Quantifying Beer Fermentation via Speed of Sound Measurements

L. Chai, P. Sundar, M. Thomas, C. Wheeler and W. Yuan

Abstract—The purpose of this research is to quantify the fermentation of beer by measuring the change in the speed of sound as it is brewing. This work is important to scientists and engineers as it illustrates the advantages and challenges in using low-cost acoustic measurements (e.g. speed of sound) for system identification. This work is particularly useful for beer brewers as a method for quantifying and tracking beer fermentation. Sending white noise through a speaker and measuring the delay between two microphones determined the speed of sound through water and 5% ethanol by volume. The time delay was computed via a weighted linear fit of the phase of the transfer function relating the two microphone signals. The magnitude-squared coherence of the two signals was used as the weights. The interpolation provided by this technique enables measurement of time delays smaller than what could be measured according to the Nyquist frequency. The percent error comparing a nominal speed of sound change and our measured speed of sound change is 43.9%. Experimental design improvements and a portable design are proposed.

Index Terms—Acoustic, beer, fermentation, measurement, speed of sound, stochastic perturbation

I. INTRODUCTION

The purpose of this research is to quantify the fermentation of beer by measuring the change in the speed of sound as it is brewing. This work is important to scientists and engineers as it illustrates the advantages and challenges in using low-cost acoustic measurements (e.g. speed of sound) for system identification. This work makes an impact on quantifying beer fermentation as it brews.

Yeast metabolizes sugars in beer to produce alcohol during fermentation. The current method of quantifying beer fermentation is to use a hydrometer to measure the density of the beer relative to water [1,2]. The ratio of the density (ρ) of a fluid to the density of water is called the specific gravity (SG), and is given by (1):

\[ SG = \frac{\rho_{\text{sample}}}{\rho_{\text{water}}}. \]  

The change in density (Δρ) of a fluid is thus given by (2):

\[ \Delta \rho = \rho_{\text{water}} \times (SG_{\text{final}} - SG_{\text{orig}}). \]  

This method requires taking a sample of beer prior to and during fermentation. Two drawbacks of this method are: (i) risk of contamination of the beer when extracting a sample to test, (ii) monitoring that is reliant on guessing the end of fermentation. Using the speed of sound measurement as a way of monitoring the fermentation process eliminates both of these risks. This technique therefore increases the probability of a successful brew and the repeatability of brewing results between batches.

These experiments used a 95% water, 5% ethanol solution to mimic the ethanol concentration in beer. Investigating the effect of colloids and carbonation on the speed of sound measurements is the subject of future work because the colloids sink to the bottom and most of the carbonation occurs after fermentation.

The beer fermentation may also be measured by the speed of sound [3]. Sending an acoustic signal through the fluid and measuring that signal at two points along its path allows measuring the speed of sound in a fluid, shown in Fig. 1, by dividing the delay length by the time delay between the two signals.

II. ANALYSIS

This section forms the analytical framework for measuring the speed of sound measuring a time delay with a known separation distance between two microphones. The time delay was estimated using two methods: cross-correlation and magnitude-squared coherence. Method 1 finds the delay as the maximum of the cross-correlation of the microphone signals [4]. Method 2 derives the delay from the slope of the phase of the computed transfer function between the two microphone signals [5].

A. Delay estimation via cross-correlation

The method of calculating the delay using cross-correlation consists of the following steps:

1) Zero the average of the signal and scale the signal so that the standard deviation is 1. Time-shift the signal
times equally such that the first signal initializes the sequence.
2) Compute the auto-correlation of each signal.
3) Compute the cross-correlation of the results of step 2.
4) Plot the cross-correlation versus the sample vector.
The time of the maximum cross-correlation value is the delay.

The resolution of this method is limited by the sampling rate. The resolution of the measured delay is 20 μs for a sampling rate of 100 kHz.

B. Delay estimation via computed phase slope

Another method of calculating the delay is to compute the transfer function between the two measured signals and measure the slope of the phase of the computed transfer function [5]. Microphone 1 will receive a signal as a function of time, i.e. \( x(t) = f(t) \). Microphone 2 will receive the same signal, but delayed by an amount \( T \), i.e. \( y(t) = f(t-T) \). The Laplace transforms of \( x(t) \) and \( y(t) \) are \( X(s) \) and \( Y(s) \), respectively, where \( s = j\omega \) and \( \omega \) is the angular frequency. Equation (3) gives the transfer function, \( H(s) \), of the two signals:

\[
H(s) = \frac{Y(s)}{X(s)} = \exp(-j\omega T). \tag{3}
\]

The Laplace transform of a time delay has unity magnitude for all frequencies and the phase, \( \phi \), of the delay is linearly related to the frequency, shown by (4):

\[
|H(s)| = 1 \text{ and } \phi(H(s)) = -\omega T. \tag{4}
\]

The time delay, \( T \), may be found via a linear fit of phase versus frequency. The speed of sound, \( c \), is then given by dividing the separation distance by the time delay, \( T \).

