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# Analysis of Vibrations and Noise to Determine the Condition of Gear Units

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## 1. Introduction

The main goal of maintenance is to maintain the characteristics of a technical system at the most favourable or still acceptable level. Maintenance costs can be reduced, operation can become more reliable and the frequency and complexity of damages can be reduced. To evaluate the condition of a technical system, it is necessary to collect precise data, and the to analyse, compare and process this data properly.

Gear units are the most frequent machine parts or couplings. They consist of a housing, toothed wheels, bearings and a lubricating system and are of various types and sizes. Durable damages in gear units are often a result of the following factors: geometrical deviations or unbalanced component parts, material fatigue, which is a result of engagement of a gear pair, or damages caused to roller bearings.

To monitor the condition of mechanical systems, methods for measuring vibrations and noise are usually used; data about a gear unit can be acquired in this way. Afterwards certain tools are used to analyse the data. Features that denote the presence of damages and faults are identified in this way.

## 2. Noise source identification

A visualization method of complex noise sources on the basis of an acoustic camera is presented. The method is based upon a new digital signal processing algorithm. This algorithm makes it possible to visualize all types of different complex noise sources from their far area. It is possible to observe monopole, dipole or quadropole noise sources, which occur simultaneously. In addition to this, reflections from hard walls, and refraction and scattering of sound waving can be observed.

The difference between the acoustic camera operation and the acoustic ray reconstruction method is great. Signals from all microphones, located along the ring or the cross of the acoustic camera, are processed in a complex way, by means of the acoustic camera algorithm. On the basis of this algorithm, delays are appropriately corrected in time domain – in relation to time, i.e. to the path length of sound waving from the elementary source to an individual microphone located in the camera – and not by means of phases as in frequency domain as in relation to the sound ray reconstruction method.

Sound waving travels along paths  $r_i$  of various lengths from the elementary acoustic source  $V(x_j)$  to an individual microphone on the ring of an acoustic camera (Fig. 1). Paths travelled by sound waving  $|r_i|$  are of different lengths and, consequently, signal delays  $\Delta_i$  of the same sound waving, produced at the elementary sound source  $V(x_j)$ , are different as well.

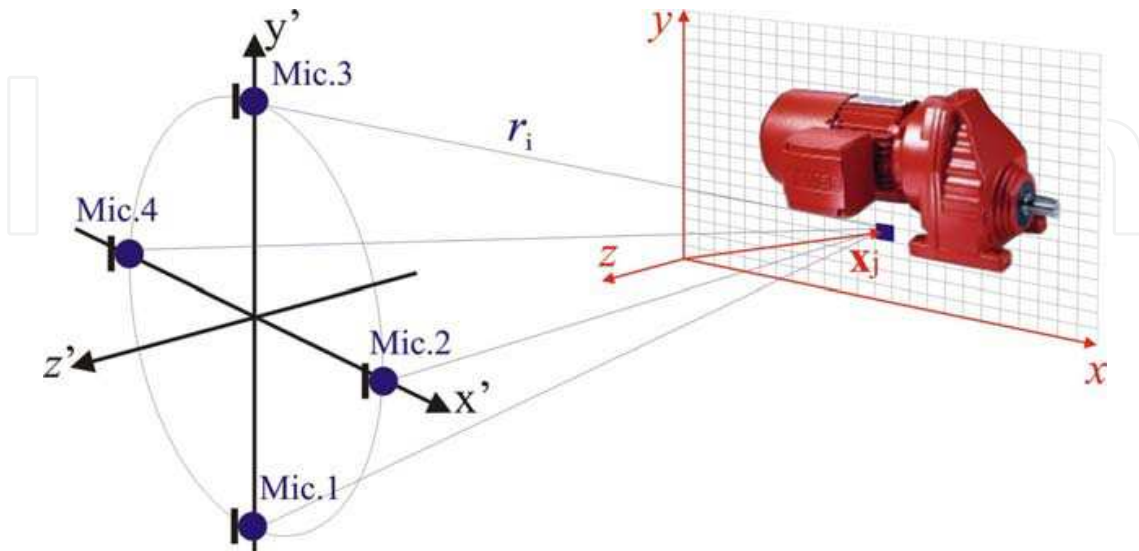


Fig. 1. Path length of an elementary source to individual microphones on an acoustic camera

Figure 2 shows an electrical signal of four microphones. The sound path from the elementary source to microphone 2 is the shortest, and the signal of microphone 2 is the fastest. The second fastest is the signal of microphone 1, the third and the fourth are signals of microphones 3 and 4. Acoustic holography calculation is based upon these delays in time.

