Effects of Frame Jitter in Data Acquisition Systems

Alexander N. Kalashnikov, Richard E. Challis, Marion E. Unwin, and Andrew K. Holmes

***Abstract*—This paper focuses on the analysis of frame jitter and the impact of data acquisition architecture on the associated disturbances to the acquired record. Frame jitter leads to the same random shift of all samples in an acquired record. It results in errors in the estimates of time intervals, and makes consecutive records slightly incoherent - compromising data averaging procedures. Two complementary algorithms are developed for the quantification of frame jitter, and their performance has been simulated and verified by experiment. They allow the estimation of the standard deviation of the frame jitter using a low-resolution instrument even in noisy environments. An expression for the minimum value of the standard deviation of the frame jitter has been obtained and verified experimentally for typical architectures of data acquisition systems. It is shown that this value could be reduced by specific improvements in the design of data acquisition system architectures.**

***Index Terms*—Jitter, data acquisition architecture, frame synchronization.**

I. Introduction

JITTER is defined as the deviation in time of instantaneous values of a signal from the position they would have had in the undisturbed signal [1]. Jitter distortion causes desynchronization in high speed circuits [2] as well as corruption of information carried by the signal [3], [4]. One of the fundamental sources of jitter derives from the effect of additive noise on the switching of trigger circuits [2]; this has been demonstrated experimentally in the context of different circuit and device arrangements [5] and by systematically attenuating the drive to a trigger circuit in a noisy environment [6]. There are also other possible influences that bring about jitter, such as changes in the raw signal used to generate timing signals, which are pertinent to the recording of physiological data [7], and physical disturbance to the signal transmission path. Jitter can be random, or it can have deterministic properties [8], [9]. In the communications field phase jitter is often considered separately from timing jitter [10], although the two are related and consideration of timing jitter *per se* has more generality. Timing jitter is usually expressed as random errors in the occurrence time of waveform samples [11] and has attracted interest in the context of signal power measurements and signal reconstruction [12]. It is present in equivalent time oscilloscopes in which successive samples of a repetitive waveform are obtained in successive repeats of the waveform; it is a major source of data acquisition errors in these instruments. However, these errors are minimized in oscilloscopes that acquire whole frames of data following a single trigger [13]. Uncertainty in the timing of this single trigger leads to successive frames being shifted relative to each other; and this phenomenon has been discussed in the context of biomedical signal processing [7], speech processing, particle physics, and sonar and radar processing [14]. In high-speed circuits the time interval between successive pulses is frequently of interest [2], and here the problems of jitter associated with individual pulses and that associated with whole frames frequently combine.

The preceding remarks imply that jitter in electronic circuits can bring about a range of malfunctions and can result in errors in data processing which may compromise the conclusions drawn in individual studies. In many situations it is important to quantify jitter and its effects, although this is frequently fraught with uncertainty due to the presence of noise and limited resolution in instrumentation. The aim of this paper is to address this problem through the development of two algorithms for the estimation of jitter in noisy environments using recording instruments of low resolution. The approach taken is to consider timing jitter as it maps into the frequency domain, thus allowing the use of the time shift theorem in the analysis. The first algorithm gives an estimate of the standard deviation of the time jitter without the need to estimate time shifts for each repetition of a signal and in a manner that reduces the influence of noise on the estimation. The second algorithm provides an estimate of the time shift of each repetition of the signal. It is more sensitive to noise than the first method, but it can be used to obtain trends in the time shift that occur, for example, during the warm-up period of digitizing apparatus. The operation of both algorithms is verified by simulation and also by analysis of real data obtained from an ultrasound propagation experiment. A consideration of the architectures of typical data acquisition equipment provides a means to establish the minimum value of the frame jitter that might be expected, and an indication as to how jitter can be reduced further in purpose built instruments.

