A Fourier transform was performed on each individually recorded ultrasonic pulse in order to determine which frequencies of sound are more strongly reflected back towards the sensor.

The frequency response for each pulse is then compared against a model response, and the difference between the two is defined as the frequency dependant reflection coefficient $R_n(f)$, where n denotes the ultrasonic pulse is being analysed.

These plots for reflection coefficient against frequency for each ultrasonic pulse are then stacked side-to-side. This creates a spectrogram for reflection coefficient $R_n(f)$ across both frequency f and pulse number n throughout the duration of a test.

The frequency of this wave is scaled from the ultrasonic range to audible frequencies using a conversion factor of $f_{sound} = \frac{f_{data}}{10{,}000}$.

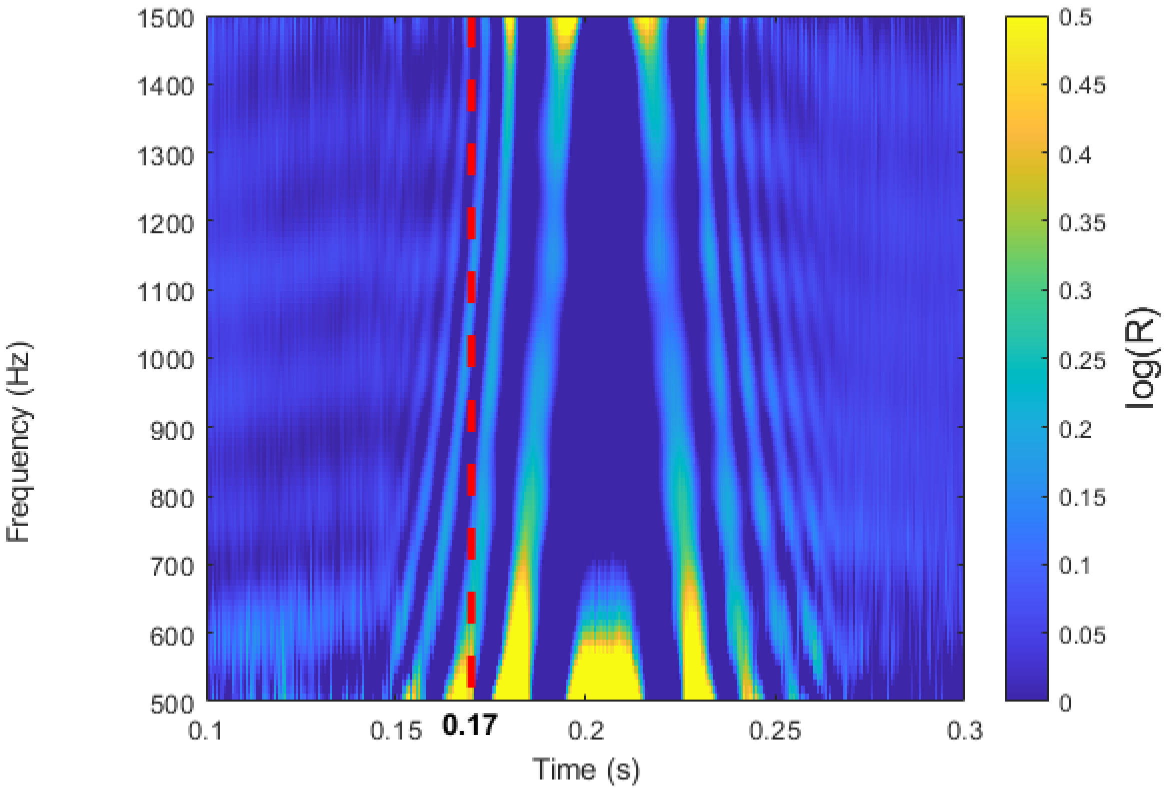

Figure 1 – Spectogram of oil film thickness for a single pass of a roller bearing, with frequency converted to within audible range.

