Experimental Study of Interaction between Ultrasonic and Audio Waves in Air

Tobias Merkel
Beuth Hochschule für Technik Berlin, Email: merkel@beuth-hochschule.de

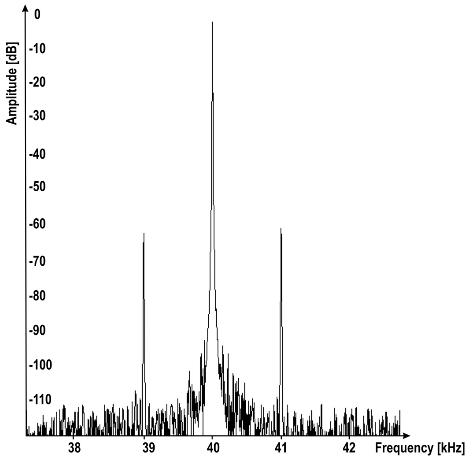

Figure 1: Amplitude spectrum of microphone signal: two side lobes can be seen at a distance of 1 kHz from the ultrasonic carrier.

The Idea

Fascinated by the principles of parametric loudspeakers – well known as audio spotlight or audio beam [1][2][3] – I wondered whether the following would be possible: If a modulated ultrasonic wave is able to transform its modulation into hearable audio waves, will the reverse effect also work? Do audio waves cause a modulation of ultrasonic waves if both waves are interfering with each other in air?

For a first investigation, I conducted a simple experiment[4]: I took an array of 14 piezoceramic transducers to generate an ultrasonic wave beam. The transducers worked at constant amplitude and a resonance frequency of 40 kHz. After travelling a distance of about 5 m, the ultrasonic wave was received by a high frequency microphone. A second wave was generated with a common loudspeaker box, therefore I used a sine tone of 1 kHz. The box was placed in such way that both waves – the ultrasonic wave and the audio wave – were superimposed in the propagation volume. Fig. 1 shows the amplitude spectrum of the received signal. In fact, a modulation of the ultrasonic wave could be observed, which was caused by the audio wave. The amplitude spectrum of the received microphone signal showed two lobes beside the carrier signal at a distance of 1 kHz.

A second observation is worth mentioning: The degree of modulation was independent of the absolute pressure level of the ultrasonic wave. The difference between the amplitudes of the carrier signal and the side lobes – about 60 dB at Fig. 1 – did not change when the amplitude of the ultrasonic wave was altered. That means that non-linear effects of wave propagation in the air are not or not only responsible for this kind of sound modulation, unlike in the case of parametric loudspeakers.

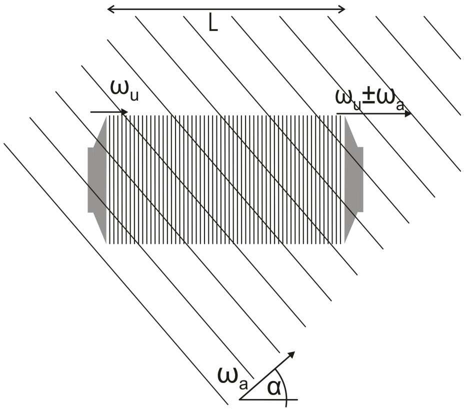

Figure 2: Experimental setup with ultrasonic wave: ω_u is modulated by crossing sound ω_a.

Sound-Sound Interaction

To find out the principles of interaction of both waves a simplified structure is used (Fig.2): The ultrasonic wave with a frequency of ω_u and a constant amplitude is sent through the room. After traveling a distance L it is received by a microphone. A low-frequency wave ω_a propagates at an angle α with respect to the x-axis. The propagation distance L must be small in this regard to obtain plane wave fronts of ω_a.

Two different effects will lead to sound velocity variation Δc of ω_u [5]: on the one hand, the motion of air particles caused by low frequency ω_a, and on the other hand, the variation in medium density.

Δc_m caused by motion of air particles

The audio wave ω_a is characterized by a motion of air particles. The particle velocity v_a is given by

$$v_a = \frac{p_a}{\rho_0 c_0}$$

(with p_a ... being the sound pressure of the audo wave, ρ_0 ... the mean density and c_0 ... the sound velocity) and causes the variation in sound velocity of the ultrasonic wave

$$\Delta c_m = v_a \cos\alpha = \frac{p_a}{\rho_0 c_0} \cos\alpha \tag{1}$$

In the case of orthogonal crossing ($\alpha = 90°$) this effect will disappear.

Δc_ρ caused by changing of the medium density

Propagation of acoustic waves is associated with a periodical change in medium pressure and density. Although the sound velocity c is often assumed to be constant there is a slight dependence from the density ρ:

$$c(\rho) = c_0 + \frac{c_0 \cdot (\gamma - 1)}{2\rho_0} \cdot \rho \qquad (2)$$

with γ being the ratio of the specific heat at constant pressure and constant volume, $\gamma = 1,4$ for air. The second term shows the wanted relationship of changing the sound velocity Δc_ρ by sound density ρ. By replacing the density with sound pressure p, we obtain

$$\Delta c_\rho = \frac{(\gamma - 1)}{2\rho_0 c_0} \cdot p \quad . \qquad (3)$$

The total variation in sound velocity Δc is obtained by adding (1) and (3):

$$\Delta c = \Delta c_m + \Delta c_\rho = \frac{2\cos(\alpha) + \gamma - 1}{2\rho_0 c_0} \cdot p_a \qquad (4)$$

That effect is expected to be very small for 'normal' audio levels. The ultrasonic pressure p_u does not have any influence.

