not be perfectly flat by the time it reaches our ears For my money, when reading specs I value a flat(-ish) frequency response over total width of frequency response, so I would rather have a monitor that produces *50 Hz to 18 kHz (± 2 dB)* than a monitor that simply states *35 Hz to 25 kHz* with no mention of tolerance. Since bass is really the most difficult area, some manufacturers provide specs that look like "*± 1¾ dB from 60Hz to 19kHz and down 10 dB at 40 Hz.*" This spec tells us not that only is the frequency response extremely flat for most of the range, but also describes how the bass rolls off down to 40 Hz, where it is still present, but at a lower level. So, choose a frequency response that you feel covers the range you need, but with an accuracy that you feel secure about.

The rest of the story

Ok, we know how to interpret frequency response specs, and we know what flat sound is. Now we get to the part of what do you actually want to listen to all day long for inspiration, comfort, appropriate volume and, of course, accuracy. Most of these factors come down to personal preference and relate to the design and manufacture of a specific device. Ported speakers sound different than sealed boxes. Open back headphones sound different than closed-back headphones. Different crossover frequencies and driver sizes affect midrange phase response, which colors the sound in a way that a frequency chart will not explain. Ribbon tweeters sound different than metal dome tweeters, but again that difference is not shown by their frequency response graph. Speaker personality and timbre must be considered along with frequency response to truly judge the type of impression a specific monitor produces.

The House Curve

After all this info about flat frequency response, some people, like hip-hop producers, may simply prefer lots of bass in their monitor system because it feels good! We make music because we enjoy the emotion, the fun, and the mood. Creating sound is our passion and our vice, not scientific experiment. Once you've chosen a monitor whose frequency response and personality you get along with, don't be afraid to play with the overall tone of the speaker. The overall treble vs. bass response is often referred to as the “house curve” of the monitor system. Most monitors allow you to slightly customize the low, mid and high frequencies of the monitor. These onboard equalizers help with accuracy and room correction, but may also be tuned for your personal taste. Once again, flat sound represents a sort of accuracy, but flat sound may not be a productive or enjoyable way to work.

If one should choose to incorporate a house curve into their monitor setup, it becomes more likely that mixes and masters will not accurately translate to the outside world. Therefore, if you enjoy working on a monitor system with hyped bass, you must be aware that outside your room the mix will have less or possibly uneven bass. Mixers who regularly use the same mastering engineers come to trust that their mastering engineer will correct this type of problem, but if you master on a hyped system, the potential for poor translation increases.

Simulated Target Curves

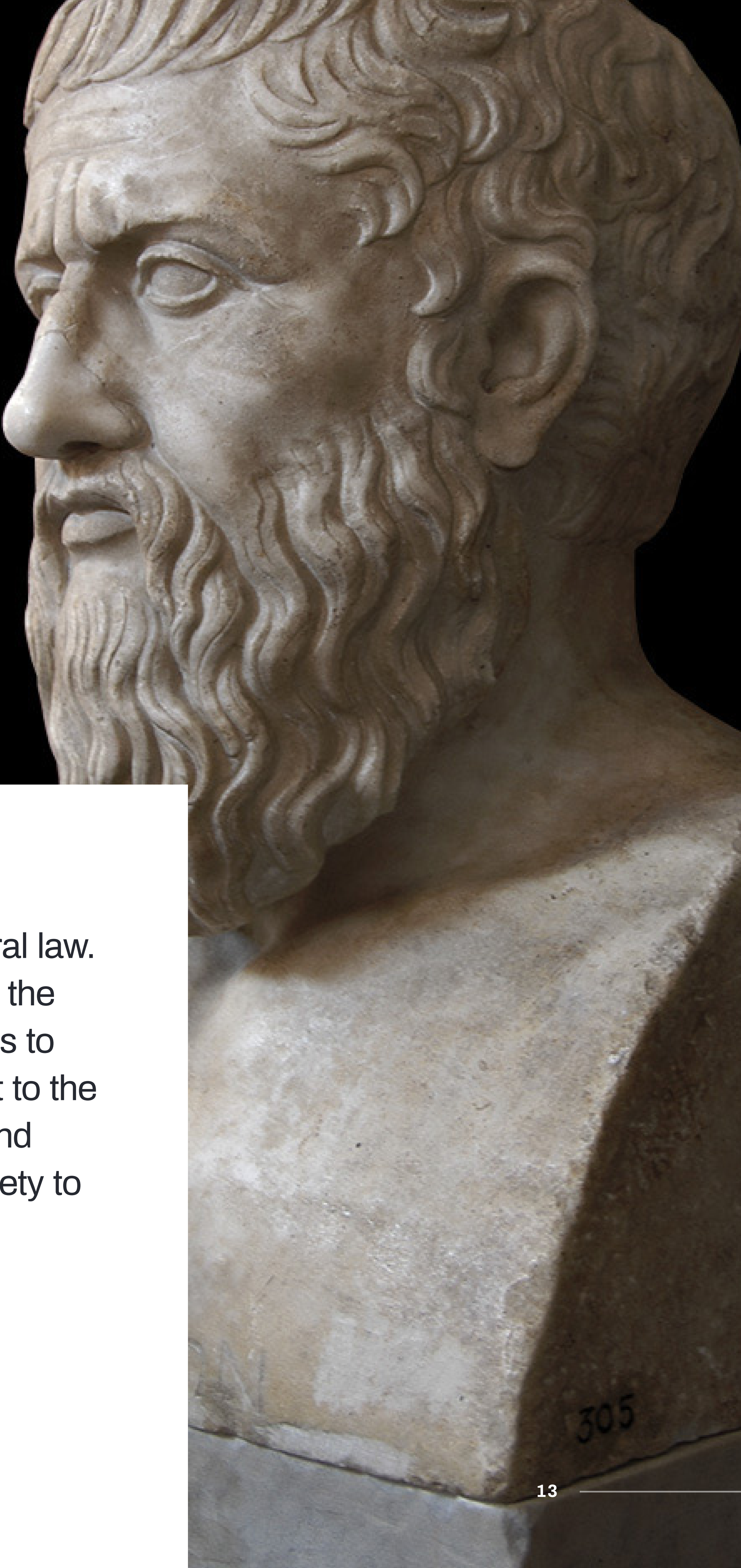
Sonarworks software provides target curves to simulate a typical domestic listening environment and also an X-Curve environment. The domestic playback curve provides an opportunity to audition what a typical home stereo is likely to sound like, while the X-Curve simulates the sound of a movie theater playback system. Sonarworks intends these playback curves to be applied after your listening environment is corrected to be flat. That is, once your system is known to be flat, you change the target curve (house curve) to simulate a specific playback environment.

Conclusion

Flat Sound

I prefer to say accurate, or flat sound is a good starting point and then fine-tune the sound to your liking. Every studio monitor, by way of its physical design and production trade-offs has its own sonic personality, but studio monitors tend to be more accurate, trustworthy and more durable than home stereo models. Try many different monitors, observe what your peers and heroes use, and at the end of the day, trust your gut. Of course, don't forget to acoustically treat your room. Experiment with creating your own house curve so that you enjoy listening in your own environment, but be aware of potential translation problems!