II. Timing Jitter and Frame Jitter

The difference between timing jitter and frame jitter can be expressed for noisy signals in the following way. We define a noisy signal $y(t)$ as

$$y(t) = x(t) + n(t) \tag{1}$$

where $x(t)$ is the true signal and $n(t)$ is the additive noise.

Timing jitter is the random shift associated with each sample τ_k

$$y(t_k) = x(t_k - \tau_k) + n(t_k). \tag{2}$$

Here we assume that jitter does not affect the additive noise, the samples of which are unknown. These random shifts are caused, for example, by the random elements of an analog to digital converter (ADC) aperture delay and the instability in ADC clock

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frequency. Frame jitter is a random shift in the whole acquired signal record, implying that each sample in any given frame possesses the same time shift as all the others

$$y(t_k) = x(t_k - \tau) + n(t_k) \tag{3}$$

or, for continuous time

$$y(t) = x(t - \tau) + n(t).$$

Frame jitter can thus be considered as a special case of the sample jitter when $\tau_k = \tau =$constant.

The particular observation that stimulated our interest in frame jitter was connected with the measurement of the ultrasonic attenuation coefficient in highly absorbing media. This coefficient is derived from the amplitude spectrum of the recorded signal in a highly nonlinear way leading to significant magnification of the otherwise negligible error [15]. In order to reduce the additive noise, coherent averaging provided by a high quality instrument was employed in the time domain. On the basis that averaging in either the time or the frequency domains should provide the same results, it holds that

$$F\{\bar{y}\} = F\left\{\frac{1}{N}\sum_{i=1}^{N} y_i\right\} = \frac{1}{N}\sum_{i=1}^{N} F\{y_i\}$$

where F is the linear operator representing time to frequency domain Fourier transformation and i is the index of a particular frame out of N. Considering the moduli on either side of this equality we might initially further expect that

$$|F\{\bar{y}\}| = \left|F\left\{\frac{1}{N}\sum_{i=1}^{N} y_i\right\}\right| = \frac{1}{N}\sum_{i=1}^{N} |F\{y_i\}|. \tag{4}$$

However, with incoherence between successive records y_i there will be a difference between the left- and right-hand sides of (4). For example, two counterphase sinewaves of the same frequency and amplitude would produce zero on the left side, but the amplitude value on the right side. In a number of experiments, we have found that even small timing differences of around 1% between records observed after processing of 1000 records led to significant differences in the estimates of ultrasonic wave attenuation coefficient. This difference appeared to be due to the above mentioned small incoherences between the acquired records that were brought about by frame jitter in the data acquisition system. Examples of experimental records disturbed by frame jitter have been presented and analyzed in [16].

The influence of frame jitter is best quantified in terms of its statistical moments. Here the mean value is not particularly significant because it applies to all repetitions of a given signal record just as the average aperture error in ADCs leads only to a permanent time shift of the waveform without any distortion. The standard deviation of the jitter time is the most useful because it provides the basis for estimating, for example, maximum time interval errors (MTIE) in the context of high speed periodic signals [14]. Because of the link between frame jitter and timing jitter it is, in principle, possible to apply algorithms designed for the estimation of timing jitter [11], [17], [18] to the estimation of the standard deviation of frame jitter. However, these algorithms are generally based on underlying signal properties, which in many situations are either unknown or are difficult to estimate, and in our view should not be necessary for the estimation of frame jitter.

The estimation techniques developed for high-speed circuits [14], [19], [20] focus on worst cases in the timing properties of pulse sequences, such as MTIE, and employ a threshold crossing technique at zero level. In many cases they are not appropriate for information signals that possess a high density of zero crossings. Additionally, they may require purpose built [14] or high-resolution [2], [20] instruments, and a massive number of test signals [2], [19], [20]. Jitter time shifts have been estimated from an ensemble of realizations that were aligned using an appropriate algorithm [7], [13] to provide the notionally best representation of the undisturbed signal. However these algorithms require solution of a least square problem in order to estimate the true signal, and this seems rather complicated for the estimation of frame jitter. The algorithms that we develop later provide a much simpler method for such estimations.

