SOURCE LOCATION BY USE OF A REAL-TIME
THIRD OCTAVE INTENSITY ANALYSER

by

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The most commonly measured quantity in acoustics is undoubtedly sound pressure. The measurement is performed directly with a microphone and the signal can be analysed and displayed in octave or third octave bands. This system is well-known and widely used.

Another fundamental quantity of great interest to acousticians is sound intensity, which describes both the amount and the direction of flow of acoustic energy at a given position.

The introduction of real time analysis using digital filtering techniques for the processing of signals from two closely spaced microphones has given a leap forward in the precision with which acoustic intensity measurements may be carried out. The use of digital filters has increased the possible frequency range and practically eliminated the inherent lack of real time capability associated with FFT-analysis of the continuing but time varying signals most often encountered in acoustics.

A 1/3 octave digital filter system is shown in Fig. 1.

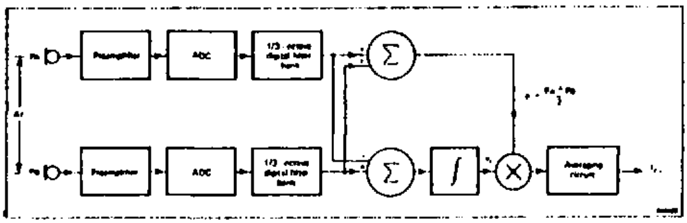

Fig. 1 Real-time digital intensity analyzer.

The signals from the two microphones are frequency analysed in parallel in real time and the sum and the difference of the two signals for each frequency band are calculated. The difference is integrated over time and divided by the distance between the microphones of the probe to yield the particle velocity V_r. The sum, and thus the pressure corresponding to a point midway between the two microphones, is then multiplied by the particle velocity to yield the sound intensity vector component in a direction given by the orientation of the probe.

The use of the two microphone technique to measure sound intensity does introduce limits to the useful frequency range of the measuring system. A principal systematic error is inherent in the approximation of the pressure gradient by a finite pressure difference. It is this approximation which sets the upper frequency limit for practical systems. The approximation error can be calculated for ideal sources, e.g. monopole, dipole and quadrupole. For a monopole source the error in dB, denoted by L_ϵ between the approximated sound intensity I_{AP} and the exact intensity I_{EX}, is given by

$$L_\epsilon = 10 \log_{10} \left[\frac{I_{AP}}{I_{EX}}\right] = 10 \log_{10} \left[\frac{\sin (k \Delta r)}{k \Delta r} \; \frac{r^2}{r_1 r_2}\right]$$

where k is the wave number and Δr is the distance between the acoustic centres of the microphones. It is seen that the approximation error is not only a function of $k \cdot \Delta r$ but also of $\Delta r/r$ (see Fig. 2). When $\Delta r \ll r$ as shown in Fig. 3 the approximation error takes the form of under-estimation of the sound intensity and this error becomes greater with increasing frequency.

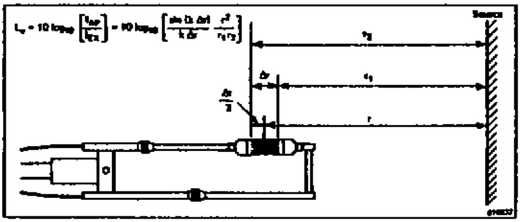

Fig. 2 Geometrical configuration.

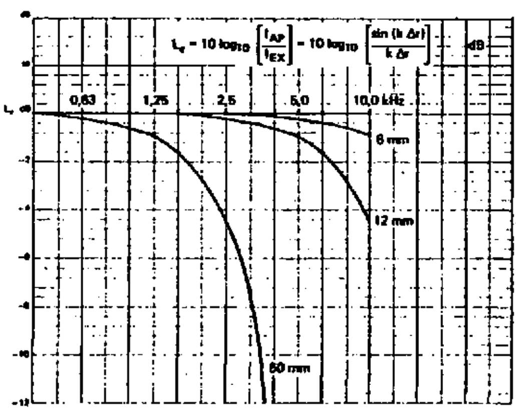

Fig. 3 High frequency limitations.

