Spring 2017

Room Acoustics: Measurement and Correction with a Portable Digital Signal Processor

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Abstract

The frequency response of a listening space is often of great interest to anyone pursuing an accurate and undistorted listening experience. Here we successfully implement sine-sweep deconvolution on a hardware platform to demonstrate an easy, digitized way to measure and guide improvements to the audio dynamics of a room.

Background

Every listening environment, from a tiny home theater to a cavernous amphitheater, has a unique frequency response that modifies sound as it traverses the space. The frequency response is a product of reflection and refraction of sound waves in the space (Figure 1). Because of its dependence on the topology of the room and relative positioning of the audio source and listener, the frequency response is often corrected by applying noise-dampening material to walls to prevent reflections, repositioning the audio source and listener to a more favorable configuration, and installing diffusers to prevent standing waves. Historically such analog techniques have been driven by subjective listening tests. Here we demonstrate a hybrid technique using a digital signal processor.

Procedure

- Began with consideration and acquisition of a suitable DSP platform. The C5515 evaluation platform from Texas Instruments was chosen early in the process for its low-power, fixed-point architecture, although this was the source of subsequent implementation difficulties.
- Before implementation, we used *Mathematica* software to try various test signals, including a simulated impulse, linear/logarithmic chirps, and audio samples. We settled on the logarithmic chirp (spectrogram shown in Figure 2), which is similar to the ascending part of a siren, because it emphasizes low frequencies.
- The test procedure is as follows. The test signal is played through speakers and recorded by a condenser microphone located at the listener's location (head in Figure 1). The frequency spectrum of the recorded signal $Y[n]$ and original signal $X[n]$ are computed via FFT, and divided magnitude-wise to yield the frequency response of the room $H[n]$. Phase was not considered since phase distortion is largely imperceptible [1]. This relies on the assumption of linearity in the room and the property of convolution in frequency:

$$Y[n] = H[n]X[n]$$

- The laborious process of implementation. The final architecture is shown in Figure 3.

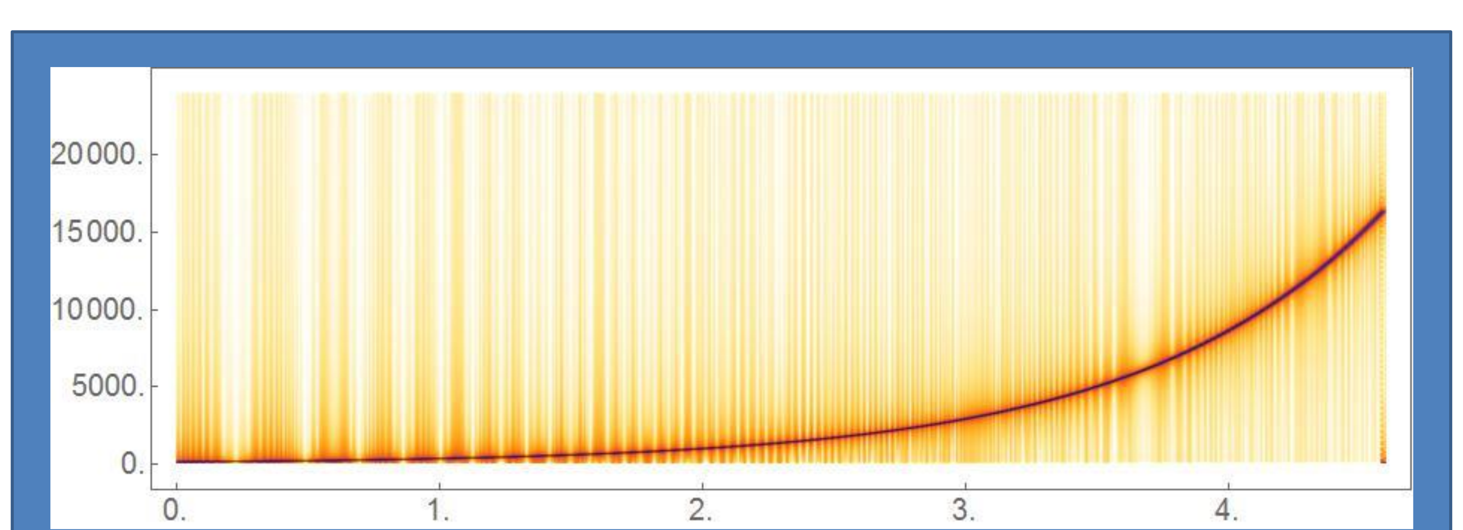

Figure 2. Spectrogram of test signal, showing that the frequency content increases exponentially over time.

