APPLICATION NOTE AN22

FREQUENCY RESPONSE AND LATENCY OF MEMS MICROPHONES: THEORY AND PRACTICE

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This application note covers engineering details behind the latency of MEMS microphones. Major components of the microphone are described. Amplitude and phase response of a microphone are explained. Phase and group delay of a microphone are defined and compared. Latency measurement is discussed including test setups, capabilities and limitations. Analog and digital PDM microphone frequency response and latency are covered. Some examples of latency and phase-critical applications are given in the end.

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PART ONE: THEORY

MEMS microphone is a transducer that converts audio pressure wave into electric output signal. Following the concept of Fourier analysis, any signal can be represented as a summation of harmonic components. In other words, performance of a microphone can be understood by looking at sine wave signals.

Generally, a microphone can be thought of as a linear system. This means that for a harmonic input pressure signal of a certain single frequency (audio sine wave of frequency f) the output signal is also a sine wave of the same frequency. It is illustrated below for a case of an analog microphone. At a given frequency and a standard input pressure wave amplitude (1 Pa = 94 dB SPL), microphone will have certain output electric wave amplitude. The RMS amplitude value is called sensitivity of the microphone at this frequency. Also, microphone introduces a phase shift to the signal. Same as sensitivity, phase shift also depends on the signal frequency.

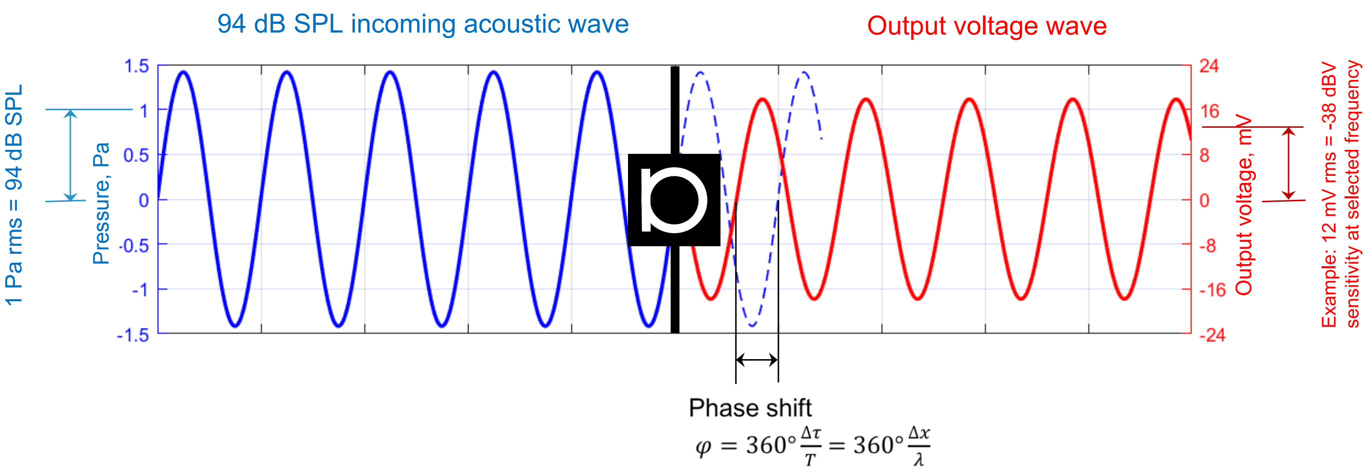
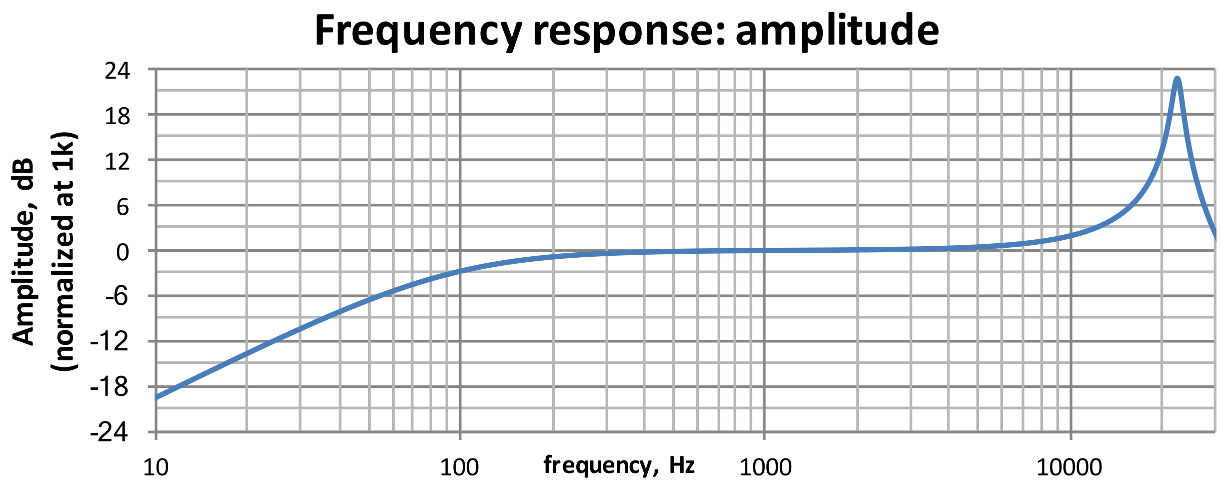


Figure 1. Input and Output of the microphone

Sensitivity and phase shift can be measured at any frequency. Frequencies of interest lay from 20 Hz to 20 kHz, which is the human audible range. The dependency of the output amplitude and phase shift on the frequency of the signal is called amplitude and phase frequency response. Typical frequency response curves are given below (Figure 2). Engineering goal is to have both amplitude and phase frequency response as flat as possible in the widest frequency range so that the output signal is as close as possible to the input signal.

