TEK TEXT

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Basic Sound System Performance Measurements

The system just doesn't sound as good as it should. The equipment seems fine, and the installation looks OK. In fact, there are no obvious problems at all, but the sound quality is still less than you expected. What do you do now?

IN SUCH A SITUATION, YOU MAY HAVE three choices:

1. Walk away from it;
2. Mess around with it;
3. Get help.

Let's think about these choices. The first one may be the wisest, unless this is your own system which you just finished installing. If not, maybe you can convince the owner that this is the best that can be done, but how long will it be until someone else proves you wrong? How long do you intend to stay in this business? Even if it is someone else's job, will the owner really believe you when you tell him a new system will solve everything? What if it doesn't? This has never happened to you? Just wait—it's a real thrill.

The second choice is probably the most common, since most people are quite confident they can find a problem if they just look long enough. But what if the problem is one you have never encountered before, or is so obscure you simply do not recognize it? What if there are several problems, all interacting with each other, so that you cannot tell which is doing what, or you just overlook some of them? Such problems are very common in complex systems like these. Again, you run the risk of failing to solve the problems, because you did not recognize or understand them well enough.

The third choice has, by far, the best chance of succeeding and making you look (sound?) good. It may also carry the highest immediate cost, both to your budget and ego. In fact, this latter hurdle may be the most difficult to overcome; no one likes to think that they may not know as much as they should. Think of such help as an investment, just like many others you must make.

HELP

This can come from many sources; these pages and others, formal courses and good old experience. For specific problems and projects, you need something a bit more tangible and immediate. Possibilities include:

1. An expert;
2. Appropriate test equipment;
3. All of the above.

While it might be possible to hire a full-time expert who can supply all the information you need, it is unlikely that you could either find or afford such a person. If you are lucky, you might find someone with some knowledge in some of the areas you need, for a reasonable salary. However, this still leaves some gaps to fill in. Maybe what you need is an occasional "gun for hire," more commonly called a consultant. Some are actually quite willing to work on such an occasional and limited basis; ask around. Most people prefer the second option, however. The prospect of becoming an instant expert just by buying the right hardware is very appealing. After all, isn't this exactly what your competition flaunts in your face? Ours is a hardware industry; you need the toys to play the game.

There may be a few tiny problems with this approach, however, such as:

1. The toys are expensive;
2. They are not easy to learn to use;
3. They do not assure you of winning the game.

In fact, there are several important system performance characteristics which even the expensive, trendy devices cannot measure. Perhaps this is not the right answer, either. That leaves the combination approach, as you have already guessed. Somehow, we must find a combination of hardware and software (expertise) which will tell us everything we need to know. This is a tall order! Many consultants do not carry that many guns.

TELL ME WHAT YOU WANT

By the way, exactly what do we need to know about the performance of a sound system? Naturally, there is considerable difference of opinion on this matter, but there is also some agreement. For example, there is compelling evidence that the most important audible characteristic of any sound system, by far, is its frequency response. Other obviously important characteristics are loudness, freedom from distortion and noise, and reliability. Furthermore, there are some hidden implications here. In particular, the frequency response is perceived by many different listeners at the same time. Naturally, they should all hear very nearly the same thing, but few sound professionals have ever really checked to see how far this requirement has been met (or missed). The same considerations also apply to other characteristics, such as loudness. All of this means that audience coverage uniformity emerges as a very important, but widely overlooked, characteristic.

So, we have more to measure than we first thought. Maybe this is

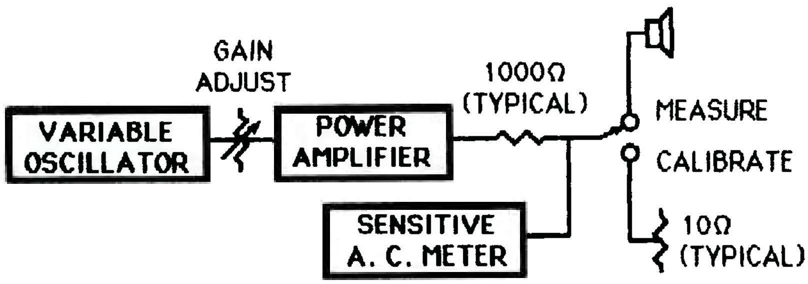


Figure 1. A simple apparatus for measuring the impedance of a loudspeaker line at all frequencies.

why a system equalization often turns out poorly; something important was left unmeasured and completely overlooked. This is ominous; could it be the situation is much more complex than we had expected and that measuring it adequately will be just too expensive? Yes and no! It is true that the factors affecting the sound heard over a system are more complex than most people realize, but they can be measured and understood. Expensive instruments are not required, either; inexpensive devices will work just fine if you only have the right ones and know how to use them.