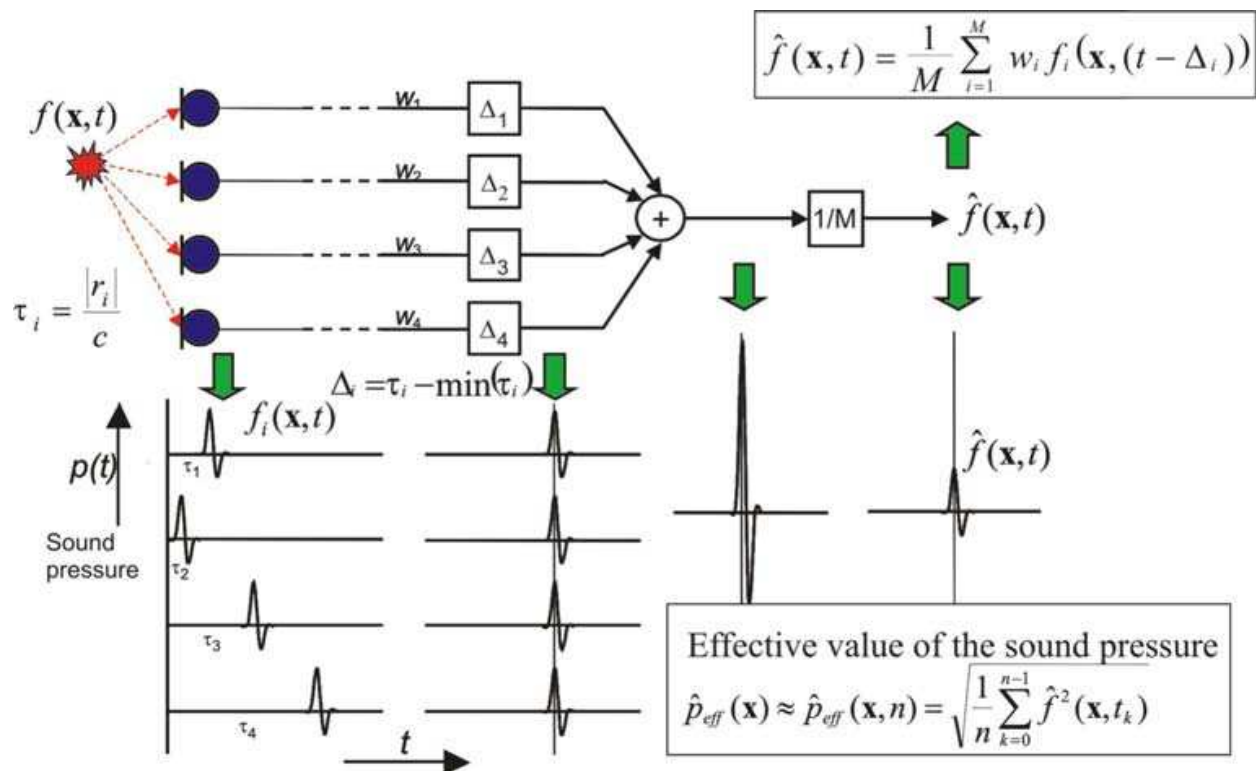


Fig. 2. Acoustic holography calculation method in relation to acoustic camera

Heinz Interference Transformation algorithm, which represents the basis for an acoustic camera, creates a pseudo inverse acoustic field with interference integrals by approximating the original acoustic source in the best way possible (by moving it forward and backward simultaneously). Time-negative reconstruction in a time positive way is realized, using this algorithm. The result is a surface of equivalent acoustic pressure at the point of greatest emission.

If we assume that sound waving from each elementary source reaches each microphone on the ring of the acoustic camera, signals arriving from different microphones can be shifted and integrated in time. For each elementary source, a new signal  $f(\mathbf{x}_j, t)$  is obtained, using the following equation:

$$\hat{f}(\mathbf{x}_j, t) = \frac{1}{M} \sum_{i=1}^M w_i f_i(\mathbf{x}_j, (t - \Delta_i)) \quad (1)$$

Afterwards the effective value of the sound pressure  $f_{\text{eff}}(\mathbf{x}_j, t)$  can be calculated on the basis of this signal:

$$p_{\text{eff}}(\mathbf{x}_j) \approx p_{\text{eff}}(\mathbf{x}_j, n) = \sqrt{\frac{1}{n} \sum_{k=0}^{n-1} \hat{f}^2(\mathbf{x}_j, t_k)} \quad (2)$$

Effective sound pressure represents a mean square value of the acoustic pressure, caused by the elementary acoustic signal at the spot of emission. The corresponding point in the acoustic image must be coloured appropriately. This depends on the position of the elementary source and on the value of its effective acoustic pressure. Areas with high effective sound pressure are usually coloured red, and areas with lower effective sound pressure are blue, which gradually fades until these areas become white. For each elementary source, the procedure must be repeated in order to obtain the entire acoustic image of the acoustic source. In case of more acoustic sources, it is possible to find out, on the basis of the acoustic image, which acoustic source at the measurement spot is the one that contributes mostly to effective acoustic pressure.

The resolution of place and time of an acoustic image, produced with an acoustic camera, has an impact upon the form of sound signals. An impulse of sound pressure has an ideal form in relation to the algorithm of an acoustic camera, and pure sine-shaped form of acoustic waving is the least favourable sound pressure phenomenon. All real sound pressure phenomena can be placed between these two forms. The sinus function, i.e. the Fourier area, represents the basis for most of the acoustic theory. This includes the theory of image method in a nearby field and the theory of acoustic ray reconstruction method. Pure sine-shaped form is very rare in relation to real sound/noise signals. Consequently, the application possibilities of the acoustic camera algorithm are much wider than the application possibilities of other algorithms developed so far.

The acoustic camera is the only acoustic source visualization method functioning exclusively in time domain; it is not necessary to use the Fourier transform to calculate the acoustic image. This means that the method using the acousting camera is not limited in the same way as are methods using the Fourier transform. Frequency analysis is part in the user system but the algorithm calculates the acoustic image first and only afterwards the Fourier transform.

The measurement system in relation to the acoustic camera of the GFaI with dRec48C192 and 32 phase coordinated microphones is presented in Figure 3. The microphones are located on a ring, in concern to the work in a free acoustic field. For an acoustic camera, prepolarised condensation microphones with linear frequency of 23 kHz (-3 dB) are used. Their response is slowly reduced from 6 dB per decade to 40 kHz. It is required to use microphones with such a high frequency area to achieve adequate resolution of the acoustic image. It is possible to achieve higher resolution in relation to higher sampling frequencies or better phase coordination of microphones.

The resolution of an analogue-digital converter is 21 bits. The highest sampling frequency is 196 kHz per channel. Digitalised signals are stored in this converter temporarily, during measurement. After data transfer to a personal computer, taking a few seconds, it is possible to calculate the sound source acoustic image.