Remember that Sonarworks Reference software allows the user to create a flat sound and then apply a custom house curve via its built-in bass boost and tilt equalizers. Flat sound ultimately describes a certain accuracy that will translate to other systems, but remember to always keep things enjoyable and productive.



> Music is a moral law. It gives soul to the universe, wings to the mind, flight to the imagination, and charm and gaiety to life and to everything.

Plato

What is Loudness?

by Brad Pack

Volume. Level. Gain. Amplitude. We engineers use a lot of different words to talk about loudness—but none of them offer an entirely accurate description. Loudness, a commonly misunderstood term, is as complex as it is important. This article will explain everything you need to know about loudness so you can master your tracks like a pro. Read on to learn about the difference between peak and RMS metering, different ways of measuring sound, and different methods for achieving loudness when mixing and mastering.

What Is Loudness?

Loudness is commonly confused with volume. The two terms, however, are entirely different concepts. **Volume** is a scientific measurement of *the quantity or power of a sound*. **Loudness**, on the other hand, is much more difficult to quantify as it is completely subjective and based entirely on your personal perception of sound. The frequency content, duration, and volume of a sound are all factors in how we perceive its loudness.

We often describe volume using Sound Pressure Level (SPL), which measures the change in air pressure caused by a sound. If you've ever stuck your hand in front of a subwoofer and felt those pulsating bursts of air, you understand the concept. We can measure SPL with a meter, but SPL doesn't gauge how subjectively powerful or *how loud* a sound will seem to an individual.

The easiest way to understand loudness is to listen to two different frequencies at the same monitor volume level. First, play a 1 kHz sine wave and adjust your monitor controller so the volume is comfortable.

Now play a 10 kHz sine wave without adjusting the monitor controller. Hopefully you will notice that the two tones do not have the same loudness, even though the volume level has not changed.

Volume can also be measured in its electrical form using the decibel system. A decibel (dB) measures the ratio between two levels: the level being measured against a fixed reference level. Decibel numbers relate to a specific scale, or reference point, like a voltage level. You can think of it like measuring temperature, where we all know that 75 degrees Fahrenheit is much different than 75 degrees Celsius. It is important that you know the scale, like C° or F°, in order to understand what the temperature measurement means.

Analog and Digital Levels

For analog audio, volume may be measured using the dBu scale on a VU meter, which represents a value referenced to .775 volts. VU meters are calibrated so that a professional audio meter reads 0 VU at a voltage of +4 dBu, or 1.228 volts.

In the digital domain, volume is measured using the dBFS scale, which stands for “decibels relative to full scale.” Digital audio signals become clipped at 0 dBFS, which is the loudest digital signal that can be represented without distortion. While there is no single standard for converting between digital and analog levels, many DAWs define analog +4 dBu, or 0 VU, as -18 dBFS, while some companies use other standards, like 0 VU = -20 dBFS. A reference level or -18 dBFS = 0 VU provides 18 dB of headroom over 0, so the maximum analog level would be +22 dBu, or 18 dB above +4 dBu (0 VU).

Peak Metering vs. RMS Metering

Now that we've determined how to accurately measure volume levels, let's talk about how that information is displayed. Most analog and digital systems use a combination of peak metering and RMS (VU) metering. For instance, Pro Tools meters, when set to display “Digital VU,” show the peak level with a single yellow tick and the RMS level with a solid green bar on the same meter display. The peak floats above the green bar and it is easy to see both values and compare the peak vs. average level as the sound plays.

Peak metering measures the maximum instantaneous level of a signal. Peak metering does not offer an accurate measurement of perceived loudness, but instead indicates how close our signal is to clipping.

Designed to emulate the response of analog VU meters, digital RMS meters react much slower than peak meters and more accurately display the perceived loudness of a signal. Sustained sounds display higher RMS levels than transients signal (like short percussion hits).

Peak + RMS Levels

To better understand RMS (VU) vs. peak levels, simply create a click track in your DAW and take a look at a meter that displays RMS + Peak. The meter will display the average (RMS) level as a solid bar and the peak level as a single bar or tick. A click track, which has a very high peak level and low average level, will show very different levels for the peak and average levels. Now try the same experiment with a string pad or flute sound. In this example, you will notice that the RMS and peak values are very close to each other.

This experiment shows us that percussive sounds have very high peak levels but low average levels, while less percussive sounds have similar peak and average levels. The example in figure 1 shows a Pro Tools meter displaying a plucked acoustic guitar next to a distorted electric guitar and you can see the differences in the peak vs. RMS level on the meters.

Pro Tools Digital VU Meter

This meter displays the average level in green and the momentary peak level with a yellow line, or tick.

Peak Level

-18 dBFS = 0 VU

RMS or VU Level

The K-System Meter

In practice, many engineers use a combination of peak metering, RMS metering, and their ears to determine the true perceived loudness of a track. However, a more sophisticated system for measuring loudness was introduced in by Bob Katz in 2000.

According to Katz in the September 2000 issue of the AES Journal article *How To Make Better Recordings in the 21st Century – An Integrated Approach to Metering, Monitoring, and Leveling Practices*; "The proposed K-System is a metering and monitoring standard that integrates the best concepts of the past with current psychoacoustic knowledge in order to avoid the chaos of the last 20 years." In other words, he found a way to set meter levels and monitor volumes that makes sense to our ears and eyes.

The idea behind the K-System is that different types of music require different amounts of headroom, but the average level of music should be standardized. Classical music or film scores require about 20 dB of headroom above their average level to allow for loud moments, dynamic pop music requires 14 dB of headroom, and very compressed pop or rock music requires 12 dB of headroom. Bob Katz proposed one meter scale for each musical style and named them K-20, K-14 and K-12. Therefore, a K-20 meter places its 0 dB (0 is the target) at 20 dB below full scale, K-14 places 0 dB at -14 dBFS, and K-12 places 0 dB at -12 dBFS. So we can say "Master this song for K-14" or "Master this for +2 dB over K-14."

To use the K-System in your studio, play pink noise with a signal generator plugin at a level that reads 0 dB on a K meter in your DAW. Adjust your monitor gain so that an SPL meter set to C weighting and slow response reads 83 dB at your listening position when only one speaker is playing. Then repeat this process for the other speaker. You may want to mark your monitor controller positions for each K scale reference level (K-20, K-14, K-12), and you might find that the -6 (76 dB SPL) is more appropriate for long days of music production. The K-scale you choose as your meter will depend on the style of music you are producing.

Reference Loudness Levels

It turns out that most humans enjoy music (or films) at nearly the same loudness level (SPL), which turns out to be right around 83 dB SPL. Therefore every movie theater is calibrated so that pink noise played at -20 dBFS produces a volume level of 83 dB SPL from each speaker in the theater. This allows a film mixer to know that their mix will translate in every theater, all around the world. When working on modern pop music productions, especially in smaller rooms, we often lower that monitor level by 6 dB to 77 dB SPL.