III. Influence of the Frame Jitter in the Frequency Domain

With both noise and jitter present the frequency domain signal is

$$F\{y\} = Y(\omega) = F\{x(t-\tau) + n\} = F\{x(t-\tau)\} + F\{n\} \tag{5}$$

where the Fourier transform for N samples of the signal is

$$F\{s\} = \sum_{l=1}^{N-1} s_l \exp(-j\omega\Delta t_l). \tag{6}$$

Invoking the time shift theorem the first term in (5) becomes

$$F\{x(t-\tau)\} = F\{x(t)\}\exp(-j\omega\tau). \tag{7}$$

If we assume that the frame jitter time is small with respect to the shortest period of the signal we can make the approximation

$$\exp(-j\omega\tau) \approx 1 - j\omega\tau \tag{8}$$

hence

$$\begin{aligned} F\{x(t)\}\exp(-j\omega\tau) &\approx F\{x(t)\}(1 - j\omega\tau) \\ &= F\{x(t)\} - F\{x(t)\}j\omega\tau. \end{aligned} \tag{9}$$

If the jitter time shift has zero mean, we get

$$F\{x(t-\tau)\} \approx X(\omega) + X(\omega)j\omega\sigma_f r_f \tag{10}$$

where σ_f is the frame jitter standard deviation and r_f is a random variable with zero mean and unity variance.

Let us consider the practical validity of (10) noting that it assumes that $\omega\tau \ll 1$ and ignores the mean time shift as it can be represented as an average aperture delay associated with the ADC. The Nyquist criterion states that the sampling frequency

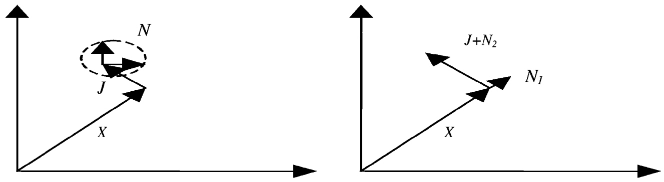

Fig. 1. Phasor diagrams of the recorded spectra (X—true signal, N—additive noise, J—frame jitter), (a) before and (b) after rotation of the additive noise components.

of a waveform must be at least twice that of the highest frequency spectral component in the signal. Theoretically, transient signals possess infinitely wide spectra, and in practice a sampling frequency which is ten or more times greater than the highest signal frequency is generally used. Recordings cannot be considered as accurate if the jitter exceeds, say, one tenth of the sampling interval. On this basis we obtain a reasonable estimate of the phase product of (8) as follows:

$$\omega\tau \leq \left(2\pi\frac{1}{10}f_s\right)\left(\frac{1}{10}\frac{1}{f_s}\right) = \frac{2\pi}{100} < 0.07.$$

The second and higher order terms that have been neglected in our approximation to the time shift theorem (8) will have amplitude of less than 0.25% of the zero order term, which is unity, and so they can reasonably be neglected. If the average time of the frame jitter differs from zero, it leads to an additional permanent delay of the frames, but does not distort them in any way. Therefore, it can be assumed to be present in x and $F\{x\}$, although not treated explicitly.

Considering the second term in (5), and assuming that the additive noise is Gaussian in the frequency domain we can represent the noise spectrum as [21]

$$F\{n\} \approx r_r\sigma_n(\omega)\sqrt{\frac{N}{2}} + jr_i\sigma_n(\omega)\sqrt{\frac{N}{2}} = r_rC + jr_iC \quad (11)$$

where $\sigma_n(\omega)$ is the noise standard deviation at frequency ω, r_r, and r_i are independent random variables $N(0,1)$, and $C = \sigma_n(\omega)\sqrt{N/2}$.