The shape of the frequency response curves comes from the design of the microphone. Moreover, amplitude and phase response of a microphone are coupled together. If a design solution impacts the performance in certain frequency range it will have an effect on both amplitude and phase curves. Knowledge of one parameter allows predicting another one quite accurately.



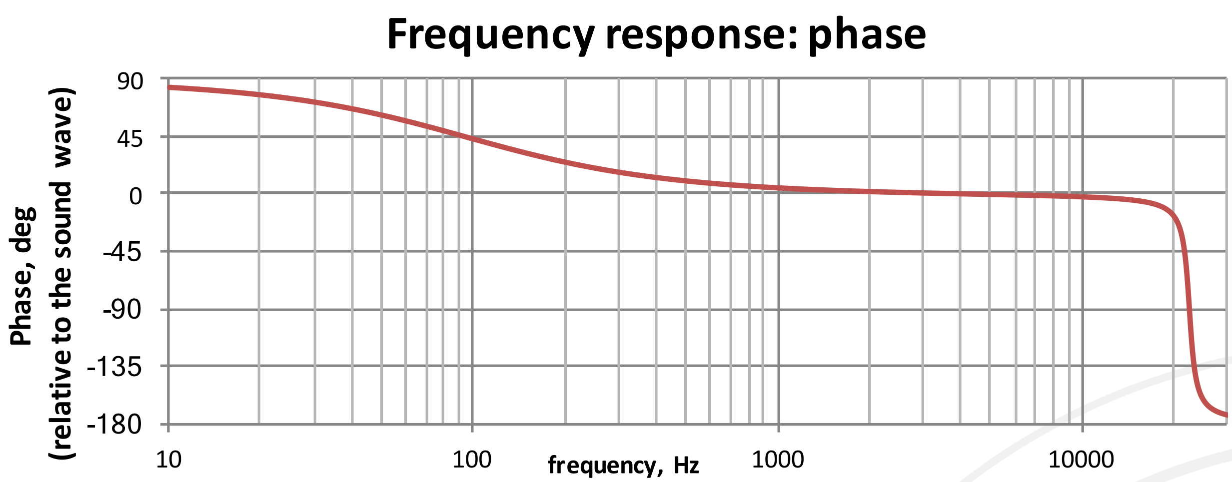


Figure 2. Amplitude and phase response of a MEMS microphone

Frequency response of a MEMS microphone is a combination of acoustic response (defined by package dimensions and MEMS design) and ASIC response. For this discussion, ASIC response is assumed to be flat and does not affect frequency response of a microphone. This assumption is usually valid for analog microphones. Typical MEMS microphone consists of several key elements that define acoustic frequency response (see the Figure 3 below). Each parameter has an impact on the amplitude and phase frequency response. The easiest way to demonstrate it is through lumped-element modelling.

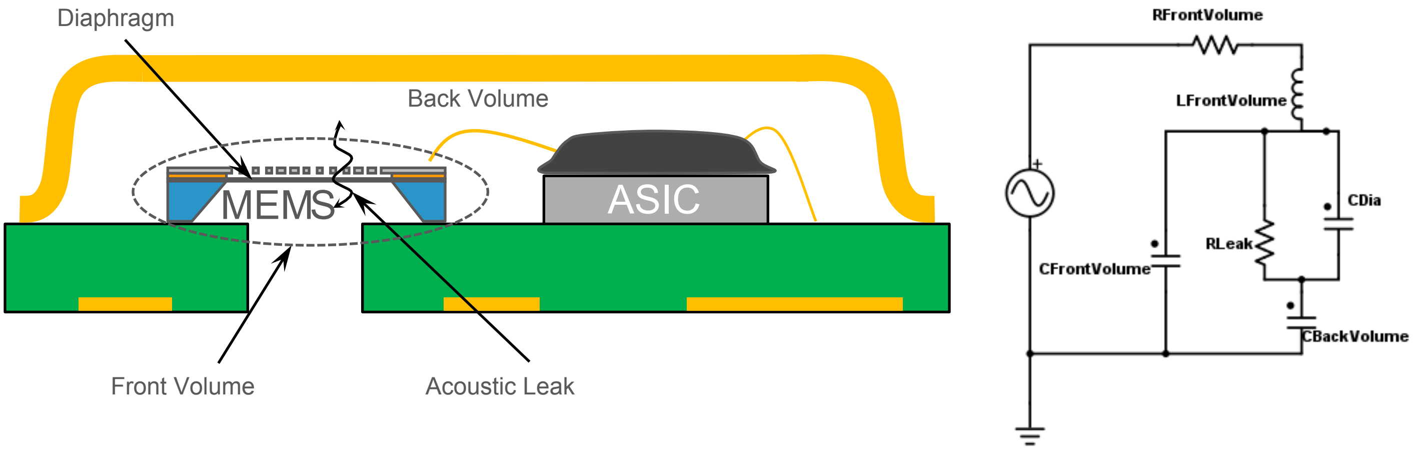


Figure 3. MEMS microphone schematic and simplified lumped-element model.

