

TRACKING FILTER DESIGN FOR PREDICTIVE MAINTENANCE OF JET ENGINE – DIFFERENT APPROACHES AND THEIR COMPARISON

KUNZ Jan¹, ČALA Martin¹, PIKULA Stanislav¹, BENEŠ Petr¹

¹Brno University of Technology, Faculty of Electrical Engineering and Communication, Department of Control and Instrumentation, Technická 3082/12, 616 00 Brno, Czech Republic, e – mail: benesp@feec.vutbr.cz

Abstract: The aim of this paper is to design, implement, test and compare several tracking filtration methods for predictive maintenance of a jet engine. Based on literature review multiple possible methods are described. Appropriate ones selected for further implementation are peak filtration, coherent demodulation, Vold-Kalman filtration and order analysis. Methods are tested to meet criteria set by an aircraft manufacturer and compared using simulated signals.

KEYWORDS: Predictive maintenance, jet engine, tracking filter, Vold-Kalman filter, peak filter, order analysis, coherent demodulation

1 Introduction

Predictive maintenance is a current trend in diagnostic of machinery and equipment. It allows to predict the state of the equipment and schedule maintenance appropriately to reduce costs and increase safety. Accordingly, the predictive maintenance is crucial for jet engines in aviation industry. This approach is common also in other areas, such as automotive (especially in diagnostics of gearboxes) [1]. It is known [2, 3] that a vibration amplitude near the actual frequency is carrying sufficient information about the condition of a rotary equipment, due to observing an imbalance of its shaft.

Therefore, for predictive maintenance purposes of a jet engines it is necessary to monitor an amplitude of a vibration component with the same actual frequency as the engine compressor or turbine shafts rotational speed have and filter out other frequency components and noise. This leads to tracking filtration, which is quite challenging in case of jet engines as they can change the rotation speed very quickly. Several different approaches to achieve this goal can be used. In the past mostly hardware solutions were used, however, nowadays it is possible to use software-based equipment to implement the tracking filtration. It brings several advantages, such as better configurability, easier parameter changes, lower price and higher robustness.

There exist several different software tracking filtration methods with different strengths and weaknesses. Therefore, it is crucial to have a specification of the tracking filter in order to select suitable methods.

2 Filter Specifications

Specifications describe a shape of a bandpass filter with time-varying center frequency f_c and bandwidth B_r (or quality factor Q). The aim of such a filter is to track the wanted frequency component of a vibration signal and remove other disturbing components and

noise. Complete list of filter specifications shows Tab. 1. Since only trend of the tracked component amplitude is needed, one average value for 2 s long block is sufficient filter output. However, the average must be available at least every 0.25 s. Center frequency f_c is calculated from an inductive sensor whose output signal contains a train of pulses with frequency proportional to the f_c . The parameter Q is user-tunable over time, whereas other parameters are fixed or dependent on a machine rotational speed.

Table 1 Filter specifications

Parameter	Value
Center frequency range f_c (Hz)	250 – 1000
Filter quality factor Q (-)	20 – 200
Corresponding relative bandwidth B_r (%)	0.5 – 5
Amplitude error A_{error} at f_c (%)	± 2
Filter bandwidth ratio $-20/-3$ dB $B_{20/3}$ (-)	< 5.3
Filter bandwidth ratio $-30/-3$ dB $B_{30/3}$ (-)	< 8.5
Center frequency accuracy (%)	0.35
Calculation time T_{calc} (s)	2 ± 0.2
Update rate f_{upd} (Hz)	≥ 4

The sampling frequency and maximum computational complexity were not specified directly, therefore both are to be chosen based on expected embedded processor speed. Nevertheless, the less computational demanding algorithm suits better as other task can be performed on the same hardware (HW). For this reason, computational complexity will be estimated for every method, however, it will not reflect software optimization, therefore the real required power may vary.

3 Methods Overview

The aim of the methods is to filter the input signal, therefore considering FIR (finite impulse response) and IIR (infinite impulse response) filters seem obvious. On the other hand, the desired output is not a filtered signal but the amplitude (a signal envelope), therefore a coherent demodulation method should be also considered.

