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PROBLEMS ENCOUNTERED WHEN MAKING SIMPLE IMPULSE MEASUREMENTS

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Introduction

Transient measurements of system response have become increasingly popular in recent years, principally because the measurements involved are at first sight simpler and less time consuming than those in the frequency domain. One type of transient method is that in which the test signal is an impulse. The development of impulse techniques has been considerably enhanced by the availability of fast, low-cost digital computers, which are invaluable for data acquisition and analysis.

We have been measuring the response of a variety of brass instruments to an acoustic impulse presented at the input end. Our aim is to locate and identify small differences between the internal bore features of nominally identical instruments. Pressure variation is measured at the input of the instrument. Each bore discontinuity causes a reflection which appears at the appropriate instant in the pressure versus time record.

In this paper, we discuss some of the problems that have arisen in this work, particularly those to do with impulse consistency. A careful detailed analysis of the experimental set-up has shown that perhaps impulse measurements are not as simple as they at first appear.

Desirable features of input impulse

The impulse response is defined as the response produced when the test signal is the delta function. One normally desires the test signal to be a good approximation to the delta function, i.e. to be as tall and as narrow as possible.

However, the following theory shows that deconvolution can be applied to the response measured using a non-ideal impulse in order to obtain the response that would have been measured using an ideal input impulse; this removes the necessity of having the ideal impulse.

If:

$x(t)$ = input signal at origin
$y(t)$ = output signal at origin
$z(t)$ = transient response,

then the following convolutional relationship exists:

$y(t) = x(t-\tau)z(\tau)$, i.e. $y(t) = x(t)*z(t)$

For a linear, causal, time-invariant system, convolution in the time domain is equivalent to multiplication in the frequency domain, i.e.

$Y(f) = X(f).Z(f)$
and $Z(f) = Y(f)/X(f)$

Thus, when independent measurements of $x(t)$ and $y(t)$ are made, division of their spectra followed by a reverse FFT should produce the impulse response.

To obtain spectra from the time signals requires windowing of the time records, followed by a forward FFT. To date, our attempts at this deconvolution process have produced unsatisfactory results. Windowing introduces an inherent error into the results. Durrani and Nightingale (Ref. 1) show that because the windowing process involves a weighting of the time domain data record, the resulting frequency domain spectrum will be affected. This means that the deconvolved time record is no longer sufficiently accurate for our purposes. Therefore, we have decided not to carry out deconvolution. So we must again ensure that our input impulse is as close to the delta function as possible. This implies a good pulse shape, i.e. a sharp rise and clean decay, with no secondary spikes or ringing. A loudspeaker source was found to be unsuitable because of too much ringing. The coupler between the source and instrument had to be chosen with care to avoid secondary reflections.

The delta impulse is infinitesimally narrow in order that as broad a frequency range as possible be covered. However, the fact that data is recorded by sampling produces a contrary requirement, namely, that the signal contains no frequency content above the Nyquist frequency to avoid aliasing. The frequency range of an electrostatic actuator source was not broad enough. A good signal-to-noise ratio is desirable. Impulse sources usually have a poor S-N ratio. It can be improved by signal averaging, which requires that the impulse should be repeatable. Also the impulse should always occur at precisely the same time in the data records, or else averaging will lead to "smearing". Exploding wires and shock tubes are unsuitable sources as the impulses produced are not repeatable (Ref. 2).

A spark source was chosen as the best impulse source for our application, being cheap and robust. Such a source has not been without some drawbacks. Unfortunately it is unable to produce negative-going impulses, a fact which will be discussed later. The main problem has been the variability of the impulses produced.

