

Relative Approach

Relative Approach	1
Introduction	1
Parameters of the <i>Relative Approach</i> analysis	2
Typical applications	4
Summary	9
Literature	9
Notes	9

Introduction

The human ear reacts particularly sensitively to quick changes in acoustic signals and to spectral structures with distinct maxima and minima. Constant or slowly varying sound levels and slow changes of frequency, on the other hand, do not trigger increased attention after listening for some time. This means that human hearing works adaptively. It does not work with absolute measurement values, but judges the distinct *patterns* contained in a sound.

In an A/B comparison in a laboratory, human hearing can detect even small level differences between two sounds. In everyday life, however, there are no such laboratory comparisons. Instead, our ears judge sounds without direct comparison to a reference sound. To do this, human hearing pays little attention to absolute sound levels but concentrates on the patterns found in a sound. It is assumed that in order to identify such patterns, our hearing creates its own set of references based on recently-heard sounds and then compares new sounds with these reference patterns.

The **Relative Approach** analysis was developed to reproduce this behavior and to perform an aurally accurate analysis. The technique is capable of separating a sound into portions containing patterns and portions without patterns, allowing judgment of sound patterns. The **Relative Approach** analysis is available in ArtemiS SUITE¹ as a function of time (adding **2D**) as well as a function of time and frequency (adding **3D**).

The basic idea of the **Relative Approach** is to determine an estimate for the current signal value by extrapolating the known signal history before the current point in time, and then subtract this estimated value from the actual signal value. The difference between the actual value and the estimated value is a measure of the variation of the signal. The analysis thus does not yield absolute, but *relative* values – in accordance with our hearing. If an acoustic signal does not contain any distinct patterns in time or frequency, the **Relative Approach** analysis will yield only very small values.

In order to differentiate the results of the relative analysis from other acoustic and psychoacoustic quantities (which measure absolute values), a new measurement unit is used: The results are given in the quantity *Compressed Pressure* with the unit *Compressed Pascal*.

¹ The descriptions in this Application Note are based on version 9.2 of ArtemiS SUITE. The general procedure also applies to other versions. However, there may be differences in the scope of functions and in the user interface.

Parameters of the *Relative Approach* analysis

As the basis for calculating the *Relative Approach* analysis, a time-frequency analysis of the time signal is required. In the Properties window of the analysis, the calculation method for this *Base Analysis* can be specified (see figure 1).

Figure 1: Properties window of the *Relative Approach* analysis

The following methods are available:

- **1/n Octave (Filter):** Partial-octave spectrum based on digital filters
- **1/n Octave (FFT):** Partial-octave spectrum based on an FFT
- **Hearing Model:** Spectrum based on the hearing model developed by Prof. Sottek
- **Loudness (FFT/HEAD):** Loudness spectrum based on an FFT using an algorithm developed by HEAD acoustics

Depending on the base analysis selected, additional options are enabled or disabled in the Properties window. For example, if the **1/n Octave (FFT)** analysis is selected, the **FFT Size** and the **Overlap** must be specified for the calculation. The window function used for the calculation is a Hanning window. Selecting the **1/n Octave (Filter)** analysis activates the **Time Weighting** selection box, where the integration time for the analysis is specified.

For the $1/n$ -octave analyses (Filter or FFT based), the **Band Resolution** of the octave or partial-octave bands can be configured.

In the **Spectral Weighting** selection box, an A, B, C or D filter is available. With these filters, different frequency ranges can be weighted approximating the properties of human hearing.

Besides the settings for the base analysis, the Properties window is also used to select the **Variation analysis**. This selection determines how the data from the base analysis are further processed to obtain the relative result. The selection box **Variation analysis** provides the options **Regression** and **Prominence 3D**:

- With the **Regression** method, the first step is an aurally accurate level transformation that accounts for the non-linear relation between sound pressure level and subjectively perceived loudness². Afterwards, an estimated value for the spectral data is calculated, based on the previous 200 ms and the neighboring frequencies. The result of the analysis is the difference between the actual data and the estimated value.
- With the **Prominence 3D** method, each sample value of the base analysis is compared to the average value of a surrounding rectangle in the time-frequency domain. The rectangle extends 1 Bark and 50 ms in both directions centered on the respective sample.

² This transformation is not required for the base analysis **Loudness (FFT/HEAD)**.

