inter.noise 2000
The 29th International Congress and Exhibition on Noise Control Engineering
27-30 August 2000, Nice, FRANCE

I-INCE Classification: 1.2

ABOUT ATTENUATION OF THE ACOUSTIC POWER OF A 1- DIMENSIONAL MONOPOLE SOURCE ARRAY

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Keywords:
SOUND SOURCE, DESTRUCTIVE INTERFERENCE, SOUND POWER, ATTENUATION

ABSTRACT
The well known method of destructive interference can be used to attenuate the radiated sound power of monopole sources. In a technical application nozzles are arranged equally spaced in a straight line. Through each of these nozzles air is piped. The air stream is isochronous interrupted due to occluding the nozzles using a mechanical device. So on the output of the nozzles an alteration of volume flux dq/dt is expected. The sound power radiated is a function of these alterations. This behavior can be modeled using a 1-dimensional monopole source array. In consequence to attenuate the radiated sound power the method of destructive interference in the near field of the sources can be applied. In the case of small distances between the adjacent elementary sources compared to the wave length of sound, a comb function of sound power attenuation occurs.

1 - INTRODUCTION
Among others the effect of the destructive interference bases on the coherence of the so named primary and secondary source. In many cases the acoustic manipulation of a given sound source, which can be seen as primary source, is reached by introducing a secondary synthetically source consisting of a loudspeaker and an electronic device. In the present case the sound sources concerned are coherent. So the mentioned effect of destructive interference can be enabled by phase shift of the elementary sources.

2 - THE MODEL
In a technical application nozzles are arranged equally spaced by d in a straight line (Figure 1). Through each of these nozzles air is piped. The air stream is isochronous interrupted due to occluding the nozzles using a mechanical device (basic frequency f_b=500 Hz). So the expected alteration of volume flux dq/dt on the output of the nozzles can be assumed as an alternating quantity with stationary RMS value. Furthermore the cross-section of the nozzles is small compared to the considered wave length. Therefrom the volume flux can be seen as the strength q_n of an acoustic monopole source.
In the frequency range the occurrence of a discrete spectrum with the spectral components $f_i = f_b \cdot i \ (i \in N)$ is expected, due to the periodical process in time domain.
In a distance r_0 the point P is situated. The sound field of the point source array aforementioned can be described by the superposition of the sound fields of the elementary sources $m = 0, 1, 2, \ldots\ n$-1. Thus to predict the complex sound pressure on the point P the well known formula

$$\underline{p} = \sum_{m=0}^{n-1} \underline{p}_m = \frac{j\omega\rho}{4\pi}\left[\underline{q}_0 \frac{e^{-jkr_0}}{r_0} + \underline{q}_1 \frac{e^{-jk(r_0-\Delta r_1)}}{r_0-\Delta r_1} + \ldots \underline{q}_{n-1} \frac{e^{-jk(r_0-\Delta r_{n-1})}}{r_0-\Delta r_{n-1}}\right] \tag{1}$$

can be applied.
Additionally in the far field of the array that is to say for points P in the region $r_0 \gg (n-1)\,d$ the simplifications:

- $\gamma_m = \gamma_0$

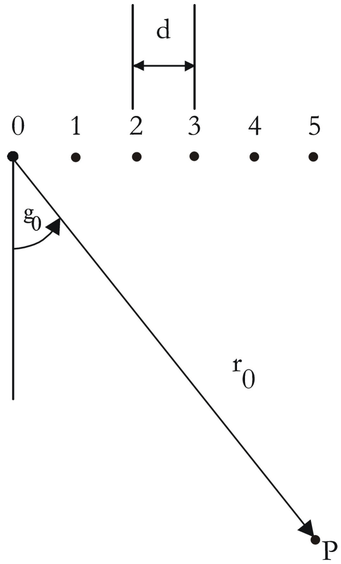

Figure 1: The geometry of the point source array.

- $\Delta r_m = 0$; for the description of the amplitude in (1): $q_m \cdot \dfrac{j\omega\rho}{4\pi\left(r_0 + \Delta r_m\right)}$

are brought in.
Now based on the fact of the coherence of the elementary sources and by use of a phase angle ωt_v which enables the enforcement of destructive interference the sound pressure at the distance r_0 becomes:

$$\underline{p} = \frac{jk\rho c}{4\pi r_0}\underline{q}_0 \sum_{m=0}^{n-1} e^{-jk(r_0 - md\sin\gamma + mct_v)} \tag{2}$$

The total sound power radiated can be found very easily by integrating the acoustic intensity I over the surface S of a large sphere ($r_0 \gg (n-1)\,d$) surrounding the sound source (premise: in the far field $\vec{I} \perp d\vec{S}$):

$$P = \int_\gamma \int_{\varphi=0}^{2\pi} I\left(r,\gamma,\varphi\right) r d\varphi \cos\gamma d\gamma = \int_\gamma I\left(r,\gamma\right) 2\pi r^2 \cos\gamma d\gamma \tag{3}$$

The sum in Eq. (2) can be treated as geometrical progression. Thereby the sound power is given by:

$$P = \frac{k\rho c}{8\pi}\left|\underline{q}_0\right|^2 \int_{-\frac{kd+\omega t_v}{2}}^{\frac{kd-\omega t_v}{2}} \frac{\sin^2\left(n\frac{kd\sin\gamma-\omega t_v}{2}\right)}{2} dx \tag{4}$$

