A Thermal Transducing Microphone

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Abstract

As sound waves propagate through the air, adiabatic compression and expansion occurs in the air due to the longitudinal travelling wave[1]. These small fluctuations in air pressure generate a small local change in temperature of the air medium. By utilising the speed of sound dependence on temperature, a high frequency ultrasonic sound wave travelling across an audio frequency wave is phase modulated on its arrival at the receiver. A system has been developed which generates a 630 kHz ultrasonic wave in air and the resulting phase modulation of the received signal is demodulated by a circuit based on a standard FM receiver chip.

Introduction

A. Gas Theory

The air in which acoustic sound waves of normal[4] magnitudes travel, undergoes adiabatic changes as the wave propagates through it. Thus there is a small local temperature rise when the air is compressed and a correspondingly small local temperature fall when expanded.

The equation describing an ideal gas behaviour is

$$PV = nRT, \qquad (1)$$

where P is pressure, V the volume, T the temperature and n the number of moles of gas. R is the universal gas constant.

An ideal adiabatic system has no heat entering or leaving the system. This system can be described by the equation

$$PV^{\gamma} = \text{constant}, \qquad (2)$$

where γ is the ratio of specific heats.

In the case of two adiabatically linked gas states we can write

$$P_1V_1^{\gamma} = P_2V_2^{\gamma} \qquad (3)$$

for the two different pressures and volumes, eliminating the constant.

If we call the difference in pressure between the two pressures P_1 and P_2, p, we can write the pressures as P_0 and P_0 + p respectively. Eq. 3 then can be written,

$$P_0V_1^{\gamma} = (P_0 + p)V_2^{\gamma}. \qquad (4)$$

If the temperatures of the two different states are called T_0 and $T_0 + \Delta T$, we can now write the equations for the two gas states using the ideal gas law (Eqn. 1) as

$$V_1 = \frac{nRT_0}{P_0}$$
$$\text{and } V_2 = \frac{nR(T_0 + \Delta T)}{P_0 + p}. \qquad (5)$$

Substituting Eqn. 5 into Eqn. 4 we can eliminate the volumes V_1, V_2 and also n, and R.

The change in temperature between the two states, ΔT, is then,

$$\Delta T = \frac{T_0}{\left(1 + \frac{p}{P_0}\right)^{\frac{1}{\gamma}-1}} - T_0. \qquad (6)$$

If we now assign T_0 to be the undisturbed air temperature of the surroundings, p as the small perturbation in pressure due to an acoustic wave and P_0 as the ambient pressure we can derive the change in temperature occurring with a passing acoustic wave. ΔT is very small and is approximately ±0.23°C with a very high sound level of 140 dB(SPL), ambient temperature of 20°C, and an ambient pressure of 101.3 kPa.

The sound speed in a medium, c_o, as a function of temperature is usually given in the form

$$c_0 = \sqrt{\gamma RT_0}. \qquad (7)$$

It can be seen that c_0 varies as the square root of the absolute temperature. Hence a variation in air temperature of ΔT will cause a variation in the sound speed, Δc, where

$$\Delta c \simeq \frac{1}{2}\sqrt{\frac{\gamma R}{T_0}}\Delta T. \qquad (8)$$

B. Phase Modulation

The time taken for an (ultrasonic) acoustic wave to propagate between two reference points will change if the air has a different temperature and hence a different sound speed.

Because the temperature changes involved are so small and hence the time delays/advances are also small it is easier to measure the phase change on arrival after traversing a path through the medium.

The speed of sound in a medium, c, is given by c = fλ where f is the frequency of the sound wave, and λ is its corresponding wavelength. Hence

$$\Delta\lambda = \frac{\Delta c}{f}, \tag{9}$$

where $\Delta\lambda$ is the change in wavelength at a fixed frequency due to the medium being at temperature $T_0 + \Delta T$.

Let the distance propagated by the ultrasonic sound waves be d, then the number of cycles, n, of the wave that occur in that distance is n = d/λ so

$$\Delta n = d/\Delta\lambda, \tag{10}$$

where Δn is the difference in the number of completed cycles. It then follows that $\Delta\emptyset$, the change in phase (in degrees) at the end reference point is given by

$$\Delta\phi = \Delta n \times 360^\circ. \tag{11}$$

Because this is a dynamic system (i.e. audio frequency sound waves will be causing the dynamic fluctuation of temperature in the gap between the transducers and hence $\Delta\emptyset$), $\Delta\emptyset$ can be thought of as phase modulation.

An example of the maximum phase modulation as a function of sound pressure level is shown in Fig. 1.

Equations (9, 10, 11) show that to maximise $\Delta\emptyset$, frequency, f, and distance of travel, d, should be maximised. It should be noted that due to physical restraints, d and f can not be increased without limit because absorption of sound waves in air is proportional to the square of its frequency,[2] and very high frequencies are difficult to couple into the air because of the high impedance mismatch.

Increasing d too much also reduces the high frequency response of the system (see Section D).

Note that log of the phase angle variation (Fig. 1) is proportional to the SPL. This means that phase angle is directly proportional to acoustic pressure, since SPL is a log quantity of acoustic pressure.

This important result infers that the microphone will behave in a linear fashion with respect to transducing the acoustic amplitude.

C. Phase Demodulation

We should now consider an equation for an ultrasonic wave propagating between two reference points. Let the driving signal, V_d, have the form

$$V_d = A_d \cos(2\pi f t) \tag{12}$$

and the signal at the receiving transducer, V_r, have the form

$$V_r = A_r \cos(2\pi f t + \Delta\phi) \tag{13}$$

where A_d is the driving amplitude, A_r is the received signal amplitude, f is the frequency of the ultrasonic wave and $\Delta\emptyset$ is the phase change due to the ultrasonic wave propagating a distance, d, and at temperature T0 +ΔT as defined before.

Multiplying Eqn.12 and Eqn.13 we get

$$\begin{aligned} V_d V_r &= A_d \cos(2\pi f t) A_r \\ &\quad \cos(2\pi f t + \Delta\phi) \\ &= A_d A_r \frac{1}{2}[\cos(\Delta\phi) \\ &\quad + \cos(4\pi f t + \Delta\phi)] \end{aligned} \tag{14}$$

where the two resultant cosine terms are recognised as the sum and difference frequency terms.

