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REAL-TIME SOUND INTENSITY MEASUREMENTS

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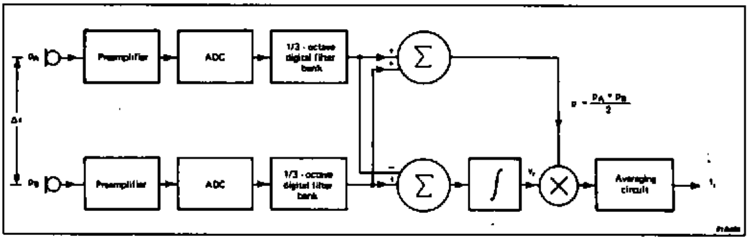

Fig.1. Real-time digital intensity analyser

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.

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 I_r in a direction given by the orientation of the probe.

$$I_r = \lim_{T\to\infty} \frac{1}{T} \int_{-\frac{T}{2}}^{+\frac{T}{2}} p(t) \cdot v_r(t)\, dt$$

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 sources. For a monopole source the error in dB, denoted by L_r between the approximated sound intensity and the exact intensity, is given by

$$L_r = 10 \log_{10} \left[\frac{\sin(k \cdot \Delta r)}{(k \cdot \Delta r)} \cdot \left(1 - \frac{1}{4}\left(\frac{\Delta r}{r}\right)^2\right)^{-1} \right]$$

where k is the wave number and Δr is the distance between the acoustic centres of the microphones and r, the distance between the geometrical centre of the microphone probe and the source. It is seen that the approximation error is not only a function of $k \cdot \Delta r$ but also of $\Delta r/r$. When $\Delta r << r$ as shown in Fig.2 the approximation error takes the form of underestimation of the sound intensity and this error becomes greater with increasing frequency.

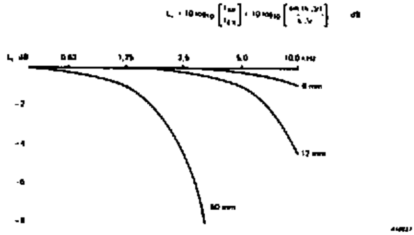

Fig.2. High frequency limitations

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 matching of 0.5°, which is typical for a B&K intensity system, is shown in Fig.3 as a function of frequency.

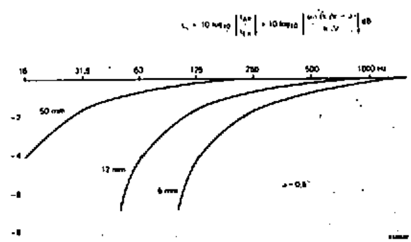

Fig.3. Low frequency limitations.

Thus the microphones used in the probe system should be selected by matching their phase responses to minimize this error. Typical phasematching of B&K microphone probe is shown in Fig.4.

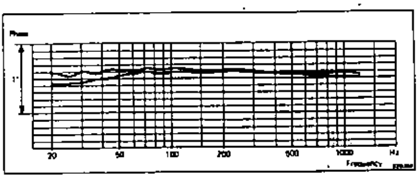

Fig.4. Typical phasematching of B&K microphone probe

If the orientation of the probe is changed by 180° then it is seen from the equation for L_r 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.

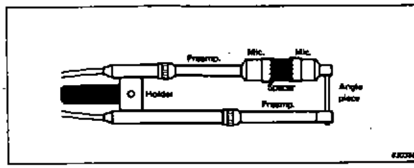

Fig.5. A sound intensity probe

To optimize both the free field amplitude response and as previous mentioned 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 1/2" and 1/4" microphones has been taken into account in the construction. The probe configuration is shown in Fig.5 and the frequency response in Fig.6.

