CHAPTER 14

How To Make Better Recordings in the 21st Century

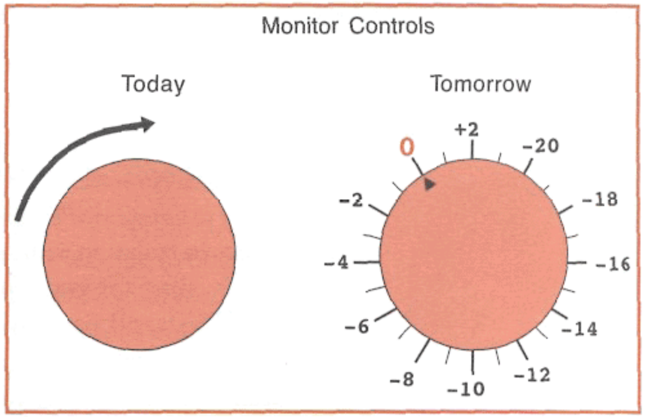
PART ONE: MONITOR CALIBRATION

I. Introduction

Calibrated monitors are the critical tools of the 21st century audio engineer. Some engineers think (mistakenly) that the need for monitor calibration is only for making of 5.1 theatrical mixes. But we'll all make better recordings if we use calibrated stereo or surround monitors. A good-sounding monitor system does not come out of the box, it takes work and care. But after the work is done, there's nothing like the pleasure of hearing great-sounding music!

What is a Calibrated Monitor System?

A calibrated monitor system is one that is adjusted to a known standard gain and frequency response. The monitor gain control is repeatable and marked in decibels. *Repeatable* means that you can return the monitor to a particular gain at any time, and *calibrated* means that the standard decibel markings on the monitor scale mean the same thing to any engineer, whether in Calcutta, New York, or Hong Kong .... This will help us collaborate, to be more consistent in our work, and to produce mixes that will perform together when later assembled at the mastering house. As we shall see, the absolute value of the numbers also defines the sound quality of the mix that will result.



Tomorrow's monitor control will be marked in 1 dB steps, and the 0 dB position will be calibrated to the SMPTE RP 200 standard (to be explained).

II. Getting Rid of Slippery Language

21st Century audio will be integrated with television, home theater, computer audio, computer games, and music playback, often all coming from a central source. During the last century most of us worked in uncalibrated listening rooms, adjusting our recording levels as we pleased, and just turning the monitor knob until it sounded "loud enough."

Try this: Put your favorite high-end effects movie into the DVD player, and adjust the loudness for a big, enjoyable presentation. Next, put one of last year's hypercompressed pop-music CDs into the same player. Watch out when you hit PLAY, because the loudness will be overbearing and in danger of damaging components and your ears. No wonder the consumers are beginning to complain. We can no longer produce recordings in isolation without regard to monitor calibration, since the same consumer equipment that plays DVDs will also play compact discs, videos, MP3s, DVD-As and SACDs!

"**Level** *is often confused with* **Gain***!*"

This is why, in the 21st century, we need to learn how to adjust our monitor gain first to a known standard, and then make the recording fit to that gain. One obstacle is the slippery daily language that we use to describe audio.

So to avoid confusion, the first step is to pick words that mean the same thing to everyone. Here is a brief glossary of the language of levels:*

VOLUME... **usually associated with an audio level control, is an imprecise consumer term with no fixed definition.** The words more properly used in the art are **Intensity** and **Loudness.**

INTENSITY... (aka SPL, Level, Pressure) a measure of the amplitude or energy of the physical sound present in the atmosphere.

LOUDNESS... is used specifically and precisely for the **perceptual level created inside the listener's brain**. Psychoacousticians can create subjective experiments that measure loudness, and have found that loudness versus intensity is quite similar across a population of listeners. However, *loudness* is much more difficult to measure in a metering system, in fact, it's best presented as a series of numbers rather than as one overall "loudness." Because of the big difference between typical metering systems and our perception, two pieces of music that measure the same on an SPL or VU meter can have drastically different *loudness*, depending on many factors, including transient and frequency response, and the duration of the sound. Exposure time affects our perception; after a five minute rest, the music seems much louder, but then we get used to it again—good reason to keep a sound pressure level meter around to keep us from damaging our ears.

LEVEL... is a measure of intensity, but when used alone means absolutely nothing, because it can

* Thanks to Jim Johnston (in correspondence) for helping to clarify some of these definitions.

mean almost anything! To avoid confusion, always accompany *level* with another defining term, e.g. *voltage level, sound pressure level. Level* is very often confused with *Gain*. Engineers can have a whole conversation about "levels" and not even know what they're talking about, unless they clearly distinguish *gain* from *level*.

SOUND PRESSURE LEVEL *(SPL)*... is one of the units of intensity. SPL measurements can be repeatable if taken in the same fashion.* 74 dB SPL is the typical sound intensity of spoken word 12 inches away, which increases to 94 dB SPL at one inch distance. While we often see language like *95 dB SPL loud*, this usage is both inaccurate and ill-defined as *loud* refers to the user's perception, and *SPL* to the physical intensity.

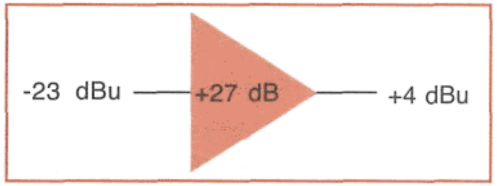
Decibels are always expressed as a ratio

A **decibel** (dB) is always a relative quantity; it's always expressed as a ratio, compared to a *reference*. For example, what if every length had to be compared to one centimeter? You'd say, "this piece of string is ten times longer than one centimeter." It's the same thing with decibels, though sometimes the reference is implied. +10 dB means "10 dB more than my reference, which I defined as 0 dB." Decibels are logarithmic ratios, so if we mean "twice as large," we say "6 dB more" [20 * log (2) = 6].

DBU, DBM, DB SPL, DBFS... are expressions of decibels with defined references. I believe the term dBu was introduced in the 1960's by the Neve Corporation, and it means *decibels compared to a voltage reference of 0.775 volts. dBm* means *decibels compared to a power reference of one milliwatt. dBFS* means *decibels compared to full scale PCM; that is, 0 dBFS represents the highest digital level we can encode.*

* SPL measurements must include the weighting curve used, e.g. A, or C, the speed of the meter (slow or fast), and method of spatial averaging (how many mikes were used and how they were placed).

GAIN or AMPLIFICATION... is always a relative term expressed in plain decibels, the ratio of the amplifier's output level to the input. It is wrong to use an absolute level (e.g. dBu or dBm or dBv) with the term *gain*. It is sufficient to say that an amplifier has, for example, +27 dB gain, and a nominal output level of + 4 dBu when fed with a given level source, as in this figure.



The meaning of Gain vs. Level. An amplifier with 27 dB gain is fed an input signal whose level is -23 dBu to yield an output level of +4 dBu. The decibels of gain should never need a suffix.

MONITOR GAIN VS. MONITOR LEVEL. Similarly, the sound pressure level from your monitor loudspeakers is often confused with the monitor gain. In fact, the term *monitor gain* is so slippery that I have started using a much more solid term that everyone seems to understand: **MONITOR POSITION.** For example, we say "the monitor control is at the 0 dB position."

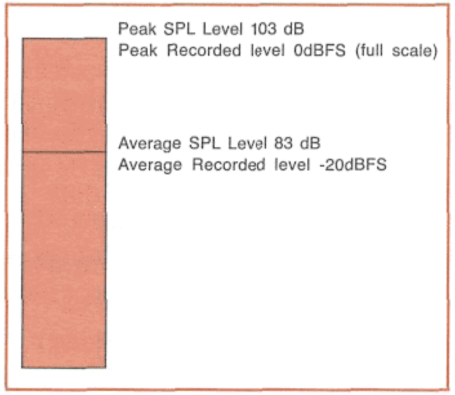
AVERAGE VS. PEAK. As we learned in Chapter 5, the instantaneous peak level of a good recording can be as much as 20 dB greater than its average (long term) level. Generally, we measure average sound pressure level with a sound level meter; sometimes we look at the peak level. For monitor calibration, the SPL meter should use the *RMS* averaging method, as opposed to a simple average (mean); simple averaging can produce as much as 2 dB error. Unless otherwise specified, when we say *average* in this book, we are referring to the RMS-measured level as opposed to the peak level.