Combining (10) and (11) yields

$$\begin{aligned} F\{y\} = Y(\omega) &= F\{x(t-\tau) + n\} \\ &\approx X(\omega) + X(\omega)j\omega\sigma_f r_f + r_rC + jr_iC. \end{aligned} \quad (12)$$

Blair [3] has shown that the terms responsible for the additive noise can be rotated in the complex plane by an arbitrary angle (Fig. 1). Consequently (12) can be rewritten for two components, one of them in phase with $X(\omega)$, and the other in quadrature

$$\begin{aligned} Y_I &= X(\omega) + r_IC \\ Y_Q &= X(\omega)\omega\sigma_f r_f + r_QC \end{aligned} \quad (13)$$

where r_I and r_Q are random independent variables $N(0,1)$. Subtracting the spectrum of the true signal we obtain the residuals

$$\begin{aligned} \Delta_I &= r_IC \\ \Delta_Q &= X(\omega)\omega\sigma_f r_f + r_QC. \end{aligned} \quad (14)$$

Finally, the frame jitter standard deviation at any frequency is obtained in terms of the variances of the components of the residuals

$$\sigma_f = \frac{\sqrt{\mathrm{Var}(\Delta_Q) - \mathrm{Var}(\Delta_I)}}{X(\omega)\omega}. \quad (15)$$

Equation (15) provides the basis for the first frame jitter estimation algorithm. The second algorithm will be for the case of negligible additive noise ($C \ll X(\omega)\omega\sigma_f$) when the jitter shift could be estimated from (13) at any frequency as

$$\tau \approx \frac{Y_Q}{X(\omega)\omega}. \quad (16)$$

IV. Estimation of the Standard Deviation of the Frame Jitter

As frame jitter results in random shifts of the same record in the time domain, we can detect its presence by observation of an ensemble of records. In order to determine the standard deviation of the frame jitter on the basis of either (15) or (16) it is necessary to know the amplitude spectrum of the true (unjittered and denoised) signal and this is not measurable directly. However both these random factors are additive in nature, thus their influence can be reduced by averaging. In our experiments we acquired 1000 frames, and evaluated the spectrum of the true signal as the average of the spectra of the individual records.

Numerical implementations of two algorithms based on (15) and (16) respectively include additional features that allow reduction in the effects of noise. The first algorithm [based on (15)] computes the ratio $(\sqrt{\mathrm{Var}(\Delta_Q) - \mathrm{Var}(\Delta_I)})/(X(\omega))$ versus ω then calculates the best straight line fit $\omega\sigma_f$. In an earlier publication we have reported a related calculation based on measurement at a single frequency [16]. The ratio and the line are considered at particular frequencies only where: 1) the variance of the quadrature component exceeds the variance of the in-phase component by at least factor 1.2; this is to ensure that the difference in the estimates of the quadrature components is due to the frame jitter, and not to their inherent variability; and 2) the amplitude of the true signal is not less than 0.1 times its maximum value—this is to cap the highest frequency used in the analysis, otherwise variability would occur at higher frequencies, which can be seen in Fig. 3(c). The choice of the parameters mentioned above was based on an analysis of a number of experimental records, and provided stable and reasonable results. The second algorithm is based on (16) and uses the frequency at which the amplitude spectrum of the signal reaches its maximum; alternatively, for dc-like signals, the maximum of the product $\omega X(\omega)$ could be used instead. The standard deviation of the frame jitter is then evaluated as the standard deviation of the individual time shifts.