If **Regression** is selected as variation analysis, the selection box **Regression Algorithm** is activated. In this box, the signal estimation of time-frequency data can be specified. The difference between the algorithms is the sequence of the time and frequency averaging. The setting **Frequency Pattern** is suitable for visualizing patterns in the frequency domain, whereas the setting **Time Pattern** is optimized for resolving temporal structures. It is also possible to combine the two methods (**Frequency + Time Pattern**). In that case, the results are calculated for both the **Frequency Pattern** and the **Time Pattern** settings, weighted according to the factors selected under **Weighting [Factor]** and then summed. Different weightings for frequency and time patterns lead, of course, to different results. To determine suitable weighting factors, it is useful to try different weightings and work interactively, comparing the result to the auditory impression.

The FFT length and the integration time of the base analysis can be specified in separate selection boxes for the two regression algorithms. The selection box for the regression algorithm **Frequency Pattern** is labeled with an **F**, whereas the one for the **Time Pattern** setting is labeled with a **T**. For the combined **Frequency + Time Pattern** method, the two settings need not be identical and can be adapted to meet different requirements in the time and frequency domains.

The settings in the Properties window greatly influence the result. To perform a truly significant analysis, the settings must not only be adapted to the type of sound to be examined, but also to the relevant aspects. Due to the wide variety of possible settings in the Properties window, **Relative Approach** analysis can be optimally configured for a wide range of applications.

For sound patterns in the time domain, the setting **Time Pattern** is suitable. Furthermore, the analysis can be optimized for the temporal pattern length by selecting an appropriate FFT length or time weighting. The shorter the pattern, the shorter the FFT length or time weighting that should be selected. A pattern in the frequency domain can be visualized using the **Frequency Pattern** method. The frequency resolution can be adapted to the relevant frequency pattern by suitable selection of the partial-octave bands (for the base analysis **1/n Octave**) or the FFT length (for the base analysis **Loudness (FFT/HEAD)**).

The base analysis is the basis for the relative calculation, and thus influences the analysis results. The setting **1/n Octave (Filter)** is suitable for patterns in both the time and the frequency domains. The base analysis **Hearing Model** is especially suited for visualizing short temporal patterns at low frequencies, whereas the base analysis **Loudness (FFT/HEAD)** in combination with the variation analysis **Prominence 3D** is very good for detecting tonal components. The setting **1/n octave (FFT)** is suitable for patterns in the frequency domain, and requires comparatively short calculation times.

Typical applications

The following example demonstrates some of the configuration possibilities for **Relative Approach** analysis. At first, the sound generated by a running windshield wiper is examined with differently-configured **Relative Approach** analyses. The sound contains temporal patterns at low frequencies and also tonal patterns between 400 Hz and 1400 Hz. Figure 2 shows an **FFT vs. time** analysis of this sound.

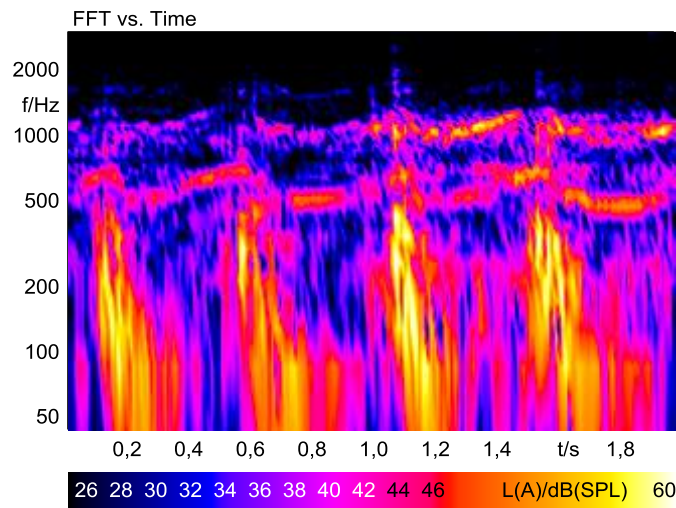


Figure 2: **FFT vs. Time** analysis of a windshield wiper sound

For the following **Relative Approach** analyses, a $1/12$ -octave analysis based on an FFT was used as the base analysis. As variation analysis, **Regression** was selected, in one case with the **Frequency Pattern** setting and an FFT size of 4096 samples and in another case with the **Time Pattern** setting and an FFT size of 1024 samples. The settings specified in the Properties window are shown in figure 3.