WHERE TO BEGIN

Since the frequency response is the most important system characteristic, shouldn't we measure it first? That is what many people do; in fact, they often do little else. But that is the mistake: many problems with a system can affect its response. These all need to be found and eliminated before the final response can be measured accurately. In fact, it is wisest to make sure the system is functioning as well as it possibly can first, and measure its response last.

The logical first step is a thorough visual inspection of the system, looking for wiring and mounting errors. Are all the loudspeaker drivers very close together and in the same plane? This is very important for good coverage uniformity, and is a common problem. Many "arrayed" full-range units have severe problems in this respect, as later tests can show.

The next step would probably be to listen carefully to the system, to detect any distortion, noise or malfunction. A simple audio oscillator can be very helpful here, especially in setting gain controls for the best signal levels and for troubleshooting. This is also the time to see that all microphones and loudspeakers are wired in the same polarity (not phase). Assume nothing—a unit may be miswired internally. An inexpensive polarity checker is very handy here, but there are ways to test polarity without a checker. For example, you can put two mics side by side, open both to the same level, and listen to what they are picking up together. If the sound is OK, so is the polarity; otherwise, it is not.

IMPEDANCE PROGRESS

Probably the most common system wiring error is a loudspeaker load with an impedance too low for the power amplifier to handle adequately. If it is a little too low, the result is increased distortion and chance of overload; a greater error can result in the sudden failure of drivers and/or amplifiers. Again, do not assume anything, especially that the manufacturer's rating is correct. A loudspeaker's impedance varies greatly with frequency, and is often considerably lower than the rating. Furthermore, as loudspeakers age, their impedance typically drops even lower.

There are impedance meters which are not expensive, but they measure at only one or two frequencies, which is not good enough. It is much better to measure the impedance over the full frequency range. No inexpensive instrument exists to do this, but you can rig one up very easily.

Figure 1 shows how to do it. The oscillator feeds the line under test through a resistor about 100 times the nominal value of the line impedance. Under these conditions, the voltage across this line will vary according to the impedance. We first calibrate the meter by putting a known value in place of the line and setting the levels so that the meter reading corresponds to the resistor value. For example, if we use a 10 ohm resistor, we might set the levels so that the meter reads 10 mV. A little experimentation will be needed to get everything right. Check to see that the reading is the same at all frequencies. Then substitute the line for the known resistor and read the level on the meter. A reading of 5 mV means 5 ohms, 20 mV is 20 ohms, and so on. Then we vary the oscillator over the audio range and watch the meter readings, especially for low values (a high impedance is seldom a problem). The actual readings may be a lot lower than you expect.

Transformers, such as those used in 70V lines, are especially bad at low frequencies. The impedance of any such line should always be measured carefully. High-level crossovers are another problem area; if the driver impedance changes strongly near the crossover frequency (usually because of a resonance), the crossover characteristics will most likely be affected. An impedance which rises with higher frequencies indicates an inductive load (which may be bad at low frequencies), while one which falls indicates a capacitive load (maybe bad at high frequencies). Numerous peaks (resonances) are another sign of trouble. Each model of loudspeaker has its characteristic impedance curve, just like a response curve. Any variation from this norm implies a similar variation in response, probably indicating a faulty unit. This is another reason to run such tests.

Any load, not just loudspeakers, can be checked in this way, but the most problems, by far, are found in loudspeaker lines. For sure, the total load on each power amplifier in a system should be measured, and it would be wise to check each individual driver. In this way, many unsuspected problems can be located before causing trouble.

GET COVERED

After we are sure all the components of a system are working properly, individually and together, then we can see how well they are doing their job of delivering sound to the audience. The parameters here are frequency response and sound level, and our first concern is that these be as uniform as possible over the listening area. How can we say what the response or level of a system is if these characteristics are different everywhere?