These two algorithms complement each other: The first provides a more accurate estimate in noisy conditions, and the advantage of this is evident in simulations whose results are presented in Fig. 2(a) where estimates of the frame jitter obtained using both algorithms are plotted versus the standard deviation of the additive noise. Here the sinewave burst signal, Fig. 2(b) was used as the notionally true signal. The second algorithm leads to a significant overestimate of σ_f. However, it does give

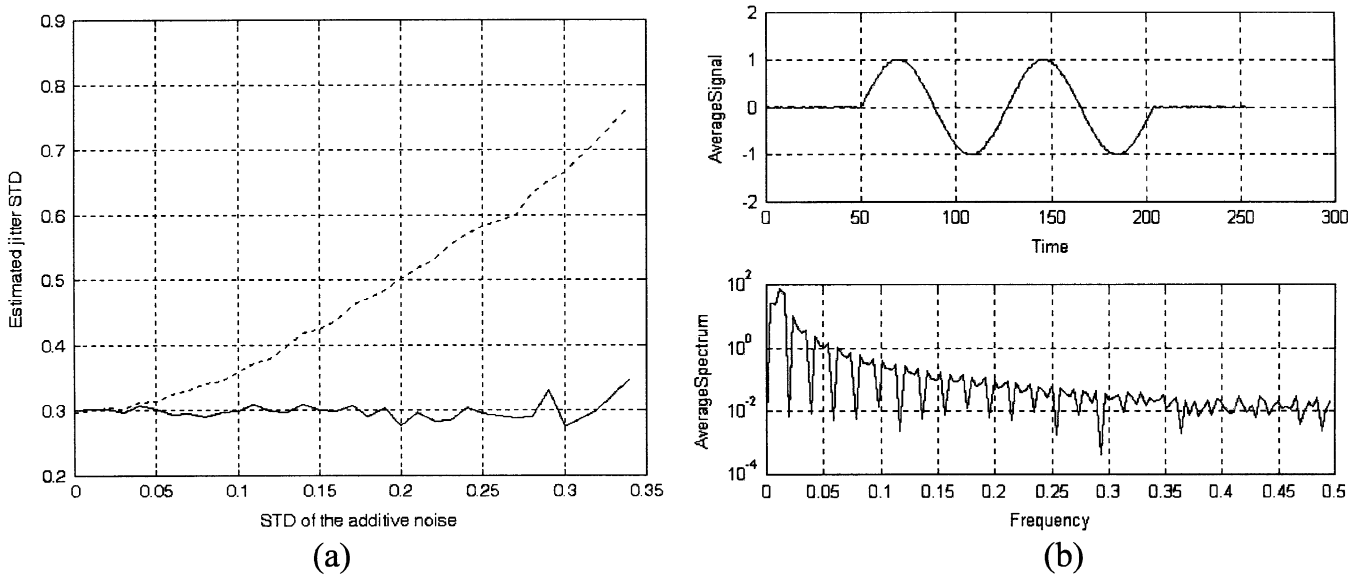

Fig. 2. (a) Simulation of the noise immunity of the two algorithms (true value of the frame jitter STD is 0.3, solid line—first algorithm, dashed line—second algorithm). (b) Test signal and its spectrum.

an opportunity to isolate jitter trends, and examples of these under different conditions were presented in [16].

Both algorithms were applied to experimental data collected from the output of a commercial ultrasonic pulser-receiver (UPR [22]) using a digital storage oscilloscope (DSO, model 9450, LeCroy Inc) operating at a sampling frequency of 400 MHz. The UPR provides a separate synchronization output that was used to trigger the DSO. The average signal computed from 1000 digitized time domain records is shown in Fig. 3(a) together with the amplitude of the averaged spectrum. This spectrum was used to represent the true signal. Fig. 3(b) shows standard deviations of the spectrum residuals (14) and the differences that appear due to the frame jitter. These differences, which are related to the amplitude spectrum of the true signal, are plotted in Fig. 3(c) where a linear trend is evident. As mentioned earlier, the experimental points for curve fitting were selected on the basis of two rules, and those selected are identified by superimposed circle and cross symbols. If the average spectrum at any particular frequency is less than the required threshold then the associated point is placed on the frequency axis. If the difference between the variances of the residuals is below the corresponding threshold then the cross is absent. The best-fit line intersects the origin as expected, and here we note that the same behavior was also observed in many other sets of data. The time shifts of individual records estimated according to (15) are shown in Fig. 3(d). They behave as a random process with zero mean, and there is no observable trend. This implies that the estimate obtained by the first algorithm applied to these data is not biased.

