Some aspects of digital processing of photoacoustic signals

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ABSTRACT

The paper presents some aspects of digital signal processing of sampled photoacoustic signal. In particular influence of DC drift and noise on photoacoustic signal amplitude detection errors is discussed. Some methods of dealing with these problems are described. Real-time sampling and stroboscope sampling are considered.

Keywords: photoacoustics, data acquisition, digital signal processing, amplitude detection errors

1. INTRODUCTION

In typical photoacoustic setups, in which light beam of modulated intensity irradiates an investigated sample, the induced photoacoustic signal has a few typical features. The most important one is that values of the frequency and phase of the modulating signal, and, as a result, frequency and phase of the photoacoustic signal are precisely known. The second is that the photoacoustic signal is usually of small amplitude and very often, especially in case of weak absorption of the light by the investigated sample, level of the photoacoustic signal is below the background noise level.

2. AMPLITUDE DETECTION IN PHOTOACOUSTIC SYSTEMS

According to the modulation method, photoacoustic setups can be divided into continuous modulation and pulsed photoacoustic systems. In case of continuous modulation, light intensity is usually a sine or a square wave. The simplest method of getting the light continuously modulated is to use a mechanical chopper. In the systems with continuous modulation , the photoacoustic response is a periodical signal - usually similar to sine wave - with the frequency equal to the light beam modulation frequency. In such a case, the most common method used in order to get the signal out of noise is narrowband filtering (usually the bandwidth of the filter does not exceed 1% of the modulation frequency) together with synchronous detection and averaging (which in some cases can last minutes or longer - in order to increase overall sensitivity of the system).[1] Typical traditional equipment used for the purpose are lock-in amplifiers or box-car integrators. It should be noted, however, that lock-in amplifier may be used only for amplitude and / or phase measurements (as a reference signal usually the light modulation signal is used), while it is not capable of retrieving shape of the signal. For such a purpose a box-car integrator can be used, although this may be a little bit troublesome, because in order to measure momentarily values of the signal at N points, N measurements are required, and for every one of them phase difference between the reference signal and moment of opening the integrator's gate must be different.

In pulsed photoacoustic systems usually pulse lasers are used as the light sources. Due to high level of the pulse energy, acoustic response is usually strong enough to be observed e.g. by means of oscilloscopes or signal recorders. To be precise, it should be stated, that time delay between consecutive laser pulses may fluctuate, and the resulting photoacoustic signal should be considered as repeatable rather than periodical. However this should not become a problem, as corresponding samples of several recordings of such a signal can still be averaged in order to increase signal-to-noise ratio.

It should be mentioned also, that pulse modulation method has some advantages over continuous modulation - e.g. it allows to cancel background photoacoustic effects (resulting from absorption of the light pulse by the cell windows or walls, etc.) if appropriate time-domain signal processing of the recorded signal is performed. The main requirement is that

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the cell must be designed in such a way, that delay of the acoustic waves resulting from the background absorptions referred to the light pulse is different than the delay of the acoustic wave produced by the absorption of the light by the sample.[2]

Due to some unknown reasons, multipoint averaging method is very rarely used in photoacoustic experiments, although applications of this method in photoacoustics and its efficiency was reported quite long ago.[3] It should be noted, that multipoint averaging circuits sample the measured signal number of times during its every period and then perform averaging of corresponding samples from every recorded period. This means that multipoint averaging circuits record shape of the signal (not only its amplitude and / or phase), which make them particularly suitable for pulsed photoacoustic experiments, although they can be used in continuous modulation photoacoustic setups as well. Further discussion in this paper is carried out with the assumption that the signal is being recorded by means of a multipoint averaging circuit in a photoacoustic system working with continuous light modulation.

Once shape of the signal is recorded, it can be used for evaluation of its amplitude (which is usually considered as amplitude of the sine component of the signal which frequency is equal to the light modulation frequency). In most of photoacoustic papers the authors use Fast Fourier Transform for this purpose.[4,5] However, in the discussed applications FFT has number of disadvantages - the most important are: broadening of the spectrum lines, inaccuracy of amplitude calculation (especially in case of small number of samples per period, significant level of background noise and DC drift - which are typical conditions in photoacoustics), and high computational requirements. From this point of view the least squares method seems to be much better for the discussed application. It should be noticed, that in case of the least squares method, digital filtering of the signal performed before amplitude calculation does not improve the final results, but can be of value if its aim is e.g. smoothing the signal shape before having it displayed to the user.

