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AN EFFICIENT ALGORITHM TO REMOVE LOW FREQUENCY DOPPLER SIGNALS IN DIGITAL DOPPLER SYSTEMS

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In color flow imaging, a high flow map rate in combination with a reasonable width of the map and good velocity resolution is essential to properly appreciate the time-dependent phenomena. The velocity resolution depends on the length of the signal segment considered in combination with the settling time of the high pass filter used to eliminate transients and low frequency artifacts. The latter can be reduced by appropriate processing. This paper presents an algorithm to suppress low frequency Doppler signals effectively and efficiently, while all the data points within the segment considered contribute equally to the average Doppler frequency computed. The algorithm is applied to computer generated Doppler signals to evaluate their time and frequency behavior. It is concluded that the proposed scheme functions adequately under various signal conditions.

Key words: Color flow mapper, high pass filter, pulsed Doppler, regression filter, signal processing, ultrasound.

INTRODUCTION

To generate a flow map a sequence of short ultrasound bursts is emitted at the pulse repetition frequency (PRF) along a given line of observation. The frequency content of the Doppler signals acquired is analyzed both as a function of depth and time to obtain the average frequency within the time window of observation over the range of investigation. Subsequently the ultrasound beam is moved to the next line until all the lines comprising the flow map are available. Since the Doppler signals assessed at successive sample volumes spaced at equal depth intervals along the ultrasound beam will be processed by the same circuitry (serial data processing) the discussion about the method of processing can be restricted to a single sample volume without losing generality. The maximum frame rate equals PRF/(N*L), where N is the number of successive bursts and L the number of lines on which the

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flow map is based. A high frame rate favors a smooth transition of subsequent flow maps, which is of particular importance in the evaluation of fast changing flow phenomena, such as within the cardiac cavities. The PRF is dictated by the required depth of interrogation, so the only way to manipulate the rate at which flow maps become available is through N and/or L. While the PRF governs the length (and position) of the field of interest, the number of lines L defines its width, leaving only N for optimization for a given size and position of the flow map. Typically for a frame rate of 8, acquired with a PRF of 4 kHz over 32 lines, N should be less than or equal to 16. This would not impose any problem if the Doppler signals were free of high amplitude low frequency artifacts due to the presence of slowly moving acoustic interfaces with a high echogenicity within a sample volume. To avoid obscuring the low amplitude Doppler signals induced by the flowing blood by high amplitude low frequency signals, high pass filtering (applied to the quadrature Doppler signal) is employed. This so-called vessel wall filter will exhibit a response time that increases as the cut-off frequency decreases with respect to the PRF. The longer the response time (expressed in sample points) the longer it will take before the output of the filter will settle. Each time the direction and/or position of the line of observation is changed transients will occur affecting the processing. Of course, it is possible to shorten the response time by selecting a higher cut-off frequency, but this will reduce the range over which velocities can be assessed (the upper bound depends on the sampling frequency, i.e., the PRF). The quadrature output of the highpass filters is fed to a frequency estimator, generally based on the complex autocorrelation function with a time lag of one sample point.

The estimate for the average Doppler frequency $\bar{f}$, expressed as a fraction of the PRF, within a time window of M points, is given as the arctangent of the imaginary and real part of the autocorrelation function [1,2]:

$$\bar{f} = \frac{1}{2\pi} \arctan\{Im(C(1))/Re(C(1))\} \qquad (1)$$

with $$Im(C(1)) = \sum_{i=1}^{M-1} (x[i]y[i+1] - x[i+1]y[i])$$

and $$Re(C(1)) = \sum_{i=1}^{M-1} (x[i]x[i+1] + y[i]y[i+1])$$

where x and y denotes the in phase and quadrature component of the Doppler signal, respectively. Assuming stationary signal conditions, the variance of the estimate is inversely proportional to M-1. The method mentioned above is rather insensitive to noise while aliasing will occur only if the absolute value of the average frequency exceeds PRF/2. To reduce the effect of response time, M will be generally less than N. It should be acknowledged that the frequency estimator only produces valid results after the transient effects are eliminated from its input.