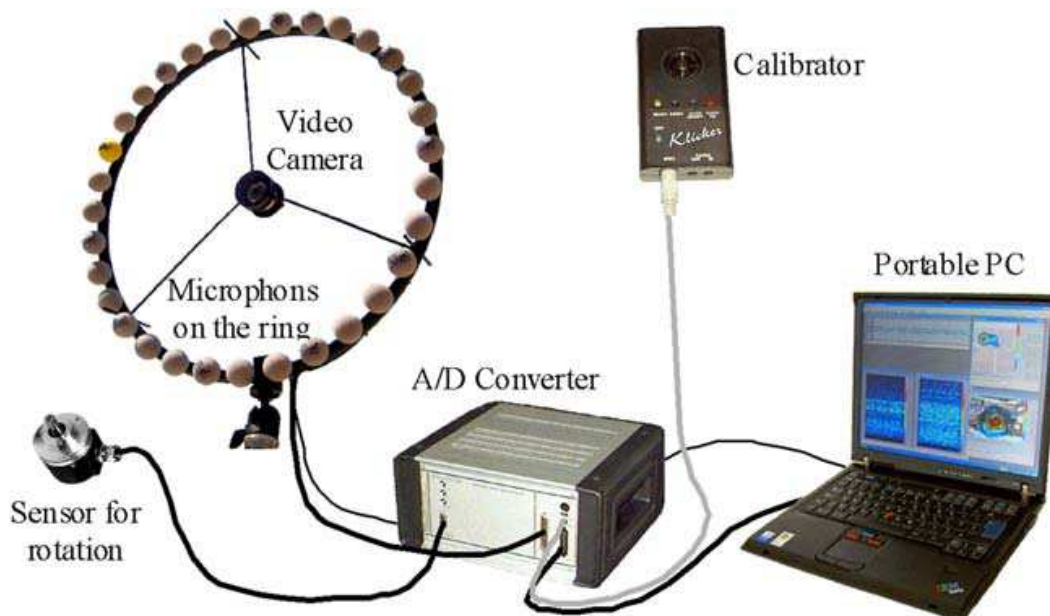


Fig. 3. An acoustic camera system GFaI for visualisation of acoustic sources

### 3. Adaptive time-frequency identification

A gear unit is a set of elements enabling the transmission of rotating movement. Although it is a complex dynamic model, its movement is usually periodical, and faults and damages represent a disturbing quantity or impulse. Local and time changes in vibration signals denote disturbance, consequently, time-frequency changes can be expected. This idea is based on kinematics and operating characteristics.

The presence of cracks in gear units is the most important factor with a negative impact upon the reliability of operation and quality of operation of gear units. It is usually possible to determine deviations from reference values on the basis of a frequency spectrum. As mentioned, it is impossible to identify changes of a frequency component in time as a gear unit is a complex mechanical system with changeable dynamic reaction; this makes the approach based on time-frequency methods more appropriate.

It often happens that some frequency components in signals are present from time to time only. In such cases classical frequency analysis does not suffice to determine when certain



frequencies appear in a spectrum. If time-frequency analysis is used, it can be determined not only in what way frequency components of non-stationary signals change with time but also their intensity levels.

Frequency analysis is often used in diagnostics, but good results are obtained more or less only in relation to periodical processes without local changes. A presence of a damage or fault leads to changes in dynamic parameters of a mechanical system. This influences the frequency spectrum. Monitoring frequency reaction is one of the most common spectral methods to identify the condition of a gear unit. With classical frequency analysis, time description of vibration is transformed into frequency description, and changes within a signal are averaged within the entire time period. This means that local changes are lost in the average of the entire function of vibrations. As a result, it is very difficult if not impossible to define local changes.

These deficiencies are eliminated with the use of the time-frequency analysis: local changes that deviate from the global periodical oscillation are expressed with the appearance or disappearance of individual frequency components in a spectrogram. A signal is presented simultaneously in time and frequency.

Individual frequency components often appear only from time to time in signals related to technical diagnostics. On the basis of classical frequency analysis of such signals, it is not possible to determine when certain frequencies appear in the spectrum. The aim of time-frequency analysis is to describe in what way frequency components of non-stationary signals change with time and to determine their intensity levels.

Fourier, adaptive and wavelet transforms and Gabor expansion are representatives of various time-frequency algorithms. The basic idea of all linear transforms is to perform comparison with elementary function determined in advance. On the basis of various elementary functions, different signal presentations are acquired.

Qian improved and concluded adaptive transform of a signal to a large extent although many authors had been developing algorithms without interference parts, which make individual transforms less useable as opposed to Cohen's class.

Adaptive transform of a signal  $x(t)$  is expressed as follows:

$$x(t) = \sum_p B_p \cdot h_p(t) \quad (3)$$

where analysis coefficients are determined by means of the following equations

$$B_p = \langle x, h_p \rangle \quad (4)$$

expressing similarity between the measured signal  $x(t)$  and elementary functions  $h_p(t)$  of transform. The original signal represents the starting point with parameter values  $p=0$  and  $x_0(t)=x(t)$ . In the set of desired elementary functions,  $h_0(t)$  is searched for that is most similar to  $x_0(t)$  in the following sense

$$|B_p|^2 = \max_{h_p} \left| \langle x_p(t), h_p(t) \rangle \right|^2 \quad (5)$$

for  $p = 0$ . The next step includes the calculation of the remaining  $x_1(t)$

$$x_{p+1}(t) = x_p(t) - B_p \cdot h_p(t) \quad (6)$$

Without giving up the generalisation idea,  $h_p(t)$  is to have a unit of energy representation of a signal.

$$\|h_p(t)\|^2 = 1 \quad (7)$$

The energy in the remaining signal

$$\|x_{p+1}(t)\|^2 = \|x_p(t)\|^2 - |B_p|^2 \quad (8)$$

The equation (6) is repeated in order to find  $h_1(t)$  that would suit best  $x_1(t)$ , etc. One elementary function  $h_p(t)$  that suits best  $x_p(t)$  is found in each step. The primary purpose of adaptive signal representation is to identify a set of elementary functions  $\{h_p(t)\}$ , most similar to time-frequency structure of a signal, and at the same time satisfy equations (3) and (4).