Microphone frequency response in the audible range can be roughly divided into three bands. In the **low-frequency** band, acoustic response represents a single-pole high pass filter. The filter is formed by the resistance of acoustic leak through the MEMS element and the capacitance or compliance of the back volume. The RC constant of the filter translates into the cut-off frequency, which is the only parameter that describes the frequency response of the microphone in the low frequency band. Accordingly, amplitude response has 20 dB per decade slope and -3 dB value at the cut-off frequency. Phase response curve will start at 90 degrees on the low-frequency side, will pass through 45 degrees point at the cutoff frequency and will approach 0 degrees at higher frequencies.

In the **medium-frequency** range, microphone frequency response is flat for both amplitude and phase. Phase shift is close to zero and sensitivity of the microphone is usually specified at 1 kHz. 1 kHz sensitivity value comes from MEMS (ex. size of the diaphragm, back plate design), acoustic design (back volume) and ASIC gain.

Acoustic package resonance occurs on the **high-frequency** side of the response. Amplitude response has a resonance peak and phase response exhibits 180 degrees inversion at the same resonance frequency. Acoustic resonance occurs in the front volume and resonance frequency shifts with the size of it. This is the reason why bottom-port microphones have resonance at higher frequencies then top-port microphones.

Latency is a critical parameter for modern audio applications. When arbitrary signal is processed by a transducer it can be delayed and its shape in time and frequency domain is altered. In some cases, delay can be calculated from the phase response.

So-called phase delay is defined only for infinite sine wave signals (see the figure below). This parameter is proportional to the phase shift and can only have a value in the range from negative half period to positive half period. Phase delay can be positive or negative.

$$\Delta\tau_{phase} = -\frac{1}{360°}\,\frac{\varphi(f)}{f}, \quad \varphi(f) - \text{phase response}$$

$$-T/2 < \Delta\tau_{phase} \leq T/2$$

Group delay is defined for AM-modulated signal pulses or wave packets. Signals of this kind are time and frequency limited. They consist of a carrier wave of certain high frequency and a slow-changing envelope signal. A short whistle is a good example. When the signal undergoes transduction it will stay AM-modulated with the same carrier frequency due to the linearity of the microphone. However, it will be delayed and the shape of the envelope can change (see Figure 4 below). Group delay has a physical meaning of the time gap between the maxima of input and output pulse envelopes. It can be shown that group delay of pulse with certain carrier frequency is the negative derivative of the microphone phase response at that frequency. Note that the phase response of a MEMS microphone is a monotonically decreasing function. This means that the derivative is negative at any frequency and group delay is always positive. Indeed, output of the microphone always shows up after the input arrives, which makes sense from cause-effect relationship perspective. Group delay of a microphone cannot be negative. Generally, group delay of a transducer is the most meaningful parameter for latency characterization.

$$\Delta\tau_{group} = -\frac{1}{360°}\frac{d\varphi(f)}{df}, \quad \varphi(f) - \textit{phase response}$$