<u>Impulse measurement apparatus</u>

The spark source and data acquisition system were first devised by John Goodwin (Ref's 2,3,4). A block diagram showing the experimental set-up is given in Fig.1. The sampling rate is variable and is set using a programmable oscillator. The ADC can accept data at rates up to 150 kHz. The programmable anti-aliasing filter ensures that all frequency components above the Nyquist frequency are removed. The spark source is actually a modified car spark plug. The plug is driven by a specially-built EHT pulse generator, which is triggered by a pulse from the DAC; the ADC is simultaneously enabled. This synchronisation between pulse production and data acquisition means that, in principle,proper signal averaging can be carried out, although we have found out that this synchronisation is not as perfect as at first assumed. This is discussed further later.

A matter for careful consideration was the means of coupling the impulse source to the instrument. The original damped mouthpiece coupler (shown in Fig. 2a),

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although convenient, had the following inherent problems:

(1) reflections in the mouthpiece cup produced secondary spikes on the decay of the impulse,

(2) the source was not non-reflecting, which caused confusing multiple reflections in the pressure versus time record,

(3) because of the close proximity of the source and the microphone, the microphone diaphragm kept breaking. A careful investigation into the cause of this breakage suggested that it was due to the very high pressures produced in the vicinity of the source.

The new long source tube coupler removed all three of these problems, as follows:

(1) there are no longer any secondary spikes,

(2) the source is effectively non-reflecting because the pulse travels from the microphone to the instrument bell and back more quickly than from the microphone to the source and back,

(3) the tube has the effect of attenuating the highest frequency components of the pulse. Thus, the excessive energy which was causing microphone breakage has been removed. The anti-aliasing filter would have had to remove this high frequency energy in any case, but would not have provided the necessary protection for the microphone.

Signal averaging and the necessity for impulse consistency

The aim of signal averaging is to improve S-N ratio. The background noise is assumed to vary randomly, so averaging over many runs is assumed to reduce it significantly. Investigations into the manner in which the mean background noise varies from run to run reveal that the distribution is more weighted towards a positive pressure than a negative pressure. Observations were made of how the running average of mean background noise changes as the number of averages increases. It was revealed that the changes are very variable at first, but after 100-200 averages, the mean takes on a more or less steady, small positive value. See Fig. 3 for an example. The reason for this positive bias is not apparent. It would have been an advantage if the spark source were able to produce negative as well as positive pulses. If, during data acquisition, pulses were alternately positive and negative, and the "negative records" were multiplied by -1, then any bias in the background noise distribution would cancel out during signal averaging. This technique has been successfully applied by Fincham (Ref. 5).
When signal averaging is carried out, it is essential that the impulses produced for each run should have consistent characteristics, that is:

(1) timing - we have already mentioned that if the impulse timing varies, the averaged record will undergo a "blurring" effect,

(2) shape,

(3) amplitude.

An inherent feature of spark sources is that the impulses produced by them are not consistent. Pulse shape undergoes small random variations, but the most noticeable inconsistency is pulse amplitude. The physical reasons for these inconsistencies are not well understood. It is known that spark formation produces ionization of air, and that "recovery" or "de-ionization" requires a certain amount of time (Ref. 6). This time is of the order of a fraction of a second. In our set-up, the time interval between successive firings is at least 1 second. An investigation into the effects of increasing this delay to 5 seconds revealed no change in impulse consistency. So we conclude that impulse variability is not caused by there being insufficient time for full recovery of the spark gap.

With our present spark source, the curved electrode has been removed from the spark plug in order to increase the spark gap length and thence the power of the impulse. In consequence, the spark can form at any position round the outer electrode ring (see Fig. 4a). It was felt that this might be contributing to impulse inconsistency, and that a more precisely defined gap would lead to more consistent impulse characteristics. For this reason tests were carried out on a spark plug with its curved electrode and central electrode each filed to a point. It was found that in the latter case the impulses were even less consistent, despite the fact that the sparks formed at the same position each time; so we see that defining the spark gap more precisely has not enhanced consistency.