The magnitude-squared coherence function was used as a weighting function for the line of best fit. This effectively filters the data to lie within the bandwidth of the actuators and sensors. The power spectral density was computed for both microphone signals and their cross-correlation; \( P_\alpha \) and \( P_\beta \) are the power spectral densities of signal 1 and 2 respectively, and \( P_{\alpha\beta} \) is the power spectral density of the cross-correlation of signal 1 and 2. The magnitude-squared coherence, \( M_{\alpha\beta} \), is given by (5):

\[
M_{\alpha\beta} = \frac{|P_{\alpha\beta}(\omega)|^2}{P_\alpha(\omega)P_\beta(\omega)}. \tag{5}
\]

This weighting function is low (near zero) outside the bandwidth of the speaker; the weighting is high (near 1) within the speaker’s bandwidth. This weighting function is applied to the phase prior to applying a linear fit. The linear fit and interpolation is used to estimate the phase, which results in increased resolution of the delay estimate. This signal processing technique is important to enable more effective processing power in a smaller device.

III. Electronics

The electronics consist of a speaker as the actuator and two microphones as sensors. The data was acquired using an Analog to Digital Converter (National Instruments NI USB-9215, [6]) that has 16-bit resolution and 100 kHZ bandwidth. A microcontroller was investigated, but not implemented in this version. The signal flow diagram is shown in Fig. 2.

![Signal Flow Diagram](image)

Fig. 2. Signal Flow diagram: The input signal goes into the DAC, which gets amplified by a power amplifier before going into the speaker. The speaker sends a sound wave through the beer, which is received by two microphones, amplified by an operational amplifier, sent through an ADC, and given as the two output signals.

A. Sensing and Actuation

A low-cost, balanced armature, waterproof speaker (Knowles 2403 260 00132, [7]) was used to drive the audio signals through the fluid. Two low-cost, waterproof, electret condenser microphones (Knowles MR-28406-000, [8]) were used as the sensors for this system. The microphones were each powered with a single AA battery, which provides 1.5 V DC. The signals from the microphone were amplified using a non-inverting operational amplifier (Texas Instruments OPA277, [9]) configuration with a gain of 100. The speaker has a higher current requirement, which was met by using a similar driving circuit utilizing a power amplifier (Texas Instruments LM675, [10]). The op-amps and power amp were powered with a power supply capable of providing ±15 V. The circuit schematic is shown in Appendix A.

B. Data Acquisition

The audio jack of a computer directly connected to the power amplifier was sufficient to provide an input signal to the speakers. The amplified input and output signals were recorded using an NI USB-9215 ADC [6]. This ADC is a 4 channel, 16-bit differential ADC capable of reading ±10 Volts (with respect to COM).

C. Microcontroller

An Arduino Uno board [11] was used, which includes an ATmega328 [12]. It has a 16 MHz ceramic resonator, 32 KB flash memory, and multiple ports for PWM outputs and ADC inputs. The highest PWM frequency is 8 MHz; the analog read resolution 12 bits, with a speed of approximately 10 kHz. One PWM channel may be used for signal generation. PWM may
be used to generate square waves, sine waves or other signals. Two ADC channels may be used for sampling signals from the two speakers separately. The signal processing is handled on-board.

IV. SOFTWARE

MATLAB is used to process the signals, and the microcontroller code will be implemented in future work. The MATLAB code is given in Appendix B.

A. MATLAB Software

The MATLAB routine was implemented, enabling simultaneous and automated data collection, analysis, and display. The signal processing flowchart is shown in Fig. 3. Signal acquisition and analysis proceeded as follows:
1. Microphone signals were recorded through the ADC (100 kHz sampling rate).
2. Microphone signals were high-pass filtered (with a cutoff frequency of 300 Hz).
3. Magnitude-squared coherence estimate between microphone signals was calculated.
4. Fluid transfer function between microphone signals was calculated.
5. Weighted linear fit was applied to phase of transfer function, using magnitude-squared coherence estimate as a weighting function.
6. Slope of linear fit was correlated to speed of sound.

V. EXPERIMENTATION

The experimental apparatus is shown in Fig. 4.

A. MATLAB Signal Output

The weighted linear regression estimation of the speed-of-sound was calibrated by submerging the apparatus in 3 L of tap water, varying the distance between microphones from 0 mm to 80 mm in increments of 10 mm, and observing the effect on the slope of the computed phase recorded test data. The magnitude of the computed transfer function is shown in Fig. 5.

B. Microcontroller Software

The software on the microcontroller includes three parts: the signal generation, acquisition, and processing. The signal generation and acquisition work in parallel, and the data is processed after it is sampled and saved in the flash memory.