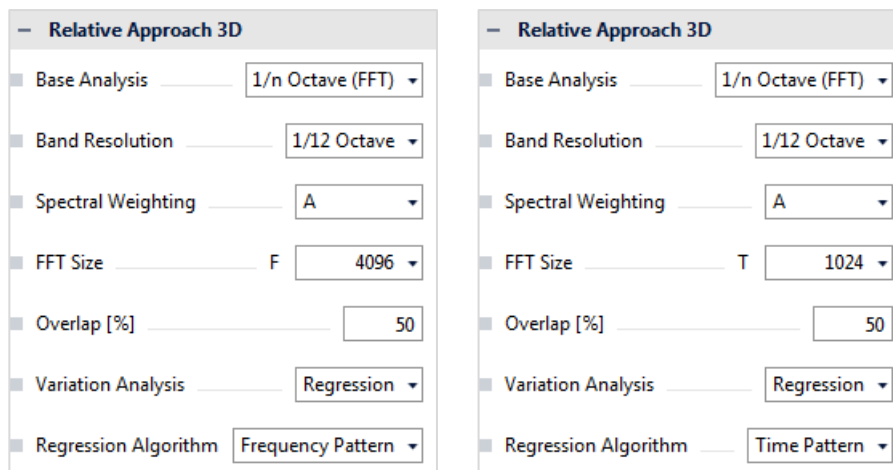


Figure 3: Left side: settings used for the analysis shown in figure 4, left diagram; right side: settings used for the analysis shown in figure 4, right diagram

Figure 4 shows the two analysis results of the windshield wiper sound. In the left diagram (setting **Frequency Pattern**), the tonal components are visible very clearly, whereas in the right diagram (setting **Time Pattern**), the temporal pattern is predominant. The settings can be selected according to the user's individual requirements, so that the relevant sound aspect is shown in the diagram.

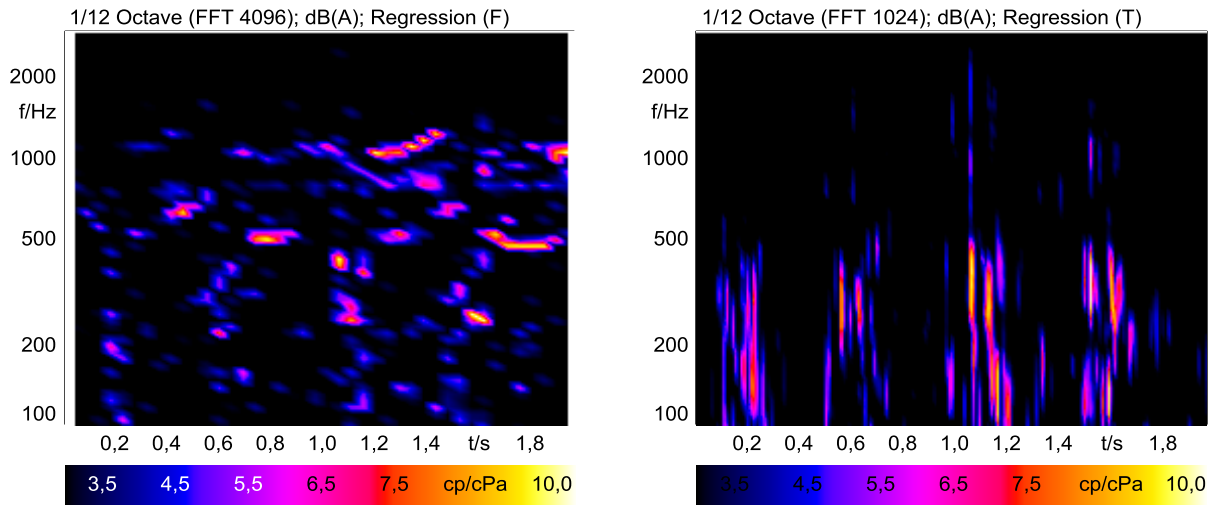


Figure 4: *Relative Approach* analysis of a windshield wiper sound, left diagram: setting **Frequency pattern**, right diagram: setting **Time pattern**

If both the tonal and the temporal patterns in a sound are of interest, both can be examined together. In this case, the regression algorithm **Frequency + Time Pattern** is used. Figure 5 shows the result.