Observing that - varies at a much lower rate than the frequency of the ultrasonic wave we can low-pass filter the result leaving

$$V_{lowpass} = A_d A_r \frac{1}{2} \cos(\Delta\phi) \tag{15}$$

If we take the second derviative of Eqn.15 with respect to $\Delta\emptyset$ we get

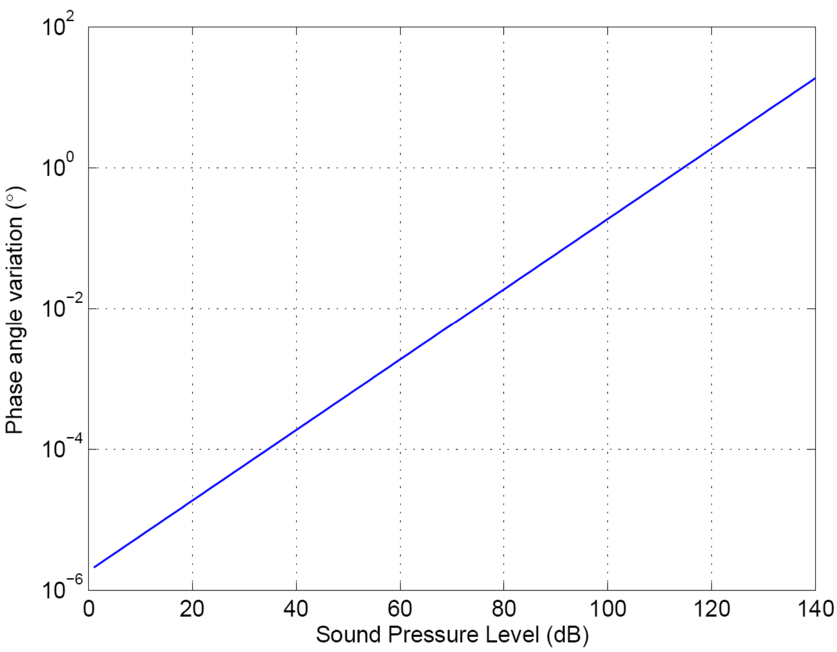


Figure 1: Phase modulation vs Sound Pressure Level

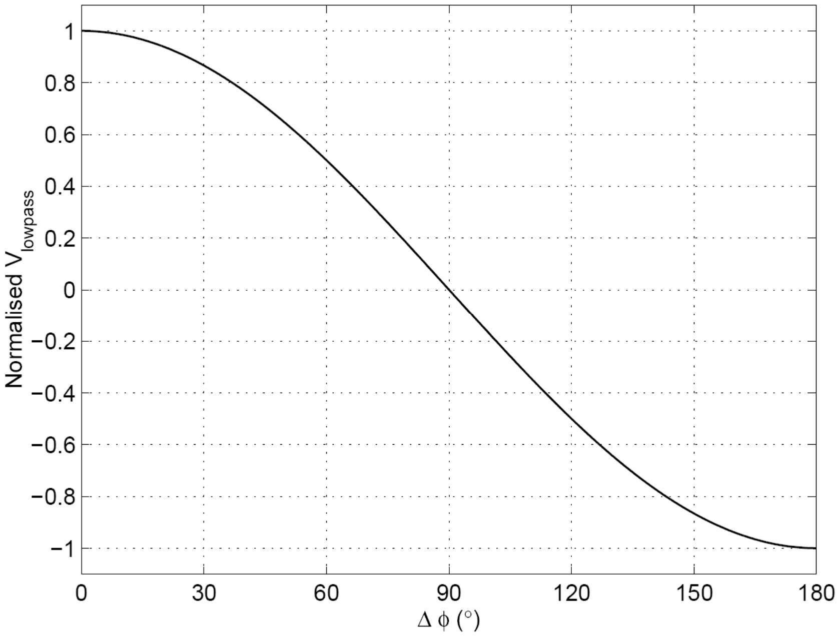


Figure 2: Theoretical $V_{lowpass}$ vs $\Delta\phi$

$$\frac{d^2V_{lowpass}}{d(\Delta\phi)^2} = -\frac{1}{2}A_d A_r \cos(\Delta\phi) \quad (16)$$

and we can see that for $\Delta\phi \approx 90°$ that $V_{lowpass}$ is approximately linear in $\Delta\phi$ since

$$\frac{d^2V_{lowpass}}{d(\Delta\phi)^2} \approx 0, \quad (17)$$

or, $V_{lowpass} \propto \Delta\phi.$

This can also be visually verified by examining Fig. 2.

D. Frequency Response

When the air medium between the two transducers changes temperature at different rates this will not produce the same amount of phase modulation to be present on the received ultrasonic wave. Due to the geometry of the prototype device, there is a non uniform frequency response. Essentially this means that there is a roll off at high frequencies which is directly related to the distance of the gap between the transmitting and receiving transducers.

If we consider the dynamic equation for the speed of sound between the two transducers, c_{actual}, with a sinusoidal plane soundwave of audio frequency f passing through the gap orthogonal to the ultrasonic wave we get

$$c_{actual} = \Delta c_{max}\sin(2\pi f t) + c_0 \quad (18)$$

where t is the time for the ultrasonic wave to transverse the gap. This means that the ultrasonic wave propagates across the gap with a sinusoidal velocity profile. The lower the frequency of the audio wave, the less the velocity profile oscillates. This is illustrated in Fig. 3.

The middle example in the figure has an interesting feature - when the speed of sound profile spends equal times faster and slower than c_0 the travel time is the same as if the ultrasonic wave had travelled at c_0 for the whole distance. This means that for certain frequencies of audio wave, irrespective of the amplitude, there should be no response of the device. The right hand picture in Fig. 3 shows that as the frequency of audio sound wave increases, increasing amounts of the enhanced sound speed cancels with the slower and there is less of a phase change at the receiving transducer.

Eqn. 18 assumes that we know the time of travel of the ultrasonic wave, but this is an unknown variable because it depends on the speed of travel. If we integrate up the travel speed with respect to time, until the predetermined distance, d, between the transducers is reached, we can deduce the travel time t_{travel}, from the following equation,

$$d = \int_0^{t_{travel}} [\Delta c_{max}\sin(2\pi f t) + c_0]dt. \quad (19)$$

The phase change can now be

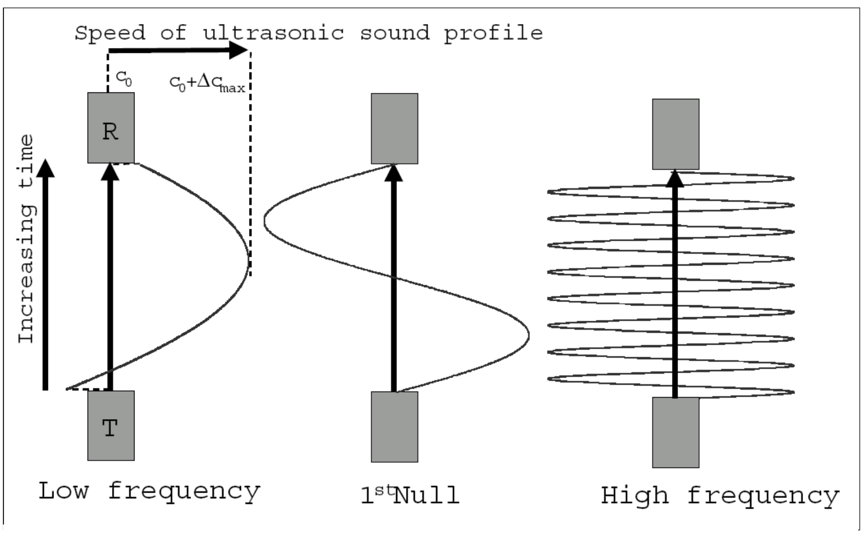


Figure 3: Speed profile of an ultrasonic wave under the influence of different frequencies of audio sound wave.

