RIR Estimation for Synthetic Data Acquisition

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ABSTRACT - Automatic Speech Recognition (ASR) works best when the speech signal best matches the ones used for training. Training, however, may require thousands of hours of speech, and it is impractical to directly acquire them in a realistic scenario. Instead, we estimate Room Impulse Responses (RIRs), and convolve speech and noise signals with the estimated RIRs. This produces realistic signals, which can then be processed by the audio pipeline, and used for ASR training. In our research, a limited corpus of speech data as well as noise sources is recorded and the RIR at 27 positions is determined using a variety of methods (chirp, MLS, impulse, and noise). The convolved RIR with the “clean speech” is compared to the actual measurements.

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Introduction – Problem statement

- To validate Automatic Speech Recognition a large corpus of data in various acoustical environments is required.
- Testing is time consuming, difficult to replicate and subject to corruption by noise.
- Auralization has been proposed but the fidelity to real rooms is not accurate enough
- There are no commercial solutions which are capable of performing high resolution data synthesis for speech recognition.
- Our proposed solution is to measure room impulse response and convolve the test vectors to synthesise measured data to the fidelity required to accurately test ASR
- The other question we want to answer is what is the best way to measure the RIR.

Introduction – Proposed approach

Three different types of excitation signals:

- Maximum length sequences (MLS)
- Sine sweeps
- White noise.

Validation signals

- 8 sentences: 4 male and 4 female from ITU-T p.863
- (ITU) P.50 signal.

Test signals

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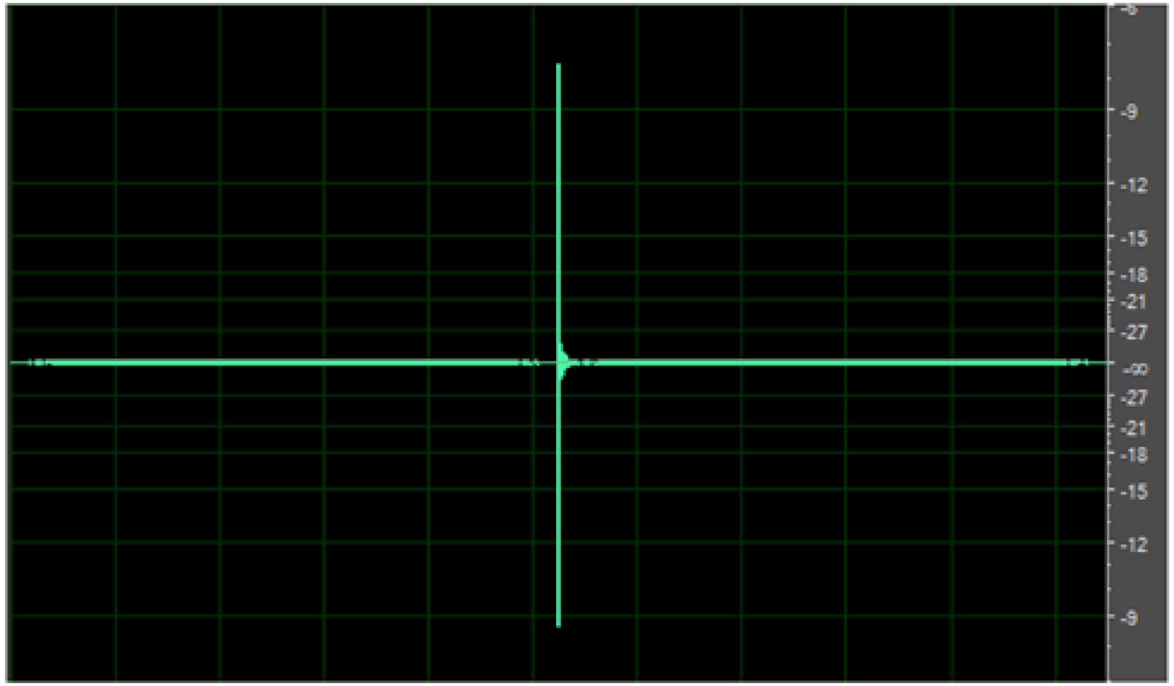
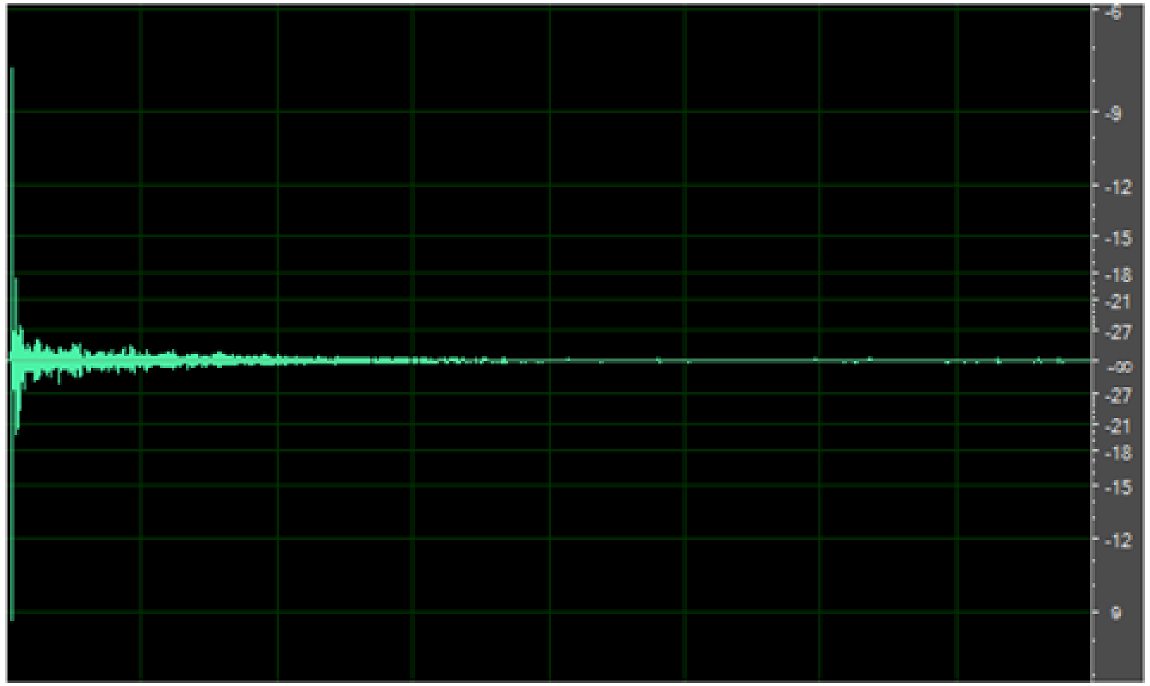
Test signal – MLS