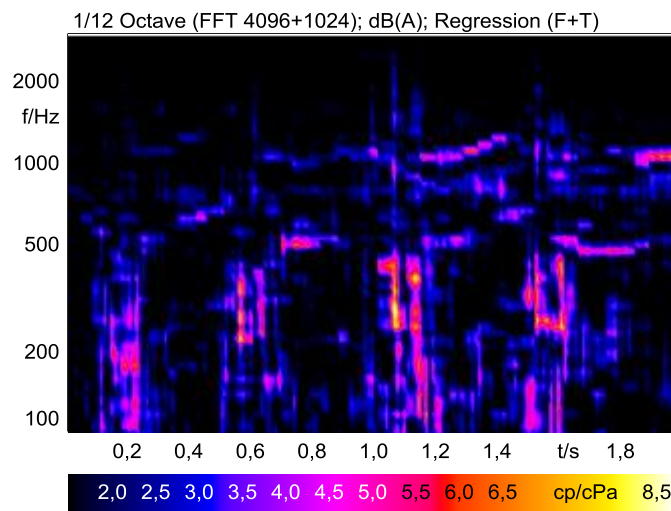


Figure 5: *Relative Approach* analysis of a windshield wiper sound with the setting **Frequency + Time Pattern**

In the next example, the interior noise of a truck is examined. When listening to the sound file, distinct rattles can be heard in the time domain between 2.3 seconds and 2.8 seconds.

Figure 6 shows the analysis of this sound. The result of the **FFT vs. Time** analysis barely reveals the rattling sound, whereas the **Relative Approach** analysis very clearly exposes each individual sound event between 2 kHz and 5 kHz.

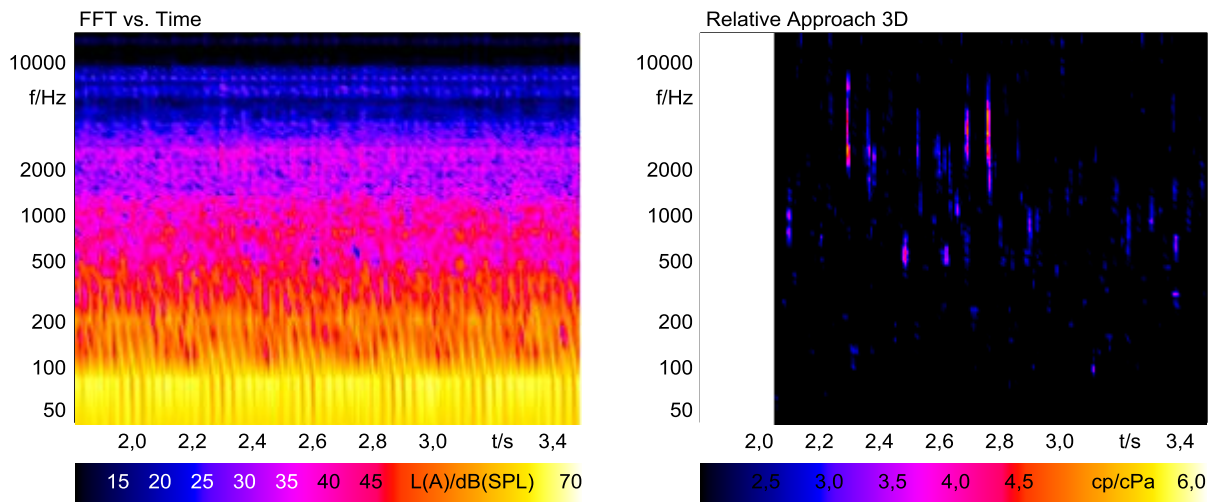


Figure 6: Analysis of a truck interior noise, left: *FFT vs. Time*, right: *Relative Approach 3D*

Figure 7 shows the Properties window for the *Relative Approach* analysis shown in figure 6.

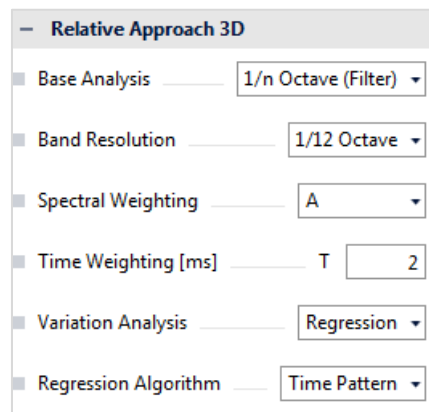


Figure 7: Properties window of the *Relative Approach* analysis used for figure 6, right

Since the relevant sound component is a temporal pattern, the variation analysis *Regression* with the regression algorithm *Time Pattern* was selected.