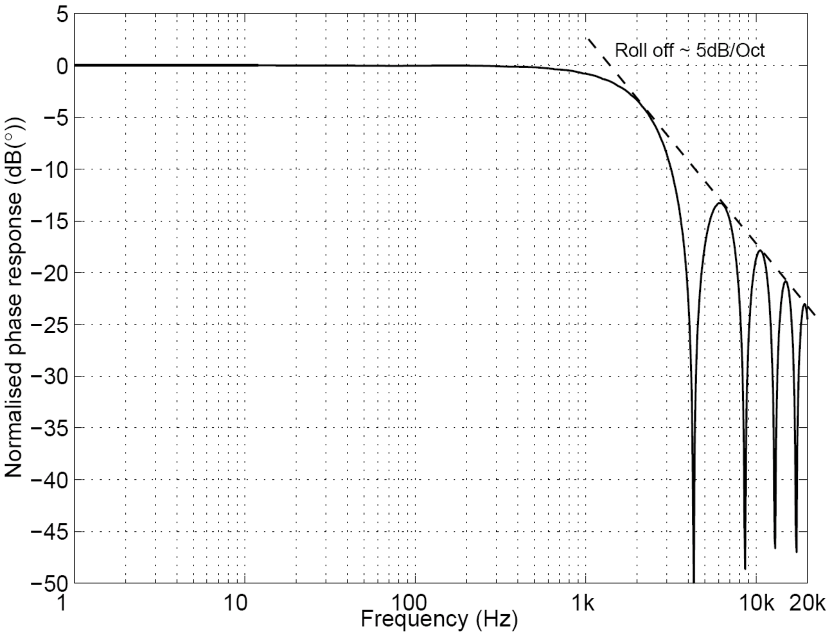


Figure 4: Theoretical Frequency Response, d=8cm

determined by subtracting the mean travel time, t_0, (since $t_0 = d/c_0$) away from the actual time of travel and determining from this small time how much more or less of a cycle has elapsed.

The integral in Eq. 19 was solved numerically with d at 8 cm for the arbitrarily chosen dimensions of the apparatus and this yielded the result seen in Fig. 4 with the response normalised so that the lowest frequency has a response of 0 dB. It can be seen that the higher frequency region containing the peaks and nulls of response tends to roll off with an envelope of approximately 5 dB/octave. The first null appears at the frequency of just above 4 kHz, which is determined by d/c_0. Theoretical plots showing the effect of changing the separation distance, d, are shown in Fig. 14(a). The predicted result of reducing the separation distance between the transducers is that the overall response of the system is lower, but the region of flat response before roll-off is extended to higher frequencies.

Experimental Setup

Ultrasonic transducers are loudspeakers that produce acoustic waves at frequencies above the

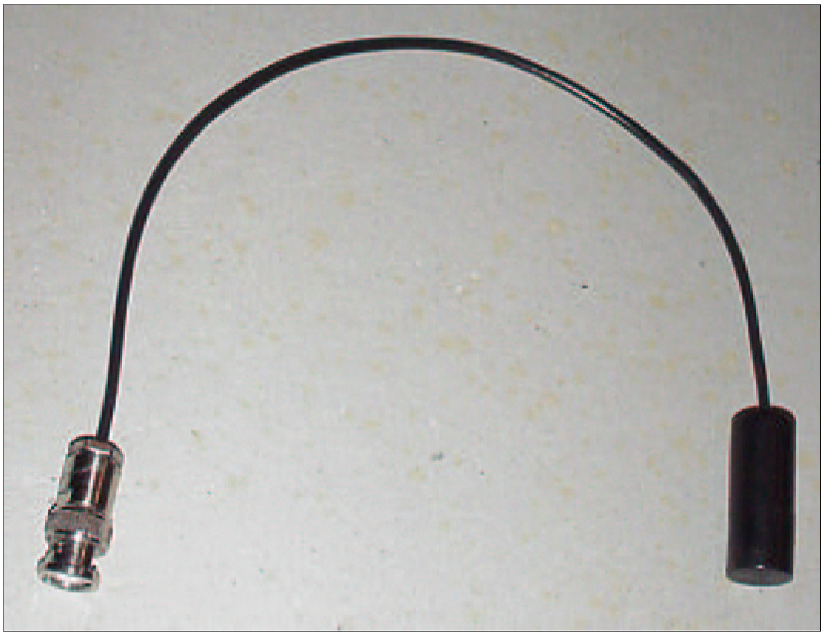


Figure 5: Ultrasonic Transducer

range of human hearing. Conventional woofer speakers are not suitable because their vibrating cone has a large inertia and the voice coil has high inductance. High frequencies are of interest because the higher the frequency of ultrasonic signal, the more sensitive the thermal microphone becomes to audio sound waves. Piezoelectric speakers use the crystal as the diaphragm and are suitable for high frequency speakers because the low inertia means the response can be very fast and crystals are very power efficient. Piezoelectric crystals generate little heat and they are capable of being driven hard, producing acoustic waves efficiently.

Air ultrasonic transducers come in two families. Firstly there is the continuous wave transducers that often have narrow (high Q) frequency bands of operation[5] and the other family is the broad-band shock type transducer with frequency ranges up to about 10 MHz that are used for producing acoustic pulses. A decision was made to use a continuous wave ultrasonic transducer so that the phase demodulation circuitry could be based on standard FM radio techniques.

Two nominally identical high frequency air ultrasonic transducers were mounted facing each other at a separation of 8 cm with one designated a transmitter, and the other the receiver. The distance was limited to 8 cm so that the receiving circuit had sufficient signal to noise ratio.

The piezoelectric transducers used are plastic sealed units about 1.5 cm in diameter and about 3.7 cm long, with BNC connectors at the end of a short lead (see Fig. 5). These transducers, sourced from the Acoustics Department at Auckland University have a nominal resonant frequency of 150 kHz.

A. The Mechanical System

A mount for the ultrasonic transducers was constructed for testing purposes. Due to the low level of acoustical signal incident on the receiving transducer because of the high ultrasonic frequencies used, it was imperative to have a mounting that kept the transducers accurately in alignment for maximum received signal strength. The ability to change the inter-transducer spacing, while still keeping this alignment was also a requirement. The apparatus shown in Figure 6 shows how these parts have been arranged in the final design.

The 1.5 cm diameter plastic sealed transducers were mounted in a PVC jacket so that the whole assembly could be slid in and out of the metallic pipe sleeve towards or away from the other transducer keeping the correct alignment when the transducer was not fully in the metallic pipe sleeve. The jacket meant that a greater range of movement could be possible since the transducers themselves are only about 3.7 cm long. The position fixing assembly was a bolt that screwed down through a hole in the metallic pipe sleeve, pressing against the PVC jacket and holding the transducers in place.

