How Earthworks Measures Microphones

By Alex Khenkin, Director of Engineering, Earthworks

How to measure microphones and the implications relating to measuring loudspeakers.

In the 1920s, they said that Victrola was so good that its sound was indistinguishable from the original. Many similar claims were made for later innovations, some of them with considerably more justification. When CDs were introduced to the public in the 80s, the history repeated itself once again – many experts pronounced CDs to be the ultimate recording media (as well as many others who pronounced quite the opposite.) It seems like the subjective boundaries of the audio perception are being constantly pushed away. Now a new, 96 kHz 24 bit media is coming; it does sound better, more realistic and "live", than CDs do. The availability of wideband and extreme dynamic range recording techniques requires careful rethinking of all components of audio systems,and has important implications for loudspeaker designers. It is becoming increasingly evident that audio perception extends below 20 Hz and above 20 kHz and that maintenance of accurate time relationships is critical to achieve an accurate sound reproduction. Consequently, measuring audio components is becoming more and more challenging task; the most difficult to measure elements in the audio chain being microphones and loudspeakers.

There is not much dispute about how to make measurements in construction or heavy industry, but there is some controversy in the field of acoustics regarding measuring techniques. This does not apply to the whole field of acoustics, though. The measurement techniques in ultrasound and underwater acoustics are well defined and documented, with standard measurement procedures and readily available equipment. It sometimes seems that this is not the case for audio. Everyone has an opinion on the subject, but they all agree that it's not easy to do.

The first thing to understand about the problems of measuring in audio is that measuring loudspeakers and measuring microphones are two very different tasks. The techniques that are valuable for one are erroneous and useless for the other. When measuring loudspeakers, one needs to eliminate the microphone as a variable, and when measuring microphones one needs to eliminate the loudspeaker as a variable.

Here at Earthworks we use the well known substitution method on a daily basis to measure microphones. The substitution method presumes a microphone with a known response in order to make the sound source errors virtually disappear. Essentially, the test is performed like this: the known microphone is placed at a specific location in front of a sound source, the source's frequency response is measured, then "nulled" (i. e. referenced) and corrected for the microphone's frequency response. Then the microphone under test is positioned in place of the reference microphone where its response can be measured and compared to the reference.

This method, however simple it might look, has pitfalls and requires a deep understanding of the principles involved. International standard IEC 60268-4 describes the conditions under which this test must be performed. The first and the most important requirement is that microphones should be measured in free field, employing plane waves or spherical waves. This means that a sound source must be small compared to the wavelength at any frequency under test and the microphone tested must be far away from the source. In a case where either the circumference of the source and/or that of the microphone exceeds the wavelength, the measuring distance must comply with:

r>d
r>d2/l

where
r is the distance from the source to the measuring point;

d is the effective diameter of the sound source;
l is the sound wavelength.

If these criteria are not met, the test results will be compromised or even meaningless. Since the standard calls for a well defined sound source, no boxed loudspeaker can be used, because each discontinuity (from the box corners down to screwheads sticking out) will act as a secondary radiator, making the source altogether too large and producing a very complex sound field which is very far from spherical. In such a situation even the slightest differences in the microphone's dimensions, shape and position will produce variations in frequency response that have no correlation with the microphone's actual performance, and which depend upon the actual loudspeaker used (should I remind the reader that any valid scientific measurement must be independent from the tools and methods used?).

IEC 60268-4 standard gives the recommendations on how to check if the sound field satisfies the above conditions:

> "Free field conditions are considered to be sufficiently realized in the region around the microphone if the following conditions are met:
>
> – within the distance of 200 mm in front, behind, left, right, above and below the position of the microphone the sound pressure level is measured at every measured frequency by means of a pressure transducer;
>
> – the axis of the transducer shall point towards the reference point of the loudspeaker.
>
> The corresponding sound pressure levels on axis positioned at different distances from the loudspeaker shall not differ by more than 0.5 dB from the calculated levels in the ideal sound field. The values at a nearly constant distance (i.e. right, left, above and below) shall not differ by more than 1 dB from the level at the reference point of the microphone."