As base analysis, a $1/12$ -octave analysis based on digital filters was used (setting *1/n Octave (Filter)*). In order to achieve a sufficiently fine time resolution for the rattling noise, the integration time for the analysis in the *Time Weighting* field was set to 2 ms.

As for the calculation of the $1/n$ -octave base analysis with digital filters, the transient effects of the filters generate artifacts at the beginning of the sound. In order to avoid that these artifacts complicate the interpretation of the analysis results or falsify the calculation of single values, the beginning of the sound is not shown in the diagram of the *Relative Approach* analysis. For this reason, the time axes shown in the two diagrams of figure 6 differ.

In the third example, two vehicle interior sounds of traction control systems (TCS) are examined. The first recording was made of an acoustically good TCS, whereas the second recording examines a noisy TCS for comparison. Figure 8 shows the *FFT vs. Time* analysis of the two recordings.

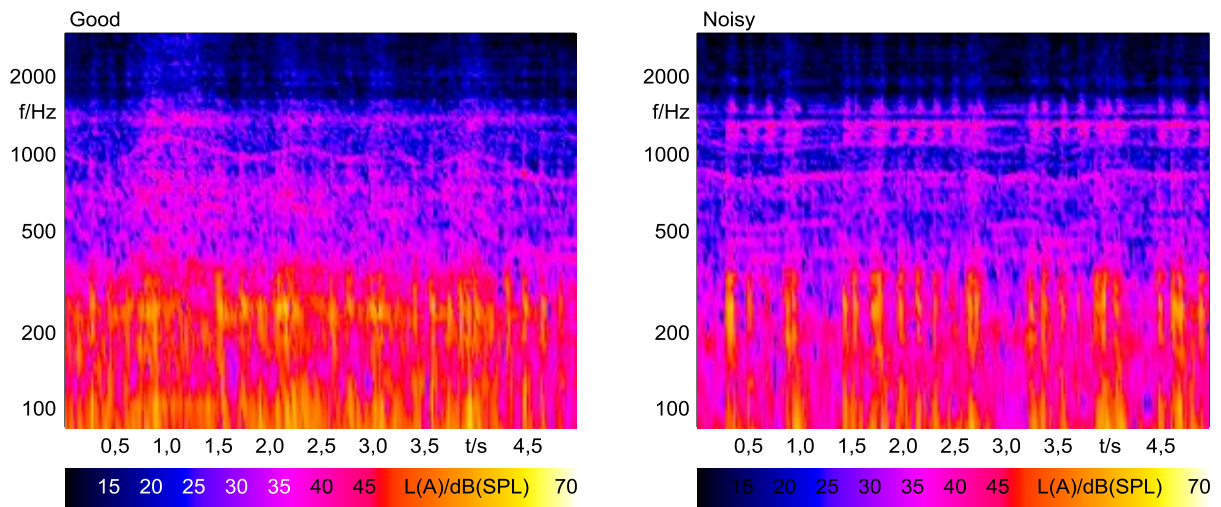


Figure 8: **FFT vs. Time** analysis of an acoustically good (left) and a noisy (right) traction control system

The result of the right FFT analysis shows traces of a pattern in the range between 1000 Hz and 2000 Hz. However, to sort the relevant pattern out of the other patterns in the diagram, considerable experience and knowledge are required. To facilitate the analysis, the sounds were filtered with a high-pass filter at 800 Hz and a low-pass filter at 2500 Hz. Figure 9 shows the FFT analysis of the filtered TCS sounds. However, in spite of the filtering in the frequency range of interest, the actual disturbing noise is still barely identifiable.