If Wigner-Ville distribution is used for both sides of the equation (3), and if equations are organised into two groups, the result is as follows:

$$P_{WV}x(t, \omega) = \sum_p B_p^2 \cdot P_{WV}h_p(t, \omega) + \sum_{p \neq q} B_p \cdot B_q \cdot P_{WV}(h_p, h_q)(t, \omega) \quad (9)$$

The first group represents elementary signal components and the second one represents cross interference terms.

A new time-dependent adaptive spectrum can be defined in the following way:

$$P_{ADT}(t, \omega) = \sum_p |B_p|^2 \cdot P_{WV}h_p(t, \omega) \quad (10)$$

As an adaptive spectrum based on representations, it is called an adaptive spectrogram. As opposed to Wigner-Ville distribution, it contains no interferences and no cross terms, and it also satisfies the condition of energy conservation.

$$\|x(t)\|^2 = \frac{1}{2 \cdot \pi} \cdot \iint P_{ADT}(t, \omega) \cdot dt \cdot d\omega \quad (11)$$

The basic issue related to linear presentations is the selection of elementary functions. When it comes to Gabor expansion, a set of elementary functions comprises time-shifted and frequency modulated prototype window function  $w(t)$ . In relation to wavelets, elementary functions are acquired by scaling and shifting a mother wavelet  $\psi(t)$ . In these two examples, structures of elementary functions are determined in advance. Elementary functions related to adaptive representation are rather demanding.

As adaptive transform permits arbitrary elementary functions, it is, generally speaking, independent from the choice of elementary functions  $h_p(t)$ .

Elementary functions, used for adaptive representation of a signal with equation (3), are very general but this is not always so in practice. It is desirable that elementary functions are localised in regard to time and frequency in order to emphasize time dependence of a signal. Also it must be possible to use the presented algorithm in a relatively simple way. In relation to adaptive representation, a Gauss type signal with its very favourable features is considered a basic choice.

The calculation of an adaptive spectrogram begins in a wide time range of a measured signal. Then the range must be decreased, depending on what the goals are. Fourier integral is among the elementary operations of searching for a suitable elementary function, and so the described calculation process is very effective. The accuracy of approximation depends primarily on the size of time-frequency interval. With narrower intervals, the representation is more accurate, but the calculation time is longer. This means that it is necessary to find a compromise between the accuracy of approximation and its efficiency.

#### 4. Practical example

The measurements were performed in the test plant of the Laboratory for Pumps, Compressors and Technical Acoustics of the Faculty of Mechanical Engineering, University of Ljubljana. The room in which the tests were carried out was not specially adjusted for performing acoustic measurements as the noise level was between 36 and 42 dB(A). This level can be achieved also in an industrial environment that is located adequately far away from intense noise sources.

A single stage gear unit with a helical gear unit with straight teeth integrated into it was used.

Two pairs of spur gear-units, built in a single stage gear-unit, were used for noise measurements. One of the pairs had a crack and the other one was without it. Thus, tests were carried out, using faultless and faulty gear units.

The aim of the measurements was to determine the presence of individual changes in a gear unit. The measured signal of a faultless gear unit and the signal of a faulty gear unit were compared to determine the gear unit condition.

Measurements were carried out under operating conditions normally associated with the relevant type of a gear unit. A ground gear pair used was a standard gear pair, with the teeth quality 6, but it had a crack in the tooth root of a pinion. It is presented in Fig. 4.

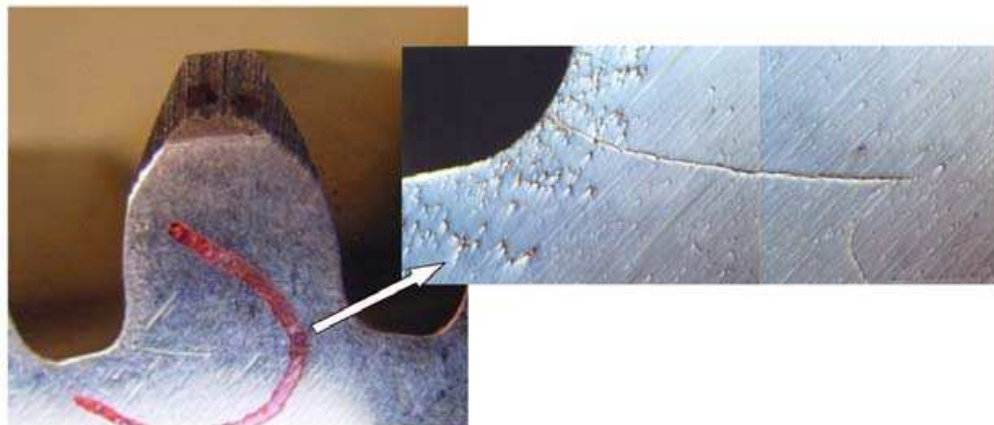


Fig. 4. A pinion with a crack in the tooth root

Adaptive time frequency transform was used to determine the presence of a crack in the tooth root, whereby the LabVIEW software tools, including the author's own software modules, were used [9].

The length of the signal of measured values was 1 s; on an average, the signal was composed of 192000 measuring points. At the time of measurement, the rotational frequency was 28,5Hz. The number of teeth of the pinion was 18, and of the gear unit 99.



In Fig. 5, the acoustic image with sound level of a gear unit is presented, where the engagement area of a gear pair can be observed as a noise source, whereas in Fig. 6, the adaptive spectrogram of noise source is presented; it is not possible to note any rhythmic pulsation of harmonics, with the exception of typical frequencies, defined on the basis of typical frequency components. Some pulsation sources are indicated (but not expressed) and their stochastics. It is very interesting to monitor the increase or decrease (even complete disappearance) in appropriate frequency components with pulsating frequency.