Another very critical requirement for this test is that a source loudspeaker must be very good,meaning that it must have a very smooth frequency response through the whole frequency range of interest. Sharp peaks and/or deep notches cannot be removed by the referencing process. I have learned this by bitter experience while designing our new directional microphone. The microphone seemed to have intermittent anomalies in frequency response around 10 kHz. After a couple days of experimenting, I looked at the frequency response of my source tweeter before referencing, and, sure enough, there was deep notch at 10.5 kHz followed by a sharp peak. I replaced the tweeter, and the apparent problem in the microphone disappeared.

Since no loudspeaker exists which can cover the entire frequency range, we use separate setups for testing different frequency ranges. The high frequency setup (1 kHz and up) utilizes a dome tweeter that is good out to 40 kHz. This is flush mounted in 1.2 x 2.4 m (four by eight feet) baffle.

Meticulous care must be used in the mounting of the tweeter. All surface discontinuities must be removed to make sure that the only radiating point is the tweeter's dome itself (Fig. 1). We use a standard microphone and DRA's MLSSATM measuring program, time-windowed to remove reflections from the baffle's edges, to measure the sound field at a certain point. This measurement is stored as a reference. Then the correction curve for the standard microphone is applied as auxiliary reference, to take the test microphone's frequency response out of the picture. The microphone to be tested is then measured in the same location and its response relative to the reference is plotted. The low to mid frequency setup utilizes a small cone loudspeaker mounted in a 2.4 x 2.4 m (eight by eight feet) baffle with the same care to avoid surface irregularities (Fig. 2), and a similar test procedure is used. In addition, we have designed a long tubular boom for a microphone stand to eliminate microphone clip reflections (Fig. 2). The dimensioning of the baffle is the main low frequency limitation for this test.

Measuring very low frequencies by the substitution method is problematical because such tests require a very large anechoic space. To measure in frequency range from zero to 500 Hz, we use a

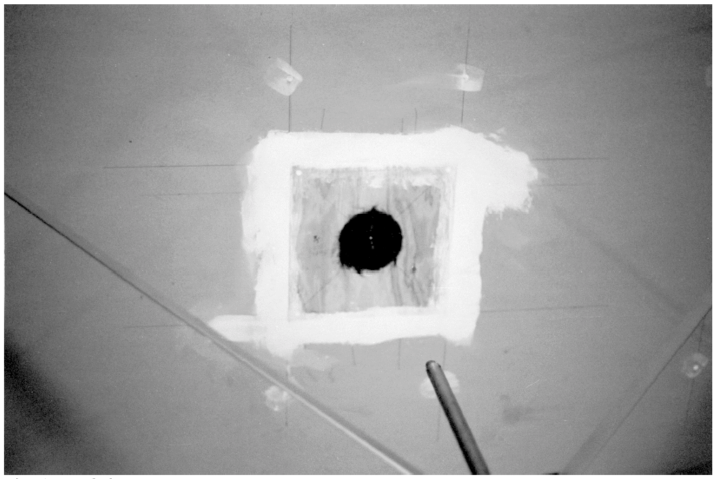

Fig. 1. *High fequency test setup.*

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Fig. 2. *Low to mid frequency test setup.*

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small, piston driven pressure chamber. This method is actually a primary frequency response calibration, since the pressure in a given size chamber at a given temperature depends only on the volume displaced by the driving piston. The high frequency limitation is defined by the chamber size, which should be less than 1/6 wavelength in any direction. Of course, this only works for pressure microphones (omnis).