3. PROBLEMS REGARDING MULTIPOINT AMPLITUDE DETECTION

Taking into consideration that photoacoustics is often used e.g. in trace detection, the signal is usually observed simultaneously with a strong noise background. Certainly, it is possible to increase signal-to-noise ratio by means of averaging and digital filtering, but in case of the measurements performed at the levels close to sensitivity limits of the setup, high level of noise cannot be avoided. The noise can be considered as a superposition of high frequency noise (responsible for random oscillations of the samples around the true signal value) and low frequency noise (DC drift, that can be noticed even within a single period of the recorded signal). Examples of such a signal are showed at fig. 1a,b.

Additional problem arises in case of stroboscope sampling in which during every period of the signal every N-th (where $N > 1$) sample of the reconstructed signal is taken. Due to the mentioned DC drift, noticeable even within a single period of the signal being recorded, the reconstructed signal shows visible steps between consecutive samples (fig. 1c,d).

It is of no doubt that the mentioned noise effects may seriously affect both - reconstruction of the true shape of the signal and detection of the amplitude and / or phase of the signal.

4. PROPOSED METHODS OF SIGNAL PROCESSING

4.1. DC drift cancelling

It is quite obvious that DC drift can significantly increase errors of multipoint amplitude detection of the sampled signal. In order to, at least in part, compensate this effect (and its influence on the amplitude detection errors) a linear approximation of the DC drift can be applied. Such a method assumes that the drift value equals to the difference between the first and the last sample of the recorded signal period. In the worst case - if the mentioned first and last points are close to the zero-crossing moments of the signal, at which the signal is changing with the speed of ($d \sin x / dx$) close to one - error resulting from the method is $1 / N$, where N is number of samples recorded per period of the measured signal (assuming that amplitude and period of the signal are normalized and equal 1.0). It should be noticed, however, that if the frequency response is known or if some preliminary calibration measurements with high level of the signal were performed, the phase of the signal being recorded can be precisely evaluated and working point of the multipoint averager can be selected in such a way, that first and last point of the recorded signal will be at maximum or minimum of the signal value, at which ($d \sin x / dx$) is zero, thus minimizing the error resulting from the method. In order to decrease influence of the high-frequency noise on the calculated DC drift value, it is possible to calculate the drift value from the averaged values of a few

