TECHNICAL UNIVERSITY OF LODZ

INTERNATIONAL FACULTY OF ENGINEERING

MEASUREMENTS

LABORATORY EXERCISE # 4

SIGNAL FILTRATION AND DIAGNOSTICS OF A MACHINE

1. Aim of the exercise

The aim of the exercise is to be acquainted with the filtration of the measured signal and the frequency analysis to break down a complex signal into its components at various frequencies. Using the spectrum analysis the diagnosis of a real machine should be performed.

2. Frequency analysis

2.1. Fourier analysis

The mathematical basis of frequency analysis is the Fourier Transform, which takes different forms depending on the type of signal analyzed. All have in common that the signal is assumed to be composed of a number (perhaps an infinitive number) of (co-)sinusoidal components at various frequencies, each having a given amplitude and initial phase.

$$X(t) = A\cos\theta = \frac{A}{2}(e^{j\theta} + e^{-j\theta}) \tag{2.1.1}$$

where $\theta = (2\pi ft + \phi)$; $2\pi f$ is a constant angular frequency (in radians/s) and φ is the "initial" phase angle at time zero.

The transformation (known as Fourier Transform) allows to present the measured signal (time depending) in the domain of frequency (mathematical understanding of the Fourier Transform nor determination of the transform constants are not the aim of the exercise).

2.2. Signal types

The type of the signal to be analyzed has an influence on the type of analysis to be carried out and also on the choice of analysis parameters. The basic divisions into different signal types are presented below in Figure 2.2.1.

For practical purposes it is sufficient to interpret stationary (time steady) functions as being those whose average properties do not vary with time and are thus independent of the particular sample measurement used to determine them. This applies to both deterministic and random signals, but in particular in the latter case it is important to realize that the results obtained from different measurements are not necessarily identical, just equally valid.

The instantaneous value of stationary deterministic signal is predictable at all points in time, while with stationary random signals it use only the statistical properties such as mean values, variances etc., which are known.

Stationary deterministic signals are made up entirely of sinusoidal components at discrete frequencies. In periodic signals all these discrete frequencies are multiplies of some fundamental frequency, the reciprocal of the periodic time. In quasi-periodic signals, the frequencies of the various sinusoids are not harmonically related. The examples of such signals are presented in Fig. 2.2.2. The approach to frequency analyzing them is basically the same. The filter bandwidth should be selected so as to separate the most closely spaced components and in that case there will only be one sinusoid component in the filter pass-band at one time. It might be argued that the closest spacing will only be known after analysis, and in some cases it may be necessary to use assort of trial-and-error process, but in many cases the likely location of frequency components will be known in advance, e.g. as harmonics of a machine rotational speed, or mains frequency.