Measurements of sound in a room require some special devices and techniques. A single frequency (as from an oscillator) will not tell us much, because the zillions (by actual count) of resonances in a room cause the sound level to change greatly with even slight changes in frequency or location. Measured this way, uniformity is nonexistent!

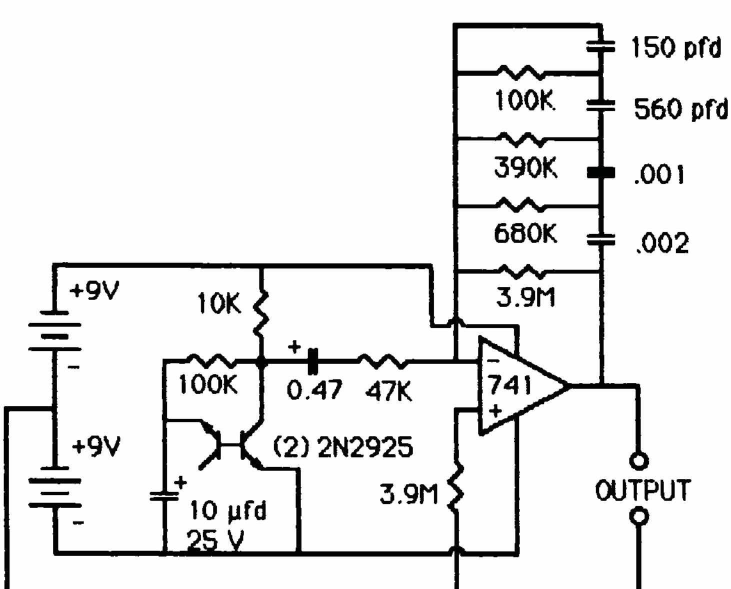
We are usually unaware of this because we hear groups of frequencies rather than individual ones, which averages out the level variations. Rather than fight nature, we should take our measurements in the same way. That means we need a signal source which produces many frequencies at once; the most commonly used source of this type is random noise.

Our ears work on a nearly logarithmic frequency scale, which means that we hear (at higher levels) nearly the same loudness in bandwidths which are the same percentages of their center frequency. If we measure random electronic (white) noise in this way, its level increases toward higher frequencies, at a rate of 3 dB per octave. Since it would be much more pleasing to both our ears and our measurements if the spectrum were flat, we usually feed the noise through a "pink" filter which attenuates the higher frequencies at a rate of 3 dB per octave.

Inexpensive pink noise generators are available commercially, or you can build one quite easily. *Figure 2* shows the schematic for a generator which is within 1 dB of ideal flatness between 50 and 15 kHz, and which rolls off gradually outside of these limits. The noise generator portion was derived from a circuit which Walter Jung published in the February 1971 issue of **db Magazine**. Other circuits are somewhat better, but this one is very simple and inexpensive, and quite satisfactory if you understand its limitations. Specifically, its output level varies some with temperature, so use it in a fairly stable environment. The first transistor should be selected for highest output without sputter. I have never found another type which works as well as the 2N2925 here, but, since they are readily available for about 25 cents each, that is no problem.

In a real pinch, you can even use an FM radio tuned off station. The noise spectrum is white with the highest frequencies rolled off by the de-emphasis circuit, which results in a very broad hump centered about 2 kHz. This is not too terrible for a rough coverage check, as the hump falls close to the frequency at which our ears are most sensitive to coverage irregularities.

Figure 2. An inexpensive pink-noise generator.



Of course, we also need some way to measure the sound levels which the noise produces over the system. Such a device is cleverly called a sound level meter (SLM). A model which is adequate for basic measurements is available from Radio Shack for under $40. Other devices are more accurate and do more, and cost more. The additional feature which is highly desirable, but not absolutely essential, is octave or third-octave filters built in. Consult the guide in the November/December 1990 issue of **db Magazine** to see what is available.