- Maximum length sequences (MLS) of length exactly 218 - 1 samples
- corresponds to approximately 5.5 seconds at sample rate of 48kHz.
- The MLS signal, after being recorded, is deconvolved into an impulse response via time-reversal convolution between the source MLS and the recording
- Microsoft’s audio lab RT60 ~ 300ms Therefore the IR is trimmed to contain 16000 samples (1/3 second).

Test signal – MLS – deconvolved signal



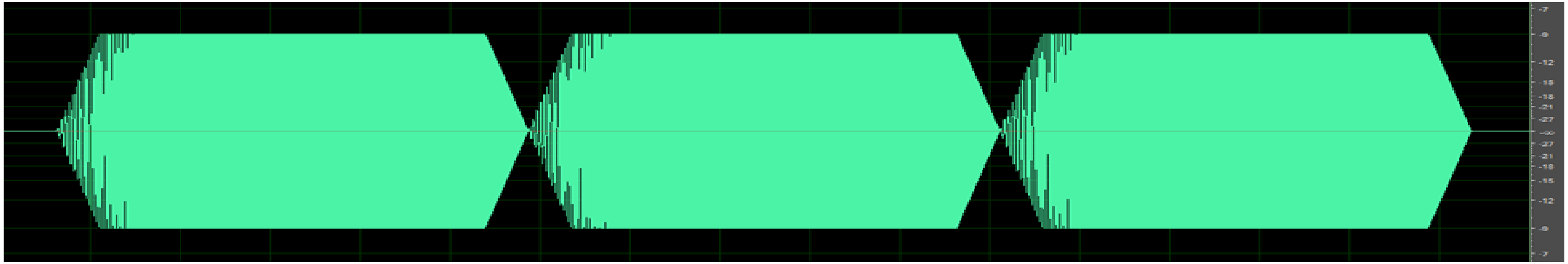


- complete deconvolved signal
- trimmed deconvolved signal

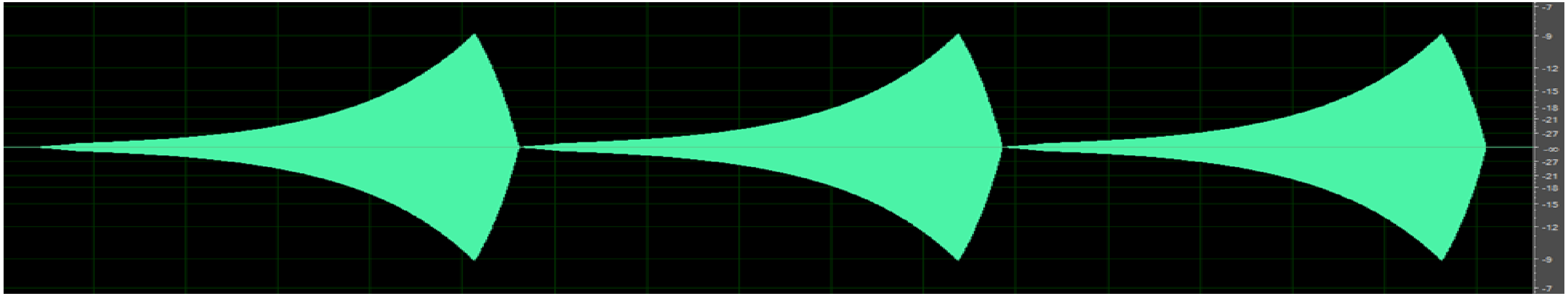
Test signal – Swept Sine

- sine signal of length of 218 samples.
- faded in and out at the end 0.5s in order to reduce distortion of the speakers due to startup transients.
- made continuous at the end points using a 1µHz bi-directional binary search pattern
- the exponential nature of the sweep is corrected by an exponentially growing amplification of the signal in the time domain before deconvolution
- immunity to room reflections and harmonic distortion of the components

Test signal – Swept sine with amplitude



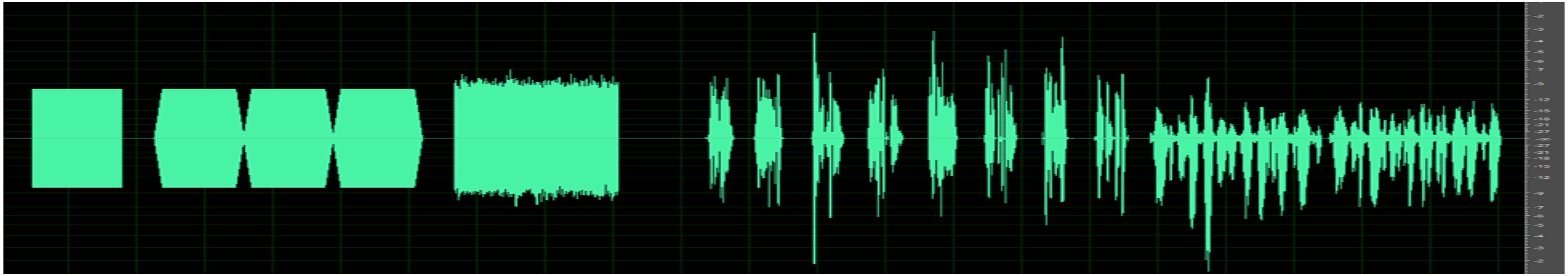
Swept sine signal for test



Amplitude corrected swept sine signal for deconvolution

Test signal – White noise & Speech

- White noise is generated for 1 second.
- The speech signals are inserted into the file following the white noise. The full excitation plus speech signal is shown below.
- MLS, swept sine, noise, real speech, finally p.50 signal

