D

"The audio application"

In the previous chapters the main focus was the Microflowns' capability to measure particle velocity. In this chapter the use for sound reproduction is examined.

The Microflown can be packaged in such way that is able to detect sound pressure instead of particle velocity. The package tunes the frequency response in such way that the pressure Microflown has a maximal sensitivity at 3kHz. This could be used to create a low cost microphone for low quality applications like toys and cellular phones.

An add-on Microflown for a pressure gradient microphone is an application for the high-end audio market. Because pressure gradient microphones are not very sensitive for low frequency sound waves (and Microflowns quite good at lower frequencies) a combination of a

Impedance tube

Microflown and a pressure gradient microphone is realised to create a high-end wide band velocity microphone-pair with a figure of eight polar pattern. Such “two-way” microphone system could be used for recording purposes.

With the use of an add-on plus pressure gradient and a (omni-directional) pressure microphone most polar patterns (directivity figures) can be assembled. This however requires a lot of hardware (i.e. sound sensors). Other methods to enable the directivity patterns are discussed in D3.

The selfnoise of a single Microflown is not low enough to create a wide-band transducer for high-end audio. If the number of Microflowns is increased, the selfnoise will reduce. This idea is used for a study to find out if a Megaflown (a combination of a lot of Microflowns) is practically feasible, see part D3. Until now these efforts resulted not in a low noise microphone but surprisingly, a thermal loudspeaker.

A very brief overview of the many ways of sound recording will be presented in part D5. The aim is to explain what type of microphone is used in what type of recording technique.

D1 THE PRESSURE MICROFLOWN

A Microflown has been packaged in such way that is able to detect sound pressure instead of particle velocity. The usable bandwidth is 3.4kHz which makes is suitable for mass markets like mobile telephones and toys. This chapter is based on the work that has been presented in [1].

Introduction

When an acoustic element is used in a battery-powered application, a number of conditions have to be met. Low power dissipation and operating voltage are required and physical dimensions are limited. It must be moisture resistant and shock proof. For a telecom application, the bandwidth is limited to 3.4kHz and the selfnoise should be in the order of 35dB(A). To be cost effective, the number of additional electrical components should be limited.

In this paper a Microflown is packaged in such way that it becomes sensitive for sound pressure in stead of particle velocity. The Microflown itself exhibits a high frequency roll off. In previous applications the frequency dependent behaviour (the roll-off) was corrected by means of an electric equaliser or by post-processing. Here the package is designed so that acts as an acoustic equaliser. It is based on the Helmholtz resonator.

The Helmholtz resonator

A tube that is terminated by a volume of air creates a Helmholtz resonator. If the tube has a length l and a surface A, the air in the pipe can be described in an electro-acoustic analogy as an inductance $L=\rho l/A$. The volume V can be described as a capacitor $C=Vc^{-2}\rho^{-1}$. Using c=330m/s as the speed of sound in air and ρ as the density of air. The resonance frequency is given by:

$$f_{Helmottz} = \frac{c}{2\pi}\sqrt{\frac{A}{Vl}} \tag{197}$$

To get an impression of dimensions: if a tube with a diameter of 5mm and a length of 7mm closes of a volume of a cubic centimetre of air, the resonance frequency will be 2800Hz. The Microflown will be positioned inside the tube and since the resonator is driven by sound pressure the assembly will be sensitive for sound pressure.

Realisation of the pressure Microflown

To be able to tune the Helmholtz housing, this experimental set up was made with a variable volume due to a movable piston. In this way the resonance frequency could be varied from 600Hz up to 4kHz. The Microflown was simply glued in the throat of the housing. Parts of the housing are depicted in Fig. 114.

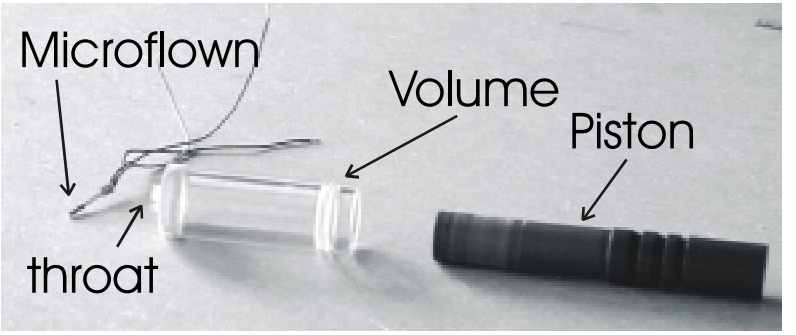

Fig. 114: Parts of the Helmholtz Housing (inner diameter of the tube is 1cm).

The packaging won’t contribute to the noise level because it only alters the acoustic performance of the Microflown.

Measurements

For comparison, a naked (non-packaged) Microflown was measured and after that put into the throat of the Helmholtz resonator.

As can be seen in Fig. 115, the frequency response changes dramatically. Due to the package, the sensitivity of the Microflown is reduced 10dB to 15dB for lower frequencies and at around 3kHz the sensitivity is increased 23dB. The term “sensitivity gain” is arbitrary here

since the naked Microflown is sensitive for particle velocity and the Helmholtz-packaged Microflown is sensitive for sound pressure.

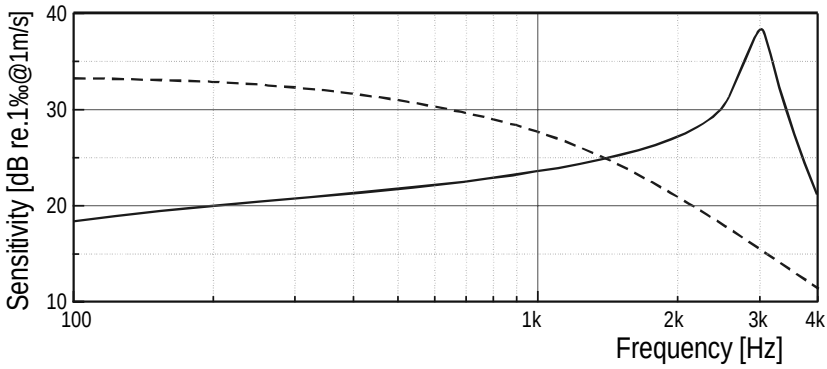

Fig. 115: Sensitivity as function of frequency for a non-packaged Microflown (dotted) and one in the throat of a Helmholtz resonator.

In a standing wave tube, the difference between sound pressure and particle velocity sensitivity becomes clear. A sound pressure maximum can be expected if the particle velocity is minimal and vice versa. The result of a measurement is shown in Fig. 116.

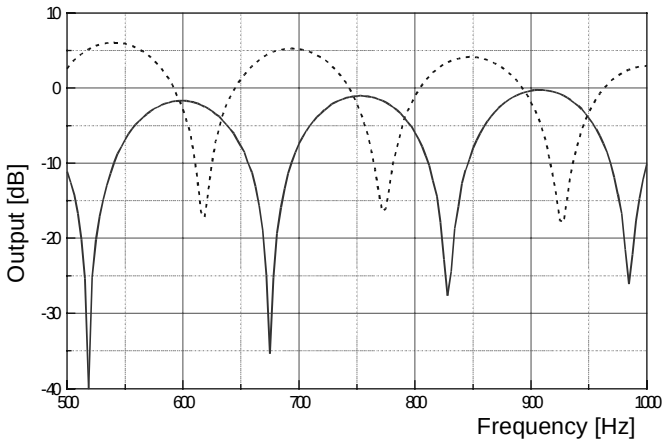

Fig. 116: Output as function of frequency for a non-packaged Microflown (dotted) and one in the throat of a Helmholtz resonator, both measured in a standing wave tube.

The signal to noise ratio at 1kHz is measured 85dB/√Hz which leads to a “A”-weighed selfnoise of 44dB(A) in a bandwidth of 3.4kHz.

Conclusion

Based on the Microflown technology, a sound pressure microphone has been realised. The Microphone has acoustic properties that are comparable to low cost electrets. The selfnoise however has to be improved from 44dB(A) to 35dB(A). This would make it suitable to battery powered mass-market applications like toys and mobile telephones.

Another possible application could be the use of the pressure Microflown in a p**u** probe. In this way the sound intensity probe would consist only on Microflowns.

References

[1] H.E. de Bree et al., The sound pressure Microflown; a novel way of transducing sound pressure, presented at the MME convention in Sweden 2000.

D2 ADD-ON MICROFLOWN FOR A HIGH-END PRESSURE GRADIENT MICROPHONE

Fig. 117: The first prototype of the add-on Microflown build in a Plexiglas housing and the Schoeps MK8 (2000).