3 - REDUCTION OF THE SOUND POWER RADIATED

The integrand in Eq. (4) is a periodic function with the period π. Strong main values of this function occur when

$$\left[\frac{kd\sin\gamma - \omega t_v}{2}\right] = i\pi,\ \ i \in Z \tag{5}$$

The upper and the lower limit of the integration interval in Eq. (4) depends on the parameters d, c, ω and t_v. However the range of the interval is always $\omega d/c$.
According to the aforementioned a reduction of the sound power occurs when the integration range lies outside of the intervals

$$\left[\left\{-m\pi - \frac{\pi}{2n}\right\}\ ;\ \left\{-m\pi + \frac{\pi}{2n}\right\}\right]\ \ m \in Z \tag{6}$$

The shift of the integration range with respect to premise (6) can be enabled by the delay time t_v. But the reached change of the phase angel $\omega d/c$ is frequency-dependent. Consequently for the effective reduction of the total power radiated an optimization over the whole frequency range $f_i = f_b \cdot i \ \ (i \in N)$ should be carried out.

4 - AN EXAMPLE

Figure 2 shows the reduction of radiated sound power by use of a time delay $t_v = 1.9$ ms. The distance d between the neighboring sources is 5 mm.

The function of the reduction of sound power is a comb function. As shown in Fig. 2 the width of the main maxima rises by increasing the frequency. But the spectral components of the elementary sources $f_i = f_b \cdot i \ \ (i \in N)$ are equally spaced. Accordingly for each quantity of t_v spectral components which are not reduced in sound power can be found.

In the most cases however the dominant part of the energy lies in the lower frequency range of the exciting spectrum. That means by a known upper cut-off frequency of the dominant part of the energy a value of the time delay t_v should be chosen which enables the reduction of all spectral components below the cut-off frequency.

The shown "comb"-function (Fig. 2) reduces all the spectral components below 8 kHz. Furthermore in the present case the significant part of energy of the exciting spectrum lies in the frequency range beneath 8 kHz. Thus the total sound power radiated is strong decreased.

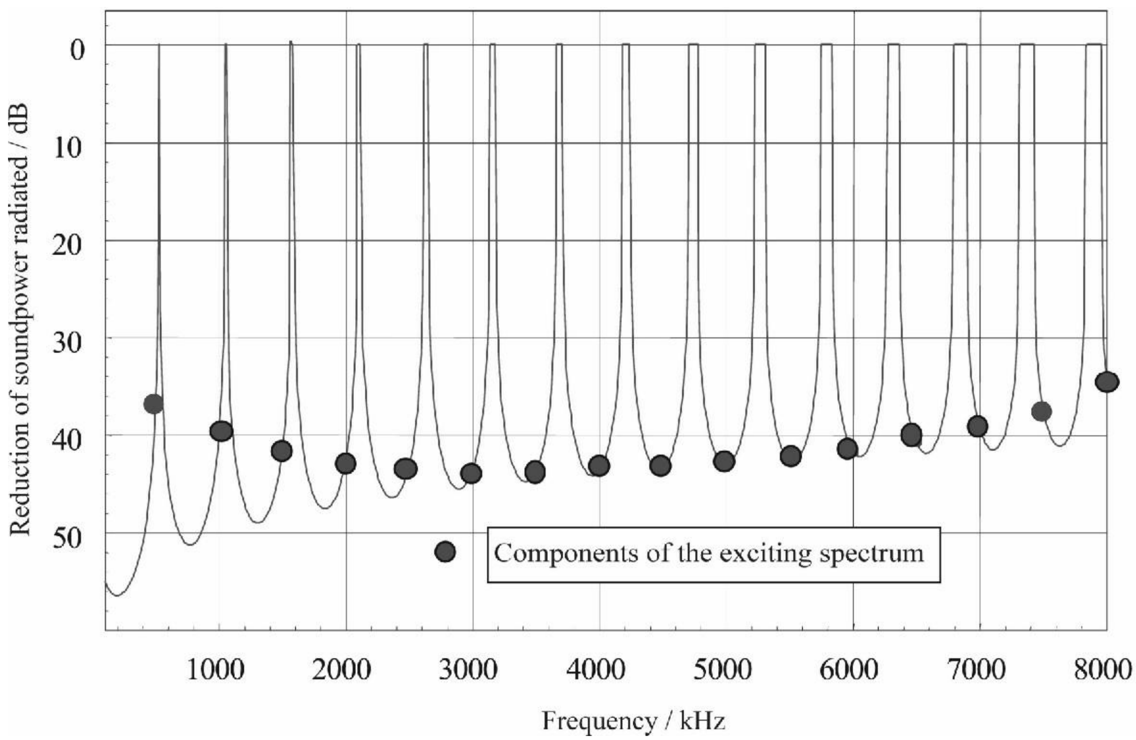

Figure 2: The reduction of sound power radiated by a point source array with linear phase shift related to the radiated sound power of the array without an phase shift. The time delay of the neighboring elementary sources is 1.9 ms.

REFERENCES

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