Although the causes of impulse inconsistency remain unexplained, a detailed investigation of the above inconsistencies has been carried out. Computer software has been written which monitors the variation of various parameters from one data run to the next. These parameters have included impulse timing, impulse shape, impulse amplitude, mean background noise level and the time interval between the impulse and later features in the time record. Tables of data for many runs (typically 200 consecutive runs) can be written away; then distributions can be plotted and various statistics (mean, mode, standard deviation, etc.) can be calculated from them. Fig. 5 shows typical impulse amplitude variation (in volts, over 200 runs).

Random fluctuations of a parameter can easily obscure a gradual increase or decrease of that parameter. Therefore, it has been useful to calculate the mean value of a parameter over the previous ten runs (along with the SD). This mean is updated after every run. This latter test can also reveal in more detail the extent of the small random fluctuations which can often be obscured when the statistical calculations are carried out over a very large number of runs. Ways of overcoming existing inconsistencies have been attempted.

Consistency of timing

Originally the timing of the impulse was defined by examining the sampled data record and finding the sample number at which the peak amplitude of the impulse occurred (the peak sample number). To overcome the timing inconsistency problem, each incoming run was examined, and runs whose peak did not occur at the predetermined peak sample number were rejected and not included in the final signal averaged version. It was soon found that for our purposes this test was not stringent enough; the timing of the impulse peak needed to be defined more accurately. We have done this by interpolation. Using Gaussian elimination, a polynomial is fitted through five sample points around the impulse peak, the peak sample number being the central point. We can then calculate pressure at 100 "inter-sample" points, and thus the peak position can be located to an accuracy of 0.01 of a sample. (We now call the peak location the "fractional peak sample number" or "FPSN").

Our observations of how FPSN varies over a series of many consecutive data runs have revealed the following causes for variation in peak position:

(1) Ambient temperature variation

We know from the Gas Laws that the velocity of sound in free air varies in proportion to the square root of the absolute temperature. Accordingly, when the measurement environment is warmer on one day than it was on the previous day, the velocity will have increased; and so the FPSN will have decreased. When two instrument responses measured on different days were being compared, this presented a great problem. Differences between two nominally identical instruments are identified by arithmetically subtracting their pressure versus time records. A plot of arithmetical difference versus time can reveal the points in the instrument at which differences occur; for instance, an extra discontinuity in one of the instruments would show up as a "blip" in the "arithmetical difference plot". If the temperature at which the two measurements were taken was different, the impulses would occur at different times and so would all the features of the standard pressure versus time record; so the difference plot would be filled with many confusing and unwanted extra "blips". We have resolved this problem by changing the sampling rate as necessary so that the impulse always occurs at the same fixed FPSN, regardless of what the ambient temperature is. Then in later plots and processing, we simply "pretend" that the sampling rate was the same in both cases.

We have found that because of the very precise interpolation of peak position, our apparatus is now extremely sensitive to small drifts of ambient temperature which may take place whilst a series of measurements is being taken. Our normal sampling rate is 46 kHz. At this rate, we found that for a temperature rise from 20.4°C to 20.6°C, the mean peak position was reduced by 0.08 of a sample (which corresponds to 1.7 microseconds).

As well as the gradual overall drift in FPSN due to a gradual ambient temperature change, there are random fluctuations due to small changes in pulse shape or momentary temperature changes. We have found that these random changes in timing usually occur within a range of 0.06 of a sample at 46 kHz (i.e. 1.3 microseconds). Data runs whose FPSN lies outside this 0.06 "acceptability window" are now rejected.

(2) An inherent fault in the computer sampling process

As mentioned earlier, the synchronisation between the triggering of the impulse and the start of data acquisition is not perfectly consistent. The precise instant at which the computer sampling process begins after the pulse has been triggered may vary by one or two sample periods. This would result in the whole data record being shifted by one or two samples. Our software can now detect when this occurs, and shift the whole record back to its "correct" position.