CREST FACTOR is the difference between the average level of a musical passage and its instantaneous peak level. For instance, if a fortissimo passage measures -20 dBFS on the averaging meter and the highest momentary peak is -3 dBFS on the peak meter, it has a crest factor of 17 dB.

III. Using A Calibrated Monitor System for Level and Quality Judgment

An experienced engineer can make a good mixdown just by listening and without looking at the meter. The key is understanding how to use the calibrated monitor control. In simple terms, the monitor level control is calibrated so that the 0 dB position produces 83 dB SPL with a pink noise calibration signal (to be explained). The recorded level of this *calibration signal* is set to -20 dBFS RMS (20 dB below full scale digital). What this means is that a comfortably loud average SPL has been set to 20 dB below the peak system level. Since the ear generally judges loudness by average level, and the most extreme crest factor anyone has measured for normal music is 20 dB, then our peak level will never overload!* Typical mixed material has crest factors from 10 to 18 dB, so this mixdown may reach peaks from -10 to -2 dBFS, more than adequate levels for 24-bit recording, as shown in Chapter 5.

What this means is that a high monitor position will permit us to produce music with high crest factor. Conversely, as you lower the monitor control position, you tend to raise the average recorded level to produce the same loudness to the ear. In the 20^{th} century, we approached this from the opposite way; as we raised the average recorded level, we were forced to turn down the monitor to keep our ears from overloading!



When monitor gain is calibrated so average SPL is 83 dB at -20 dBFS, and you then mix by the loudness of the monitor, then the music will never overload and you will never have to look at a record level meter!

Monitoring by the numbers

Judging Loudness. If we become familiar with how various known recordings reproduce on our calibrated system, and the monitor position we use to reproduce those recordings, then we can judge the absolute loudness of any master in the making just by noting the monitor position, without having to compare it with other known recordings.

Judging Sound Quality. As the average level increases and approaches the peak level, more compression and peak limiting will be required to keep the medium from overloading. As we described in Chapter 10, some amount of *compression* can enhance a recording, but extreme compression is self-defeating, it lowers the crest factor and dilutes the clarity, impact, spaciousness,

* Assuming the mix engineer's ears have normal sensitivity to loud sounds. While no mix engineer works without glancing at the peak meter, you get my point.

and liveliness of the presentation. It's ironic that mastering engineers are being asked to do some damage to recordings in the name of loudness. Of course, the point where damage occurs is subjective and depends a lot on the music and the message, but we all agree there is such a thing as too much.

Work to a predetermined and fixed monitor gain. In the 21st century of mastering, we should work to a predetermined and fixed monitor gain; if the music becomes too loud, turn down the amount of processing or the output of the processors rather than turn down the monitor! We should use the measured *position* of the monitor control as a guide to the sound quality we are probably going to produce. In other words, if we find the monitor control drifting down too far, our recording is also probably deteriorating. 0 dB position is typically necessary to reproduce audiophile classical and acoustic jazz recordings that have used no compression or limiting. I've found that -6 dB position (corresponding with a crest factor of about 14 dB) is the lowest monitor gain that still produces a high-quality musical product with typical pop music, and most of the pop music recorded in the last century until about 1993 sounds "just right" at the -6 dB position. Slowly but surely, as we are forced to turn the monitor below -6 dB to keep a comfortable loudness, the sound quality is reduced. By working hard, I can make masters geared for -7 or -8 dB monitor position that still sound pretty good.* But some current hypercompressed pop CDs exceed this loudness by as much as 6 more decibels!

* Some monitors are marked in "SPL," which designers think is very sophisticated. However, it's very misleading. **This is a classic case of confusing gain with level.** The 83 marker is meaningless after calibration.

Monitor gain for mixing versus mastering. Mixing and mastering should be collaborative processes. I recommend that you be conservative with average levels during mixing, so as not to deteriorate the recording, for we cannot restore quality that has been lost. When mixing pop music, set your monitor position from 0 dB to no lower than -6 dB to make a recording that falls in line with the vast majority and still has good clean transients; it will help you produce a recording with life and acceptable dynamic range for home and car listening. You will still be able to be creative with compression and other effects—a fixed monitor gain is liberating, not limiting. When such a well-made recording arrives for mastering, we have much more freedom; we will raise the apparent loudness if we can do so while preserving or enhancing the recording's virtues, but the clarity and beauty of the recording will not have been ruined prior to arrival at the mastering house.

{ *A fixed monitor gain is liberating, not limiting.* }

Different Size Rooms. Note that room volume and number of loudspeakers affect the apparent loudness of a system. The more loudspeakers, the louder the system for the same monitor control position. I determined these recommended monitor control positions in a large stereo mastering room with loudspeakers 9 feet from the listener. In an extra large theatre, as much as 2 dB additional gain may be needed, whereas in a small remote truck with loudspeakers a couple of feet from the listener, as much as 2 dB less gain may be necessary. Set your standards accordingly.

IV. Setting Up and Calibrating the System

Summary of Essential Tools

Now that we know the benefits of having a calibrated monitor, let's see what tools we need to construct a good-sounding, calibrated monitor system.

- A great room, whose dimensions, wall construction and layout have minimal obstructions/reflections between the loudspeakers and the listener, with low noise and *good isolation from the outside world.*
- For surround sound, five matched "satellite" loudspeakers and amplifiers with flat frequency response (preferably good down to 60 Hz), high headroom, each capable of producing at least 103 dB SPL before clipping. To repeat the adage from Chapter 6, *high headroom monitors are necessary* to make proper sound judgments: if our monitors are compressing, we cannot judge how much compression to use in the recording.
- One (preferably two) subwoofers, capable of extending the low frequency response of all the satellites down to about 25 Hz, and producing at least 113 dB SPL at low frequencies before clipping.
- A low distortion monitor matrix with versatile and flexible bass management, capable of repeatable, calibrated monitor gains, and of down mixing and comparing sources from 7.2 through mono. With this, we can confidently produce recordings that can be interchanged with the rest of the world, and sound wonderful on systems large and small.
- A monitor selector to feed the matrix, with both digital and analog inputs.
- Measurement/calibration equipment:
  Preferable: A calibrated 1/3 octave real time analyzer (RTA) and microphone(s), with multiple memories, selectable response speed, and ability to integrate several microphone locations (spatial averaging).
  Alternate (less accurate): A high quality sound level meter with calibrated microphone, selectable filters and response speed.
  Test Signals: If using a sound-level meter, then you need RMS-calibrated sources of **filtered pink noise**. If using a 1/3 octave RTA, then you can use ordinary wide-band RMS-calibrated pink noise.
- **And let's not forget the most critical ingredient: Knowledge. The services of a trained acoustician may be needed on first-time setup,** to perform anechoic and early-reflection analysis of the room and loudspeakers, interpret the causes of measured frequency response errors, their audible significance, and suggest acoustically-based cures.

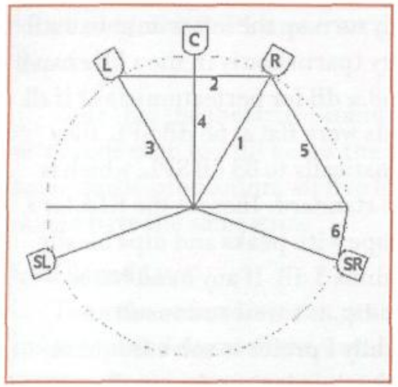
Placing the Main Loudspeakers

The ideal reproduction system should have no obstacles in the path between all the loudspeakers and your ears. This certainly turns most recording consoles and outboard racks into serious problems and is the reason why my rack gear is in the back corner, and my listening couch is placed in **front** of *the computer and DAW. This forces me to go behind* the ideal listening position when doing heavy editing, but all critical listening and remote control of transports and processors can be accomplished from the couch where there is little or no acoustical interference between loudspeakers and ear.

The Rope (Clothesline) Procedure

Tom Holman* describes how two pieces of string can be used to set up your monitors at the proper distances and angles to conform with the ITU 775† recommendation, illustrated below.

