Pulse Processing in ArtemiS

ArtemiS provides the user with a wider range of analysis functions, many of which allow their results to be calculated and plotted against revolution speed instead of time (e.g. FFT vs. RPM, order spectrum vs. RPM etc.). To calculate these analyses, information about the current revolution speed is required. This information can be recorded and stored in three different ways:

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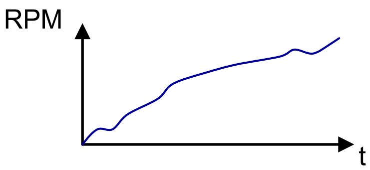
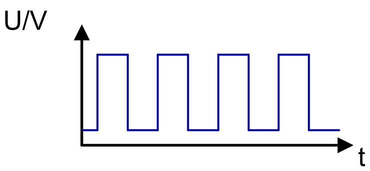
1. Pulse channel
The pulse signal is recorded via a separate pulse input of the recording hardware. Only two possible states are stored: “0” and “1”.



2. Direct storage
The revolution speed curve is stored directly in a signal channel. For such a recording, the hardware must have an input supporting DC signals.



3. Trigger signal
As with the pulse channel, the revolution speed is recorded as a pulse signal. However, the signal is not stored as a digital bit sequence (“0”, “1”), but as an analog voltage curve.

Figure 1: Storage methods for RPM information

A detailed explanation of the advantages and disadvantages of these different storage methods can be found in the section “Reference Quantity/RPM” of the ArtemiS online help.
This Application Note describes peculiarities to be dealt with when storing RPM information in a pulse channel, such as TTL logic, possible error sources in the recording of revolution speeds and missing pulses in RPM recordings.

TTL Logic

HEAD acoustics offers measurement front-ends such as SQuadriga, which provide separate pulse inputs apart from the normal inputs for microphones, acceleration sensors etc. These pulse inputs allow the connection of RPM sensors delivering a TTL-compatible signal. TTL stands for transistor-transistor logic and describes a signal form as shown in figure 2.

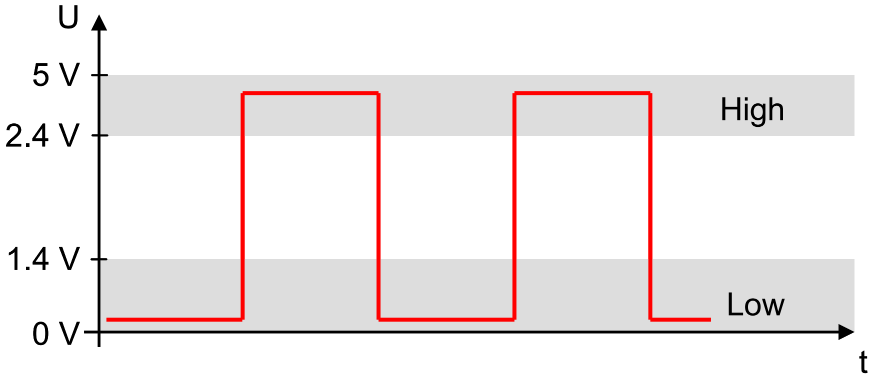


Figure 2: Example of a TTL signal

Such a TTL-compatible pulse signal is stored as a “pulse channel” (see above) that is represented, for example, by a specific bit in an audio signal stream. Only two states can be recorded: “low” (≙ ”0”) and “high“ (≙ “1”). SQuadriga recognizes voltages between 0 and 1.4 V as “low” and voltages between 2.4 V and 5 V as “high”.

These signals are then translated into RPM information according to the settings made in ArtemiS. The possible settings are described in the sections “Reference Quantity/RPM” and “Subchannels and pulses” in the ArtemiS online help.

A TTL-compatible pulse signal is required, for example, by the following front-ends from HEAD acoustics: SQuadriga, Octobox and HMS III/IV. Front-ends like SQlab III and DATaRec 4 are also capable of processing rectangular signals with other voltage limits.