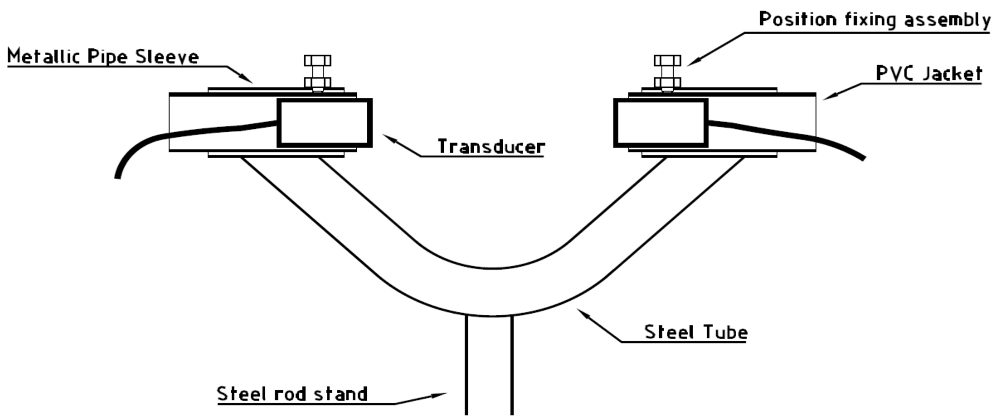


Figure 6: Mechanical schematic representation

B. The Electronic System

A driving signal for the ultrasonic transducer was generated by using a feedback arrangement to lock into the crystal resonance at approximately 630 kHz. A copy of this signal along with the signal received at the other transducer were sent to the demodulator using a typical radio like arrangement based on the Philips SA605 IC. The signal was then low-pass filtered to remove the 630 kHz carrier frequency then coupled from the filter and amplified with a Brüel & Kjær (B&K) measuring amplifier type 2636 with adjustible voltage gain from -30 dB to 100 dB. This external preamplifier also had a selectable 22.4 Hz-22.4 kHz bandpass filter which could be used to further condition the signal before measuring.

For a comprehensive description of the electronic design see the masters thesis entitled `A Thermal Transducing Microphone', author, Bevan Diprose.[3]

Calibration of the Microphone

The microphone was calibrated against a known temperature measuring device, a thermocouple. Because the microphone does not respond to temperature at a single point, but over the complete volume between the transducers, it is important to keep the temperature of air homogeneous over the gap. With that thought, the transducers were subsequently mounted in a copper pipe at the arbitrary separation of 8 cm, with the thermocouple sitting approximately in the centre of the pipe (see Fig. 7). The microphone was found to be inherently less noisy as a thermometer than the thermocouple, and has a much shorter response time. It was therefore decided to bury the whole system in a sand bed having a very high thermal inertia. This caused the temperature of air in the pipe to vary slowly, aiding temperature homogeneity throughout the volume. This also meant that the thermocouple's output could justifiably be low-pass filtered with a long time constant to get rid of measured fluctuations that were not real temperature changes.

The whole system was then mounted on an electric hot plate, so that heat could be injected when required. The hot plate was large, so that it would apply relatively spatially uniform heat to the box containing the sand.

Both the signals from the thermocouple and the microphone circuit were routed to a 12 Bit 2 channel DC capable analogue to digital converter interfaced to a PC.

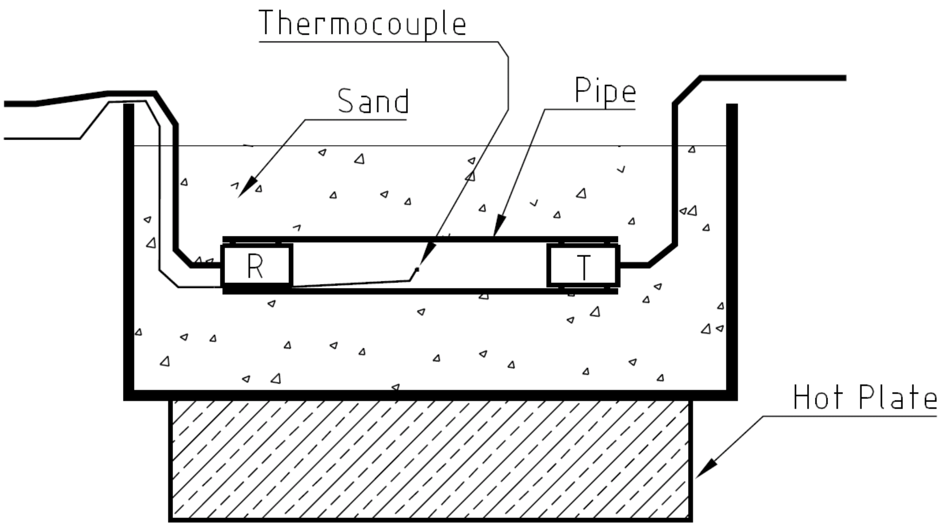
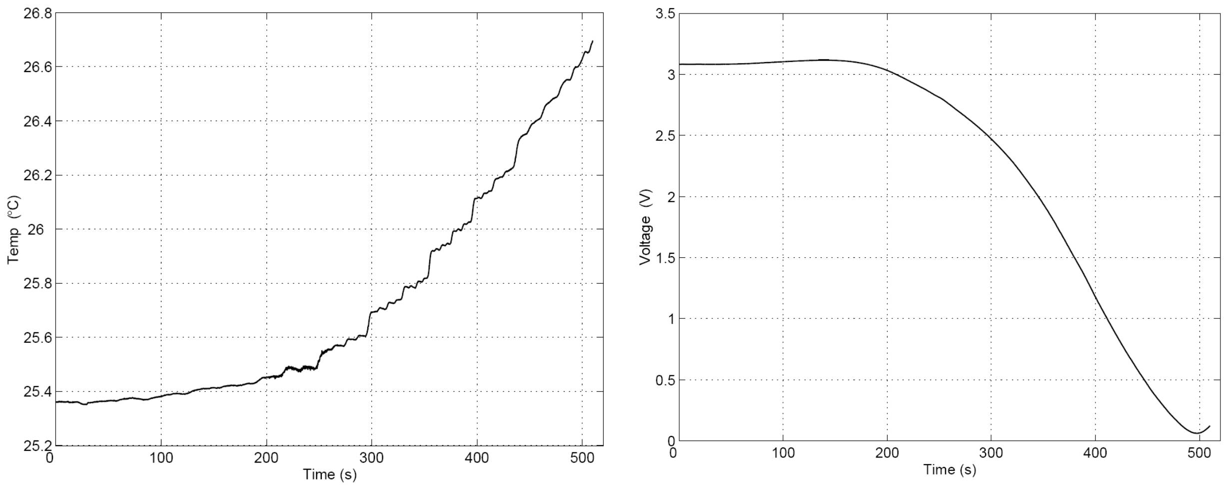


Figure 7: Transducer calibration setup

A. Method

Power was supplied to the hot plate for approximately 1 minute, after which it was switched off for the



(a) Temperature of air gap **(b) Microphone Voltage Output**

Figure 8: Raw data from experiment

remainder of the experiment.

The temperature from the thermocouple, and voltage from the microphone circuitry were then continuously sampled simultaneously and the average (per channel) recorded at a rate of 1 sample per 100 ms over the next 15 minutes or so. The end of the sampling was actually determined as the time when the output voltage of the microphone was observed to reverse direction. This occurs because the output of the phase demodulator varies cosinusoidally with phase (see section on Phase Demodulation).

B. Results

The temperature of the air medium measured with the thermocouple as a function of time is shown in Fig. 8(a).

