

Time Scale Re-Sampling to Improve Transient Event Averaging

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ABSTRACT

As the drive to make automobiles more noise and vibration free continues, it has become necessary to analyze transient events as well as periodic and random phenomena. Averaging of transient events requires a repeatable event as well as an available trigger event. Knowing the exact event time, the data can be post-processed by re-sampling the time scale to capture the recorded event at the proper instant in time to allow averaging. Accurately obtaining the event time is difficult given the sampling restrictions of current data acquisition hardware. This paper discusses the ideal hardware needed to perform this type of analysis, and provides analytical examples showing the transient averaging improvements using time scale re-sampling. These improvements are applied to noise source identification of a single transient event using an arrayed microphone technique. With this technique, the averaging is performed using time delays between potential sources and microphones in the array. As a result, the relative time information needed is contained within the measured data and a separate trigger event and event time are not required.

INTRODUCTION

The desire to improve the noise and vibration characteristics of automobiles, as well as to further understand issues such as sound quality, has led to the recent development of many new data analysis techniques. Nearly all methods for analyzing transient events, including wavelets and Wigner-ville distributions as well as many others, are performed on time domain data. Typically these techniques are applied as post-processing of data, providing a great amount of flexibility in the analysis process. Many of these techniques compute results in the form of a spectrogram which is a time versus frequency distribution plot. If the transient data were first averaged, additional insight into the

structure of a transient event may be gained from the spectrogram computing algorithms.

Averaging transient data enhances a transient event because it reduces components of the data which are not correlated with the occurrence of the transient event. These components which are reduced in averaging include both periodic and random phenomena, which may add "noise" to the transient analysis results and completely mask some characteristics of the transient event. Averaging transient data also improves the statistical estimate of the characteristics of a transient event. A better statistical estimate of the characteristics of a transient event are especially important if the amplitude of the event is not exactly consistent from average to average.

While transient averaging is desirable, it requires a very repeatable excitation. This excitation must also be of a form which allows it to be used as a trigger in the acquisition of the response data. The actual time at which the excitation occurs, known as the trigger event time, must be measured very accurately to preserve frequency information near the Nyquist frequency in the averaging process. This type of accuracy is not possible at a sample rate of 2 samples per cycle of the highest frequency of interest. The data must therefore be oversampled to obtain a higher sample rate and thus determine the trigger event time more accurately. The disadvantage of oversampling with many data acquisition systems is that the same sample rate is used for all channels, resulting in an acquisition of much more response data than is necessary. The preferred acquisition hardware would allow a much higher sample rate on the trigger event channels than is used on the response channels.

Once accurate estimates of event times are established, the response data can be re-sampled to align response data samples in time with the start of the excitation. The synchronized transient events can then

be averaged in the time domain prior to applying a transient analysis technique. The re-sampling can be done either in the time domain or the frequency domain.

One application of time scale re-sampling is noise source identification of a single transient event using an array of microphones with a delay-and-sum technique. This method does not require a separately acquired trigger event or accurate event time, and thus does not rely on specialized hardware for extremely high sample rates. It is based on removing time delays, computed from the speed of sound and the distances between potential source points and microphones in the array, from microphone responses and summing them for an enhanced source signal. This technique requires accurate time delays and accurate synchronization of the microphone signals before the summation. Time scale re-sampling greatly improves this synchronization and, hence, the accuracy of the technique.

TIME SCALE RE-SAMPLING THEORY

If the trigger event and the response data are both oversampled, re-sampling is not necessary. If, however, the trigger event is sampled at a higher frequency than the response data, it is possible that the response data will have to be shifted backwards in time to the instant at which the trigger event occurred. The re-sampling necessary to synchronize the response data with the trigger event may be done very efficiently in either the time or frequency domains, as discussed in the following subsections. To accomplish this, the response data is shifted in either the positive or negative directions of time to align a data sample with the time instant at which the trigger event occurs.