All these aforementioned methods are real-time, so for one input sample they provide one output. However, from the filter specification at section 2 it is visible that not only real-time methods can fulfil this criteria, because a block of signal can be selected and processed at once. Therefore, we can include other methods such as a Vold-Kalman filter or a frequency filtration based on DFT (Discrete Fourier Transform).

All these methods will be briefly described and their pros and cons will be mentioned in following sections.

3.1 FIR filter

These type of filters does not have feedback coefficients, therefore they are always stable [4]. However, for creating a narrow bandpass filter to fulfil the specs (Tab. 1) hundreds of coefficients are needed – $f_c = 250$ Hz and $Q = 200$ (20) yields in 4017 (343) coefficients. The filtration itself is not much computationally demanding as it just requires multiply-and-accumulate operations [5]. However, every filter parameter change yields in recalculation of all coefficients within the update period (less than 0.25 s), which requires a computational power beyond the limit of current embedded processors. Using a bank of filters reduces the computational power needed but requires a lot of fast memory as millions of coefficients need to be stored and accessed quickly. These limitation makes this method inappropriate for this application.

3.2 IIR filter

This filter type represents the digital implementation of analogue filters [4]. On one hand, these filters have feedback coefficients, so stability needs to be checked as they may be unstable. On the other hand, the desired filter shape can be done by less coefficients than FIR filters. For instance, the specs (Tab. 1) can be achieved by 7th to 12th order using standard design method (Chebyshev in this case).

Similarly to the FIR filters also all IIR coefficients need to be recalculated as filter parameters change. However, in this case the recalculation is less demanding because of fewer coefficients, so it could be done on current embedded processors. Nevertheless, the filter needs to settle down to initialize its internal states after every recalculation. The settling time for 1 % error in case of 7th order IIR filter is 14.57 time constants (τ) [6] and for 12th order even longer, so it requires additional computational power. In conclusion, these two requirements makes this method impractical for this application. However, a special type of IIR filters can suit better.

3.3 Peak filter

The principle of this type of IIR filter is similar to a comb filter [7], so it is suitable for narrowband filters. The principle is based on subtracting (adding) original and delayed signals, which can amplify (suppress) the frequency f_c , where the delay is 180° [8]. Unlike the comb filter, which is using time delay, the peak (notch) filter is using all-pass filter to create the delay. This allows to amplify (suppress) only one frequency and not its higher harmonics as in case of the comb filter [7].

The frequency response of the filter is defined by an all-pass filter, therefore we have selected a representation which is defined by a center frequency f_c , a quality factor Q and a gain G (Tab. 2) because these parameters are used for filter specification (Tab. 1). As this representation of the peak filter has the gain G at the frequency f_c , a static gain of the filter needs to be adjusted to fulfil the specification for this application. The representation of the all-pass filter is a second order IIR filter, so very narrow bandpass or a very high attenuation can be achieved only by cascading several filters. To fulfil specified parameters two filters in a cascade are sufficient.

Table 2 Peak filter coefficients with $K = \tan\left(\pi \frac{f_c}{f_s}\right)$ and $V_0 = 10^{\frac{G}{20}}$, where f_c and f_s are center and sampling frequencies (Hz), and G is gain (dB) [7]

b_0	b_1	b_2	a_0	a_1	a_2
$1 + \frac{V_0}{Q}K + K^2$	$2(K^2 - 1)$	$1 - \frac{V_0}{Q}K + K^2$	$1 + \frac{1}{Q}K + K^2$	$2(K^2 - 1)$	$1 - \frac{V_0}{Q}K + K^2$

Moreover, this representation of the all-pass filter has some coefficient not dependent on Q . This means that only some, not all, coefficients need to be recalculated when only one parameter changes. If the change of the parameters is small, we can keep the same internal states of the filter. This produces small error especially during rapid changes of f_c but significantly reduces the required computational power.

In conclusion, this method requires significantly less HW resources than classic IIR filter due to lower order of the filter and recalculation of only some coefficients when one parameter changes. This also reduces the necessity of settling the filter if the change is small. On the other hand, the fixed frequency response which can be only moved (f_c change) or stretched (Q change) does not limit the usage of this method in this application as the filter is defined by those parameters.

3.4 Coherent Demodulation

This method is based on an orthogonality of harmonic functions. This means that the integral (average value) of a product of two harmonic functions with different frequency is zero [1]. However, when the frequencies are equal, the integral is proportional to the amplitudes of those signals and a phase delay between them.