In Fig. 7, the acoustic image of a gear unit with a crack in the tooth root is presented, where it is possible to note the engagement area of a gear pair as a noise source, and in Fig. 8 and Fig. 9, the adaptive spectrogram of noise source is presented. Rhythmic pulsation of some frequency can be observed. This is typical for meshing frequency 515 Hz. Pulsation is expressed only in relation to the presence of a crack. Pulsation reflects a single engagement of a gear pair with a crack.

To determine the presence of a crack in the tooth root, adaptive transform was used for vibration analysis. In relation to adaptive spectrogram, adaptive representation for signal decomposition, prior to Wigner-Ville distribution, was used.

A fine adaptive time-frequency resolution is characteristic of an adaptive spectrogram due to limited features of elementary functions. Consequently, time-frequency resolution of the transform is adapted to signal characteristics. As an elementary function, it is possible to apply Gauss function (impulse) and linear chirp with Gauss window. If a signal contains linear chirps resulting from a linear change in the rotational frequency of a gear unit, it is possible to use an adaptive spectrogram to determine in what ways a possible frequency modulation is reflected in the time-frequency domain. The transform calculation time increases, along with the larger amount of data and the increased number of cycles required to search for an adequate elementary function.



Fig. 5. An acoustic image with noise source of a gear unit of a faultless gear

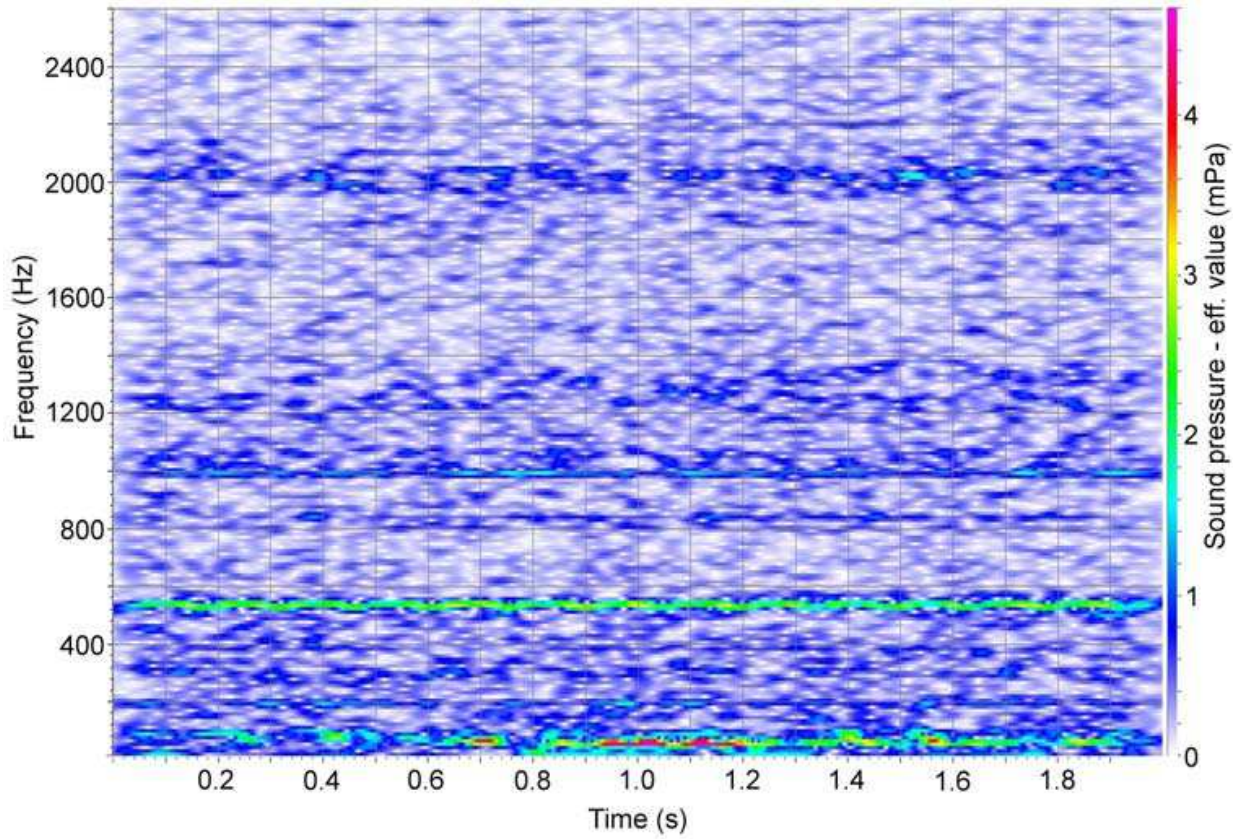


Fig. 6. A adaptive spectrogram of a gear unit of a faultless gear



Fig. 7. An acoustic image with noise source of a gear unit with a crack in the tooth root