Pressure gradient microphones are not very sensitive for low frequency sound waves. The Microflown however has a high signal to noise ratio at these lower frequencies. A combination of a Microflown and a pressure gradient microphone is realised to create a high-end wide band velocity microphone-pair with a figure of eight polar pattern. The electronics, realisation and test results of this complementary “add-on microphone” are presented here. With the use of the wide band microphone pair, the Blumlein recording technique is tested. This chapter is based on the work that has been presented in [1].

Introduction

The bandwidth of sound covers three decades: 20Hz-200Hz, 200Hz-2kHz and 2kHz-20kHz. A conventional, high-end pressure gradient microphone can cover only the last two decades. The signal to noise ratio of a pressure gradient microphone is low at lower frequencies. This is related to the fact that at lower frequencies the signal is decreasing (due to the reduced pressure gradient) and noise is increasing (due to 1/f noise). Compared to a pressure gradient microphone, the Microflown has a relative high signal to noise ratio in the lower two decades ($S/N_{re.1Pa}$>100dB/√Hz). The polar pattern of both types of microphones is similar; a figure of eight. To reach a full bandwidth, the Microflown will be used to pick up frequency sound waves up to 250Hz, whereas a high quality pressure gradient microphone (a MK8 Schoeps) is used to cover the 250Hz to 16kHz bandwidth. The preamplifier of the add-on Microflown is designed to operate on the standard 48V Phantom powering.

The add-on and pressure gradient combination can be compared with a two-way loudspeaker system. One loudspeaker takes care of the lower tones (like the Microflown) and the other transmits the high tones.

Some pressure gradients exhibit a ‘flat’ frequency response down to lower frequencies. Such microphones have an equaliser build in to boost the lower decade. The signal to noise ratio however will not change due to this, because both signal as noise are amplified.

A preliminary and brief test of the velocity microphone assembly (the add-on Microflown and the high-end pressure gradient microphone) has been investigated for the Blumlein recording technique; a method that requires the use of two figure-of-eight microphones.

The Microflown

This paper presents the first steps of the Microflown technology in the professional audio. Instead of sound pressure, the Microflown is capable of measuring particle velocity, which is closely related to (sound) pressure gradient. So in an audio perspective the Microflown can be seen as a pressure gradient microphone (with a figure-of-eight directivity pattern) that has a good signal to noise ratio from 0Hz up to 1kHz. For frequencies higher than 1kHz the frequency response has a decay, see chapter 2 and A1: *"Model of the Microflown"*.

Preamplifier of the add-on Microflown

In professional audio the Phantom powering is a well-known standard. (DIN EN 61936 & DIN45596 cover details). Initially, it was specified as 48 volts (P48) provided through 6.8k resistors. Similarity of these resistors is quite critical, the two resistors must be matched within 0.4% tolerance. There are also systems defined at 24 and 12 volts (P24 and P12), but those are more rare than the traditional 48V feed. Those lower voltage systems use lower resistance values for powering. Most modern condenser microphones will operate at a wide range of Phantom power voltages.

The name "Phantom power" is related to telecommunications: a Phantom line is a configuration in which a telegraph signal is imposed on a balanced voice pair by using the ground return. This method was also suitable for powering microphones in studios.

The Microflown should be used as an add-on to the gradient microphone preferably without the use of additional powering. One simple way of achieving this is by placing the resistors in series with the two Phantom resistors.

The sensitivity of a Microflown element itself is defined by the differential resistance variation of both sensors due to a particle velocity of 1m/s; that is 146dB PVL re. 50nm/s. At a power dissipation of 40mW to 60mW the sensitivity of a non-packaged Microflown is about 35‰; when packaged, the sensitivity rises 15dB (5.5×) to 200‰. At 94dB (the acoustic reference level) the particle velocity is 2.5mm/s, the differential resistor variation at this velocity is 0.5‰.

Add on

The simplest solution for a preamplifier would be a connection of each of the phantom resistors to one of the resistors of the Microflown, see Fig. 118. The output will be “a balanced output” or a differential voltage, amplified by e.g. a mixing table. Due to 0.5‰ differential resistor variation per 94dB PVL the output will be 4.2mV/94dB. The capacitor cuts-off the signal for frequencies above 250Hz.

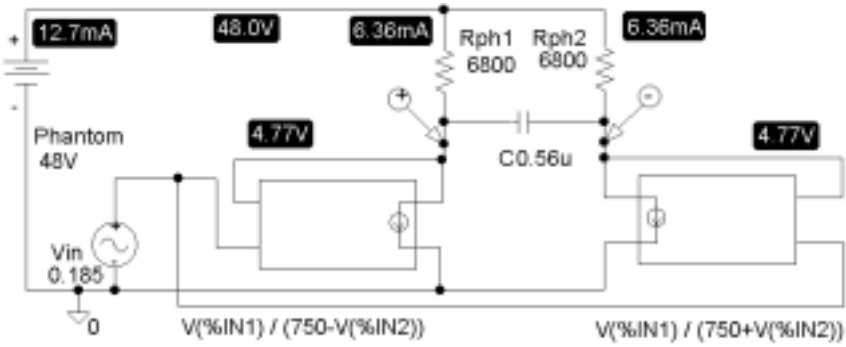

Fig. 118: Simple circuit for phantom powering. One box represents one sensor-wire of the Microflown.

The sensitivity of the simple solution that is presented in Fig. 118 is too low compared to that of the MK8 (10mV/Pa or 10mV/94dB). The circuit that is presented in Fig. 119 has a higher sensitivity that can be adjusted by R_{gain} up to 32mV/94dB. The sensitivity for higher frequencies can be attenuated with capacitors Clp1 and Clp2 (1st order low pass filter).

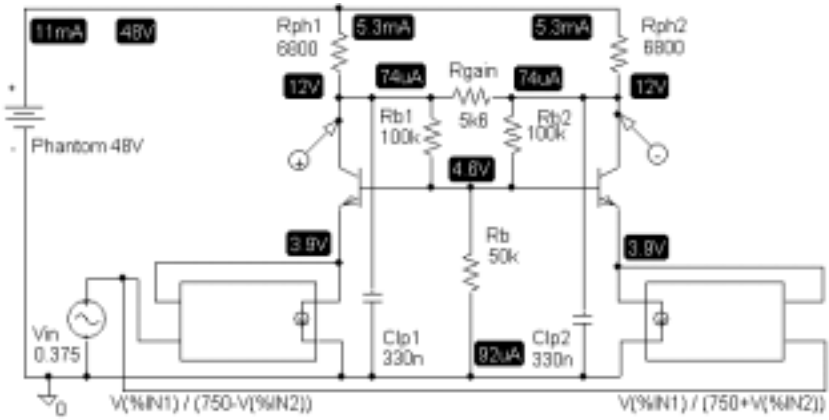

Fig. 119: The Phantom preamplifier.

Calculations show that at low frequencies the Microflown should have a theoretical maximal S/N (P=20mW per sensor, $T_{sensors}$=600K, the sensitivity at 94dB is ΔR/R=0.5‰) of (see chapter 4: "*Electronics and noise*"):

$$\frac{S}{N} = 20Log_{10}\frac{1}{\sqrt{4k}}\cdot\frac{1}{\sqrt{Bw}}\cdot\sqrt{\frac{2P}{T_{sensors}}}\cdot\frac{\Delta R}{R} \tag{198}$$
$$= 20Log_{10}\frac{1}{\sqrt{4\times 1.38\cdot 10^{-23}}}\cdot\frac{1}{\sqrt{1[Hz]}}\cdot\sqrt{\frac{0.04[W]}{600[K]}}\cdot 0.5\cdot 10^{-3} = 115dB/\sqrt{Hz}$$

The simulated noise output of the circuit that is shown in Fig. 119 is 21.5nV/√Hz, the sensitivity (low frequency) 10mV/94dB and the simulated theoretic maximal S/N of the Microflown plus preamplifier is thus 113dB/√Hz. The A-weighed noise output voltage of the Microflown is given by:

$$V_{noise} = 21.5nV\times\sqrt{\int_0^{100kHz}\frac{W_A^2}{1+f^2/250^2}df} = 250nV \tag{199}$$

Using W_A as the A-filter. The theoretic minimal A-weighed selfnoise of the Microflown in this (limited) bandwidth is 2dB(A). The Mk8 has a sensitivity of 10mV/Pa and a selfnoise of 17dB. The A-weighed rms noise output voltage thus is 1.4µV.