LUFS / LKFS

The K-meter has been further refined and a modern standardized approach to measuring perceived loudness is called the LUFS or LKFS system. Originally called Loudness Units relative to Full Scale (LUFS) and renamed to Loudness, K-weighted, relative to Full Scale (LKFS), this loudness standard was created in 2006 as to standardize audio levels for video formats. Most broadcast, film and video game companies have adopted LUFS / LKFS as the standard for measuring loudness and a typical film mix level spec would be stated like: -27 dB LKFS ±2 LU.

In order to represent perceived loudness more accurately than ever before, LKFS/LUFS measures levels over a longer period of time and with a flatter frequency response than a standard VU meter. Even streaming audio services are getting in on the action. Here's what Spotify recommends regarding loudness levels:

"Target the loudness level of your master at -14 dB integrated LUFS and keep it below -1 dB TP (True Peak) max. This is best for the lossy

formats we use (Ogg/Vorbis and AAC) and will ensure no extra distortion is introduced in the transcoding process… Negative gain is applied to louder masters so the loudness level is at ca – 14 dB LUFS. This process only decreases the volume in comparison to the master; no additional distortion occurs."

All this technical speak means that a song with an average level on a K-meter or -14 dB and a peak level of -1 dBFS will play on their service exactly at the volume it was uploaded at. A louder song will simply have its volume lowered to match this reference level. All streaming services use similar rules so that all the songs play back at a similar volume and the listener does not have readjust their volume for different songs. In practice, most well produced top 40 pop songs have a K level of about -8 dB during their loudest chorus section and an overall K level of about -11 dB. It is also recommended to leave 1 dB of peak headroom on masters so that conversion to mp3 or other lossy formats does not introduce extra distortion.

Achieving Loudness in Mastering

Since most streaming services adjust, or normalize, the level of every track to -14 dB LUFS, mastering engineers can't simply use a maximizer or limiter to push our masters as loud as possible. Instead, we have to create the correct impression of loudness using mastering tools like equalization, saturation and dynamic range processing to create the intensity and loudness that suits our music.

Tools for Measuring LUFS Levels

While some DAWs like Steinberg's Cubase and PreSonus Studio One are equipped with LUFS meters, most DAWs only provide basic peak and RMS meters.

Don’t worry; there are plenty of highly accurate and surprisingly affordable, or even free, options for LUFS meter plug-ins. Listed below are a few of our favorites.

Loudness Meters

- Nugen Audio VisLM or Mastercheck Pro;
- iZotope Insight 2;
- Waves WLM Plus;
- Meter Plugs K-Meter and Dynameter;
- Youlean Loudness Meter;
- Klanghelm VUMT and VUMT Deluxe.

Conclusion

Now that you know what loudness is, how to measure loudness, and what LUFS your mixes should be at, the only thing left to do is put these new skills to use. Calibrating your room for loudness is one important step towards consistent mastering and mixing, and perfecting your monitors' frequency response with a tool like Sonarworks Reference 4 may be the next logical step. Happy mixing and mastering!

> Once someone asked me three words that best describe me and I said 'Loud, Louder, and Loudest'!

Anastacia

Calibrating Your Listening Room for Loudness?

By Eli Krantzberg

When it comes to mixing and mastering music, our aesthetic sensibilities can be fickle. What sounds right to us one day might not seem right the next day, or when listened to at different volumes, or through different monitors, or in different listening environments. Our perception of sonic qualities, including instrumental balances and overall intensity, are influenced by a number of variables. As simple as it sounds, monitor volume is one of the main variables that we can control to help us achieve consistency in our mixes.

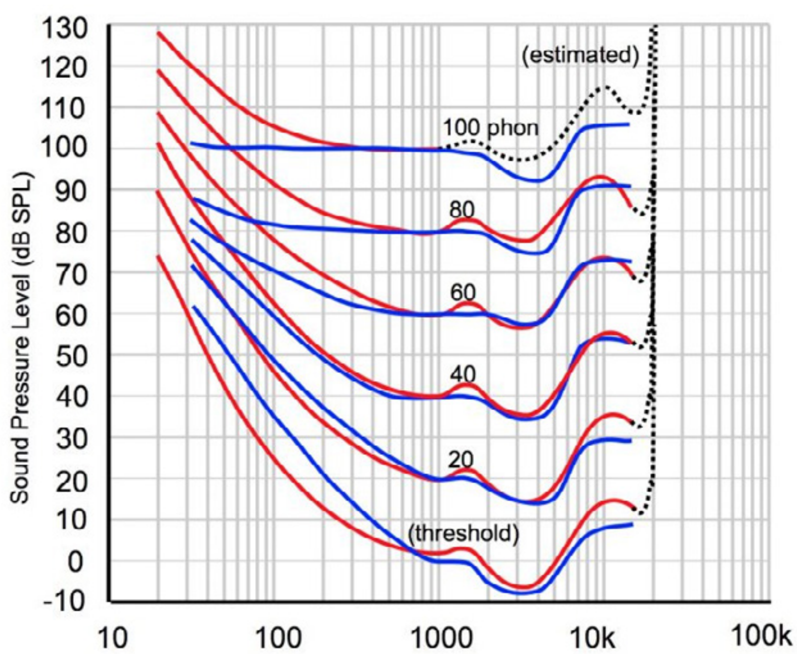
One of our main goals while mixing and mastering is to achieve a certain amount of musical impact or feeling, which we call loudness. While volume describes the actual sound pressure level (dB SPL) in a room, loudness can be thought of as how intense, dense, powerful or “loud” the sound feels to you. For example, a recording of an explosion played back at a low volume still sounds like a “loud” sound to our brain, while a recorded whisper played back at a high volume still feels like a soft sound.

Both volume and loudness are important considerations and during mixing and mastering, where our job is to create the appropriate intensity, or loudness, of the music. Maintaining a consistent, or calibrated, listening volume helps us remain focused and consistent when making adjustments and decisions that affect the loudness of our project.

Is Monitor Volume Important While We Mix?

At different playback volume levels our brains perceive the loudness of different frequencies unevenly. For instance, when monitoring at low volumes, our ears and brain focus more on the midrange while the low and high frequencies are not perceived with the same emphasis. At higher listening volumes, the low and high frequencies appear more prominent while the midrange frequencies command less of our aural attention. Objectively, and in terms of physics, the actual frequency balance stays the same at all volume levels, but the human brain's interpretation of the frequency balance changes.

In the 1930's two acoustic researchers, named Harvey Fletcher and Wilden Munson, studied the phenomenon of human perception of frequencies vs. loudness. They developed a loudness contour, called the Fletcher-Munson Curve (fig. 1), which indicates the volume (dB SPL) levels across the frequency spectrum necessary for the listener to perceive a constant loudness level when presented with pure steady tones. This calculation was refined in 1956 and is now used as the ISO 226 standard for Equal Loudness Level Contours.



