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WHITE PAPER

The Basics of Through-the-Air Audio Quality Test System Characterization

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Executive Summary

Implementing a through-the-air audio quality test system generally involves some form of system level characterization. This paper discusses some essential aspects of audio system characterization, including the important issues in test fixture design. Although this discussion is primarily intended for mobile phone production test applications, the concept may be useful for any application that requires audio characterization, such as entertainment systems and MP3 players.

Background

The complexity of mobile phone production testing is increasing rapidly due to the almost constant addition of phone features and enhancements that require verification during production. At the same time, there's a continuing push to reduce test time and cost, compounding the challenges today's test engineers face. Under these circumstances, it may seem reasonable to rely more on the design qualification process to eliminate the need for some parts of the testing process. However, most handset manufacturers still operate under a 100 percent testing requirement. No design qualification process, no matter how rigorous, can warrant 100 percent defect-free components, particularly if those components are electromechanical transducers, which can be damaged

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during assembly or handling. As a result, audio quality measurements remain fundamental to the ever-growing suite of tests that must be performed.

The Continuing Need for Traditional Audio Quality Test

There are two basic categories of audio quality tests:

- *Audio only test.* This type of test can be conducted at preassembly, at the circuit board level, or after final assembly. This configuration usually sends the stimulus signal from the audio analyzer through audio components on the DUT and back to the audio analyzer for measurement (e.g., an audio loopback mode test). The test measures audio quality parameters such as distortion, frequency response, etc.
- *Combined audio/RF test.* This type of test, usually conducted after final assembly, involves verifying audio stimulus signal over the complete RF transmit-receive path. It requires a communication analyzer with audio analyzer capabilities.

While combined audio/RF tests usually provide a useful picture of the DUT overall, verifying correct modulation/demodulation, as well as the integrity of the audio signal, they don't necessarily reveal the condition of the electromechanical transducers, i.e., the speaker and the microphone. In fact, it's possible for a transducer with some mechanical defects to produce valid results in combined audio/RF tests.

Traditional audio tests are still essential to validate the condition of the transducers. Through-the-air audio test measurement parameters, such as Total Harmonic Distortion (THD) or Total Harmonic Distortion Plus Noise (THD+N), can reveal subtle defects in the transducers that may be hard to detect with other methods. They also provide the manufacturer with an objective measure of audio quality for total quality control purposes.

This white paper will explore some of the fundamentals of conducting an audio quality test.

Designing the Audio Quality Test System: Test Fixture Characteristics

A good test fixture is essential to achieve repeatable, meaningful results. The most important function of the fixture is shielding ambient acoustical noise, which means the fixture should be enclosed in a material that acts as a noise shield. To reduce noise as much as possible, the enclosure should be made with material that would produce maximum transmission loss, which is governed by *mass law*, approximated here as [3]:

$$TL = 20\log_{10}(m_s \times f) - 48$$

where TL = random coincidence transmission loss (dB)

m_s = mass per unit area (kg/m²)

f = frequency of the sound wave (Hz)

Material [3]	m_S (kg/m²-per mm)	A (Hz-mm)
Aluminum	2.7	12900
Steel	7.7	12700
Glass	2.5	15200
Lead	11.0	55900
Plywood	0.6	21700

Here, either steel or aluminum would be the practical choice.

Mass law is affected by the *coincidence effect* at higher frequencies and by the *resonance effect* at lower frequencies [3]. Both effects should be considered when designing the fixture.

