Room Impulse Response Estimation for Synthetic Data Acquisition

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DEVICES & STUDIOS

XBOX

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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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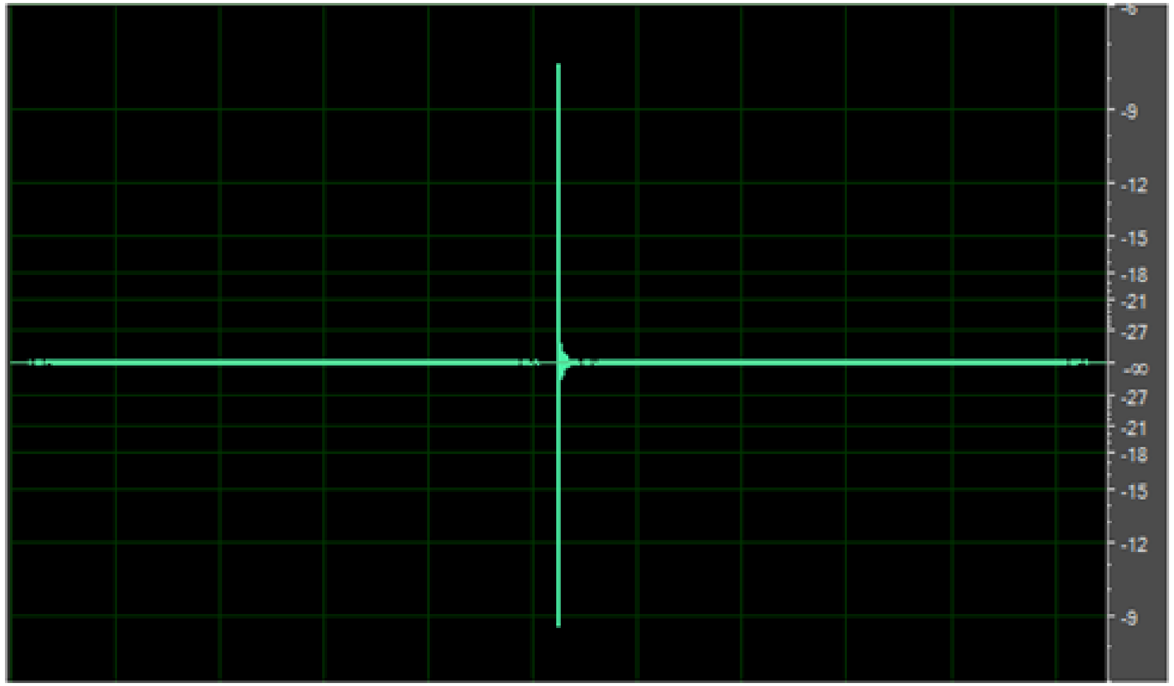
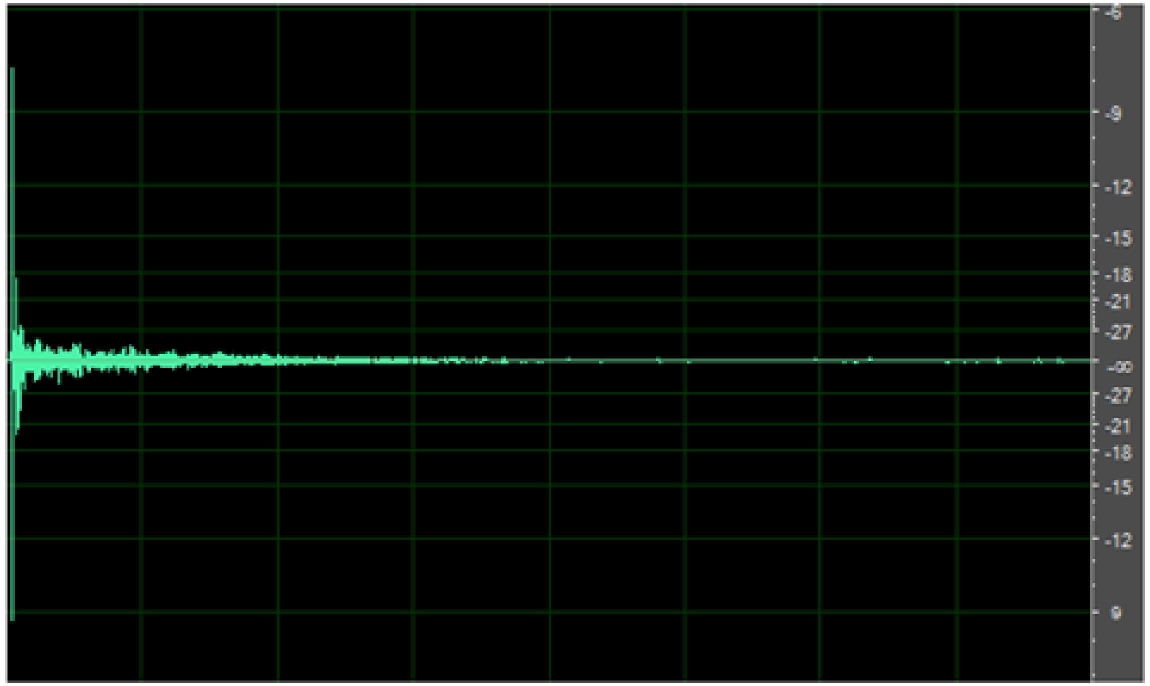
Test signal – MLS