To avoid the phase dependency the input signal can be multiplied by two harmonic signals shifted by 90° (sine and cosine functions) with the same frequency as the measured one. The products, called in-phase and quadrature components, are then filtered by a low-pass filter to obtain the DC values and fulfil required parameters [6]. Then the identity of sine and cosine functions is used to cancel the phase dependence and obtain the envelope of the input signal, which is the desired output for our application.

This method is not computationally demanding as it requires just a harmonic signals generation, low-pass filtering and a few arithmetic operations. This can be easily processed by embedded processors. A disadvantage can be the fact that tracked signal must have zero mean value to work properly.

3.5 Vold-Kalman Filter

There are two generations of the filter but only the newer – called second generation filter (sometimes also called an angular-displacement) – will be considered in the text because its output is directly wanted signal envelope. The filter can track sine component with variable center frequency f_c and bandwidth. Big advantage of zero phase delay is balanced by necessity to calculate only signal blocks rather than ability to process signal samples sequentially over time.

The filter is defined by data and structural equations

$$\mathbf{y} = \mathbf{x}e^{j\Theta} + \boldsymbol{\eta} \quad (1)$$

$$\nabla^2 x_n = x_n - 2x_{n-1} + x_{n-2} = \varepsilon_n \quad (2)$$

The data equation (1) describes measured signal \mathbf{y} as a sum of tracked component envelope \mathbf{x} modulated by rotating phasor Θ and noise $\boldsymbol{\eta}$. The rotating phasor is integral of the tracked frequency f_c with respect to the sampling frequency f_s

$$\Theta = \frac{2\pi}{f_s} \text{cumsum}(\mathbf{f}_c). \quad (3)$$

The structural equation (2) specifies smoothness of adjacent samples within the tracked component. In this case, filter of 2nd order is used. Filter result as output envelope \mathbf{x} is given by minimizing error terms $\boldsymbol{\eta}$ and $\boldsymbol{\varepsilon}$ has following form of a system of equations

$$\mathbf{x} = (\mathbf{A}^T \mathbf{R}^T \mathbf{R} \mathbf{A} + \mathbf{E})^{-1} \mathbf{C}^H \mathbf{y}, \quad (4)$$

where \mathbf{E} stands for identity matrix and powered H is conjugate transpose. Tuning of the filter is done by square matrix \mathbf{R} with weighting factors at main diagonal – higher r leads to narrower bandwidth [9]. Matrix \mathbf{A} contains elements of Eq. (2) along main diagonal and matrix \mathbf{C} contains rotating phasor Θ on main diagonal.

Computational complexity of the Vold-Kalman filter is high because Eq. (4) must be solved as a sparse system of equations, subsequently increased by a processing using 2 s long blocks with 0.25 s shift. With optimized implementation available from [10], calculation time following the specifications can be achieved using currently available HW. The big advantage of VKF is the ability to tune filter parameters arbitrarily inside the block.

3.6 Fourier Transform

Another way of the signal processing using blocks is the Discrete Fourier transform (DFT). This method converts signal from time to frequency domain. When f_c remains constant, obtaining the signal amplitude is an easy task. However, when the frequency is changing within one processed block, then for correct amplitude estimation all spectral lines, where the frequency have appeared, has to be taken into account. It can be achieved with a small error on a signal with slowly changing frequency [11]. This approach leads to a filtration not corresponding to required parameters in Tab. 1. Without the knowledge of a speed of the frequency change this method is inappropriate for our application, especially when fast changes of the frequency are expected. But we can modify the input signal to suit better the DFT algorithm using an order analysis.

Such an approach requires resampling of the input signal according to the frequency profile of the tracked component. Result have the same number of samples per signal period, regardless of the actual frequency. After resampling the signal looks like one with a constant frequency, thus stationary signal suitable for DFT. Therefore, it can be processed by the DFT without any limitations. To fulfil specifications, rectangular window must be used which results in systematic amplitude error of the output. The amplitude is then calculated from selected number of spectral lines given by the quality factor Q .