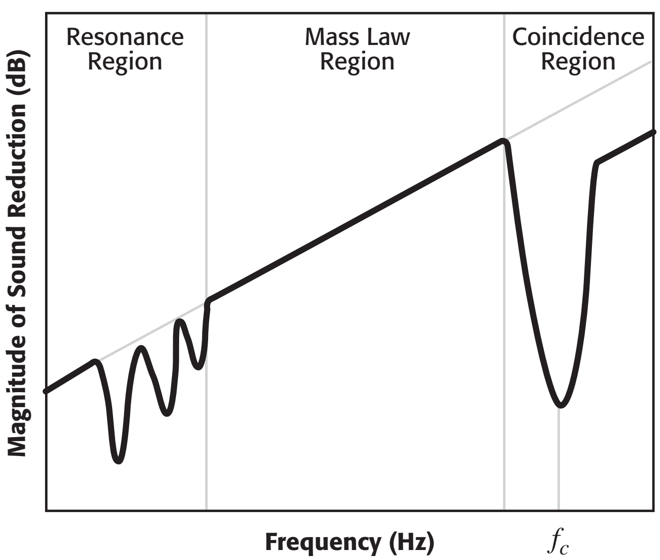

Figure 1. Typical sound reduction

Coincidence effect. High frequency waves cause ripples or "bending" waves that travel longitudinally along the wall of the fixture enclosure. The frequencies of these ripples differ from those of incident waves, except at a certain frequency called the coincidence or critical frequency (f_c). At this critical frequency, the sound energy is transferred very efficiently through the walls of the fixture enclosure and the transmission loss described by mass law no longer holds. The value of f_c is described by [3]:

$f_c = A/T$

where A = constant of material (Hz-mm) (see previous table for values)

t = material thickness in (mm)

Material thickness t (mm)	f_c (Hz) of Steel	f_c (Hz) of Aluminum
4	3175	3225
3	4233	4300
2	6350	6450

The goal here is to have an f_c that's beyond any frequency of interest in the test. Note that the thinner the enclosure material is, the higher f_c will be. For the frequency range of 200Hz–4kHz, aluminum or steel walls that are 3mm thick or less would suffice.

Resonance effect. Assuming the fixture is rectangular, consider the two facing walls of the fixture enclosure. For this example, imagine that a sound wave originates from one wall as if there is a speaker in it. This sound wave is reflected back by the facing wall [1]. If the distance between the two facing walls is half the wavelength, then we have the reflected wave in phase with the incident wave, resulting in a classic resonant or standing wave.

For each pair of facing walls inside the fixture, the resonant frequency f is calculated as [1]:

$$f = \frac{c}{2l}\, n$$

where c = speed of sound (~ 344m/s), l = distance between two facing walls (m) and n = 1, 2, 3, etc. (order of harmonic).

For a rectangular fixture, the resonance frequency is estimated as [4]:

$$f = \frac{c}{2}\sqrt{\left(\frac{n_x}{l_x}\right)^2 + \left(\frac{n_y}{l_y}\right)^2 + \left(\frac{n_z}{l_z}\right)^2}$$

where n_x is the harmonic resonance between x-wall and its facing wall, i.e., n_x is 1 for the first harmonic, and l_x, l_y, and l_z are the length, width, and height of the fixture in meters.

When choosing the fixture's design, it's important to keep in mind that the resonance frequency depends on the ratio of length, height, and width of the fixture. Unfortunately, there's no one standard ratio, although a perfect cube should be avoided. One of the commonly used ratios (R. Walker, BBC, 1996) is if [4]:

$$1.1\frac{l_y}{l_z} < \frac{l_x}{l_z} < 4.5\left(\frac{l_y}{l_z}\right) - 4$$

Another major challenge in test fixture design is to minimize cross-coupling, which occurs when the stimulus signal at the test speaker is directly coupled into the test microphone instead of going through the DUT. The best remedy for this is to use an acoustic coupler to direct the sound from the test speaker to the DUT microphone. If the signal path through the acoustic coupler is sealed properly, it can significantly minimize the stimulus signal leakage that causes cross-coupling.

It's also important to minimize the ambient signal levels inside the fixture by adding material that will absorb or dampen sound. A variety of acoustic damping materials are commercially available, with varying levels of sound absorption or reduction, depending on the type and thickness of the material.