- Maximum length sequences (MLS) of length exactly 2^{18} - 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





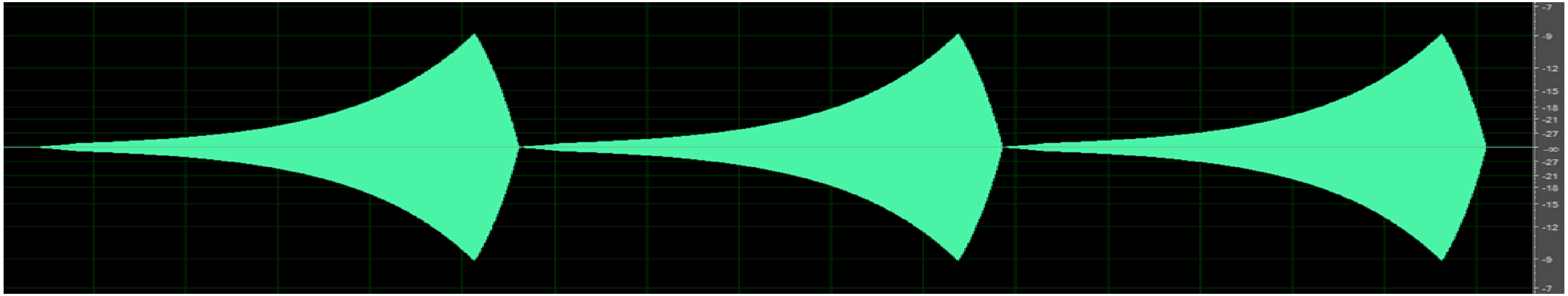
- complete deconvolved signal
- trimmed deconvolved signal

Test signal – Swept Sine

- sine signal of length of 2^{18} 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 correction

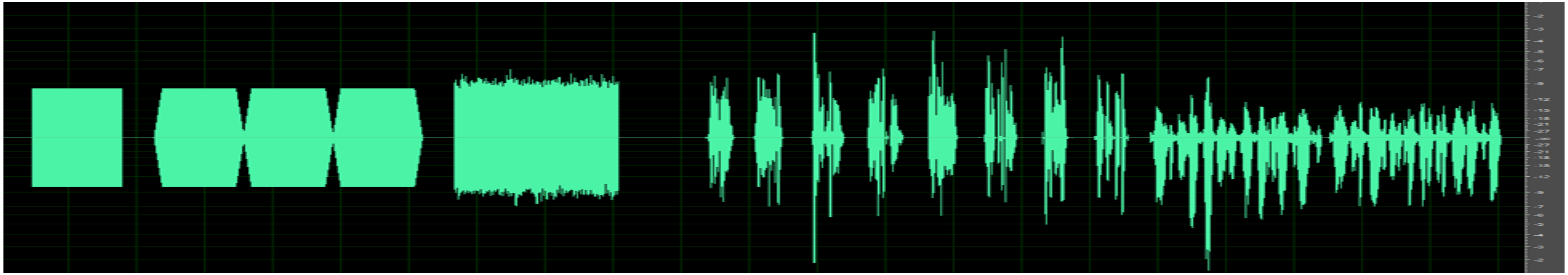
Swept sine signal for test



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

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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.
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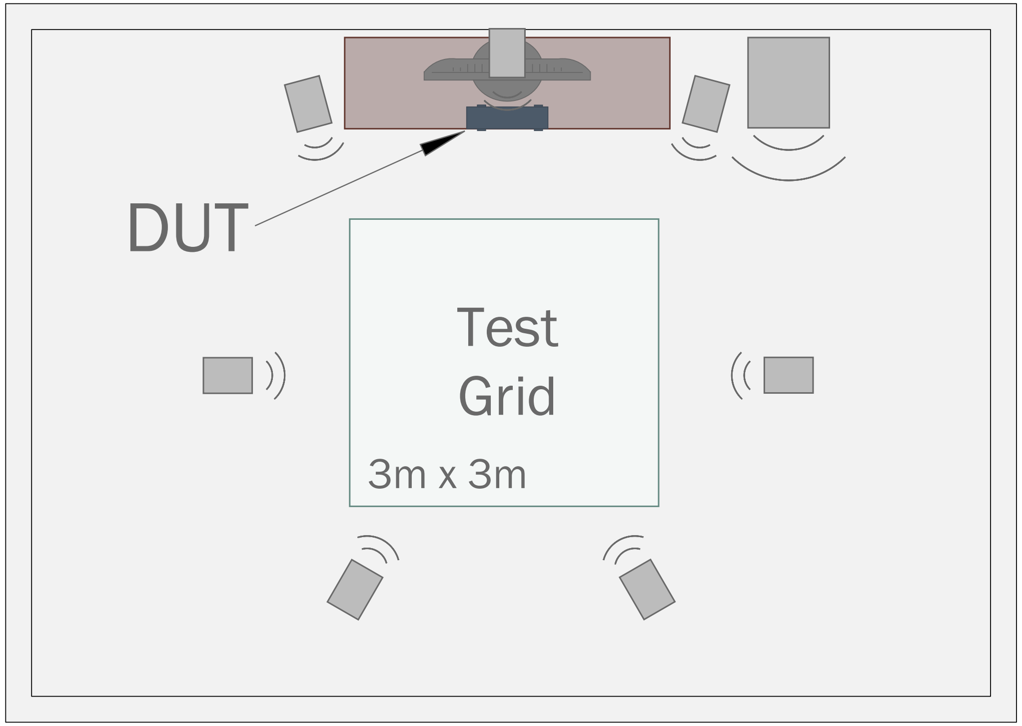