The voltage output of the microphone circuitry as a funtion of time is shown in Fig. 8(b).

It is observed that the microphone voltage varies extremely smoothly with time compared to what is seen from the thermocouple reading. It would appear that the microphone is less noisy than the thermocouple because it is known that the temperature is not fluctuating, but steadily rising because of the very high thermal inertia of the system. The recorded data from the thermocouple was subsequently low-pass filtered to remove these non-real temperature effects.

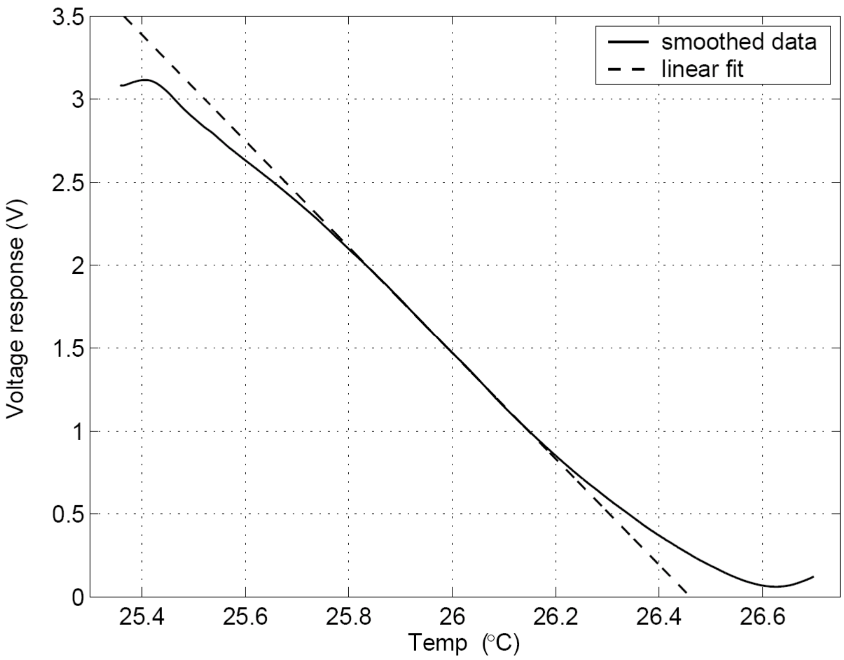


Figure 9: Microphone voltage vs filtered thermocouple temperature

To deduce the sensitivity in Volts/°C we plot the voltage output of the microphone against the known temperature, and calculate the slope of the linear region. The plot of Microphone output voltage against the low-pass filtered thermocouple temperature (Fig. 9) shows a very good correlation with that of Fig. 2, and the absolute value of the slope in the linear region in the centre, the device sensitivity, is determined to be 3.2 V/°C at a separation between ultrasonic transducers of 8cm.

To determine the uncertainty in the sensitivity would require repeated experiments and this was not deemed necessary at this stage.

We can now calculate an estimate of the equivalent acoustic sensitivity in V/Pa as quoted for conventional microphones. This can be obtained by working with Eq. 6 and deriving, p (ΔP), the acoustic pressure from the artificially produced change in temperature, ΔT.

The rearranged equation has the form

$$p = P_0 \frac{(T_0 + \Delta T)}{T_0}^{(1-\frac{1}{\gamma})^{-1}} - P_0, \qquad (20)$$

with the variables as defined originally. Substituting standard values we have T_0 is 293 K, P_0 is 101.1 kPa, and γ is 1.4.

The approximate sensitivity to temperature has already been determined as 3.2V/°C, and since sensitivity is defined as $\Delta V/\Delta T$ in this case we can choose for the moment ΔV to be 3.2V, and ΔT as being 1°C.

Substituting ΔT, T_0, P_0 and γ in eqn. 20, we obtain p ≈ 1.2 kPa. Sensitivity to pressure then follows,

$$\frac{\Delta V}{\Delta P} = \frac{\Delta V}{p} = \frac{3.2\ V}{1.2\ kPa} \approx 2.7\ mV/Pa, \qquad (21)$$

with an 8 cm separation between transducers and at close to DC frequencies.

Testing the Microphone

Having determined the sensitivity of the thermal microphone, it is now possible to acquire some proof that the microphone responds to acoustic waves due to the thermal interaction with the ultrasonic wave and not via some other coupling (e.g. direct coupling in to the receiving ultrasonic transducer). An incident acoustic wave at low frequency (chosen because predicted response drops off with increasing frequency, see

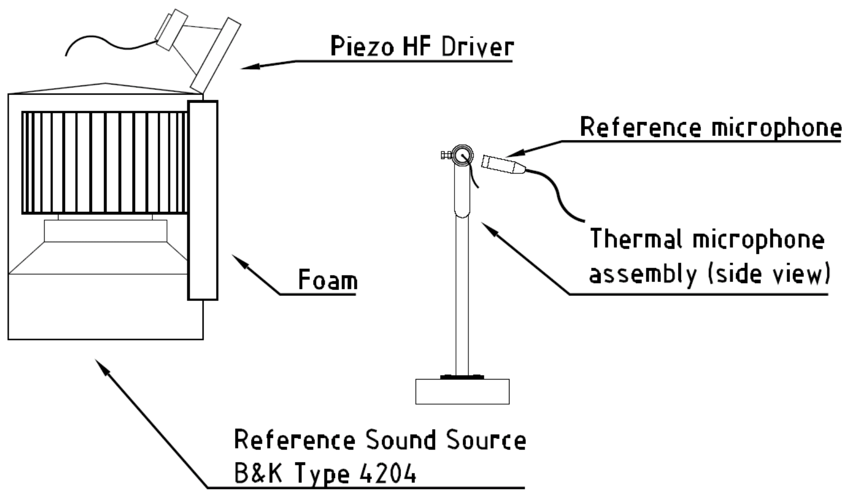


Figure 10: Physical setup of experiment

previous discussion) should appear at approximately the same level as the signal received via a reference microphone, once they are normalised with their respective sensitivities. If the two levels are not similar, this would be good evidence that the thermal microphone is not working via the predicted interaction.

The experimental work for this section was all carried out at the Acoustics Research Centre, University of Auckland using the B&K 2133 dual channel, real-time, frequency analyser.

A. Experimental Setup

Fig. 10 shows the setup used for experimentation. The sound source was composed of the B&K Type 4202 Reference Sound Source and a small piezoelectric high frequency driver used in parallel. The B&K Reference Sound Source produces a quoted sound level of approximately 90.9 dB(A) at 1 m. The spectrum is approximately pink (equal power per octave) from 100 Hz to 10 kHz. This sound is produced by a specially designed high speed, high inertia fan, so that the spectrum remains relatively constant.

Due to some air motion produced by the rotating fan, a foam baffler was required over the front of the reference sound source. Air flow, doppler shifting the ultrasonic signal between the transducers produces much higher phase changes than temperature induced phase change. This principle is sometimes used in ultrasonic anemometers to determine wind speed, but is an unwanted side effect for this experiment. The foam prevented air flow in the direction of the thermal microphone.