Equal-loudness contours (red) (from ISO 226:2003 revision)
Fletcher–Munson curves shown (blue) for comparison

Given that we perceive the blend of frequencies differently at various volume levels, how loud should we monitor when we are mixing or mastering? From the Fletcher-Munson Curve graph (fig. 1), we see that a monitoring level of 85 dB SPL (in a large room) provides the flattest hearing curve. If you are working in a smaller space, like a typical bedroom studio, the flat response is probably closer to 79 dB SPL. Now that we are aware that our sensitivity to bass and treble frequencies becomes more flat as volume increases, we can infer that monitoring too softly will cause us to want to increase the bass and treble content of our mixes (because at low volumes the midrange is most prominent and bass and treble seem too low).

Calibration

Our goal is to find a trustworthy, or calibrated, monitoring level which sounds well balanced in your listening environment. To calibrate your monitoring level, you will need either a software or hardware SPL meter. I use an iPhone app called SPLnFFT Noise Meter, but you could also use a dedicated SPL meter, like the Velleman analog meter or the Extech digital meter.

Start by bringing your monitor levels down for now–we will set monitor volume as the final stage of the overall adjustments. Set up a signal generator plugin in your DAW to output pink noise at -20 dBFS and verify that level on the master fader output meter. If you use a stereo signal, keep your output panning at 100% L/R, but if you use a mono signal generator, make sure to center the pan knob.

If you have a software control panel, like UAD's Console application, you can double check that the level reaching the hardware is showing as -20 db. Once this is established, set the output gain of your audio interface all the way up at 0 dB.

For techies:

We typically reference -20 dBFS for digital levels because that level matches the analog world's headroom standards, where -20 dBFS is roughly equal to +4dBu or 0VU. Note that -20dBFS = +4dBu is the SMPTE/AES standard. Pro Tools and the BBC default to -18 dBFS = +4dBu, the EBU standard. Among mastering studios, -14dBFS = +4dBu is also common.

Grab your SPL meter or launch your SPL app and position it in your listening position, where your head would usually be, with the microphone pointing up. If the app or SPL meter gives you the option, select the C weighting scale and slow response time, as these are the most accurate settings for this type of measurement. Pan the pink noise to one speaker and slowly bring up your monitor volume until the SPL meter is registering 82 dB SPL. Pan the signal to the other speaker and verify that you have the same 82 dB reading (less than ± .5 dB between the left and right speakers is close enough). When you pan the pink noise back to the center you will notice the meter will now read 85dB, which is a 3 dB increase because both speakers are playing. You should also notice focused center image, which will confirm that your speakers are in polarity with each other. For mixing or mastering pop music, an 85 dB playback level may feel too loud, so it is common to readjust this level down to 79 dB. This level (79 or 85 dB) is now your calibrated monitor level. On monitor controllers with a digital readout it is easy to store and recall this precise setting, otherwise you may want to place a mark on your monitor controller to indicate this playback level.

With your monitor level set and both speakers matched, you can now mix, confident that you are working at an optimal and consistent level.

Because of the Equal Loudness phenomenon, it is vital to keep your monitoring level consistent when mixing. Of course you will find it useful to occasionally monitor at quieter levels to verify that the important elements of the mix remain audible even at low levels. Many audio interfaces and monitor controllers have an adjustable DIM function, which allows you to press a button and drop the monitor volume by a preset amount. DIM is useful for checking balances at a lower listening level and also simply to allow you to have a quiet conversation while the music is playing. Switching off DIM will instantly set your monitors back to your calibrated monitoring level.

If you regularly mix or master on headphones, level calibration is not as straightforward. We can't easily measure the SPL level of our headphones, so we must rely on our musical sensibilities. For headphone calibration, I suggest you listen to well-mastered reference songs at various levels until you feel that you have found a level that provides the proper feeling of impact, intensity and a full-range frequency response. Then, try to mark your headphone monitor level control at this calibrated level. As with monitor speakers, the more consistent you are with your listening level, the more consistent your mixes and masters will turn out.

Mastering Tip

To set your optimum mastering monitor level, import some reference tracks that have been mastered and listen to them at your calibrated level. Then make small (2 or 3 dB) level adjustments to find your optimum calibrated mastering level. Now you can confidently compare your masters to commercial reference masters for their loudness, impact and feeling without relying on level meters.

Mixing Tip

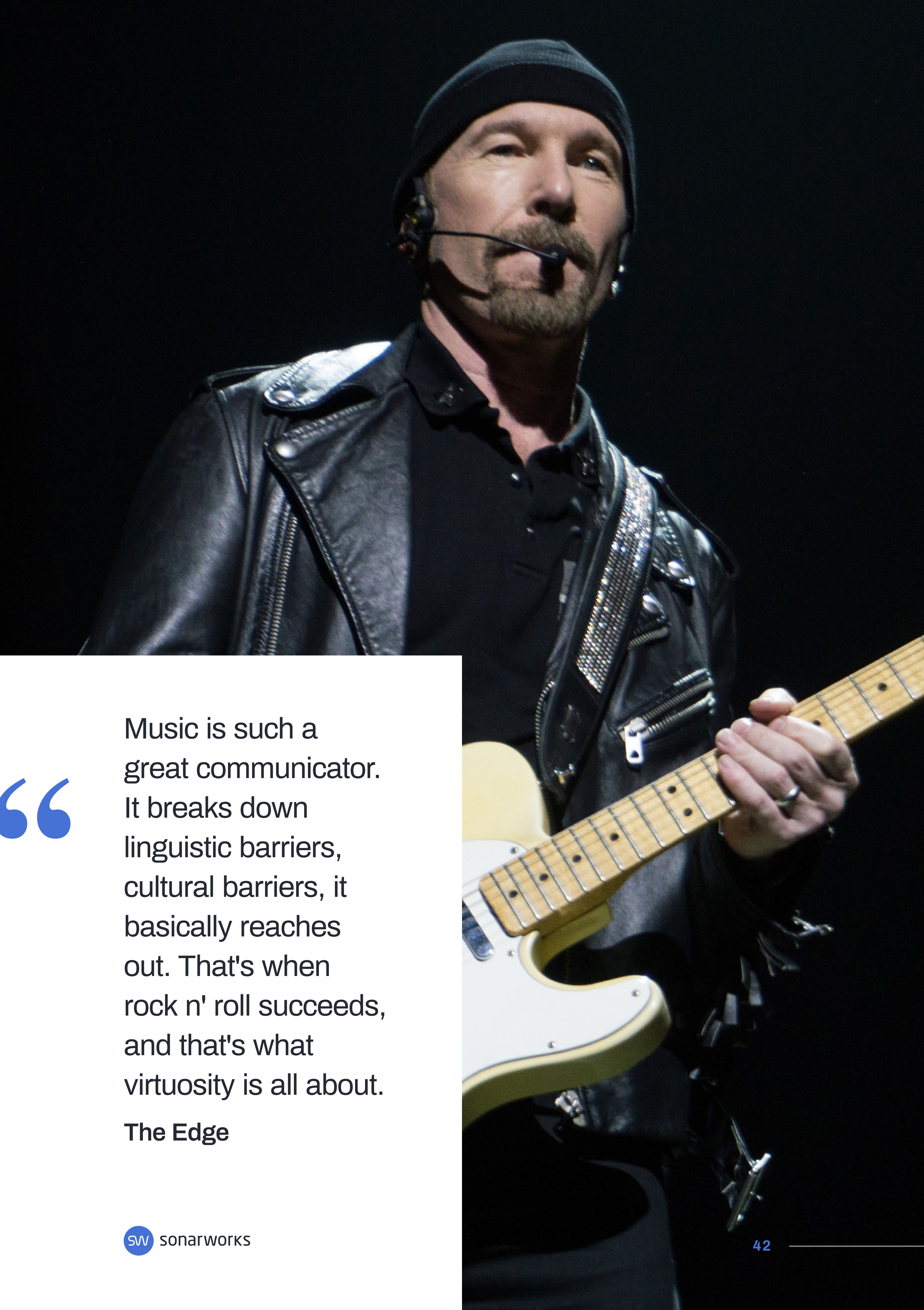
To set your optimum monitor level for mixing, import some reference tracks that have been mastered and lower their playback level in your DAW by 6 to 8 dB and listen to it at your calibrated level. Then make small (2 or 3 dB) level adjustments to find your optimum calibrated listening level. By lowering the level of mastered tracks by 6 or 8 dB you can more fairly compare your unmastered mix to mastered reference songs.