Data analysis

- 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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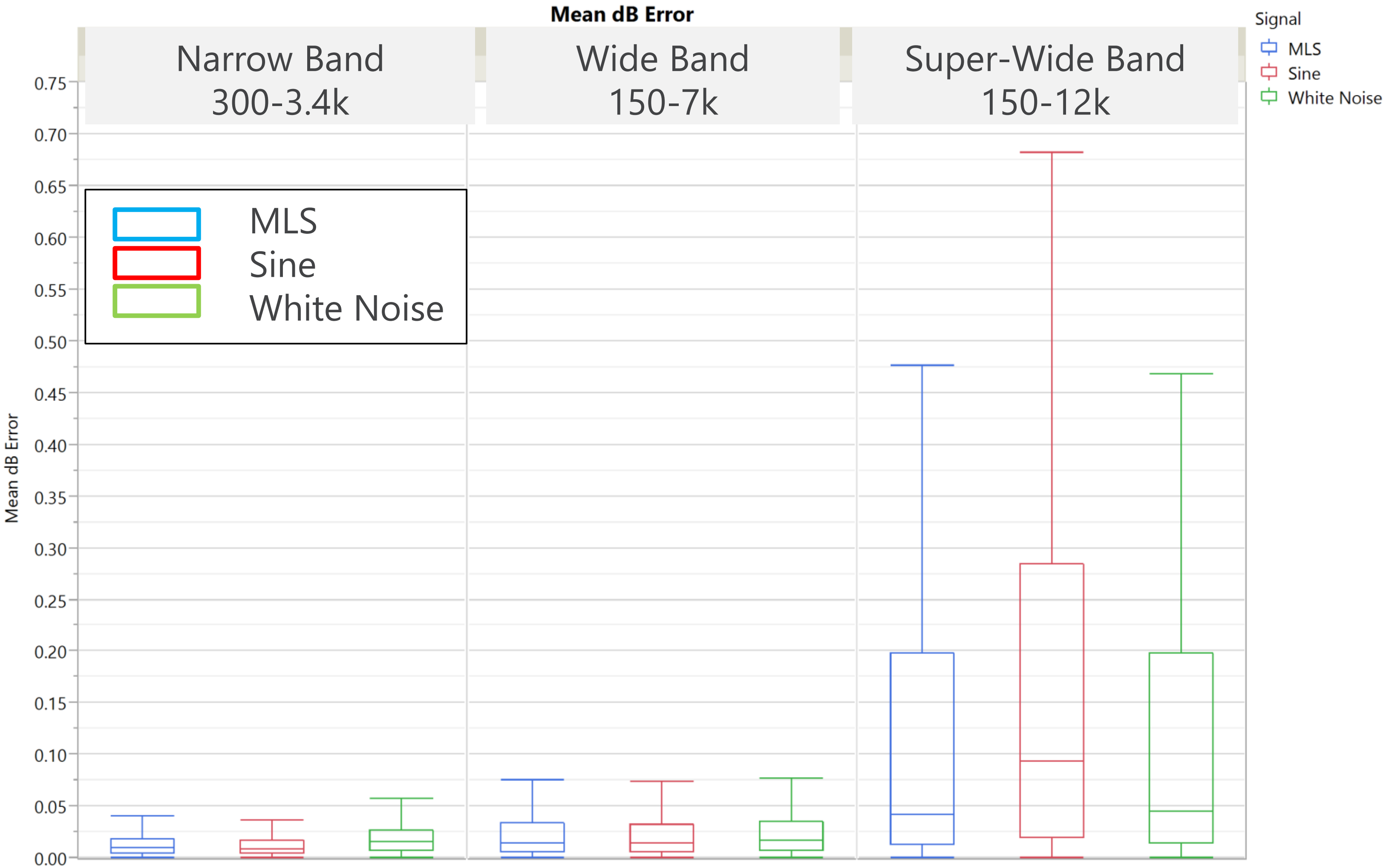
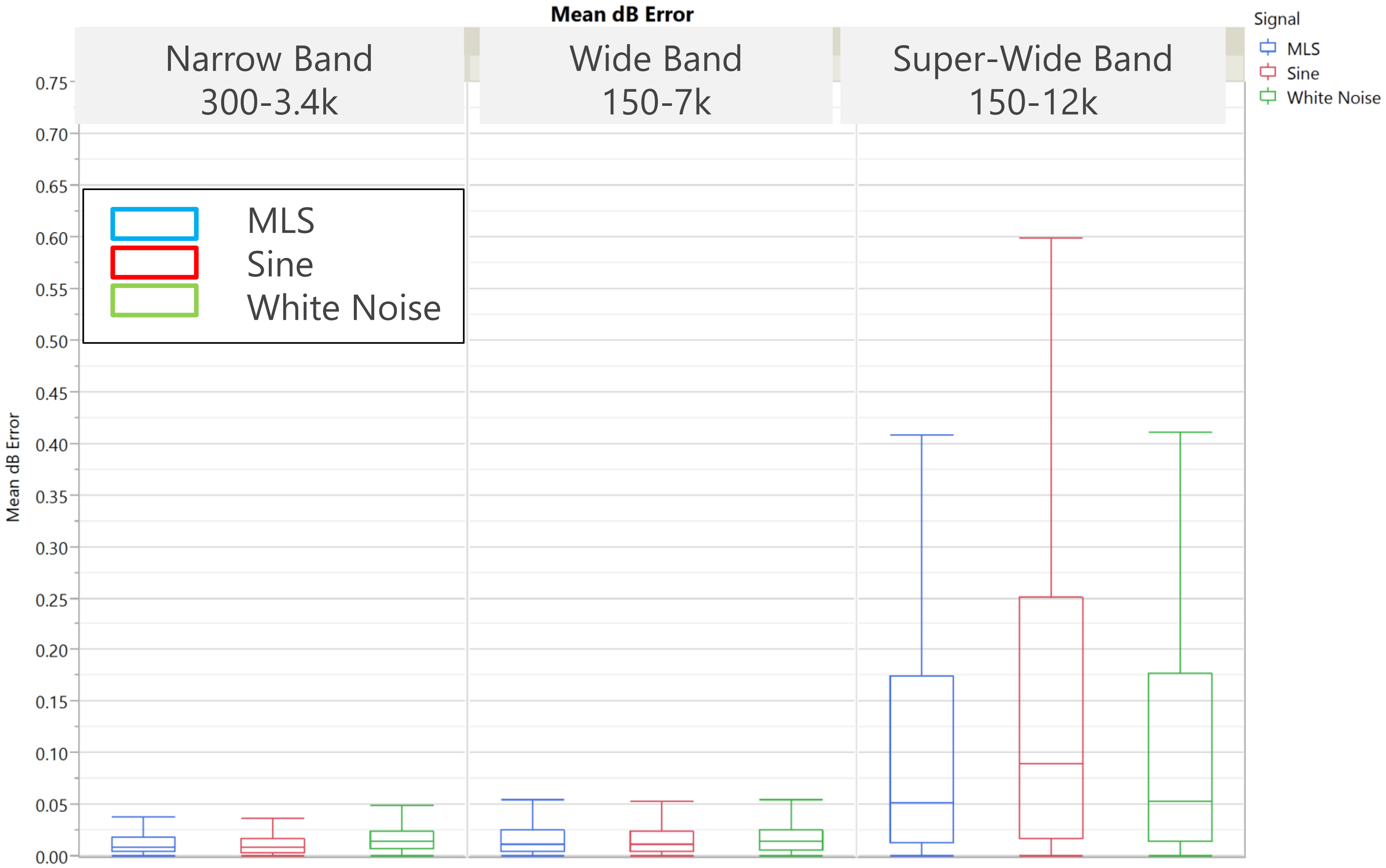
All Mics – All positions

