Microphone Intermodulation Distortion Sequence

Introduction

Microphone distortion is very difficult to measure because typically the loudspeaker used to measure the microphone will have greater amplitude response irregularities and distortion than the microphone. By weighting the output signal from the generator with the reciprocal response of the loudspeaker’s fundamental, it is possible to produce a constant sound pressure level versus frequency at the microphone position. If separate test tones are fed to two separate loudspeakers, the loudspeakers’ harmonic distortion will have no influence on the measured intermodulation frequency components. Consequently, only the distortion of the microphones will be measured (Fig. 2). Ref. Brüel&Kjær Application Note “Audio Distortion Measurements”

The purpose of this sequence is to measure the Intermodulation Distortion (IM) of a microphone. Unlike traditional IM distortion measurements performed on loudspeakers, this technique requires the test stimulus to be played through two separate source loudspeakers, one for the playback of the Master (fixed) tone and one for the playback of the Slave (sweep) tone. If a single source loudspeaker were employed, it would generate its own IM products which would be impossible to separate from that of the microphone under test. The sequence’s default stimulus parameters are a 100 Hz tone (Master) and a stepped sine sweep from 10 kHz to 224 Hz (Slave). Stimulus levels should be set to maintain a 4:1 ratio of Master to Slave. Sequence defaults are 94 dBSPL (Slave) and 106 dBSPL (Master).

The test sequence plays the Master Tone through one source speaker (typically a subwoofer) and the Slave through another (typically a full-range speaker or dual-concentric design) with the two speakers placed as physically close to one another as possible. The stimulus playback is captured by the DUT mic and the Recorded Time Waveform is passed through a HarmonicTrak analysis step with its Distortion Tab configured for IM analysis. The final Display Step shows the DUT’s response to the Slave sweep, IM2, IM3 and Total IMD.

Figure 1 - Final Display of Microphone IM Distortion sequence
Software and Hardware Requirements

Software
SoundCheck 14 or newer
SoundCheck Module 2019 IM Distortion

Hardware
Power Amplifier - Listen SCAdm (p/n 4060) or similar
Audio Interface/Microphone Power Supply – Listen AudioConnect (p/n 4050) or similar
Reference Microphone – Listen SCM-3 (p/n 4004) or similar
Fullrange/dual concentric loudspeaker
Subwoofer

Hardware Setup & Calibration

The full-range speaker and subwoofer may either be self-powered or passive. In the example below, the subwoofer is powered and the full-range is passive.

Before proceeding, open the Microphone IM Distortion sequence in SoundCheck. You will likely encounter a Relink dialog, indicating that the Subwoofer Signal Path is not present in your System Calibration. It is recommended that you select “Add to System Calibration” from the “System Signal Path to Use” dropdown and add the Subwoofer calibrated device file to your System Calibration when prompted.

1. Calibrate your reference microphone as described in the SoundCheck user manual. You will use the reference microphone for calibration of the two source speakers.
2. Set up the hardware as shown in Figure 2. Place your calibrated reference mic at the DUT location (typically up to 1 meter away, on the axis of the full-range speaker).
3. Copy the Subwoofer Equalization sequence from the Microphone IM Distortion sequence folder to your SoundCheck/Sequences/Calibration/Calibration-Output folder.
4. Calibrate the full range speaker using the Speaker Equalization sequence as described in the SoundCheck user manual. Calibration frequency range=100 Hz-10 kHz. Use the Source Speaker signal path.
5. Calibrate the subwoofer using the Subwoofer Equalization sequence (run from the System Calibration editor). This process is similar to speaker equalization performed above. Use the Subwoofer Signal Path.
6. Replace the reference microphone with the DUT microphone.