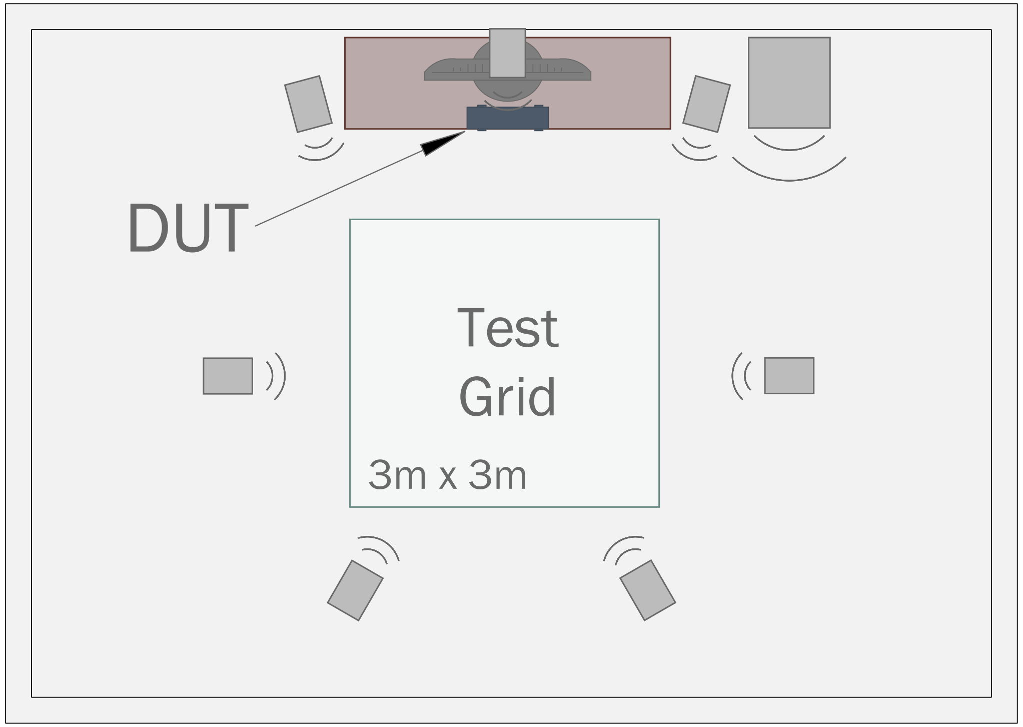


Room and Test setup

DEVICES & STUDIOS

Room Set-up

- 27 positions in a 3x3x3m grid ranging from 1m in front of the device to 3m back and 3m high.
- For each position each of the 7 JBL speakers also play the test signal as the RiR will change for each robot position



Data analysis

- Room Impulse Responses (RIR) are calculated by deconvolution
- RiRs are convolved with the speech section of the test signal, producing synthetic data for each position and configuration. 6510 RiRs were generated for analysis.
- A 8192 point FFT analysis compares the synthetic data to the directly measured data
- the mean-error is computed across 6 bands using the formula below.
  - Narrowband (300-3.4kHz),
  - Wideband (50-7kHz),
  - Super Wideband (50-12kHz),
  - Subwoofer Band (20-120Hz), and
  - Full Band (all points).