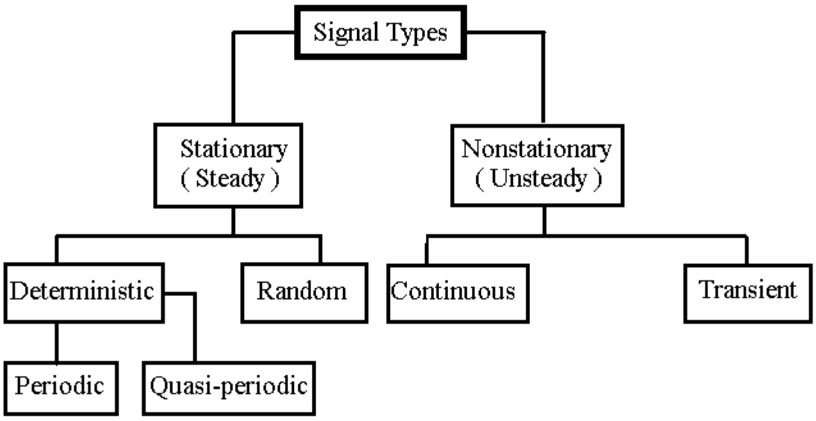

Fig.2.2.1. Division into different signal types

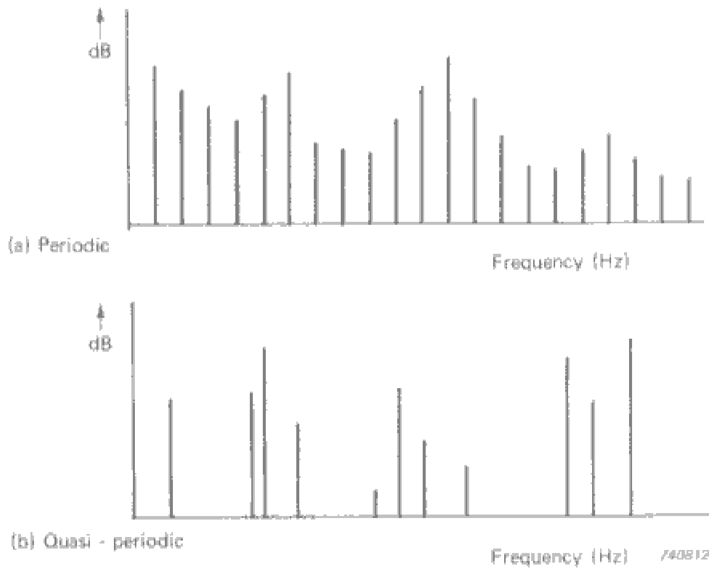

Fig. 2.2.2. Typical periodic and quasi-periodic spectra (B&K)

Normally, a constant bandwidth analysis on a linear frequency scale will be most appropriate to the analysis of deterministic signals, since harmonically related components will then be equally separated and resolved.

In contrast to deterministic signals, random signals have a spectrum, which is continuously distributed with frequency, as show in Fig. 2.2.3.

Non-stationary signals may be roughly divided into continuous non-stationary signals (of which a good example is speech) and transient signals which may be defined as those which start and finish at zero. Transient signal is treated and analyzed as a whole, whereas a continuous non-stationary signal will normally be analyzed in short sections, each of which will often be quasi-stationary. For example, a continuous train of speech can be divided up into short individual sounds: vowels, consonants etc. The process of dividing up such a continuous signal into short sections is called " time windowing " because the total signal can be considered to be viewed through a window, which only transmits the portion of interest.

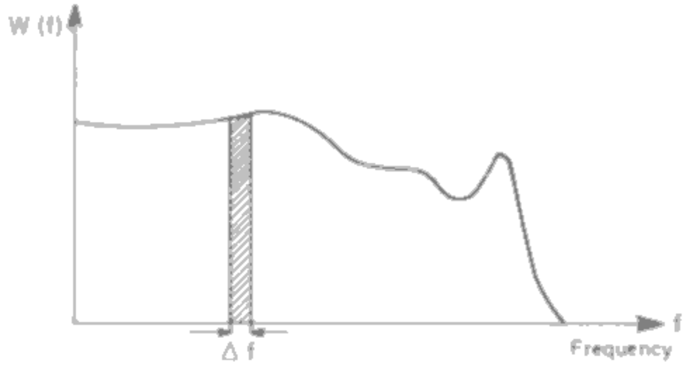

Fig. 2.2.3. Continuous spectrum of a stationary random signal (B&K)

2.3. Filters

It is well known that the measuring signal can consist not only of useful quality, but also of disturbances and noises. The process of filtration is one of the ways of eliminating the disturbances and noises from the measured signal.

Process of filtration allows also transmission or attenuation of the signals in certain range of frequencies (band-pass filter or in the second case pass-band filter). Set of pass-band filters allows dividing the measuring signal into given subintervals (frequency bands). A number of those subintervals and their widths are dependent on used set of filters.

The classical method of obtaining the signal frequency spectrum is to pass the measured signal through a number of analog filters with different center frequencies (or one filter whose center frequency is moved over a frequency range) and measure the transmitted power at reach frequency.

A basic choice to be made is between constant absolute bandwidth and constant relative (percentage) bandwidth where the absolute bandwidth is a fixed percentage of the tuned center frequency.