Here's a step-by-step embellished recipe. All speakers are equidistant from the center of an imaginary circle, with the center front being 0°, front left and right speakers at +/- 30°, and the surround speakers at +/- 110° (ITU accepts surrounds between 100° & 120°). Start with a long piece of rope or clothesline (which doesn't stretch so easily) a little longer than 3 times the length of the proposed distance to one loudspeaker. Tie one end to a mike stand located at the center of the circle (the prime listener). Run the rope to the approximate proposed position of the right front speaker, and



The ITU 775 recommendation for 5 channel loudspeaker placement.

put a piece of black tape on the string to mark the radius of the circle (see **1**). Then fold the long rope at the tape and add two more pieces of tape to mark three identical length sections. This radius is our "standard length," and equals 60° of angle when it runs between two points of the circle.

Spread the marked rope to create an equilateral triangle (see **1, 2, 3**), and now mark the floor at the points for the left front and right front speakers. Cut the rope at the first tape to leave a radius that can swing from the central mike stand. To find the center speaker location, fold a standard length of the remaining rope in half and mark its midpoint. Use that rope to find the midline between the LF and RF speaker and temporarily mark the floor there. Then cross the radius rope over this centerline and mark the position for the center speaker at the end of the radius rope (see **4**).

How to find 110° without a protractor? Use a standard length rope reaching from RF (see **5**) and temporarily mark the spot where it meets the radius rope. This is at 30°+60°= 90°. Now divide a standard length rope in thirds (see **6**), run it from the 90° spot and mark where this 1/3 distance meets the radius rope. This is 90°+20° = 110°, for SR. Do a mirror image of this procedure to find SL, and you're done!

Physically place the subwoofers just in front of, and slightly outside the centerlines of the satellites. Later you may "tweak" the position of the subwoofers for the flattest response at the listening position and best integration with the satellites.

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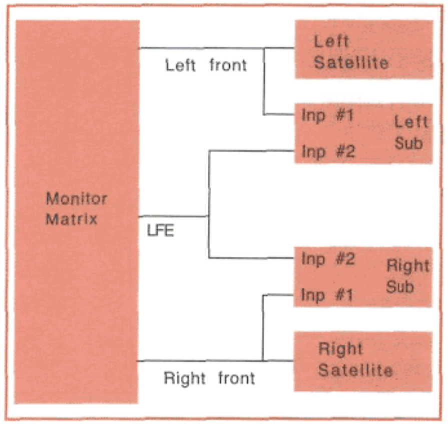
* Holman, Tomlinson [2000] *5.1 Surround Sound: Up and Running*. Focal Press.

† International Telecommunication Union, specification ITU-R BS.775-1

Connecting and calibrating the system levels

The 5.1 monitor system has six outputs, which should be connected to the inputs of the corresponding loudspeaker/amplifiers. I'm going to be describing a system using true stereo subwoofers. One way to connect such a system takes advantage of a subwoofer with two inputs (which most of them have), as illustrated below. You will be using some of the bass management built into the sub and some built into the monitor matrix.

You will choose the low-pass setting on the subwoofer which produces the most seamless "splice" to the satellites; ideally as low as 40 Hz, but some systems need as high as 80 Hz. This depends on the low frequency response of the satellites.[1] Start with the frequency recommended by the manufacturer and later you can tweak according to your room response measurements, as I will explain. Set the woofer polarity to normal and the initial phase setting to 0 degrees (if the woofer has a continuous phase control). The phase control on the subwoofer lines up the apparent distance of the sub with that of the satellites. Leave the woofer phase at 0° if your monitor matrix has delay compensation—if the sub is closer than the satellites, add time delay to the sub based on 1 ms = 1 foot. Later this can be fine-tuned, preferably using time-delay spectrometry, or the real-time analyzer. If your room geometry does not permit the surrounds to be the same distance from the ear as the front speakers, then you can delay the appropriate sets of speakers to match.



Connecting a monitor matrix with stereo subwoofers. By using the dual inputs of each sub, we can still have a mono LFE signal (the .1 channel) and stereo bass from the front main speakers.

Now let's check the integrity of each connection. **Turn the monitor gain control down all the way!** Feed a calibrated, **uncorrelated***, **5-channel** pink noise source at a level of -20 dBFS RMS into all digital inputs of the system, advance the monitor gain and the trim adjustment on each loudspeaker just a small amount to verify it's operating. Then, solo each output in turn and verify it's getting to the correct speaker.

SMPTE RP 200 Level Calibration

Now we'll be producing some loud test signals, so we suggest putting on earplugs. Place a calibrated measurement microphone pointing directly upwards, at ear height at the central listening position. Connect this to your 1/3 octave RTA. Set the RTA to an averaging time between about 3 and 10 seconds, and wait at least that long before taking any reading. **Turn the loudspeaker trim controls down all the way!** Set the master monitor level to the 0 dB (reference) position. Now, solo ONLY the Left loudspeaker. Slowly turn up the left trim gain until the midband energy (particularly in the 1 kHz band) reads 68 dB SPL (68.2 dB for perfectionists).[2] If all the individual bands were flat at 68 dB SPL, they would sum mathematically to 83 dB SPL, which is the SMPTE RP 200 standard. Inspect the RTA for a general smooth shape with peaks and dips ideally less than plus or minus 3 dB. If any band has a significant peak or dip, it's time to consult an acoustician! Generally I prefer to solve frequency anomalies with acoustic solutions first rather than equalization. Don't be concerned at this time about

*Uncorrelated means there is random, or no continuous relationship between channels. Correlated means there is some relationship. If the same, mono source is fed to all channels, then they are 100% correlated.

the absolute flatness of the high end, which will be rolled off.

Repeat this procedure for each of the 5 main loudspeakers, sending pink noise one channel at a time. If 68 dB is not an easy value to "read" with your RTA, then you may, for example, raise the pink noise to -18 dBFS RMS, which should result in 70 dB SPL per 1/3 octave band and (if all bands were equal) would sum to 85 dB SPL broadband. Remember, it's far more accurate to use the midband level measured with a 1/3 octave analyzer than a wideband SPL measurement, due to variations in microphone off-axis response, low frequency room resonances, filter tolerances, and so on. The alternative is to use a sound level meter with a band-limited 500 Hz to 2 kHz signal calibrated to -20 dBFS RMS, to read 83 dB SPL. If only full-range pink noise is available and an RTA is not available, an alternative method (though less accurate, with as much as 2-3 dB possible error) is to use a wideband SPL meter set to C weighting, slow response,

Note that the theatrical standard adjusts the surrounds each to 3 dB below the fronts, but for home music production, all five loudspeakers should have the same gain.

Total Sound Level

The subwoofers have not yet been calibrated and are turned down all the way. Five uncorrelated sources should sum approximately 7 dB higher than an individual channel. Release the solo button and verify that all five main speakers are operating, and the SPL in the midband rises about 7 dB (+/- 1 dB). If not, then one or more of your cables may be wired out of polarity, speaker distances or level calibration could be off, or a component is defective.

Phantom Center Check

Now let's check the **phantom center** produced by an in-phase mono signal when listening at the central position. This confirms the front main speakers are in polarity and there are no acoustic anomalies. Turn the pink noise off and turn the monitor control to about –10. Change the pink noise source to mono, that is, the same signal to all channels. Solo both left and right front loudspeakers. Now remove your earplugs, turn on the **mono** pink noise and verify the phantom center appears as a fairly narrow virtual image at the physical location of the center loudspeaker. You might tweak the angles (toe-in) of the speakers until the phantom image is narrow in the critical midband. If the image is off-center, recheck the left/right gains and speaker distances. Try tweaking one channel's trim up or down slightly to recenter the image, then return to the previous section and recheck the measured left/right gains to verify they match acoustically within +/- 0.1 dB in the 1 kHz band. Loudspeakers must be well-matched to produce an excellent phantom center.

Now compare the sound of the phantom center with that of the center speaker itself, by alternating between soloing the center or the two sides. The center speaker should sound a little brighter, but the position of the pink noise should not change if you are sitting in the center and all speakers are equidistant from the listener.

Bass Management

Integrating a subwoofer or pair of subwoofers to extend the response of a stereo system is an art and a science. Extending that idea to 5.1 is serious science, with its own set of compromises. We're going to start by creating and verifying an exceptional full-range 2-channel system, then extending it to 5.1. Since we are using stereo subwoofers, it is logical to set the bass level on a per-speaker basis, but the two subs couple with each other and the distances between them and from the walls affect the total bass response. It's not an easy affair, and you should approach it systematically.