Test System Setup and Characterization

Transducer selection is the next important aspect of fixture setup. Frequency response is one of the most important specifications for the test speaker and the test microphone. Frequency response should be flat (within ±3dB) for the frequencies of interest, usually 200Hz–4kHz. The test microphone will generally require a preamplifier, which should also have flat frequency response for the frequencies of interest and typically have a gain around 20dB. Other specifications, such as power level, noise performance, sound pressure level, etc., are also important, and should be considered carefully.

Given that the mobile phone transducers are, by necessity, physically small, their sound production quality is somewhat limited. Therefore, specialized expensive transducers usually aren't necessary. There are a variety of perfectly adequate, reasonably low cost transducers readily available.

The size of the test microphone and the test speaker should generally be small to allow for convenient mounting inside the test fixture. The sound pressure level decreases 6dB for every doubling of the distance, so it's critical to pay attention to the distance (d) between the test speaker and the DUT microphone, and the test microphone and the DUT speaker when mounting these items inside the fixture. These distances should be equal and, in most cases, d should be from 2mm to 15mm.

Although each component in the system has a set of manufacturer's specifications, the overall system should be treated as a black box and must be characterized as a system. Characterizing the individual parts of the system can also reveal useful insights into the system's overall behavior.

Three basic procedures are generally applicable for any type of characterization:

- Determine system noise.
- Determine a test signal level for a chosen mid-band frequency.
- Using this signal level as a reference level to quantify frequency response.

As an example, let's take part of the system, the speaker and the microphone (with pre-amplifier), as shown in ***Figure 2***. The following test equipment will be needed for characterization:

- Keithley Model 2015 DMM/Audio Analyzer
- Sound pressure level (SPL) meter (preferably with 30–120dB range)

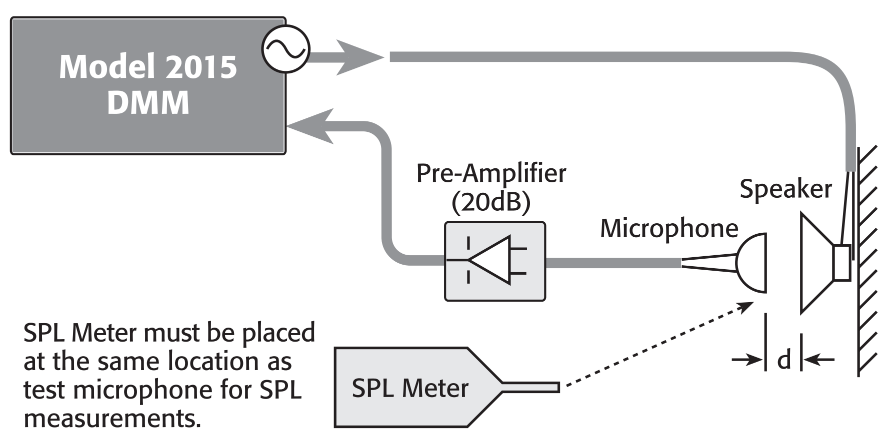

Figure 2. Characterizing a sub-system (speaker and microphone + preamp)

We can simply tape the speaker on a fixture wall and align the microphone directly in front of it. The microphone can be mounted on a supporting stand, but avoid using a microphone support that could deflect sound, e.g., a facing wall. To ensure the same sound pressure levels, the SPL meter's microphone should also be placed at distance *d* (in the same physical location as the microphone occupied) when measuring SPL.

System Noise

With the Model 2015's output turned off, measure V_{RMS} using the Model 2015 and the sound pressure using the SPL meter. This provides a measure of the noise level in the system and the environment. In a reasonably quiet room, the sound pressure should be around 40dB, and V_{RMS} can be a few millivolts as the microphone picks up and amplifies ambient noise.

Test Signal Levels

Generally, it's safer to start with low signal levels to avoid damage to the device [2]. Also, note that the higher the test signal, the more potential it has to create cross-coupling effect and produce resonance conditions inside the test fixture.