$$Error_{mean} = \sqrt{\frac{\sum_{i=1}^{N} E_i}{N}}$$

Results

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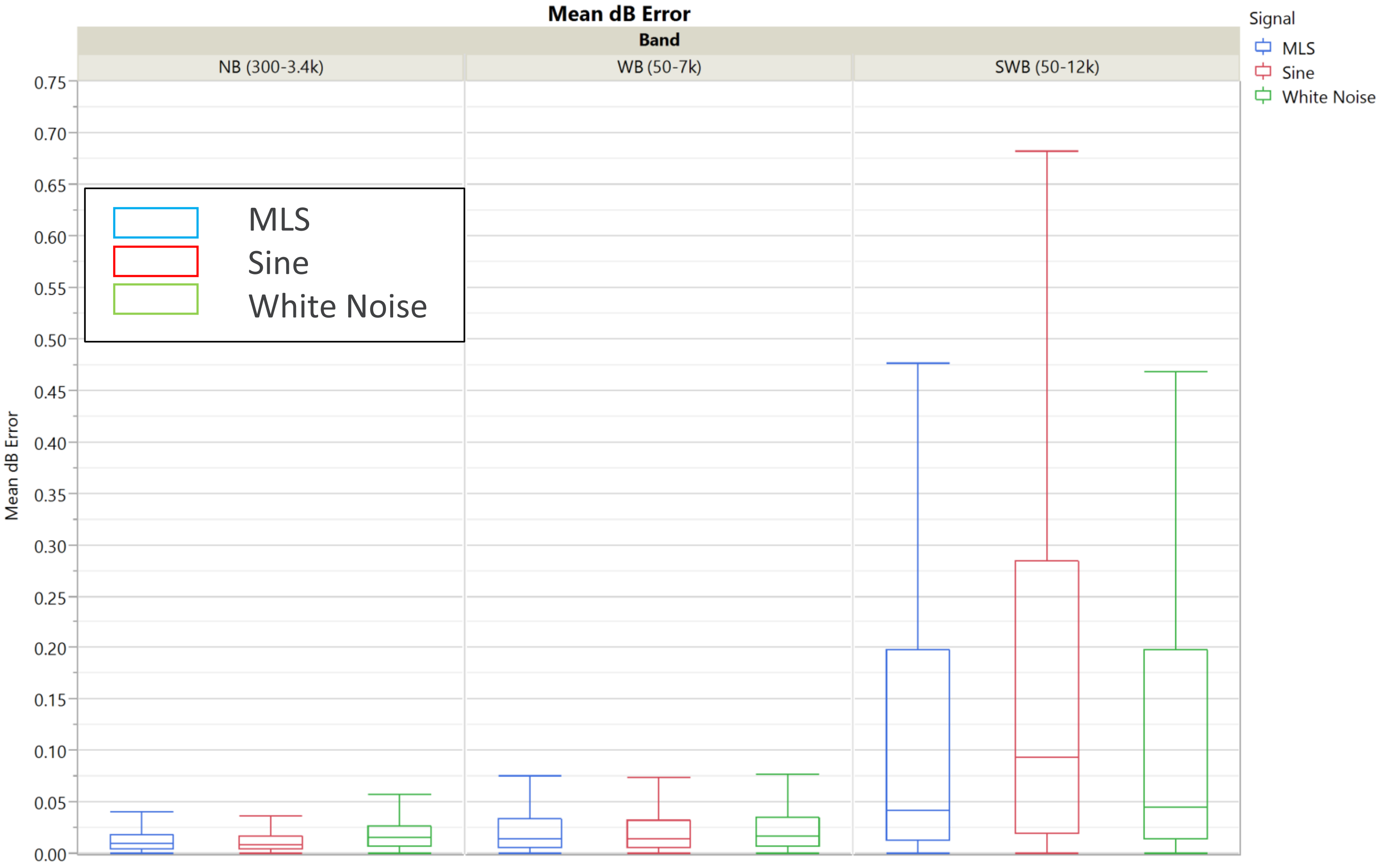
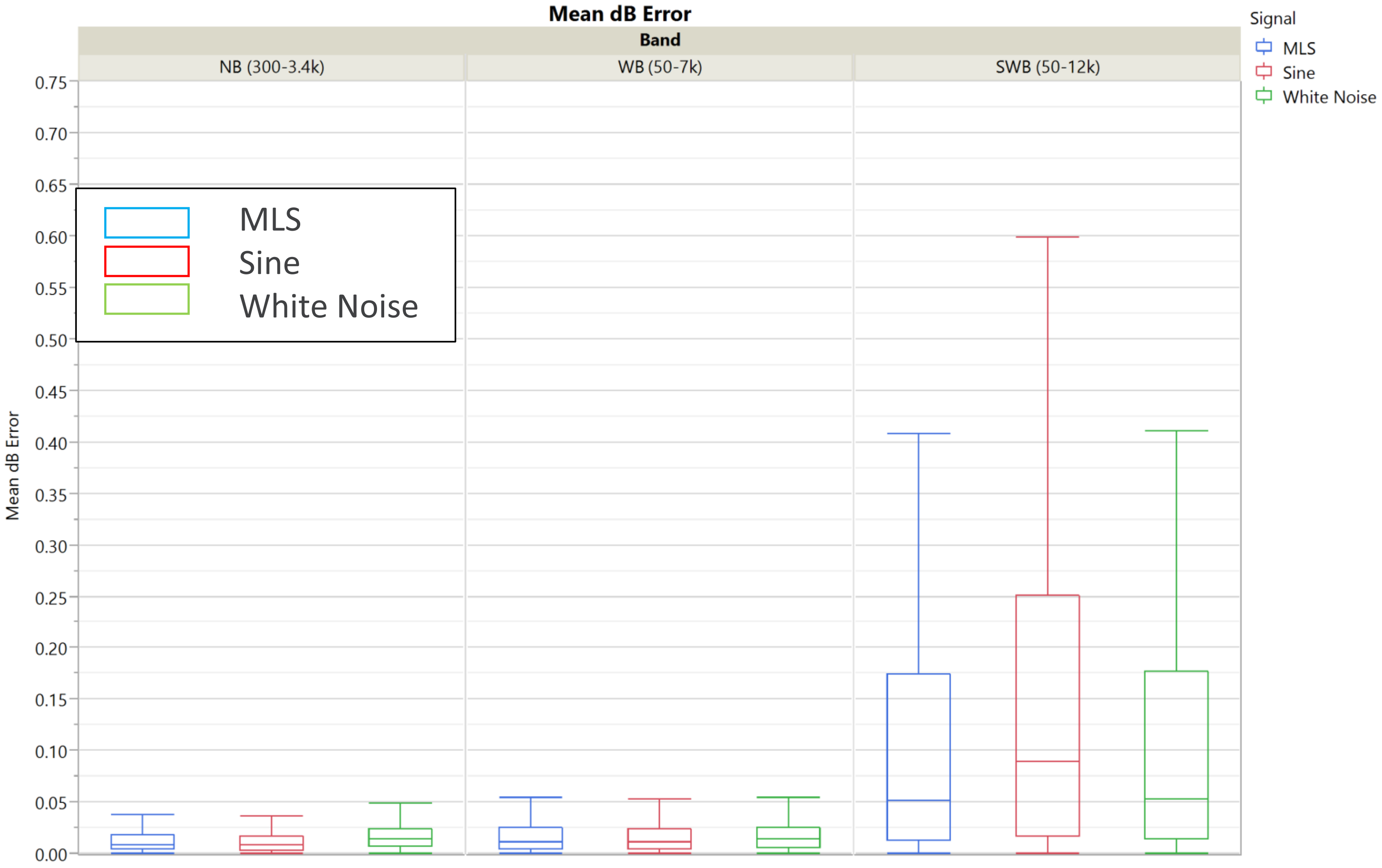
All Mics – All positions

- Overall summary
- Good in telephony bands, max error <0.7dB (outliers hidden)
- Error <0.5dB for MLS (excluding hidden outliers)
- MLS best method in Narrow Band, and is never worse than White Noise