Experiments

Changing the sound velocity over distance L will result in a change phase of the ultrasonic wave. This is measured by Δt referred to the original ultrasonic signal. With (4) and under the condition that the propagation distance L is very small in comparison to the wave length λ of the audio wave we obtain

$$\Delta t = L \cdot \frac{2\cos(\alpha) + \gamma - 1}{2\rho_0 c_0^3} \cdot p_a \qquad (5)$$

To produce the ultrasound wave, a piezoceramic transducer with a sound pressure of 110 dB and a frequency of 40 kHz was used. A high frequency condenser microphone was placed at a distance $L = 20$ cm. This arrangement was placed on a revolvable board for easy adjustment of the angle α. The audio wave was blasted by a common loudspeaker. The distance between the ultrasonic transducers and the loudspeaker was about three meters to obtain nearly plane wave fronts of the audio signal.

Figure 3 shows the results of the experiments. The audio wave – with a pressure level of 88 dB – was sent through the ultrasonic wave field according to figure 2. The averaged maximum of phase differences Δt was assessed at several angles α by using different frequencies ω_a.

Generally there is a good analogy between calculated and measured curves. The measured values are nearly independent of frequency ω_a. The pressure level of the ultrasonic wave has no measurable influence on the measured results. It was varied between 110 and 76 dB (this was the lowest level for that the measurements for the demodulation process were usable); phase shift Δt did not change in this range.

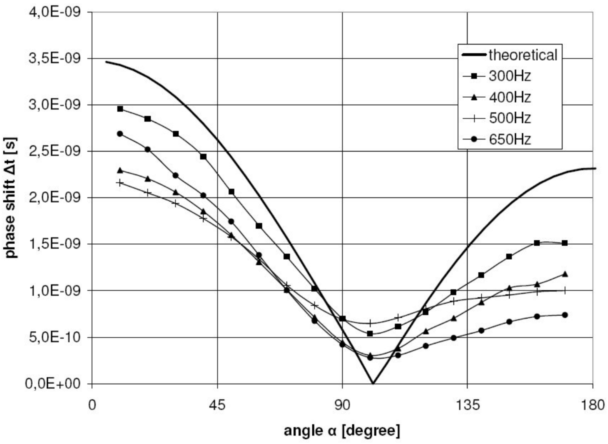

Figure 3: Measurement results: maximum phase shift Δt over angle α compared to calculated values according to (5)

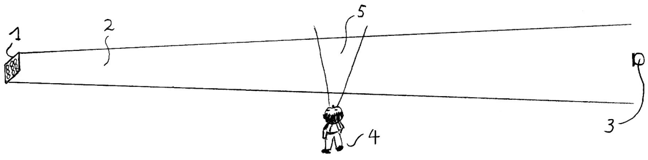

Figure 4: The idea of "virtual" Microphone

The fact, that nearly all measured phase shifts are smaller than the calculated values, is due to acoustic shadows of the experimental equipment. Also the condition that L must be small compared to audio wave length λ_a is only poorly fulfilled.

"Virtual" Microphones

These investigations are made with the aim to develop a new kind of sound recieving system (Fig. 4). The ultrasonic beam (2) is sent over a long distance. It will be modulated by any audiosource (4). The main advantage is that no technical equipment is needed at the place of sound reception (5). That's why we call it "virtual" microphone. The soundsource cannot destroy the microphone and, in the case of speaking persons, the microphone has not to be fastened and will not constrain the movement of the person.

To put this microphone in practice high requirements are to fulfill: Ultrasonic waves with very low phase noise are necessary. The sound aquisition system must work with high dynamic range because the modulation effects are very small. For real time demodulation of ultrasonic signal a very capable DSP is needed working with high accurate algorithm for demodulation.

References

[1] P. J. Westervelt. Parametric acoustic array. *J. Acoust. Soc. Amer.*, 35:535–537, 1963.

[2] D. J. Berktay, H. O.; Leahy. Farfield performance of parametric transmitters. *J. Acoust. Soc. Amer.*, 55:539–546, 1974.

[3] J. Yoneyama, M.; Fujimoto. The audio spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design. *J. Acoust. Soc. Amer.*, 73:1532–1536, 1983.

[4] T. Merkel. *Die Umkehrung parametrischer Schallabstrahlung.* DAGA 2008, Dresden, 2008.

[5] L. Naugolnykh, K. A.; Ostrovsky. *Nonlinear Wave Processes in Acoustics.* Cambridge University Press, 1998.