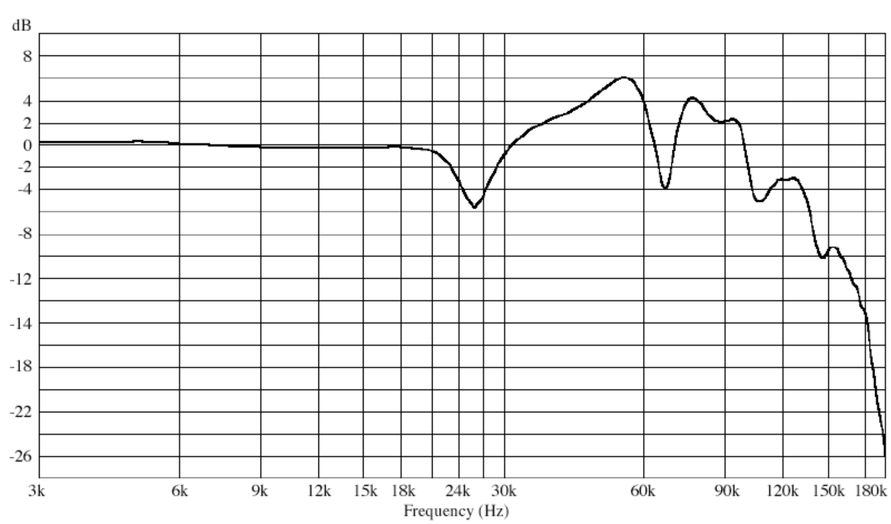
We have developed another technique, which we believe to be of primary standard nature. This is our proprietary system for impulse response measurements using a spark as a very short pressure impulse event. This way we can obtain the actual impulse response of the microphone. We then derive frequency response using FFT. We use a 1 MHz sampling rate, eliminating the need for an anti-aliasing filter (any microphone's response is down so much at the Nyquist frequency that the microphone itself works as an anti-aliasing filter). This method has many advantages to it when compared to any other calibration methods. First and most important, the response is obtained from an actual acoustic event, thus revealing any problems the acoustical structure of the microphone might possibly have. Second, it's relatively fast – about a minute per microphone. Third, it is suitable for any type of microphone. Fourth, it also serves as a good test of any RF problems with a microphone. The shortcomings are that it can be used for high frequency testing only, and it cannot be used for measuring the absolute sensitivity of the microphone. While there is not much novelty in the test itself, to make the actual spark machine is a real challenge. In designing our spark system we had to make sure that the actual sound pressure pulse was short enough so that it's frequency content was well-defined well beyond 200 kHz.

All this long and complicated routine is employed in order to provide the best measuring tool possible, so that you can make the most out of loudspeaker measurements which require the same attention to detail in order to get meaningful and consistent results. In loudspeaker measurements, there are two different tasks. The first task is to measure and select components (drivers), and second is to measure a loudspeaker as a final product. These tasks are very different and need a different approach. When measuring components, one should eliminate any variable except the component itself, so that decisions in a selection process are based on actual performance of the drivers and not side effects like edge reflections. For driver testing we use similar 1.2 x 2.4 m baffle with the cutaway for a 35 x 35 cm auxiliary test board. The test board is easily replaceable, and for every driver model we measure, we make a new board in which the driver fits perfectly flush and snug. This auxiliary board with a driver in it is then flush mounted into the baffle, and all slots are filled with soft modeling clay to make a seal and a smooth transition from one to the other. This type of care is absolutely necessary; even slight irregularities in the surface can (and will) bring errors into driver's response, especially if these irregularities have a symmetrical pattern. Placing a tweeter a millimeter under or over the surface changes the response by several decibels and makes the tweeter look unusable in hi-end applications. Of course, this baffle is not large enough to measure low-frequency drivers, and we plan to build a much bigger one.