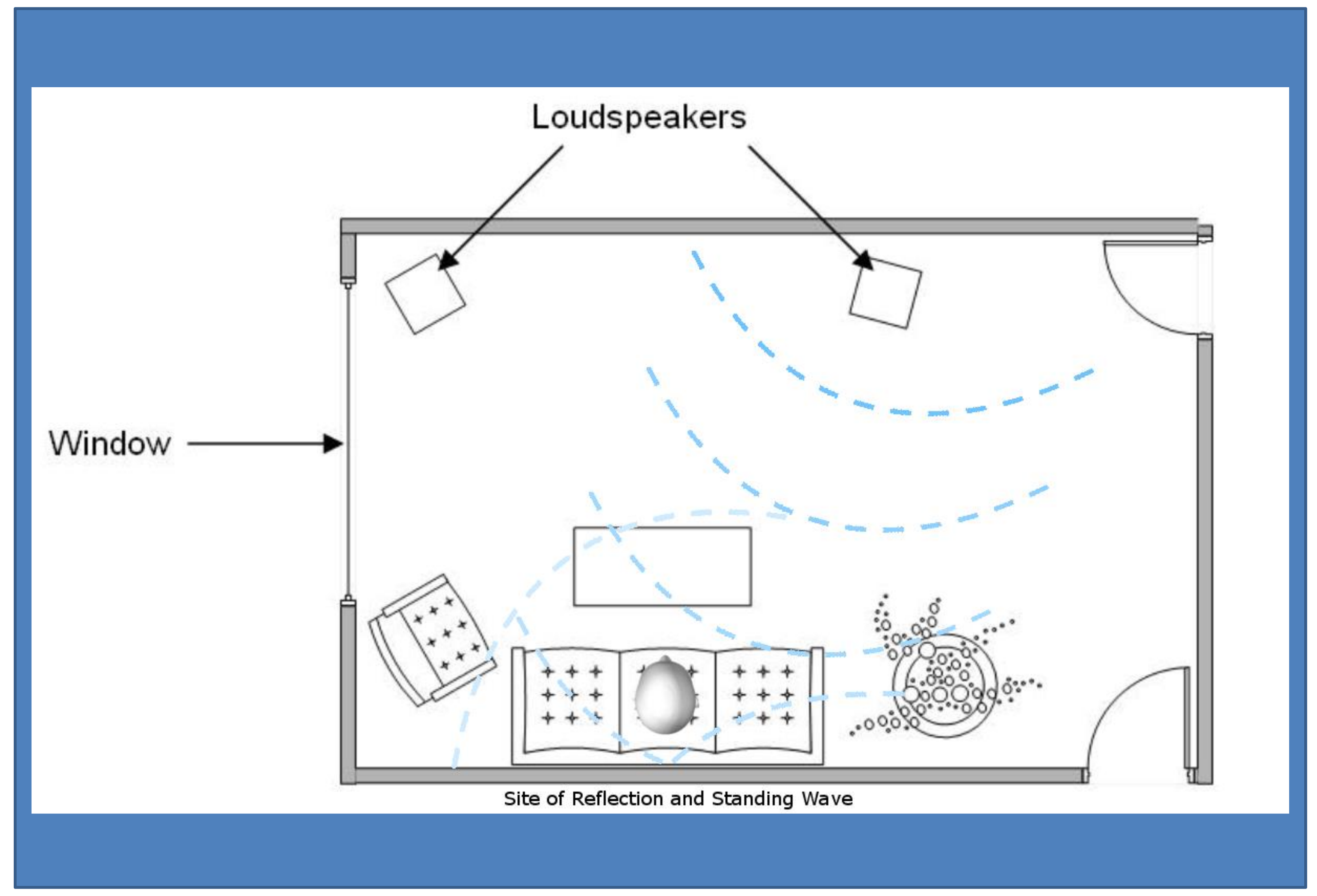

Figure 1. Prototypical listening setup, highlighting distortive effects of reflection and standing waves