$$\Delta\tau_{group} > 0$$

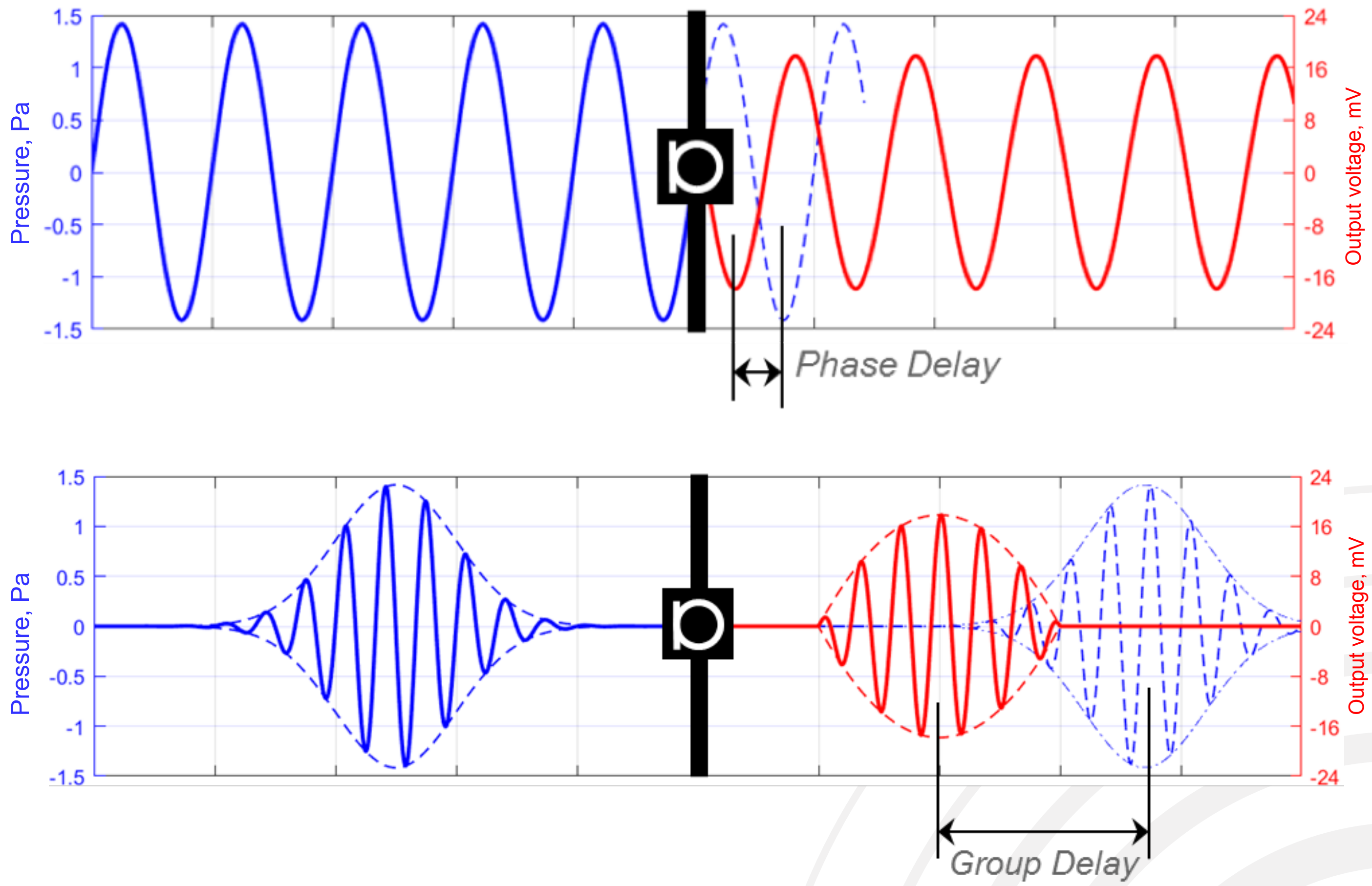


Figure 4.
Phase delay of a sine wave (top) and group delay of an AM modulated pulse (bottom).

Phase and group delay of a transducer can be derived from the phase frequency response. Typical frequency response of a MEMS microphone was explained above. Figure 5 below shows typical curves for delay of a microphone. Note that group delay only has positive values unlike phase delay. Also, group delay is large in low-frequency range due to low frequency roll off of a microphone. Then it reaches the minimal value in the midband. Finally, group delay has a peak at the package acoustic resonance frequency due to phase inversion.

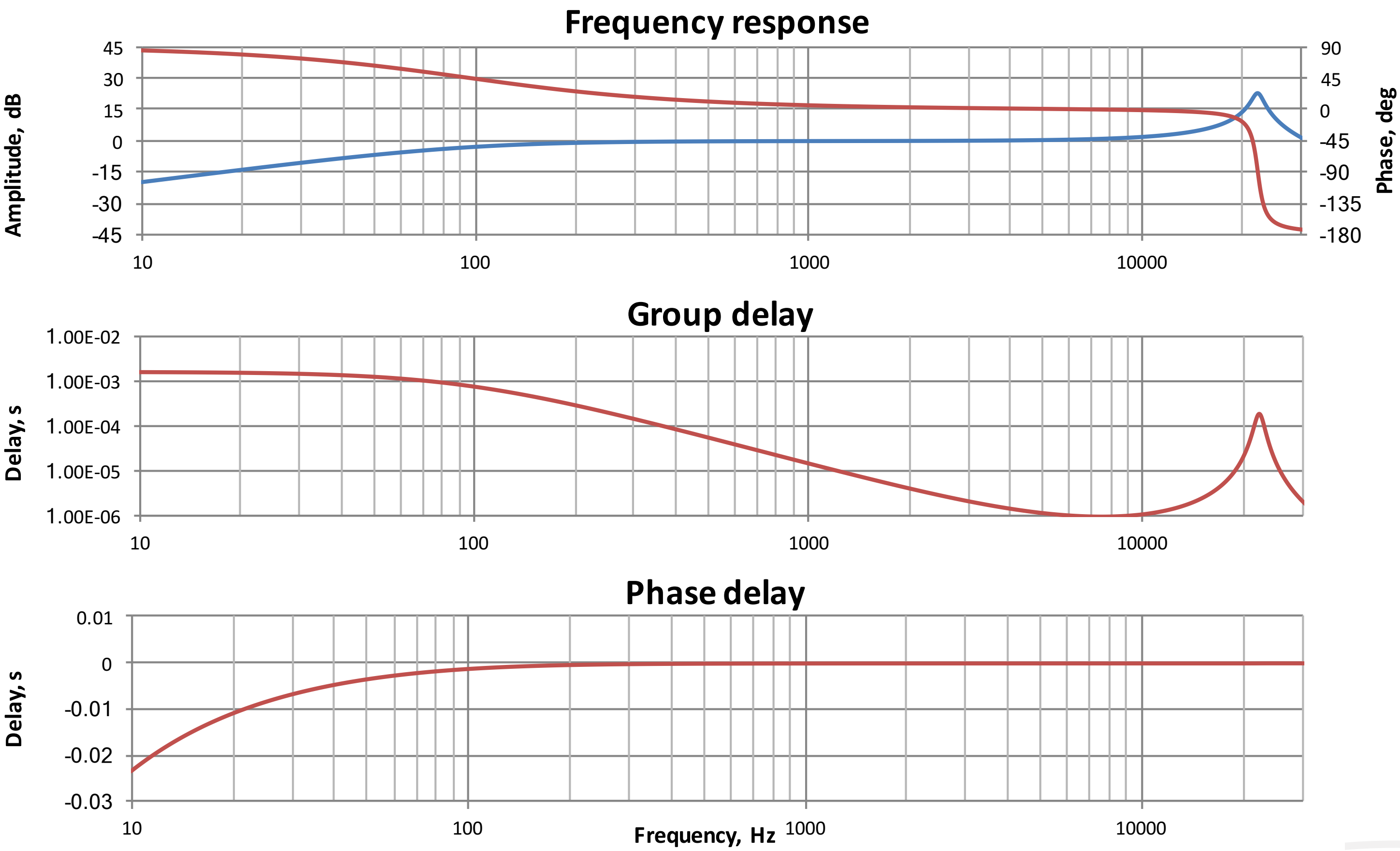


Figure 5. Frequency response, group and phase delay of a typical MEMS microphone.