If transducers were perfect and signal processing ideal, the approximation error could be reduced by making the microphone spacing as small as possible. However, intensity measurements based on the two microphone technique are highly sensitive to differences between the phase responses of the two microphone channels. Phase mismatch has the greatest influence for small values of microphone spacing and for low frequencies. This sets the lower frequency limit for the system. The approximation error due to a phase mismatch of $0{,}5^{\circ}$, which is typical for a B&K microphone probe, is shown in Fig. 4 as a function of frequency.

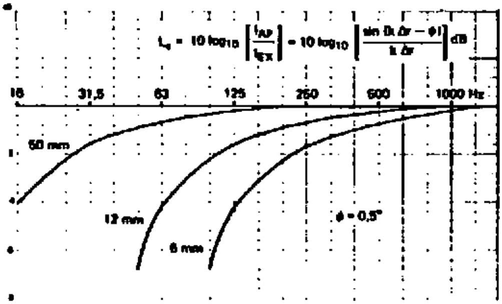

Fig. 4 Low frequency limitations for a monopole source.

Thus the microphones used in the probe system should be selected by matching their phase responses to minimize this error.

If the orientation of the probe is changed by 180° then it is seen from the equation for L_{ε} that the intensity will be overestimated instead of underestimated. An average of the two intensity measurements made at low frequencies with a 180° change in probe orientation for the second measurement will considerably reduce the approximation error.

The physical configuration used in placing two microphones close to each other is also very important. To obtain good performance above 2 kHz requires careful consideration of the geometry. The variation in the effective acoustical separation Δr as a function of frequency has been investigated for different configurations. A comparison of variation in Δr is shown in Fig. 5 for a side-by-side configuration and a face-to-face (slit grid) configuration.

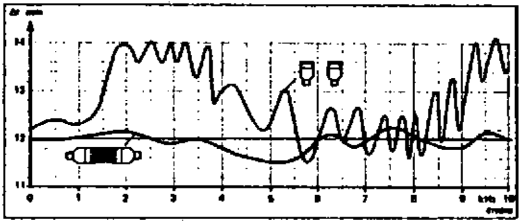

Fig. 5 Two ½" microphones used in different configurations.

To optimize both the free field amplitude response and the phase response, a new probe has been designed using B&K 2633 type preamplifiers. Also the need for a fast calibration method, easy change of microphone spacing (3 different lengths of spacing are supplied), as well as a choice between ½" and 1/4" microphones has been taken into account in the construction. The probe configuration is shown in Fig. 6.

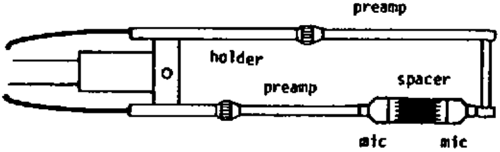

Fig. 6 A sound intensity probe.

Both when using the system for sound power determination and for acoustic source and sink location it is important to notice the directional characteristics of the probe (see Fig. 7).

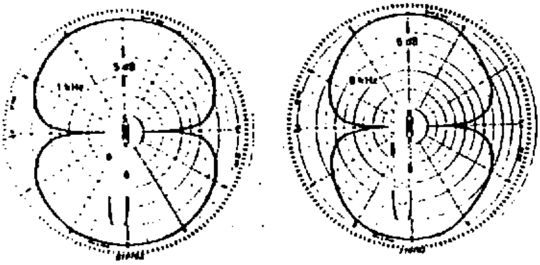

Fig. 7 Directional characteristics for ½" microphone probe.

For sound power determination it is seen that although the probe may be directed up to 60^0 off axis to the direction of maximum sound intensity, the measured sound intensity decreases by only 3 dB from the maximum level. However, for source location extremely good directional sensitivity is given by searching for intensity minima (detected by change in the brightness of the channels).

In addition it should also be noticed for noise source location that phase mismatch (at low frequencies) introduces an angle error in location of the direction to the source. As shown in Figs. 8 and 9, this error is minimized considerably by phase matching of the system and by choice of a large spacer.

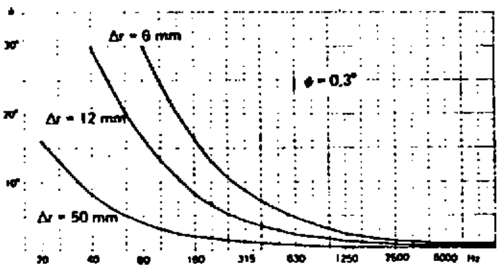

Fig. 8 Angle error as a function of spacing and frequency.