Constant bandwidth gives uniform resolution on a linear frequency scale and this, for example, gives equal resolution and separation of harmonically related components and this will facilitate detection of a harmonic pattern. However, the linear frequency scale automatically gives a restriction of the useful frequency range to (at the most) two decades.

Constant percentage bandwidth, on the other hand, gives uniform resolution on a logarithmic scale and thus can be used over a wide frequency range of 3 or more decades. It is thus both natural and efficient to analyze spectra dominated by structural resonances on a logarithmic frequency scale with a constant percentage bandwidth.

It is worth paying attention particular to two special classes of constant percentage bandwidth filters, octave and third octave filters, since they are widely used, in particular for acoustic measurements. The former have a bandwidth such that the upper limiting frequency of the pass-band is always twice the lower limiting frequency, resulting in a bandwidth of 70.7%. This can be derived as follows:

If: f_l = lower limiting frequency
f_u = upper limiting frequency
f_o = nominal center frequency

Then $$f_u = 2f_l \qquad (2.3.1)$$

and f_o = the geometric mean = $\sqrt{f_u \cdot f_l} = \sqrt{2f_l^2} = \sqrt{2}f_l$. (2.3.2)

The absolute bandwidth is equal to $B = f_u - f_l = f_l$ (2.3.3)

and the relative bandwidth: $\partial = \frac{f_u - f_l}{f_o} = \frac{f_l}{f_o} = \frac{f_l}{\sqrt{2}f_l} = \frac{1}{\sqrt{2}} = 70.7\%$ (2.3.4)

Internationally standardized center frequencies are laid down in International Electrotechnical Commission (IEC) Recommendation 225 that specifies a set of contiguous filters based on reference center frequency of 1000 Hz.

Thus it can be seen that it is possible to cover 3 decades in frequency with 10 octave bands ranging from 22.5 Hz (lower limiting frequency for 31.5 Hz center frequency) to 22,5 kHz (upper limiting frequency for 16 kHz center frequency).

Third octave filters are obtained by dividing each octave band into three geometrically equal sub-sections, i.e. $f_u = 2^{1/3} f_l$ and by coincidence this is equal to one-tenth of a decade since

$$\log_{10}(2^{1/3}) = 1/3 \log_{10}(2) = 1/3 \cdot 0.3 = 0.1 = 1/10 \log_{10}(10) = \log_{10}(10^{1/10})$$

By the same procedure as for octave filters, the percentage bandwidth of third octave filters can be derived as:

$$\frac{2^{1/3} - 1}{2^{1/6}} = 23.1\%$$

Practical filters deviate from ideal filters in several ways (Fig. 2.3.1.).