PART TWO: MEASUREMENTS

Delay of a microphone is derived from phase frequency response. Frequency response of a MEMS microphone is usually measured by acquiring audio signals side-by-side with a high-quality reference microphone. It is critical to apply correction of the reference microphone frequency response in order to measure frequency response of the MEMS microphone relative to the audio wave (see Figure 6 below).

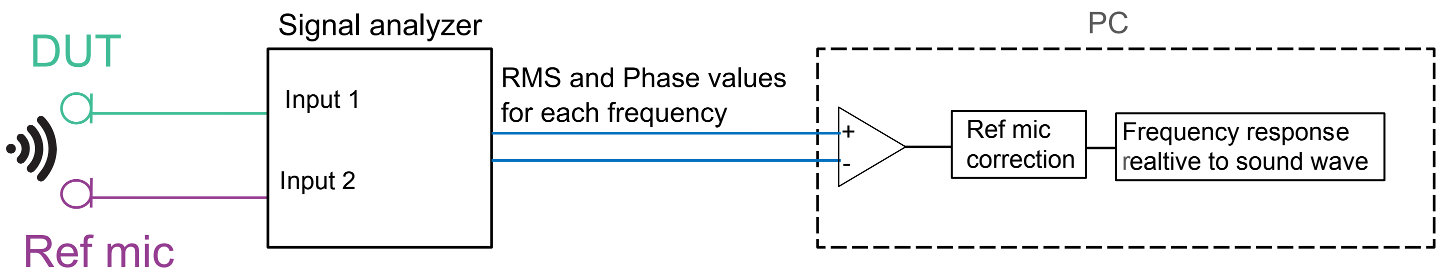


Figure 6. Signal path diagram for frequency response measurements.

From the acoustic point of view, measurement is usually done in two steps: low-frequency data is collected using pressure cavity test setup and high-frequency data is collected in the anechoic chamber (see Figure 7 below). Pressure cavity setup might be valid up to 2-3 kHz. Audio wavelength must be much larger than the cavity dimension. In this case acoustic pressure can be assumed to be uniform in the cavity. On the other hand, anechoic setup allows avoiding reflections in the chamber and free-filed acoustic measurement can be done in the high-frequency range, from 1 kHz up to 20 kHz. For anechoic setup, it is critical to position MEMS microphone under test as close as possible to the reference microphone. The spacing between them must be much smaller than the audio wavelength so that both devices are exposed to the same audio field. Also, signal-to-noise ratio must be considered during the test. The signal must be not too low to be well above the noise floor and not too high to avoid distortion.

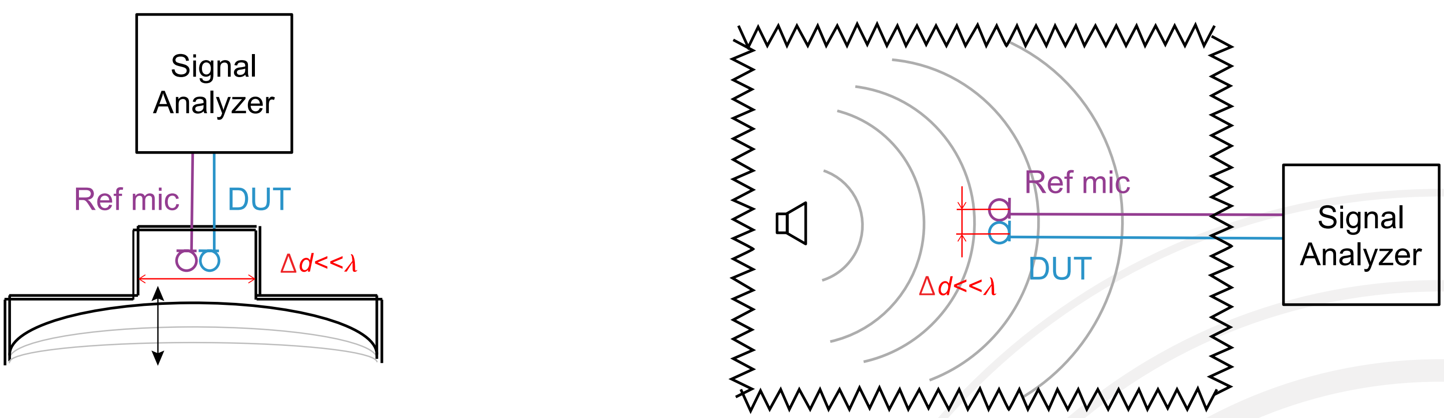


Figure 7. Acoustic measurement setups: pressure cavity (left), anechoic chamber (right).

PART THREE: EXAMPLES

Figure 8 below depicts an example of measured frequency response and delay curves for a typical analog microphone. In this example, ASIC response can be assumed flat, the features of the microphone frequency response come from acoustic design. Please note that the group delay is calculated as a numerical derivative of experimental phase response data.

This results in high group delay uncertainty. In other words, it is expected that the group delay curve looks so noisy. Minimal group delay value in the mid-frequency range is ~5 us.