The foam, while helping reduce the air flow, also attenuated the level of high frequency components of the reference acoustic waves. This problem was alleviated by mounting a dedicated high frequency driver just above the sound source and pointing it towards the thermal microphone. The H.F. driver was driven from an amplified electronic pink noise source (uncorrelated to the reference sound source) and radiated signals over the approximate range of 1.5 kHz to 20 kHz.

The thermal microphone was placed about 40 cm away from the two sound sources. This is not ideal, because the quasi-spherical waves produced by the H.F. driver are not of large enough radius to `appear plane' in the air gap between the ultrasonic transducers (plane wave assumption was used in the theory derivations), and high frequencies when radiated from the driver are less homogeneous with respect to angular direction. A minimum distance of 1.85 m is normally recommended to minimise the effect of these two problems (IEC Standard recommendation to ensure measurements are not in the near-field of the loudspeaker). The recommended distance was varied experimentally, but after the unfortunate demise of several H.F. drivers, was reduced to 40 cm to preserve the remaining H.F. driver. Using larger distances means having to increase the sound level output in order to have the same sound pressure at the measuring location. This puts more stress on the speaker drivers, and puts them at risk of burning out.

The sound arriving at the thermal microphone was measured with a standard B&K Type 4165 reference microphone placed in the sound field just behind the thermal microphone. The thermal microphone does not physically block the sound passing through the air gap, making an auxiliary simultaneous measurement possible. The arriving sound was measured in the centre of the two ultrasonic transducers, but back about 2 cm further from the sound source as to not disrupt the ultrasonic sound path.

The experiment was conducted in the anechoic chamber to ensure that there was a quiet noise free environment, that only direct sound is used in measurements, and to ensure a place of relatively constant temperature - free from air flow so that the thermal microphone does not respond to ambient air currents.

B. Noise Floor

The noise floor of the thermal microphone was measured after the B&K pre-amplifier and found to have a level of 80.5dBA obtained by averaging over 4 minutes and 12 seconds (4 times the standard 64sec offered by the B&K analyser). In contrast, the reference microphone's noise floor level is only 18.5dBA after its preamp. The noise floor spectrum of them both can be seen in Fig. 11. The ramp effect seen on the thermal microphone's curve is a phenomena of the demodulation and is predicted from radio theory.

C. Frequency Response

Having determined the noise floor, it is now possible to determine the frequency response and distinguish between actual signal and the noise inherent in the system.

The configuration of sound sources in the experimental setup attempts to produce a random noise source containing all frequencies between 20 Hz and 20 kHz at high enough level so that all the individual frequencies can show above the noise floor of the thermal microphone. This task was not easy to achieve. The noise floor level rises with frequency and the thermal microphone was also found to be relatively insensitive to high frequency acoustic waves. This means that high frequencies have to be at a very high level to be seen above the noise. Broad-band high frequencies, while being easily produced, cause significant power to be dissipated in the speaker windings, enhancing the possibility

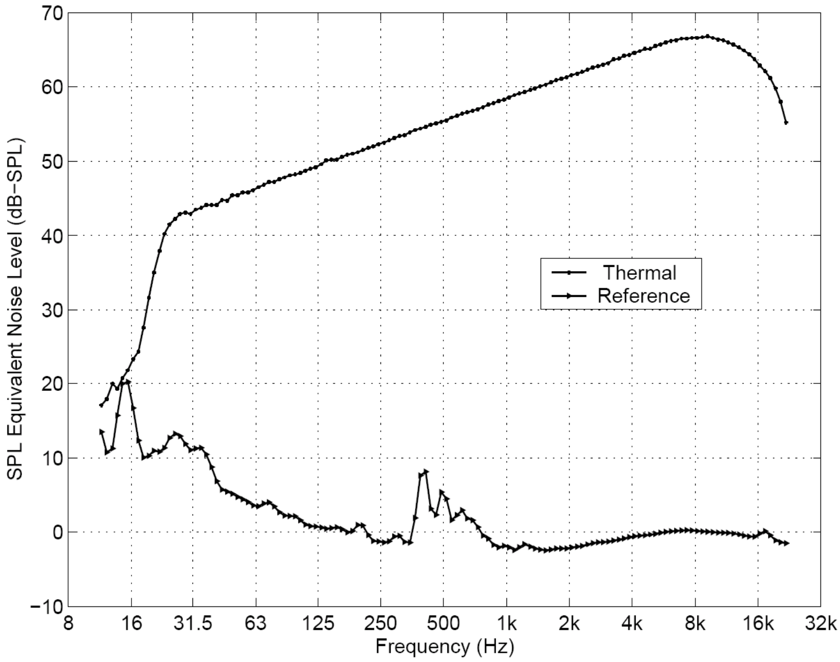


Figure 11: Noise Floor of Thermal and Reference Microphones

of speaker failure. The amount of energy dissipated increases with frequency so boosting the high frequencies can drastically increase the power dissipated in the driver. It is not apparent on listening to broad-band noise that the driver is being overloaded because the normal telltale signs of distortion (harmonic generation) are masked by the broad-band nature of the sound itself. After the destruction of some H.F. drivers, the microphone was moved so that it was close to the H.F. driver, allowing it to be operated at a lower, safe level. The distance of 40 cm was adopted as a compromise.

The standard reference microphone and the thermal microphone were connected to the frequency analyser and the spectrum generated by both devices were measured as the sound sources were activated. As for the noise floor measurements, the signals were averaged over 4 minutes and 16 seconds in 1/12 octave bands.

The spectrum of the reference microphone and the thermal microphone in response to the sound source are shown in Fig. 12. The noise floor of the thermal microphone also shown indicates

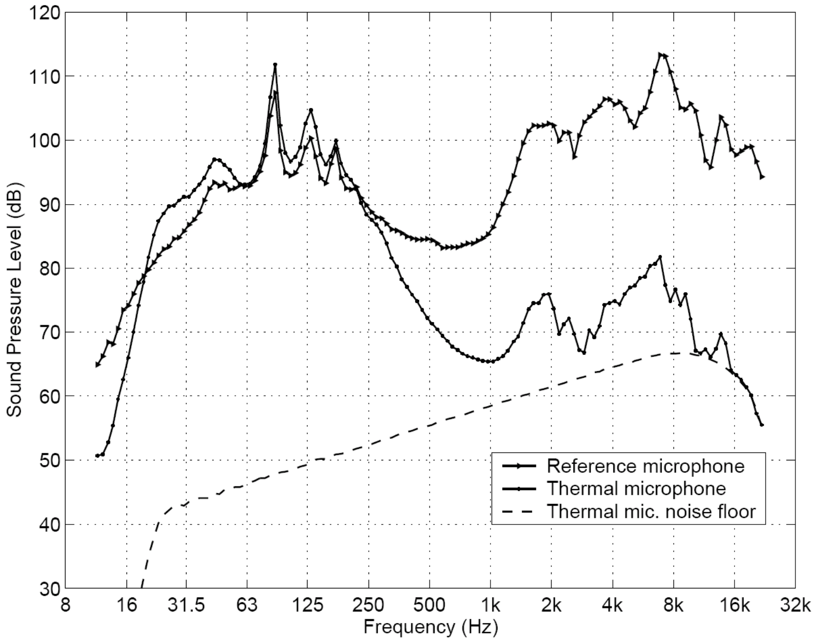


Figure 12: Response to sound source

that frequencies above 14.5 kHz did not appear above the noise for the thermal microphone. The inability of the H.F. driver to produce high frequencies at high SPL is (as previously discussed) the cause of this. The thermal microphone in preliminary testing (with better H.F. drivers) was found capable of transducing signals above 20 kHz, so it is unfortunate that the full audio frequency response can not be determined from these results.