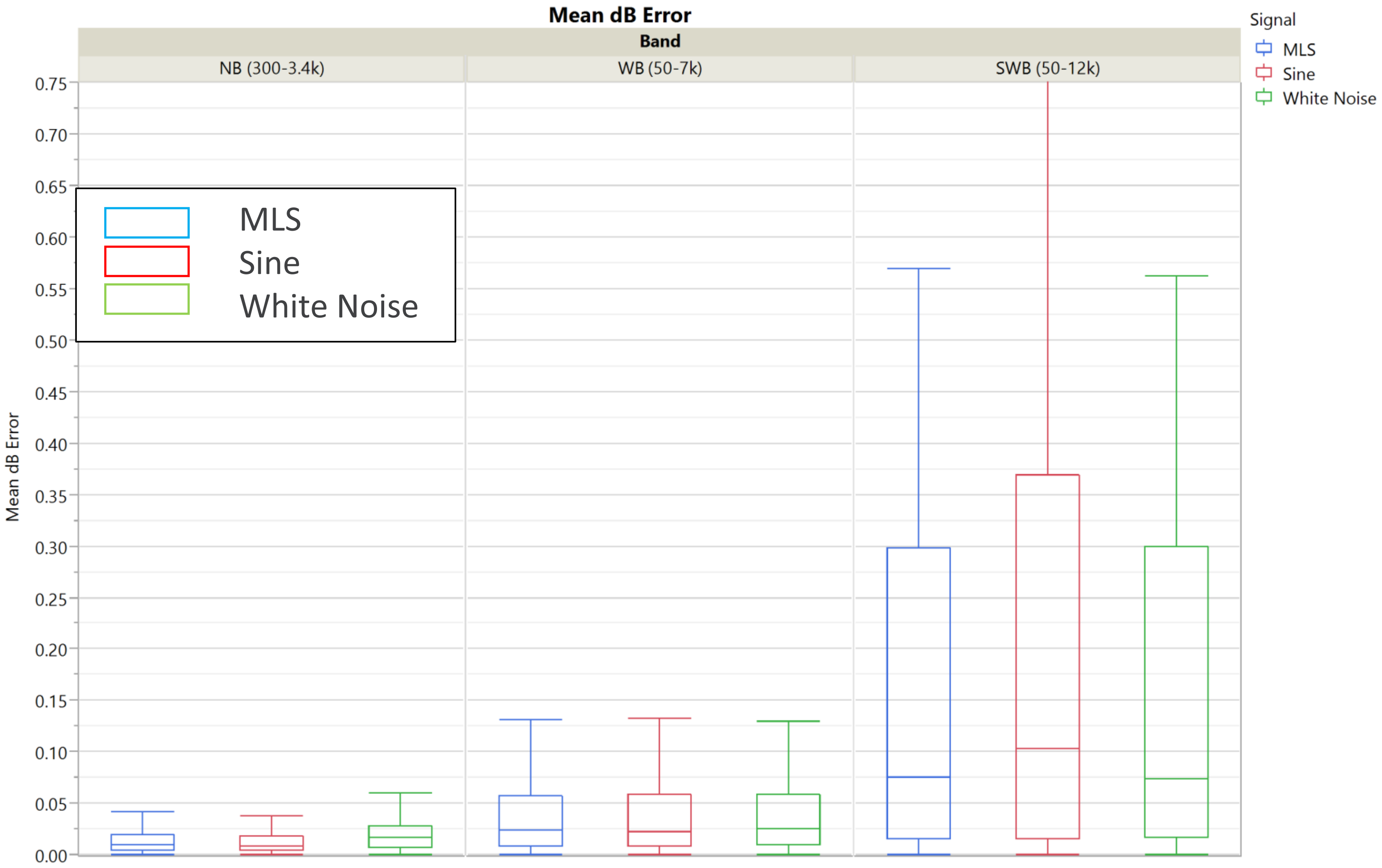
Production Array Mics – All positions

- Device mics only
- Excluding subwoofer
- MLS better in Narrow Band marginally
- Very good performance – average error of 0.05dB for MLS in Super-Wide Band