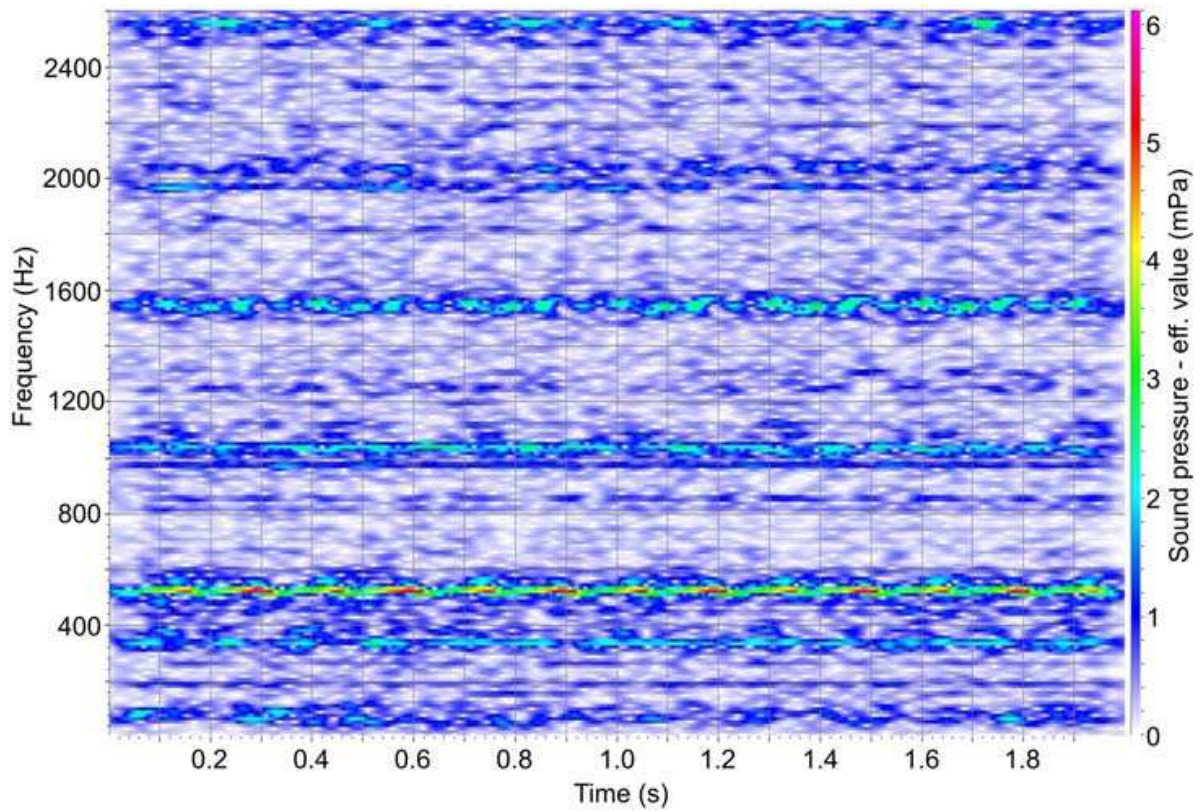


Fig. 8. A adaptive spectrograms of a gear unit with a crack in the tooth root; position 1

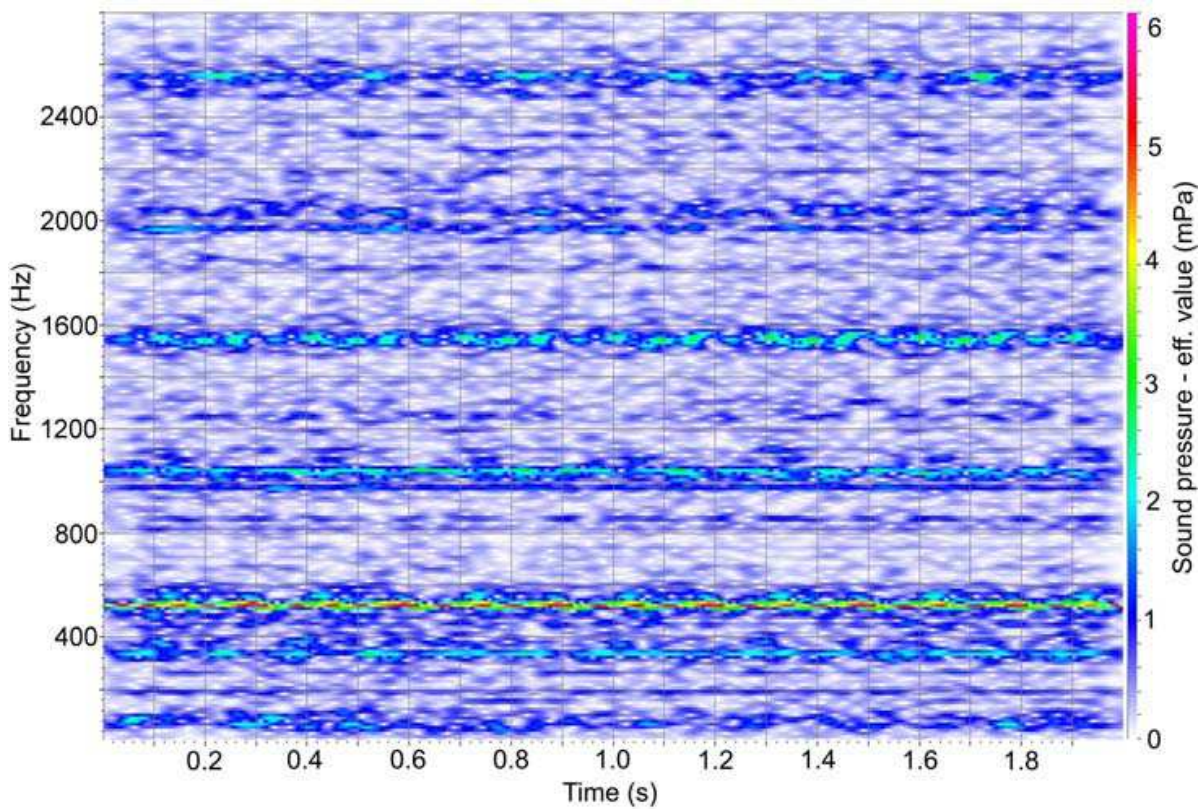


Fig. 9. A adaptive spectrograms of a gear unit with a crack in the tooth root; position 2

The vibration signal of measured values was 1 s long and composed of, on an average, 50000 measuring points. At the time of measurement rotational frequency was 20 Hz. Adaptive spectrograms in relation to Gabor transforms are presented for comparison. The length of the window is 6800 points, which is 15% more than the length of the period of one rotation of a gear pair.

Calculation time required for adaptive spectrogram is at least 10 times longer than the calculation time for the Gabor transform, but the resolution of the adaptive transform is, on an average, two times better.