- 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



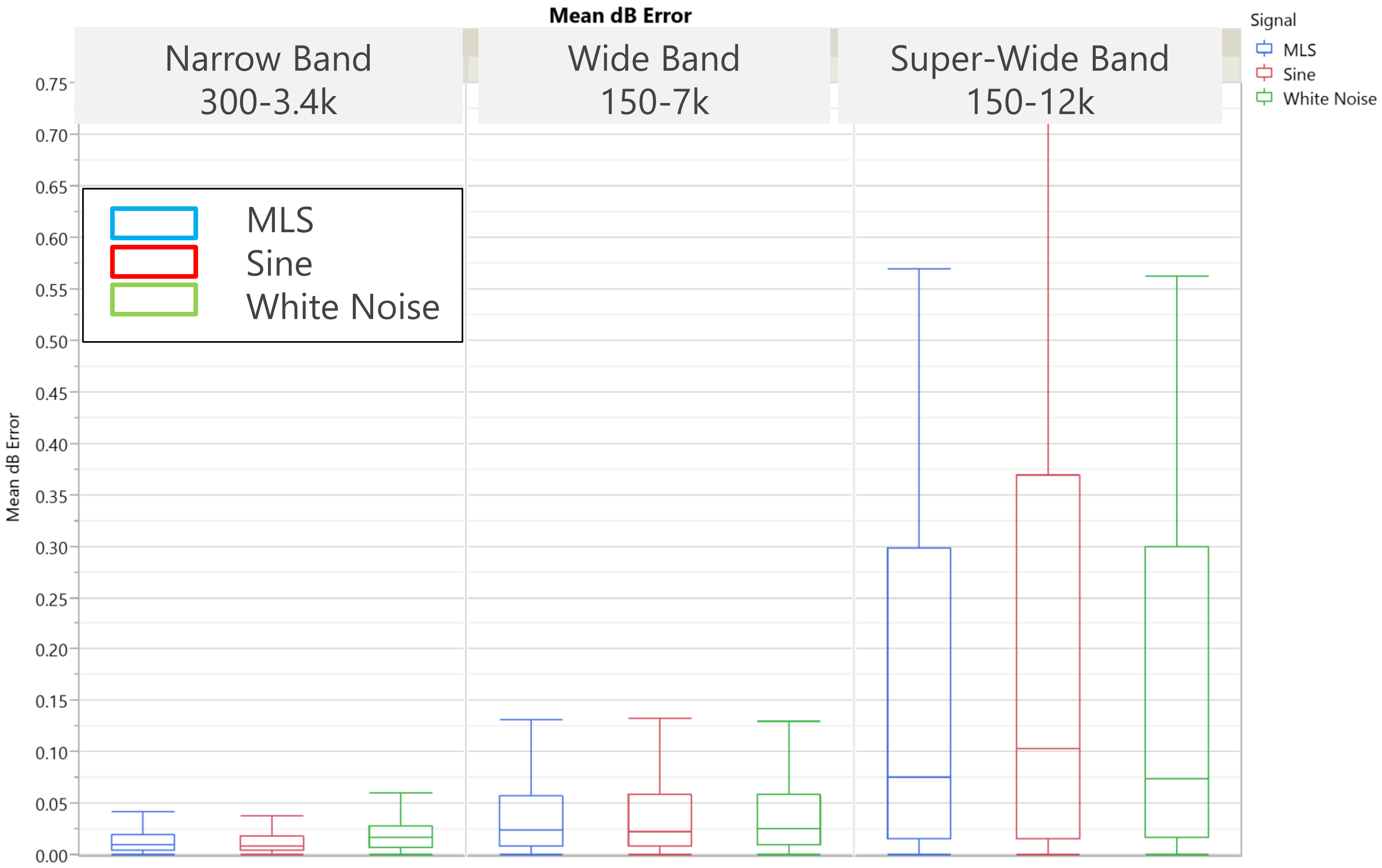
Production Array Microphones – All Positions