It is clear from above that the response time of the highpass filter plays a crucial rule in the assessment of the average Doppler frequency. To demonstrate its contribution let us assume a first order recursive digital filter having an infinite impulse response (IIR filter) with a filter constant of k=1/8, while at time t=0 a sudden

change in input amplitude of A=2047 occurs (full scale step for a 12-bits input). The transfer function of this filter can be expressed as:

$$H(z) = (1 - z^{-1})/\{1 - (1-k)z^{-1}\} \qquad (2)$$

where $$z = \exp(j2\pi f/PRF)$$

while its cut-off frequency can be approximated by PRF*k/6 [3]. Then the output will gradually decay as $A(1-k)^t$ (t=0, 1, ...). Setting the output value to A/e gives the response time of the filter expressed in sample points (t=7.5 or approximately 1/k), while setting the output to one gives the time required to eliminate the effect of the full scale step (t=57). It is obvious that a settling time of 57 sample points for a full scale step is far too long compared to the time available. The settling time can be reduced by selecting a higher filter constant, thereby accepting a higher cut-off frequency. Setting k to 1 transforms the recursive filter into a stationary echo-cancellor [4]. The settling time is then only one sample point while the cut-off frequency is approximately PRF/4. Another possibility is offered by digital filters with a finite impulse response (FIR filter). For such a filter the settling time is identical to the length M of the impulse response used. Moreover, with a FIR filter a steeper roll-off can be obtained. Generally, a lower cut-off frequency and/or a sharper roll-off will lengthen the impulse response. But also the FIR filter has the disadvantage that the first M output samples have to be skipped for frequency estimation, since these samples still contain (low frequency) artifacts, prolonging the time required to assess the average Doppler frequency. Presently, autoregressive techniques [5] do not provide a viable alternative, because the number of sample points only permits models with a low order, while a quite high order is required to model the narrow spectral peak at the zero frequency.

The present paper will discuss an algorithm to keep the time required to obtain a valid result for the output of the frequency estimator as low as possible. To achieve this another filter type is proposed that efficiently eliminates high-amplitude low frequency Doppler signals, while it is unsusceptible to transient effects, i.e. all the sample points within the time window of observation contribute equally to the frequency estimate. Computer generated Doppler signals are used to demonstrate its characteristics. The results are compared with those obtained employing a cascade of two IIR filters of the first order, as introduced above.

FILTER ALGORITHM

The signal window considered to estimate the average frequency will generally have a length of the order of a few milliseconds. For such a window, low frequency signals will approach a straight line. Therefore, the low frequency Doppler signals can be eliminated by fitting a regression line to the data points within the window and considering only the deviations between the local value of the line and the data. This approach can be followed for both the in phase and quadrature component of the Doppler signal. Denoting the expression for the regression line (for the in phase component) as:

$$l[i] = m_x + id_x \qquad i = -(N-1)/2,..,0,..,(N-1)/2 \quad (N \text{ odd}) \qquad (3)$$

then the input for the frequency estimator will be x[i]-l[i]. In the above expression m_x is the mean of the N sample values within the window and d_x is the direction coefficient. The latter may be derived using the classical least square error approach. However, the objectives of the procedure is to suppress the low frequency components end to pass rapid fluctuations unaltered. This can be achieved by the requirement that the sum of the deviations for a negative index should equal the sum of the deviations for a positive index, i.e.:

$$\sum_{i=1}^{(N-1)/2} \{x[-i] - (m_x - d_x i)\} = \sum_{i=1}^{(N-1)/2} \{x[i] - (m_x + d_x i)\}$$

which gives:

$$d_x = 4 * \sum_{i=1}^{(N-1)/2} (x[i] - x[-i])/(N^2 - 1) \qquad (4)$$

The same procedure can be followed for the quadrature component. A drawback of the above mentioned procedure is that the parameters of the regression line can only be estimated after all the data within the window are available, requiring temporary storage of data.

The cut-off frequency of the regression filter, as introduced above, is related to the length of the window: the longer the window, expressed in number of sample points, the lower the cut-off frequency will be. It can be anticipated that the regression filter will exhibit a second order frequency characteristic. The transfer function of the regression filter will be presented in the Results section together with the transfer function of a cascade of two IIR filters of the first order.

SIGNAL SIMULATION

The output of the regression filter depends on the frequency as well as the phase of the input signal. For a low frequency signal the best fit will be obtained if the phase of the input signal equals zero in the center of the window of observation. This can only be true for one of the quadrature components, affecting the amplitude and (slightly) the phase of the filtered Doppler signal. For wideband signals the error should be negligible. Moreover, the main objective of the regression filter is to eliminate the influence of transient and low-frequency artifacts on the average Doppler frequency rather than to obtain a correctly filtered signal. The frequency estimator should extract the mean frequency of the true Doppler signal irrespective of the level of low-frequency (stationary) Doppler signals and of background noise.