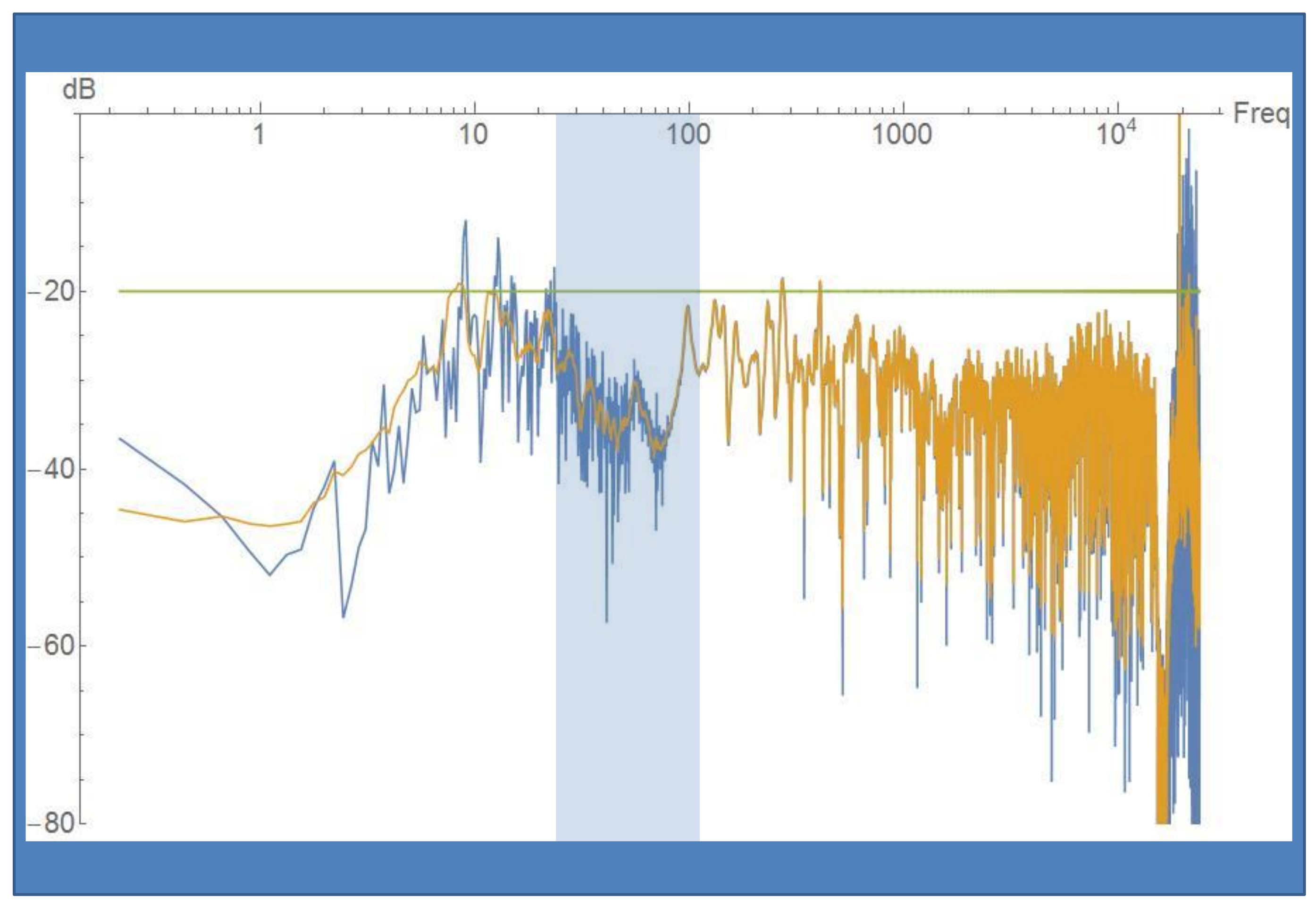

Figure 4. Measured frequency response for a listening setup, no treatment.