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Consistency of pulse shape

Consistency of pulse shape has been investigated by writing away 10 consecutive single data runs, one immediately following the other, and comparing them. These tests reveal that the general shape is always the same, but small variations occur. When larger variations in shape occur, they are accompanied by a larger change in FPSN (i.e. timing) or amplitude; as each run is already tested for its acceptability with respect to pulse timing and amplitude, this means that the pulses with adverse shape will automatically be detected and rejected.

Consistency of amplitude

Impulse amplitude is monitored in terms of voltage input into the ADC. The ADC voltage range is +/-10 volts. The amplitude acceptance window is usually set at the upper 2 volts of this range. The original reason for this was the desire for maximum resolution. However, we have found that for our purposes the resolution is sufficient even when the window is at the low end of the voltage range; so the position of the amplitude window can now be flexible. Distortion caused by ADC overload is prevented by setting the upper amplitude limit just below the maximum possible ADC voltage. In practice, about 70% of the impulse amplitudesoccur within a 1 volt range of the ADC. The total percentage variation of amplitude is usually 25-30%. Through experience, we have learnt that on some days, amplitude variation is worse than on other days; for example, on one occasion the standard deviation of a 200-run amplitude distribution was 0.60 volts, and a few days later it was 0.49 volts. It was originally thought that the extent of the variation was dependent on ambient conditions, such as calmness of weather and presence of background noise. However, our statistical tests show that ambient background noise does not have a significant effect on amplitude variation; the range of mean background noise level is only a very small percentage of the range of impulse amplitude (between 4% and 8%). Additionally, readings are sometimes more consistent during stormy weather than during calm weather. Thus, the cause of variation of consistency between days remains obscure.

Discussion

We have discussed the necessity of a consistent delta-function-like impulse. We have found that our present spark source produces well-shaped pulses, but its consistency is poor. Tests show that the consistency of a pulsed loudspeaker source is far superior, but the loudspeaker remains an unsuitable source for our application for the reason already mentioned. Rather than seek a new source, we have sought to "improve" the consistency of the present source by rejecting those runs whose impulses lie outside quite stringent limits of acceptability in amplitude and timing. Normally, at least 70% or 80% of the impulses produced are acceptable.

We have also solved the major problem of compensating for variations in ambient temperature.

With our present set-up, we have achieved our goal, namely, we can adequately detect small variations between a test instrument and an ideal prototype, even when the two measurements are done on different days with different ambient conditions. An example of this is the comparison of a euphonium in a "normal" state with the same euphonium with one of its valves misaligned by a small amount (3mm). Fig. 6a shows the time record for the normal euphonium, with the position corresponding to the misaligned valve identified. Fig. 6b shows the arithmetical difference between the two records and we note the existence of a prominent "blip" at the position of valve misalignment.

References

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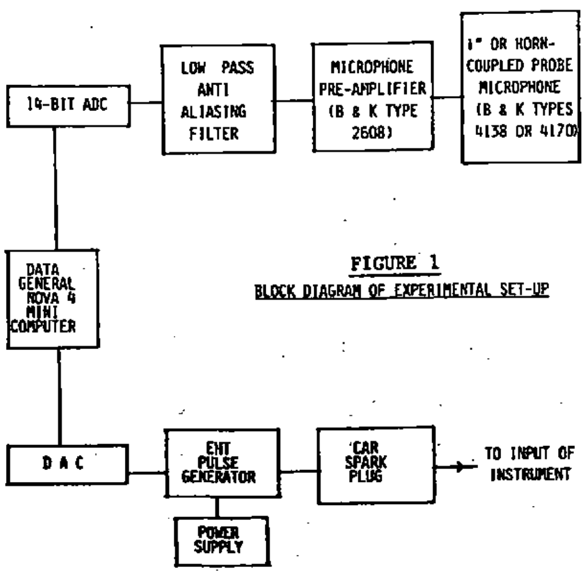