The computational complexity of this method is quite high as it requires a period search, resampling using interpolation within the period, and DFT. On the other hand, the resampling can be done during the signal acquisition and DFT algorithms are well optimized. For these reasons this method can be performed on embedded HW.

3.7 Summary of the Proposed Methods

From theoretical analysis it is visible that using FIR or IIR filters for tracking filtration is too computationally demanding to be processed on a current embedded HW, therefore these filters will not be tested.

The other four methods can be processed, however, it is important to mention different algorithm complexity, which is summarized in Tab. 3. From real time methods the coherent demodulation is less computational demanding than the peak filtration as the demodulation does not require coefficient recalculation. Furthermore, using peak filter or coherent demodulation allows to compute the amplitude during data acquisition as they are real-time. On the other hand, the Vold-Kalman and the order tracking filtration can be processed only after all data for one block are acquired. This differences has to be taken into account when suitable method is selected.

All four aforementioned methods have been implemented in LabVIEW to be tested. Firstly, we have tested whether the methods are able to fulfil the specs, then we tested the methods on simulated signal to reveal their limitations.

Table 3 Algorithm complexity of implemented tracking filter methods

Method	Algorithm complexity
Peak Filter	N
Coherent demodulation	N
Vold-Kalman	N^2
Order Analysis	$N \log(N)$

4 Designed Filters Parameters

Following graphs in Fig. 1 and 2 show typical frequency responses for all mentioned methods tuned to meet the specifications. The list includes coherent demodulation (in graphs named Coher), order tracking (Order) – DFT with the signal resampling, peak filter (Peak) and Vold-Kalman filter (VKF). Graph in Fig. 1 shows the narrowest settings, Fig. 2 reflects the widest one. All limits for -3 dB, -20 dB and -30 dB, as well as amplitude error of max. 2 % (approximately 0.17 dB) are satisfied.

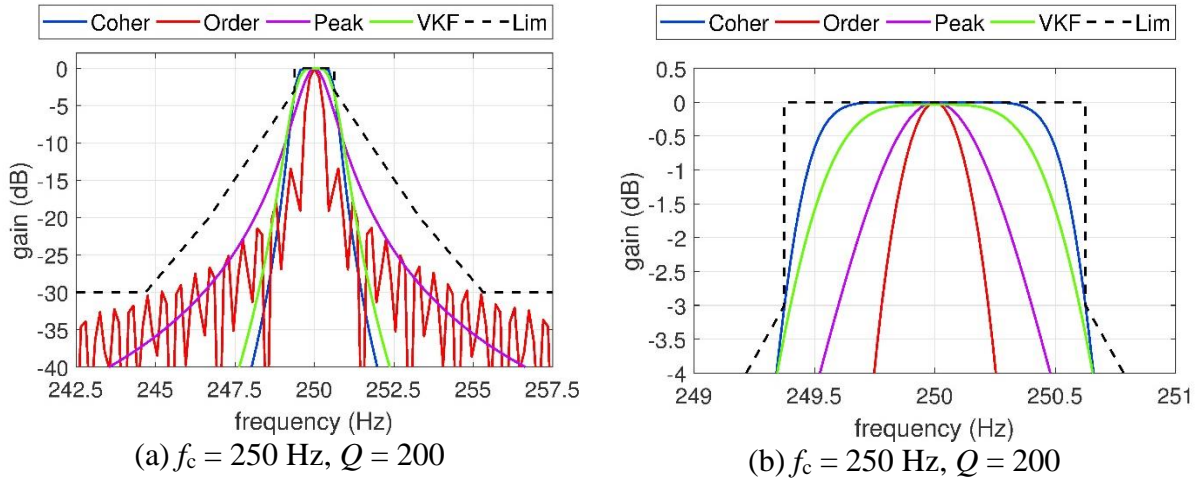


Fig. 1 Frequency responses of all filters for $f_c = 250$ Hz and $Q = 200$: a) an overview, b) detail near -3 dB.

The lower limit of sampling frequency f_s is given by filtered range of 1000 Hz plus one half of the widest bandwidth 1025 Hz. Therefore, minimum f_s must be at least twice as much the value ($f_{s\min} = 2050$ Hz) according to the Nyquist theorem. For purposes of analog anti-aliasing filter, the $f_{s\min}$ must be higher. Additionally, higher limit influences computational complexity and for some cases also stability. Based on all considerations, f_s was set to 20 kHz.