Set the Model 2015's output to 1kHz, 50mV and measure V_{RMS} and SPL. Then, increase the stimulus by 6dB (doubled to 100mV). Notice that the V_{RMS} measured should be doubled and the SPL measured should be increased by 6dB—indicating the linear operating region of this subsystem. Repeat this process at least three or four times.

If we arbitrarily assume the Model 2015's output has no limitation and keep repeating this process, we will reach a point where a 6dB increase in the stimulus is no longer met by a 6dB increase in measured signal, indicating the non-linear region of this subsystem. In practice, we may not reach this region because the Model 2015's output has finite levels. On the other hand, we are more interested in lower signal levels.

After verifying the linear region or at least part of it, we should choose a stimulus signal that will produce measured signals at least 20dB above noise; i.e., if the noise SPL is 40dB, the stimulus signal level chosen should produce at least 60dB of SPL. In most manufacturing test applications, the test signal used ranges from tens of millivolts to a few hundred millivolts, producing SPLs from 65dB to 85dB.

Frequency Response

Once the stimulus or test signal level is chosen, we can characterize the frequency response of the system. Performing a frequency sweep with the audio analyzer (2015) over the chosen frequency range will produce a picture of how the system behaves at those frequencies.

In some cases, the system may exhibit frequency-dependent gain. In that case, it's important to verify that it isn't caused by saturation or resonance of any component. The sweep may be repeated at different signal levels. Generally for the entire linear region, the frequency response should remain linear; i.e., the gain should remain constant for a given frequency. Having this frequency response data is crucial to choosing the test frequency range or interpreting the test results.

Overall Test System

Every test system is unique, so it's essential to have a "golden phone" as a benchmark against which to compare subsequent test results. The golden phone is a DUT with known good performance and functionality, or which is otherwise determined to be a suitable reference DUT.

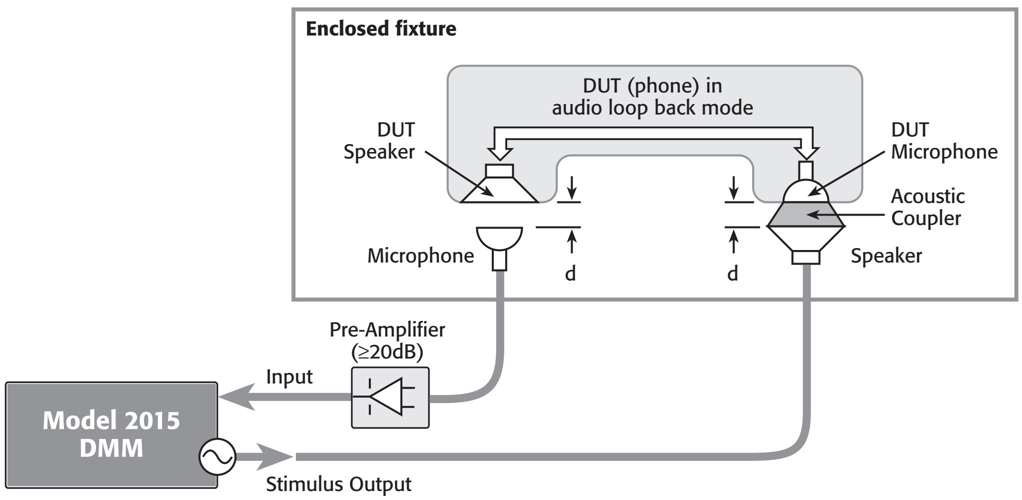

Figure 3. Over-the-air audio test system (DUT in audio loop back mode)

The overall test system can be characterized following the same general procedure described in the speaker and microphone example. In addition to those steps, the cross-coupling should be verified by performing a frequency sweep with the DUT power off or audio loopback mode off.

Test Parameters

Once the characterization process is complete, design and test engineers can typically work together to define the specific test criteria. Having a complete set of characterization data with a golden phone will simplify this task significantly.