Calibrating your room for loudness is an essential step towards consistent mixing and mastering. Calibrating your monitors for a flat frequency response is equally important and tools like Sonarworks Reference 4 create the optimum EQ curve for your monitors in your listening environment. Calibrating your monitors for proper loudness and frequency response, puts you well on your way to creating professional sounding, competitive mixes and masters.

Chapter 1

Key Takeaways

- [ ] Strive for Flat Sound in your listening environment
- [ ] Experiment carefully with a house curve for you monitors
- [ ] Learn to decipher manufacturers specifications
- [ ] Practice listening at a calibrated loudness for the best frequency presentation and consistency
- [ ] Get a feel for loudness meters, like the K Scales and LUFS meters
- [ ] Use an SPL meter to calibrate your listening room
- [ ] Learn to listen for loudness, or density and power, not just volume
- [ ] Use Sonarworks Reference software to develop a flat in-room response for your monitors



> Music is such a great communicator. It breaks down linguistic barriers, cultural barriers, it basically reaches out. That's when rock n' roll succeeds, and that's what virtuosity is all about.

The Edge

Chapter 2

Producing Masters That Translate

Confidence is security for an artist. If we feel confident that our vision will be delivered to our audience and that they have a fighting chance of interpreting our vision nearly the way we imagined, then we can sleep a bit better. If we can't trust that our sonic picture is seen in the proper light, then what's even the point? Producing masters that translate to our intended audience seems to many people to be a secret art that only a few mastering engineers or mixers have the key to.

The truth is that if we work in a trustworthy environment and create inspired productions that feel right to us in our calibrated, true listening space, then the audience will receive a proper presentation of our vision. It is really as simple as that. Work in a calibrated space, create excellent sounding productions and export files that remain as true to the original source as possible. The following articles explain how this works and points you in the direction of producing masters that you will be proud of on any playback system.

6 Tips for Creating Masters that Translate Well to All Systems

By Nick Messitte

The topic at hand requires more than just this article to be complete—there are books, podcasts, and courses devoted to this subject. Nevertheless, this brief article highlights some important tips, and points you toward further techniques that are well worth your time to explore. Here are six tips to get you started.

1. Treat your room and calibrate your speakers

You need to face this simple truth: you are far better off spending money (and effort) on room treatment and monitors than on any other piece of gear. How can you appreciate all the sonic wonders of your beautiful gear if you can't trust what you're hearing?

So, you need to understand the basics when it comes to your room. Room dimensions, construction materials, surface finishes, and other physical elements all influence the acoustics. As an experiment, run Sonarworks calibration software on your speakers in an untreated room and check out the corrective room curve suggested by the software. If this curve is anything but (mostly) flat, the curve is telling you just how much influence the room exerts over your monitors.

When setting up a new room it is advisable to consult an acoustician, but their services aren't inexpensive and their processes may seem complicated. However, most companies that manufacture room treatment products have acousticians and consultants on staff. These professionals can make basic recommendations to their customers regarding off-the-shelf treatment options. A few of the acoustic companies with consultation services are GIK Acoustics, Prosocoustic, and Auralex Acoustics, and there are many others. Many companies like these offer online analysis tools, and are also happy to review photos or sketches of your room in order to provide expert advice for specific room treatment options.

As for your speakers, there are a couple of good guidelines (note: speaker placement is covered in depth in our blog post by Barry Rudolph here). Start by setting up your speakers about 20% of the room-length away from the front wall, and symmetrically placed between the side walls. Set up your monitoring position in an equilateral triangle, and aim the speakers to focus on a point just behind your head, when you are seated in the sweet spot.

These are good starting points, but you will need to refine the room setup with your ears. Use reference material for this—listen to songs you know inside and out. Key in to specific mix elements that reveal acoustic problems: stereo width and imaging, mix details like vocal reverbs, bass presence and tightness and the tonal balance at various listening levels. If what you hear doesn't stand up to the test of "does this sound the way I know it should," then you have been informed of the shortcomings of your current setup. Use mixes that you are very familiar with and also make use of test recordings, like those from audiophile companies or from the Audio Check website. In particular, the MATT test will let you hear with a simple audio test what frequencies cause your room to ring, reverberate and smear the sound.

2. Maintain a calibrated monitor level during mixing and mastering

When you turn up your monitors, a funny thing happens to the lows and highs: they stick out more, feel larger and louder. Turn the dial too low and you get the opposite—these frequency ranges seem harder to hear while the midrange remains present.

So, it's best to maintain a proper and consistent level while mixing and mastering. You will not make objectively good judgments if you continuously fiddle with the level—especially when comparing the master to the original mix or your mix to reference mix.

Research shows that for typical project rooms, a consistent listening level of around 76 to 79 dB SPL (or as high as 82 dB in a large mix room) presents a properly balanced frequency response. Working at the appropriate volume allows you to make decisions you can trust and ones that will translate to the outside world. Try starting at 79 dB and see if that is comfortable and if you feel comfortable with more or less volume, step it up to 83 or down to 76 dB. The actual level is not as important as being consistent and mixing or mastering at the same level every day. (Be sure to read our blog post "Calibrating a Listening Room for Loudness" by Eli Krantzberg)

Beyond a calibrated monitoring setup, you must also level-match your mix or master with any reference for a true comparison. Level matching helps you to make sure you're making things sound better, not just louder. When I'm mixing for a client and they want to compare my mix to a commercial release, I simply turn down the commercial reference song by about 8 dB to give the impression of the same loudness. This way we are comparing frequency balances and mix quality and not simply reacting to loudness differences. Check out the Perception plugin from MeterPlugs, which allows you to compare level-matched versions of a raw mix to its processed master.