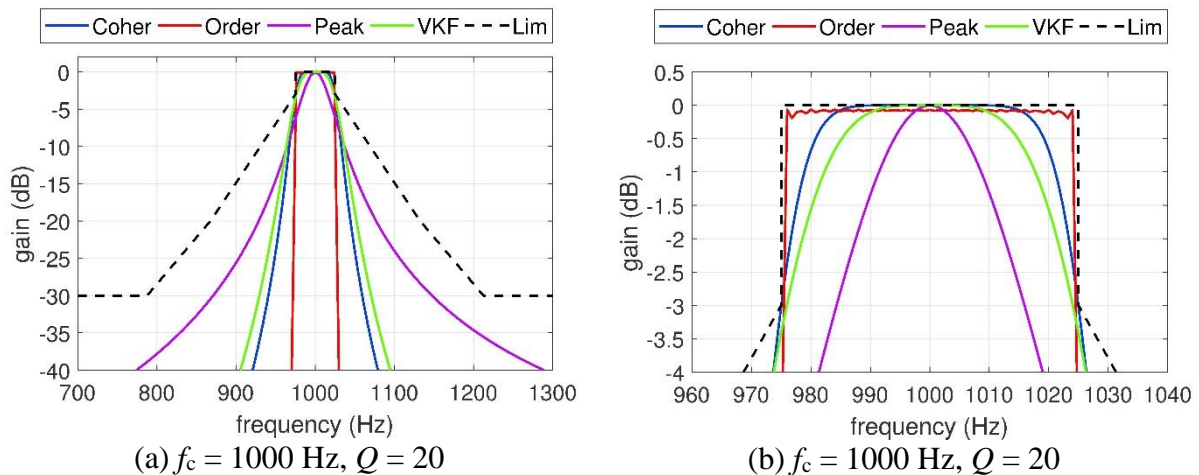


Fig. 2 Frequency responses of all filters for $f_c = 1000$ Hz and $Q = 20$: (a) an overview, (b) detail near -3 dB.

5 Verification by Simulations

Four different scenarios were used for testing the methods to show their behaviour. These scenarios are a simulation of a typical cases in real situations. Therefore, these signals reveal properties and limitations of tested methods which help select a proper one for desired

applications. For a better understanding of the methods behaviour, the results are displayed graphically. For instance, an amplitude step which shows how fast the methods are or a frequency sweep where an error during fast change of f_c can be seen.

Scenario 1 simulates an amplitude change of the signal by a step, therefore in Fig. 3 settling of each filter is visible. The detail 3b shows that all filters are settled sooner than 3 s. The fastest are the order tracking and the Vold-Kalman filter because they do not have a phase delay.

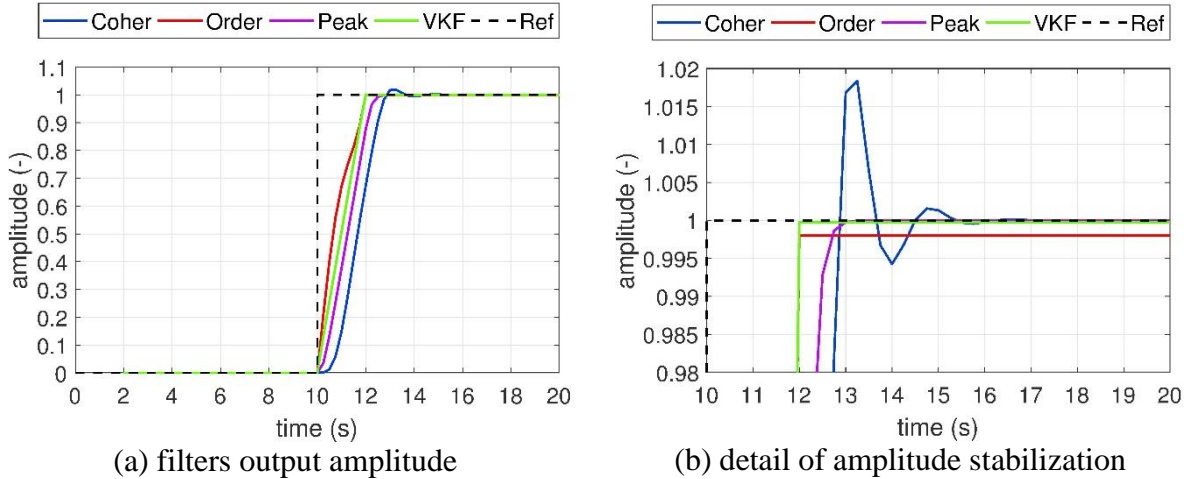


Fig. 3 Response on step amplitude change at $t = 10$ s for $Q = 200$, $f_c = 500$ Hz.