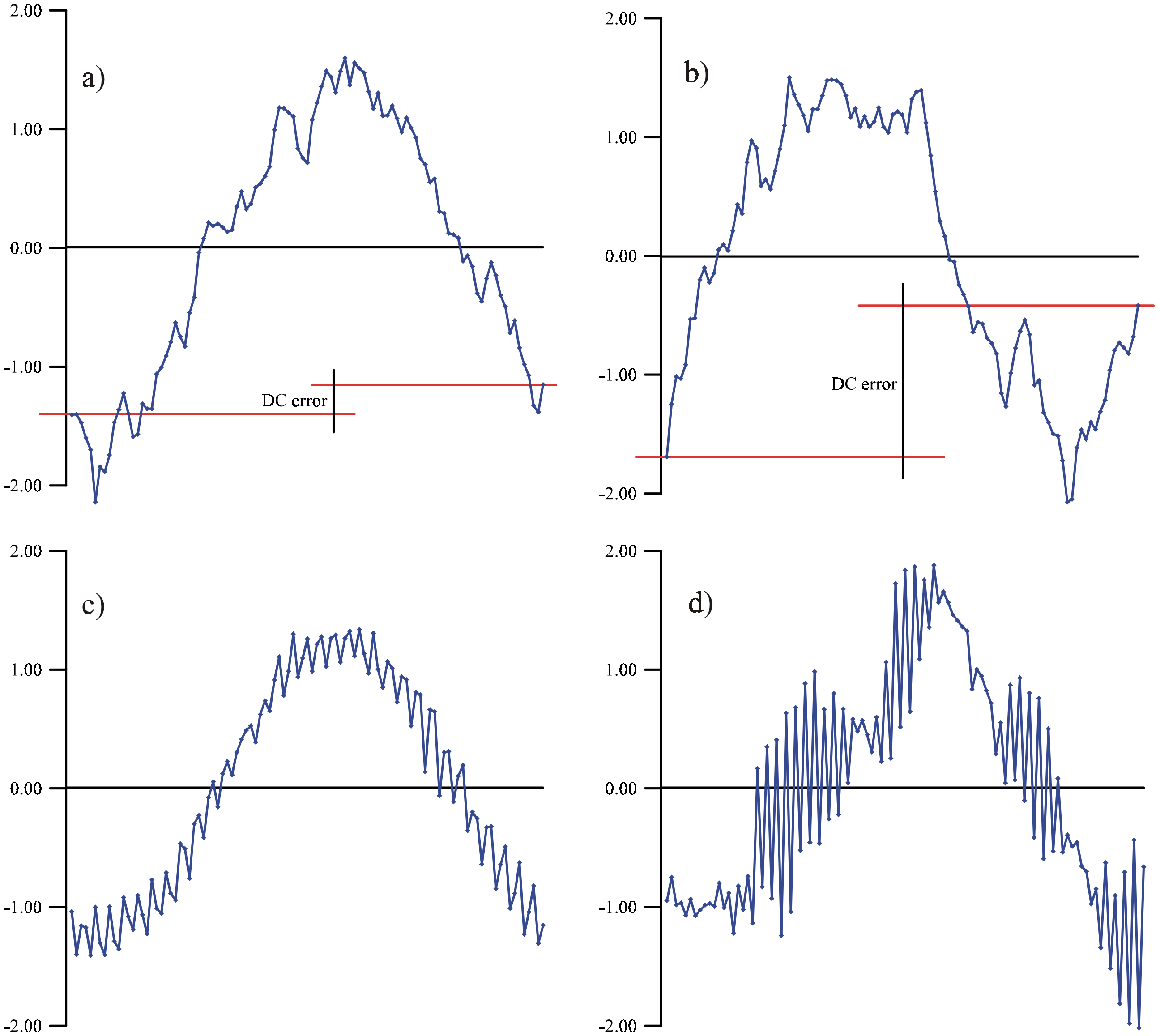

Fig. 1. Examples of some measured photoacoustic signals: a) a signal with relatively small DC drift, b) with significant DC drift, c) with stroboscope effect at relatively small DC drift, d) with stroboscope effect at higher level of DC drift.

samples taken at the beginning and at the end of the recorded signal period. Number of samples taken into such an averaging operation should be selected accordingly to the total number of samples per period. Once the DC drift between the first and the last sample of the recorded period is known, and assuming its linearity, the drift can be canceled by recalculating values of all the samples as:

$$s_{i\ (new)} = s_{i\ (old)} - \frac{i * DC}{N}, \qquad \textbf{(1)}$$

where:

s_i - i-th sample value,

DC - DC drift between the first and last sample,

N - number of samples per period.

The above algorithm implements whole-period DC drift compensation, but having precise knowledge about zero-crossing points of the signal it is possible to implement similar DC drift compensation for every half of the period separately. It should be noticed, however, that around zero-crossing points value of (*d sin x / dx*) is close to one, hence it may turn out (especially at small number of samples per period) that gain from such a half-period DC drift compensation is smaller than errors resulting from the method.

4.2. Stroboscope effect removal

The mentioned DC drift, which in case of stroboscope signal recording gives result of sudden steps between the consecutive samples, can be easily removed by means of a moving-average filtering in which number of the samples being averaged is N if signal recording was implemented as sampling every N-th sample of the reconstructed signal. It should be noticed that in case of stroboscope signal recording, multipoint amplitude detection should be preceded by moving-average filtering, otherwise amplitude detection errors can be much higher.