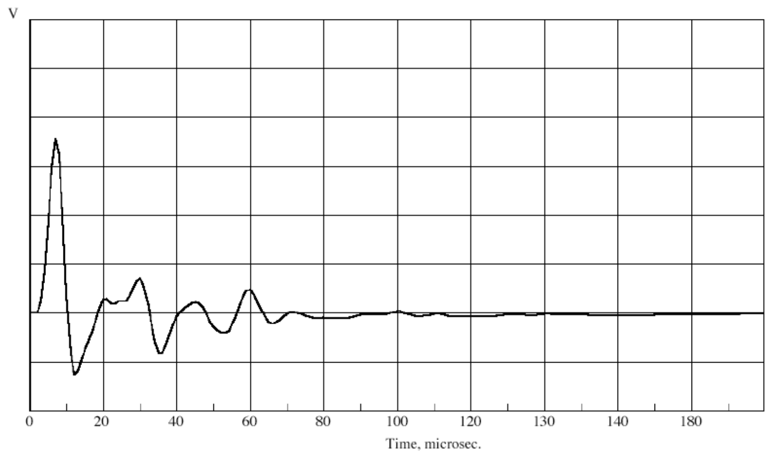
The second task is to measure a loudspeaker as an assembled device. At this stage the speaker is placed in anechoic chamber and measured as a whole, edge reflections (often incorrectly called "diffractions") and all. It's only when measuring a finished loudspeaker that one really wants to see those reflections and their effect on the speaker's response.

The choice of a microphone can have a very strong effect on the results (now we are back to that "scientific measurement must be independent from the tools and methods used" business). There are four questions one should ask about the microphone to be used:

- is this microphone free-field calibrated?
- is its frequency response smooth through the whole range of interest?
- does it have its actual response data so that it can be taken into account?



a) Frequency response.



b) Impulse response.

Fig. 3. *Measured frequency and impulse responses of Gefell MK 301 measurement microphone with the cap on*

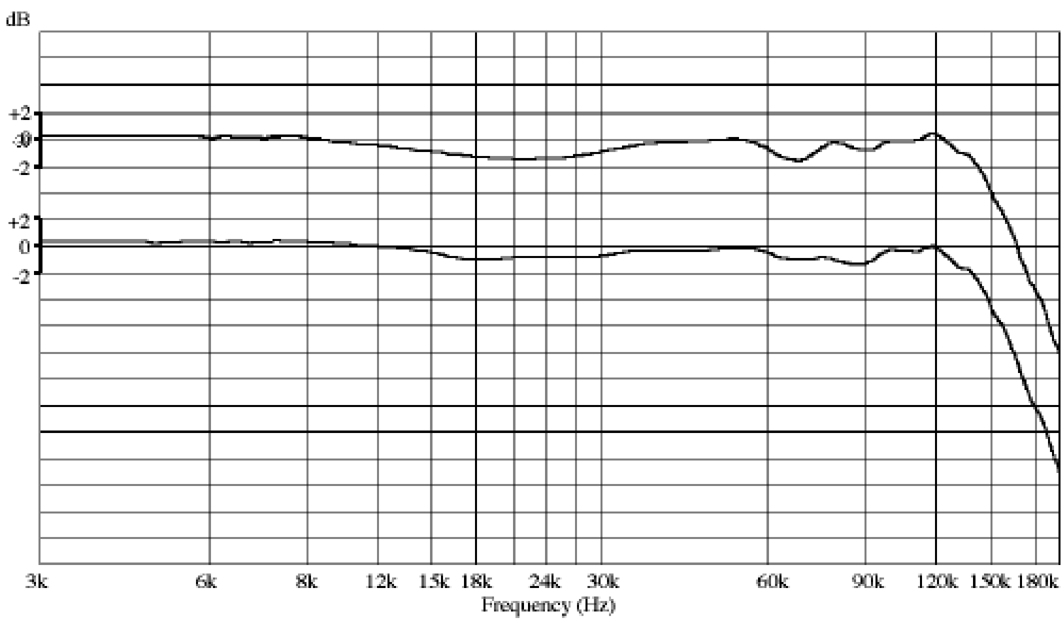


Fig. 4. *Measured frequency responses of Gefell MK301 measurement microphone with the cap off: threads open (top) and covered (bottom).*

· • does its shape lack any reflective artifacts (like abrupt steps, protruding rings or switches, etc.)?

If the answer is positive all four times, there is a very good chance the microphone will give a truthful representation of the speaker measured. Attention must also be given to the microphone's mounting. Microphone clips should be placed as far from the tip as possible. I even use an absorptive material around the microphone stand to eliminate reflections. Although Earthworks microphones have the stepped shape, the step is very gradual and the reflections due to its presence are about 40 dB under the signal (actual measurement).