START HERE

Let us assume you have only the simplest equipment and let's see what we can do with it. Feed noise over the system at a moderate level and set your SLM to A-weighting. This setting rolls off the lower frequencies rather strongly, similar to your hearing at lower sound levels. It also lets the meter read mostly the frequencies where coverage uniformity is most important. Set the sensitivity to get an on-scale reading at a slow meter speed. Slowly walk through the listening area, holding the meter away from you and not getting between it and the loudspeakers, or the meter between you and the loudspeakers. Walk from side to side and from front to rear, along several different paths, and note how the reading changes. With such a broad frequency range, the reading should be nearly the same at all locations, within 3 or 4 dB. If it varies more than that, your coverage is not as uniform as it should be, which is a very common problem. Even if the reading is consistent, you still may have coverage problems. If you are in a reverberant room, for example, then you may be reading only reverberant sound, which has almost the same level everywhere, while the direct sound from the loudspeakers varies by an unknown amount.

While making your traverses, listen carefully for any changes in the noise. Sudden changes across a limited distance, especially if they are repetitive, are probably caused by a poor overlap between adjacent loudspeakers. Your ears are more sensitive to such irregularities than this type of instrument. On the other hand, your ears will probably not notice a gradual change in level, which the meter will clearly indicate. Listen also to the nature of the sound source; it should be well-defined and stable. Problems here are further indication of poor loudspeaker layout.

PIECE BY PIECE

There are other techniques which will tell us much more about the coverage. One of the most useful techniques is to feed noise over only one loudspeaker at a time, and measure its coverage. Find the point where the level is the highest, then see where the level drops off 3 dB from this, and, for a better understanding, mark such points on a floor plan. Connect these points, and you will have drawn the -3 dB contour, which is the limit of really good coverage. For even more information, do the same for the -6 dB points. Then repeat all of this for each loudspeaker, at least roughly.

Now examine each coverage area. Individually, each should cover a portion of the audience evenly and not spill much outside this area, or onto walls or unoccupied floor. Together, they should fit neatly to cover all the listening area and little else, with little mutual overlap. Furthermore, the actual level within each area should be the same as in the others.

Tedious? Yes. Revealing? Absolutely! Disturbing? Quite possibly. Certainly this procedure will tell you what is right and wrong about the coverage, and may well reveal that the coverage is not what you expected. It will also tell you, of course, how well your improvement efforts are working. It may show, for example, that no matter what you do with your loudspeakers, you cannot cover all of the audience area evenly. Bad news: you need different loudspeakers. This may, however, explain a few things.

Likewise, this technique is not limited to high-frequency coverage. Coverage of low-frequency units can be measured simply by turning off everything else and using the C-weighting on the meter. However, since this weighting is almost flat, the meter will now pick up a great deal of low-frequency ambient noise. Check the level of this first with the system off, then set the system level 10 or 20 dB higher. Be careful not to overload anything!

At low frequencies, level variations occur much more gradually because of the longer wavelengths involved. For the same reason, low-frequency loudspeakers are almost non-directional, so changes in their location and aiming have less effect. It is often impossible to get really good coverage in this region.

BAND TOGETHER

Even more can be learned by using octave band filters. Such a limited frequency range reveals level variations much more easily and quickly. Additionally, full-range units can be tested with all drivers on, because the filters can discriminate between them.

It is convenient, but not essential, to have the filters built into the SLM. In fact, nearly the same results can be obtained with an octave band graphic equalizer. Simply wire it into the noise signal path, then turn one filter all the way up and the others all the way down. This will produce an octave band of noise, good enough for coverage measurements. Uniformity should be checked in several bands—typically at 4 kHz and an octave or so below each crossover frequency.

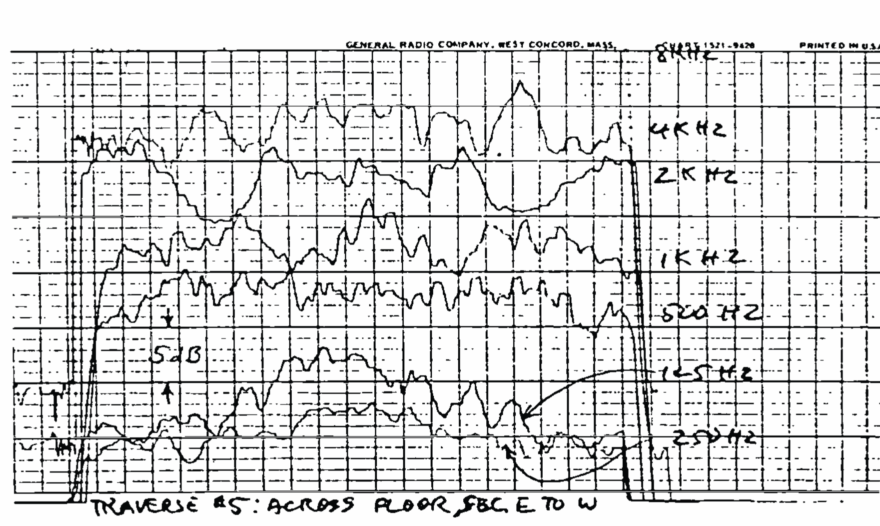
Filters are also very useful for setting levels, both between the various drivers in a full-range unit, and between various units covering the same frequency range. An equalizer may be a bit less accurate in this application, however, as the various filters may put out slightly different levels at their maximum settings. If these levels are measured, however, and adjusted to be the same, then there should be no problem.