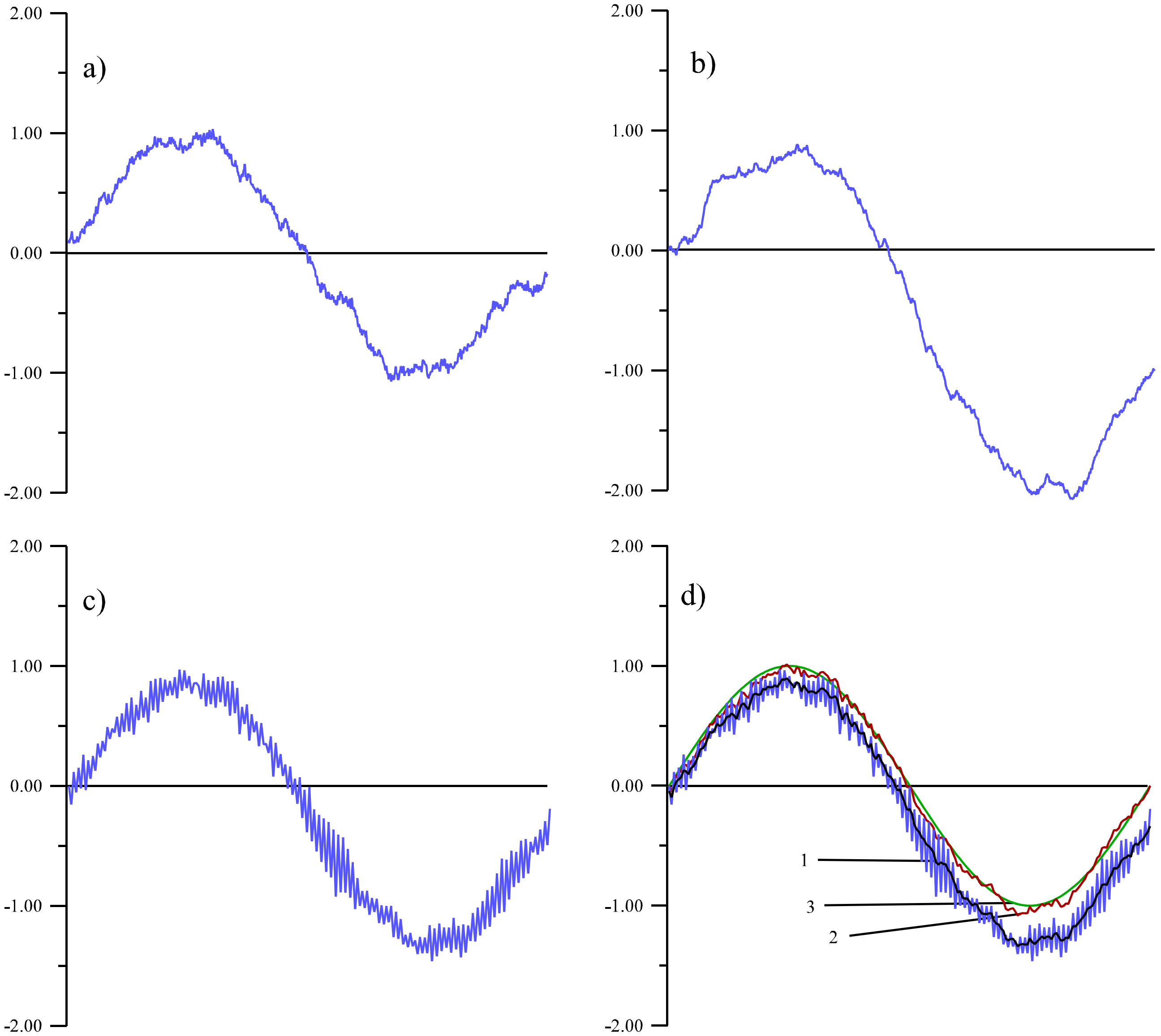

Fig. 2. Examples of the simulated photoacoustic signals: a) with small DC drift, b) with stronger DC drift, c) with stroboscope effect, d) signal from (c) after passing through a moving-average filter (1) and DC drift cancelling (2), and compared with the basic sine wave used as a start point of all the simulations (3).

4.3. High frequency noise reduction

High frequency noise can be removed by means of digital filtering e.g. in a low-pass FIR or IIR filter. However, taking into consideration that influence of such a noise on the least squares method is negligible, such a digital filtering may be useful only if the sampled photoacoustic signal is to be visualized.