The relative frequency response shown in Fig. 13 is obtained by subtracting the reference microphone's response to the sound source from that of the the thermal microphone for each measured frequency component. This assumes that the thermal microphone responds linearly to sound pressure level, and this linear response was assumed from theoretical predictions (see section on Phase Demodulation) and the sensitivity plot (see Fig. 9).

The response of the thermal microphone from approximately 16 Hz to 350 Hz is within 6 dB of the 0 dB response relative to the reference microphone. This confirms that the thermal microphone is indeed transducing acoustic sound waves due to their thermal interaction on the ultrasonic wave produced by the microphone system. If the audio level acquired from the thermal microphone had been markedly different from that acquired from reference microphone at low frequencies after normalising for their different sensitivities, questions could have been legitimately raised as to the method by which the audio sound waves are transduced by the thermal microphone. Because the thermal microphone's sensitivity was only a ball-park figure, this variation of around 5 dB is acceptable. The slight fluctuations of level over these frequencies is small compared with the overall trend of response and further testing is needed to determine if they are real.

Outside the 16 Hz to 350 Hz range the output varies significantly from the 0 dB mark. The roll-off at the low end will be due to the high-pass filter in the preamplifier and the high end roll-off was also predicted, but the position and rate of this is somewhat different from that expected.

Fig. 13 also shows the comparison between the predicted response and the actual response. The offset between the two responses in the Y-axis is arbitrary because the theoretical response was scaled so that at low frequencies the response was 0 dB. The actual response position also depends on the calculated sensitivity which was only approximately determined. It must also be noted that the

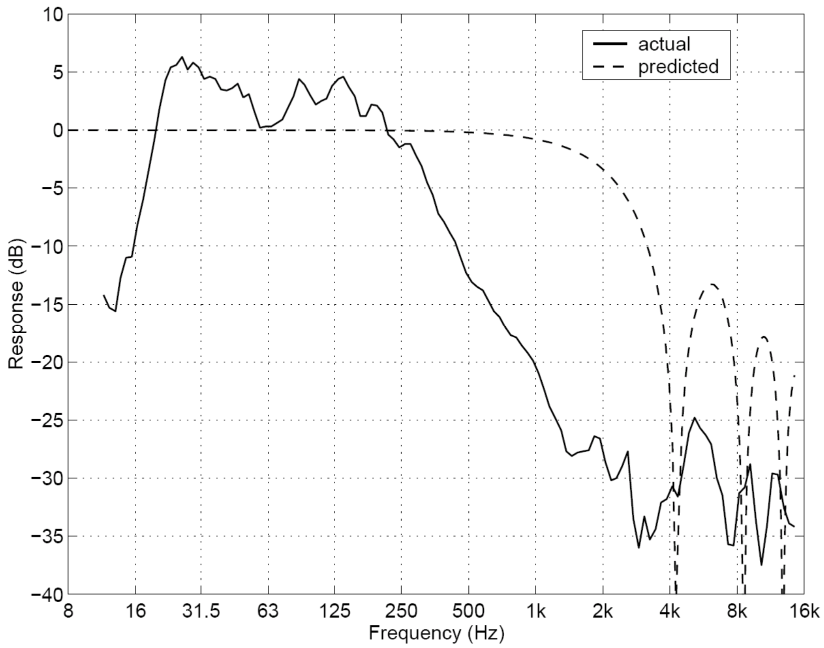


Figure 13: Comparison of predicted response with actual

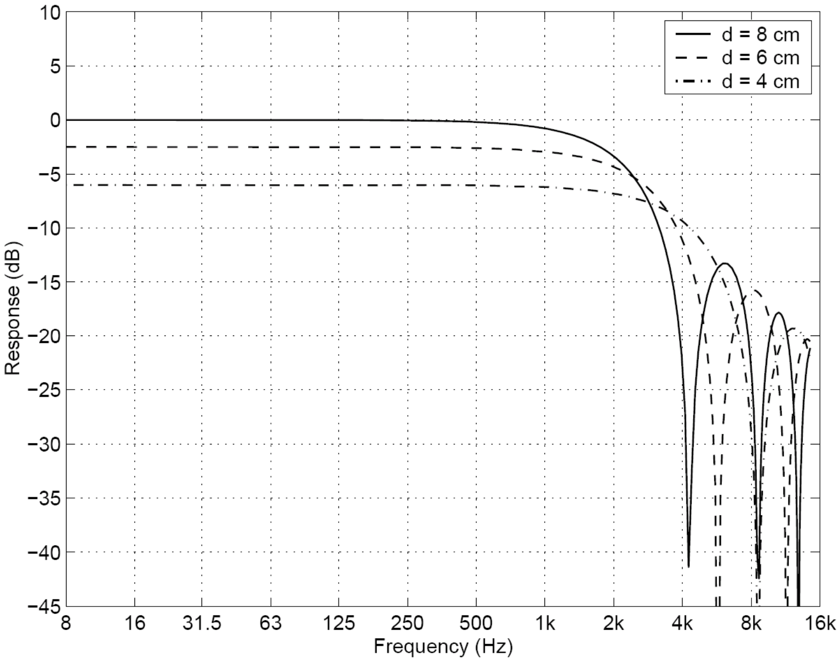
theoretical prediction does not include the high pass filter in the B&K preamplifier circuitry, which causes the roll off below 22.4 Hz.

The two most important deviations of the actual result from that predicted is firstly the roll-off in response that starts at around 200 Hz and falls steadily until about 3.5 kHz, and secondly the predicted peaks and nulls at higher frequencies do not really match. The region between 250 Hz and just over 1 kHz rolls off at a rate of approximately 10 dB/octave. If this frequency range can be considered as the high end roll off region than this roll off rate is approximately twice that predicted roll-off envelope of 5 dB/octave.