If a system's transfer function is perfectly linear, its response to a sinusoidal stimulus signal of *f* Hz will be identical in shape with the stimulus; i.e., in the frequency domain, both the stimulus and the response signals will have the same frequency (*f*). However, if the system isn't perfectly linear, any non-linearity will show up as energy at harmonics of the fundamental (stimulus) [2]. Distortion measurements are one of the most widely used methods of measuring non-linearity, which, in the case of a system with electromechanical transducers, could mean a defective transducer.

THD+N is the most commonly used distortion parameter because it measures the linearity of the DUT while taking into account the effects of both harmonic distortion and noise [2]. A low THD+N value not only indicates the harmonic distortion of the DUT is low, but that noise in the system is also low.

The frequency-dependent nature of the system and the DUT makes it advisable to use a frequency sweep to measure distortion over a selected frequency range. The Model 2015 can perform this sweep while simultaneously acquiring voltage and distortion data. In the example described in this paper, a ten-point frequency sweep took only 542ms to complete, including the GPIB bus-transfer time. ***Figure 4a*** illustrates simultaneous THD+N and V_{RMS} frequency response measurements using the Model 2015.

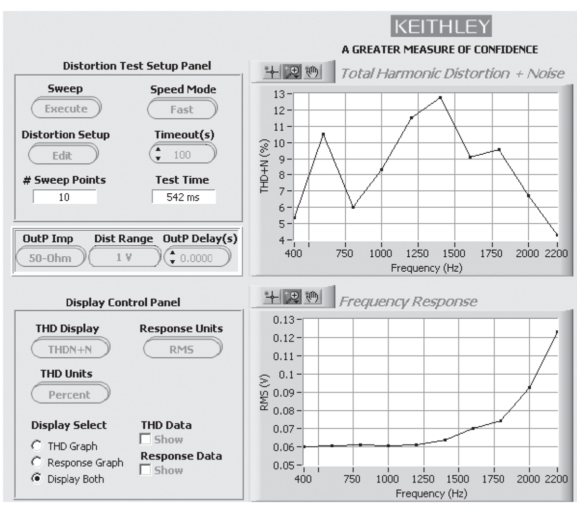

Figure 4a. Ten-point frequency sweep using Model 2015: Measuring THD+N(%) and frequency response (V_{RMS})

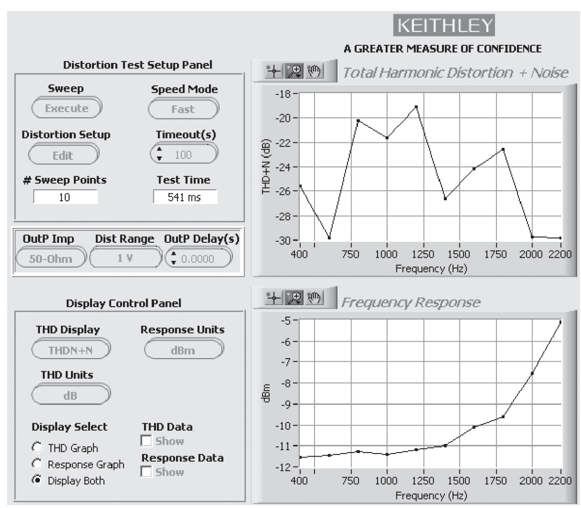

Figure 4b. Ten-point frequency sweep using Model 2015: Measuring THD+N(dB) and frequency response (dBm)

Conclusion

This paper outlines the basic steps to follow to create an audio test system capable of fast, multi-frequency audio quality measurements using inexpensive components and the Model

2015. Although conceptually simple, this type of traditional audio test can reveal subtle component defects that other test methods can overlook.

Although not addressed in this discussion, it's highly desirable to incorporate and design the same test fixture with additional RF design parameters to take RF measurements as well as audio. This will reduce the device handling time and can significantly reduce the overall test time.

References

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