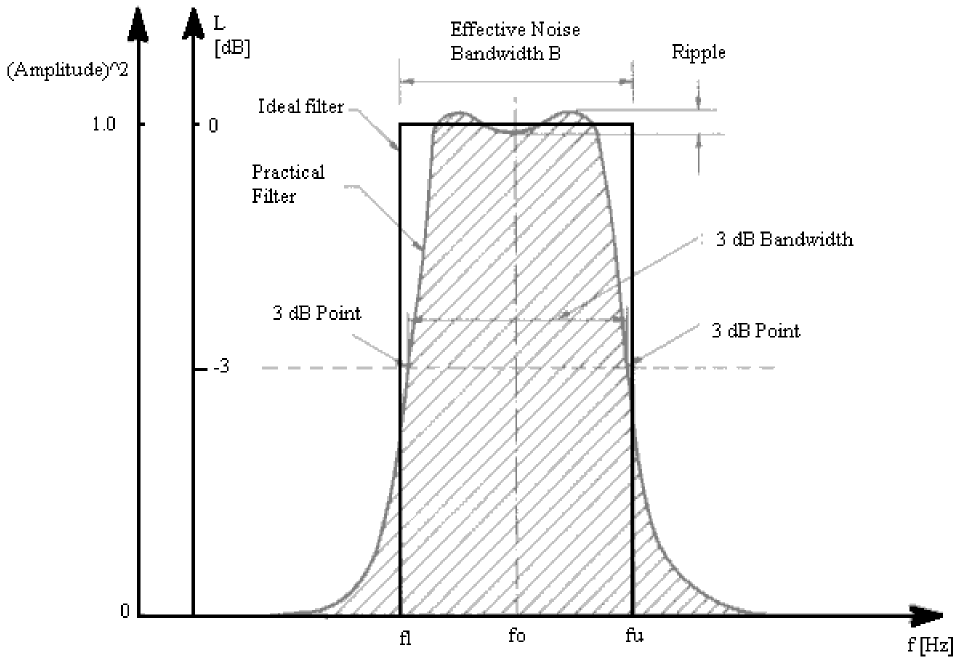

Fig. 2.3.1. Practical vs. ideal filter

Normalized middle frequencies of presented filters (each octave consists of 3 third octaves) are gathered together in Table 2.3.1.

Table 2.3.1. Middle frequencies of octave and third octave filters

Octaves	Hz	16	31.5	63	125	250	500	1000	2000	4000	8000	16000
		12.5	25	50	100	200	400	800	1600	3150	6300	12500
Third		16	31.5	63	125	250	500	1000	2000	4000	8000	16000
octaves	Hz	20	40	80	160	315	630	1250	2500	5000	10000	20000

3. Diagnosis of a real machine

Due to existing clearances and not balanced rotating elements, inaccuracies during production process and mutual interactions of cooperating elements, in real working machines there occur mechanical vibrations. Each one of these reasons generates vibrations of specified frequency, i.e. not balanced or bended shaft rotor may generate 1 impulse per one its turn. Frequency of generated vibrations is equal to frequency equivalent to shaft rotational speed. Damaged bearing system may generate vibrations of frequency equal to a product of rotational frequency and number of balls in bearing. Mechanical vibrations generated by particular parts are summed, strengthened or weakened; effect of their activity is delayed and deformed. These vibrations are revealed as acoustic signals, which propagate in environment, and we receive them as a noise caused by the machine. As the exploitation proceeds, vibrations of machine increase due to the getting worn, what more, participation of particular “vibration sources” in total vibrations also changes.

While making measurements of mechanical vibrations either acoustic ones, we transform them both in domain of time and frequency. More expressive is the result in frequency domain because it is easier to find the main reasons of vibrations. Knowing the construction of given machine we are able to identify the individual parts generating the maximum vibrations and to start repairing machine these parts. Example of spectrum analysis for helicopter vibrations with visible resonant effects is shown on Fig. 3.1.

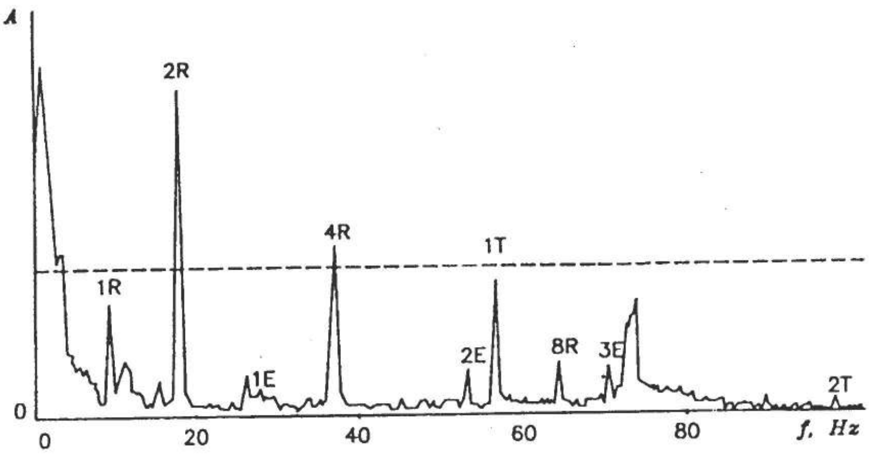

Fig. 3.1. Harmonic distribution of amplitudes of helicopter vibrations:
1R, 2R, 4R, 8R vibrations from the main rotating shaft;
1E, 2E, 3E vibrations of the second order of the lever;
1T, 2T – vibrations due to the not balanced tail rotor.