FREQUENCY DOMAIN RE-SAMPLING - The most efficient method to re-sample the data is to use FFTs. The response is effectively time shifted by transforming the response to the frequency domain, applying the appropriate phase shift, and transforming back to the time domain. The phase shift is described by $e^{j\omega\tau}$ and is equivalent to a delay in the time domain. This procedure essentially shifts the time signal by the delay τ in the time domain, with the start of the time block shifted to the end of the time block and the end of the time block shifted to the start. As a result, there should be at least τ seconds in the time block at both the start and end of the transient event, so that the shifted transient event remains intact. A disadvantage of transforming to the frequency domain can be leakage effects, if the transient is not completely observed, i.e., does not go to zero at both ends of the data block. The advantage of this technique is that the shifted data is valid out to the Nyquist frequency.

TIME DOMAIN RE-SAMPLING - An efficient method to re-sample the data can also be implemented in the time domain through the use of an FIR filter which introduces a simple time delay in the response data.

An efficient FIR filter which is designed to perform this time delay is based on common interpolation filters [ref. 1]. Interpolation filters are commonly used to digitally up-sample data. A good interpolation filter will have a very small ripple in the passband and a very steep roll-off. These interpolation filters approximate the analog sampling process by positioning the center of the interpolation filter at the instant in time at which a data sample is desired. The filter coefficients which are aligned with data samples are then convolved with these data samples to determine the value of the data at the desired instant in time. The data should also be acquired such that the transient does not start at the beginning of the time block and does not continue to the end of the time block. This will reduce any filter transient effects which may otherwise be introduced into the data. A limitation of this process is that the data is not valid out to Nyquist due to the roll-off of the interpolation filter. An advantage of this method over the frequency domain method is that there are no leakage effects introduced in this process.

RESPONSE DATA UPSAMPLING - The response data can also be upsampled to a time axis with a finer delta-t resolution, which can be done efficiently by upsampling by an integer number. New data samples are calculated which fall at fractions of the current delta-t's. This procedure, while not providing the accuracy of re-sampling to a specified time axis, can still provide a drastic improvement in the accuracy of the averaged results. This is due to the fact that the data has smaller delta-t's, thus any time shifting necessary to average the data can be done with less approximation than with the original time history.

Upsampling may be done in the time domain using a digital filter by inserting evenly spaced zeros between original data values and applying a low pass filter to this zero inserted time history. This will result in a time history which only has frequency content out to the original Nyquist frequency or less. In the frequency domain, upsampling may be done by performing an FFT, appending zeros to both the positive and negative frequencies, and then performing an inverse FFT. This inserts new data values evenly spaced between the original data values.

TIME DOMAIN AMPLITUDE ACCURACY - Transient data which is to be analyzed in the frequency domain requires that the data be sampled at a rate of two samples per cycle or greater for the highest frequency of interest. The FFT will give an accurate amplitude estimate for all frequencies up to Nyquist with this sample rate.

If the data is to be analyzed in the time domain, two samples per cycle are not enough to statistically characterize the amplitude of the signal accurately. It is recommended that if time domain amplitude is important, the data be sampled at a rate of at least ten points per cycle of the highest frequency of interest. This sample rate gives a maximum amplitude error of

5% with an average error of 1.6%. If twenty points per cycle are acquired, the maximum amplitude error will be 1.3% with an average error of .04%. The data may be acquired at two points per cycle and upsampled by an arbitrary amount to improve the amplitude resolution of a signal. This process is subject to the same limitations (i.e., filter effects, leakage, etc.) as the time and frequency domain re-sampling procedures.

TRANSIENT AVERAGING WITH A TRIGGER

To ensure accurate time domain averaging of any type, it is necessary to determine very accurately when a trigger event begins. Once this has been done, averaging may be performed to suppress the characteristics of the signal which are not correlated with the trigger event, and to enhance those characteristics which are correlated with the trigger event.

For transient averaging, the event timing is done by recording the exact moment in time at which an excitation is input to the system. Note, however, that a trigger event will always be recorded as having occurred later in time than it actually happened, due to the delay between when the trigger event occurred and the next instant in time at which the event channel is sampled. For transients, typically the trigger events are quantities which vary rapidly, although some trigger events (such as a throttle position in an automobile) may vary slowly. As an example, if the engagement characteristics of an A/C compressor clutch are to be analyzed, a typical event which may be monitored is the electrical signal which energizes the clutch. For brake engagement, the monitored event may be the depressing of the brake pedal. Other events may be monitored through electrical signals from the vehicle's computer, sensor outputs from sensors which are already installed on the vehicle, or through sensors which are mounted specifically for trigger purposes.