The numerical values obtained by using both algorithms are presented in Table I.

The use of curve fitting in the first algorithm gave an improvement in the results compared to the earlier version which was based on a single frequency [16]. The second algorithm leads to a significant overestimate of σ_f notwithstanding the relatively low amplitude of the noise. This is due to the high energy in the low frequency components of the test signal which are only marginally affected by the frame jitter, the influence of which increases with frequency. An important implication of this is that even noise of low amplitude becomes significant in the quantification of time shifts. In contrast, the first algorithm provides consistent results, and the high frequency sinusoidal signal seems to be a better alternative to the rectangular pulse as a test input.

V. Influence of the Data Acquisition Architecture on Frame Jitter

It is perhaps obvious that the standard deviation of the frame jitter depends on the sampling frequency and the instrument used. Data acquisition instruments, and whole systems containing them, can essentially be divided into two distinct categories which represent two types of basic architecture. The first of these employs separate devices—one for the signal source and one for data recording and digitizing. The operation of the two parts of the system is synchronized by a trigger signal between the source and the recorder, Fig. 4; a system of this kind was employed in the experiments described in this work and previously in [16]. Even with an adequate trigger signal the actual event of digitization may still not be properly synchronized to the signal source, there being a random delay between the trigger and the next closest sampling clock edge in the digitizing sequence that will capture a whole frame of data samples, Fig. 5. Now the repetition frequency of the frames is generally very much less than the ADC sampling frequency within each frame. An important implication of this is that the time difference between the trigger and the start of a frame will be uniformly distributed, yielding the following expression:

$$\sigma_f = \frac{1}{f_s\sqrt{12}} \approx \frac{0.289}{f_s} \tag{17}$$

where f_s is the sampling frequency. Using this expression for the experimental arrangements described above, the standard