Objective Subwoofer Measurement: Put your earplugs back on and send uncorrelated pink noise at -20 dBFS RMS to the LF system: left satellite and sub. Turn up the left subwoofer's trim gain until the RTA shows the low end is in the same ballpark as the rest of the frequencies. You may see amplitude anomalies near the splice point, indicating some parameters are not yet optimized. Then check the polarity of the sub; the position that produces the most bass is the correct one; if the result is ambiguous, temporarily set the sub's cutoff frequency as high as possible and recheck the polarity. The next part is the most time-consuming, where art and science really combine, for the ideal splice will happen only when the low-pass frequency, high-pass frequency, subwoofer amplitude, time delay and phase are just right. Take your time, "focusing" each parameter until the flattest response is obtained at the splice point. If you must compromise, remember, the ear finds peaks more objectionable than dips. Now take a spatial average of the response over a few listening positions around the sweet spot, and continue working until you're satisfied the left sub is integrated according to the RTA.

You may have to move the subwoofer around to produce the flattest extreme low end; the closer the sub is to walls or corners, the higher the amplitude of the deep low bass. If you move the sub, then you will have to readjust its time delay.

Next, if your room is symmetrical, it makes sense to try placing the right subwoofer as a mirror image to the left. Though occasionally, this is not a good idea if the subs both end up at the peak or null of a standing wave (expert acousticians apply here). Repeat the above process with the right loudspeaker system. Now send a mono pink noise source to all channels and solo both the left and right system (including the sub), turning the master monitor down until the 1 kHz band reads 68 dB, and see if the bass response with both channels operating is still within tolerance. Don't be surprised to see a heavier bass response than with the individual channel reading. If it rises, even as little as a dB, consider spreading the subs further apart to reduce their coupling, but then again, if they approach the walls, the low bass will go up from wall proximity. This interaction is at different low frequencies, so hopefully you will find a position with the least compromise.

Subjective Assessment, Stereo First

We have not yet set the bass management for the center speaker or the satellites, but now is a good time to check out the sound of the full-range stereo pair with bass management. It would be nice to discover a definitive piece of music that confirms your subwoofers are now perfectly integrated with the rest of your system. Since a subwoofer is not supposed to be a "boom machine" for most music, it really should be conspicuous by its absence rather than its presence. And that's the first way to listen. Listen to music with the subwoofers on and off. They should not feel "lumpy," they should simply add a sense of weight to the extreme low end. If the crossover frequency is 60 Hz or below, then you may hardly notice a difference except for the solidity of the sound. That's the way it should be!

Finding the right recording to evaluate bass is difficult because recordings of bass are all over the map. It could take days to check your subs by using a variety of recordings. An excellent way to evaluate a full range system is with a recording of a string bass whose level is very naturally-recorded. I have been using one of my own stereo recordings as a bass test record: my recording of Rebecca Pigeon, "Spanish Harlem" on Chesky JD115

This song, in the key of G, uses the classic I, IV, V progression. Here are the frequencies of the fundamental notes of this bass melody:

49 62 73

65 82 98

73 93 110

If the system has proper bass response, the bass should sound natural; notes should not stick out too far or be recessed. Start with the subs turned off and verify the lowest note(s) are a little weak. Then turn the subs on and verify they restore the lowest notes without adding any anomalies. Verify that the addition of the subs does not move the instrument forward in the soundstage (an indication the bass level is set too high) or become vague in its placement (an indication the subwoofers are too far apart). It's that simple. Then, take a break and enjoy Rebecca's performance for its natural acoustic reproduction of voice, string and percussion instruments, and the acoustic depth of a good recording hall. If you get this sound quality, then you are off to a good start with an excellent 2-channel stereo system.

Bass Management for Center and Surrounds

Our next job is to smoothly extend the low frequency response of the center and surround loudspeakers. Once again insert uncorrelated, calibrated level pink noise, with the master monitor to 0 dB position. Solo the center loudspeaker, and set the bass management to feed the low frequencies of the center speaker to the subwoofer(s). Adjust the highpass frequency of the center loudspeaker to the same frequency used for the left and right (if the center speaker is the same model as the sides). Then tweak the bass management level trim of the center (the amount of energy from center redirected to the subwoofer) until the total bass response is as flat as possible with the RTA. Determining a correct bass level from the two surrounds is a bit more complicated, since they are electrically summed into a single mono bass (unless the bass management is sophisticated enough to redirect the left surround's bass to the left sub and vice versa). Soloing each surround in turn, adjust the bass-management trim from each one for flattest response, then check the bass response from both surrounds at once with both uncorrelated and mono pink noise. Favor the response with mono pink noise since we are assuming that in typical music recording the bass will be in phase in both surrounds.

LFE Gain Setting

The LFE, or .1 channel is an auxiliary channel designed to increase the headroom of the bass channels. This is because when extra bass is desired below about 50 Hz, the ear (which is insensitive to bass) could require digital levels as much as 10 dB hotter than full scale digital! In a properly-designed 5.1 system, this headroom is taken care of in the design of the subwoofer. If in doubt, check with the manufacturer. To meet the RP 200 standard, the individual RTA bands **for the LFE channel only** should read 10 dB higher than the 1 kHz band. That is, 78 dB SPL if the 1 kHz band is at 68 with -20 dBFS RMS pink noise. **Solo** the LFE output and adjust the level of the LFE channel trim until the 50 or 63 Hz band reads 78 dB.

This completes the monitor calibration. Now you're on the same page as the most advanced 21st century mastering engineers. To speak the same language, tell all your fellow engineers: "My monitor system is calibrated with 0 dB reference SMPTE RP 200." Now sit back and enjoy your calibrated multichannel reproduction system!

V. Taking it Beyond: Monitor Equalization?

My philosophy is to avoid monitor equalization unless absolutely necessary. I believe that we should do everything possible to fix room-induced problems acoustically, and to relocate subwoofers and/or satellites if necessary for more linear response. Equalization, if performed, should be done by a skilled and experienced acoustician who understands the trade offs of electrically equalizing the direct response when a room anomaly is the root cause. When EQing, remember that the ear responds to the direct and room sound differently than an RTA. Finally, consider the tradeoff of additional noise and distortion if an equalizer is added to a system.

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1 If the satellites are good down to 40 Hz, so much the better, because the stereo imaging will probably be more coherent with a lower crossover frequency. However, when mastering for Dolby Digital, it is important to make a test listen with a mono crossover at 100 Hz to be compatible with consumer bass management systems. Many authorities recommend a 4th order (24 dB per octave) low pass on the woofer and a 2nd order (12 dB per octave) high pass on the satellites.

2 Holman shows an individual band SPL of 70 dB SPL, but note that this was taken with a pink noise signal of –18 dBFS. If the source noise is higher, then we must expect a higher output SPL. Measurements will be much more repeatable from room to room when you measure the 1 kHz band, as described in the text. So, determine the level to use when measuring the 1 kHz band by subtracting 14.8 (which all but perfectionists round to 15 dB) from the official broadband SPL. For example, if the source of pink noise is at -20 dBFS RMS broadband, the broadband SPL would be 83 dBC, and set the monitor gain until the 1 kHz band reads 68.2 dB. If the source of pink noise is at -18 dBFS RMS broadband, then the broadbadn SPL would be 85 dBC, and the 1 kHz band 70 dB (70.2). This is partly explained in a footnote to the SMPTE RP200 specification.