To evaluate the behavior of the regression filter (in combination with the correlation estimator), a quadrature Doppler signal s[i], consisting of 8000 points and containing a stationary signal v[i], a true Doppler signal d[i] and a noise component n[i] were generated on a computer. The true Doppler signal had a rectangular frequency distribution over a specified frequency range. Since the true Doppler signal can be easily shifted in the time domain to any arbitrary center frequency the only parameter of importance for generation is the frequency range. Therefore, the Doppler signal considered can be expressed as:

$$s[i] = A_v\, v[i] + A_d\, d[i] \exp(2\pi j \bar{f}_d i) + A_n\, n[i] \qquad (5)$$

where A_v is the amplitude of the stationary or vessel wall signal, A_d the amplitude of the Doppler signal and A_n the amplitude of the background noise. The frequency f_d, expressed as a fraction of the PRF, is the center frequency of the true Doppler signal and the parameter to be derived by the estimation procedure. For all simulations, the frequency range of the vessel wall signal was maintained at 0.01 of the PRF while its average frequency was set at zero. The input data of the filter are presented on a 12 bit scale (amplitude ranging from -2048 to 2047) offering a dynamic range (72 dB) equivalent to the dynamic range observed in Doppler systems. The 0 dB level corresponds to an RMS amplitude of 1024. In all simulations the true Doppler signal had an amplitude of -40 dB while the white noise component had an amplitude of -60 dB.

Both the input and output of the filters are in integers. To reduce truncation errors the intermediate filter results have to be presented more accurately. Therefore, the coefficient of direction of the regression line is presented with a 3-bit mantissa while for the recursive filter a mantissa corresponding to the filter constant (e.g., for a filter constant k=1/8, a 3-bit extension) is used. Of course, the arctangent conversion is executed in floating point. The results of the signal processing are qualified using the mean bias and the RMS error in the estimate as observed over a large number (the length of the signal file divided by the window length) of subsequent signal segments.

RESULTS

Transfer function

A recursive filter of the first order has insufficient roll-off to suppress the high amplitude low-frequency signals. To increase the roll-off, two recursive filters of the first order are cascaded. The cut-off frequency of this cascade can be approximated by PRF*k/4, where k (k<1) is the filter constant [3]. For practical reasons, k will be a power of 2, since then multiplication can be replaced by shifting the data. However, to have a free choice in cut-off frequency, we will here allow any value for k. The transfer function can be verified experimentally by applying at discrete frequencies a monochromatic signal to the filter and evaluating the input-output relationship. To be rather independent of the duration of observation of the output, this evaluation is performed in quadrature, i.e., two filters are considered simultaneously, one for the in phase and the other for the quadrature component. Allowing some time for settling, the amplitude of the output (and input) is then given by the square root of the sum of squares of the in phase and quadrature component, irrespective of the actual phase of the input signal. The same procedure can be followed for the regression filter. However, the output of the regression filter will be related to the phase of the input signal as well. To eliminate this influence (and residual truncation noise), the input/output relationship is evaluated for a large number of points in time (typically 10*N, where N is the length of the signal segment used for the regression filter).

Figure 1a gives the transfer characteristics of a regression (N=17) and recursive filter having the same cut-off frequency (at the -3 dB level). The filter constant of the recursive filter is adjusted to the cut-off frequency for the regression filter. As expected, the regression filter exhibits a second order characteristic (roll-off of 12 dB/octave). The transition from pass band to rejection band appears to be very sharp. The transfer characteristic of the recursive filter is much smoother resulting in considerably less suppression of low-frequency signals for the same cut-off frequency. Figure 1b depicts the transfer characteristic of the regression filter for

