High-pass filtering extends the dynamic range for recording pulse shapes

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When a roughly flattopped pulse electrical pulse is recorded by a digitizer, signal clipping can destroy information that is crucial to the experimenter. This is particularly true in release-overtake shock experiments, where the occurrence time of a change in slope at the top of a photomultiplier pulse is sought. We have invented a simple and accurate method to record the pulse that is robust to clipping. The digitizer records the signal through a high-pass circuit. The inverse of the high-pass filter is applied via fast Fourier transform during analysis. The insensitivity to clipping allows one to record the change in slope signature at a higher gain, and thus with an improved signal to noise ratio. The technique is generically applicable to any recording of a roughly flattopped pulse. © 1995 American Institute of Physics.

I. INTRODUCTION

In many experiments, the shape of a pulse is recorded by an electronic digitizer for later analysis. In the particular case of a roughly flattopped pulse, any signal clipping may produce a total loss of information important to the experimenter. This is particularly true for an important kind of shock experiment called the "release-overtake" experiment. We will use this application as an example to demonstrate a simple signal processing method that accurately records the top of the pulse in spite of significant signal clipping.

The release-overtake experiment promoted by McQueen¹⁻³ is used to measure the speed of sound in compressed materials. In this experiment, a thin impactor generates a shock and release (rarefaction) wave both traveling forward through the sample (Fig. 1). The depth at which the release catches up to the leading shock is a measure of the sound speed in the compressed sample. The moment of overtake is detected most sensitively optically, from behind the target. Behind the sample is a fluid which produces light under shock. When the shock enters the fluid the light intensity jumps immediately to a roughly constant level (Fig. 2). When the release wave catches up to the leading shock in the fluid, the light intensity drops sharply, producing a change in slope denoted as the catch-up point. Measuring the time difference T_{cu} between the start of the light pulse and the catch-up point is the goal. Different impactor thicknesses produce different T_{cu} 's; this information is used to determine the sound speed in the sample.

McQueen reports² that the light intensity I varies with the impactor velocity U_f approximately

$$I \sim U_f^{8}. \tag{1}$$

(This is consistent with $I \sim T^4$ by the Stefan-Boltzmann law, and typically observed shock behaviors $T \sim P$, and $P \sim U_f^2$, where T is temperature, and P is the shock pressure.) We accelerate our projectiles with a two-stage gas gun, initially propelled by gunpowder. The variation in powder burning produces a final projectile velocity variation of 10% in some cases. Because of the high exponent of Eq. (1), this yields a variability in signal intensity of ~2:1. An unknown sample equation of state may also contribute a significant uncertainty through the variation in P, through $I \sim T^4 \sim P^4$. Because the useful portion of the signal lies at the top of a pulse, any overloading is a catastrophic loss of the catch-up signature. This forces us to be especially conservative in choosing the nominal vertical gain of our digitizing recorder. However, the 8 bit vertical resolution is insufficient at the reduced nominal level to determine the catch-up time to the desired precision.

As a solution, we have invented a simple method to eliminate the catastrophic sensitivity to clipping. The signal is recorded through a high-pass circuit. The inverse operation is applied numerically during data analysis. We will demonstrate that the method is robust to clipping and preserves the high frequency information needed to determine the change of slope at the catch-up point. As a generic method, it can be applied to any pulse experiment where the pertinent information lies on top of a roughly flat pulse.

II. METHOD

Our optical signal is detected with photomultiplier tubes (PMTs). Coaxial cables lead signals to Tektronix DSA 602 transient digitizers. Coaxial cable delays are used to increase the total cable length to 300 ns to make the appearance of spurious echos obvious. The vertical resolution of the digitizers is 8 bits, or 256 vertical points. The sampling interval is 2 ns.

The traditional way to handle signals of high dynamic range is to use redundant channels with different vertical gain settings. We assign 2 PMTs to observe each of 6 target locations, with a high and low optical attention preceding each PMT tube. We must be mindful that in spite of the increased dynamic range of the digitizer using this method, the PMTs have their own dynamic range limitation. Excessive light will cause PMT saturation. However, this saturation is more forgiving than the clipping of the digitizer. The change in slope at the catch-up point, though weaker, can still be observed in the saturated signal.

Our method is based on the idea that the catch-up point is a change of slope, from a slope that is initially small. Therefore, if one records the derivative of the signal, the signal immediately preceding the catch-up will always be



FIG. 1. Target design. Impact of a thin projectile initially generates two shocks traveling in opposite directions. One shock reflects off the impactor rear and becomes a forward traveling release (R) wave, which eventually overtakes the leading shock (S). When the shock leaves the sample (T) into the indicator fluid (L), it emits light related to pressure detected by photomultiplier tubes (PMTs). Different sample thicknesses produce different catch-up times, information which is used to deduce sound speed in sample.

near zero, and this can be placed at the center of the digitizing range for any gain.

We can generate the derivativelike signal by a high-pass circuit. This is trivially implemented by inserting a small capacitor in series at the digitizer input (Fig. 3). The 50 Ω impedances of the digitizer input and cable with the capacitor form a high-pass network with a time constant $T_{\rm RC} \approx 30$ ns. Due to stray inductances, the behavior of the filter is more complex than a simple RC circuit. We do not attempt to model its effect by circuit theory. Instead, we measure the actual transfer function for the filter of each channel. This is done both accurately and conveniently by fast Fourier transforms of the digitized signals.

