Recommended Equalization Procedures

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I have found it interesting that in the 29 years that I have been an acoustical consultant, most technical papers on equalization discuss the effects of proper equalization without discussing technique. It is also interesting that, without many papers on technique, there are so many firms that manufacture "equalizers" or equalization hardware. There are some consultants who offer an education program for this type of work, but the cost of this education is typically expensive, opinionated or limited in scope. I believe that it is more a disservice to the audio industry to provide so many tools to the trade without providing techniques for their use.

The goal of sound system "equalization" is the adjustment of the system by physical or electronic means to obtain a system capable of providing sufficient gain and sound quality to satisfy the client. The client may not be very critical of sound quality, or they may be very knowledgeable and selective, so the first step in equalization is to know the client's requirements before attempting equalization. Be aware that the designer or consultant, if they exist on a project, is not the client, but is a representative of the client. They were contracted to provide a system design concept that complies with the desires of the client. Consultants provide assistance in commissioning the system to ensure that the system provides the best quality possible and that the client gets what they purchased.

Read about Electro-Voice® equalizers:
EQ-131
EQ-231
EQ-215

In this paper, I will discuss some of the procedures used in the inspection, adjustment, and balancing of a sound system. Some aspects of this procedure will not apply to the various sound system types, but I recommend the reader examine each procedure to determine its application to their system. Masking systems are very different from stadium reinforcement systems, but both require procedures for successful equalization. Consider each procedure listed and determine how it may help you in completing your next sound system, no matter what type of system it may be.

Recommended Equipment

Before touching those equalizers, it is recommended that some basic tests on the system be made to ensure that wiring is in order and electronic products are operating properly. This procedure avoids the embarrassment of having the "commissioning authority" arrive to realize that a major piece of equipment has failed. This condition also avoids the added bonus of having to reimburse someone for the time lost or additional trips to the project.

Read about Electro-Voice® Dx38 and ACOne signal procesors.

The following test equipment is what I consider to be the minimum recommended for system adjustment:

1. Analog voltmeter with dB scale face.
2. Pink noise generator.
3. Audio oscillator (10 Hz to 10,000 Hz.)
4. Frequency counter (10 Hz to 10,000 Hz.)
5. Flat response omni-directional microphone.
6. 1/3-octave real-time analyzer.
7. Compact disk player with various well-recorded disks (country, classical, pop.)
8. Sensitive ears attached to an open mind (MOST important of all.)

The following are not considered acceptable versions of the above equipment:

1. Bob the Tweak. No matter what he says, get a meter.
2. Cheap noise generator where you can hear the noise repeating a "random" cycle.
3. Bob whistling—the frequency must be stable.
4. Bob guessing (I think that's 437 Hz…)
5. Any microphone other than a measurement microphone.
6. Heavy rack-mounted device with nixie tube display. (Aging myself, eh?)
7. No heavily processed music. (No Rage Against the Machine, except for limiter settings.)
8. "I'll fix it by equalizing it so it sounds right."

The following test equipment allows faster or more accurate system adjustment:

1. Hand-held 1/3-octave real-time analyzer.
2. Pulse generator and oscilloscope.

3. Time-energy-frequency (TEF) analyzer.
4. Computerized testing system.

I have found that an analog VTVM with a dB scale is the best meter to use in audio system level measurements due to the intermittent nature of the test signals (pink noise). Digital meters typically have a frequency limit of 1000 Hz which is not sufficient for the frequencies we are measuring.

Installer's Basic Tests

The following procedures are recommended in the following order:

1. Confirm the polarity of every driver that will be installed in the system in the shop before installation at the jobsite. There are some package polarity testers that will perform this test, but the best way is to observe a positive polarity pulse, and observe the waveform using an oscilloscope. The polarity of the first observation of the wave (small as it may be) defines the true polarity of the transducer. Measure one driver for each frequency range in this way and use it as a reference. Put the additional drivers adjacent to the reference driver and connect that driver in parallel to the reference driver with wiring connected in true polarity. Connect both drivers to a small amplifier and a pink noise generator. Reverse the polarity and the level from the two drivers should audibly reduce at a position exactly between both drivers. Make sure that the reference driver is properly wired, or you will have to rewire all other drivers to match the reference. Any drivers that reduce level when wired in polarity are not properly wired, and require correction. True driver polarity may also be lost, so confirm proper polarity before correcting driver wiring. Do this for all drivers used in the system. Package loudspeakers are also subject to polarity errors in wiring. Verify in a similar manner that all package loudspeakers are wired properly.

2. Verify that the wiring of the system has been implemented as shown on the designer's drawings. Notify the designer prior to their arrival of any changes to the original wiring scheme if it is required for the system to function properly. Discuss the variance and why the change was made to avoid surprises that may require rework.