Add on

The total noise output is expected to be: √(noise Microflown^2 + noise microphone^2) = √(250nV^2+1.4μV^2) = 1.42μV, which will lead to a selfnoise of 17.06dB(A) for the total system. (The noise of the add-on Microflown can be reduced if the signal is suppressed on the mixing table for frequencies above the 500Hz). So it can be expected that the noise of the total system (Microflown plus MK8) will be not noticeable higher than the noise of the MK8 itself.

Sound recording techniques: the Blumlein configuration

The add-on Microflown has been realised and its sensitivity is adjusted to that of the pressure gradient microphone. The full-band velocity microphone assembly is tested in a Blumlein configuration. The results of this recording technique are discussed in paragraph 6.

The Blumlein configuration is named after Alan Dower Blumlein who was born in England, 1903. He has become well known in the audio community for the creative nature of his stereo contributions. The configuration consists on two figure-of-eight microphones that are angled 90 degrees to each other. The midpoint of the stereo soundstage should be at 45 degrees of both microphones that are in phase, at 90 degrees in the polar plot of Fig. 120.

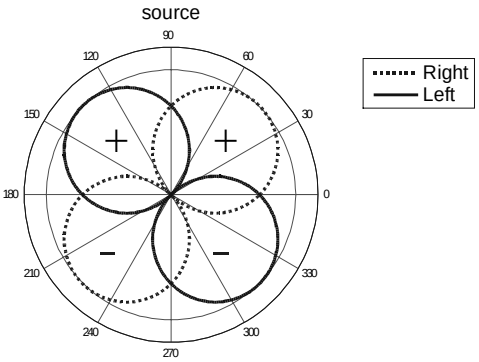

Fig. 120: A Blumlein configuration consisting of two figure-of-eight microphones rotated 90 degrees to each other. The sound source is located at the 90 degrees direction.

The Blumlein technique is not the most common microphone technique in use today, because (among other considerations) the pressure gradient microphones that must be used for the technique are not capable to pick up low frequency sound waves.

The technique itself is best suited when both the sound source and the recording environment are favourable. In reality, however, such ideal conditions are not often present. Recording engineers and producers encounter the common problem of uninteresting or even unpleasant acoustics within modern performance spaces. Because the Blumlein technique faithfully reproduces these poor acoustics and blends them into the stereo image, other techniques must be employed to control their influence in the recording. Thus, although the Blumlein technique often yields excellent results, it is also susceptible to many real-world problems that can affect its performance [2],[3].

A musician was recorded who was playing the drums. The drum set consisted on two base drums. The full band microphone pair was configured according the Blumlein configuration. The Microphones where positioned 80cm above the ground and on one meter distance from the drum set. During the recording the add-on Microflowns where switched on and of every five seconds. One can clearly hear the difference in the lower frequencies.

Measurements

The frequency response of the microphones is measured in a standing wave tube (SWT), an anechoic room and a tube with a passive sound absorber at the end.

In an anechoic room and in the tube with the acoustic damping at the end the calibration is relatively simple since the specific acoustic impedance is (ideally) a real constant. However, for low frequencies it is difficult to damp sound waves. Due to reflections for the lower frequencies, the specific acoustic impedance is not constant anymore. In other words for lower frequencies the conditions become non-anechoic and the calibration method fails. Therefore, for lower frequencies an alternative calibration method is deployed additionally. This method is based on full reflection instead of full absorption: a standing wave tube, see chapter AV *"Calibration methods"*.

The add-on Microflown and the pressure gradient microphone were measured in a standing wave tube with a length of 8 meters and a

diameter 16cm, the distance *(l-x)* was 4 meters. In a standing wave tube with such diameter the maximum frequency that can be used is about 1200Hz. The reference sound pressure microphone was a one-inch B&K 4144 sound pressure microphone with a sensitivity of 30.5mV/Pa. A Tascam MA8 preamplifier was used to power the microphones.

Frequency response

The anechoic frequency response is shown in Fig. 121. The sensitivity of the Schoeps MK8 is dropping at lower frequencies and the sensitivity of the Microflown is high at lower frequencies. Due to absorption difficulties below 50Hz the anechoic room cannot be used properly.

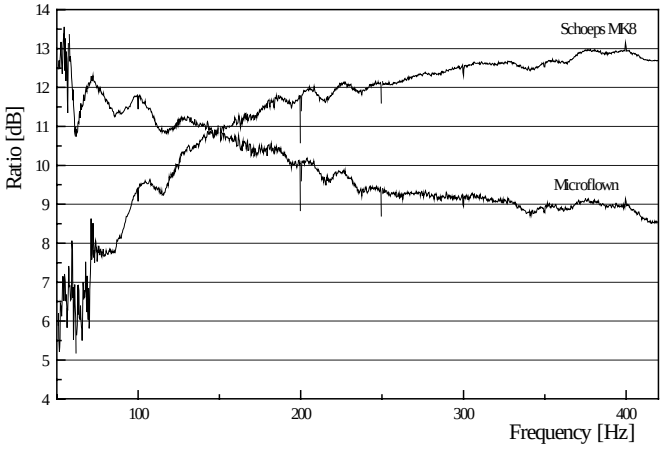

Fig. 121: Frequency response of a Microflown (with low pass-filter) and a pressure gradient microphone in an anechoic environment (Bw=500Hz).

The measurement results of the tube with a passive sound absorber at the end (Fig. 122) again show a “flat” low frequency behaviour of the Microflown (below 250Hz) and a low frequency roll-off of the MK8.

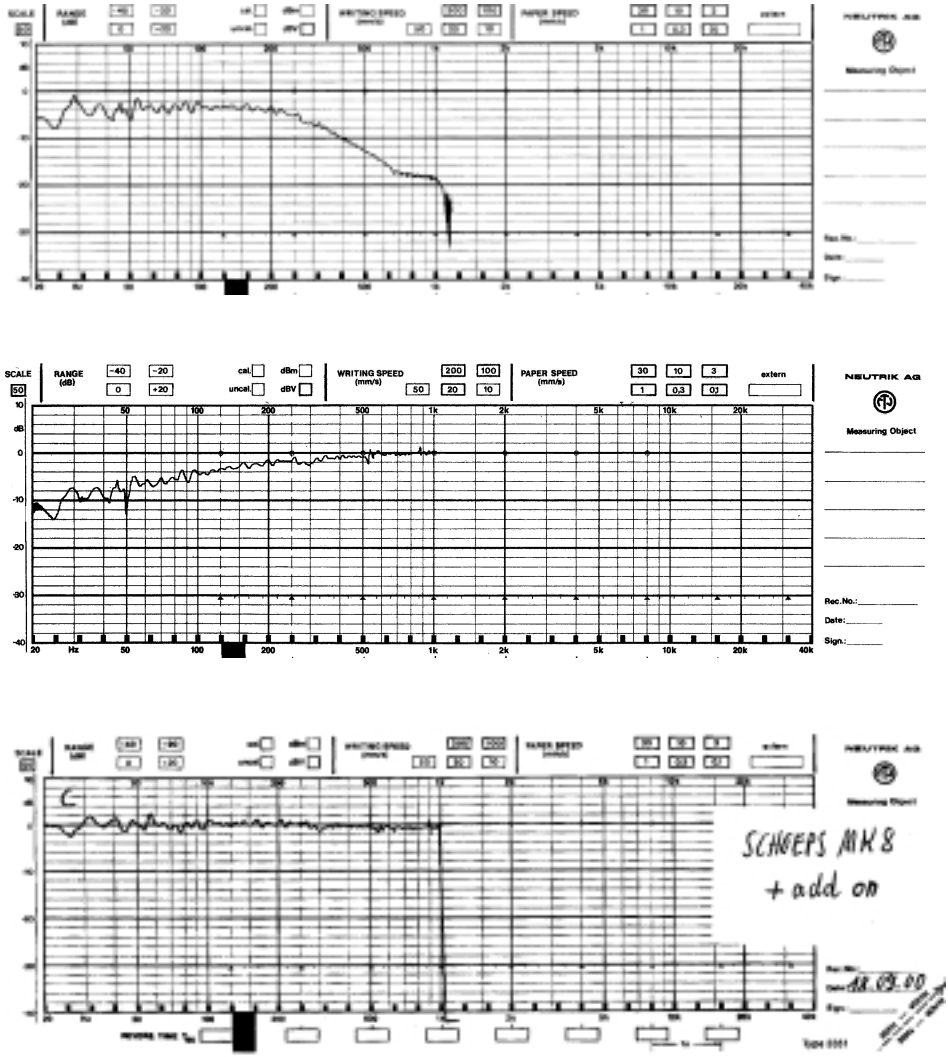

Fig. 122: Frequency response of a Microflown (with low pass-filter, upper curve) and a pressure gradient microphone (middle curve) in a tube with passive damping ate the end Bw=1kHz. The frequency response of the combination of both microphones plus proper filtering is shown in the lower curve (Schoeps 2000).