Theoretically, if enough averages are taken of an event, the resulting averaged response will be the impulse response of the system due to the excitation source which is used in the averaging process.

TRIGGER EVENT TIMING ACCURACY - For accuracy, trigger event times must be acquired at a sample rate which is considerably higher than twice the highest frequency which is to be analyzed after the averaging process. This implies that on an acquisition system which samples all channels with the same sample rate, much more response data must be acquired than is minimally necessary from Shannon's sampling theorem. This accuracy with which the trigger event times must be recorded depends on the highest frequency of interest for analysis. For example, many acquisition systems may specify a phase accuracy of $\pm 1^\circ$ between channels. This specified phase accuracy is important in cross channel DSP calculations, such as cross-powers or frequency response functions. To achieve an accuracy of $\pm 1^\circ$ phase in the averaging of transient data, the trigger event times must be recorded

to within $\pm 1^\circ$ phase of the highest frequency of interest. This implies that the event channel must be sampled at a rate of 180 samples per cycle of the highest frequency of interest! Obviously, this sample rate may not always be possible. As the sample rate decreases, the high frequency components of the response are averaged out as though they were noise.

Several strategies for acquiring the trigger events accurately may be used. One acquisition strategy is to acquire the event channels at a very high sampling rate, while acquiring the response channels at a sampling rate which is commensurate with the maximum frequency of interest. This strategy is much more efficient in terms of data storage than sampling all channels at a very high rate, but still uses considerable storage space to describe the event times. This is because an ADC is used for the trigger event channel when an analog input may not be necessary. For example, if a micro-switch is used as a trigger device, an acquisition system with a 16 bit ADC records 16 bits of information for the signal. However, a micro-switch only has on and off positions, which can be described completely with 1 bit. Therefore, the preferred acquisition strategy is to use an acquisition system which has digital inputs sampled at a very high rate for the trigger events, and analog inputs for the response channels. Digital inputs are 1 bit, where the bit is set to on or off. The response channels are digitized with ADC channels which have 12-16 bits of accuracy and are sampled at the minimum rate necessary for the frequency range of interest.

Another issue which must be addressed is the synchronization of the event with the response data if there is not a response data sample occurring at the time the event time is recorded. The solution to this obstacle is discussed in the next subsection of this paper.

PROPOSED TRIGGER EVENT RECORDING SYSTEM - A data acquisition system is proposed which minimizes the storage space for event times and is based on a very accurate embedded clock. This proposed acquisition system uses an event monitoring channel to record the exact time at which a trigger event occurs, instead of recording a bit for every sample time of the event. This type of event time recording requires much less storage space, since only the time at which a trigger event occurs is recorded. This system also "stamps" the response channel data with a periodic time stamp to ensure that the event time and the response data can be synchronized through re-sampling. The accuracy of the time stamp clock must be sufficient enough to provide the desired accuracy of trigger event timing. This implies that if an accuracy of $\pm 1^\circ$ phase at 20 kHz is desired, the clock has to be accurate to within approximately 150 nano-seconds. This type of clock accuracy is becoming available and should not be considered absurd! If this acquisition system is used to acquire data on rotating equipment, the tachometer

signals are treated as event channels and only their zero crossing times recorded by time stamps.

This type of data acquisition system also allows separate acquisition modules to acquire and process data as though all data were acquired on one acquisition system. Using separate modules requires the recording of a periodic event signal or synchronization signal on each acquisition module. The modules are either synchronized through a cable which supplies a clock signal, or each system has its own clock. If each system has its own clock, the clocks are synchronized before and after data is acquired, and then all data is re-sampled to a common time axis. The time scale re-sampling is done using the re-sampling algorithms discussed above. Once the data is re-sampled to a common time axis, all multiple channel DSP analysis techniques, such as cross-power analyses, are applied to the data.