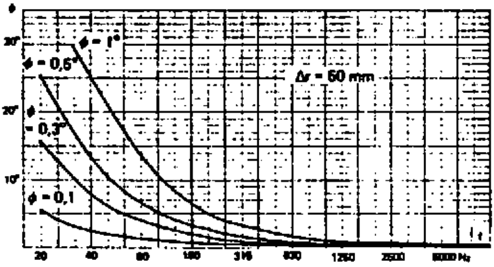

Fig. 9 Angle error as a function of phase matching and frequency.

For the electronic part of the system matched components are used in the following four blocks: Preamplifier - Input Attenuator - Anti-aliasing Filter and Sample and Hold. This means that the only thing which must be done before each new measurement is to calibrate the amplitude of the two channels.

The digital filter system has the great advantage that there is no phase mismatch between the two channels. A digital filter has a phase curve similar to the corresponding analog filter, so in a single digital filter the phase function can be very badly behaved. In our system this is no problem as long as the function in both filter channels is exactly the same. This would be the case even if two physically separate digital filters were used, but in fact here it is absolutely sure because it is the same filter unit being time shared between the two channels.

Looking at the integrator circuit we are faced with a special problem. The integrator can be considered as a special digital filter, but here we are in a situation where we have no counterpart to phase match against. Unfortunately there is no ideal digital integrator so we have to make a compromise. Due to the importance of the phase variation we have chosen the very simple digital filter given by the equation:

$$H(z) = \frac{1+z^{-1}}{1-z^{-1}}$$

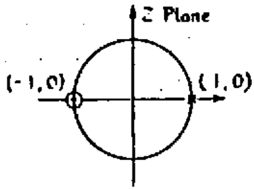

Fig. 10 Digital integrator. Zero and pole configuration.

As can be seen from Fig. 10, the filter has a pole in (1,0) and a zero in (-1,0). A calculation of the amplitude and phase gives:

$$|H(\omega)| = \left|\cot \frac{\omega T}{2}\right|$$

$$\angle\emptyset = -\frac{\pi}{2}$$

so we have exactly the required phase curve and an amplitude curve which is very easy to correct.

The only unpleasant feature is that if an error signal in one way or another has arisen in the integrator, it will stay there forever. So the integrator is always automatically cleared before a measurement is started.

The number of applications of the sound intensity analysing system is legion. The system not only performs many standard measurements easily and efficiently, but also opens up new measurement possibilities.

Consider, for example, the investigation of diesel motors. The standard method involves the laborious technique of wrapping various parts of the motor with lead sheeting in order to locate and identify the various sound sources. The total investigation could easily last weeks. With the sound intensity analysing system the principal sources can be located and ranked in order of importance in a matter of minutes.

As already mentioned, source location is very easily performed with this system because of the directional characteristics of the sound intensity probe, where the measured sound intensity is a function of the angle. The minima of the measured sound intensity are especially sharp and well-defined, which makes it very easy to locate acoustic sources. Another example taken from the motor industry is the control of noise inside the passenger compartments of vehicles. By simply tracing the sound intensity flow lines with the probe the sources and sinks of acoustic energy can be found: that is, the places where noise enters and leaves the compartment. Maps of these sources and sinks are extremely helpful when tackling noise reduction problems.

The Sound Intensity Analysing System is also eminently suitable for sound power determination, as sound power is originally defined in terms of sound intensity.

$$\iint_S \vec{I} \cdot \vec{dS} = P_a$$

What we have to do is to make a linear integration of the sound intensity over a surface enclosing the source, which is quite simple because of the digital approach of the system. The measurement can be done on site, that means, no special room (e.g. semi-anechoic chamber) is required, even if the environments are noisy or the machine is coupled to other machines. The method still works because the contribution from all sources not situated within the enclosing surface will automatically be integrated to zero according to Gauss' theorem. A practical measurement will only take a few minutes.

To summarize, the B&K Sound Intensity Analyzing system type 3360 opens new horizons for acoustical measurements. The instrument operates both in sound pressure (from 1,6 Hz to 20 kHz) and sound intensity mode (from 3,2 Hz to 10 kHz) in real time. But in many applications there is a distinct advantage in measuring the vector quantity sound intensity rather than the scalar quantity sound pressure.