The software converts the data from the time domain to the frequency domain via the Fast-Fourier Transform (FFT). This function is programmed in Assembly to improve efficiency. The function is limited to 256 sampling points. Test data was down-sampled from 100 kHz to 10 kHz and then rescaled to the precision of the microcontroller to test its signal processing capability. The maximum error of the computed phase lag (comparing MATLAB to microcontroller) is less than 10% and is much lower in the frequency range within the bandwidth of the speaker and microphones.
The same testing sequence (0 to 80 mm spacing in intervals of 10 mm) was then repeated for a 5% Ethanol / 95% water mixture (by volume) for a total volume of 3 L, to mimic the ethanol content of beer as it brews. A comparison between the 5% ethanol solution and pure water is shown in Fig. 7.

The normalized change in speed of sound, \( \Delta c \), is given in (6), where \( c_{5\%} \) is the slope for the 5% ethanol mixture and \( c_w \) is the slope for tap water:

\[
\Delta c = \frac{|c_{5\%} - c_w|}{c_w}.
\]  

The measured change in the speed of sound \( \Delta c = 1.64\% \), and the nominal change in the speed of sound [13] is \( \Delta c_{\text{nom}} = 1.14\% \). This gives a percent error (relative to the nominal value) of 43.9%. This is only an error in measuring small differences in the speed of sound, not in measuring the actual speed of sound. The nominal speed of sound [13] uses tabulated data, which could fluctuate based on temperature or tap water composition. This is more of a basis of comparison than a perfect reference. These measurements were not tested against a proven device (e.g., a resonance tube) with our fluid.

\[ B. \text{ Sensitivity Analysis} \]

The slope of the phase, \( \vartheta \), is linearly related to the microphone separation, \( \delta \), shown in (7):

\[
\vartheta = A + B\delta, \quad \text{where}
\]

\[
A = 2.396 \times 10^{-7}, \quad B = 1.005 \times 10^{-7}.
\]

The sensitivity analysis of parameters \( A \) and \( B \) is shown in Fig. 8.

\[ VI. \text{ Conclusion and Future Work} \]

This work shows that using a speaker that provides white noise and measuring the delay between the two microphones determined the speed of sound through water and 5% ethanol. The future work aims to improve the compactness and portability of the device and to integrate it with common fermentation vessels. There are several improvements that may be implemented, mostly in mechanical and electrical hardware integration.

\[ A. \text{ Electronic improvements} \]

The next iteration of this device will include a printed circuit board (PCB) that contains the signal driving and conditioning circuitry (see appendix A). An advantage of using a PCB is that the components are centrally located and easily accessible. Low-pass filters will be included in series with the output signals. The speaker and microphones will be powered with a lithium-ion battery and a series of voltage regulators and voltage references. Additionally, this PCB will interface with the microcontroller that will process the signal. The speed of sound (and therefore beer doneness) may be displayed on an LCD screen.

\[ B. \text{ Mechanical improvements} \]

This device will be integrated with a 5-gallon bucket, a common fermentation vessel used by home brewers. It will be mechanically coupled to the lid of the pail via a pair of threaded rods and suspended in the fluid. The device enclosure has spaces that will allow the fluid to be in full contact with the device (speaker and microphones) throughout the fermentation process. Individual holders made from a food grade plastic hold the speaker and microphones. The speaker
is spaced 20 mm from the first microphone, and the first microphone is 50 mm away from the second microphone. Several spacers control the spacing between each element. A CAD model of the device is shown in Fig. 9.

Fig. 9. CAD Model of the final device

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REFERENCES


Lauren A. Chai received her B.Sc. in mechanical engineering from the Massachusetts Institute of Technology (MIT). She is currently pursuing her SM in the Department of Mechanical Engineering at MIT. She is researching passive actuation of rigid-elastic origami lattices for assembly processes.

Prithvi Sundar is a first-year graduate student in the Department of Mechanical Engineering at MIT. He holds a B.S. in Mechanical Engineering from the University of California San Diego. He is currently performing research with the Comprehensive Initiative on Technology Evaluation (CITE).

Marcel A. C. Thomas (SB ’12, SM ‘14) is a PhD student in mechanical engineering at MIT in the Precision Compliant Systems Lab. Marcel is currently working on using over-actuation to compensate for undesirable mode shapes in photolithography machines.

Charles M. Wheeler holds a B.S. in Mechanical Engineering from the University of Colorado at Boulder. He is a Master’s student in Mechanical Engineering at MIT, working in the Precision Compliant Systems Lab. His research develops robust, scalable processes for 3D tissue assembly using sheets of cultured cells.

Wenzhen Yuan is a PhD student in Department of Mechanical Engineering working with Professor Edward Adelson in Perceptual Science Group. She is currently working on tactile sensors and their application in robotics. She has a broad interest in machine perception.