Scenario 2 Fig. 4 simulates jet engine run-up condition from Fig. 4b where rotational speed is rapidly increasing. This shows ability to track correct amplitude of a non-stationary signal which is a key property of tracking filters. Again, methods based on classic filter (coherent demodulation and peak filter) show settling. In addition, peak filter cannot track the amplitude correctly during rapid f_c change because of optimizations explained in section 3.3. A rectangular window used in order tracking causes slight underestimation in some cases (example visible in Fig. 2b), which can be also seen in Fig. 4a. A more narrow setting of filters results in higher errors since settling times become longer.

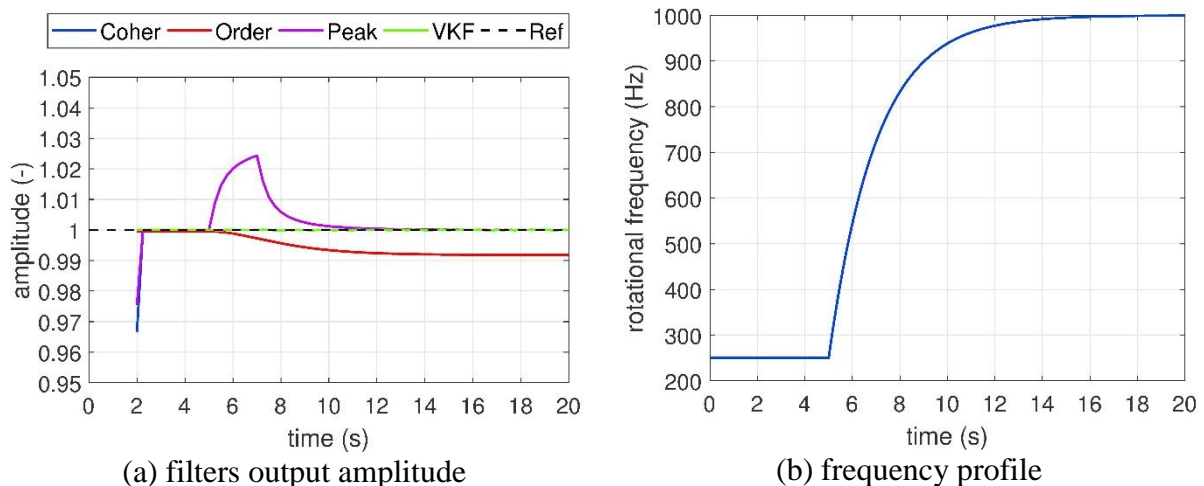


Fig. 4 Response on change of rotational frequency starting at $t = 5$ s with $\tau = 2$ s for $Q = 20$ (jet engine run-up simulation).

Scenario 3 simulates a discrimination of two close harmonics in the signal, typical for a real-world signal containing responses of multiple mechanical parts like gears, bearings, etc. The testing signal is composed of a measured one with frequency f_c and a disturbing signal

with frequency $f_d = 1.02f_c$. The frequencies are so close to each other that the methods with widest settings ($Q = 20$) are not able to separate measured signal from the disturbing one (Fig. 5a). However, when the quality factor is increased ($Q = 200$) all methods are able to separate those signals (Fig. 5b). From the results one can see the importance of a proper parameters settings as it have a significant effect on the results.