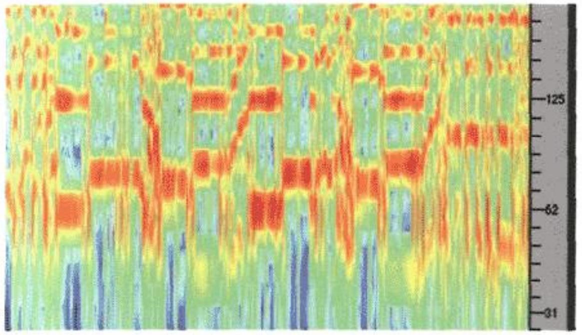


Figure C8-01: *SpectraFoo™ spectrogram of the bass frequencies of several measures from a rock piece. Read it like an orchestra score, time runs from left to right. Red represents the highest levels. Note the bass runs in the 62-125 Hz fundamental range are paralleled by second and third harmonics.*

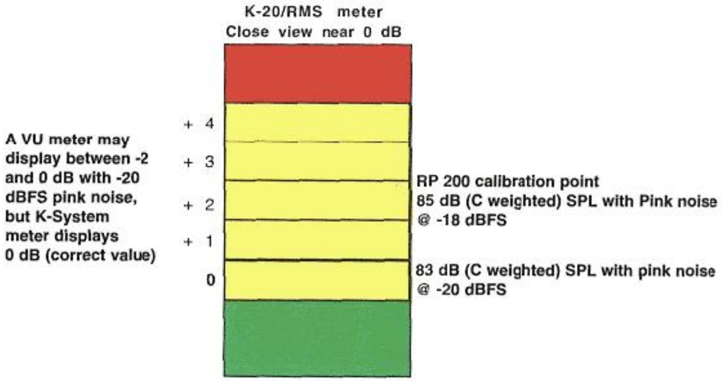


Figure C15-02: *A K-20/RMS meter in close detail, with the calibration points.*

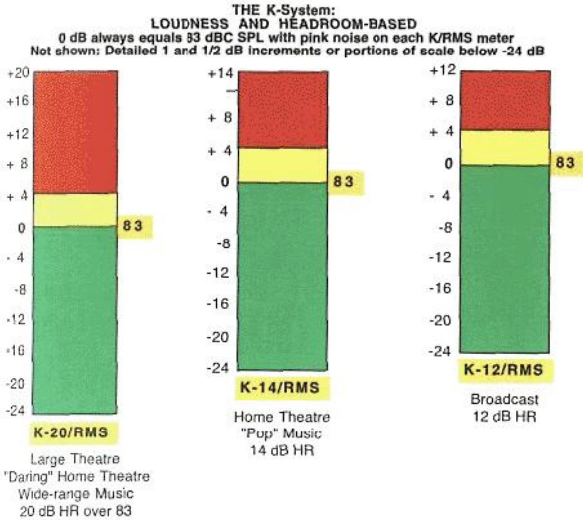


Figure C15-01: *The three K-System meter scales are named K-20, K-14, and K-12. I've also nicknamed them the papa, mama, and baby meters. The K-20 meter is intended for wide dynamic range material, e.g., large theatre mixes, "daring home theatre" mixes, audiophile music, classical (symphonic) music, "audiophile" pop music mixed in 5.1 surround, and so on. The K-14 meter is for the vast majority of moderately-compressed high-fidelity productions intended for home listening (e.g. some home theatre, pop, folk, and rock music). And the K-12 meter is for productions to be dedicated for broadcast.*

Figure C15-03: *A K-14/RMS Meter as implemented in Spectrafoo*

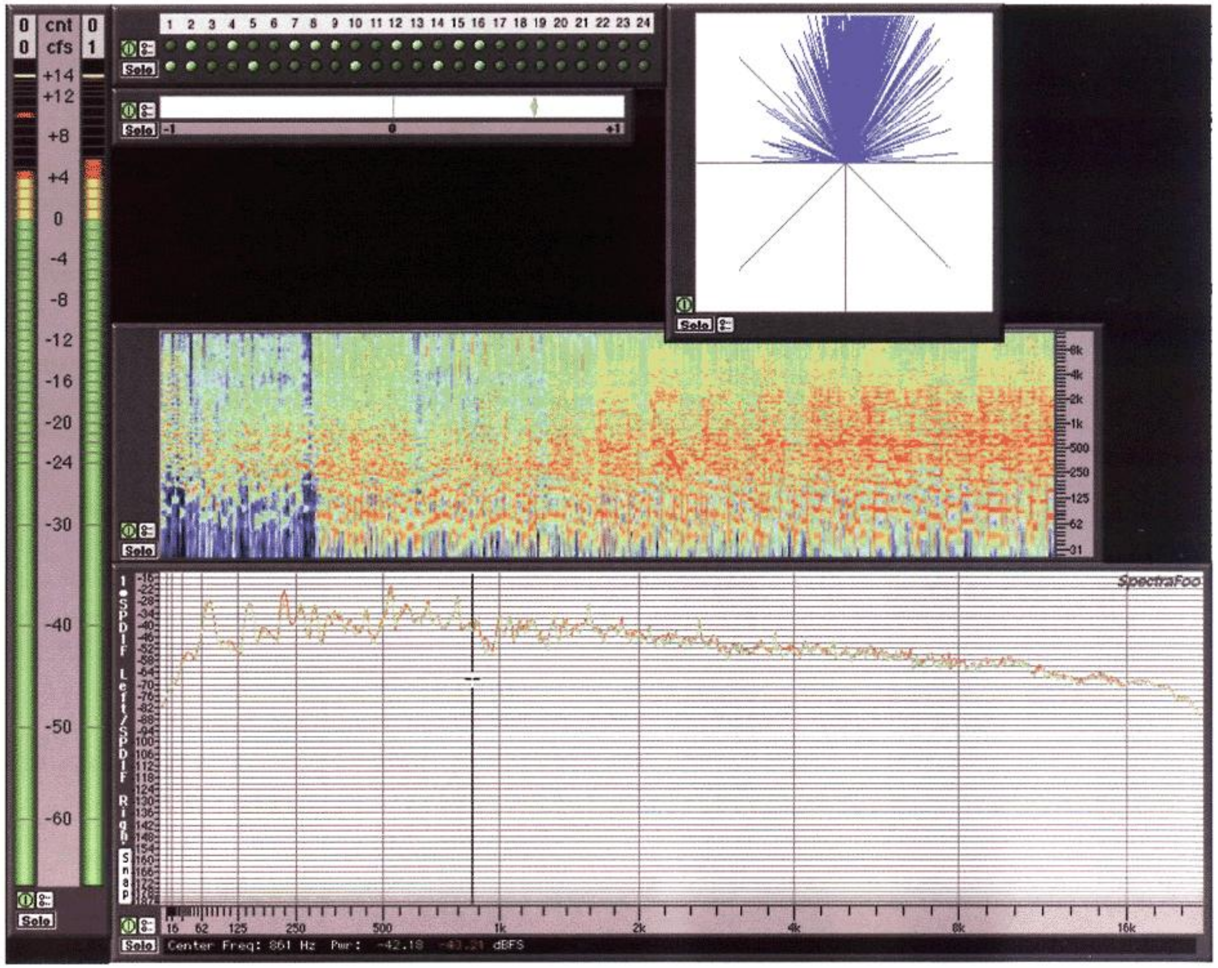


Figure C16-01: *SpectraFoo during a moment of musical action. From left to right at top: K-14 Meter, bitscope, and stereo position indicator. Directly below the bitscope is a phase/correlation meter. In the middle of the screen is a Spectragram, quiet section at left part, then the song begins. At top right is a stereo position indicator, and at the bottom, the Spectragram, left channel in green, right channel in red.*