Because the ripples in the frequency response of both Microflown as MK8 are found at the same frequencies, one can conclude that the ripples are caused by non-anechoic conditions (Fig. 121 and Fig. 122).

Add on

The standing wave tube measurements are presented in Fig. 123 and Fig. 124. They show the frequency response of both probes in a bandwidth of 17.5Hz to 1100Hz. The coherence in this bandwidth is unity apart from the minima. As expected, the frequency response of the pressure gradient microphone shows a roll-of below 250Hz. The add-on Microflown has a flat low-frequency response.

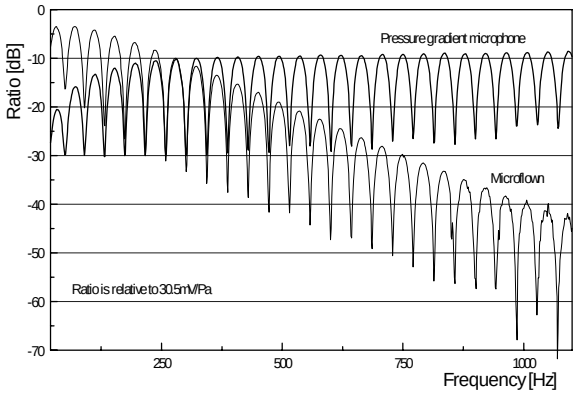

Fig. 123: Frequency response in a standing wave tube of a (low pass filtered) Microflown and a pressure gradient microphone (Bw=1100Hz).

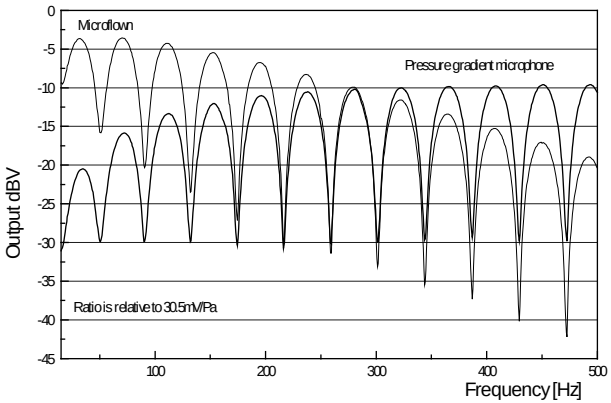

Fig. 124: Frequency response in a standing wave tube of a (low pass filtered) Microflown and a pressure gradient microphone (Bw=500Hz).

Polar pattern

At 19Hz, 107Hz, 980Hz and 2kHz the polar patter of the add-on Microflown is measured in the standing wave tube. Only half the plot was measured. To reveal the figure of eight response (for 19Hz and 107Hz), the figures are presented on a linear scale in Fig. 125. The polar plots shown in Fig. 126 are represented on a logarithmic scale. The lateral rejection of the Microflown (ratio between on- and off-axis response) is at 19Hz, 30dB and at frequencies above 100Hz the rejection is more than 50dB, the polar patterns of both add-on Microflown and the MK8 are similar.

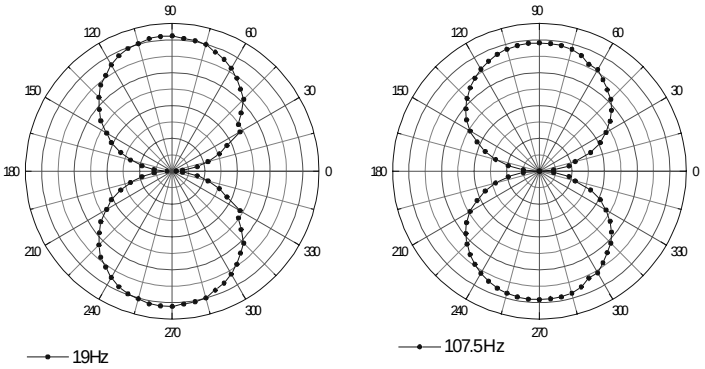

Fig. 125: Polar patterns of the add-on Microflown (linear scale).

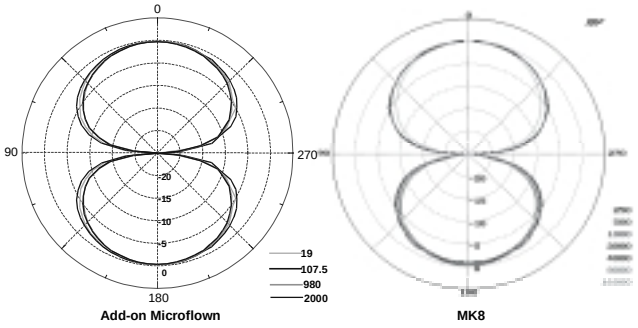

Fig. 126: Polar patterns of the add-on Microflown (left) and the MK8 (right), logarithmic scale.

Noise properties

The noise of the add-on Microflown and the Schoeps MK8 were measured, both the A-weighed noise as the spectral density. The density is measured in a 1Hz bandwidth. The preamplifier was adjusted in a such way that the sensitivity of the add-on Microflown and the Schoeps MK8 were 0dBV per Pascal. The high gain was used to enable the noise measurements.

An external third order low pass filter has been used to filter frequencies above 250Hz (The –3dB point of the filter was on 250Hz) to enable A-weighed noise measurements for the add-on Microflown.

The A-weighed noise output of the add-on Microflown was measured: 39.8µV, the A-weighed noise output of the Schoeps MK8 was measured: 172µV. The A-weighed selfnoise of the add-on Microflown is thus: 6.2dB(A) and the A-weighed selfnoise of the Schoeps MK8: 18.9dB(A). The total noise of the two microphones will be $\sqrt{(39.8\mu V^2+172\mu V^2)}=176.5\mu V$, resulting in a 19.15dB(A) selfnoise.

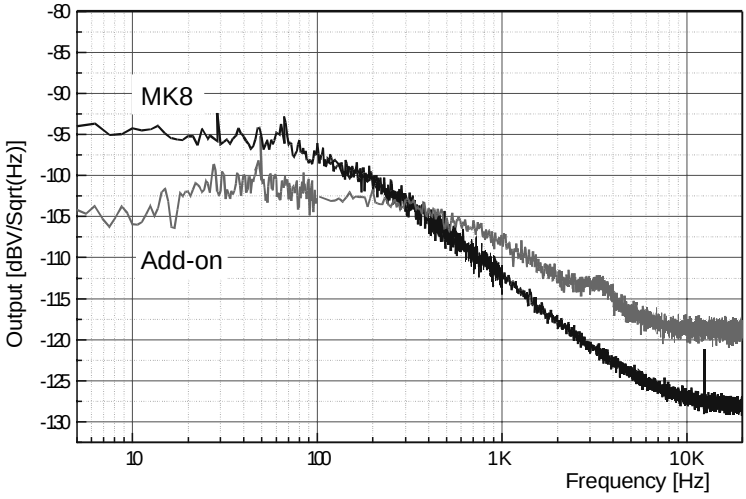

Fig. 127: Noise density of the add-on Microflown and the Schoeps MK8. The sensitivity of both microphones was adjusted to 1dBV per Pascal.

The A-weighed noise figures are higher than expected. The Schoeps MK8 should have a 17dB(A) selfnoise. A not completely silent measurement set-up could be the cause this extra 1.9dB. The noise of the

Microflown is also higher than expected. The selfnoise should be 2dB(A) using only first order filtering. The low pass filtering however was exercised with an external post-filter. The (third order) filtering was adjusted so that the –3dB point was on 250Hz (corner frequency 600Hz). It should also result in a selfnoise of -1dB(A). The selfnoise of the Microflown could be higher than expected due to hum (and a little background noise that also increased the measured selfnoise of the Schoeps MK8).

To get a good insight into the noise behaviour of the Microflown no external and internal filtering is utilised for the spectral density measurement. As can be seen in Fig. 127, the spectral noise density of the add-on Microflown is about -102dBV at lower frequencies. With a sensitivity of 1V/94dB, the signal to noise ratio is measured 102dB/√Hz in stead of the 113dB/√Hz that was expected from theory. As can be seen the noise density is increasing at lower frequencies. A possible explanation is that some acoustic noise influenced the noise measurements.

Fig. 128: Realisation of a Microflown and a Schoeps MK8 in one housing (Schoeps 2000).