For diagnostics different types of analyzers of frequency spectrums are used. Lately it is more common to make spectrum analysis by measuring signals using PCs and sophisticated software. In this case the signal measured by microphone is transmitted to digital-analog converter and saved in PC memory, next analyzed by appropriate calculating code.

The classical method of obtaining the frequency spectrum of a signal is to pass it through a number of analog filters with different center frequencies (or one filter whose center frequency is moved over a frequency range, like in this experiment) and measure the transmitted power at each frequency.

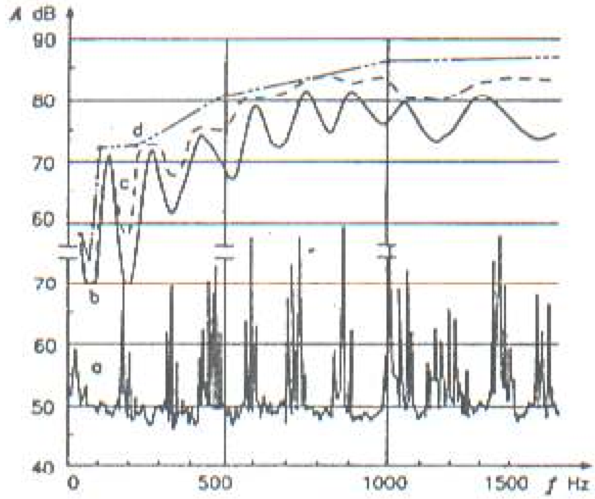

Fig.3.2. Identical acoustic signal recorded by means of different type analyzers: a – narrow-band filter with bandwidth 2Hz, b – narrow-band filter with bandwidth 4%, c – third octave filter , d – octave filter

On Figure 3.2 the spectra obtained by means of different type analyzers: narrow-band filter with bandwidth 2Hz, narrow-band filter with bandwidth 4%, third octave filter and octave filter of the same acoustic signal are shown.

Some devices have also possibility to measure values by means of linear filter with bandwidth significantly exceeding the range that we are interested in, i.e. in acoustic for audibility band ranging from 20 Hz up to 20 kHz usually the linear filter transmits signals for band from 2 Hz to 40 kHz. Linear filters make it possible to measure the average value of sound intensity in the whole range of frequencies of the analyzed spectrum.

4. Description of the test rig

The stand to be used in the experiment (Fig. 4.1) allows to carry out the spectral analysis of acoustic noise emitted by examined objects and to test the analyzer by means of test signals emitted by the decade generator.

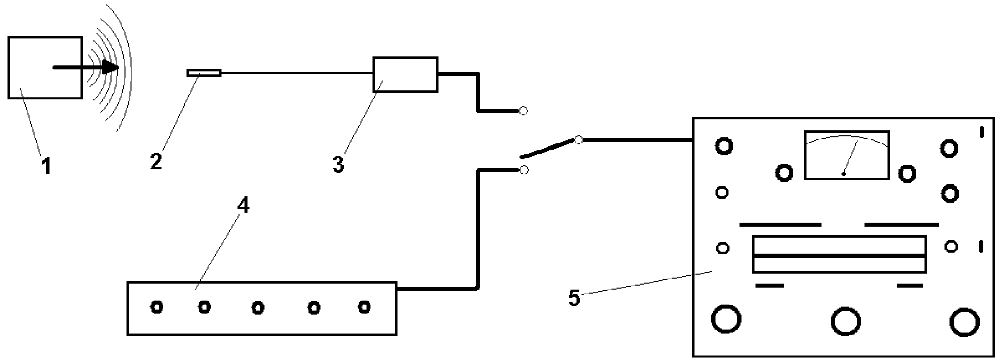

Fig. 4.1 Block scheme of the test rig: 1 – object under testing; 2 – microphone; 3 - microphone preamplifier; 4 – decade generator; 5 – frequency analyzer Type 2120 (B&K)