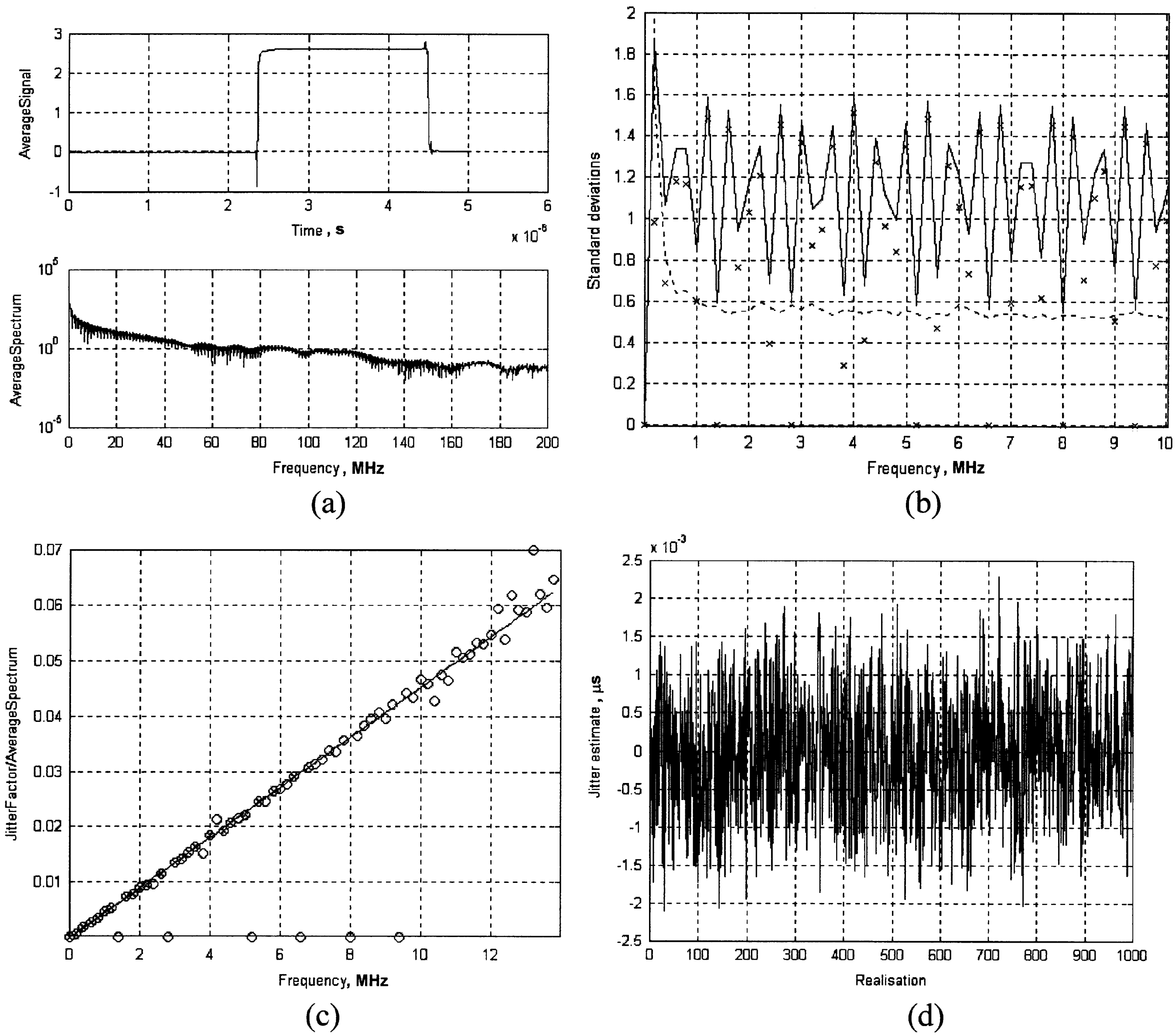

Fig. 3. Application of the algorithms to the experimental analysis of a rectangular pulse. (a) Averaged pulse and the averaged spectrum. (b) Standard deviations of the spectrum components (solid line—quadrature, dashed-in-phase), and their difference (crosses). (c)Jitter factor—experimental points and regression line. (d) Estimated time shifts.

NUMERICAL VALUES OBTAINED BY USING BOTH ALGORITHMS

σ_f	Comment	Algorithm 1	Algorithm 2
Rectangular pulse (Figure 2)	High low-frequency content	0.73 ns	0.85 ns
Acoustical spectroscopy signal [16]	Narrow band signal	0.72 ns	0.73 ns

deviation of the frame jitter should be 0.72 ns, and this is almost equal to the value estimated by the first algorithm (see Table I). This type of system is used in much equipment that supports data acquisition, and for such systems (17) establishes the minimum value of the frame jitter standard deviation that could be expected even if the trigger waveform and the digitizer response to it were in all practical senses perfect.

The second architecture employs rigid synchronization between the test system frame generator and the higher frequency digitizing clock, illustrated diagrammatically in Fig. 6. This type of system can be found in some commercial devices, particularly in the ultrasonic field (like EUI [23]). In a large group of tests on such systems and using the first algorithm applied to experimental records, we have estimated values for σ_f that were up to five times lower than the limiting value implied by (17) for a system with a sampling frequency of 320 MHz. It is therefore clear that this architecture is greatly superior to the first in relation to its frame jitter amplitude. Its advantage is particularly

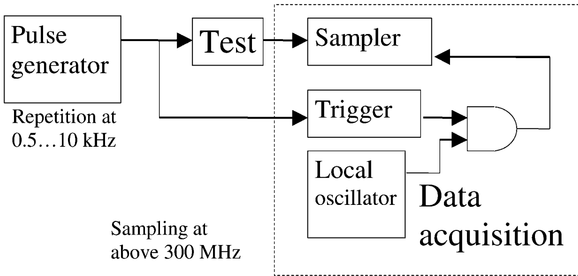

Fig. 4. Generic architecture of the first data acquisition system: the pulse generator and local oscillator are independent.