ANALYTICAL EXAMPLE

Analytical data was created to demonstrate the improvements in the averaging process for re-sampled data. Three analytical examples are presented. The first example uses simple sine waves, while the second example uses impulse response functions which are transient in nature. The third example shows the advantages of averaging a transient event which contains periodic noise prior to computing a spectrogram.

SINE WAVE EXAMPLE - The first example using sine waves is actually an example of synchronous averaging. In synchronous averaging, a tachometer signal related to the rotation of the shaft is monitored to determine the start of a revolution. Data is then acquired for a set number of revolutions. This data is synchronized to the start of a revolution and may then be averaged in the time domain [ref. 2-4]. This averaging procedure enhances the information which is related to the rotation of the shaft and suppresses information which is not correlated with the rotation of the shaft.

This example was created by superimposing sine waves from each Δf which had a random phase and an amplitude of 1, known as a pseudo-random signal. This signal was created such that the trigger signal was always received after the trigger actually occurred. This is the situation encountered when the data is tape recorded on a DAT recorder or not sampled at a high enough rate. A Monte-Carlo study was performed on this scenario where 50 averages were taken with randomly generated portions of a Δt time delay. The random distribution was a normal distribution for both the phase of the sine waves and the portions of a Δt at which the trigger time was incorrect. Time domain averaging was performed with the 50 time histories and an FFT performed to determine the error in magnitude as a function of normalized frequency. This

procedure was applied 500 times and the results averaged.

The error in magnitude is a non-linear function which approaches approximately 5 dB at the Nyquist frequency, as seen in Figure 1. This corresponds to an amplitude error of approximately 40% at Nyquist! The same process was then performed with a time domain interpolation algorithm employed to shift the signals back to the actual trigger time. With this procedure, the amplitude is now within ± 1 dB out to approximately .75 of the Nyquist frequency. This procedure is sensitive to the characteristics of the interpolation filter used, including both the roll-off of the filter which in this case led to the amplitude error above .75 of Nyquist, and the passband ripple of the filter which may cause amplitude errors. The same process was then performed using the frequency domain phase shifting procedure, which resulted in nearly perfect amplitude estimates all the way to Nyquist. This curve is indistinguishable from the actual curve, as shown in Figure 1.

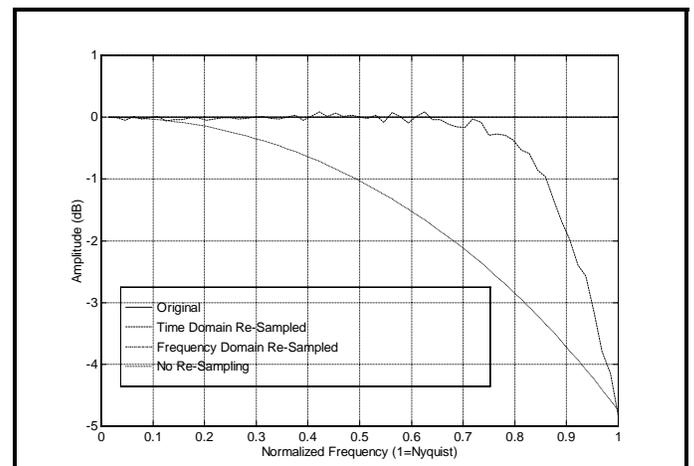


Figure 1: Effect of re-sampling of sine waves.

Figure 1 demonstrates that both re-sampling algorithms preserve much more information near the Nyquist. It can also be seen that if the amplitude errors are to be minimized without re-sampling, the data must be acquired at a sample rate of five to ten times higher than is minimally necessary.

IMPULSE RESPONSE FUNCTION EXAMPLE - The second example which is presented is the result of taking 50 averages of a transient signal. This transient signal is the impulse response of a system and was averaged by randomly generating a trigger time. The trigger signal was not recorded until the next Δt after it occurred. The responses due to the impact were then averaged without any re-sampling, and by using the frequency domain re-sampling procedure. The results of this study are shown in Figure 2.