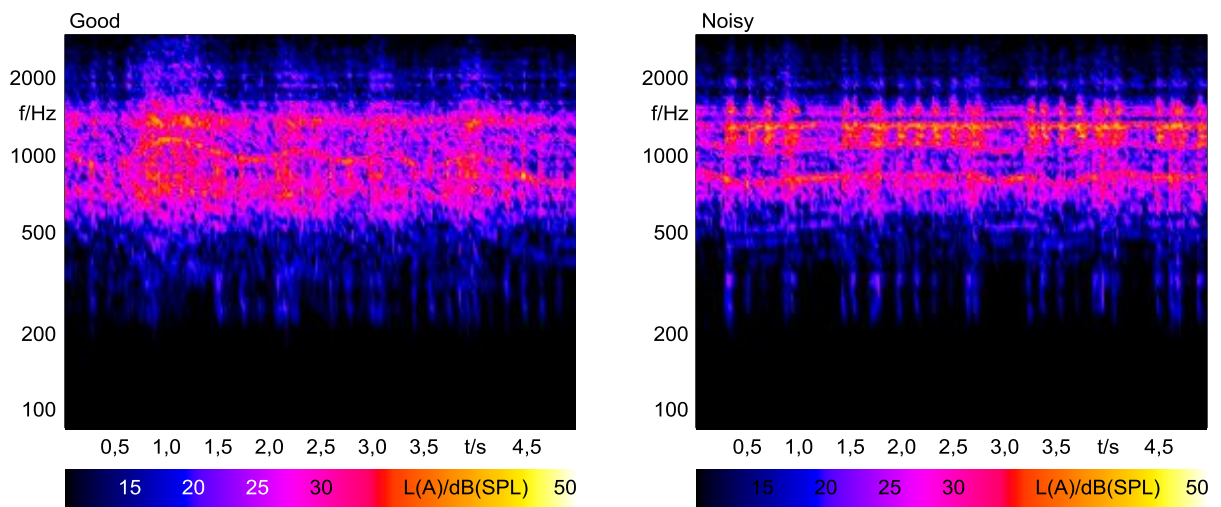


Figure 9: Filtered **FFT vs. Time** analysis of the filtered signal of an acoustically good (left) and a noisy (right) traction control system

Using the **Relative Approach** analysis greatly facilitates the interpretation. Figure 10 shows the result of a **Relative Approach** analysis of the filtered signal. The relevant noise pattern that distinguishes the good TCS from the bad one is now clearly visible. For the diagrams in figure 10, the base analysis **1/n Octave (FFT)** was used. This analysis requires only a short calculation time, so the user obtains a quick overview of the patterns contained in the noise.

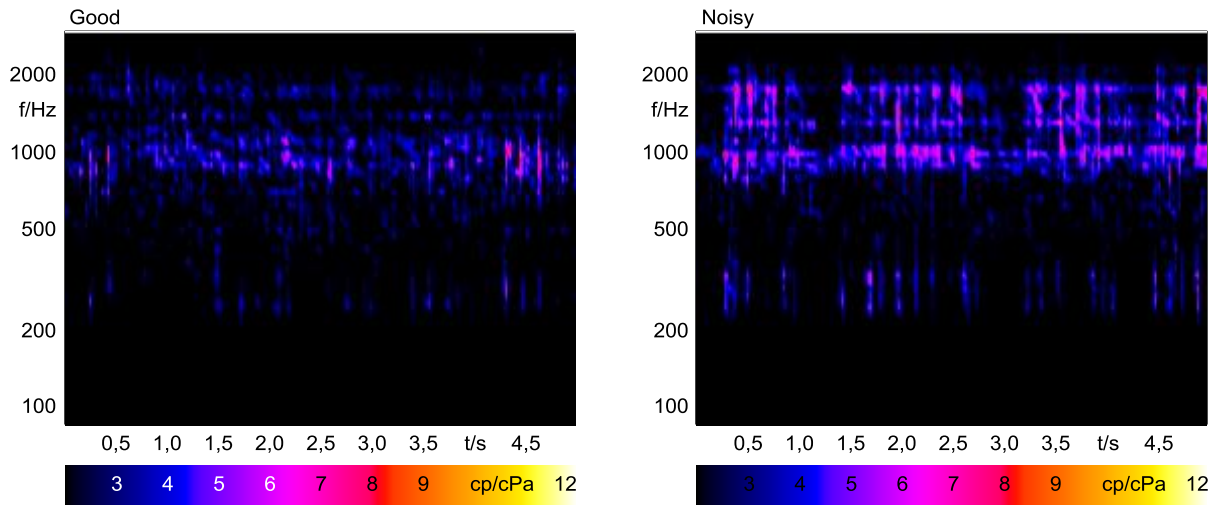


Figure 10: **Relative Approach** analysis, base analysis **1/n Octave (FFT)**, of the filtered signal of an acoustically good (left) and a noisy (right) traction control system

For comparison, figure 11 shows the results of another **Relative Approach** analysis of the TCS sounds, this time using the base analysis **1/n Octave (Filter)**.³ This analysis takes longer to calculate, but allows a more detailed examination of the noise pattern.