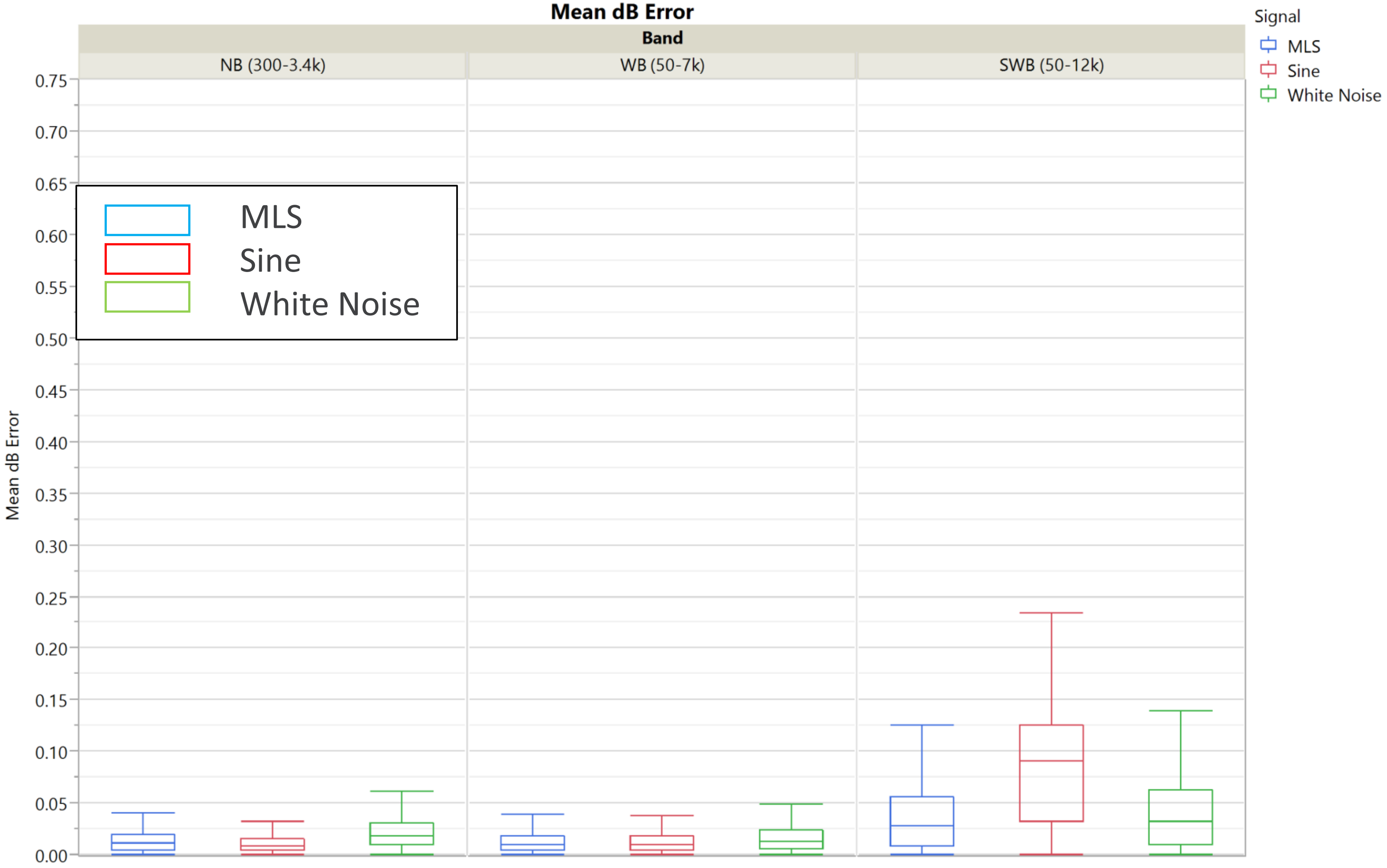
Prototype Mics– All positions

- Device mics only
- Excluding subwoofer
- Sine quite bad, exceeding plot range!