Unfortunately, the described methods are not very suitable for time-domain measurements. The main problem is that there is no easy way available to take time properties of the measuring system into account, or to null its impulse response (in fact, I don't know any). MLSSATM and other measurement systems do not yet offer the option of importing the test microphone impulse response for a time-domain reference. Thus, the only available way to obtain a meaningful time response of the driver is to use a measuring system with a time response that is significantly better than that of the driver to be measured. When measuring state-of-the-art tweeters it's becoming harder and harder to assume that the measuring system's time properties are much better. Moreover, the weakest link in this chain is usually a microphone. Even using Earthworks M55 measurement microphone, which has exceptional time domain characteristics, one still has to take its impulse response into consideration. The biggest advantage of Earthworks microphones, in this case, is that their acoustical structure creates only minimal reflections. This is vital for time-domain measurements.

Many measurement microphones have some kind of slotted cap covering the diaphragm. This cap is often used to shape the frequency response, as well as being a protective cover. Unfortunately, it is seldom a minimum phase equalizer, and tends to degrade the original impulse response of the microphone element by creating a complex set of reflections. In addition, many measurement microphones have an oversized rear chamber with significant internal echoes.

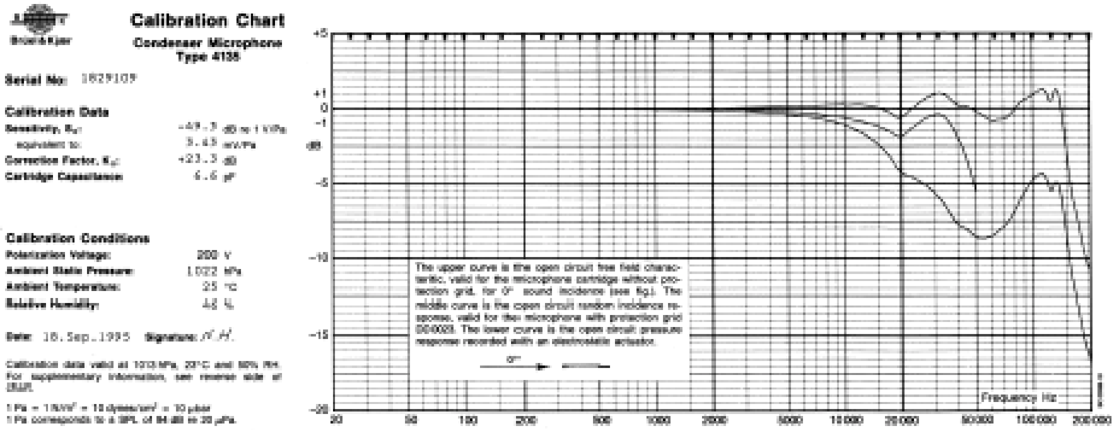
These echoes make impulse response anomalies even worse. All this results in time-domain measurements being seriosly compromised. We have studied 1/4" measurement microphones by B&K and Gefell, and concluded that the only meaningful way to use them was to take the caps off1 (Fig. 3). Then, even the exposed threads had an influence on the response (Fig. 4)! And I had to cover the threads to bring the measured responses closer to those specified by their manufacturers (Fig. 5). The reason for this is that the supplied responce curves of the microphones are not actual acoustic measurements. The manufacturers use an electrostatic actuator and a correction curve for presumed effect of microphone shape to obtain a response chart for their microphones, a method that has no means to reveal the influence of anomalies in the external acoustical structure. It's a bit like deriving the response of a finished loudspeaker from the responses of its components instead of measuring it.

Measurements in audio are not easy to perform, but providing that a careful scientific approach is used and appropriate standards are consulted, there should be no possibility for results to be ambiguous or controversial. The ruler must be the same no matter who is using it.