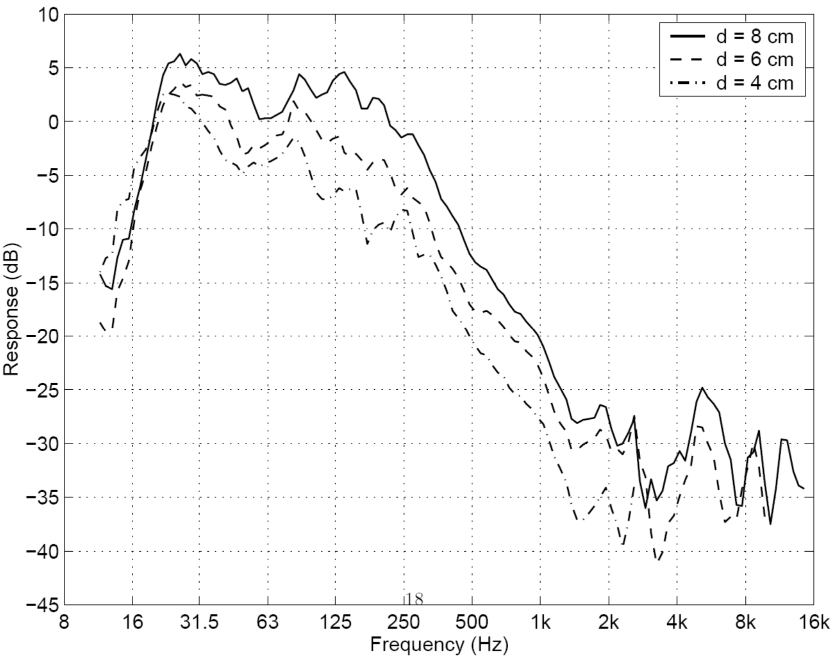
Figures 14(a) and 14(b) show the comparison between the predicted frequency response and the actual results as the distance between the transducers were varied. Again, no account was taken in the theoretical calculations of the 22.4 Hz high pass filter on the preamplifier. Note that unlike the result for d = 8 cm, the actual results for d = 6 cm and d = 4 cm do not cover the range all the way up to 14.5 kHz. This is because the responses for those distances hit the noise floor at lower frequencies. It appears that the predicted overall drop in level as the transducers are moved closer together is evident in the actual results. The prediction of a slightly increased range in frequency before roll off from the flat region as d is reduced is not evident, and again the predicted peaks and nulls do not match.

Because the sound source used for testing was non ideal (due to the incidental audio acoustic wave not being plane between the ultrasonic transducers), it is not possible to definitively discard the theoretical predictions. Increasing directivity of the H.F driver as it produced audio waves at higher frequencies may have caused a significant difference in the recorded sound pressure level between the reference and thermal microphones. The higher frequency quasispherical waves would also appear very non planar to a passing ultrasonic wave in the gap because of the audio wave's short wavelength giving rise to a lower response at the output. Diffraction of the audio wave around the ultrasonic transducers could also have possibly had an effect on the results. These reasons may together explain why the actual results deviate more from those predicted at higher frequencies.

The geometry of the ultrasonic system which differs from the point



(a) Predicted response



(b) Actual response

Figure 14: Effect on response of changing ultrasonic transducer separation, d

source and receiver assumed theoretically in derivations of frequency response may explain why the distinct nulls in the response were not really evident. The transducers have finite diameter and hence the acoustic path length and the response to audio waves have some uncertainty and this means that the predicted sharpness of the null points will be blurred. Sampling in 1/12th octave bands also means that further frequency averaging is taking place as part of the analysis.

Conclusion, Implications and future Considerations

A prototype device has been constructed that can detect audio range acoustic waves in air due to the phase modulation imposed on an ultrasonic wave transversing a high SPL audio sound field. As far as can be ascertained, there has been no previous work employing this principle for construction of a microphone making this project unique.

Considerable and time-consuming effort was expended on the electronics, firstly in design and construction, and then attempting to increase the system's sensitivity, and to reduce the noise floor.

The microphone was calibrated with respect to a slowly varying temperature source and then placed in a reference sound-field and the response measured. The results appear to verify that the mechanism behind transducing the audio sound wave is of the predicted thermal origin. Future work needs to be done with the device to verify, or rethink aspects of the theory related to the predicted frequency response as difficulty existed in obtaining suitable sound sources for testing.

The microphone's poor frequency response and high noise floor make it an inappropriate substitute for the microphones in standard use today, but it may have application in non standard areas, such as the measurement of infrasound, very high level sounds or as a high speed thermometer.

Future Modifications

The main area in need of future development would be in the lowering of the noise floor. This noise floor has been determined to arise from the electronics, and more specifically appears to come from the demodulator. Investigations need to be carried out to confirm that this is indeed the case, and then consideration of a larger dynamic range phase demodulator may be the first step to reducing the noise floor. Alternatively, obtaining an ultrasonic transducer capable of higher frequency operation would increase the phase deviation in the

(Continued on page 27)

Dear Sir,

I've been thinking about the phenomenon of "road rage". Some experiences in Auckland traffic in the past few weeks have made me wonder if the sound quality of horns might be an unnecessary trigger.

I imagine that horn levels and timbre have been chosen so that the sound penetrates and provokes a quick response/reaction in the face of potential danger.

Essentially, horns are designed to startle other road users! So they are loud, and have qualities we would normally associate with unpleasantness i.e they are harsh, aggressive, and intruding.

However, in many cases - perhaps a majority - horns are not used for alerting to a potential danger but for other forms of communication. The more this communication has a negative flavour the more likely, I suspect, it is to nourish road rage. My conjecture is that the typical car horn sound quality adds a negative flavour.

I therefore suggest that there is a case for an additional signal (e.g. a softish "tootle" sound) on cars which can be used in situations where we wish to communicate without discomforting.

I'd like to find out if there are others who would support this idea?

George Dodd

(Continued from page 26)

received signal, allowing the SPL equivalent noise floor to drop while still working with the same demodulator.

Applications

Because the measured characteristics (frequency and noise floor) of the thermal microphone do not compare well to conventional microphones, it is not suitable for studio recording. However, because of its near solid state construction it could be used in more rugged situations where there is rain or dust in the air. Since the ultrasonic system does not impinge significantly on the acoustic signal path there are possible uses in non-obtrusive sound measurement. Measurements of high level sound directly in front of speakers could be taken in live settings and used to control an aspect of the signal path, or explosions could be monitored at close range. The response without the B&K preamplifier goes down to DC and therefore the microphone may have applications to meteorological infrasound measurement.

When determining the sensitivity of the microphone to temperature variations, it was noted that the output gave an extremely clean signal, obtained when the volume of air between the transducers was heated. One application could therefore be found as a high speed thermometer. Because this device measures the temperature of the air directly, and does not have to wait until some metal or other substance is heated up to the same temperature, it has extremely low thermal inertia. This could be valuable when speed of measurement is very important such as measurements of rapidly fluctuating air temperatures.

References

1. **V.A. Shutilov**. Fundamental Physics of Ultrasound, chapter 2, pages 29,30. Gordon and Breach Science Publishers, New York, USA., 1988.

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3. **Bevan C. Diprose**. A thermal transducing microphone. Master's thesis, The University of Auckland, 2003.

4. **Normal magnitude** refers to acoustic waves with amplitudes small when compared to the ambient pressure. Large amplitudes behave in a non linear fashion eg. shock waves

5. **Common nominal** frequencies are 40 kHz, 150 kHz and 225 kHz although lately other frequencies have been released ❏

Reading English Language…

Aoccdrnig to a rscheearch at an Elingsh uinervtisy, it deosn't mttaer in waht oredr the ltteers in a wrod are, the olny iprmoetnt tihng is taht frist and lsat ltteer is at the rghit pclae. The rset can be a toatl mses and you can sitll raed it wouthit porbelm. Tihs is bcuseae we do not raed ervey lteter by it slef but the wrod as a wlohe.

Ceehiro.