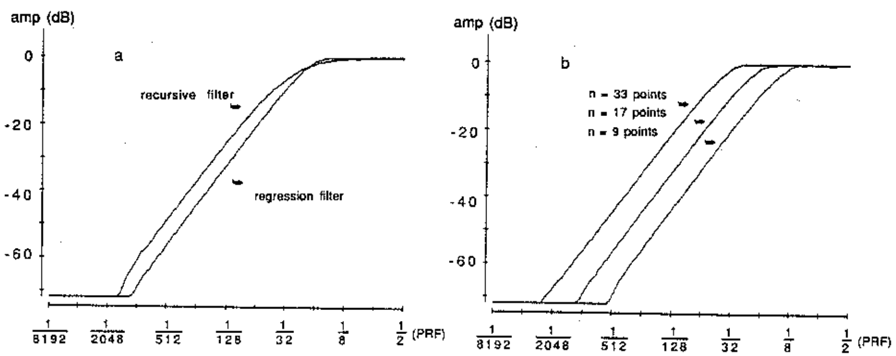

Fig. 1 In the left figure the transfer characteristics of a regression filter and a cascade of 2 recursive filters of the first order, having the same -3 dB cut-off frequency, are compared. The right panel gives the transfer characteristic of the regression filter for a window length of 9, 17 and 33 points, respectively. In both figures the amplitude and frequency axes are on a logarithmic scale where the frequencies are expressed with respect to the PRF.

N=9, N=17 and N=33 sample points. As anticipated, the cut-off frequency shifts downward for an increasing length of the signal segment. It can be concluded that the cut-off frequency at the -3dB level is in the order of 0.87*PRF/N (actually 0.85*PRF/N for N=9, 0.86*PRF/N for N=17 and 0.9*PRF/N for N=33).

Frequency estimation

As stated before, transients may affect the average signal frequency estimated with a correlation frequency detector. To evaluate this effect, computer generated Doppler signals, containing background noise, high amplitude low-frequency signals and a true Doppler signal with a given center frequency and bandwidth are fed to both types of highpass filters and subsequently processed by the frequency estimator. The cut-off frequency of the recursive filter is set according to the length of the signal segment used for the regression filter (N=9, N=17 and N=33). To obtain comparable conditions, the recursive filter is applied to signal segments as well. Therefore, the memories of the recursive filter are set to zero at the beginning of each signal segment. This will introduce transients lasting for some samples, depending on the cut-off frequency considered. The amplitude of the vessel wall signal is either 0 dB, -20 dB or -∞ (no vessel wall signal present). In all simulations, the center frequency of the vessel wall signal is zero, while its frequency range equals 0.01*PRF (rectangular frequency distribution). The center frequency of the true Doppler applied signal is set at 0.2 or 0.4 times the PRF, while the frequency range (rectangular frequency distribution) equals 0.1*PRF (Figure 2).

Table I summarizes the results obtained for the given signal and processing conditions. Presented are the estimated average frequency and its RMS-error as observed over a large number of signal segments. Using the expressions for the cut-off frequency given above the response time of the recursive filter is of the order of 0.29*N. For the recursive filter, either all the sample points within a signal segment or the last N/2-1 points are used for frequency estimation. The latter case reflects the situation that some time (approximately 1.5 times the response time) is allowed for settling of the filter.

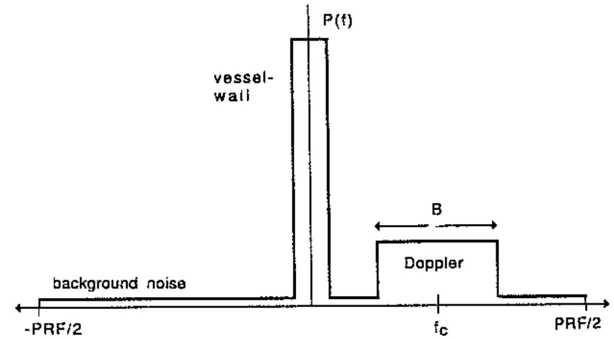

Fig. 2 Spectral representation of the simulated Doppler signal used to evaluate the performance of the regression filter. The spectral width of the stationary signal is 0.01 of the PRF, while its amplitude (specified in the time domain) is either 40 or 20 dB with respect to the amplitude of the Doppler signal.

Table I. Estimated mean frequency and its RMS error (expressed as a fraction of the PRF) for a true Doppler signal with a uniform distribution over B=0.1 around a center frequency of f_d=0.2 and f_d=0.4, respectively, for a regression filter and a recursive filter both having the same cut-off frequency. The number of sample points N within the time segment is set at N=9, N=17 and N=33, respectively. For the recursive filter type either all N points or the last N/2-1 points contribute to the estimate. The amplitude of the vessel wall signal is -∞ (no vessel wall signal), -20 and 0 dB, whereas the amplitude of the true Doppler signal is -40 dB and the amplitude of the background noise is -60 dB (SNR=20 dB).