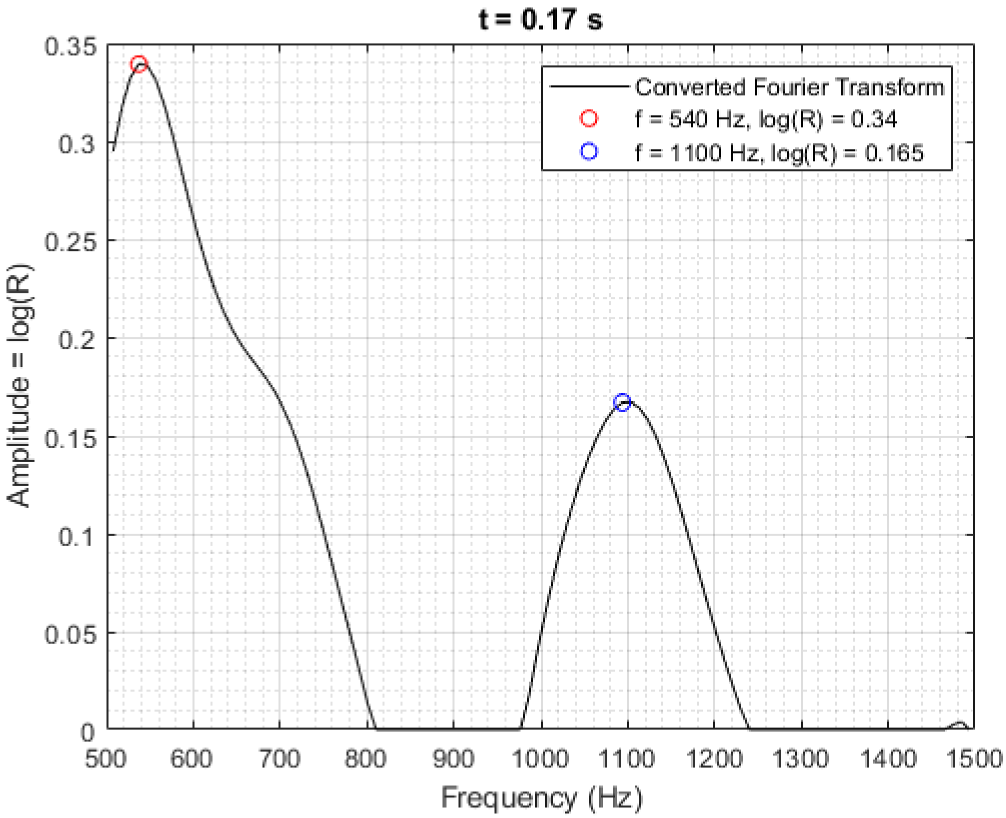

Figure 2 – Time segment of Figure 1 taken at 0.17 s, with two example frequency components highlighted.

The natural logarithm of $R_n(f)$ was also taken to represent the amplitude of each wave in order to make the subtle changes in frequency more prominent and audible.

Each frequency is also given a random set phase angle $\phi(f)$, which allows for a continuous wave between pulses for each frequency without having complete silence at the interface between pulses.

These sine waves are then summed for each pulse to give a final waveform S_n.

$$S_n\left(t_n\right) = \sum_i \left(log\left(R_n(f_i)\right) \bullet sin\left(2\pi f_i t_n + \phi(f_i)\right)\right)$$

The duration and sample rate of the tone was determined by the time array t. A sample rate of 44.1 kHz was used for this array as an industry standard for audio sampling.