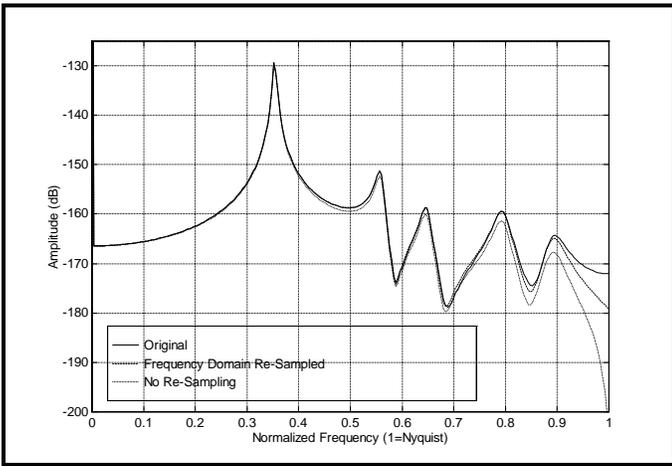


Figure 2: Frequency response after 50 averages.

Clearly it can be seen that as the frequency approaches the Nyquist frequency, there is an error in the amplitude estimate of the averaged data where no re-sampling was performed. Again, re-sampling the data results in about a 3 dB improvement in amplitude at the highest frequency mode.

IMPULSE RESPONSE WITH NOISE - This example consists of periodic noise of four sine waves of constant frequency and amplitude added to a system's impulse response. This example simulates data acquired on a rotating machine where the transient event is not related to the rotation of the machine. Figures 3 and 4 show the time domain signals before and after averaging with the frequency domain re-sampling.

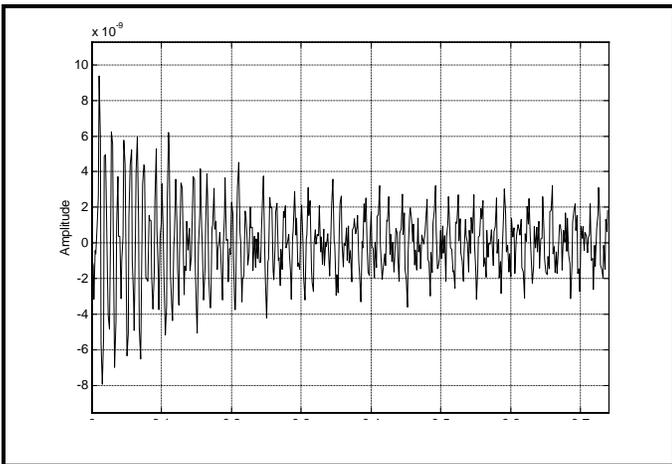


Figure 3: Transient signal with periodic noise.

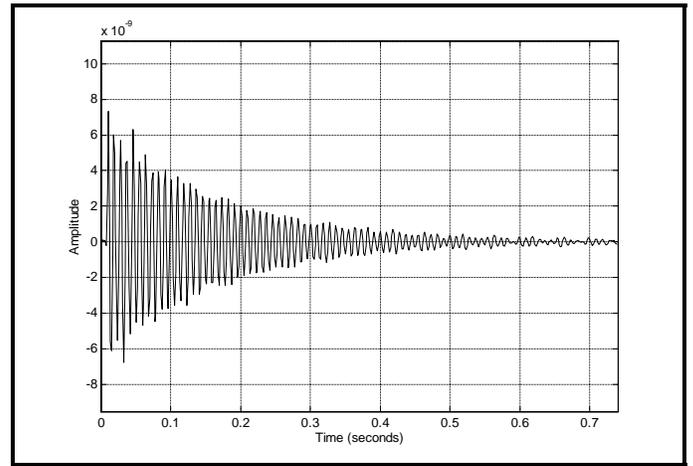


Figure 4: Averaged transient signal with noise.

Figure 4 clearly appears to look more like the impulse response of a system than Figure 3. The 200 averages taken to generate this plot have effectively reduced the amplitude of the periodic components. Frequency domain representations of these signals, along with the averaged with no re-sampling results, are shown in Figure 5.