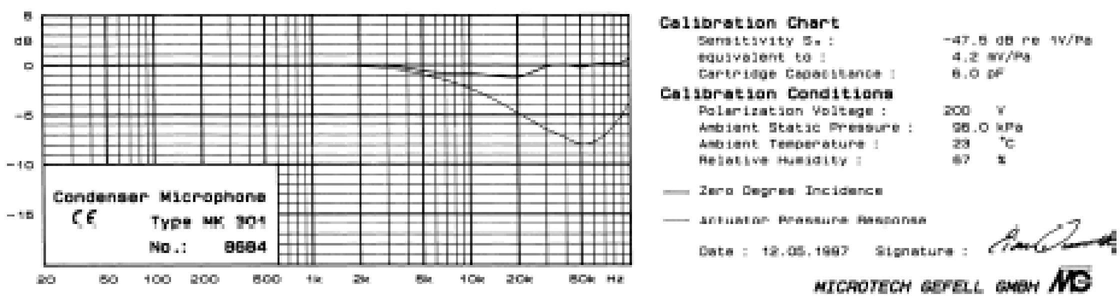
I would like to thank all the people here at Earthworks who helped me with the work that led to this article, especially David E. Blackmer.

Alex Khenkin received Masters Degree in Acoustical Engineering from Moscow State Institute of Radiotechnics, Electronics and Automation in Russia. He joined Earthworks Engineering Department in 1996.

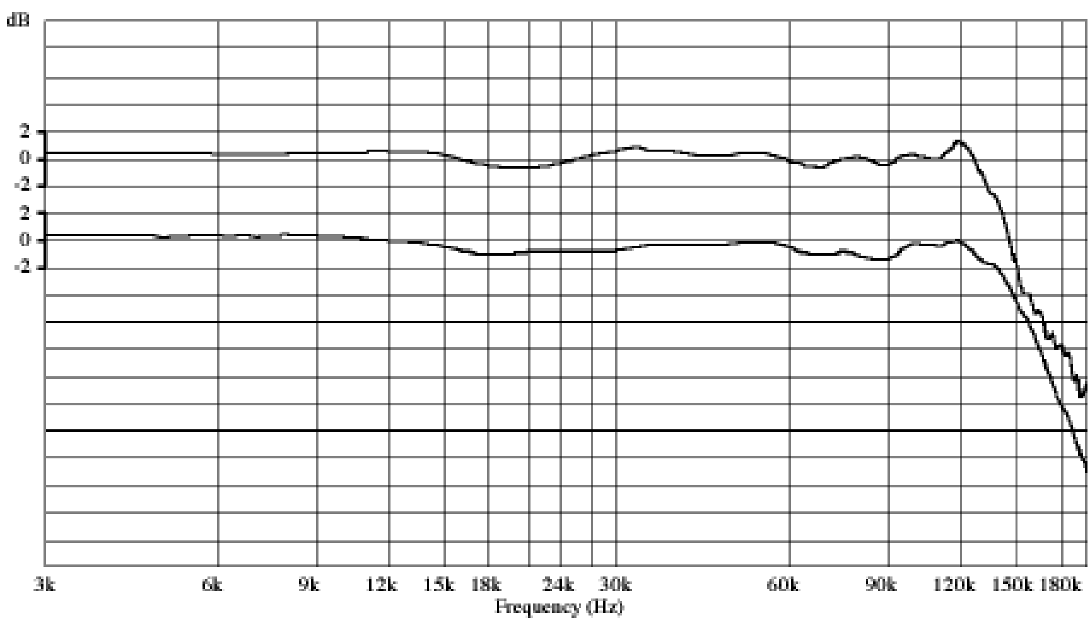
1 In fact, the responses supplied by the manufacturers are valid only when the caps are taken off. (A. K.)



a) Manufacturer's frequency response chart for B&K 4135 microphone.



b) Manufacturer's frequency response chart for Gefell MK301 microphone.



c) B&K 4135 (top trace) and MK 301(bottom trace) measured by spark test; both microphones are cap off.

Fig. 5. *Manufacturers' response curves (a, b) and measured response curves (c).*