GEORGE AUGSPURGER TALKS ABOUT "MONITOR SPEAKER VOICING: IS IT WORTH IT?"

The announcement for tonight's program gave us a title to the effect of, "Monitor Speaker Voicing: Is It Worth It?" From my standpoint as a listener, yes. It is definitely worth it. I think some of the things that can be accomplished with equalization of control room speakers, or hi-fi speakers in your home, are little short of miraculous. On the other hand, I do not pretend to be a professional mixer or studio engineer. I'm just one of several million listeners at the consumer end. And I think there is an area here where we have to respect the particular methods that a particular mixer or engineer is used to working with... he may prefer not to equalize the speakers he uses. Although we have done equalization of monitors in existing studios, we prefer to include the service as part of a program which involves studio design or remodeling.

As far as my own background in this field goes, about four years ago, I believe it was, Tom Hidley and I made a couple of experiments over at TTG. What we did was simply take two of the monitor speakers, do a little quick and dirty equalization on one of them, and then call in one of the mixers. We said, "Here, now listen to a couple tracks on this speaker and then listen to it on *this* speaker and tell us what you think." And his comment was, of course, that the tape sounded the way it should on the unequalized speaker, and didn't sound right on the equalized speaker. This was exactly what we had expected because the mix had originally been made on the unequalized speakers. Well, the next question we asked him was, "Alright, if you had to work on this *equalized* speaker, what would you do?" And the answer was that he would have mixed the final tape differently! If you have a technique which will actually force you to turn out a different product, then it's something that you simply cannot ignore, one way or the other.

So, at that point, Tom Hidley and I and Larry Phillips (then) of JBL began doing some experimenting with techniques that were available. Later, after Perception, Inc. was formed, we spent some time with Dr. Boner in Austin, Texas; we obtained a license for the complete Boner equalization process in various forms, and we gradually worked out our own method of adapting these procedures to playback systems.

We, in equalizing monitor speakers, normally feed one-third octave bands of noise into the system, sequentially, one at a time. The important thing about doing it this way, to me, is that you have a signal you can listen to, and correlate what you measure with what you hear. We do the actual measuring with Brüel-Kjaer one-half inch condenser microphones. These have the advantage that they are flat in the range of interest; you don't need to apply any correction to them. And instead of using a single microphone, we use three microphones in a simple little array, so that we can sum the signal level from three locations simultaneously and get a space average at the mixer's normal location. Because of the wide variations we were getting from point to point, we felt that we were able to get a more accurate picture by going through an averaging technique rather than using a single microphone.

What curve do we equalize to? Well, we talk to the client first. The final curve will depend not only upon the kind of work the client does, the kind of sound that he is used to, but also the limitations of the particular job. You run into situations where there's simply no point in trying to push low frequency response down below 63 Hz, let's say. Because to do it, you're going to blow out the amplifiers and speakers.

The curves that we try to arrive at are generally pretty flat. One of the things that came as a surprise to me, as a matter of fact, was that after taking a little time to go over the pros and cons of the various curves, the overwhelming majority of studio people say, "Let's start out with a generally flat curve, and then if we later want to change it, we can." And the great majority of them seem to be satisfied with what is essentially a flat curve after the equalization is done. The only deviation from this is at the extreme top end, which is almost always rolled-off to some degree. Typically, if the client has no preference, we will try to maintain the curve flat out to 5,000 or 6,000 Hz, with a gradual roll-off above this region.

Finally, after we've gone through the measurement routine, we then *listen* to the monitor speakers, first just comparing them with pink noise input. (We try to do this when the client is out of the room; no matter how closely matched two loudspeakers are and how much time you spend equalizing them, you'll *never* get them to sound *exactly* the same on pink noise on a direct A-B test. An A-B test allows your ear to make incredibly fine discriminations.) This is a test for our own purposes to see if they really sound close enough to indicate that the basic EQ has been done properly. If they do, then we listen to the two simultaneously through each of the individual one-third octave bands—and this separates the men from the boys. If the speakers are well-matched to begin with, and if you've done a pretty good job equalizing, you actually can hear a nice, solid, well-formed mono image sitting halfway between those two speakers all the way from about 200 Hz up. If you find the image shifting from one side to the other, then you forget what you've measured and adjust the filters until the image is as nearly centered as possible.