THICKER PLOT

Extensive readings taken by these techniques and carefully plotted will show a great deal more about actual loudspeaker coverage than you ever suspected, if you are willing to go to the trouble. There is a way to get even more information and save considerable time and trouble at the same time, but, of course, it is not free. It involves tape recording the sound during the traverses, rather than reading the levels directly, then playing back the recording through filters into a chart recorder. The resulting graphs, such as those in *Figure 3*, are much more informative than any direct readings could be.

For example, the graphs shown are plotted from a traverse across the floor of a large arena, which had a well-designed central loudspeaker cluster. The 125 Hz band shows a rise in level near the center of the floor, because that area is closer to the cluster and the low-frequency units were not directional enough at this frequency to offset this effect. At 250 and 500 Hz, their directional characteristics are adequate to provide very uniform cov-

Figure 3. Plots of actual loudspeaker coverage uniformity.



erage. The high frequencies show moderate irregularities about halfway to each side, because this is the transition region between horns. At any given location, the level may be up at one frequency and down at another. Readings taken over a broader frequency range would not reveal such irregularities, even though they are clearly audible, which illustrates the value of this particular technique. Actually, the coverage shown is quite good, yet its shortcomings are clearly revealed by this measurement technique.

The filters needed for this method are no problem, as we have already seen. The recorder used was a Sony Walkman Professional; any good-quality recorder without a limiter will do. The chart recorder? Ah, that is the problem; they all seem to cost $2,000 or more. That is really not bad for what they can do, but if your budget is not quite ready for that, there is an alternative. Some consultants are willing to take your recording and plot it out in this way for a reasonable fee. Ask around if you are interested.

ANOTHER DIMENSION

This set of plots has also given us some information on another matter. In some locations, the sound level was up in one frequency band and down in another. This, of course, amounts to a response irregularity, which changes with location. In other words, the response may not be uniform over the listening area, even if the overall level is. Coverage uniformity comes in two dimensions: overall level and frequency response. Just when you thought you were beginning to get a handle on all of this, Murphy strikes again.

Unfortunately, response irregularities are even more pervasive and difficult to control than level irregularities. Anyone who has ever carried a real-time analyzer around in a listening area while pink noise was playing over a system has seen the truth of this. The response is literally different in every seat.

This has several annoying implications. First, there is no location where an "average" system response can be measured. Second, the only way to determine the actual average system response is to measure the response in a number of locations and then average the results together. Third, the response in any given location will be different from this average, perhaps considerably so.

All this means that the system frequency response is difficult to measure. Forget about just taking a reading with a real-time analyzer; that will tell you something about the response at one location, which will be different at all others. If we could take many such readings at different locations and average all of them together, we might have something useful, but how do we go about such a thing? Furthermore, how do we go about it if we do not have a real-time analyzer and do not want to spend the money for one?