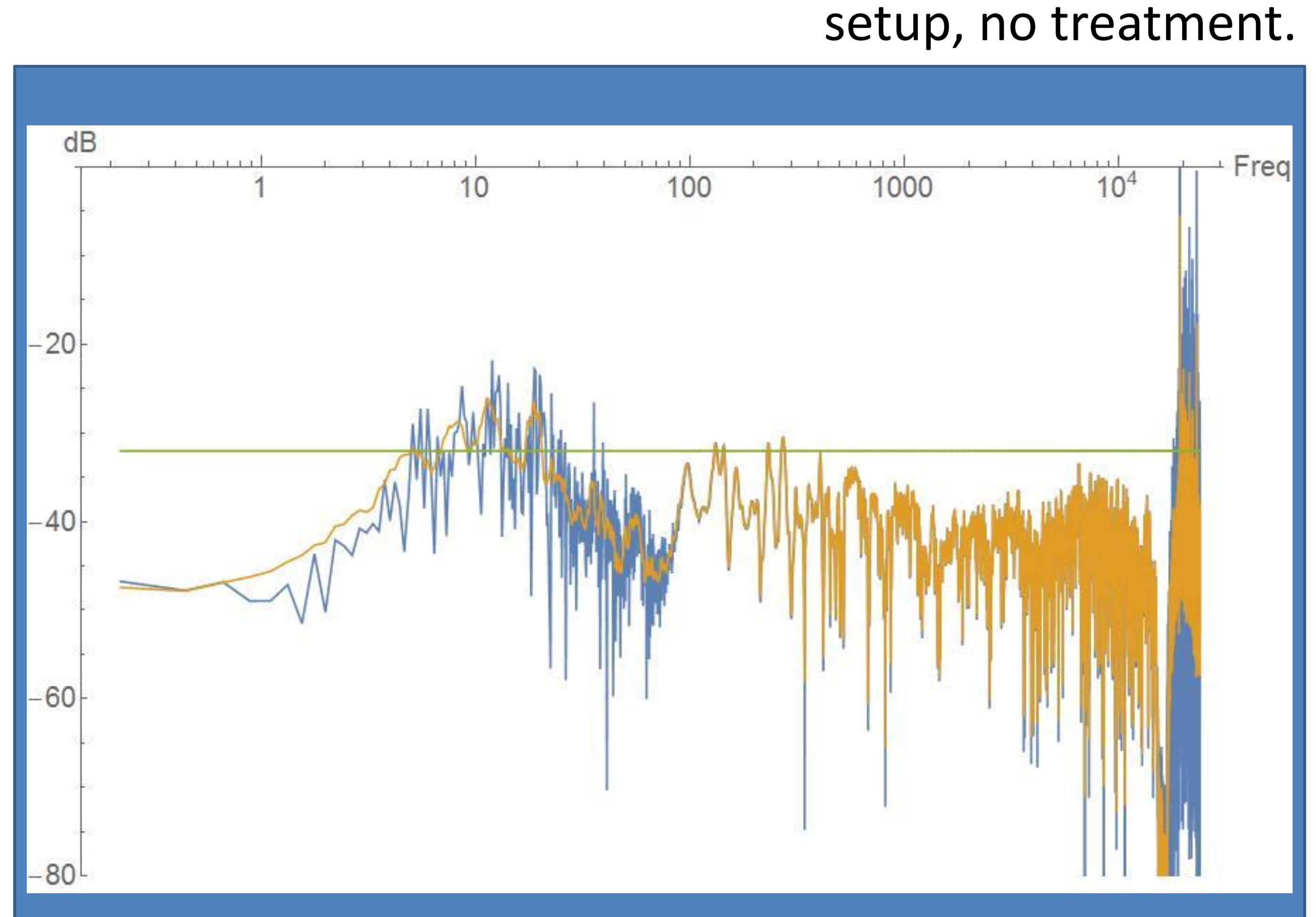

Figure 5. Response of same room with treatment. Note the boosted low frequencies.