		f_d = 0.2 B = 0.1			f_d = 0.4 B = 0.1		
A_v dB	N	regres. filt.	recursive filter N	recursive filter N/2-1	regres. filt.	recursive filter N	recursive filter N/2-1
-∞	9	0.205 (0.022)	0.209 (0.021)	0.201 (0.035)	0.404 (0.020)	0.403 (0.020)	0.396 (0.071)
	17	0.201 (0.017)	0.203 (0.016)	0.201 (0.021)	0.401 (0.017)	0.401 (0.017)	0.399 (0.046)
	33	0.200 (0.012)	0.201 (0.013)	0.201 (0.017)	0.400 (0.013)	0.400 (0.013)	0.401 (0.017)
-20	9	0.205 (0.022)	0.047 (0.056)	0.073 (0.068)	0.403 (0.034)	0.087 (0.134)	0.146 (0.159)
	17	0.200 (0.017)	0.043 (0.045)	0.083 (0.059)	0.400 (0.017)	0.069 (0.106)	0.161 (0.149)
	33	0.192 (0.015)	0.039 (0.038)	0.081 (0.056)	0.395 (0.016)	0.057 (0.089)	0.151 (0.141)
0	9	0.202 (0.026)	0.000 (0.015)	0.003 (0.016)	0.398 (0.054)	-0.001 (0.015)	0.004 (0.028)
	17	0.161 (0.041)	0.000 (0.009)	0.004 (0.013)	0.342 (0.088)	0.000 (0.009)	0.004 (0.024)
	33	0.060 (0.048)	0.000 (0.007)	0.002 (0.007)	0.104 (0.122)	0.000 (0.007)	0.002 (0.008)

If the generated Doppler signal is processed by the regression filter, the estimated mean frequency is close to the center frequency of the true Doppler signal. Only for an extremely high amplitude of the vessel wall signal, in combination with a long signal segment a significant deviation is observed. For N=33, the cut-off frequency equals 0.027*PRF. At a roll-off of 12 dB per octave, a signal with a frequency of 0.005*PRF will be suppressed by only 24 dB. That is why for a low cut-off frequency the vessel wall signal may contribute to the estimated average frequency. The transients introduced by resetting the internal memories of the recursive filter have a tremendous effect on the estimated frequency. Only in the absence of a vessel wall signal, the mean frequency is estimated correctly. For a vessel wall signal with a moderate amplitude (-20 dB), a reduction of the number of points contributing to the autocorrelation function improves the results (in terms of mean frequency) but still a significant underestimation is observed. On the other hand, a reduced number of points enlarges the RMS error in the estimate, as is confirmed by the results obtained in the absence of a vessel wall signal. For high amplitudes of the vessel wall signal, the estimator output is fully dominated by the vessel wall signal, irrespective of the center frequency of the true Doppler signal.

DISCUSSION

To reduce the effect of high amplitude low frequency Doppler signals as induced by stationary or slowly moving tissue interfaces (e.g., vessel walls), high pass filtering of the Doppler signals is required prior to frequency estimation. To achieve an acceptable frame-rate in two-dimensional color flow imaging, only short time segments are used to estimate the average frequency of the filtered Doppler signal. However, transients introduced by repetitive switching the direction/position of the ultrasound beam may have a considerable effect on the estimated average frequency. This has been illustrated by a high-pass filter consisting of two recursive filters of the first order. In the presence of a stationary or slowly moving reflector inducing a low frequency Doppler signal with a moderate or large amplitude compared to the amplitude of the true Doppler signal the transients fully determine the output of the frequency estimator. The mean Doppler frequency will be correctly estimated only if very long signal segments compared to the settling time of the filter (at the expense of the field of vision and/or the frame rate) are used. To circumvent the problems associated with transients, an alternative filter (denoted as regression filter) is introduced. It is based on the deviation of the input signal from the regression line within a signal segment of a given length. The number of points within the signal segment fully determine the cut-off frequency of the regression filter. The transfer characteristic exhibits a rapid transition from pass band to rejection band while the roll-off is 12 db per octave. To keep the simulation realistic all computations are in fixed point, except for the arctangent conversion in the frequency estimator. Despite the independent filtering of the quadrature components and the truncation of the intermediate results the estimated mean frequency is correct over a wide range of signal conditions and segment lenghts.