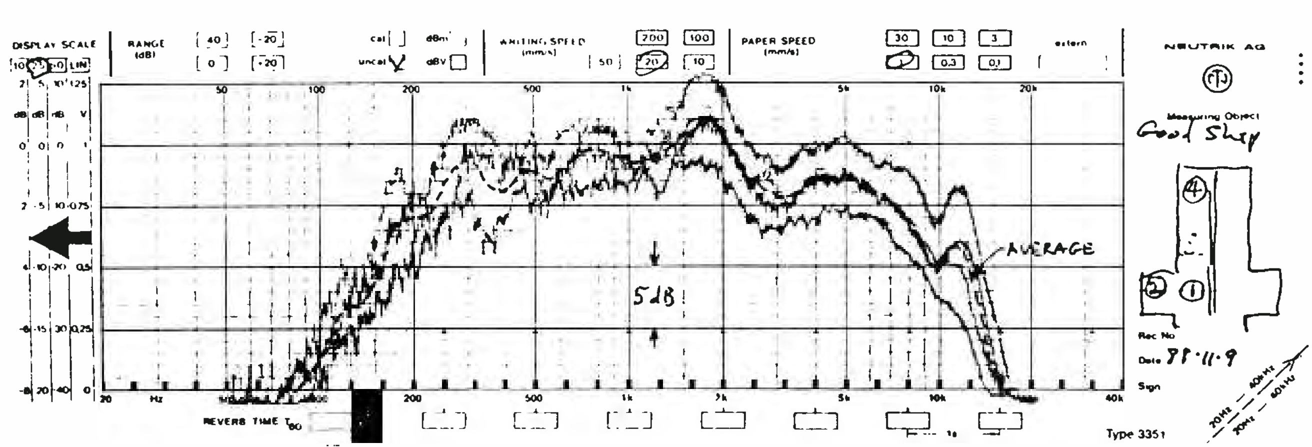
THE RIGHT RESPONSE

As before, there really are ways to make accurate and meaningful measurements of the system frequency response with a minimum of equipment, if you are willing to go to the trouble. How do you think it was done before there were any real-time analyzers? (Yes, such measurements were actually made for many years before then, but only by a few dedicated professionals.)

The minimum test equipment needed is a pink-noise generator and a filter set. The same equalizer you use for response correction can be used for measurement, if you are careful. It can be any bandwidth—octave, two-thirds or third-octave—of the graphic (boost and cut) type. As always, the narrower the bandwidth, the better the results. Rauland-Borg used to manufacture a device designed just for this application, which worked very well (trust me; I designed it). It is no longer available, but you can approximate its function with any graphic equalizer.

The trick, once again, is to set all the equalizer bands to minimum, except for one, which is set to maximum. The danger in this is that the various filters in the equalizer may not be matched well enough to produce all the same level. Check this first by feeding the noise generator through the equalizer to a VU meter. It will be easier to read the meter, especially at lower frequencies, if you temporarily slow down its response speed by connecting a large (at least 1000 fd) capacitor across its terminals.

Figure 4. Plots of actual system frequency response, measured by continuous sweep.



Set all the bands to minimum except one, which is set at maximum, and read the resulting level. Try starting with the lowest band, and adjust overall gains (not the band) until you get a "0" reading on the meter. Visually average the meter fluctuations over a few seconds. Then reduce this band to minimum, turn up the next one, and read the level without changing any other gains. If this reading is higher than before (usual), reduce the filter setting until you get the same reading, and note (mark) the resulting setting. Repeat this for all of the filters, and you will have them calibrated to the same level. Such filters will not be as good as those in most analyzers, but they will be quite sufficient for our purposes.

THE SETUP

Set up the system by feeding the pink noise into the power amplifiers, set to produce a moderately high sound level over the loudspeakers (about 80 dB). Place the main system mic (if there is one; otherwise use a flat-response omnidirectional type) in the seating area, at ear level and pointed toward the nearest loudspeaker. Feed its output into the slow-responding VU meter, with gains set to produce a mid-scale reading. Turn off the noise generator temporarily to be sure that the background (ambient) noise level is much lower. Insert the filter set (equalizer) into the system either after the noise generator or before the meter. Theoretically, the results will be the same either way, but there are practical differences. With the filter after the noise generator, you will hear each noise band individually, which may be useful for detecting overload, distortion, rattles, and so on, in the amplifiers or loudspeakers. Filtering before the meter will help discriminate against ambient noise if it is too high.

Run the lowest band on the equalizer up to maximum, with all others at minimum, and read the level on the meter (if it is not too low). Plot this as a point on frequency graph paper. Then turn this band down, run the next one up to its calibrated level, and take and plot the new reading. Repeat this for each band in turn, then connect the points on the graph for clarity. Next, move the mic to an entirely different location in the listening area and take another set of readings. The plots will be clearer if each is made in a different color. Take readings at six to twelve locations scattered throughout the listening area.

This may take a couple of hours, but there is little other cost, and the information obtained is highly valuable.