An electrical test pulse is applied to each channel, with and without the high-pass filter and recorded by the digitizer as $V_{\text{with}}(t)$ and $V_{\text{without}}(t)$, respectively. The fast Fourier transforms (FFT) are divided to produce the frequency response of the filter

high-pass(f) =
$$\frac{\text{FFT } V_{\text{with}}(t)}{\text{FFT } V_{\text{without}}(t)}$$
. (2)

If $V_{\text{PMT}}(t)$ is the PMT signal to be found, and $V_{\text{rcrd}}(t)$ is the recorded digitizer signal, then without clipping the PMT signal is obtained

FFT
$$V_{\text{PMT}}(t) = \frac{\text{FFT } V_{\text{rerd}}(t)}{\text{high-pass}(f)}.$$
 (3)

The steps in the recording process are illustrated in Fig. 4. The source PMT signal (a) is converted to a derivativelike signal (b) by the high-pass circuit. This signal suffers clipping, because a high gain is deliberately used to improve the signal to noise ratio around the catch-up point. We have determined that large amounts of clipping can be tolerated by



FIG. 2. The shock entering liquid generates light related to pressure by $I \sim P^4$. When the rarefaction wave catches up to the leading shock, the intensity changes slope. The time from the start of the light to catchup (T_{cu}) is the measurement of interest.



FIG. 3. (a) Electrical configuration. The photomultiplier (PMT) generates a signal from the shock generated light. The capacitor and internal 50 Ω impedances of the recorder and cable form a high-pass filter with a time constant of $T_{\rm RC}$ =30 ns. The transfer function of the filter high-pass(f) measured by applying the test pulse with (b) and without (c) filter and dividing their Fourier transforms FFTs.

the process without adversely affecting the catch-up signature, as long as the catch-up point is not closer than about $T_{\rm RC}$ to the pulse start.

After the experiment, the recorded derivativelike signal is "integrated" by dividing its Fourier transform by highpass(f) by Eq. (3). This signal [Fig. 4(d)] is sufficient to determine the catch-up point, since all the high frequency information, except that under the clipped region, is present. However, for mostly cosmetic reasons, the signal can be further processed to bring a closer resemblance to the original in low frequencies as well. The difference in baselines [Fig. 4(d)] is defined as M. A triangle of area M is added to the clipped signal [Fig. 4(e)]. When this is integrated via Eq. (3), the result [Fig. 4(f)] is very close to the original in both low and high frequency information.

The only portion of the signal significantly altered by the clipping is directly under the clipped region, which is approximately a time constant T_{RC} from the pulse start. We



FIG. 4. The steps in recording the experimental signal; (a) photomultiplier signal; (b) after the high-pass circuit; (c) digitizer clips signal, (d) after numerically applying the inverse of the high-pass filter. Catch-up time $T_{\rm cu}$ can be obtained from this signal. Optionally, further processing will restore proper shape to the tail. The baseline offset M is the area that needs to be restored to the derivative signal (e). (f) The final result after applying the inverse of the high pass filter to (e).

4704 Rev. Sci. Instrum., Vol. 66, No. 9, September 1995

High-pass filtering

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FIG. 5. The effect of clipping on the reconstituted signal is tested numerically. (a) Bold curve: source PMT signal. (b) Bold: simulated high-passed signal obtained by multiplying a Fourier transform of PMT by high-pass(f), which was measured from an actual filter. The signal was then clipped at either 0.1 or 0.05 V and integrated by division of its Fourier transform by high-pass(f). These reconstituted curves are light curves in (a). The curves are shifted vertically for clarity. Data for a short time after shock is altered, but change in slope at the catch-up point remains.

have numerically tested the ability to resolve the catch-up point close to the pulse start on the severity of clipping. For the PMT pulse, we used an experimental signal [Fig. 5(a), bold] measured previously with direct coupling between the PMT and digitizer. The simulated derivativelike signal [Fig. 5(b), bold] was obtained by applying Eq. (3), and artificially clipped at either of two levels. This was then integrated via Eq. (3) and shown as light curves against the original curve in Fig. 5(a).

The results show that this method is very robust to severe clipping. Only the region directly clipped was affected. All other regions retained the high frequency information content needed for detecting the change in slope.

We observe that our digitizer does not appear to be "blinded" by overloading; it recovers promptly.

Figures 6 and 7 show the result to the signal of Fig. 5 when the missing area M lost to the clipping is restored. M is given by the baseline offset in the integrated result (Fig. 6).



FIG. 6. Steps in restoring the low frequency information, on the data of Fig. 5. The integrated clipped signal has a baseline shift, defined as M. A triangle of area M is added on top of the clipped region. This is then integrated and shown in Fig. 7.

The width of the triangle is the region of clipping; the height is calculated from the area and width. Figure 7 shows excellent agreement between the original and restored curves for both high and low frequency information, except in the exact region of clipping. We speculate that agreement can be further improved by using a more complex shape instead of a



FIG. 7. Comparison of the original, and reconstituted signal when the missing area destroyed by clipping to 0.05 V is restored by a triangle as described in Fig. 6. The curves are offset vertically for clarity. The inset shows detail around the pulse start and catch-up point. The change in slope at the catch-up is faithfully preserved, as well as the entire pulse tail. The portion directly under the clipped region (<320 ns) suffers some distortion.

Rev. Sci. Instrum., Vol. 66, No. 9, September 1995

High-pass filtering 4705

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triangle, perhaps matching the slopes of the derivative signal at the boundaries. This would be warranted if the experimenter had prior knowledge of the typical behavior of the front of the pulse, such as its rise time.

III. DISCUSSION

This method provides an easy and accurate method to make signal recording much more robust to clipping, for signal details that lie on top of a roughly flat pulse. This in turn allows the nominal signal level to be increased, producing a superior signal to noise ratio. This technique is not limited to shock physics, but to any experiment involving pulses. The use of fast Fourier transforms with digitized data makes the signal processing trivially simple, yet precise.

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