You are ready to start the sequence.
**Sequence Logic**

<table>
<thead>
<tr>
<th>Type</th>
<th>Step Name</th>
<th>#</th>
<th>Out</th>
<th>In</th>
</tr>
</thead>
<tbody>
<tr>
<td>Mes</td>
<td>Enter Master Level</td>
<td>1</td>
<td></td>
<td>// Creates Memory List value for Master (sweep) level</td>
</tr>
<tr>
<td>Mes</td>
<td>Enter Slave Level</td>
<td>2</td>
<td></td>
<td>// Creates Memory List value for Slave (fixed tone) level</td>
</tr>
<tr>
<td>Sti</td>
<td>Intermodulation</td>
<td>3</td>
<td>Source Speaker</td>
<td>// Creates the IM stimulus used in Analysis</td>
</tr>
<tr>
<td>Sti</td>
<td>Master – 10k-224Hz</td>
<td>4</td>
<td>Source Speaker</td>
<td>// Creates the sweep used in Acquisition</td>
</tr>
<tr>
<td>Sti</td>
<td>Slave – 100 Hz</td>
<td>5</td>
<td>Subwoofer</td>
<td>// Creates the fixed tone used in Acquisition</td>
</tr>
<tr>
<td>Acq</td>
<td>Play &amp; Record</td>
<td>6</td>
<td>Subwoofer</td>
<td>DUT Mic</td>
</tr>
<tr>
<td>Ana</td>
<td>IM Distortion</td>
<td>7</td>
<td></td>
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</tr>
<tr>
<td>Dis</td>
<td>IM Distortion Final</td>
<td>8</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure 2 – Typical Hardware Setup for Microphone IM Distortion Sequence**

![Diagram of hardware setup](image)
Notes on Configuring the IM Stimulus

It is important not to inadvertently measure at harmonic multiples of the test tones. This will include unwanted harmonic distortion components. A good rule of thumb is to measure more than N times above the fixed tone (f,) for IM distortion. N is the greatest absolute value of the negative distortion order.

The default sequence is configured to use a sweep from 10 kHz to 224 Hz with R40 resolution and 150 ms step duration plus a fixed tone at 100 Hz. When editing the stimulus parameters, please be aware of the following:

The sequence contains three stimulus steps:

1. **Intermodulation**: Used in the IM Distortion Analysis Step. Contains both the Master and Slave settings.
2. **Master - 10k-224Hz (R40)**: Used in the Acquisition Step (Source Speaker Signal Path). Uses the Master settings.
3. **Slave – 100 Hz**: Used in the Acquisition Step (Subwoofer Signal Path). Uses the Slave settings.

Any edits to default stimulus parameters should be made to both the Analysis stimulus (#1) and Acquisition stimulus (#2 and #3) steps. If the sweep length changes (#1 and #2), make sure to adjust the duration of the Slave tone (#3).

Notes on Signal Path Gain

Your DUT Mic may have a very low sensitivity value, in which case the Max FSD value of your acquisition may not be optimal for accurate IM measurements. If you are using a Listen device that supports Gain Auto-Read and Auto-Range, you should enable them in System Calibration and Acquisition respectively for the DUT Mic Signal Path. If your input signal path does not include a Listen Auto-device, look at the Max FSD value (it is displayed in a table in the Final Display step). If it is below -30 dB re 1 FSD, add as much available signal path gain to get the FSD value as close to (but not to exceed) 0 dB. Whatever gain value is used on the signal path, it should be manually entered into the Gain field of the DUT Mic Signal Path in System Calibration.

Suggestions for Further Sequence Development

This sequence has been designed for simplicity and has been written for a basic two channel system. Ways in which you could modify or further develop the sequence include:

- Modify the sweep range and/or resolution
- Enable Gain Auto-Read (System Calibration) and Auto-Range (Acquisition) if using compatible Listen Hardware (e.g. AudioConnect, SoundConnect 2, AmpConnect)
- Add an Autosave step
- Add a Limits step
- Postprocess the IM Fundamental to derive the DUT Mic’s 1 kHz sensitivity