Active loudspeaker crossover and EQ, a very practical FV-1 application

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Powered loudspeakers need sharp crossovers to remove the breakup region from the woofer, and protect the tweeter from damaging near-resonance excitation. Analog active crossovers have certain limitations, as even precision components produce slight variations from unit to unit on the production line. The FV-1 can be used to accept a mono input (both inputs driven in parallel), and provide separate low and high outputs, performing the crossover operation with digital precision.

Further, the location of the tweeter is often on the face of the speaker box, but the crossover frequency emission point on the bass driver is usually recessed somewhat. The FV-1 can impose a delay in the tweeter path to correct for the resulting time delay. The variables for the crossover and alignment delay can be formulated once, and used in each unit; the frequency and delay values determined by experimentation and listening tests. As a final touch, the bass end may be extended digitally. The code to accomplish these tasks can be programmed into an inexpensive EEPROM that drives the FV-1, and the same code can be used in each production unit.

This may produce a line of loudspeakers that perform well, but only if the drivers are perfectly uniform from unit to unit, and lot to lot. In reality, the amount of adhesive used in constructing each driver, the exact voice coil wire diameter, the diaphragm material thickness (affecting both mass and stiffness), magnet charge, amplifier gain, and many other production variables lead to a certain non-uniformity across a given production run, no matter how precise the *crossover* may be.

Loudspeakers all sound different, because they are *built* differently, with different materials, different drivers, box and port sizes, box corner rounding and so forth, but two speakers of the *same* kind, should be *identical*. The benefit of matched speakers is obvious, as in a 1dB loss in one tweeter over another. What is often overlooked however, is the loss of sharp *imaging* that results from driver variations. If you take a pair of your studio monitors and feed them with mono program, and the image is a sharp plane right down the middle of the stereo field, then you've got a pretty well matched set of speakers. If the image is blurred, which is most often the case, then you're most likely listening to a pair of poorly-paired drivers. Most studio monitors deliver a blurred 'space' of sound around the middle of the stereo image.

If the drivers are constructed from materials that are reasonably stable with humidity, using hydrophobic polymers or metals (not paper), then the opportunity to individually EQ each speaker on the production line exists with the FV-1, greatly improving the precision imaging capability of your product. The high pass and low pass crossover filters can easily be modified in both frequency and amplitude, but these simple variations can be augmented with precision parametric equalization. An example of an FV-1 program that allows bass extension, 24dB/oct crossovers, a time alignment delay and 7 bands of parametric equalization is provided in the example program area of the Spin Semiconductor website, as ACT_XOVER.spn.

Several issues need to be understood when designing such EQ corrected loudspeakers. First of all, the product most likely cannot be made *flat* over the entire audio band, as this would require excessive power to reach the very low frequencies. Build your speaker the way you like it, with crossover values that produce a nice, durable product, with *no* EQ; this is your product, be proud of it. To make more units that sound the *same* however, you will need to make subsequent units match the 'golden unit' through added EQ.

The spectrum of the golden loudspeaker will most likely not be perfectly flat, although through selection of drivers and crossover values you will most likely be targeting as flat a response as possible, down to the natural roll off at the bottom end. Further, it is not necessary, nor is it in any way practical, to test your speakers in a fully anechoic chamber. The whole point of EQ matching is to produce *sameness*, not flatness. Therefore, any test environment that is reasonably quiet and most importantly, *stable*, is sufficient.

A typical test setup can be as simple as a wooden frame, defining the edges of a 1 meter cube, supporting multiple layers of glass wool and heavy cotton fabric, leaving a port on one side in which the speaker is inserted and connected to power and test signals. It is extremely important however, that the speaker is inserted to the exact same position each time the process is performed! This can be done by placing the loudspeaker under test onto a platform that protrudes just within the chamber, and a set of stops that guarantee consistent positioning. A test microphone (nice quality, small condenser mic) is placed through the padding on the opposite side of the chamber, so that it is fixed directly in front of the speaker, between the woofer and tweeter, perhaps 0.5 meters (or more) from the face of the speaker. A greater distance would be nice, but this means a larger chamber, and less tolerance to external noises during the tests.

The speaker is then stimulated with an exponentially swept sine wave, traversing 20Hz to 20KHz in perhaps 2 seconds, while the microphone signal is recorded for analysis. Care must be taken to not overdrive the unit, as low frequency distortion components can adversely affect the measurements. Several sweeps can be averaged to better define the speaker's response, and remove the influence of that fork lift that occasionally rumbles by in the production shop. (And that guy pounding boxes together with a hammer!). The amplitude response of the unit under test is then compared with the stored response of the 'golden unit', whereupon a comparison can be made, and EQ parameters can be prescribed.

The analysis software for such a process is not a simple thing, and will be carefully written with concerns for each model in mind. The resulting frequency, Q and gain/loss for each band, as well as driver levels and so forth, can be downloaded directly to the device under test with a four pin connector; a ground line, two logic lines for accessing the EEPROM, and a toggle line to one of the program select pins of the FV-1, inciting it to re-load the EEPROM code. These control lines can have pull up resistors internal to the unit, and be available at the rear of the speaker for this purpose only.

The use of all 7 bands will probably be unlikely; go after the gross variations first, and only the little ones when they have a rather low Q. do not attempt to correct very high Q resonances, do this through a better selection of drivers! Sharp variations can change with time and variations in humidity, and are best left alone. The software can be human interactive, with an operator adjusting EQ bands manually, or fully automatic, in which case the software becomes significantly more complicated. In any case, program modifications can be downloaded quickly, and productivity can be high in a shop that produces perhaps 100 to 300 units a day.

It has been my experience that such matching of loudspeakers provides a remarkable listening experience, despite the room in which the listening takes place. It is as though our hearing mechanism can (to some limited extent) quickly remove the room, and leave us with the speakers and the music to enjoy. When the mono program, which is so often the main character in music productions, is right there in your face, you can reach out and almost grab it!

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Notice

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