3. Some consultants are "quirkish" about termination methods (I use 4.7K ohm 2 watt resistors on the outputs of all power amplifier channels). Provide termination resistors as defined in the construction documents. Resistors may seem labor intensive or inappropriate to the installer, but will mean a lot if a consultant requires them to stay up from 10:00 p.m. till 7:00 a.m. installing those resistors.

4. Turn off all power amplifiers (VERY important).

5. Insert constant-power-per-octave (pink) noise to a console input. Adjust the console output for a "0 VU" indication on the console. This output level could be anywhere from -10 dBu to +8 dBu depending on the console. Distribute this console output to all systems and sub-systems and adjust the output of the first electronic component in each sub-system to 0 dBu output level. If the first electronic component after the console has insufficient input headroom to receive this signal, insert a pad to reduce the input signal level to the desired 0 dBu level. Most all signal processing equipment has an output capability of +18 dBu, so this procedure will provide the electronic system with approximately 18 dB of headroom before clipping.

6. Set all equalizers flat with high pass and low pass controls, if available, so the equalizers have minimum effect on system response. Do not bypass the equalizers, for some devices have different gains when in the bypass mode. The jargon on these switches is always interesting (BYPASS IN... let's see, that means that the bypass is operating—right?)

7. Adjust compressors to be in the system (not bypassed) but set with a compression ratio of 1:1 and threshold controls if available for +10 dBu for minimum effect on system dynamics.

8. Adjust the output level of all line level electronic products to read 0 dBu at the output terminals. The only exceptions to this procedure may be electronic signal delays, digital signal processors (DSPs) or crossover networks discussed in detail below.

9. Electronic signal delays and DSPs have multiple input and output level controls. With 0 dBu input voltage, adjust the input control so that the input signal indicator (if present on the unit) flickers an indication of only 10 dB of headroom. Pink noise has a peak level or "crest" factor of typically 13 to15 dB, and with this input adjustment flickering at -10, there is approximately 20 dB of headroom for average program level when set in this manner. Adjust the output level controls to obtain as close to 0 dBu output level as possible. The internal gains of a digital delay or DSP sometimes does not allow 0 dBu output to be obtained. If this is the case, adjust the output level to the nearest 5 dB increment below 0 dBu (-5, -10, etc.). This 5 dB increment is not a necessity, but is easier to remember and document than various signal levels with limited effect to system noise.

10. Crossover networks by their nature divide the energy at their input terminals to two or more frequency ranges. With pink noise input band-limited from 62.5 Hz to 16 kHz and a crossover frequency of 1000 Hz, a properly terminated passive symmetrical crossover network would have the same voltage at its high frequency output terminals as its low frequency output terminals due to the distribution energy through a pink noise filter (3 dB per octave). Pink noise properly filtered has similar spectral characteristics to music, therefore similar levels would be approximated out of a crossover network only if the crossover was set to 1000 Hz. More output level would be present if the crossover frequency was lowered to 800 Hz. Some crossover networks compensate their gain to obtain similar voltage at all outputs. Other crossover networks sometimes do not have sufficient gain to compensate for narrow band-limited outputs such as those typical for mid-frequencies. Like the delay units or DSP, adjust the input level with 0 dBu pink noise input until the device indicates10 dB of headroom. After this has been completed, adjust the outputs to the nearest 5 dB increment at or below 0 dBu.

11. After making the above adjustments, turn down all power amplifier level controls and turn on the power amplifiers. Adjust the output gain of each low impedance (8 Ohm or lower) power amplifier to +4 dBu with pink noise input. Adjust the output of constant voltage (70 volt) amplifiers to indicate: +18 dBu, or approximately 10 volts on an analog meter. This voltage level is approximately 10 dB below their rated output voltage.

12. Replace the pink noise generator with a sine wave oscillator and adjust its output so that rattles or vibrations occur from the loudspeakers. Locate and correct any resonances or vibrations that occur at the offending loudspeaker. These rattles can cause immense slowdowns in commissioning, as damaged loudspeaker drivers can also sound like rattling chains. Fix those rattles before discussions regarding removal or replacement of drivers is suggested.

13. With pink noise source to the system, walk through the areas covered by the system or sub-systems and listen for noticeable shifts in frequency response or quality. Note if these differences are due to acoustical considerations such as major wall reflections or possibly due to driver wiring errors. In any areas of question, note the devices directing sound to that area and turn off all other devices in the system. In multi-way systems (bi-amplified, tri-amplified) listen to each spectrum (low, mid or high-frequency) individually and invert the polarity of one of the two devices covering that area. Verify the polarity of any offending drivers with adjacent drivers covering the same frequency range to similar areas, and modify any wiring errors. If the condition persists, note the area of concern and notify the designer prior to their arrival so that their time may be concentrated towards this variance.