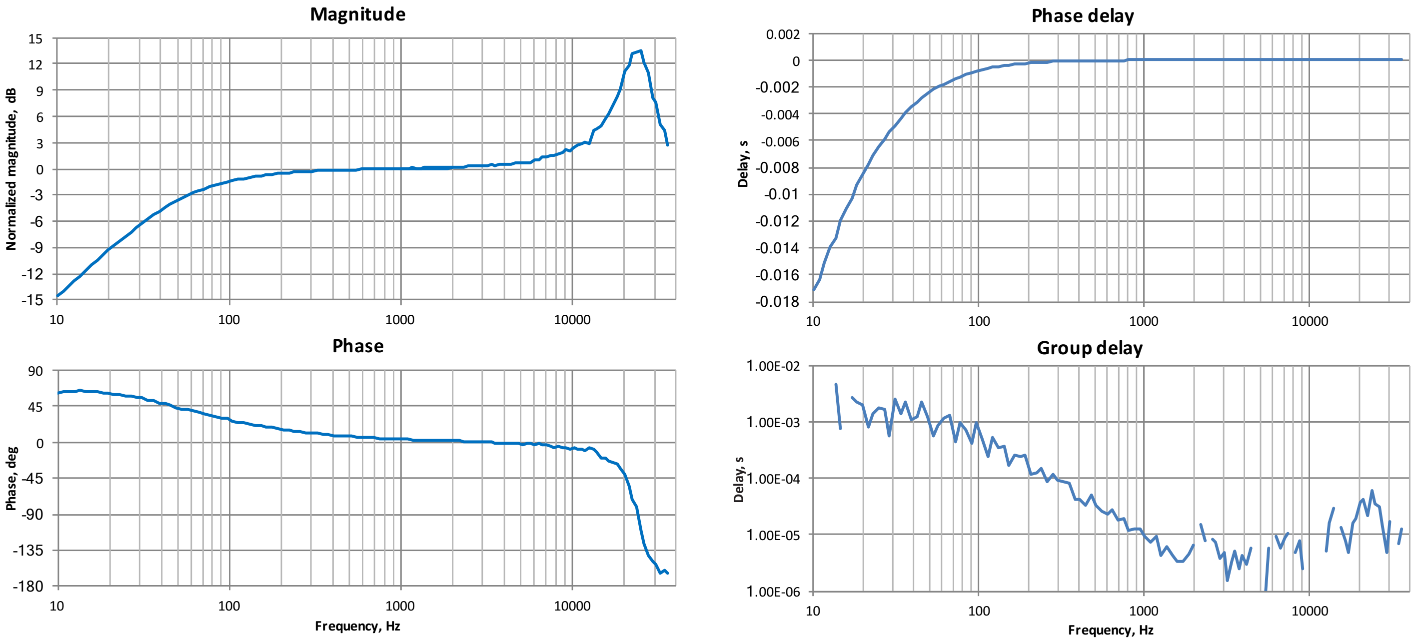


Figure 8. Typical Frequency response and delay curves of an analog microphone

Digital ASICs introduce delay. It takes a certain amount of clock cycles to process a signal. This effect alters the phase response curve and dominates the delay in the mid-frequency range. Figure 9 below shows a frequency response and group delay of a typical PDM microphone Microphone-level frequency response is a combination of acoustic and ASIC response. In this example, ASIC has low frequency roll-off close to acoustic LFRO corner. Both filters overlap and result in low frequency mic-level response that cannot be expressed as a one-pole high pass filter. Constant digital ASIC delay results in linear ASIC phase response (note that frequency axis has logarithmic scale). Acoustic resonance in high-frequency range introduces a peak in mic-level amplitude response and phase inversion at corresponding frequency.

Microphone group delay is dominated by LFRO effect up to 1 kHz. Minimal delay in mid-frequency range is dominated by the ASIC digital delay. Group delay peaks in high-frequency region due to acoustic package resonance.

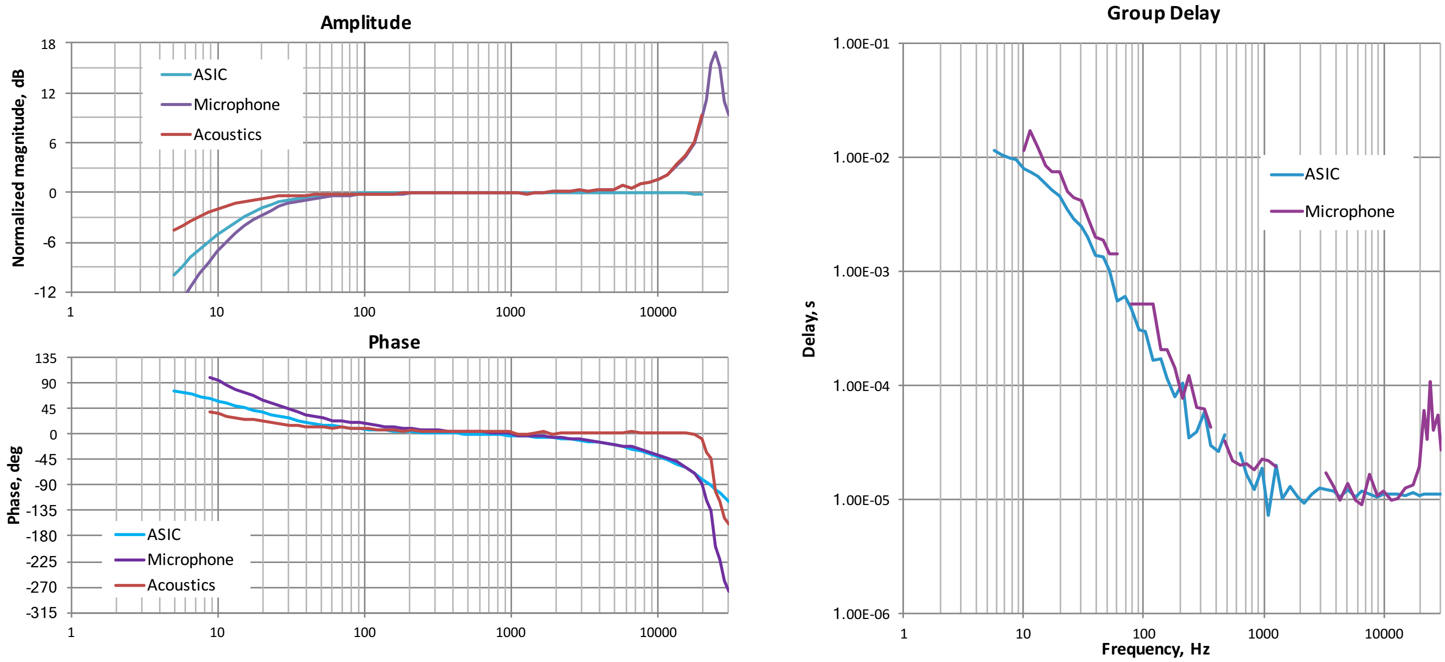
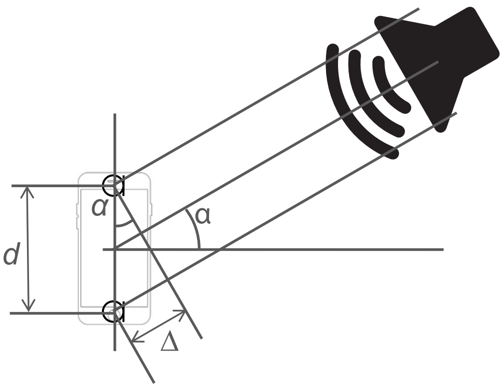