To estimate the coefficients of the regression line for the in phase and quadrature components of the Doppler signal, all the samples within the time window should be available. This requires intermediate storage of the sample points in a memory. The size of the memory should be greater than the number of sample points along a line (e.g., 128) times the number of sample points in time (e.g., 33). Since data processing is executed after data acquisition for a given line of observation has been completed, a high line rate can only be achieved if a second memory is simultaneously filled with data from the next line. As an alternative, while Doppler data processing is in progress, echo information may be assessed along the same

line (or along another line for electronically steered ultrasound systems) assuming that the processing rate per sample position is sufficiently high.

Since the regression filter proposed exhibits a transfer characteristic with a roll-off of 12 db per octave, the cut-off frequency of the regression filter should be set at a considerably higher frequency than the maximum Doppler frequency induced by the slowly moving tissue interfaces. This reduces the ability to assess low blood velocities. A sharper roll-off may be obtained by replacing the regression line by a curve with a higher order at the expense of hardware complexity and processing rate per sample volume.

The RMS error in the estimate for the Doppler mean frequency appears to be better than $\{B/(12N)\}^{1/2}$ [6,7], where B is the bandwidth of the true Doppler signal (not its effective bandwidth) and N the number of sample points within a time window. The background noise will hardly contribute to the RMS error because of the signal-to-noise level used in the simulation (SNR = 20 dB). The RMS error in the frequency estimate for Doppler signals with a center frequency of 0.4 shows the same behavior. This result should be independent of the center frequency of the true Doppler signal at least if there is no aliasing [7]. Only for a very high vessel wall amplitude the RMS error is larger than expected, probably due to residual errors in the filtering procedure. The RMS error can be reduced by increasing N but this will reduce the cut-off frequency of the filter as well. If long signal segments are allowed, for instance, if the frame rate required is low or if only the velocity distribution along a single line is assessed, independent setting of the cut-off frequency and RMS error is still possible. This can be achieved by splitting up the available signal segment in subsegments with a length in accordance with the desired cut-off frequency. The mean frequency, as obtained for each of the subsegments, are averaged to reduce the RMS error in the estimate.

Since the Doppler signals are processed segment by segment the filtered output is not a continuous function of time anymore. This makes the filtered signal unsuited for audio-output and/or spectral decomposition. This will require additional circuitry for the audio-gate.

CONCLUSION

Traditional Doppler highpass filters are susceptible to transients, heavily affecting the estimated mean frequency and its RMS error. The regression filter that we introduced avoids transient effects. The estimated mean frequency is in close agreement with the center frequency of the imposed true Doppler signal over a wide range of signal conditions. Its cut-off frequency can be varied simply by adjusting the length of the signal segment considered. Incorporation of a regression filter to reject high amplitude low frequency signals favors the achievable frame rate of a flow mapper while the RMS error in the estimate is minimal. For a given cut-off frequency, a further reduction in RMS error can be obtained by averaging the outputs of the frequency estimator for subsequent signal segments. However, this approach is only applicable if a high cut-off frequency is required and/or a low frame rate is allowed.

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EXPERIMENTAL EVALUATION OF THE CORRELATION INTERPOLATION TECHNIQUE TO MEASURE REGIONAL TISSUE VELOCITY

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A newly developed correlation interpolation method to measure the regional velocity of moving tissue is evaluated in an experimental setup. Pulsed ultrasound echo signals (center frequency 3.5 MHz) are received from a rotating Agar disk containing scattering particles. When averaging over a depth range of 2.2 mm at a pulse repetition frequency (PRF) of 930 Hz, the standard deviation of the measured displacement between 2 successive pulses was found to be ±6 μm. In a second series of experiments, the angular velocity of the disk is estimated from the displacement, as measured simultaneously in two different regions located on separate echo lines (PRF = 465 Hz per line). The exact position of both regions in respect to the center of rotation was found to be irrelevant. The accuracy of the calculated angular velocity was found to be better for large angles between the two lines of observation than for small angles.

Key words: Correlation; echo; experimental evaluation; heart; rotation; signal analysis; tissue; ultrasound; velocity.

1. INTRODUCTION

During the ejection phase of the left ventricle of the heart, the volume of the cavity decreases. Because the volume of the wall of the left ventricle remains virtually constant, the wall thickens during this phase. Total thickening of the left ventricular wall was experimentally found to be 10 to 30 percent [1-4]. In case of poor regional myocardial performance, for example, due to regional ischemia, wall thickening decreases considerably [1,3]. Thus, left ventricular wall thickening appears to be a useful method to assess regional myocardial performance. To obtain a reliable

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