5. SIMULATIONS

In order to evaluate influence of the described effects on multipoint amplitude detection errors a method of computer simulations was used. To simplify the problem, it was assumed that a pure sine wave of a well-known amplitude will be used as the noise-free photoacoustic signal. In order to minimize errors resulting from limited number of samples, horizontal resolution was set to 1000 samples a period. High frequency noise component of the test signals was simulated by random values from (-0.5, 0.5) range multiplied by a k_H coefficient (telling about the level of the noise referred to the sine wave amplitude). DC drift was simulated in a little bit different manner. At the beginning of the signal period the drift was assumed to be zero. Then for every consecutive sample, the DC offset for a given sample was calculated as the DC offset value for the previous sample, to which a random value of uniform distribution from (-0.5, 0.5) range multiplied by another k_L coefficient was added. Similar technique was used in order to produce artificial stroboscope sampling signal. Stroboscope sampling effect was simulated by generation of the signal in *N* passes, so that in every pass every *N*-th sample was produced (starting the passes from sample number 1, 2, ..., *N*-1). In such a way DC offset was gradually changing between every *N*-th sample, while showing rapid steps between consecutive samples (exactly like in real stroboscope sampled signals). In the described simulation method it was possible to adjust smoothly strength of both noise components (by means of k_L and k_H), while having precise information about noise-free signal component. Examples of the simulated signals are given at fig. 2a,b,c, while an exemplary result of proposed digital processing of the photoacoustic signal referred to the noise-free sine wave is presented at fig. 2d.

Table 1. Averaged errors (%) of multipoint amplitude detection for different options of photoacoustic signal processing vs. level of high-frequency k_H and low frequency k_L noise.

k_H	k_L	signal + noise	DC drift removal		filter
			whole period	half period	
0,01	0,01	1,41	0,85	0,96	1,42
0,032	0,01	1,44	0,86	0,96	1,44
0,1	0,01	1,42	1,11	1,13	1,42
0,3	0,01	1,45	2,53	2,17	1,45
1	0,01	1,77	7,53	6,32	1,78
3	0,01	3,53	23,45	19,33	3,53
0,01	0,032	4,55	2,6	2,64	4,56
0,032	0,032	4,59	2,63	2,65	4,6
0,1	0,032	4,5	2,67	2,68	4,51
0,3	0,032	4,5	3,53	3,26	4,51
1	0,032	4,57	7,89	6,71	4,57
3	0,032	5,54	23,86	19,65	5,55
0,01	0,1	14,1	8,15	8,14	14,11
0,032	0,1	13,94	8,03	8,02	14
0,1	0,1	13,92	8,2	8,23	13,93
0,3	0,1	14,07	8,39	8,29	14,09
1	0,1	14,31	11,23	10,38	14,33
3	0,1	14,2	24,6	20,66	14,22
0,01	0,3	41,22	25,79	25,86	41,28
0,032	0,3	41	25,53	25,5	41
0,1	0,3	40,79	25,55	25,54	40,84
0,3	0,3	41,89	25,44	25,41	42
1	0,3	41,66	26,71	26,53	41,7
3	0,3	41,7	33,5	31,2	41,7

In order to evaluate amplitude detection errors resulting from the introduced noise components, for given combination of k_L and k_H number of simulations was performed. Every simulation was started by generation of a sampled signal with the method described above. Then the same produced test signal was subjected to number of operations including whole-period drift cancelling, half-period drift cancelling, and digital filtering (narrowband bandpass), and to the signals resulting from these operations (as well as to the test signal) multipoint amplitude detection by means of the least squares method was applied. Thereafter absolute values of relative errors of the amplitude evaluations were calculated by referring the obtained amplitude values to the known amplitude of the pure sine wave that was used as the basic noise-free photoacoustic signal in all simulations. In such a way every single simulation resulted in four unsigned values of relative errors. For every combination of k_L and k_H values of these errors were averaged in a series of 5000 simulations. Values of such averaged errors for different values of k_L and k_H are given in table 1. Analysis of these values indicates that in most cases use of the whole-period DC drift cancelling significantly reduces amplitude detection errors. Only combinations of high level of high-frequency noise with simultaneous absence (or small level) of low-frequency noise (causing DC drift), which is very unlikely (if not impossible) in photoacoustic measurements, shows increase of the errors after DC drift cancelling. It should be noticed that signals processed with half-period DC drift cancelling and whole-period DC drift cancelling have very similar level of amplitude detection errors, while the half-period DC drift cancelling requires more calculations and, especially at small horizontal sampling resolution, may introduce higher level of errors resulting from the method. Digital narrowband band-pass filtering does not show any amplitude detection error improvement.

6. CONCLUSIONS

Preliminary discussion of influence of noise on multipoint amplitude detection of photoacoustic signals showed that especially low-frequency noise, resulting in DC drift of the recorded signal, has severe impact on amplitude detection errors. This influence can be significantly reduced by means of the whole-period DC drift cancelling method presented in the paper. Low-pass digital filtering of the recorded signal does not decrease amplitude detection errors, and may be of value only in case of visualization of the recorded signal.

7. REFERENCES

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