The microphone (2) with associated preamplifier (3) processes acoustic signals into electrical signals with adequate amplitudes. The noise analyzer (5) transforms the signals from time domain into frequency domain. The Frequency Analyzer Type 2120 (B&K) is a constant percentage bandwidth analyzer for use in the frequency range from 2 Hz to 20 kHz in conjunction with the internal filters. Four basic configurations are available: 1 – constant relative bandwidth filter (1 %, 3 %, 10 % and 1/3 Octave); 2 – tunable band stop; 3 – tunable high pass filter; 4 – tunable low pass filter. Internationally standardized weighting filters A, B, C and the D for sound measurements are included. The decade signal generator (4) allows to generate sine signals in the frequency range from 1 Hz to 20 kHz with 1 Hz resolution.

5. Description of the exercise

The experiment consist of two parts:

1) Investigation of the characteristics of particular band pass, linear and weighting filters built in the analyzer.
2) Spectrum analysis of sound emitted by machine under test.

5.1 Investigation of filters characteristic

Course of investigation is as follows:

1. After consultation with an instructor one has to set on one type of examined filter (first 1/3 octave filter, next 10 % filter) and its nominal centre frequency should be set up. Next the calculations of the theoretical boundary frequencies and the relative bandwidth of the investigated filters should be performed:

   Ex. third octave filter: $f_l = \frac{1}{\sqrt[6]{2}} f_o;$ $\quad f_u = \sqrt[6]{2} f_o;$

   Relative width of band: $\delta = \frac{f_u - f_l}{f_o}$ is constant and equals 0.232 for 1/3 octave filters and 0.10 for 10 % filters.

   The results of calculations should be written in Table 3.1.2. (theoretical side).
2. The standard signal from the decade generator should be provided to the direct input of the analyzer.
3. After consultation with an instructor the amplitude (A_{in}) of the input signal should be set up. Changing input signal frequency (f) and keeping constant A_{in} one should record the output signal amplitude on filter (A_f). Next the attenuation L_i of the investigated filter for particular frequencies should be calculated:

   $$L_i = 20log\left(\frac{A_f}{A_{in}}\right)$$

   All the results have to be presented in the Table 3.1.1. of the report.
4. After having done the log-linear graph of $L_f(f)$ the values of lower f_l and upper f_u boundary frequencies (i.e. frequencies at which attenuation is 3 dB, i.e. $L_f = -3\ dB$) of examined filter should be found. Basing on experimental values of lower f_l and upper f_u boundary frequencies and the theoretical equations (eq. 2.3.2-2.3.4) the experimental center frequencies and relative band widths of the examined filter should be calculated. The results of calculations should be written in table 3.1.2. (measured side).
5. The results obtained from measurements should be compared with theoretical parameters of the filter.
6. The second log-linear graph presenting the dimensionless characteristics for examined filter should be prepared $L_f(f/f_o)$.

7. The third log-log graph (i.e. f axis should be logarithmic one) presenting the characteristic for examined filter should be prepared $L_f(f)$. The answer to the question: *which one of the presented graphs is the most useful for filter analysis* should be answered in the conclusions.
8. The steps 1-6 for 10 % filter should be performed. New results for the filter should be presented at the same (above described) graphs, prepared earlier for the third octave filter.

5.2. Diagnostics of a machine

The object of diagnostics is a grinding machine. Its diagnostics should be performed on basis of the spectral analysis of Acoustic Emission (AE) of this machine. Work order is as follows:

1. After consultation with instructor the microphone should be connected to the analyzer. The microphone should be placed in an appropriate position.
2. One should set the filter type switch to the "linear" position.
3. Grinder should be turned on.
4. Total level of sound L emitted by the machine under diagnostics should be recorded from the meter.
5. From transformed dependence (Appendix A): $L = 20log\left(\frac{p}{p_o}\right)$ (p_o=20 μPa – level of pressure reference for acoustic signals) the acoustic pressure equivalent to measured overall AE p should be calculated.
6. Weighting levels of sound emitted by the machine should be recorded from the meter by using filters ***A*** and ***C***.
7. Using third-octave filter the mean values of signal attenuation L at the output of the filter for succeeding center frequencies should be read and the results should be recorded into Table 3.2.1.
8. The highest noise levels measured using the third-octaves from Table 3.2.1. should be shown in Table 3.2.2. Next, using 10 % filter, the dominating frequencies inside these third-octaves should be measured and recorded into Table 3.2.2.
9. The final results should be presented together at one graph $L(f)$ for both sets of filters (frequency axis in logarithmic scale).
10. From the graph mentioned above, the bands of maximum values of L should be described and discussion of final results should be completed on the basis of the design data of the grinder and a hypothesis about sources of AE should be presented.