With the completion of these procedures, the installer will have found and noted almost all of the system problems that may cause delays in system commissioning or equalization. Upon arrival at the jobsite the designer will be able to immediately start tests to obtain the maximum quality from the system.

Consultant's Tests

The following procedures are typically performed by a consultant on a consultant-designed project. These tests may also be performed prior to the consultant's arrival to speed up the commissioning process and expose any additional problems. We recommend the contractor consider these methods for design-build projects:

1. With pink noise input to the system, adjust the power amplifiers to obtain a consistent sound pressure level in the room at each frequency range reinforced by the system or sub-systems. Adjust power amplifier gains to obtain a reasonably flat frequency response in the room from 100 Hz to 3000 Hz. Overall levels for this test should be approximately 85 dBA.

2. Physically move or align all cluster devices to minimize spectral changes in areas covered by more than one loudspeaker. Center the analyzing device (your ears?) between the two sources observing the greatest destructive impact on the reinforced signal to align the loudspeakers. Alternating between Energy-Time-Curve (ETC) and Energy-Frequency-Curve (EFC) displays on a TEF analyzer are extremely fast ways to perform these adjustments. Use of the SMAART program also provides ease in measuring alignment and frequency response of the system. Without an analyzer, determine a physical position in the room that is not centered in the room, and not directly on axis with any loudspeaker drivers. Turn off all drivers except those directing sound to that position, and play music program through those drivers. Adjust the drivers physically to obtain best music and voice quality. Note the alignment between drivers, and repeat the physical arrangement with all other drivers in the cluster.

3. Electronically align all loudspeakers in a cluster with signal delays to improve voice and music quality and minimize interaction between loudspeakers. I have demonstrated that 1/8" variations in high frequency driver positions noticeably affect the system quality in the region covered by two loudspeakers. Alignment of clusters in this method results in minimum response variations within the areas covered by two devices. Alignment minimizes the area affected, and reduces the phase variations in that crossover region.

4. Turn off all power amplifiers except those powering the main loudspeaker system or cluster. Set the equalizers and filters to obtain "acceptable quality" sound in the space from the main loudspeakers. When complete, sample the sound with a spectrum analyzer with memory (or scratch pad) and save for later.

5. For each sub-system (such as choir monitor systems), turn off the main loudspeaker power amplifiers and turn on amplifiers for each sub-system individually. Adjust power amplifier level controls, then equalizers to obtain a similar frequency response from each sub-system as the main loudspeakers. After these rough adjustments, turn on the main loudspeaker power amplifiers and observe the effect the main system has on low frequency response. With these observations, adjust the sub-system equalizers to compensate for low-frequency build-up or other spectral changes in areas covered by two or more loudspeaker systems. Adjust monitor systems so that the sound quality does not change when walking from the main seating area into areas covered by monitor systems to prove to the performers that their systems have the same or better quality and loudness as any other seat in the house. This will reduce complaints when monitor systems are set properly.

6. Adjust signal delays for sub-systems using delays for signal alignment (such as under-balcony distributed systems). Adjust delays and power amplifier levels to obtain best source location effect in areas covered by these sub-systems. I have found that best results in delay systems have been achieved when the time offset of the distributed system matches the main system sound at the edge of the delay loudspeaker coverage nearest the main loudspeaker. Compensate also for different sound levels required for natural sound in the areas covered by the sub-systems. Under-balcony delay systems require proper delay adjustment and proper level reduction for different zones in the sub-system. An estimated drop in level of up to three dBA is usual for a deep under-balcony delay system. Use the expensive analyzer (your ears) to determine the best attenuation of the delay system.

7. Set compressors and signal processors to optimum effect. Compressors are typically set to provide no greater than a 3:1 compression ratio with the threshold set to the point where approximately 1 dB peak signal attenuation occurs with a 0 dBu pink noise input signal. Greater compression ratios or lower thresholds may be required for system protection. I recommend the use of a limiter to protect the system and allow the compressor to provide a minimum effect to system quality before protection is required.

Establish Effective Procedures

The above information is provided to inspire the sound system professional to initiate a procedure for setting up and adjusting sound systems in an organized fashion. Many testing procedures using more specialized equipment and computers is possible depending on the amount of equipment, time or money available to the project.

Much discussion regarding how to perform certain procedures is in order. I promote the sharing of all procedures that have been of value with all other people in the business. Re-invention of the wheel is not typically the most efficient use of resources, and sharing of any procedures that were helpful would help us all to do our work more effectively and efficiently.