There are two magic factors which make this whole thing possible. One of them is the fact that the technique of making acoustical measurements in band widths nominally a third of an octave wide enables you to come up with a set of measurements which correspond very well with your subjective interpretations of the sound of those loudspeakers. This is primarily because the critical bands of the ear are very nearly a third of an octave wide. The other reason that this technique works is that in small rooms, the tonal balance you hear is always localized at the source. In other words, you do not and you *cannot* consciously separate out what the loudspeaker is doing from what the room is doing in terms of tonal response. In a typical listening situation in a home listening room, or a control room, the reverberant field is what you hear, in terms of tonal response. The reason I keep emphasizing tonal response is that there are other little clues that your binaural hearing give you, and anyone who says, well, you just go in with a pile of filters and you can make any room sound like any other room is wrong; you can't. But if you can get essentially the same subjective tonal response, then even though you can tell, you're listening to a 604 instead of a Tannoy and, even though you obviously know that you are sitting in a fairly live room instead of an anechoic chamber—the chances are that you will still arrive at essentially the same mix, and that's the important thing.

I have jotted down ten questions, complaints, criticisms, etc. that crop up again and again, so I'll go through these now and maybe give you some brief comments of my own:

(1) Altec says monitors *should* be voiced; JBL says their monitors *may* be voiced; Electro-Voice says their monitors don't need voicing; Tannoy says the whole thing is a silly American bunch of nonsense! First of all, Electro-Voice's basic point is well-made. If you are after a certain general tonal characteristic, it is obviously going to be easier to get if you start out with a speaker that pretty well approximates it to begin with. If you are lucky enough to pick exactly the right speaker and exactly the right room, then you don't need any voicing at all. I have some clients that feel this way very strongly, and I'm not about to argue with them. On the other hand, from my personal standpoint, I would feel better to assist the room acoustics to some degree.

(2) "I've heard speakers equalized, and I've heard speakers unequalized, and the equalized systems sounded worse."

There are, I think, four possible answers to this. Number one is simply the question of personal preference; maybe the equalized systems *do* sound worse to that person. That's one of the arguments you can't refute. Number two, perhaps the equalization has been performed incompetently. There are some people doing equalization who shouldn't be doing it—just as there are some piano tuners that should be locked up. The third thing is, even with considerable experience and confidence and the best intentions in the world, there is a strong subconscious pressure to believe the instruments, rather than what you hear. I think you continually have to try to find some way to resolve what differences there may be and to find out why, and what you can do to come to some sort of satisfactory solution. The fourth possibility is that there are some situations, very rare, in which equalization doesn't seem to work the way it should. I'm thinking of Benson Sound in Oklahoma. Larry Benson had me come down to equalize his Altec 9845's. Before I arrived he had experimented with placing those monitors in every conceivable place in the control room, which I think is a marvelous thing to do. Unfortunately for me, by the time I got there, he had found *the* place! And almost everything I tried with the filters made the 9845's sound worse—no question about it. So once in a great while, yes, that situation will come up.

(3) "Even after equalization, different control rooms still sound different." Yes they do; and, yes, they will. I've already mentioned this; the point is that the *tonal balance* should be nearly the same after equalization.

(4) "The monitors don't sound exactly the same, even after equalization." I have three quick comments here: Number one, I would like to emphasize, again, that any equalization techniques have built-in limitations. None of them are as accurate as your ear. The normal method of using third-octave bands of noise has a maximum confidence factor, of something like ±1 dB. Yes, it's a painful thing to try to explain to the client that those little 1-dB glitches that you've drawn for him really don't mean anything. You can take any curve that I show you and arbitrarily take any point in that curve and move it 1 dB one way or the other, and your curve, statistically, is just as valid as mine. And that's just one of the limitations we have to live with. The second thing is that when you say that two monitors don't sound the same which two are you talking about? I can get the front left and the front right awfully close. However, if I use the same measuring gear, and do the front speakers and then do the back two speakers of a quadraphonic room, the back speakers are not going to sound the same as the front speakers. The third point is that the closer you get two speakers matched to each other, the easier it is to hear the little differences that remain. A speaker that just has some blatant overpowering characteristic will sound very much the same in almost any room. You can identify it just like that. As loudspeakers get better and smoother, and more neutral in sound, it's much easier to pick up, in an A-B test, very small differences between them.