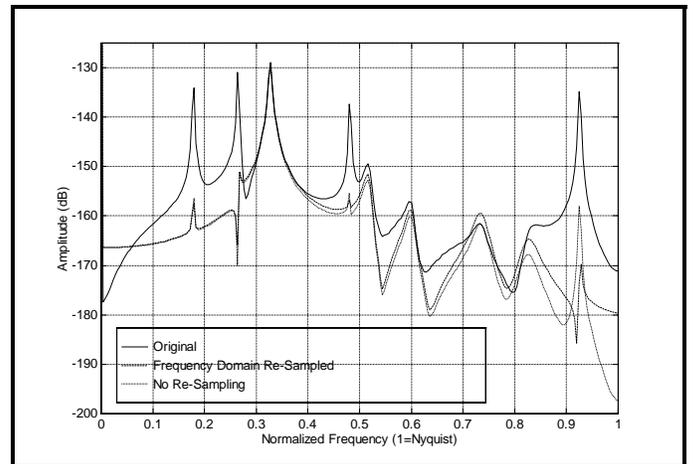


Figure 5: Frequency domain plot of transient with noise.

Figure 5 again shows that the averaging process has reduced the amplitude of the periodic noise. The periodic noise peaks are the peaks which have been reduced by large amount. It can also be seen that the averaged with no re-sampling procedure reduces the amplitudes of both the noise and the transient's components of interest.

To further analyze the averaged and non-averaged transient events, spectrograms of one acquisition period and of the averaged signal were computed. These spectrograms, given in Figures 6 and 7, show the need for averaging when noise is present. The single acquisition spectrogram is the spectrogram which would be generated with no averaging. All of the modes and the rates at which they decay are difficult to

see in this spectrogram, which is shown in Figure 6. The periodic components mask much of the transient signal's characteristics.

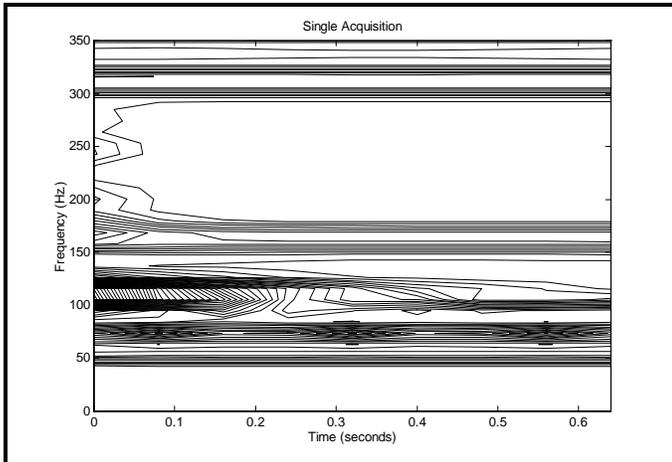


Figure 6: Single acquisition spectrogram of signal.

By averaging the signal before computing the spectrogram, the characteristics of the transient event are much more easily seen. The averaged data spectrogram is much easier to interpret even though traces of the periodic components are still visible. Figure 7 shows this averaged data spectrogram.

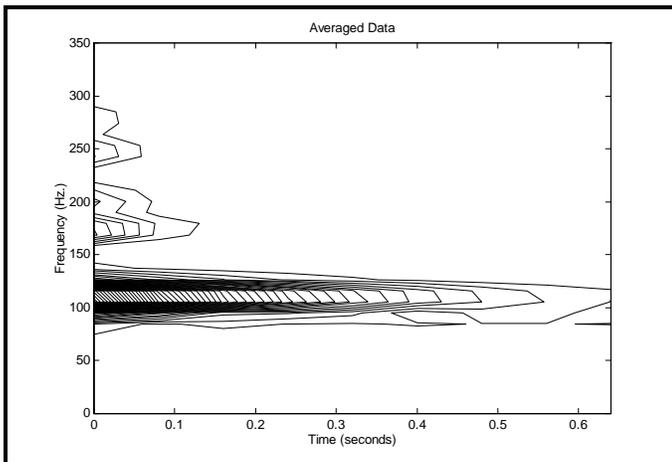


Figure 7: Averaged data spectrogram.