FIGURE 1

BLOCK DIAGRAM OF EXPERIMENTAL SET-UP

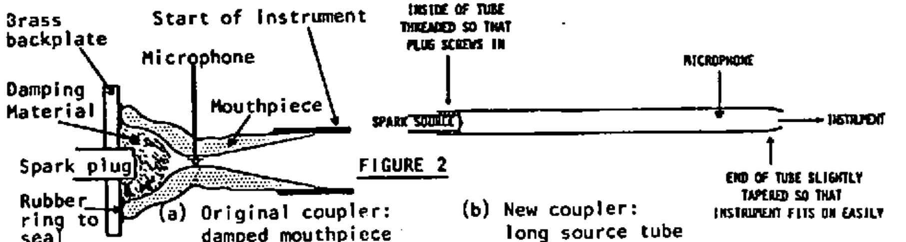

FIGURE 2

(a) Original coupler: damped mouthpiece

(b) New coupler: long source tube

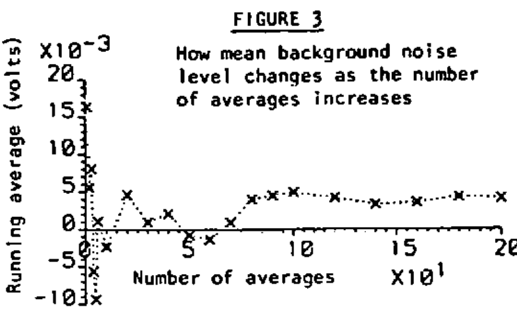

FIGURE 3

How mean background noise level changes as the number of averages increases

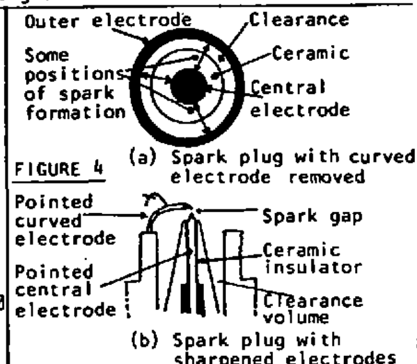

FIGURE 4

(a) Spark plug with curved electrode removed

(b) Spark plug with sharpened electrodes

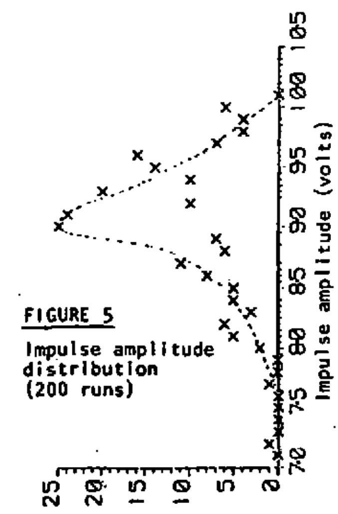

FIGURE 5

Impulse amplitude distribution (200 runs)

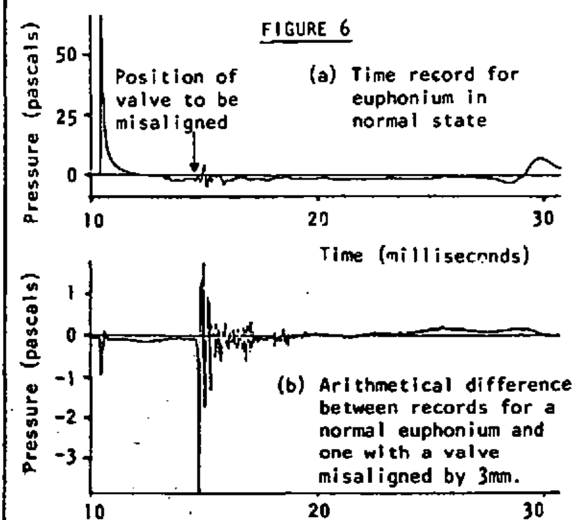

FIGURE 6

(a) Time record for euphonium in normal state

(b) Arithmetical difference between records for a normal euphonium and one with a valve misaligned by 3mm.