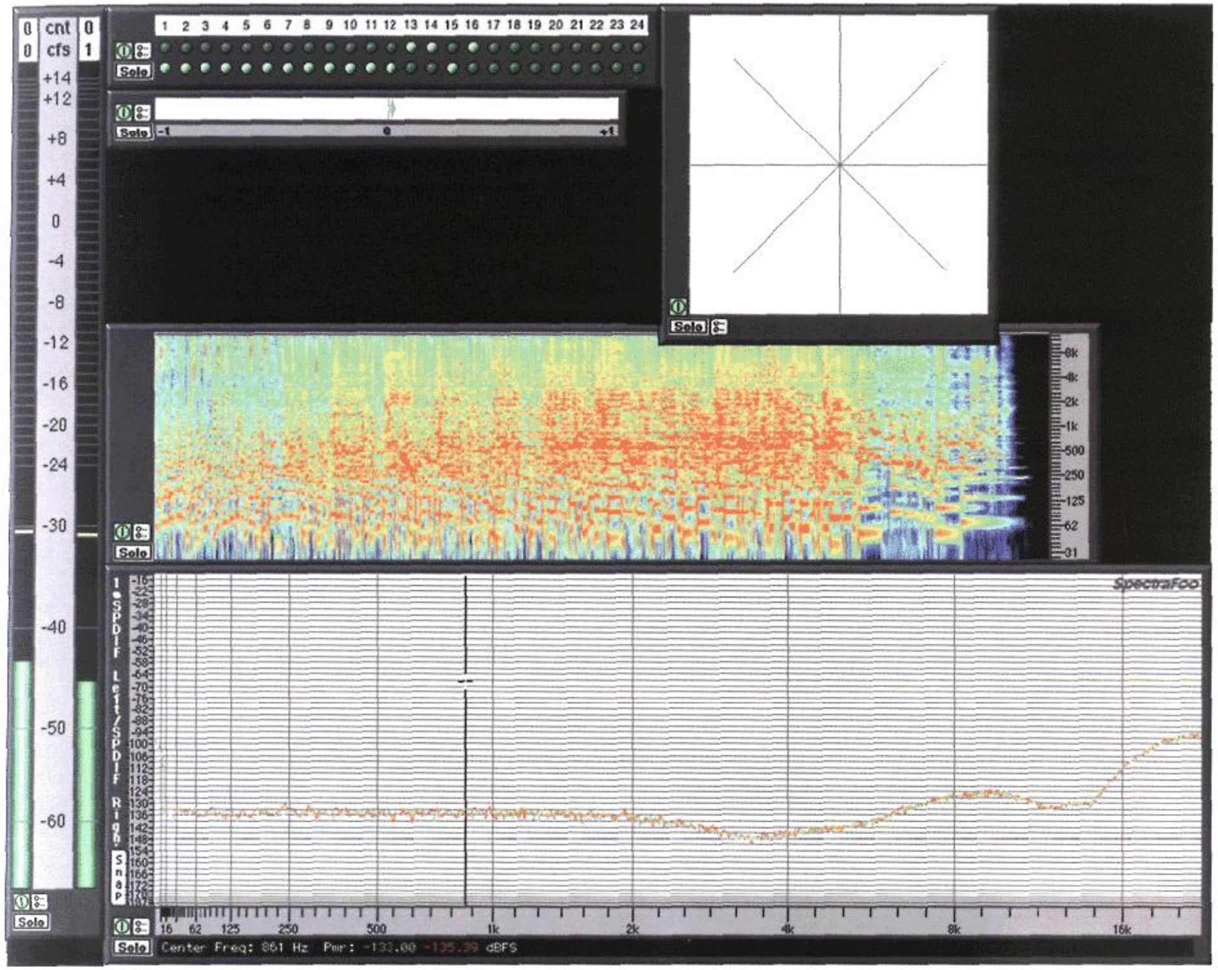


Figure C16-02: SpectraFoo during a pause in the music. Only the bottom four bits are toggling on the bitscope, and the characteristic curve of POW-R dither type 3 is revealed on the Spectragram. The last notes of the music "fading to black" can be seen at the right of the timeline on the Spectragraph.

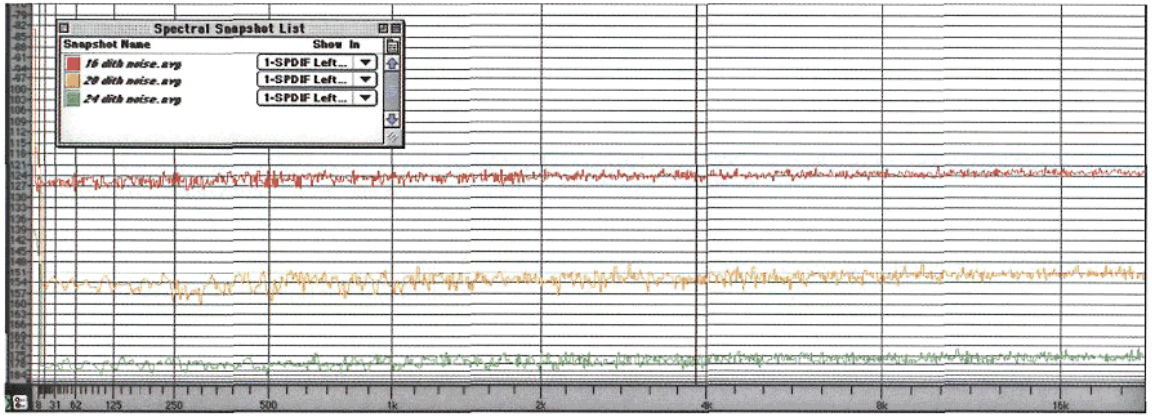


Figure C16-03: *Comparing 16, 20, and 24 bit flat-dithered noise floors (red, orange, green traces, respectively).*

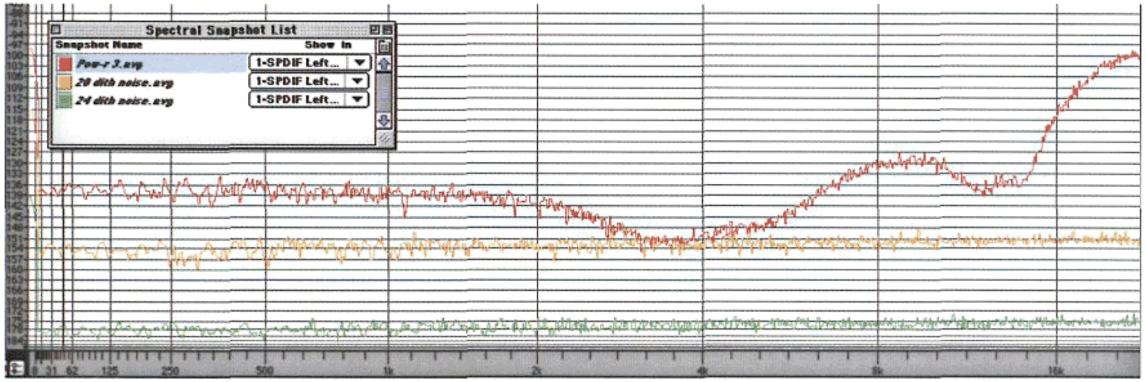


Figure C16-04: *POW-R type 3 at 16-bit (red trace) noise floor, with 20-bit flat dither (orange) and 24-bit flat dither (green) for reference.*

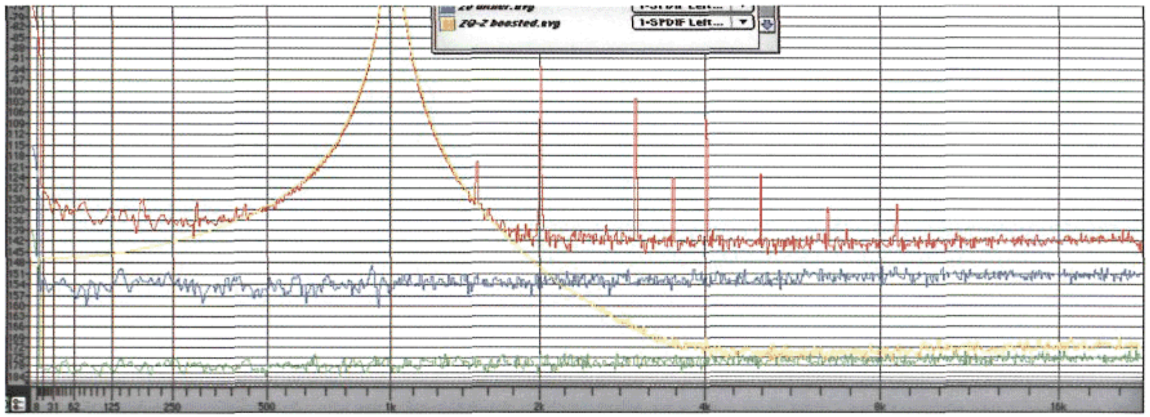


Figure C16-05: *Distortion and noise performance of Millennia Media NSEQ-2 analog equalizer in tube mode (red), 20-bit random noise floor for reference (blue), 24-bit noise floor (green), and Z-Systems ZQ-2 digital equalizer (yellow).*