3. Produce a well-balanced, dynamic, and appropriately loud product

The goal of mastering is to show off a mix in the best light, both by itself, and in the context of an album or playlist. This means masters must sound balanced in frequency content as the listener moves from song to song. If the lows are too overwhelming, or the highs are too harsh, listeners will find them challenging to enjoy without fiddling with volume and EQ controls. Refer to our two previous tips, and you'll be on your way to a well-balanced master, since you can trust what you are hearing in your room.

We also need to take great care in ensuring proper dynamics–both during a master and from song to song. We don't want to crush dynamic range and kill a song's impact, while we don't want so much dynamic range that the soft parts of a song fall below the ambient noise or our listening environment. This is especially true for listening in cars, outside with earbuds or on built-in phone or laptop speakers.

It's not uncommon to apply volume automation during mastering to keep the overall dynamic range in check. Say, for example, we have a pop song where the mix jumps up 6 dB from the verse to the chorus. We could simply use volume automation to slightly (and tastefully) bring up the level of the verse to flow more musically into the chorus. Dynamic range considerations may vary from genre to genre, so you must stay attuned to what's appropriate for the style. Every client seems to wants a loud, banging master, but achieving loudness is not simply a matter of compressing and limiting or maximizing to high heaven. Rather, you must make other musical adjustments to achieve a desired target level. Keep in mind your distribution chain as well. For instance iTunes recommends leaving 1 dB of peak headroom to allow for the slight level increase when creating mp3s from .wav masters. Leaving this headroom gives the mp3 encoder a better chance of producing a less distorted mp3 file.

As an exercise in achieving loudness, apply a limiter or maximizer to achieve the loudness you desire. Is the mix distorting or feeling pushed too much? If so, start adjusting different EQ bands (before the maximizer plugin) to see what frequency ranges cause the mix to overload. Perhaps the sub bass needs to be pulled back or the lead vocal level is impairing your ability to make a loud master. I find that I'm constantly balancing the maximizer against EQ changes as I refine a master that. Each song has a sweet spot where the mix sounds musically balanced and appropriately loud. As always, compare your master to a reference in the same genre to ensure you are creating a musical and competitive master.

4. Use headphones for technical and creative checks

In professional studios, engineers tend to do the bulk of their creative work on monitor speakers and then check for technical quality issues on headphones. Headphones often highlight technical issues, including distortions caused by clipping or limiting, as well as clicks and pops that are often caused by edits, plugins, automation or clock issues. Sometimes, small noises in an otherwise great mix become much more apparent after the mastering process. After a QC pass, you may choose to repair technical issues with something like iZotope RX or the various dedicated repair and restoration tools provided by mastering DAWs.

Eschewing headphones during the creative part of mastering might not be practical when working in a suboptimal room or when volume is an issue. You might not have the best sounding room or accurate monitors when you're starting out, so go ahead and make decisions through headphones. Just be aware of the inherent risks of only using headphones, as presented in this article. If you rely on headphones for anything other than quality control, be sure to make extensive use of reference masters. Also, don't forget to try a Sonarworks headphone profile for your specific headphones, which will produce an accurate and trustworthy frequency response for your specific headphone setup.

5. Decide if you should reference material outside your room

Pros know their rooms and monitors inside and out. If a master plays well in their room, it will sound great on other systems. But, many engineers still provide a boombox or home stereo system for their clients to reference. Clients may be disoriented in the listening environment of a mastering room, so a boombox or Bluetooth system may be welcome reference point.

Mastering engineers who have tens or hundreds of thousands of dollars to make it happen can invest in proper construction, acoustical treatment, power conditioning, and high end wiring. Every little bit matters and the smallest elements all contribute to the final sound. That said, many engineers don't have access to those kinds of funds, so they must make use of all the tools at their disposal, including multiple reference monitor systems.

Listen on your iPhone. Listen on your car stereo. Listen on your boombox. Then go back and do it again. Don't just bring your master to these playback systems—bring along the original mixes and other reference material as well. Level match as best you can (the volume knob is your friend), and make quick, broad decisions as to whether your master stacks up against the competition. On a car stereo system, I don't mind if the vocal feels a bit too loud or soft, but does the low end hold together? On a phone speaker you won't hear the bass or kick drum accurately, but can you hear the lead vocal and important elements of the track? I find that when listening outside of my studio, I listen for the main musical elements and the meaning and emotion of the song. I don't worry too much about the small mix details like subtle effects or pannings.

6. A question: Should I master while I mix?

The consensus seems to be no, you shouldn't. These are two (too) different processes. Different mindsets and different goals. Some engineers, however, do master while they mix, and can do it successfully. These people usually have a lot of experience, a great deal of confidence, and the ability to quickly change mindsets. The master-while-you-mix process may finish up with a great mix that is also a competitively loud master, but that is still only part of the story.

I'd recommend you leave mixing and mastering as two separate processes. Mixing addresses the arrangement, production and emotional needs of the song, while mastering prepares a great mix to fit in with the world that it belongs to. You mix a song to convey its meaning, emotionality, intelligibility, story, and other unique facets. You master to compete and fit in context among songs of a musical genre or on a specific album and also to sound well-balanced in different of monitoring environments. Don't forget that mastering also includes album sequencing, technical quality control, proper encoding of metadata, as well as proper encoding for distribution or manufacturing.

To combine mixing and mastering into one step is dangerous. You ought to think nothing of metadata while mixing just as you ought not think of sidechain-compression on the bass while mastering. The mastering mindset is one of improving an already established product and preparing it for distribution, whereas mixing is like creating a single planet out of a universe out of chaotic, disparate elements.

Clients do expect your mix to sound “like a master,” so you may find yourself applying some mastering processing to your mix in order to achieve a competitive level and some added polish to your mix. Keep in mind that a little goes a long way and your perspective will change once you’ve been away from the mix for a day or two.