Fig. 10 shows Gabor spectrogram; no rhythmic pulsation of harmonics can be noted, with the exception of typical frequencies, determined on the basis of power spectrum. When it comes to adaptive spectrogram (Fig. 11), with a higher level of energy accumulation in the origins, it is possible to note some pulsation sources but they are not very expressed. It is very interesting to monitor how appropriate frequency components with rotational frequency of 20 Hz increase or decrease or even completely disappear. This phenomenon is typical of the 3rd harmonic, 1530 Hz is expressed, only in relation to the presence of a crack. The phenomenon is much more expressed in relation to the adaptive spectrogram (Fig. 13) than in relation to the Gabor spectrogram (Fig. 12).

The spectrogram evaluation can be based on an average spectrogram, which represents an amplitude spectrum of a Fourier or adaptive transform of a measured signal, and on observing pulsating frequencies of individual frequency components.

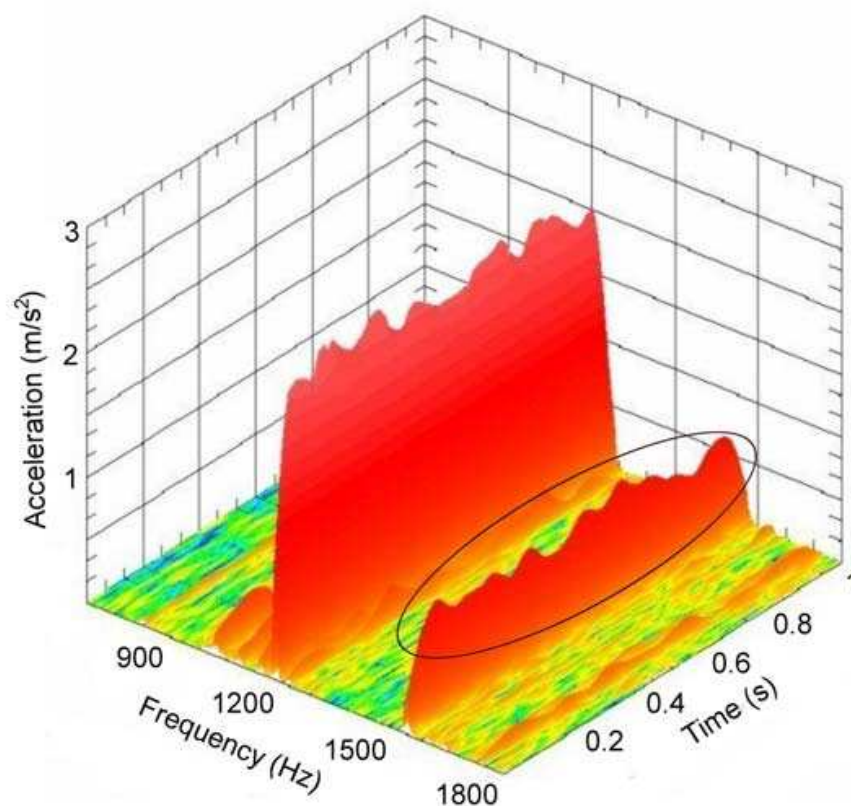


Fig. 10. Gabor's spectrogram of a faultless gear unit



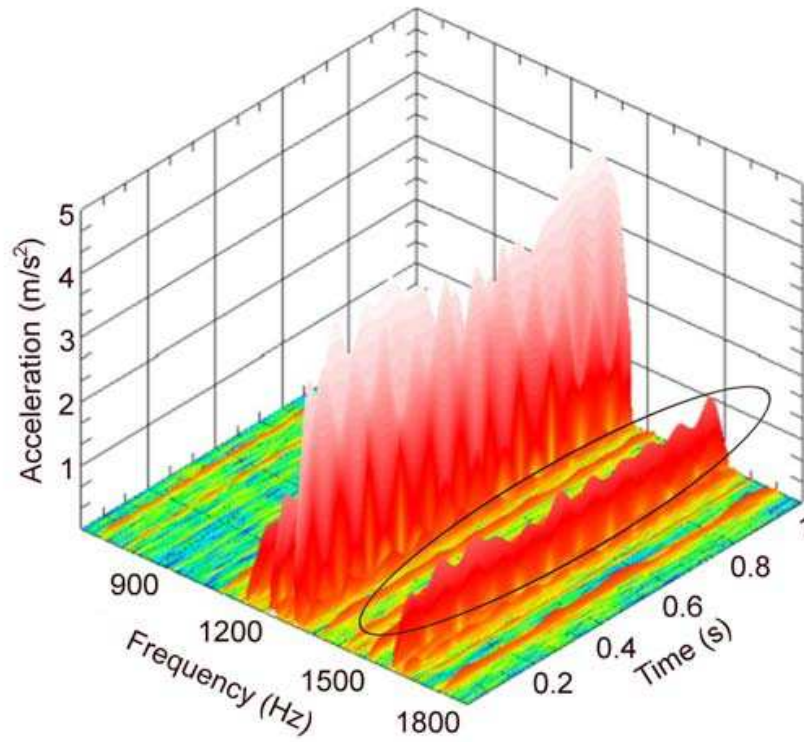


Fig. 11. Adaptive spectrogram of a faultless gear unit

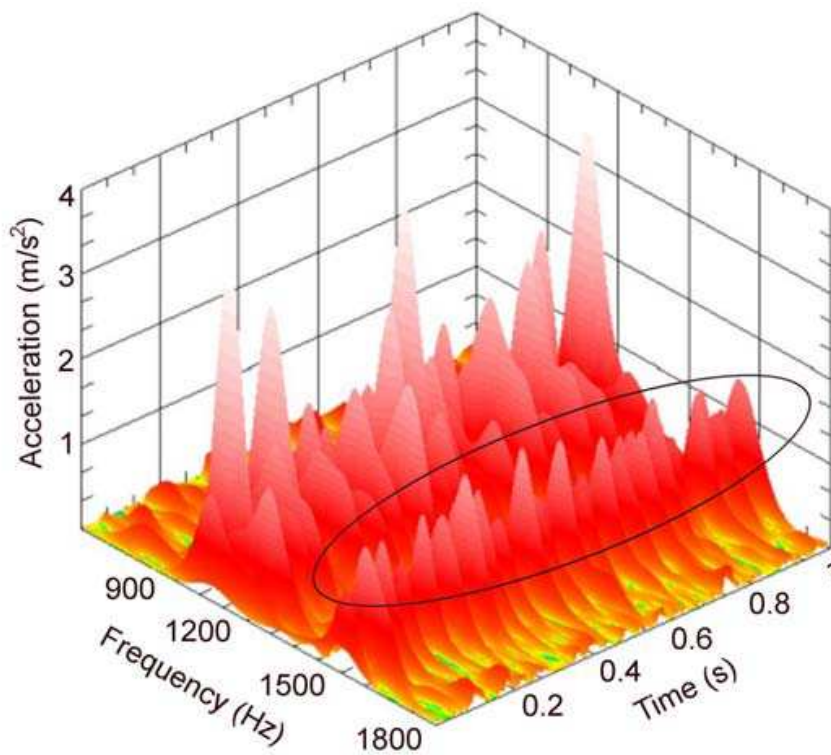


Fig. 12. Gabor's spectrogram of a gear unit with a pinion with a crack



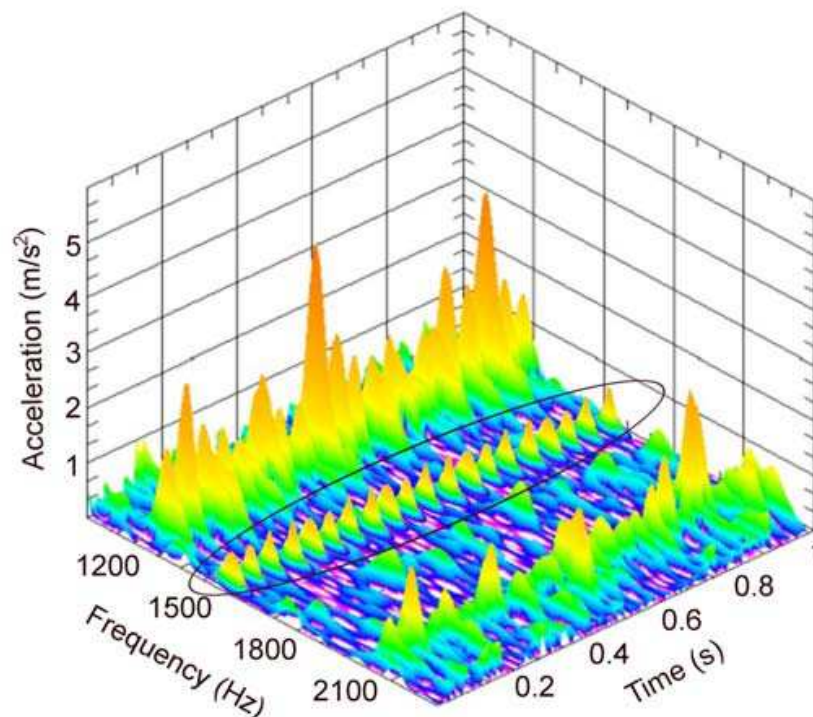


Fig. 13. Adaptive spectrogram of a gear unit with a pinion with a crack

## 5. Conclusion

The resolution in time and in place achieved with the use of an acoustic camera with its specific algorithm, which functions in time domain, and of specifically located microphones for acoustic source visualization is better than with any other acoustic system.

Industrial gear units were used for noise analysis, the purpose of which was to identify faults. The use of the presented methods can improve both, the safety of operation and the reliability of monitoring operational capabilities.

The reliability of monitoring life cycle of a gear unit is improved with the use of appropriate spectrogram samples and the achievement of a clear presentation of the pulsation of individual frequency components, which, along with the average spectrum, for a criterion for evaluating the condition of a gear unit.