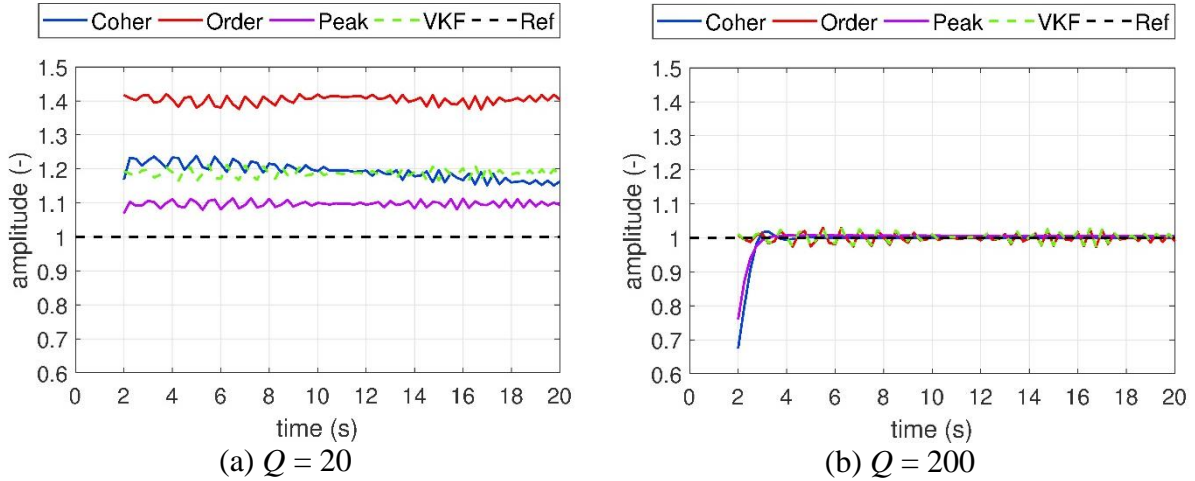


Fig. 5 Filters output for different filter quality factor Q . Signal contains two harmonics with an unitary amplitude. Frequency of first harmonic is linearly changing from 250 to 300 Hz and frequency of the second harmonic is always 2 % higher. (a) the bandwidth is too wide, filters pass both signals; (b) filters are selective enough, second component is filtered out.

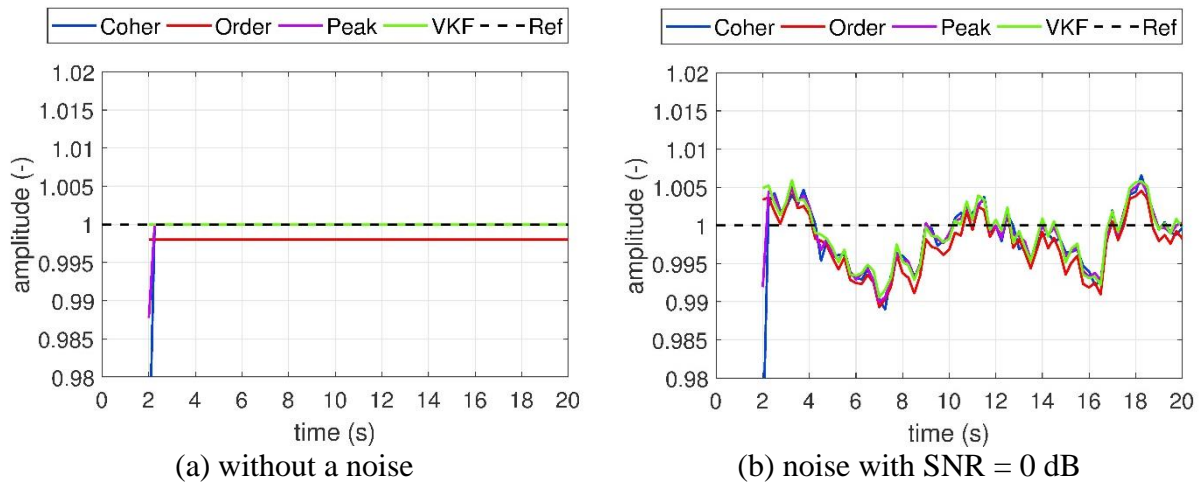


Fig. 6 Impact of a noise level on filters performance, maximum bandwidth selected ($Q = 20$).