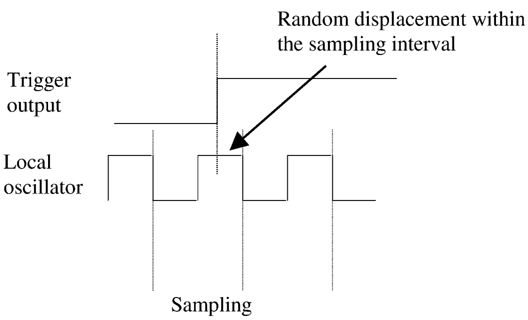

Fig. 5. Appearance of the random time shift between the trigger event and start of sampling that is evenly distributed within the sampling interval.

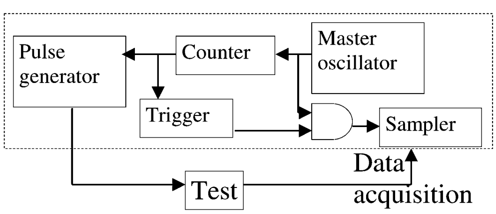

Fig. 6. Generic architecture of the second data acquisition system: the pulse generator is triggered by the master clock.

significant in situations where it is required to accurately measure time delays from a trigger event, for example, when measuring wave group velocity or phase velocity and its dispersion. It is also significant when using coherent averaging to improve signal-to-noise ratio (SNR) since frame jitter introduces partial incoherence between notionally identical signal components in successive frames, leading to erroneous loss of energy in the higher frequency spectral components.

VI. Concluding Remarks

Frame jitter is the timing error that arises in data acquisition instruments in which a frame of data is acquired in response to a single trigger. It differs from the timing jitter that is present in an equivalent time digital oscilloscope, which results in random independent timing errors associated with every digitized sample. In contrast, frame jitter leads to the same timing error for every acquired sample in the same frame. This error compromises accurate measurements of time intervals and complicates the use of averaging because the consecutive records become incoherent.

Our analysis of frame jitter in the frequency domain showed that it could be considered as an additive random term masking the true signal. The phasor of this term is orthogonal to the phasor of the true signal, and its modulus is proportional to the modulus of the true signal phasor and frequency. The result is that distortions caused by the frame jitter increase with frequency, and cannot be reduced by enlarging the signal amplitude, although coherent averaging can still reduce its detrimental effects.

We have described two algorithms which complement each other, and should be used simultaneously in order to quantify the influence of frame jitter by determining its standard deviation. The first algorithm provides a more accurate estimate if there is no trend in the time of arrival of different frames, even in noisy conditions. The second algorithm leads to significant overestimation of the frame jitter in noisy conditions, but could be used to detect the presence of the above-mentioned trends. A significant advantage of both algorithms is that they do not require an instrument with a high resolution for the quantification of jitter.

A consideration of the architecture of common types of data acquisition instruments showed that the minimum obtainable value of the standard deviation of the frame jitter is related to the sampling interval in just the same manner as the standard deviation of the quantization noise relates to the value of the quantization box (17). This applies to the cases where the sampling clock of the digitizing instrument is not fully synchronized with the test signal generator. The influence of frame jitter can be reduced significantly by provision of this synchronization.

The results presented here are applicable to the design of digitizing instruments and the assessment of their quality. The results also apply to timing measurements related to wave propagation and data transfers more generally, and can be used to establish the number of coherent averages that would be required to achieve measurements with uncertainties that were below a specified threshold.

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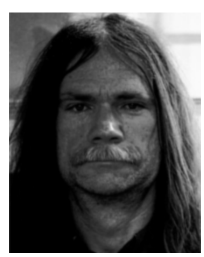

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