Figure 9. Typical Frequency response and delay curves of a digital PDM microphone

PART FOUR: APPLICATIONS

Consider the simplest beam-forming algorithm. By measuring the delay between the outputs of two microphones on the device, it is possible to compute the direction of the sound source (see the Figure 10 below). For example, signal delay between microphones is maximal when the sound is coming along the vertical axis (along the line connecting the microphones). Alternatively, there is no difference in the time of arrival for a signal that travels along the horizontal axis (perpendicular to the line connecting the microphones).



$$\alpha = \mathrm{asin}\left({}^{\Delta}/_{d}\right) = asin\left(\frac{\tau c}{d}\right) = asin\left(\frac{\varphi}{360° f}\frac{c}{d}\right)$$

$\alpha - sound\ source\ angle$
$\Delta - path\ length\ difference$
$d - microphone\ spacing$
$\tau - signal\ time\ delay\ between\ microphones$
$c - speed\ of\ sound$
$f - audio\ frequency$
$\varphi - phase\ difference\ between\ microphones$

Figure 10. Simplest beam former.

The calculation above will only work if the microphones are identical. The algorithm won't work in case microphones introduce unequal phase shift from input to output. Even if the same microphone model is used there is always manufacturing variability that leads to specification tolerance. Beamforming algorithms usually work in the range up to 1 kHz. Corresponding microphone specification is low frequency roll-off. Microphone phase mismatch is small for well-matched microphones with tight LFRO specification. This allows achieving high angular resolution for the beam former. On the contrary, wide LFRO specification allows microphones to have big phase mismatch that leads to poor angular resolution of the algorithm (see Figure 11 below). Table 1 gives a performance example for operation at 200 Hz.

	Nominal Response	Case 1 Small Variation	Case 2 Medium Variation	Case 3 High Variation
Magnitude at 35 Hz, dB	-3	±0.4	±0.75	±1.2
LFRO (-3 dB point), Hz	35 Hz	±3	±6	±10
Max phase mismatch 200 Hz, deg	N/A	1.7	3.3	5.5
Beam former angular error, deg	***0***	***±5***	***±9***	***±15***