Conclusion

- A Microflown that is used as a pick up for low frequency sound waves has been realised. This “add-on Microflown” is used complementary to a sound pressure gradient microphone (Schoeps MK8) that rolls off at frequencies below 250Hz to get a full bandwidth “figure of eight” response.
- A preamplifier that enables the Microflown to operate on the standard 48V Phantom powering has been designed, simulated and tested.
- Due to the limited bandwidth (250Hz) the selfnoise of the add-on Microflown is very low: 6.2dB(A) compared to the A-weighed selfnoise of the Schoeps MK8: 18.9dB(A). The selfnoise of the Microflown-microphone pair therefore will increase just 0.3dB.
- Preliminary tests show that the Blumlein recording technique is possible with the full-band microphone assembly. Compared to a Blumlein recording with only two pressure gradients, more low frequencies were observed.
- A standing wave tube appeared very useful for the low frequency calibration of (pressure gradient) microphones.
- The Microflown can be a good extend to reach a full “figure of eight” bandwidth. The measured selfnoise of the Microflown is higher than expected however low enough for this application.

Discussion

- An additional higher-order low-pass filter has been used to cut-off the high frequencies. This filter simulates equalising equipment that can be found on mixing consoles. It would be more elegant to build this filter within the preamplifier.
- Regarding the available Phantom power it is also possible to use two Microflowns in one housing. This will result in a 3dB lower selfnoise. Both Microflowns can be placed perpendicular to each other to create a more omni-directional (or better toroidial) polar pattern.
- If the pressure gradient microphone is replaced by two spaced (miniature) microphones and a Microflown in the middle, a

system is created that is capable of detecting full band sound pressure and pressure gradient. The polar pattern of such microphone assembly can be varied from full-band omni, cardioid, super cardioid to full figure to eight.

- For frequencies up to 600Hz the signal to noise ratio of the add-on Microflown is higher than the Mk8. If the size of the pressure gradient microphone reduced in a manner that the usable bandwidth will be shifted one octave, the add-on can be used for frequencies up to 500Hz and the pressure gradient up to 30kHz.
- Regarding the power consumption, it is possible to operate an add-on Microflown and a regular pressure difference microphone at one 48V Phantom input.
- The noise spectrum provides information about the signal to noise ratio.

References

[1] H.E. de Bree, Add-on Microflown for a high-end pressure gradient microphone, presented at the 109th LA AES September 2000 (no. 5181).

[2] R. Streicher & F. Alton Everest, The new stereo soundbook, Audio Engineering Associates California, 1998, ISBN 0-9665162-0-6.

[3] J. Wuttke, Microfonaufsätze, Auflage 1998, Dr. Ing. Shoeps GMBH.

D3 UNIDIRECTIONAL MICROPHONES BASED ON MICROFLOWN TECHNOLOGY

A unidirectional microphone (i.e. a microphone that has a non-omni polar pattern) can be seen as an assembly of two types of microphones: a certain percentage omni (or pressure sensitive) and the rest is figure-of-eight or bi-directional (velocity sensitive). A cardioid for example consists of 50% omni and 50% velocity sensitivity.

Compared to velocity microphones, pressure microphones are of very high quality. They have broad band sensitivity (at least 20Hz-20kHz) and are low noise transducers (less than 20dB(A)). The velocity microphones on the other hand do not have such excellent features. Their bandwidth is limited (250Hz-16kHz) and the noise figures are usually a few dB less good than the pressure microphones.

Due to the fact that the bandwidth of velocity microphones is limited, also the non-omni microphones microphone assemblies (like cardioids and super cardioids) do have directionality difficulties at lower frequencies; after all, they consists on a certain percentage velocity sensitivity. On the other hand, directionality is sometimes required. Coincident recording techniques such as the “XY-stereo” configuration use two cardioids or super-cardioids. So one should used tricks to reduce low frequency problems.

With the use of an add-on Microflown, that has been presented in [57], a velocity microphone can be constructed that has a wide band (20Hz-16kHz) sensitivity with also good (signal to) noise properties and directionality. This assembly can be used for Blumlein recording techniques. If such microphone assembly (add-on plus pressure gradient) is combined with a pressure microphone, (almost) all possible polar patterns can be created, see Fig. 129. These principles are well understood. Two other ways of creating a directional microphone will be presented here.

The first idea for a microphone assembly that has wide band directional properties is based on a (velocity) Microflown and (sound pressure) microphone for low frequency directionality, for the mid-band two separate pressure microphones are used in such way that they create

a cardioid microphone. For high frequencies only one pressure microphone will be used that becomes directional due to shielding.

The second idea is based on a velocity gradient method. Two Microflowns will be used in such way that the polar patter becomes unidirectional. This is a method that can only be used for lower frequencies (lower than 1kHz). So this assembly must be seen as an add-on for a directional microphone.

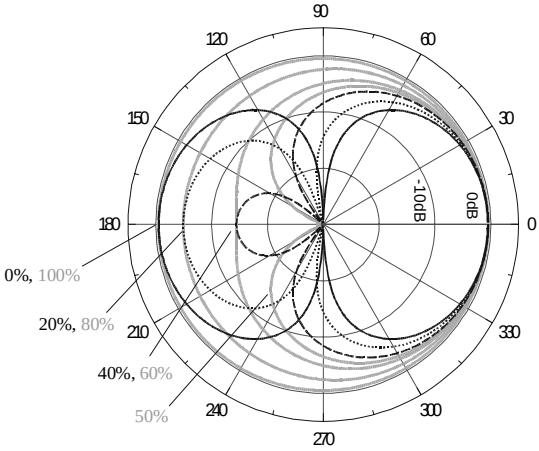

Fig. 129: Sum of a *x*% sound pressure and (100%-*x*%) velocity transducer. A ratio of 50/50% creates a cardioid.

Method I: Add-on plus microphone pair

The microphone assembly that is presented in this paragraph consists on two miniature pressure microphones that are positioned on both sides of a Microflown. The method has three operation modes in that are different for different bandwidths. The result however will be a full bandwidth microphone with a cardioid polar pattern. The microphone positions are shown in Fig. 130.

For low frequencies the Microflown the Microflown has a high signal to noise ratio and will be used to generate signal with a figure-of-eight polar pattern. The sum of both (pressure) microphones is used to create an omni-directional signal. The sum of these two signals (pressure and

velocity) can be used to generate a microphone assembly that has a cardioid shape of polar pattern.

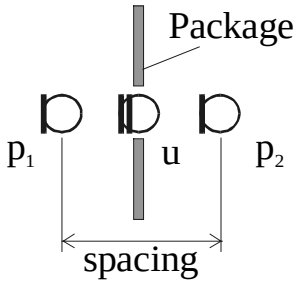

Fig. 130: The realisation of the Microflown with microphone pair.

For mid-range frequencies, the Microflown signal is suppressed and the velocity signal is obtained from the pressure gradient that is generated by subtraction signal of the two pressure microphones. According to the linearised equation of momentum conservation the particle velocity can be generated by:

$$\begin{aligned} u(x,t) &= -\frac{1}{\rho_0}\int \frac{d}{dx} p(x,t)dt \approx -\frac{1}{\Delta x \cdot \rho_0}\int p(x+\Delta x,t) - p(x,t)dt \\ &= -\frac{1}{\Delta x \cdot \rho_0}\int p_1 - p_2 dt \end{aligned} \tag{200}$$

Using ρ_0 as the density of air and Δx as the spacing. Again the sum signal of both microphones is used for the pressure reception.

For high frequencies, the signal of only one pressure microphone is used. The package of the Microflown that is located behind the microphone will act as an obstruction leads to directionality of the microphone.

Method II: Unidirectional Microflown pair

The Microflown has a figure-of-eight sensitivity. This means that the directivity pattern is cosine shaped, and that the phase shifts 180^0 between forward and backward travelling sound waves. This phase shift becomes clear if the output signal is compared to the output signal of a

sound pressure probe, see Fig. 131. If the output signals of both type of microphones are summed, a sound wave from behind (Fig. 131B) will result in no signal at all. This idea is used to create a sound probe that is sensitive for sound waves in one direction.

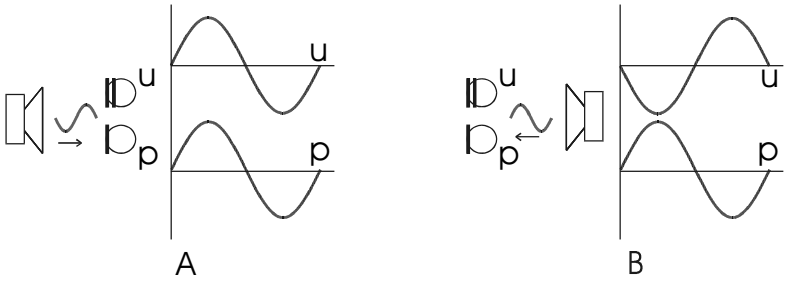

Fig. 131A: The phase shift is zero if the sound source is in front of the Microflown and the sound pressure probe. (B) If the sound source is located behind, both probes are 180^0 out of phase.