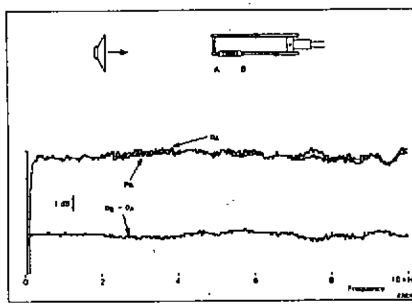

Fig.6. Amplitude response of the probe

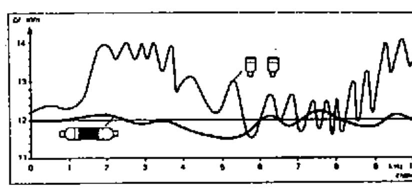

Fig.7. Effective separation as a function of frequency

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.7 for a side-by-side configuration and a face-to-face (slit grid) configuration.

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.8), which is a cosine function.

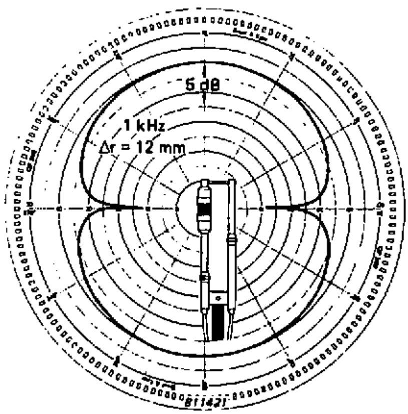

Fig.8. Directional characteristics of the probe

For sound power determination it is seen that although the probe may be directed up to 60° off axis to the direction of maximum sound intensity, the measured sound intensity level 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).

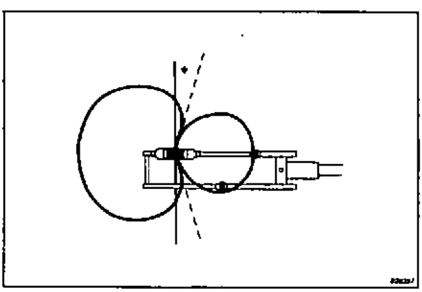

Fig.9. Unsymmetrical directional characteristics due to lack of phasematching

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 Fig.9. This error is minimized considerably by phase matching of the system and by choice of a large spacer.

For the electronic part of the system matched components are used in the following four blocks: Preamplifier – Input Attenuator – Antialiasing Filter and Sample and Hold. This means that the only thing which is needed before each new measurement is to calibrate the amplitude of the two channels by means of a pistonphone.

The digital filter system has the great advantage that there is no phase mismatch between the two channels. In fact it is the same filter unit being timeshared between the two channels. The octave Chebyshev Filters fulfil the ANSI S1.11 class II filter standards, the 1/3 octave Chebyshev filters the ANSI S1.11 class III standards, which are the most strict filter standards as shown in Fig.10. In practice it is impossible to distinguish between analogue and digital filters except that the stability of the digital filters is far more superior.

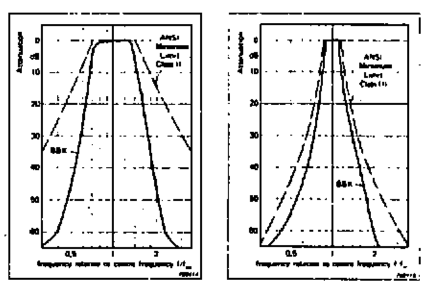

Fig.10. Filter characteristics of the analyser

Unfortunately an ideal digital integrator does not exist so in the design of the integrator circuit a compromise has to be made. Due to the importance of the phase response a very simple digital filter given by the equation:

$$Y_n = X_n + X_{n-1} + Y_{n-1}$$

has been chosen.

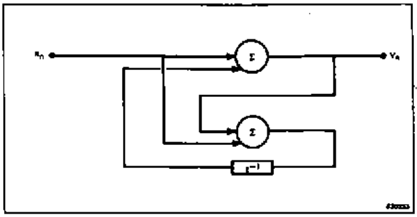

Fig. 11. The digital integrator

A calculation of the amplitude and phase gives:

$$|H(\omega)| = |\cot \omega/2f_s|$$

$$< H(\omega) = -90°$$

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

The only consequence of an ideal integrator 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 are 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 an 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.

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.

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