6. Final remarks

The report should consist of:

- Clearly formulated aim of the experiment.
- Short description of course of the experiment (2-3 phrases).
- Tables with recorded results.
- Graph drawn on millimeter paper (or printed from PC) according to hints given in the instruction.
- Discussion of the obtained results and authors own remarks and suggestions. Discussion should consist of:
    - ❑ Comparison of obtained band filter parameters (for octave and third octave filter), one with another and with the theoretical values.
    - ❑ Identification of the machine parts, which can be the source of maximum sound levels in the analyzed band of frequencies.

- Presentation of diagnostic research conceptions concerning technical condition of the machine being tested on the basis of executed spectrum analysis.

Questions and problems to consider:

- What does the process of filtration and its practical applications consist in ?
- Principal conditions for process of filtration being carried out.
- What are the main types of filters in classical acoustic measurements ?
- Characterization of different filters.
- What is the differences between an ideal and a real filter ?
- How do we define filter limiting frequencies ?
- Filter application in the metrology and everyday life (natural filters in nature).

Literature:

The description of the laboratory exercise was prepared based on the following materials:

Bruel & Kjaer documentations and printed materials.

Descriptions of the test rig prepared by the Metrology Group for laboratory exercise for students of Mechanical Department, Technical University of Lodz, Poland.

Jenkins G.M., Watts D.G., 1968: *Spectral Analysis and its Applications.* Holden-Day, S.F.

Lyons R.G., 1997: *Understanding Digital Signal Processing*. Addison Wesley Longman Inc.

Randall R.B., 1977: *Application of B&K Equipment to Frequency Analysis*. B&K, Naerum Offset, Denmark.

Test rig

The test rig is located at the Institute of Turbomachinery (Metrology Group), Technical University of Lodz, Poland.

APPENDIX A

Basic information about acoustics

Sound is a rapid change of pressure in the resilient surrounding. Temporary pressure changes in the surrounding, called acoustic pressure, are defined as effective value of temporary difference of pressure value, which has appeared in a particular point of the surrounding under the interference of acoustic vibrations and statistic pressure value (which exists in a particular point before the vibrations have been caused).

Human ear starts to react to the sound with frequency of 1000 Hz corresponding to acoustic pressure p_o = 20 μPa. This value has been accepted as principal value of acoustic pressure, and it is so called sensitivity threshold of human ear (for different frequencies the principal values of acoustic pressure are different).

Hence, the sound with the pressure above p = 20 Pa already cases pain in human ear. Because the rate of change is very big ($p/p_o = 10^6$), for practical purposes relative measure of acoustic pressure has been implemented, so called acoustic pressure level L:

$$L = 10\log\left(\frac{p}{p_o}\right)^2 = 20\log\left(\frac{p}{p_o}\right) \quad [dB] \tag{A1}$$

In the Table A1 acoustic pressure p [Pa] and level of sound intensity L [dB] dependence is graphically illustrated and examples of common sound intensities are given.

p	L[dB]
100 Pa	134
10 Pa	114
1 Pa	94
100 mPa	74
10 mPa	54
1 mPa	34
100 μPa	14
20 μPa	0

Table A1

dB	
140	pain boarderline
130	take off jet aircraft (20m)
120	take off the jet air cratf(100m)
110	music band
100	pneumatic hammer
90	heavy transport
80	vehicle cabin
70	medium traffic
60	loud conversation
50	office
40	suburb
30	library
20	muffled room
10	acoustic vessel
0	threshold of hearing