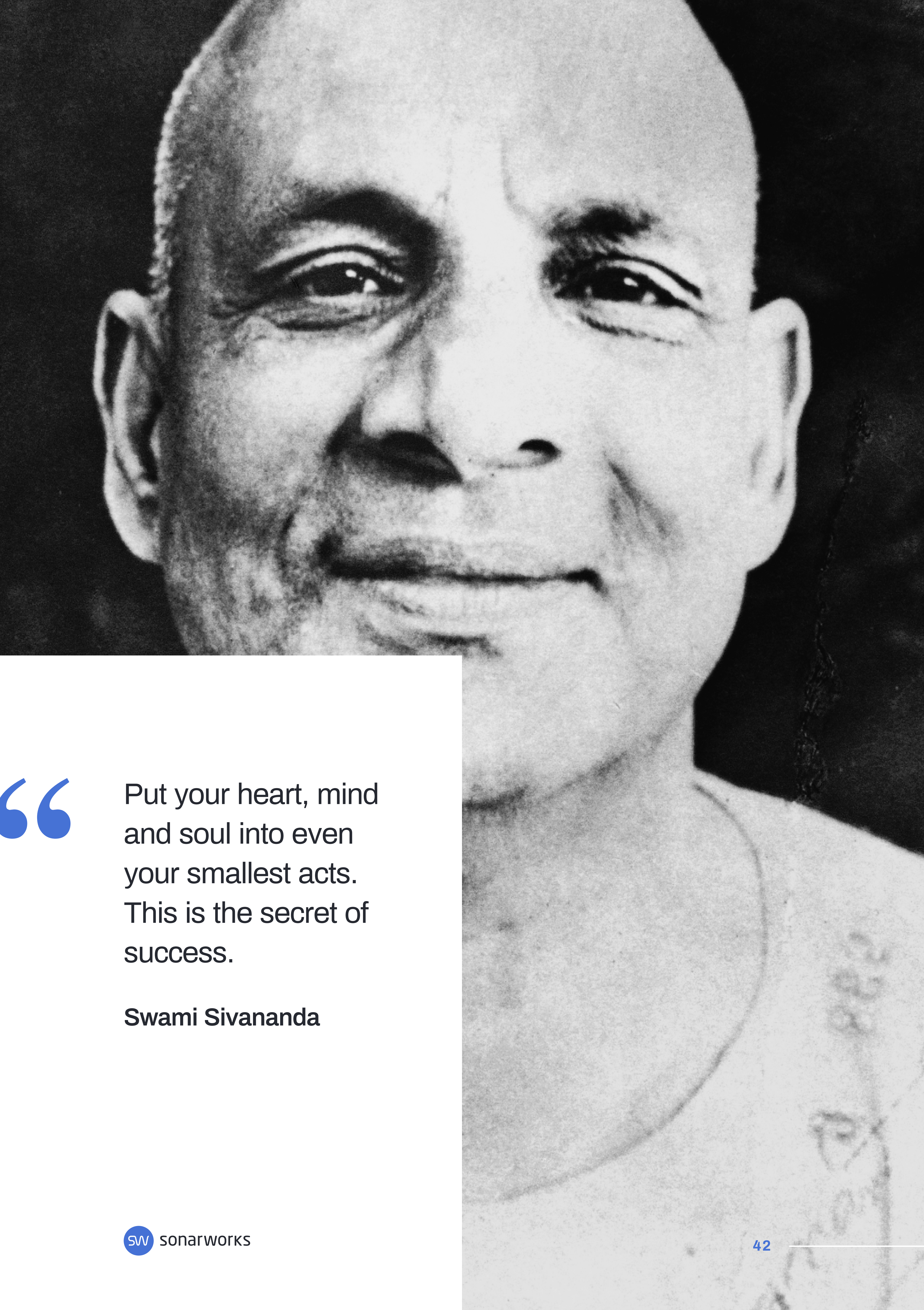
The Simple Truth

Let’s end this with the secret of making mixes that translate. If your room and monitor system is properly tuned and sufficiently accurate and you work at a proper and consistent listening level and you occasionally check your work outside of your room, masters that sound great in your room will automatically sound great everywhere.

Conclusion

Creating mixes that translate on all systems takes time and practice. However, it's a lot of fun! I recommend practicing the process every day. Take one mix each day that you have on your drive—it doesn't matter which—and try to make it sound like a mastered product that works on all reference systems, using the tips we've discussed. Try to keep the process fast; don't spend more than half an hour mastering a song during this practice routine.

Keep track of the elements of your masters that consistently surprise you, like too much bass or certain frequencies that feel too loud or too soft. Take this knowledge back to your mastering setup and try to figure out if you are not hearing those same problems in your room and, if that's the case, why can't you hear those problems. At the end of a few weeks, you'll be surprised at how much better your masters have improved!



> Put your heart, mind and soul into even your smallest acts. This is the secret of success.

Swami Sivananda

Creating Masters for Streaming and Selling

by Adam Kagan

Here's the situation: You, as a mastering engineer, just completed mastering a few songs for a new independent artist, and they love your work. Excellent job! Now they've asked for a few deliverables. First they would like copies of the songs for upload to all the streaming services. Second, they would like a copy of each song for their music videos, which will wind up on YouTube. Third, they would like to manufacture a couple hundred CDs to sell during their upcoming tour and also give away to their crowdfunding supporters. Your job, since you've gotten this far, is to create versions of the masters that are suitable for each of the above uses. Let's examine each case to see what's really needed.

Compression vs. Compression

When recording, mixing and mastering music we apply dynamic compression, which is a way to musically control dynamic and tone. During lossy encoding, like mp3 or AAC, data compression is applied, which reduces physical files size. For this article, compression refers only to data compression.