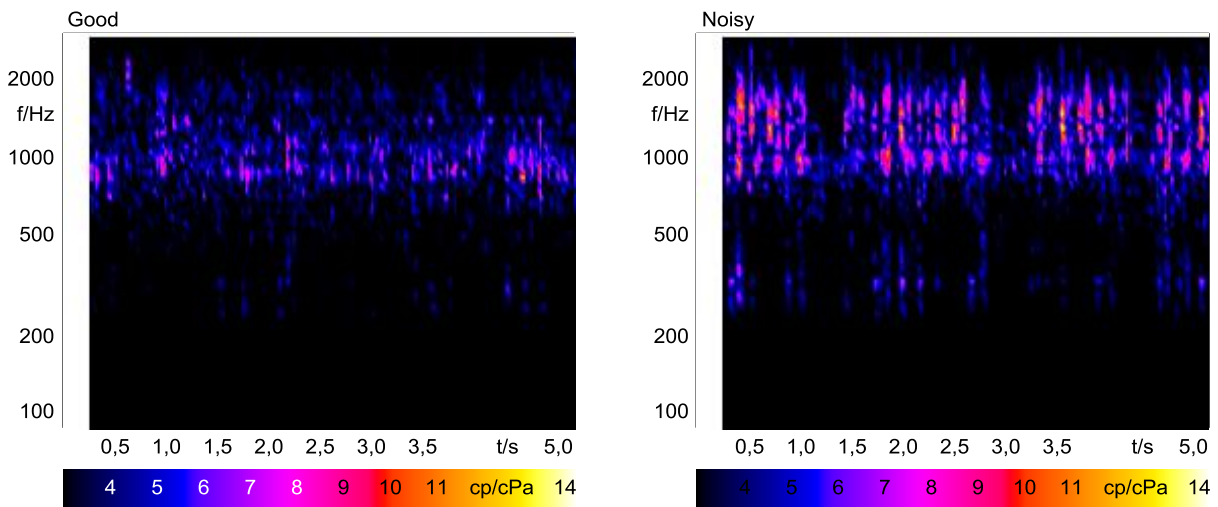


Figure 11: **Relative Approach** analysis, base analysis **1/n Octave (Filter)**, of the filtered signal of an acoustically good (left) and a noisy (right) traction control system

³ Again, the analysis results at the beginning are not displayed in order to prevent the transient effects of the filters from distorting the result.

Summary

The human ear reacts particularly sensitively to distinct patterns contained in a sound. With the **Relative Approach** analysis, such sound patterns can be identified.

For optimal adaptation to the respective analysis requirements, the **Relative Approach** provides a wide variety of configuration options. With the base analysis **1/n Octave (FFT)** and the resolution **1/12-octave**, the user can first get a quick overview of the patterns contained in the sound. For sound patterns in the frequency domain, the **Regression Algorithm** should be set to **Frequency Pattern** and a sufficient window size should be chosen for the FFT, whereas sound patterns in the time domain can be better represented with the **Time Pattern** setting and a small window size.

For more detailed examination, the base analysis **1/n Octave (Filter)** can be used. It requires a longer calculation time, but also gives a more detailed analysis.

Following these basic rules, the user can adapt the **Relative Approach** analysis to the respective application.

Literature

K. Genuit

Objective Evaluation of Acoustic-Quality Based on a Relative Approach
Inter-Noise'96, Liverpool, England, Conference Proceedings (Book 6, pp. 3233-3238)

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Models of Signal Processing in Human Hearing
ELSEVIER, International Journal AEÜ of Electronics and Communications, 59 (2005) 157 – 165
(Dedicated to Prof. em. Dr.-Ing. E. h. Hans Dieter Lueke on the occasion of his 70th birthday)

Notes

For calculating the analyses presented in this Application Note by means of a Pool Project, you need the following ArtemiS SUITE modules: **ASM 00** ArtemiS SUITE Basic Framework (code 5000), **ASM 01** ArtemiS SUITE Basic Analysis Module (code 5001) and **ASM 16** ArtemiS SUITE Advanced Psychoacoustics Module (code 5016). If you want to calculate the analyses by means of an Automation Project or a Standardized Test Project, you may need other modules. Your HEAD acoustics representative will gladly provide you with further information.

Do you have any questions or comments?

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We look forward to receiving your feedback!