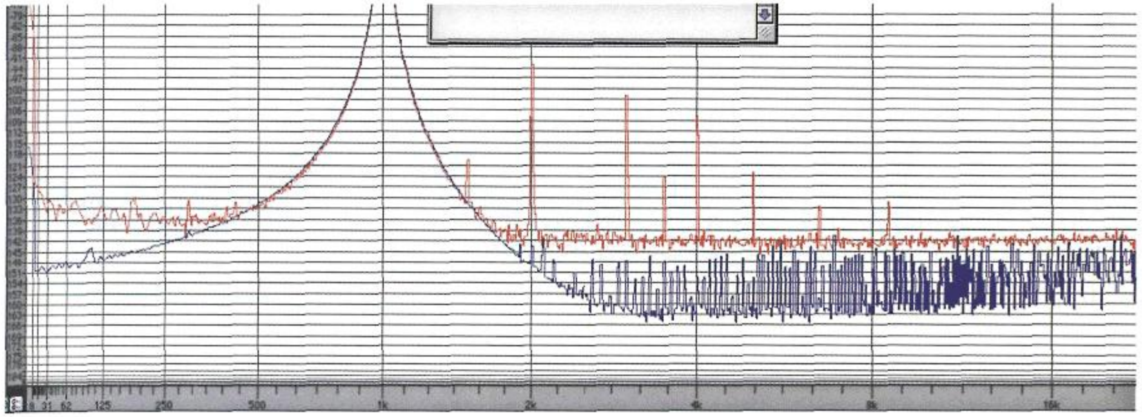


Figure C16-06: *Distortion and noise performance of analog Millennia Media NSEQ-2 (red trace), versus Digital Z Systems set to truncate at 20 bits, no dither (blue trace).*

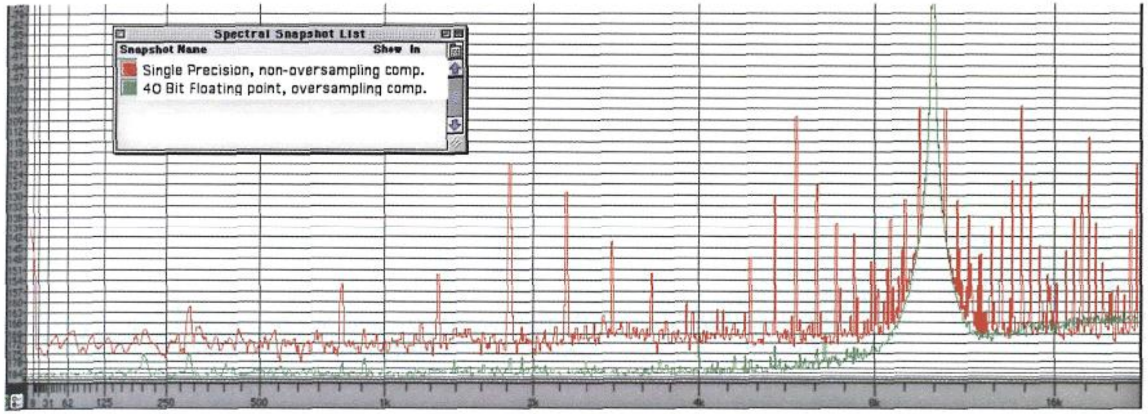


Figure C16-07: *Comparing two digital compressors, both into 5 dB of compression with a 10 kHz signal. Red trace: Single Precision, non-oversampling. Green: 40-bit floating point, double-sampling and dithered to 24-bit fixed level.*

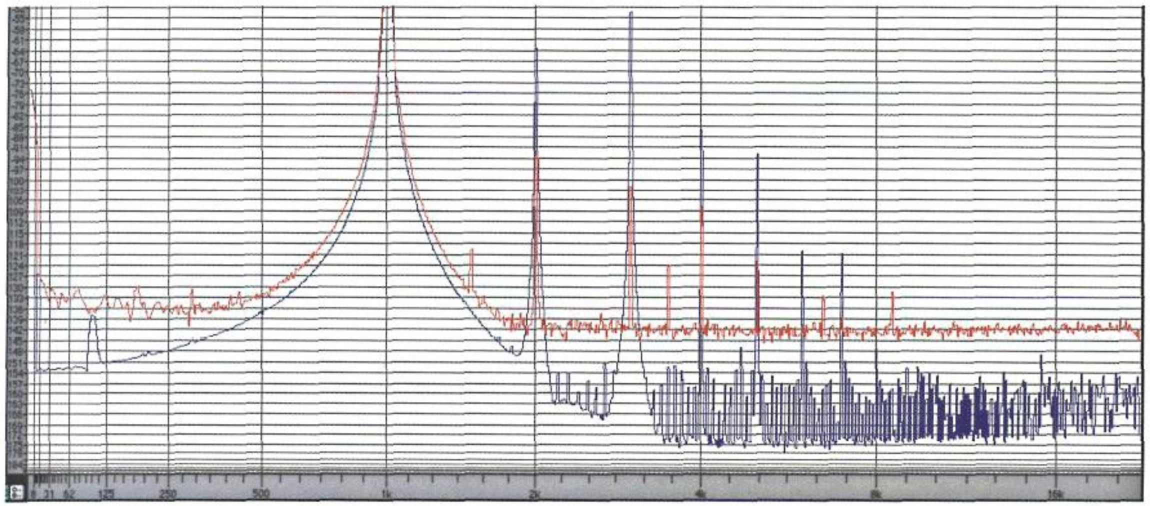


Figure C16-08: *Comparing Cranesong HEDD-192 digital analog simulator (blue trace) to NSEQ (red).*

Figure C16-09: A simple 10 dB boost applied in two different types of processors. In red, a single-precision processor, whose distortion is the result of truncation of all products below the 24th bit. And in blue, the output of a 40-bit floating point processor which dithers its output to 24 bits.

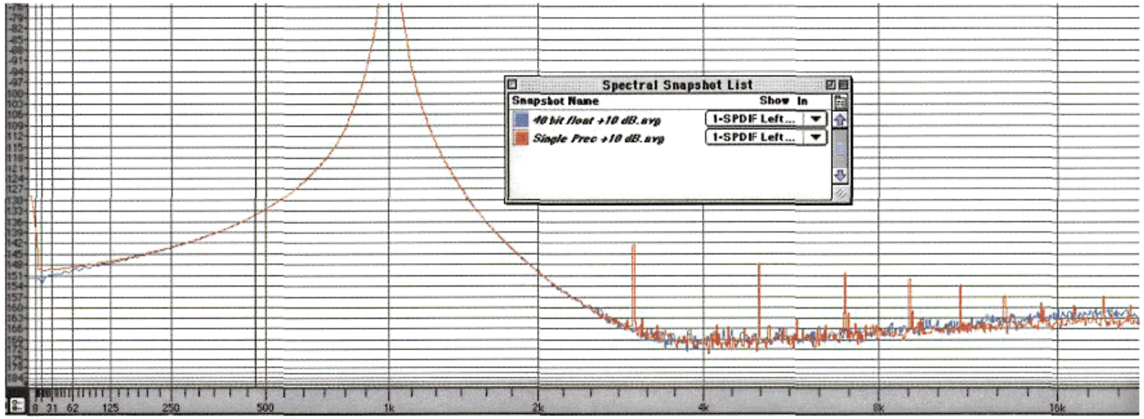


Figure C16-10: Compares two excellent-sounding digital dynamics processors, the oversampling Weiss DS1-MK2 (green trace), which uses 40-bit floating point calculations, and the standard-sampling Waves L2 (red), which uses 48-bit fixed point. The switchable safety limiter of the Weiss, which is not oversampled, is shown in orange.