Scenario 4 shows resilience of the methods to noise that is present in the signal due to a combustion of a fuel. A constant-frequency of 500 Hz sine signal in Fig. 6a was used to show state without a noise. In comparison with a signal in Fig. 6b, where a strong Gaussian white noise with $SNR = 0$ dB is added, results start to fluctuate. Nevertheless, all methods can handle such a noisy signal within the specified limits ($\pm 2\%$) due to averaging the result for 2 s even when filter selectivity was the lowest. Increasing the selectivity would decrease amount of a passed noise but also would add a more significant distortion due to a longer settling.

Numerical values of a root mean square error (RMSE) defined as

$$\text{RMSE} = \sqrt{\sum_{i=1}^N (o_i - r_i)^2}, \quad (5)$$

where o_i is method output and r_i is reference amplitude. RMSE shows a methods' performance. Values are summarized in Tab. 4, where lower value means more accurate output. VKF performed the lowest figures almost at all scenarios, mainly due to possibility to tune the parameters even within the block and nearly no transitions. On the contrary, coherent demodulation seems to perform the worst, mainly due to settling. In a summary, differences among methods are small, all methods fulfil specifications, therefore are suitable for such an application.

Table 4 Numeric comparison of filters using RMSE criterion.

Scenario	Q (-)	Coher	Order	Peak	VKF
1	20	0.094	0.494	0.034	0.002
	200	0.894	0.336	0.337	0.021
2	20	0.033	0.053	0.065	0.000
	200	0.398	0.053	0.711	0.012
3	20	1.661	3.448	0.833	1.604
	200	0.397	0.128	0.285	0.128
4 no noise SNR = 0 dB	20	0.033	0.016	0.012	0.000
	200	0.398	0.016	0.128	0.002
	20	0.046	0.039	0.035	0.034
	200	0.394	0.045	0.129	0.037

CONCLUSIONS

In this paper four different methods of tracking filtration are proposed and tested to select a proper one for a predictive maintenance of a jet engine. Two of the methods are on-line, namely a coherent demodulation and a peak filtration, so they process the data sample by sample. Whereas the other two methods, a Vold-Kalman and an order tracking filtration, are processing a block of the data at once.

The methods have been implemented in LabVIEW and tested on different signals to reveal their limitations. Firstly, frequency responses (Fig. 1 and 2) for each method and different settings were generated to check, whether the methods are able to meet the criteria defined by an aircraft manufacturer (Tab. 1). Then, the methods were tested by four different scenarios which were designed to simulate real working conditions. The root mean square error (RMSE) was used to compare quantitatively the accuracy of the methods and results are presented in Tab. 4. In addition, the behaviour of the methods under different working conditions were presented also graphically.

Results show that all four methods fulfil the criteria, however, there are differences. The offline methods does not have a phase delay so they do not need to settle, whereas the on-line methods need some time, defined by the filter, to settle down when a significant change in the input signal occurs. On the other hand, the online methods provide the output immediately, which can be crucial for some applications. Furthermore, as the peak filter does not reinitialize its inner states when parameters are changed, the rapid change in frequency causes an error (Fig. 4), but allows much less computational power. From the same figure it is visible that the order tracking method can have small constant error due to the frequency response of used window as was mentioned in sections 3.6 and 5 and is also visible in Fig. 2.

Despite minor differences in results, all methods are able, when set properly, to filter the higher harmonics out of the measured signal (Fig. 5) and also suppress noise (Fig. 6). One can see from RMSE results that the Vold-Kalman is the best method almost in all cases. However, it is the most computationally demanding one (Tab. 3) as it has to solve the same number of equations as the number of samples in the signal.

The other three methods are quite comparable, however from the data it is visible that the order tracking method is the best for the narrowest filters ($Q = 200$). It results from the fact that this method does not have a transition band, whereas the on-line methods have. Furthermore, the coherent demodulation is the most vulnerable to noise.

Every method has its pros and cons therefore no one can say that one method is the best for all applications. Finally a method has to be selected according to specific requirements for a desired application. In our case, we preferred the computationally least demanding algorithm. Among considered methods, it is valid for coherent demodulation and peak filter. The first one can handle fast frequency changes better, as it is common in the jet engines. In conclusion the coherent demodulation was selected as it suits the best to our application.

Acknowledgment

Authors gratefully acknowledge financial support from the Ministry of Industry and Trade of the Czech Republic under project TRIO FV20043 Advanced Technology of Modular Control and Diagnostics Systems for Aircraft Engines.

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