- 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



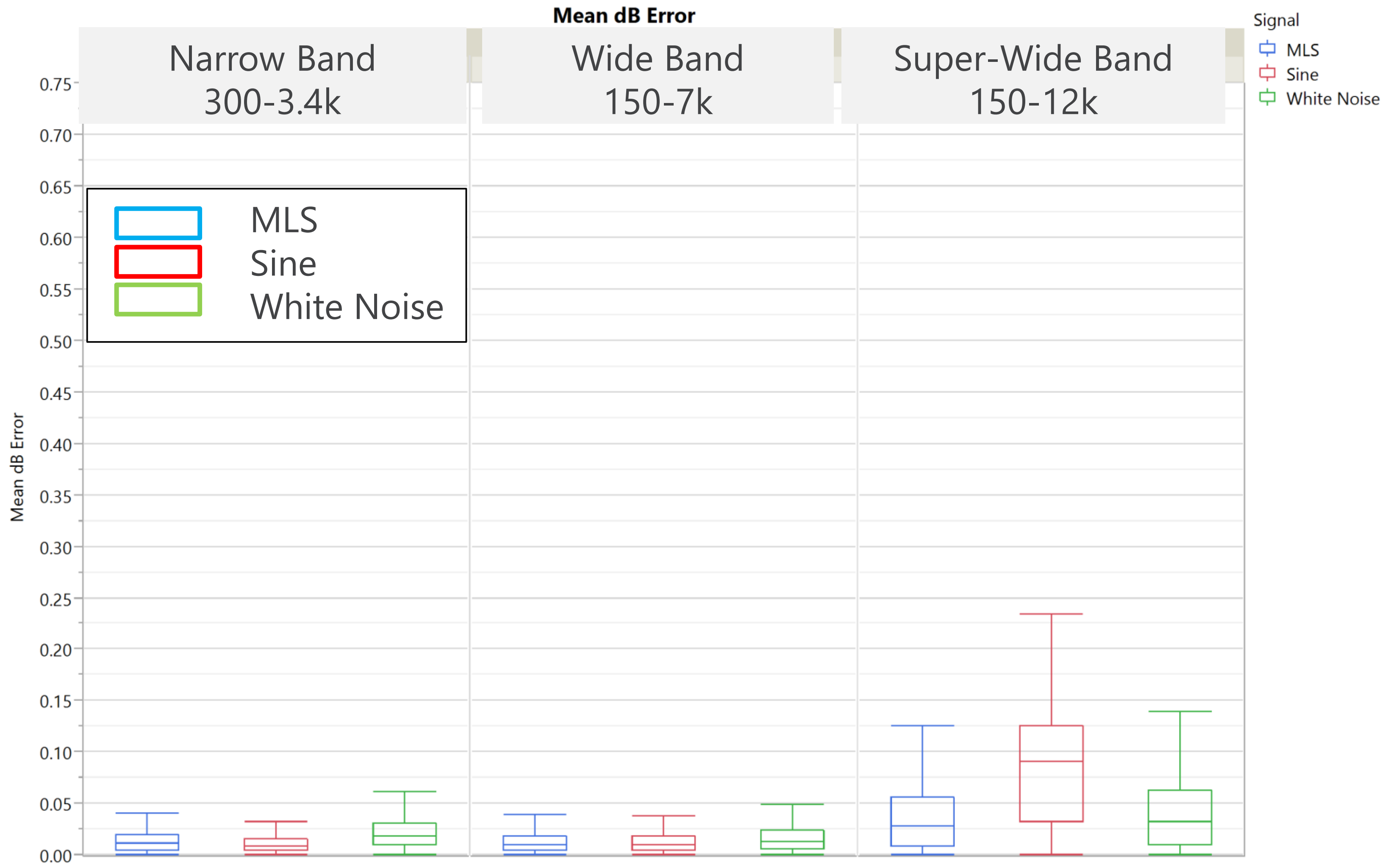
Prototype Microphones – All Positions

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