ON THE AVERAGE

Examine the resulting plots to see how closely they track each other. If they all fall within a 5 dB window, you have either an incredibly good loudspeaker system or a very reverberant room, which gave you misleading readings. It is normal to have greater variations at lower frequencies because of the lack of directional control in loudspeakers in that region. Greater variations at high frequencies are also common, and indicate inadequate coverage. One or more plots which are consistently higher or lower than most indicate poor level matching. Wide variations over the mid-range strongly indicate an inadequate loudspeaker design.

Draw an average through all the plots. This, finally, is the overall system response. Examine its shape; it is not unusual for the level to drop off rather sharply near the frequency extremes, but large irregularities elsewhere are a sign of trouble. Crossovers are a common problem area; consider these carefully. All in all, your first carefully measured system response may look considerably worse than you expected.

This method will give you as much information about your system response as will a real-time analyzer with built-in memories. In fact, it can do even better, because the number of plots is not limited by the memories, and because they are produced in hard copy. The results are certainly nothing to be ashamed of, but there are ways to do even better. One way would be to use a bandwidth narrower than third-octave, but such devices are rare and expensive. Another way is to analyze and equalize, using filters which can be set to any frequency, rather than being limited to fixed points. Such filters are called parametrics, and they are readily available at reasonable prices. It is even possible to build the necessary sweepable filter easily and inexpensively, if you are interested. As before, the only real price for more information and better results is more time and trouble expended.

A CLEAN SWEEP

In this case, the procedure is to set up the system, with the equalizer installed as before, and set the filter bandwidth to one-third octave. Then the filter center frequency is slowly swept, and the resulting level variations are read on the meter. The exact frequencies of peaks and dips can be found and plotted, as well as enough other points to establish the true shape of the response curve. After several such curves are plotted and averaged to get the overall system response, the needed corrections can be seen very clearly.

All of this is very helpful, but tedious. Fortunately, a recording swept analyzer, which consists of a tunable filter coupled to a chart recorder, can do all of this for you.

Just seeing which band of an equalizer the feedback falls in is not good enough, since the bands are too broad and overlap too much

Currently, only one manufacturer (Neutrik) makes a version well-suited to this application, and it costs about $3,000, but another manufacturer (GenRad) used to make a similar device which is still widely available on the used equipment market. Both are capable of making several types of useful measurements, including reverberation time. *Figure 4* shows an example of such an automated response plot.

As before, it is possible to record the noise at various locations, then send the recording to someone who has the equipment to analyze it and plot out the frequency response. The tricky part here is that the response of the recorder probably is not flat, but a test recording, consisting of pink noise fed directly into the recorder, can be analyzed to compensate for such shortcomings. It can work, if you have time to wait for the results.

THE HOWLING

The final system measurement is of the feedback frequencies, which, like many other characteristics, are often not measured at all. In fact, it is not necessary to measure the feedback frequencies unless feedback is a real problem in a particular system, and a special effort must be made to control it. Then, more information may be helpful.

Just seeing which band of an equalizer the feedback falls in is not good enough, since the bands are too broad and overlap too much. If you have an oscillator with an accurate tuning indicator, you can tune it to the same pitch as the feedback and read the scale, but that is usually not close enough, either. A better method is to read the frequency directly with an inexpensive frequency counter. There are even ways to measure all the frequencies at which a system can feed back, without actually making it do so, but these require a bit of specialized equipment and techniques which almost no one seems to know about. Furthermore, while it is possible to perform this analysis via a recording, such a technique would be difficult and probably impractical.

NOW WE KNOW

However, we have seen that many very useful performance tests can be carried out easily and inexpensively. In fact, these tests provide a great deal of information on exactly why the system sounds the way it does, much of which cannot be obtained at all with highly sophisticated and expensive test equipment. While the most basic tests can be tedious, they provide excellent results for a minimal equipment cost. Moreover, even better results can be obtained much more quickly and easily for a relatively modest investment. Also, if you have only occasional need for full system testing, or would rather not try to handle it all yourself, there are consultants who can provide such services.

We have not yet said as much as we need to about how to interpret the results of these tests, and have said almost nothing about how to correct the problems they reveal. All of that would fill another article, as it probably should. If you just cannot wait, more information is available from the referenced papers, or you can contact the author directly at (512) 837-7252.

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