The summation of the output of an omni-directional pressure microphone and a particle velocity microphone will not result in a full rejection in one direction but in a cardioid shaped polar pattern, see Fig. 132.

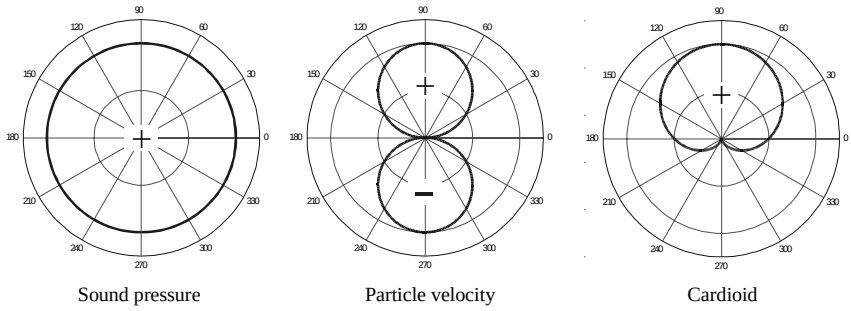

Fig. 132: The summation of a sound pressure microphone and a particle velocity microphone will result in a cardioid shaped polar pattern (linear scale).

Two closely spaced Microflowns can be used to obtain a sound pressure signal that has a $\cos^2(\theta)$ shape polar pattern. This way of sound pressure determination is already shown in [55]. The method is somewhat similar to the pressure gradient method that is used in p-p sound intensity

probes [60]. The velocity gradient method is based upon the linearised equation of mass conservation:

$$\frac{\partial \rho(r,t)}{\partial t} + \rho_0 \cdot \frac{\partial\big(u(r,t)\big)}{\partial r} = 0 \tag{201}$$

Where ρ is the density of the medium, r as the radial distance and u as the particle velocity. For sound levels below 135dB it is assumed that the increase in pressure p and increase in density are linearly related: $p=c^2\rho$. Therefore:

$$\frac{1}{c^2}\frac{\partial p(t,r)}{\partial t} = -\rho_0 \cdot \frac{\partial\big(u(t,r)\big)}{\partial r} \tag{202}$$

Using c as the speed of sound. Integrating both sides will lead to:

$$p(t,r) = -\rho_0 \cdot c^2 \cdot \int \frac{\partial\big(u(t,r)\big)}{\partial r} dt \tag{203}$$

In practice, the particle velocity gradient, $\partial(u(t,r))/\partial r$, can be approximated by measuring the velocities, u_a and u_b, at two closely spaced points, with a separation of Δx:

$$p = -\frac{\rho_0 \cdot c^2}{\Delta x} \cdot \int (u_b - u_a)dt \tag{204}$$

This approximation is valid as long as the separation (Δx) is small compared to the wavelength of interest. The particle velocity component of the sound wave is calculated by the sum of both Microflowns.

In fact one should use six Microflowns to calculate sound pressure. A three-dimensional extension of Eq. 2.14 is the use of the divergence of the particle velocity to calculate the sound pressure:

$$p = -\rho_0 c^2 \cdot \int \frac{\partial u}{\partial x} + \frac{\partial u}{\partial y} + \frac{\partial u}{\partial z} dt \tag{205}$$

However, here the one-dimensional realisation has been used. One can deduct that the sound pressure determination of this method has a “second order figure of eight” polar pattern ($cos^2(\theta)$ shape). (One way to look at this is that pressure gradient generated by (omni directional) pressure microphones results in a figure-of-eight response and so a gradient generated by elements that have a figure-of-eight response for them selves will generate the square of a figure-of-eight response.)

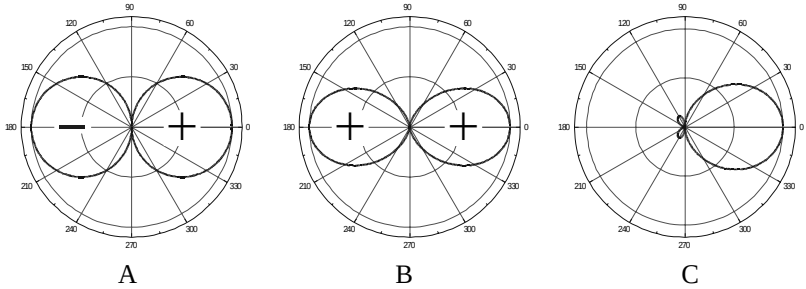

Fig. 133: Linear polar plot of A: the particle velocity, B: the calculated pressure and C: the summation of both.

The signal to noise ratio of the Unidirectional Microflown Pair

The particle velocity gradient signal that is generated is proportional to the frequency as long as the spacing is small compared to the wavelength. To obtain the pressure, the velocity gradient should be integrated as shown in Eq. 2.14. In other words, integration means division by *f* for the frequency response and 90 degrees phase shifting for the phase response. Due to this the noise of the Microflown, will be increased for lower frequencies.

The signal of the UMP

To be able to reach high signal to noise levels at low frequencies, the spacing between both Microflowns should be as large as possible. The spacing of the two Microflowns however should be smaller than half the wavelength.

A Microflown spacing of *d*=15cm seem to be sufficient. A simulation of the frequency response of the non-integrated and the integrated velocity gradient is depicted in Fig. 134. The integration is established by an amplifier with a gain of A=750/f for f>10Hz. Suppose that the output of a certain Microflown 1 is given by $\mu_1 = \mu \cos \omega t$ then the output of the second can be written as $\mu_2 = \mu \cos(\omega t + kd)$. Using $k=2\pi f/c$, f the frequency and c the speed of sound. For plane waves the ratio of the sound pressure and particle velocity is given by: ρc. In order to get the

sensitivity and dimensions right, Eq. 58 should be divided by this value. The output of the UMP will be given by:

$$\begin{aligned} Output &= \mu_1 + \mu_2 - \frac{c}{d}\int \mu_1 - \mu_2 dt \\ &= \mu\cos(\omega t) + \mu\cos(\omega t + kd) - \mu\frac{c}{d}\int \cos(\omega t) - \cos(\omega t + kd)dt \\ &= \mu[\cos(\omega t) + \cos(\omega t + kd) + 2\frac{c}{d}\sin(\frac{kd}{2})\frac{\cos(\omega t + kd)}{\omega}] \end{aligned} \quad (206)$$

For small d the amplitude of the sum of μ_1 and μ_2 can be estimated by: $2\mu\cos(\omega t)$. Eq. 59 then can be simplified to:

$$Output \approx \mu\cos(\omega t)[2 + 2\frac{c}{d}\sin(\frac{kd}{2})/\omega] \approx 4\mu\cos(\omega t) \quad (207)$$

If the free field sensitivity of the Microflowns is μ=10mV/Pa, the output of the UPM will be approximately 40mV/Pa.

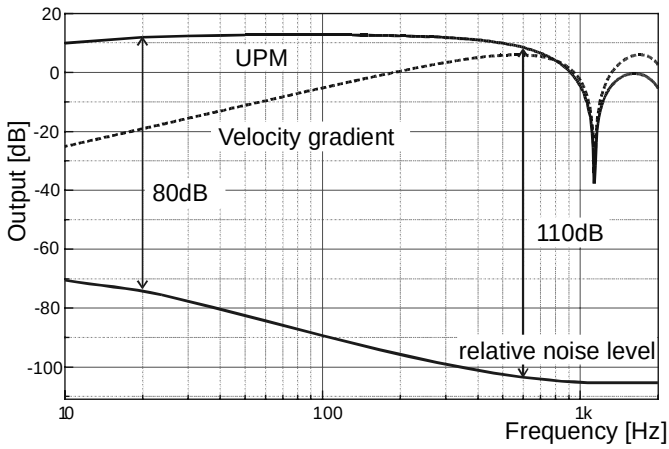

Fig. 134: Expected signal and noise levels of the UMP.

The range of polar patterns that can be expected is shown in Fig. 135.