Possible Error Sources in Pulse Data Acquisition

If RPM information is stored as a pulse signal, the current revolution speed must be calculated from this signal. For this purpose, the time interval Δt between two rising edges[1] of the signal is determined (see figure 3), and the current revolution speed is calculated with the following formula:

$$\text{rev. speed [rpm]} = 60 \cdot \frac{1}{\Delta t} \cdot \frac{1}{\text{pulse factor}}$$

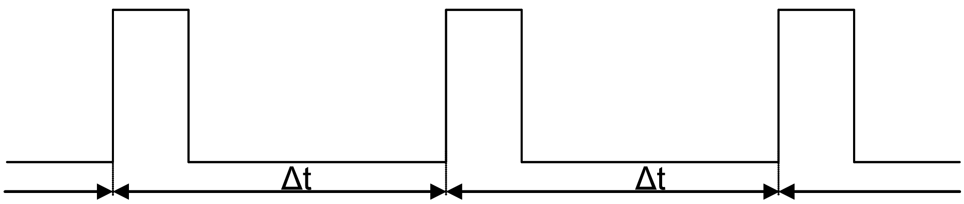


Figure 3: Analog pulse signal

[1] It is also possible to use the falling edge for the calculation, see “Preparation of the reference quantity” in the ArtemiS online help.

The term 1/Δt determines the number of pulses per second (pulse frequency), which is converted into pulses per minute by the factor 60. The value must then be divided by the pulse factor. The pulse factor specifies how many pulses are recorded per revolution. The final result is the current revolution speed with the unit [rpm] (revolutions per minute). Therefore, the more precisely the time interval Δt can be measured, the more accurate the resulting RPM value will be.

However, in digital signal processing, it is not the analog pulse signal that is evaluated, but the discrete values of the signal sampled with a defined sampling rate f_s. When stored as a pulse channel, the sample values are a 1-bit signal that can only have a value of "0" or "1". Figure 4 shows an example of how a pulse signal is encoded in a digital pulse channel.

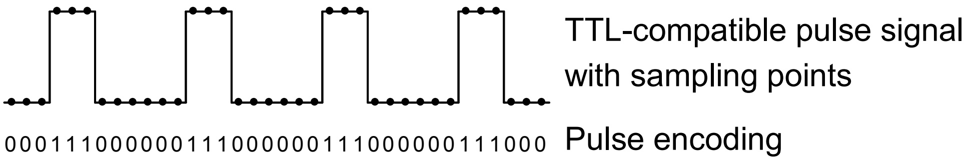


Figure 4: Sampling and encoding of an RPM signal in a pulse channel

Figure 5 shows three examples of sampling a pulse signal with different sampling rates. The sampling positions are marked by the points. The sampling rate decreases from the first to the third example. To determine the pulse period Δt, the number (n) of the sampling points between two successive "0" -> "1" transitions is counted, i.e.