(5) People say, "Why should I have the systems equalized, because even after they're equalized, they tend to drift." This is true. It depends to some extent upon the particular speakers that are used, to a much greater extent, upon how hard they are used, and, it is a function of the loudspeakers, not the equalization process itself. The studios that run their monitors at a reasonable level stay reasonably stable. You can come back in six months, check the curves, and come out with something very close to what you had done previously. Studios that concentrate on rock, where people want to run the monitors at 110 and 115 dB—in these you can get everything tweaked up—come back two weeks later and find variations of 2 dB. And that's just the name of the game!

(6) This comes up occasionally when a fellow says, "Well, hey, now that I've paid for all this fancy equalization, when I put a tape on and switch the EQ in and out, I really can't hear much difference." Quite often this is true. You may find that the overall balance between lows, midrange and highs, has not been changed very much in the process of equalization—and in that case, you won't hear a dramatic difference switching in and out. But, still, I say that equalization is worth it because if I listened a little longer I would begin to hear the ranges of frequencies of the particular instruments, the particular type of program material, where the equalization really cleaned the thing up.

(7) This is one that comes up occasionally, not from the client, but from people that do equalization. They say, "I measured the curve and then I went back to the equalizers and measured their response electrically to develop an inverse curve. But when I equalized to the inverse curve, I didn't come out with flat response. Somehow the electrical measurements and the acoustical measurements did not agree." It isn't that there is something mysterious about acoustics or that rooms are nonlinear, it's simply inherent in whatever measuring techniques you are using. There are a couple of reasons that can be given for this.

One is that the filter shapes you are using for measuring and the filter shapes you are using for equalizing are different. That's probably the primary cause. The other is the statistical nature of the measuring system itself. You have to develop a feel for your measuring equipment and your equalization equipment to understand how all these things are interacting.

(8) "Alright, I did all the voicing, and my acoustical measurements show that I have a beautiful flat curve; but then I go through, and I measure a sine wave response of the filters, and it looks like a picket fence." Most of the filters that are used for equalization, are so-called minimum-phase devices; that is, for a given curve, the filters will insert the least amount of phase shift to provide that particular curve. But there is a clinker here; the curve that we're talking about is the actual sine wave electrical response of the filters. Remember, when you're doing acoustical measurements, you are averaging the entire response over roughly a third of an octave, and there are an infinite number of actual curves within that one-third octave that will give you the same total power within the band. If you aren't careful, you will find that you have inserted ripple to the extent of 3 or 4 dB, and all sorts of excessive phase shift simply because it didn't show up on your acoustical measurements. Is this good or is it bad? Well, you really don't know, do you? The assumption is that, all things considered, we're better off to get the curve that has the least amount of perturbations—the smoothest electrical curve that will still manage to get us the acoustical curve we're after.

(9) "All those filters introduce excessive phase shift." Actually, it isn't all those filters. The technique of using individual filters tuned at centers a third of an octave apart works out to be a very flexible technique, a very powerful technique. What you actually have when you get done, is *one* filter. With most available filter sets, these all combine into a single minimum-phase filter which happens to produce that particular electrical response. If you wanted to, you could get out text books and design a single filter that would give that response, but it's easier to do it by turning little one-third octave knobs individually. The phase shift is highly unlikely to be more than 45° at any point, and usually considerably less than that.

(10) "Which voicing system's best?" I would say that it depends upon the techniques that are congenial to the person who's doing the job. To me, my system is perfection, but I have seen many people work with other systems and do extremely acceptable jobs.

JBL

Reprinted by permission of George Augspurger.
This is a transcription of part of a panel discussion on "Monitor Speaker Voicing" held at the October, 1973 meeting of the Los Angeles Section of the Audio Engineering Society.
George Augspurger is associated with Perception, Inc., Los Angeles.
PIB1/73 Printed in U.S.A.
James B. Lansing Sound Inc., 3249 Casitas Avenue, Los Angeles, California 90039