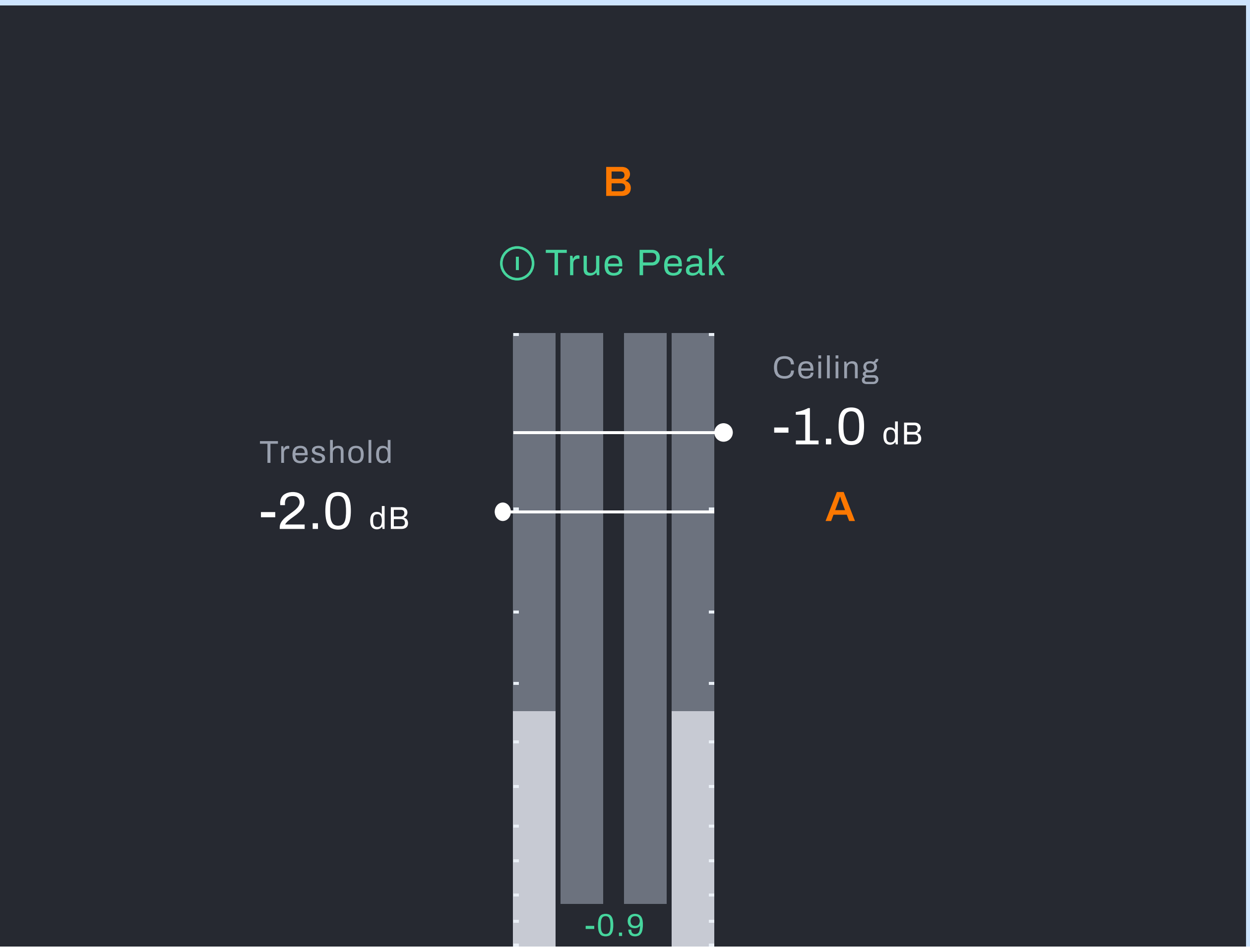
YouTube and Streaming Video

Audio for video is probably the simplest case, so let's get to that first. For video, you should deliver a 16bit, 48kHz .wav file version of your master and include instructions to the video editor to encode the audio as PCM, uncompressed audio. Every video editing workstation is different, but the editor needs to choose an export option, or codec, that embeds uncompressed PCM audio into with the video file. Do not let them export using a lossy AAC or mp3 codec.

When the high-resolution video with PCM audio gets streamed via YouTube (or other streaming services) the highest resolution playback setting should stream uncompressed audio and video. If the user chooses a lower quality playback setting, YouTube will stream a more compressed, lower bitrate and lower quality video and audio file. If your original video file contains an already lossy file, it will be transcoded again for lower bitrate streaming, so the viewer hears an audio track that has been twice damaged by lossy encoding. Demand that video editors keep this in mind! I have personally spent a bit of time learning the encoding options of a few common video editing programs so that I can gently offer some suggestions to a video editor as to what encoding format may be best for a given situation.

Headroom and Lossy Encoding

You should be aware that creating a lossy-encoded version (mp3 or AAC) of a music file will introduce some distortion, and thereby raise the volume of the audio file by a little bit. A 320 kbps mp3 may raise the volume by 0.5 dB, while a 128 kbps mp3 may raise the volume by as much as 1.5 or 2 dB. Knowing this, it is important to maintain a ceiling of at least 1.0 dB (True Peak) for your mastering maximizer/limiter when exporting for lossy encoding, which includes mp3, AAC and use in streaming videos.



Ozone Maximizer

A Ceiling set to -1 dBFS

B True Peak is On

Optimize for Online Distribution

Streaming and download are the two most likely ways your music will be distributed online. In most cases, your music will be available for streaming on services like Spotify, Apple Music, Tidal, Deezer, etc. For purchased music, you will most likely use a service like iTunes, Amazon Music, 7Digital, eMusic, BandCamp, etc. For those of us who are not signed to a major record label, we must use aggregators, like Tunecore, Distrokid, to submit our music to streaming and sales platforms. While most aggregators allow you to upload almost any version of your song (mp3, .wav, flac, etc.), it's best to provide a version that will play well on all the sites.

Each streaming site measures the loudness (LUFS) of your song and matches your song's level to that of all the songs on the platform. In order to avoid processing your audio, most songs are simply lowered in level to meet a specific LUFS measurement. However, not all services use the same LUFS standard and not all sites measure loudness the same way. So what are we to do? For better or worse, we don't have much of a choice here. When using aggregators, we are able to provide one master that gets distributed to all the platforms. But really, that's okay as long as our master is competitive with other songs in our specific genre. Everyone's song will be treated the same way on each platform, so the playing field stays level.

Optimize for iTunes

The iTunes store is an exception to the above description. iTunes will currently accept MFiT (Mastered For iTunes) masters from independent artists via Distrokid. The deal with MFiT is that an MFiT certified mastering engineer will provide iTunes with a 24 or 32 bit .wav file. MFiT also specifies that you leave some headroom for mp3 encoding. To become certified in creating MFiT masters, take some time to read and understand the Mastered for iTunes guide from Apple.

Mastering Plugin Resources

It is important to know exactly how a particular lossy encoding or streaming platform will change the level and sound of your master.

To that end, the following products allow you to audition your master in various encoded formats to check for loudness problems on many platforms:

- Ozone 8 Codec Preview;
- Nugen Mastercheck;
- Expose by Mastering the Mix;
- Sonnox Codec-Toolbox.

CDs, Vinyl and Online HD

While it's becoming less and less common for independent artists to manufacture and sell CDs, the idea may still appeal to many, especially those of you who play live shows or have an audiophile audience. When preparing to manufacture CDs, make sure to print the final master at 16bit, 44.1 kHz. If you intend to press vinyl, contact the pressing plant or cutting room to see if they would prefer an unmastered or a 24 bit file instead of the 16 bit CD master. Vinyl cutting often works better from a master that hasn't already been maximized for CD.

If you are pressing compact discs, be sure to check with your CD plant to see if they require a Red Book audio CD, disk image file or a special DDP file for duplicating or replicating your CDs. The plant will also need your artwork delivered in a specific format, so make sure to get that info before you complete your graphic design elements.

Online sales of uncompressed .wav files seems to be available from only a few sources, like HDTracks, AIX Records, Chesky Records and artist sites like Bandcamp and Soundcloud.

These sites will sell the highest resolution version of the music you can provide. Even if you provide uncompressed files to these sites, you should be aware that many listeners will still convert the downloaded files to mp3, so it is still advised to leave 1 dB headroom on your .wav file masters.

It's a Wrap

DIY distribution has become the norm and it behooves all of us to learn about the best way to create and distribute our masters. The business side of distribution and revenue collection is a vast issue, but creating masters that sound great on all the various platforms and media should become second nature. Remember that as streaming services mature, they may periodically update their delivery recommendations, so do your homework and try to stay on top of the state of things.

Chapter 2

Key Takeaways

- [ ] Set up your acoustic space to optimize for flat sound
- [ ] Listen at a calibrated volume in your room or headphones
- [ ] Produce dynamic, punchy mixes
- [ ] Check your masters for technical errors and noises on headphones
- [ ] Use reference material to make sure your masters sound competitive
- [ ] Understand your distribution media and their inherent qualities and flaws
- [ ] Stay abreast of the best practices for digital distribution services

If you're looking to get the best out of your studio, then try this...

Reference 4 software removes unwanted sound coloration from your speakers and headphones. Our Reference 4 calibration software makes your headphones and monitors coloration free, so you can hear what's really happing in your mix.

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