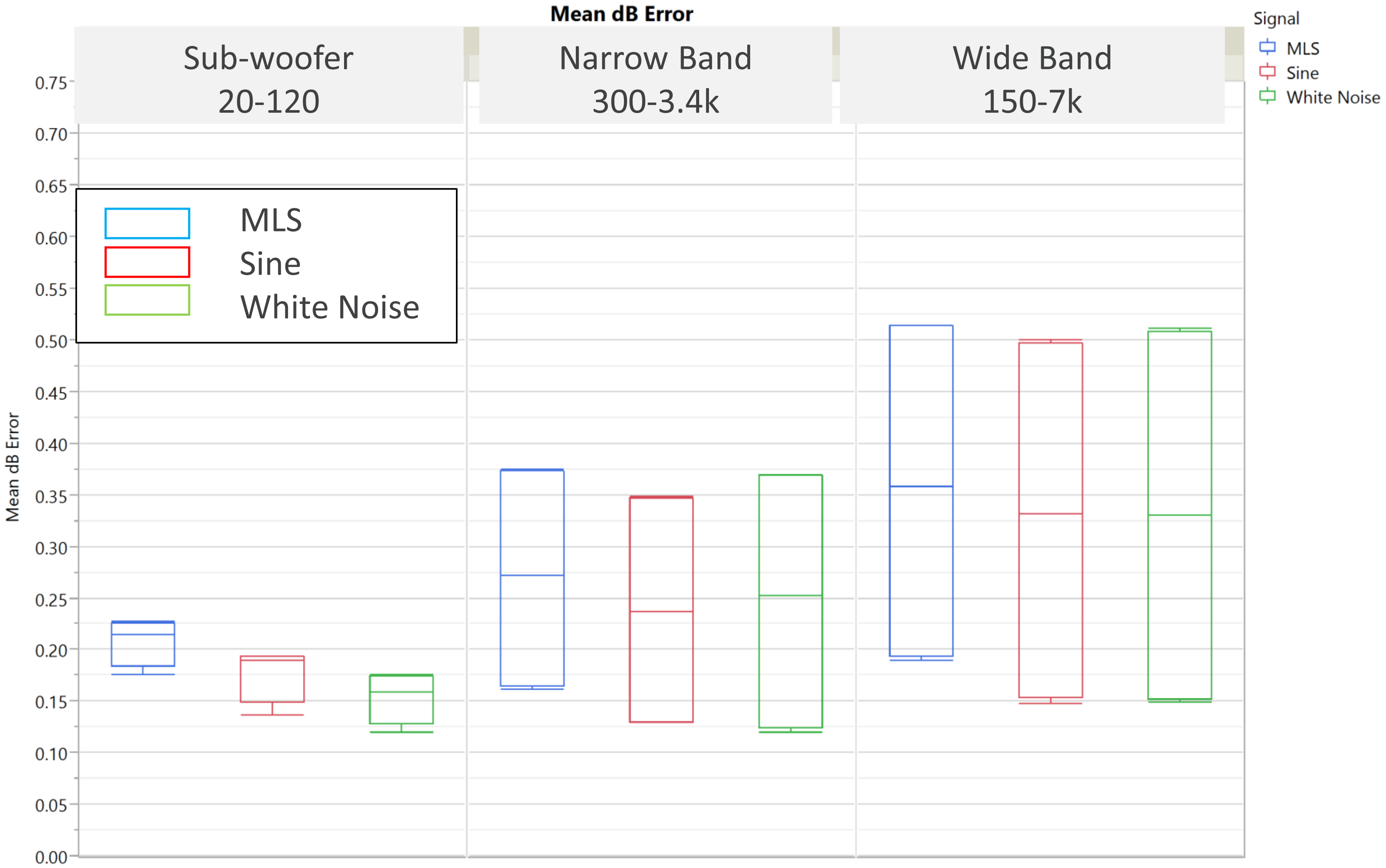
Reference Mics – All positions

- Excluding subwoofer
- MLS better in all categories

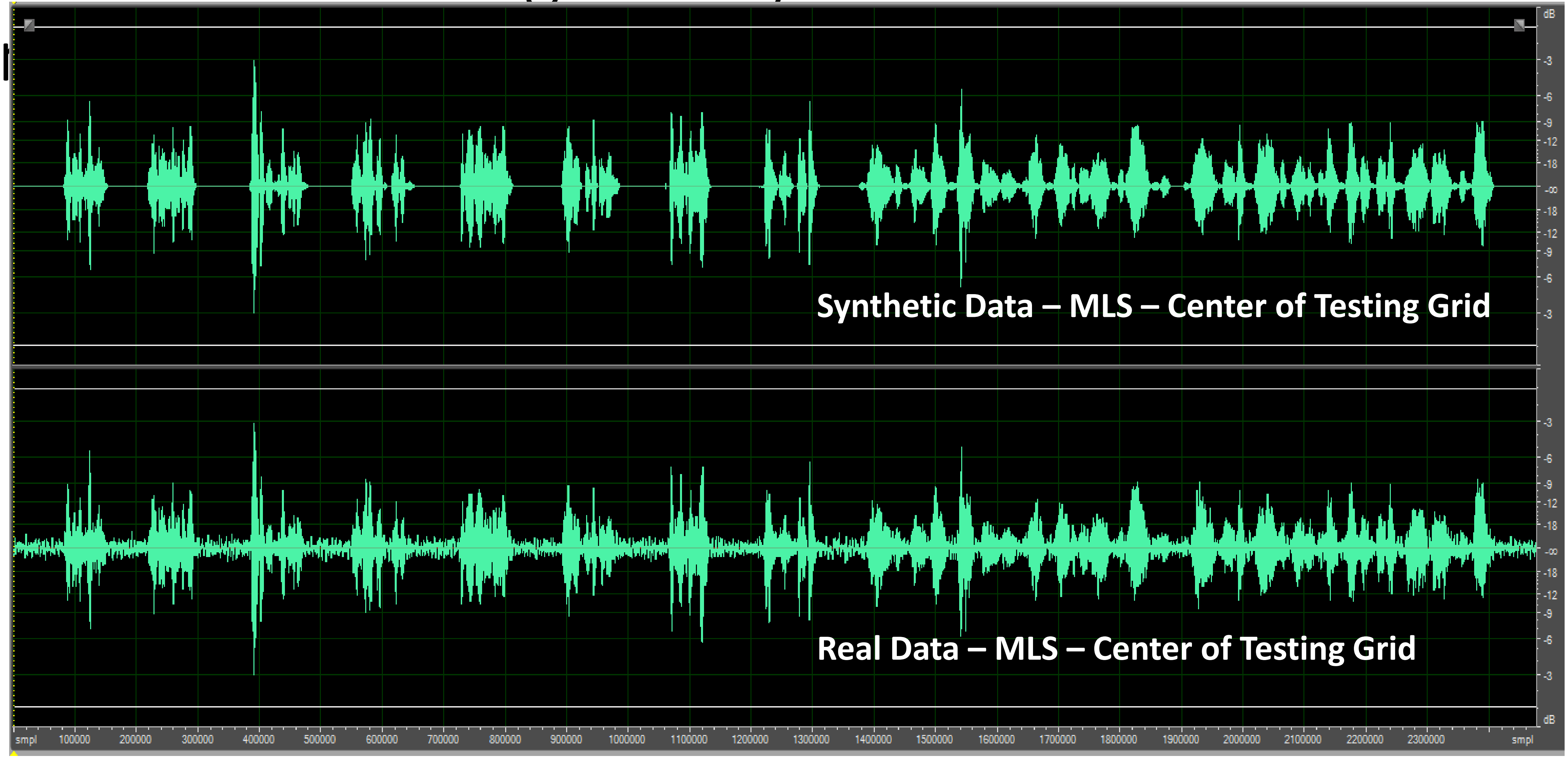


Reference Mics – Subwoofer

- Subwoofer only
- sine is better in NB/WB, noise is better in sub-woofer band
- Still good performance for all signals
- Possibly due to limited spatial averaging



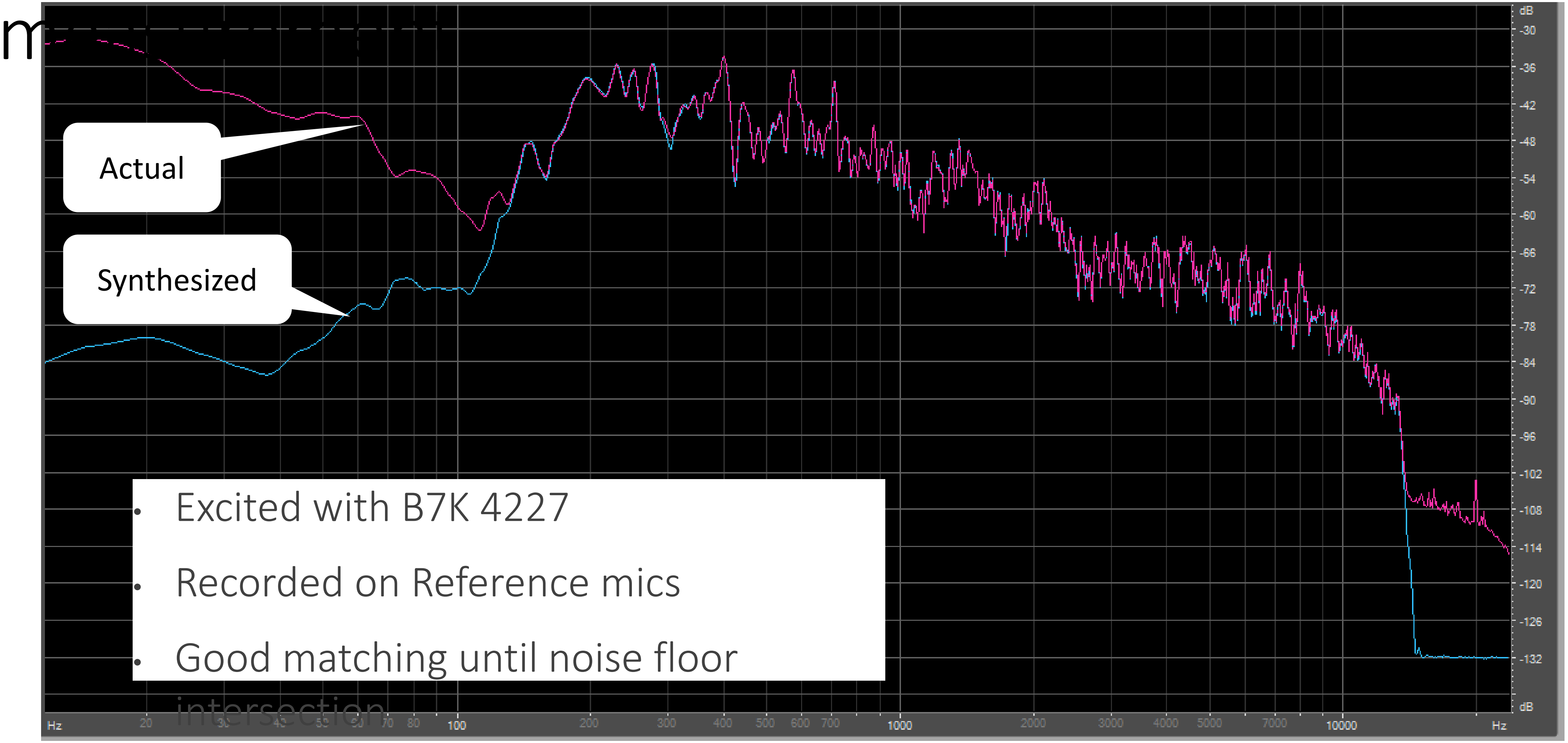
Results – Time signal – synthesized vs actual