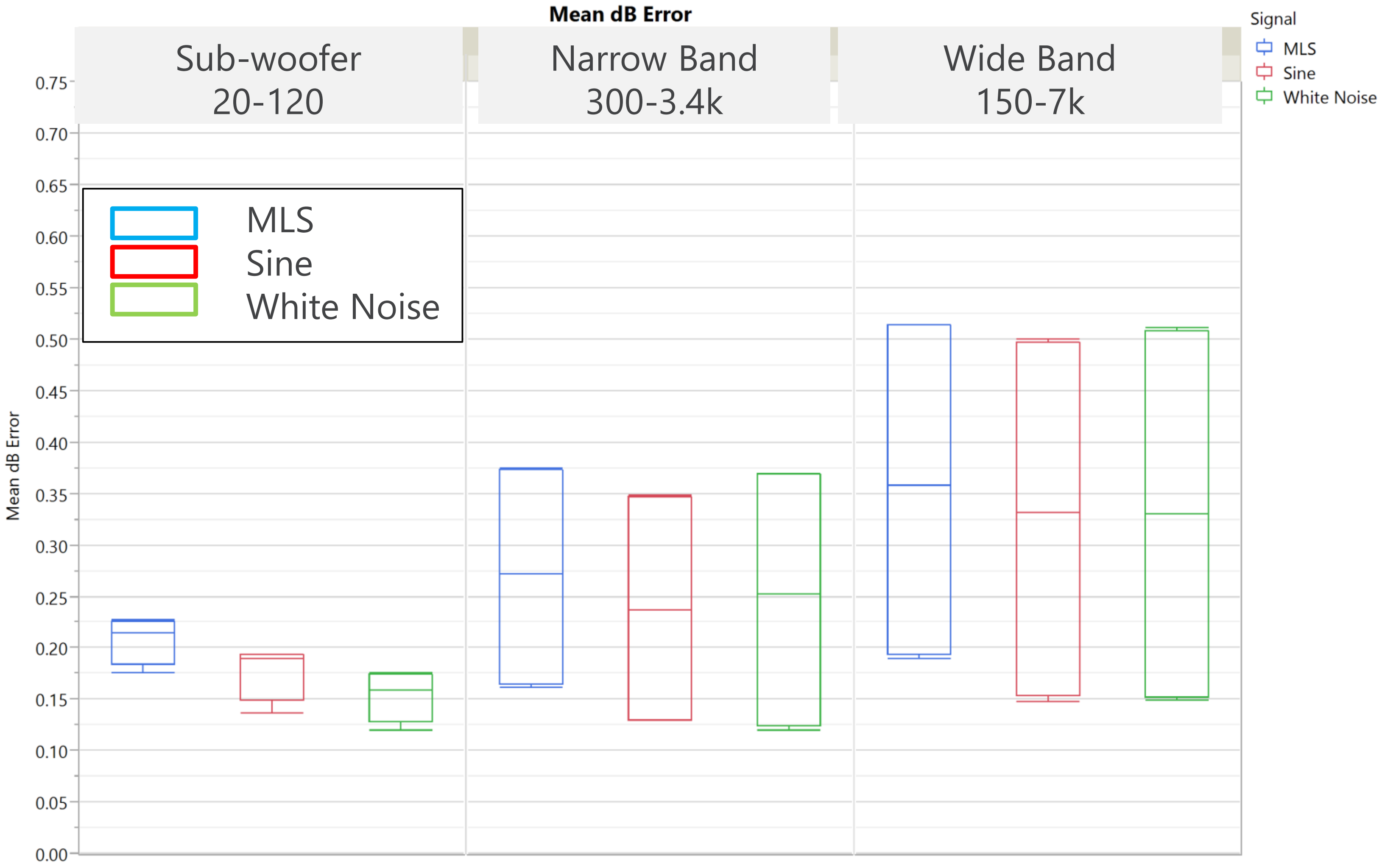
Reference Microphones – All Positions

- Excluding subwoofer
- MLS better in all categories

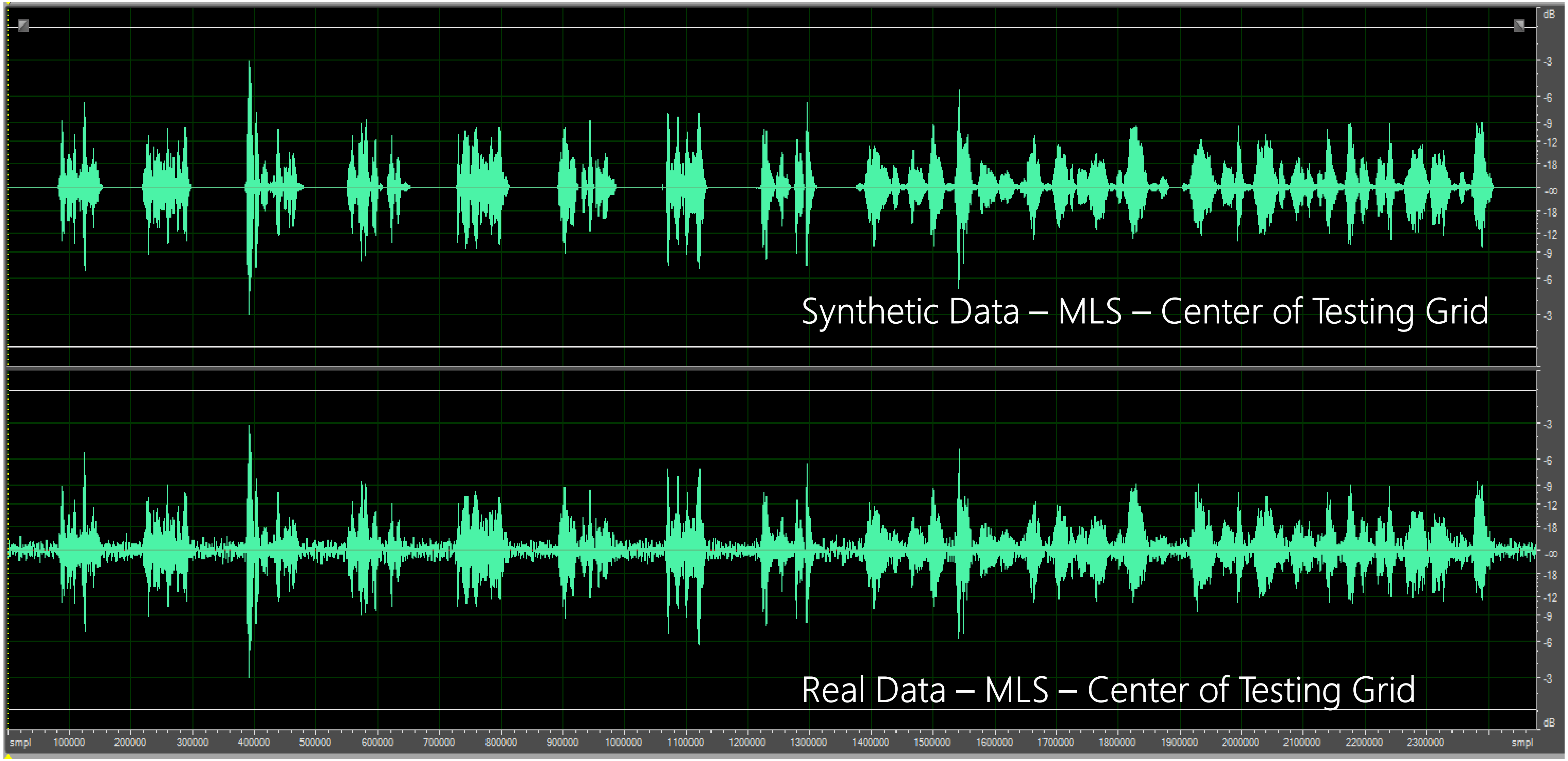


Reference Microphones – Subwoofer