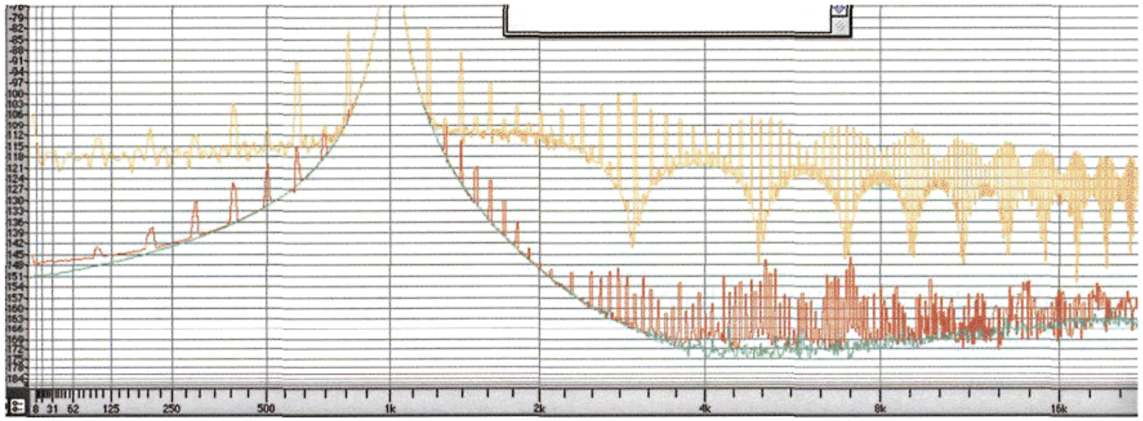
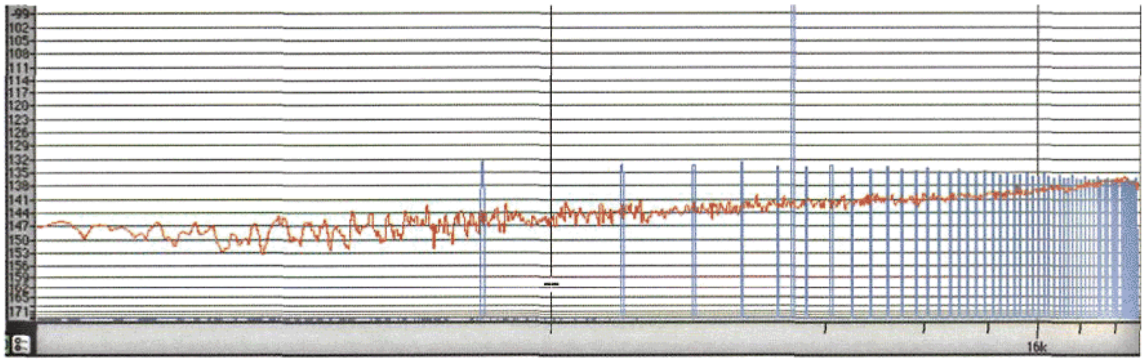


Figure C19-01:

Jitter testing:

16-bit J-Test signal (blue trace) overlayed with the Noise floor of UltraAnalog A/D converter (red trace) which together define the limits of resolution of my jitter test system.



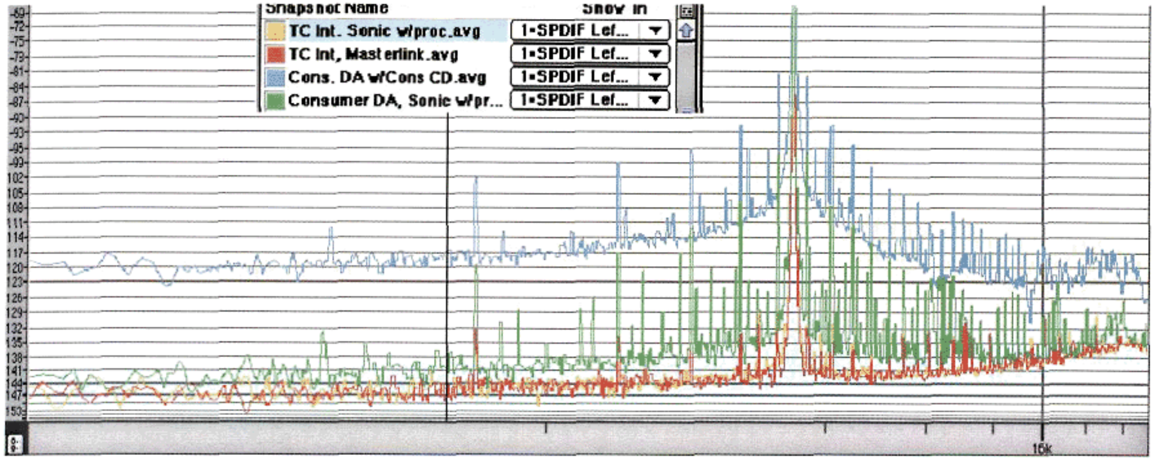


Figure C19-02:
Jitter measurements with J-Test signal:
Orange Trace: TC DAC jitter on internal sync, fed from Sonic Solutions.
Red: TC DAC jitter on internal sync, fed from Masterlink.
Blue: Consumer DAC fed from consumer CD Player.
Green: Consumer DAC fed from Sonic Solutions.

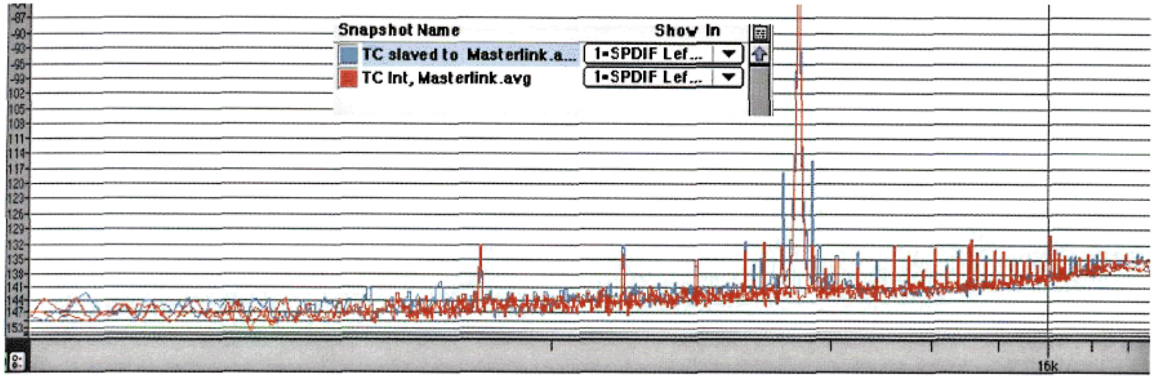


Figure C19-03:

Jitter measurements, demonstrationg how different clocking methods may produce different sound with the same source transport.

Masterlink transport feeding J-Test Signal to TC D/A.
Blue: TC D/A slaved to Masterlink transport via AES/EBU.

Red: TC D/A on internal sync.

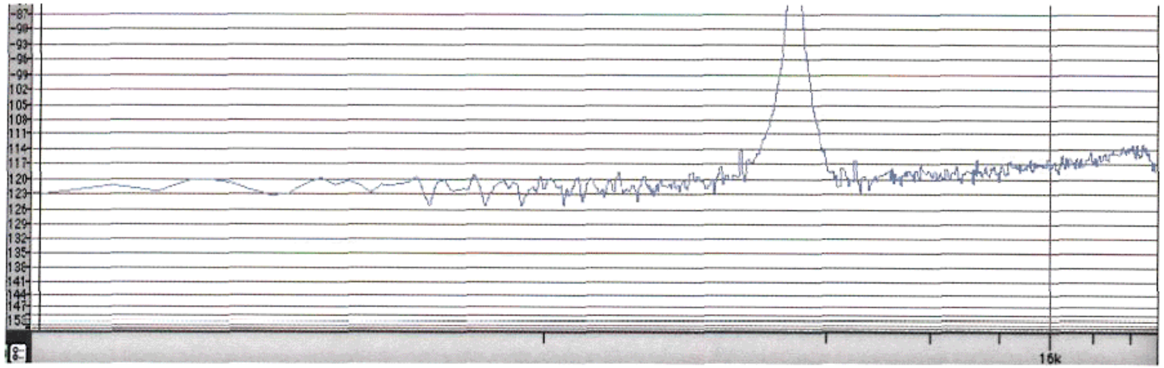


Figure C19-04:

Jitter Measurements:

J-Test signal feeding Weiss DAC on AES/EBU sync

View from the bridge. *Digital Domain's Mastering studio. Visible in front of the listening couch are: Rolling rack with Weiss EQ-1 LP Equalizer, Weiss DS1-MK2 dynamics processor, and Digital Domain DD-2 K-Stereo Processor; One pair of Dorrough meters; Reference 3A (satellite) loudspeakers on sand-filled stands plus Genesis Servo-controlled subwoofers.*