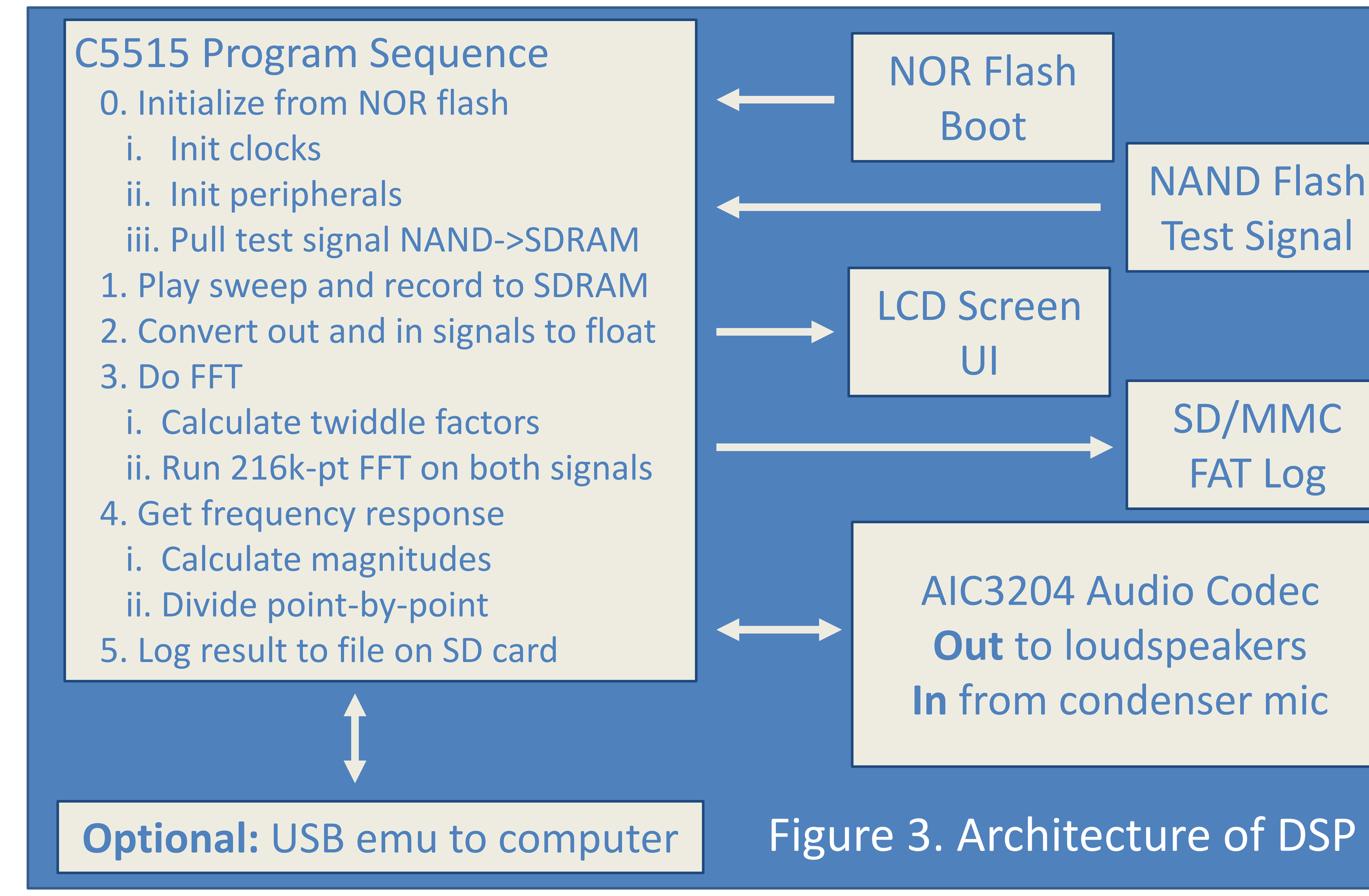

Figure 3. Architecture of DSP

Discussion

The implementation of the algorithm into the DSP was successful in that it matches calculations made by Mathematica when given the same data. However, one drawback is that the algorithm was ultimately implemented using floating-point calculations, whereas the DSP only has fixed-point hardware support. The limitation was due to fixed-point FFT algorithms discarding a bit per stage to prevent overflow, which proved unacceptable for a 216k FFT. Using a floating-point FFT slows execution time significantly; future work might include a port to the C6XXX DSP platform which contains floating-point hardware, albeit at a higher cost and power consumption.

As a formal verification of the hardware-implemented measurement algorithm, we sought to calculate and compare the frequency response for a typical listening setup with and without damping. Figure 4 shows the frequency response of the untreated room in blue. In Yellow is the same signal but lightly smoothed by an 8-tap FIR filter. The green line is arbitrarily placed as a visual aid to help compare heights across the response. Lastly, note the dB/log axes.

Since human auditory perception is strongest at low frequencies and bottoms out at 20 Hz, one is usually interested in correcting the range from 20 Hz to 1000 Hz [2]. In the response, we see a considerable dip around 80 Hz (highlighted in Figure 4), corresponding to a wavelength of 14 feet, which was hypothesized to be due to standing wave phenomena or destructive interference with reflected waves at that location.

To investigate this frequency dip, we applied rudimentary damping behind the microphone in the form of a thick blanket that would absorb incoming sound waves. The same test, when run again, yielded the response in Figure 5. It is clear that the low frequency was boosted with respect to the high frequencies, but it is unclear if this is because of standing wave reduction in the low frequencies, which create troughs in the response, or by reducing reflection of higher frequencies, which distort soundstage and introduce unwanted reverberation [1]. Regardless, it is rewarding to see a physical change to the space reflected in its digital characterization.

Having demonstrated that the measurement is robust enough to react to the introduction of relatively minor damping to the space, future work will focus on calibration to the frequency response of the microphone itself, characterizing more acoustically interesting spaces, and improvements to the code base.

Acknowledgements
We thank the software team of Enercon Technologies in Gray, ME for their aid in hardware acquisition; the authors of the KissFFT library and C5515 CSL, which great reduced development times; and the TI E2E community for clarifying what TI support would not.

References
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[2] Pate, Tyler. "Audio Characterization Primer." SLAA641 (2014): n. pag. Application Report. Texas Instruments, Aug. 2014. Web.