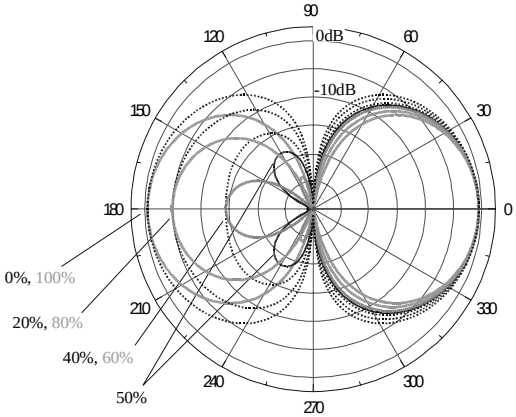

Fig. 135: Sum of a *x*% calculated sound pressure and (100%-*x*%) mean velocity.

The Noise of the UMP

Due to the integration the noise output of the sound pressure determination will be dominant for lower frequencies. After all it is multiplied by A=750/f for 10Hz>f>750Hz.

The signal to noise ratio of the system can be estimated from the signal to noise ratio of a single Microflown. Suppose that a Microflown has a S/N of 100dB/√Hz for frequencies below 750Hz. The UMP will have a four times higher signal level but about a A=750/f higher noise level.

So at 750Hz the signal to noise level of the system is about 110dB, at 75 Hz this is reduced 20dB to S/N=90dB and at 20Hz it is reduced 30dB to S/N=80dB. The signal to noise ratio of a UMP that is based on Microflowns with a supposed signal to noise ratio of 100dB/√Hz is depicted in Fig. 134.

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D4 THE MEGAFLOWN

The selfnoise of microphones is a important figure. For studio applications it should be less than 20dB(A). The Microflown that is used for measurement purposes has a selfnoise in the order of 40dB(A). A Microflown that is useful for broadband studio applications must have a 20dB better signal to noise ratio than a measurement Microflown.

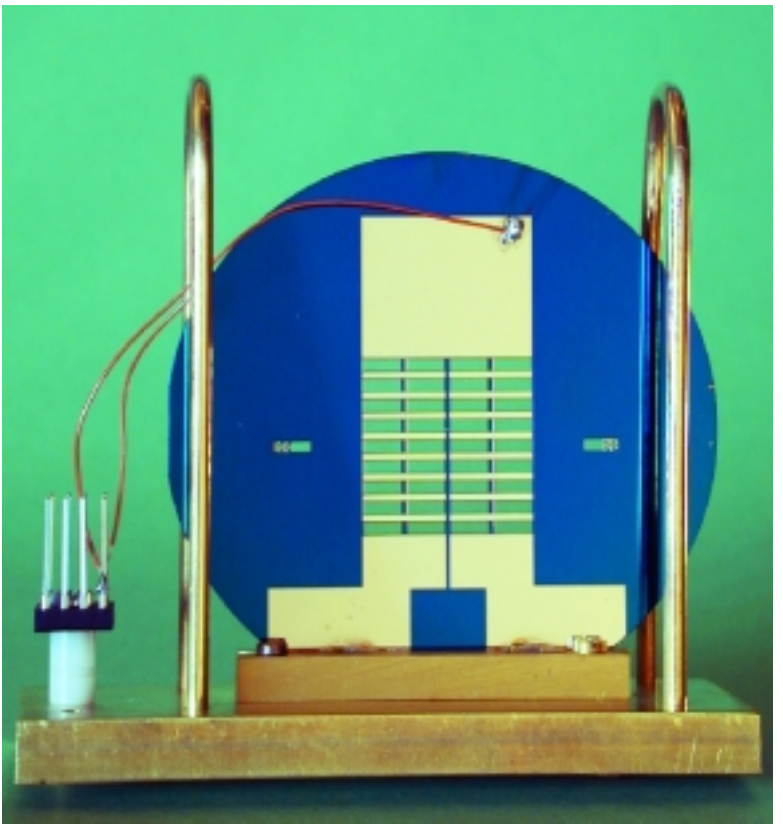

Fig. 136: A Megaflown is made by a hunderd through the wafer Microflowns (Doekle Yntema, Philip Ekkels).

Megaflown

A pressure gradient studio microphone has a selfnoise in the order of 20dB(A) and a bandwidth of 16kHz. What must be done to create a Microflown that has the same capabilities?

A Microflown of the year 2000 has at 40mW a sensitivity of 50‰. The operating temperature is 650K at this power dissipation. A packaging can be designed that has a gain of 20dB. In this case the sensitivity increases to 500‰. The Performance of this Microflown is (see Eq. (4.52)):

$$Performance = 20Log_{10}\left\{\frac{50\cdot 10^{-9}}{\sqrt{4\times 1.38\cdot 10^{-23}}}\times\sqrt{\frac{0.05}{650}}\times 0.5\right\} = 29.4dB$$

For a bandwidth of 20kHz and a diffusion corner frequency (f_{diff}) in the order of 1kHz and a heat capacity corner frequency ($f_{heat\ cap}$) of 8kHz the Noisefactor is 70dB, see Fig. 4.16. The “A” weighed selfnoise is thus (at least) 80-29.4=40.6dB.

A possibility to reduce the selfnoise is to increase the operation power without increasing the temperature. This can be achieved by increasing the effective length of the sensors by a factor of 100. Fig. 136 shows the first efforts to create such Megaflown. This particular realisation did not operate due to heating problems. Much effort is spent to solve these (and, of course) other problems so the expectation is that within the year 2001 a prototype will be ready.

The hundred Microflowns are made at once in a similar way as the through the wafer Microflown is made, see chapter 3, Fig. 3.10 and chapter B, Fig.77.

D5 STEREO RECORDING TECHNIQUES

In this chapter a very brief overview of the many ways of sound recording will be presented. The aim is to explain what type of microphone is used in what type of recording technique. Much information is from, and detailed information can be found in *"the new stereo soundbook"* written by R. Streicher & F. Alton Everest [1].

If microphones are put as close together as possible they create a coincident pair. In short, there are three ways to do this: the "*Blumlein configuration*" that uses two figure-of-eight microphones, the "*XY technique*" that uses two unidirectional microphones and at last there is the "*MS technique*" that uses a figure-of-eight and a microphone of choice.

Another way to record stereo sound is by the use of an artificial head that contains of two microphones are positioned inside the artificial ears. The last "way" of recording stereo sound is by using multiple microphones that are spaced. These ways of recording are not discussed here, more information can be found in [1].

Coincident-microphone stereo technique

If two microphones are placed as close together as possible the pickup is called a "coincident pair". In this way any differences in the time of arrival of sounds are kept to a minimum. The microphones are angled symmetrically on either side of the midpoint of the stereo soundstage, so that each microphone emphasises one side of the stereo image.

The Blumlein Configuration

The Blumlein configuration is named after Alan Dower Blumlein who was born in, England, 1903. He has become well-known in the audio community for the creative nature of his stereo contributions. The configuration consists on two figure-of-eight microphones that are angled 90 degrees to each other. The midpoint of the stereo soundstage should be at 45 degrees of both microphones that are in phase, 90 degrees in the polar plot of Fig. 120.

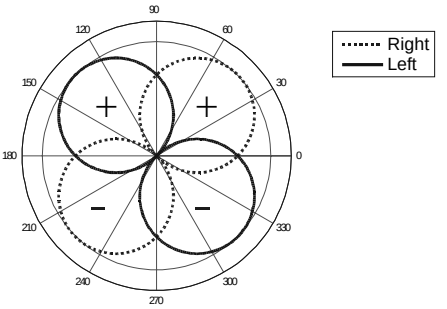

Fig. 137: A Blumlein configuration consisting of two figure-of-eight microphones rotated 90 degrees to each other. The sound source is positioned at 90 degrees.

Because the rear lobe of the left microphone, see Fig. 120, aims at the right side of the audience, and vice versa, the contribution of these rear lobes to the stereo perspective will be reversed from the actual left/right forward presentation. Although this is of little significance with ambience or reverberation, this crossing of channels may be of importance when part of the direct sound field (for example, antiphonal choirs or principal action in a radio play) is located to the rear of the stereo pair.

In a favourable performance space, the contribution of the rear lobes of the Blumlein pair can be appreciated, while in an unfavourable environment (one which produces too much reverberation or excessive noises) results will be less pleasing, and another technique for making the recording should be selected.

The Blumlein technique is not the most common microphone technique in use today because it is best suited for use when both the sound source and the performance spaces are favourable. In reality, however, such ideal conditions are not often present. Recording engineers and producers encounter the common problem of uninteresting or even unpleasant acoustics within modern performance spaces. Because the Blumlein technique faithfully reproduces these poor acoustics and blends them into the stereo image, other techniques must be employed to control their influence in the recording. Furthermore, the figure of eight microphones that are used nowadays are not sensitive at lower frequencies. So for this bandwidth other (omni) microphones have to be utilised. Thus, although the Blumlein technique often yields excellent results, it also is susceptible to many real-world problems that can affect its performance.