Table 1. Relation between beam former accuracy and microphone matching

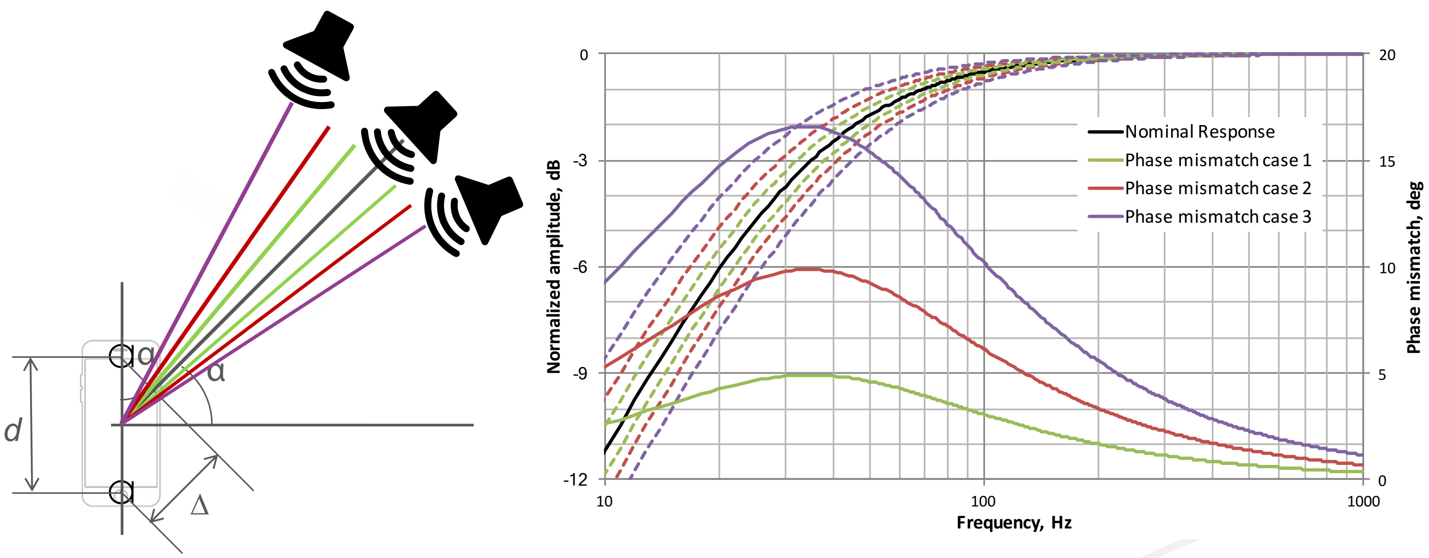


Figure 11. Beam former accuracy limitations due to microphone matching

Active noise cancellation (ANC) is another common application where microphone latency is critical. The concept behind it is straightforward: unwanted audio noise is sensed with a microphone, inverted and combined with desired audio signal. Combined signal is played through the speaker. This way unwanted noise is cancelled and only desired audio is left (see Figure 12 below).

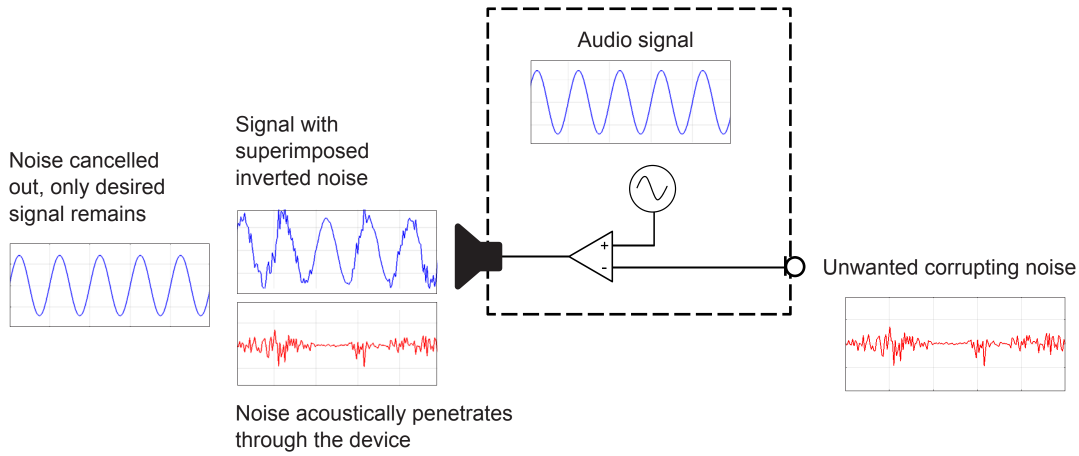


Figure 12. The concept of active noise cancelation (ANC)

Microphone and ANC circuit group delay must be short enough so that inverted noise overlaps with original noise and cancelling is effective. The human ear only perceives cancellation if temporal mismatch is under 10%. If a microphone introduces a significant group delay, cancellation will be ineffective (see Figure 13 below). Active noise cancellation algorithm requires microphones with group delay as low as possible. Note that this is a different requirement from the beam former case where relative microphone mismatch has to be minimized and absolute microphone delay is not that critical.

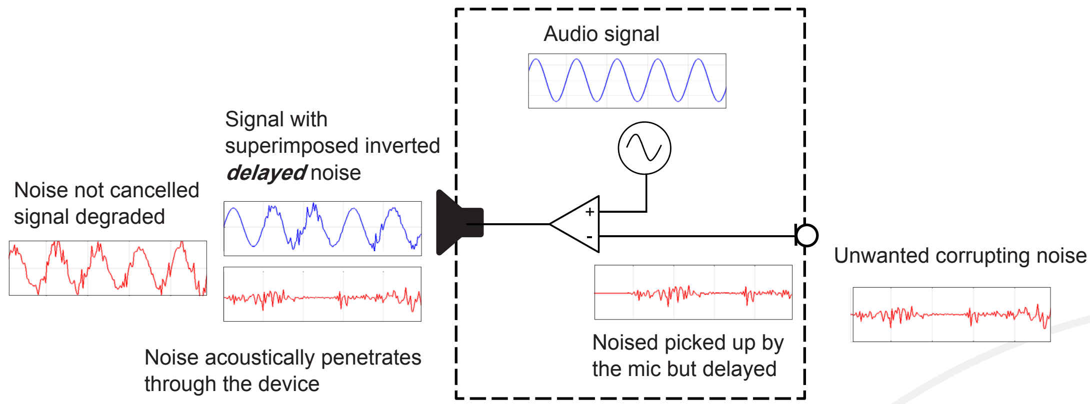


Figure 13. Active noise cancellation does not work with a long delay microphone.

The examples above demonstrate simplified algorithm applications. Actual signal processing done in modern consumer electronics devices, such as Blind Source Separation, is far more advanced but still comes down to the same concepts. The general goal of such algorithms is sometimes referred to as 'The Cocktail Party problem'. Multiple sound sources are active at the same time and are recorded with a mic array. The goal is to identify each individual source and optimize performance accordingly by directional selection, suppressing noise, cancelling echoes and other means. Microphone specifications vary from one application to another but general requirement is common: microphones must be as identical as possible and group delay must be as short as possible.

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