$$\Delta t = n \cdot \frac{1}{f_s}$$

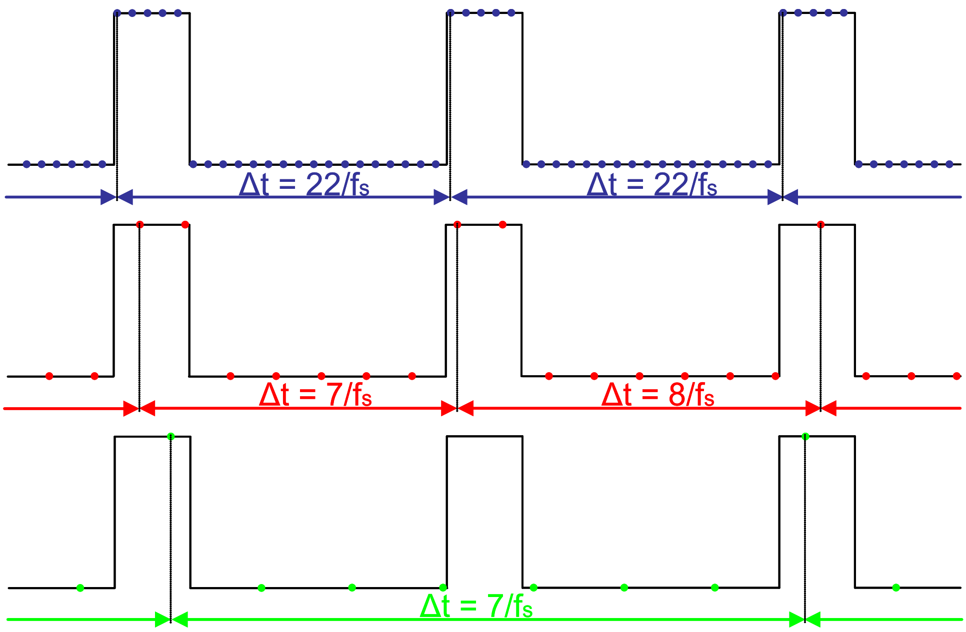


Figure 5: Pulse signal sampled with different sampling rates

It is easy to see that the accuracy of the Δt measurement depends directly on the sampling rate. The lower the sampling rate is compared to the pulse rate, the less accurately the period Δt is determined. If the sampling rate is too low, a wrong RPM value is the result. In the middle ex-

ample of figure 5, Δt is sometimes determined as $7/f_s$ and sometimes as $8/f_s$, even though the actual distance between the rising edges of the analog pulse signal remains constant. The calculated revolution speed varies between two or more values (this is called a "jitter effect"). This error is caused by a too low sampling rate.
In the last example of figure 5, the sampling rate is even lower, resulting in some of the pulses being not even detected and evaluated, causing Δt to become much too high, which in turn would result in a completely wrong value for the revolution speed.

The sampling of the signal leads to a systematic error, because the time of the rising edge in the pulse signal cannot be determined exactly.

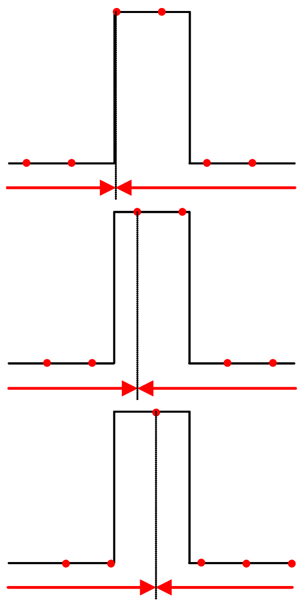


Figure 6 illustrates this systematic error. A pulse is sampled three times with the same sampling rate f_s, but the sampling points are shifted differently in each case. In the first example, the sampling point that marks the "0" -> "1" transition used for determining Δt almost exactly matches the actual rising edge of the analog signal, whereas the point is shifted to later times in the second and the third example. The maximum possible error in locating the rising edge depends on the sampling rate and can be calculated as $\frac{1}{f_s}$.
The lower the sampling rate compared to the pulse frequency, the higher this error becomes. The higher the sampling rate, the lower the jitter effect is.

Figure 6: Sampling of a pulse signal

In addition, it must be made sure that the distance between the sampling points is considerably shorter than the pulse width of the signal. This ensures that all pulses are sampled and none are left out as shown in the third example of figure 5.
In some cases, it can be useful to sample the pulse channel with a higher sampling rate than the audio channels. For example, if only acceleration channels are recorded, for which a sampling rate of 8 kHz is sufficient, the pulse channel should be sampled with a higher rate (oversampling). For example, the DIC6B module of DATaRec 4 allows the pulse inputs to be sampled with 16 or 32 times the sampling rate of the normal audio channels.
Some measurement setups, however, deliver a very high pulse frequency. This is the case, for example, when the pulse signal used for calculating the revolution speed of an engine running at 6000 rpm is sampled at a gear wheel with 360 teeth and the setup is designed so that each tooth generates a pulse. Such a configuration would result in a pulse frequency of 36,000 pulses per second. A sampling rate high enough for this pulse frequency is only possible with special hardware. If such hardware is not available, the measurement setup must be modified so that it delivers a lower pulse frequency.
Another source of error can be a measurement setup delivering a too low pulse frequency. A low pulse frequency means that there are long temporal gaps between the pulses. The long de-

lay between the pulses would make the RPM measurement sluggish and thus inaccurate. This delayed availability of RPM information would also have a negative effect on real-time displays of RPM-dependent analyses, whose results would be delayed as well. Most software applications for recording and analysis therefore specify a lower frequency limit (e.g. 1 Hz in the HEAD Recorder). As soon as the pulse frequency falls below this limit, i.e. the distance of the pulses becomes too high, the software displays a revolution speed of 0 rpm.