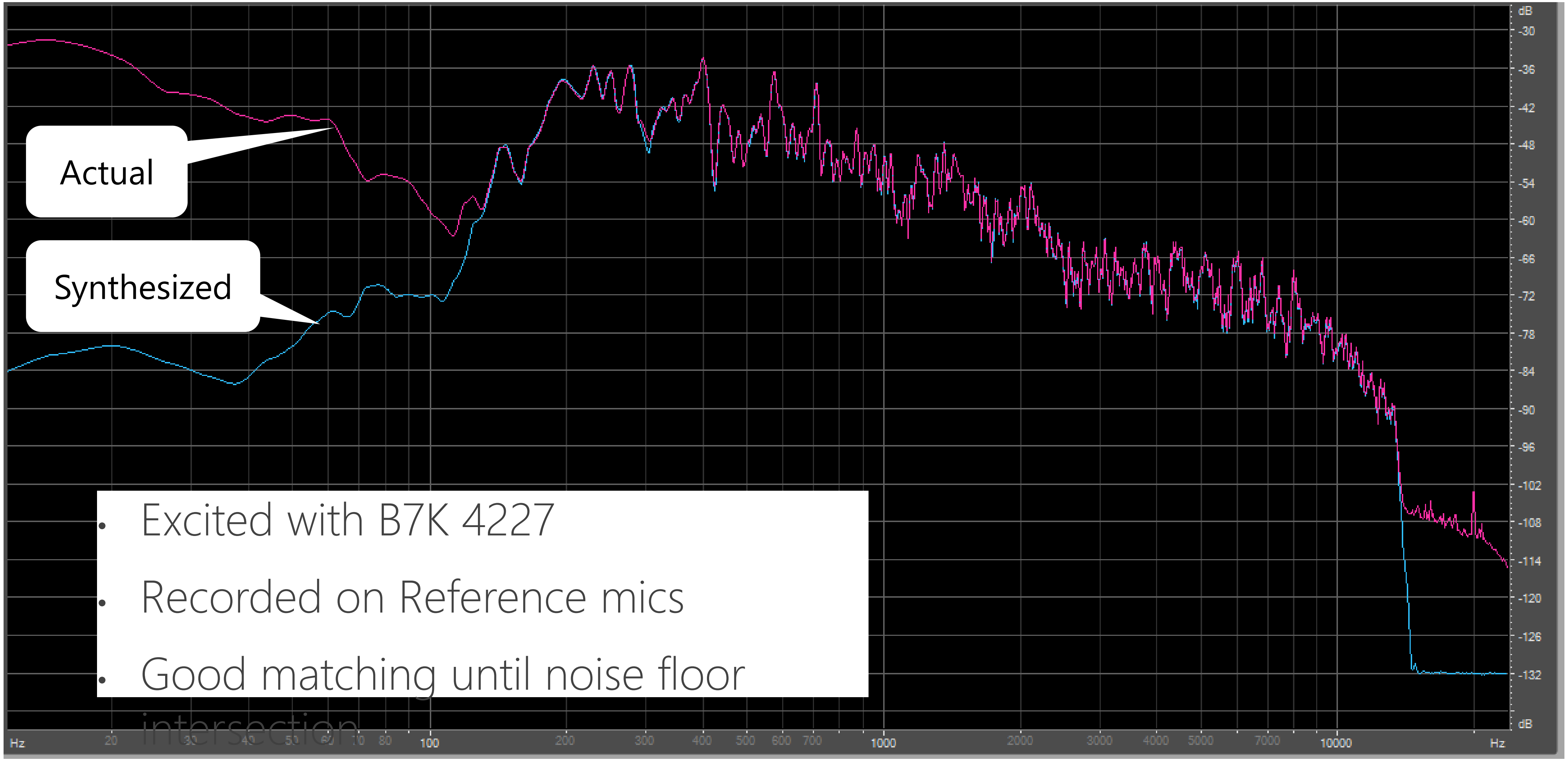
- Subwoofer only
- sine is better in NB/WB, noise is better in sub-woofer band
- Still good performance for all signals
- Possibly due to limited spatial averaging