XY Stereo techniques

When the two microphones are cardioids or super cardioids, the response is primarily to the front half of the sound field, see Fig. 138. Because these do not have rear lobes, as do bi-directional microphones, the pickup of unwanted sound from the rear (ambience, reverberation, audience noise, etc.) is reduced.

As with the Blumlein technique of crossed bi-directional patterns, crossed cardioids provide a strong sense of lateral spread across the stereo soundstage. Because this technique also results solely from differences in intensity between the microphones, the two signals have phase integrity, in a highly accurate and stable image for the listener. Because the microphones pick up little sound from the rear, they can be located further from the sound source and still maintaining a good direct-to-reverberant ratio. This can be of particular advantage if the ensemble is quite large.

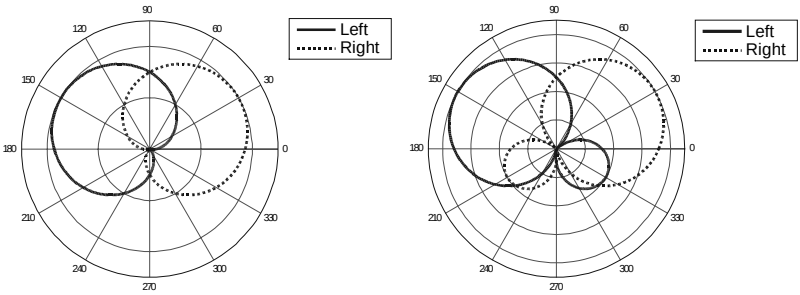

Fig. 138: Two directional microphones are arranged as a “XY-stereo” configuration. Left a XY arrangement using cardioids, right a XY arrangement using super-cardioids

Although other polar patterns could be used for XY pickups the most common configuration uses cardioid microphones, angled between 60 degrees and 135 degrees. The angle determines the ”width” of the stereo image, and is suggested by the actual width of the soundstage relative to the placement of the microphones. Because the polar pattern of a good cardioid microphone provides uniform frequency response to about 90’ off-axis, the resulting stereo image can encompass a broad span.

The mid/side technique

Among the many theories of stereo production proposed by Alan Blumlein perhaps his most innovative was a hybrid form of stereo microphone pickup. It was not until the 1950s, however, that this approach was realised in a practical way by the Danish State Radio [1].

The concept of this ingenious mid/side technique is simple. The Mid (or M) signal is the discrete monophonic pickup provided by a microphone of any kind, aimed with its most sensitive direction to the centre of the sound source. The Side (S) information is provided by a figure-of-eight microphone positioned so that the axis of minimum pickup is aimed to the centre of the sound source, see Fig. 139, Fig. 140 and Fig. 141.

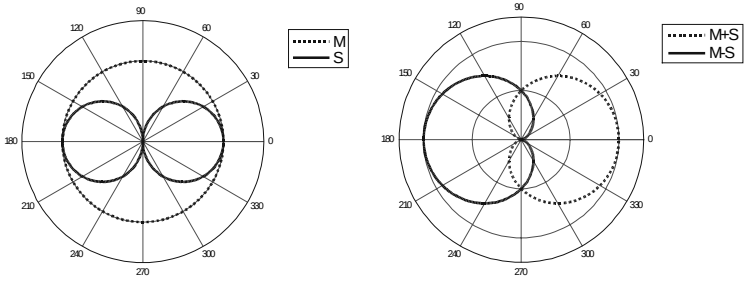

Fig. 139: A MS assembly consisting of a omni-directional microphone providing the M (mid) signal and a figure-of-eight microphone providing the S (side) signal. Right the result of the MS technique.

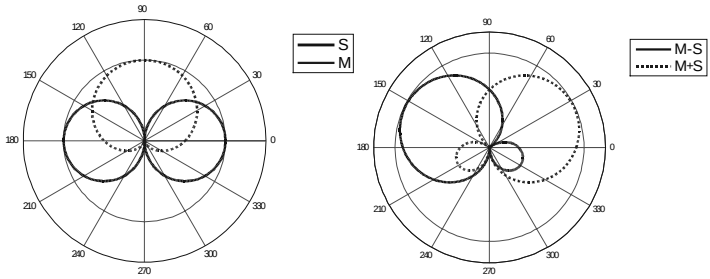

Fig. 140: A MS assembly consisting of a cardioid microphone providing the M (mid) signal and a figure-of-eight microphone providing the S (side) signal. Right the result of the MS technique.

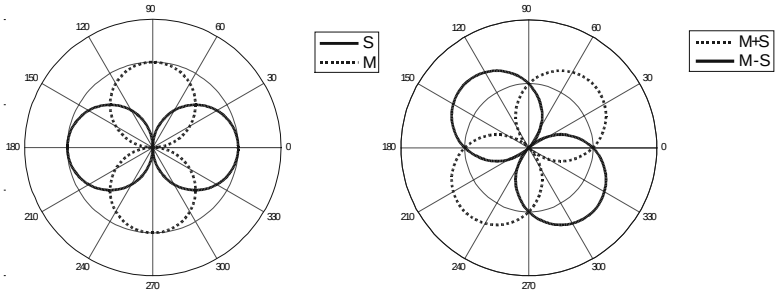

Fig. 141: A MS assembly consisting of a cardioid microphone providing the M (mid) signal and a figure-of-eight microphone providing the S (side) signal. Right the result of the MS technique: a Blumlein pair.

Because any polar pattern may be used for the Mid microphone, an variety of polar patterns can be created for the resulting stereo pickup. In all cases the Side microphone must be a figure-of-eight. In Fig. 139, Fig. 140 and Fig. 141 the signal ratio M:S equals 50:50. This ratio may be different resulting in an even larger variety of polar patterns.

While the Mid/Side technique achieves extreme versatility with the stereo image, it still preserves absolute monophonic integrity. This preservation occurs because when the two stereo signals are summed to obtain a mono signal only the Mid component remains. This is an important consideration for broadcasters [1].

Considerations of coincident stereophony

A major benefit of all coincident techniques is that the angular fidelity of the pickup closely matches the original source. Lateral position within the reproduced sound stage is independent of the distance of the sound source from the microphones, because the intensity differences between the two signals are determined primarily by the polar pattern of the microphones. Thus, a person talking at a position 45º off the centre line generates a reproduced sonic image at that angle, whether two feet or twenty feet from the microphones.

Much has been written concerning the sonic advantages and disadvantages of coincident microphone techniques. The most frequent criticism is that the sound they produce is very “dry” or “analytical” and lacks the sense of “spaciousness” which can be achieved by other techniques. Sometimes, coincident techniques also are chastised for having a too “narrow” image that is “confined” between the two loudspeakers.

For some people the coherent precision of the coincident microphone techniques, particularly MS, simply sounds too analytical indeed, “too correct”. Although the soundstage may be very accurately reproduced, this is simply insufficient to satisfy their sonic and/or emotional desires. They desire something more: “spaciousness”.

Another method to increase spaciousness is to introduce some incoherence into the stereo pickup by the creating some signal delay between the two channels. Achieved by either electronic means or physically, by introducing a slight spacing between the microphones, the result is claimed to “add a sense of space” to the recording. Whether this “space” is considered phase distortion or sonic enhancement lies fully in the ear of the listener.

In the early days of the development of stereophonic techniques, researchers experimented with two very different approaches. Alan Blumlein focused on coincident microphone techniques in England, while at the same time in the United States, spaced microphones were being used for experimental recordings of the Philadelphia Orchestra at Bell Laboratories.

Time-of-arrival (phase) effects play an important role in the preservation of directional cues for accurate stereo imaging, and the early research work done at Bell Laboratories has led to contemporary mixing engineers using spaced microphone techniques to excellent advantage.

By placing two microphones not on the same spot but relatively close together, a near-coincident configuration is established. Much of the phase integrity can be preserved; therefore, these configurations remain largely dependent on intensity differences for their stereophonic information. The spacing between the microphones, however, introduces phase differences that can be quite significant to the sound produced from the system.

When compared to the coincident pickups near-coincident techniques tend to provide an increased sense of ”space” around the performers; this is similar to adding ”air” or providing a more ”open” sound. This effect is due solely to the phase anomalies (i.e. comb filters) introduced by the time-of-arrival differences between the two microphones. This ”phasiness” is considered by some people to be pleasing and a favorable improvement over the more analytical sound produced by coincident techniques.

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