When it comes to life cycle design, it is necessary to use an adequate method or criterion to monitor the actual condition of a device and particularly of its vital component parts, which can have a considerable impact upon the operational capability. If faults and damages are detected in time, it is possible to control the reliability of operation to a great extent. The prediction of the remaining life cycle of a gear unit is improved with the use of reliable fault identification methods.

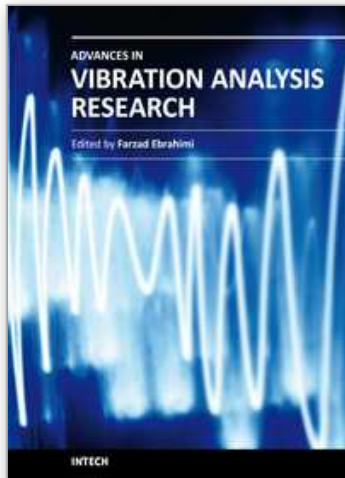
In this contribution, fault identification in industrial gear units is based on vibration analysis; it increases the safety of operation and, consequently, of monitoring operational capabilities.

The life cycle of a gear unit can be monitored more reliably with the use of appropriate spectrogram samples and a clear presentation of the pulsation of individual frequency components that, in addition to the average spectrum, represent a criterion for evaluating the condition of a gear unit. Adaptive time-frequency representation is clearer, without

increased dissemination of signal energy into the surroundings, and it enables reliable fault identification.

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## **Advances in Vibration Analysis Research**

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Vibrations are extremely important in all areas of human activities, for all sciences, technologies and industrial applications. Sometimes these Vibrations are useful but other times they are undesirable. In any case, understanding and analysis of vibrations are crucial. This book reports on the state of the art research and development findings on this very broad matter through 22 original and innovative research studies exhibiting various investigation directions. The present book is a result of contributions of experts from international scientific community working in different aspects of vibration analysis. The text is addressed not only to researchers, but also to professional engineers, students and other experts in a variety of disciplines, both academic and industrial seeking to gain a better understanding of what has been done in the field recently, and what kind of open problems are in this area.

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