Results – Time signal – synthesized vs actual measurement

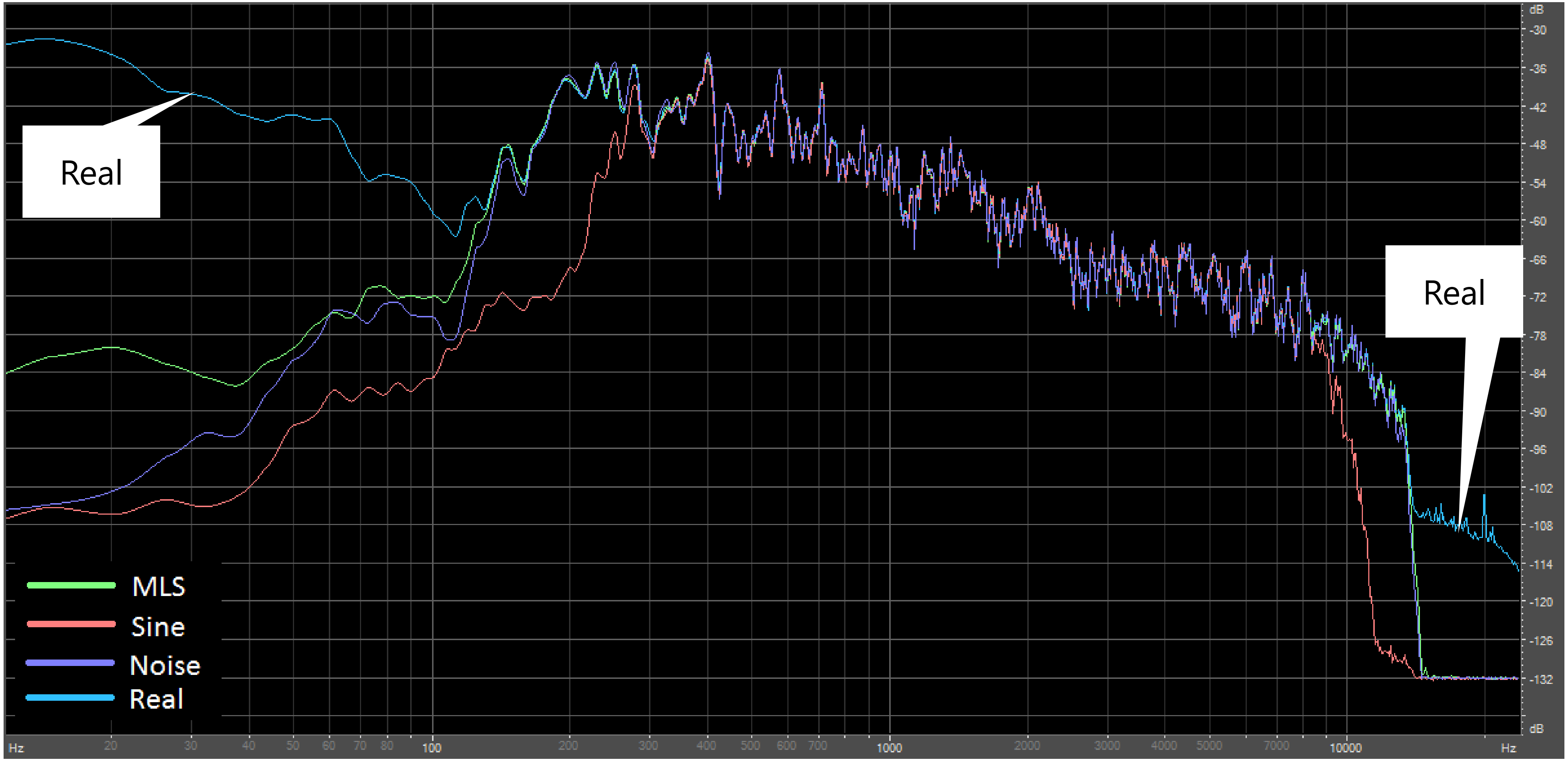


Results – FFT synthesized vs actual measurement



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

Results – Center of Test Grid



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!

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