Synthetic Data – MLS – Center of Testing Grid

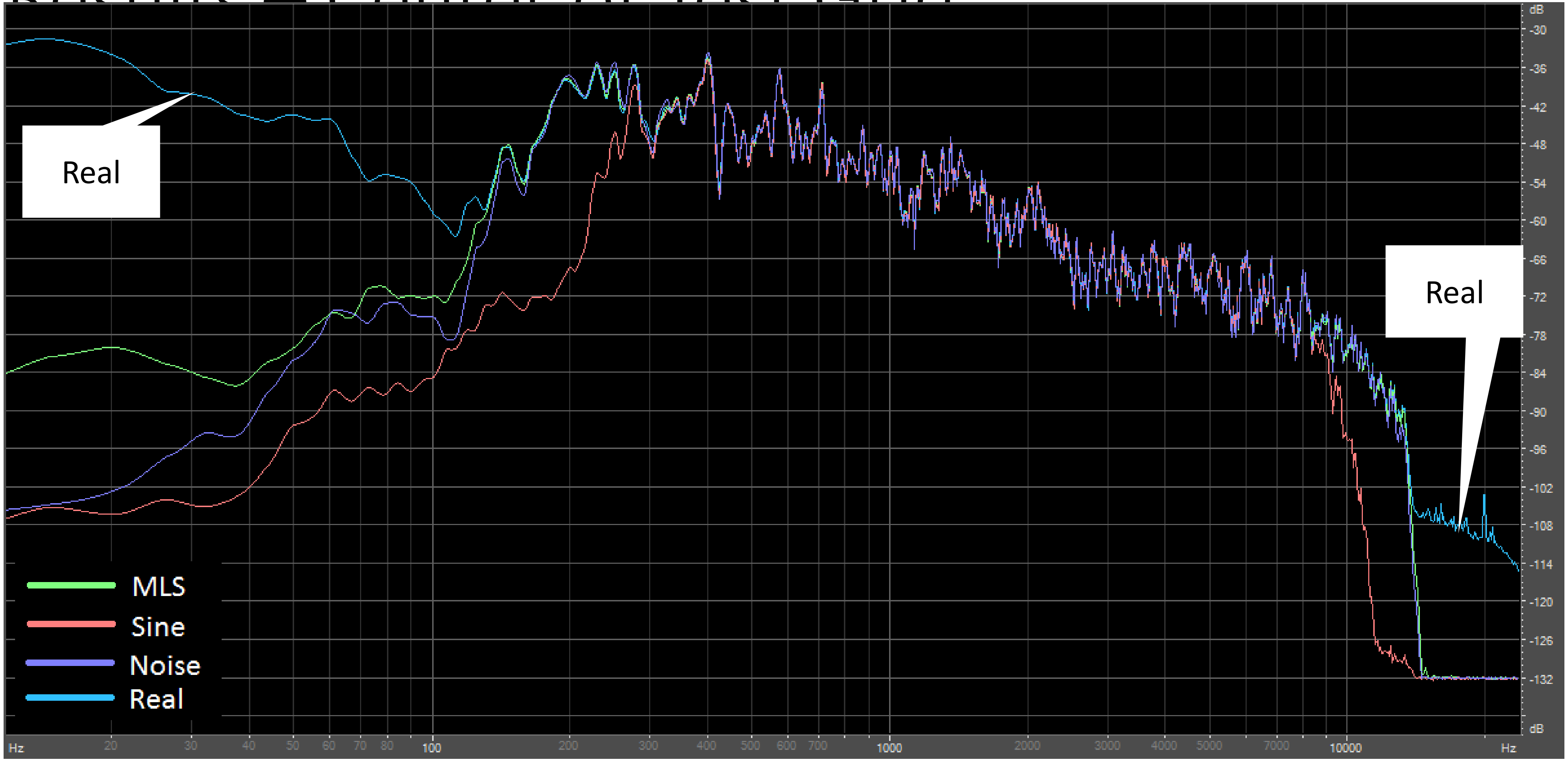
Real Data – MLS – Center of Testing Grid

Results – FFT synthesized vs actual measurement



- Excited with B7K 4227
- Recorded on Reference mics
- Good matching until noise floor intersection

Results – Center of Test Grid



Real

Real

Conclusions

- Of the three signals all provide reasonable performance
- The MLS test signal provides the best overall performance
- We now need to validate this approach on a full corpus run which is planned in the future.

Questions ?

THANK YOU!

DEVICES & STUDIOS

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