Gaps in RPM Acquisition (e.g. Caused by Missing Gear Teeth)

In the automotive area, cases are common where one or two teeth are deliberately missing on a gear wheel, e.g. for marking the top dead center. Acquiring RPM information from such a gear wheel would normally lead to errors in the calculation of the revolution speed. The missing pulses would not be recognized as such, but would be interpreted as a sudden drop of the revolution speed (see figure 7).

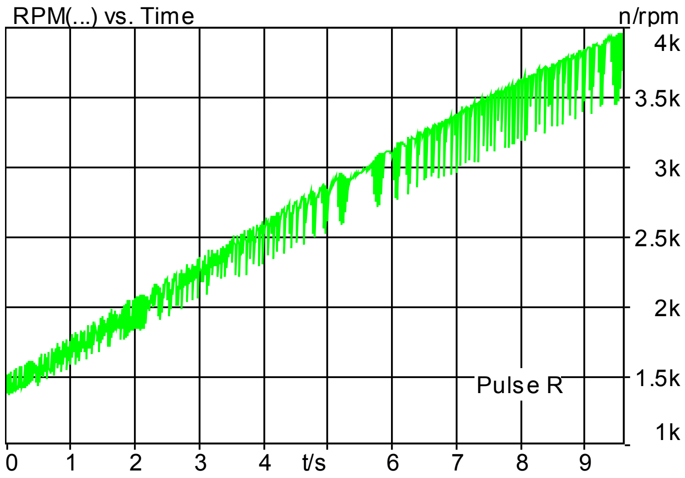


Figure 7: RPM calculation with missing pulses, uncorrected

However, this error can be avoided by means of a suitable parameter selection for the calculation of the current revolution speed in ArtemiS. For this purpose, the “Digital Channel Attributes” tab in the “Data Set Info” dialog of the file must be opened.[2]
In this case, the two fields “pulse factor” and “pulse step” must be matched as follows:
Double-clicking in the “pulse factor” field opens a dialog where the pulse factor and the unit of the reference quantity can be specified. In the field “Impulses per Revolution”, the actual value must be entered, i.e. if the full gear wheel would have 60 teeth, but two teeth are missing to mark the top dead center, the value to be entered is “58”. In addition, the value in the “pulse step” field must be set to “58” as well. That way, the RPM calculation is averaged across 58 pulses and the two missing pulses are not accounted for (see figure 8). Unlike the uncorrected signal, an RPM signal corrected this way can be used very well for calculating RPM-dependent analyses.

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[2] In order to open the „Data Set Info“ dialog, you have to press the [Shift]-key while clicking on the “Dataset Info” button on the Property Page of the time signal.

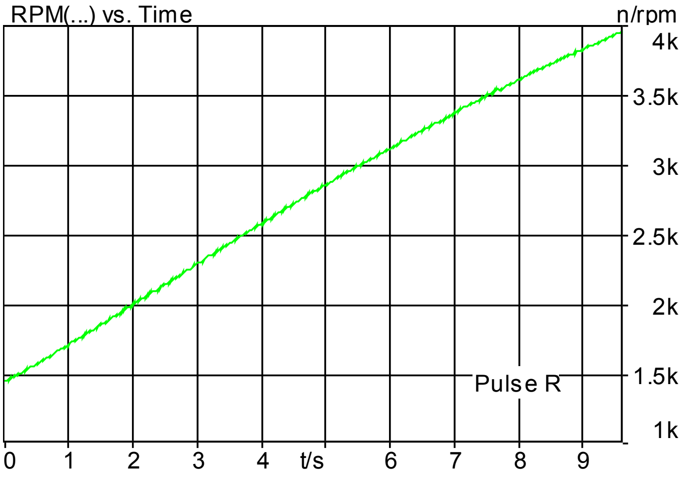

Figure 8: RPM calculation with missing pulses, corrected

Do you have questions for the author? Contact us at imke.hauswirth@head-acoustics.de.
We look forward to your feedback!