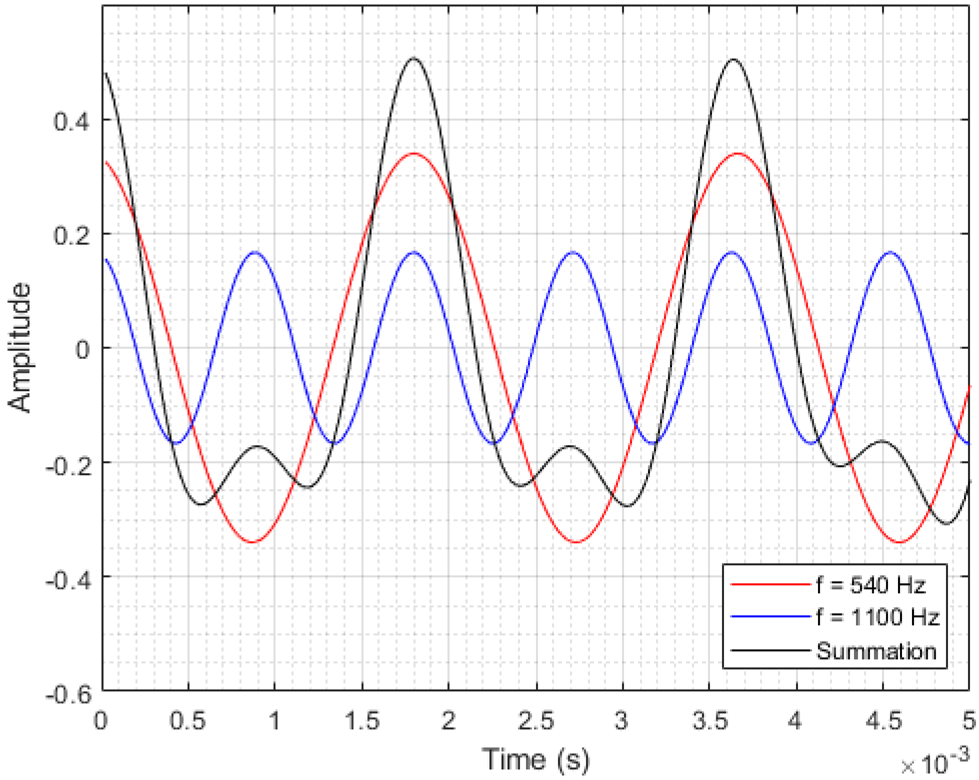

Figure 3 – Frequency components from Figure 2 reconstructed into a tone.

Owing to the high pulse rate of 9 kHz for the ultrasonic recordings, if the time vector t directly represented real-time data acquisition, then only 4.9 samples per pulse would be available. This severely limits the quality of sound producible through reconstructing the soundwave directly to real-time. A new pulse rate of 10 Hz was instead used to give 4410 samples per pulse. This allows for the soundwave to carry information for audible frequencies.

The soundwaves for each pulse are then concatenated end-to-end in order to produce a sound file that represents the duration of an entire test.

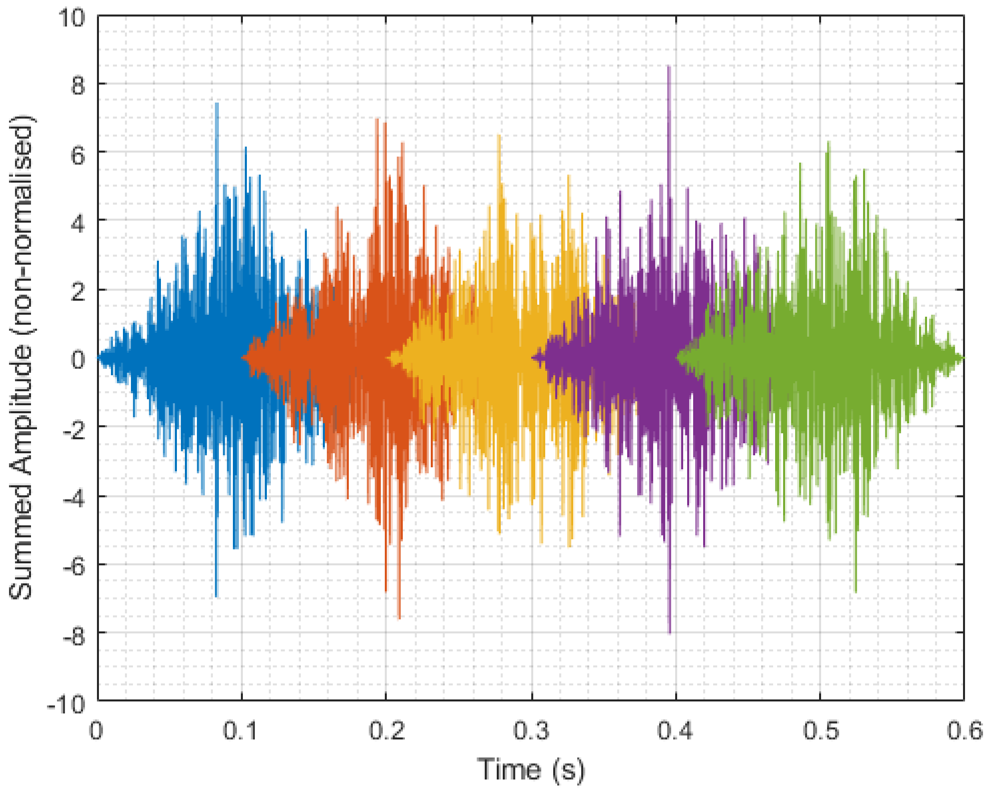

Figure 4 – Concatenated and merged tones for the first 5 individual ultrasonic pulses of a given test.