TRANSIENT AVERAGING OF A SINGLE EVENT

In noise and vibration applications, there are often sources producing transient signals which are not repeated, such as squeak and rattle noise sources. To identify these transient noise sources, a direct time domain acoustic array technique was developed [ref. 5] which uses spatial-temporal averaging. No recorded event or explicit event time is necessary, thus no specialized hardware is required. The technique is a temporal array method based upon a summation of arrayed microphone signals which have been shifted by

known source-to-microphone time delays, known as a delay-and-sum technique. The response time histories are shifted by the appropriate time delays in the time domain, or equivalently, the phase is shifted in the frequency domain. The removal of time delays or phase shifts from response signals forms the basis for all temporal array methods, such as the widely used beamforming technique [ref. 6]. These techniques have been in use for many years in the areas of satellite tracking, underwater acoustics, etc., and were the first application of array processing. A very simple application of a temporal array method is the estimation of the distance of a lightning bolt from a given location. By measuring the time between the lightning bolt and the sound of thunder, and knowing the speed of sound, it is possible to determine how far away the lightning bolt is.

Since temporal array methods are essentially a superposition of pressure waves, where a zero point crossing of a pressure wave is required to resolve it, they are limited to a source resolution of $\frac{1}{2}$ wavelength of the radiated sound source. Therefore, the techniques are only useful for noise sources which radiate frequencies whose wavelengths are small compared to the dimensions of the source. As an example, at 1000 Hz, the wavelength of sound is approximately 1 foot, so 1000 Hz sources can be resolved to within 6 inches. Since transient events tend to be impact-like in nature, they typically contain higher frequency content.

The advantage of acoustic arrays in this application, versus one or two microphone response locations, is the ability to acquire and process a large amount of spatial information. By processing spatial data, signals radiating from locations other than the selected candidate source location may be attenuated. The set of candidate source points are selected at locations suspected of producing transient noise. For a source image map output, the area of interest should be discretized to the desired source resolution. If a signal is emanating from a given candidate source point, the shifted microphone signals should sum in phase for an enhancement. If a signal is not radiating from that candidate source point, the shifted microphone signals should average to a background noise value. The amount of signal enhancement above the background noise depends upon the amount of spatial information used in the averaging process. If a total of N microphones are used, the signal is potentially enhanced above the background noise by a factor of \sqrt{N} .

The time delays, τ_{ij} , are calculated from the speed of sound c and the geometry of the microphone and candidate source point positions by

$$\tau_{ij} = \frac{r_{ij}}{c}$$

where r_{ij} contains the distances between the i candidate source points and the j microphones in the array.

The result of this technique is a set of enhanced pressures, one for each candidate source point, as a function of time. This is expressed mathematically as

$$P_i(t) = \left| \frac{1}{N} \sum_j X_j(t - \tau_{ij}) \right|$$

where $P_i(t)$ is the enhanced pressure for candidate source point i , N is the total number of microphones, X_j is the j^{th} microphone response, and τ_{ij} is the time delay between the i^{th} source point and the j^{th} microphone point. These enhanced pressures may be plotted on a time scale to compare the magnitudes for each source point as a function of time, or may be plotted at each individual time point spatially as a source image map. It is also possible to create a "movie" of the transient event, where each source image map at a point in time is one movie frame.

A weighting factor w_j may also be added in the summation of the above equation to define the shape of the array's enhancement, or to "calibrate" the technique to actual source magnitudes. Applying a factor of $w_j = 1$ for each microphone is equivalent to assuming that the array is placed in the farfield of point sources, thus seeing only planar waves. Applying a factor of $w_j = 1/r_{ij}$ is equivalent to assuming that the array is placed in the nearfield of point sources, thus detecting the curvature of spherically radiating waves. However, it should be noted that to get the time domain amplitude accurate within an average of 1.6%, at least 10 samples per cycle of the highest frequency of interest is needed, as was discussed earlier. This may be obtained by upsampling the enhanced pressure results calculated above.

RESAMPLING - The accuracy of the technique is dependent upon the accuracy of the time delays and the synchronization of the responses. It was shown [ref. 5] that time delay errors larger than ± 1 delta-t resulted in a form of spatial leakage and a breakdown of the technique. For this reason, it is very important to have an accurate speed of sound and accurate microphone and source point positions, which may be achieved with the use of a coordinate digitizer [ref. 7]. Once the time delays are computed, it is important that the microphone time domain signals are synchronized or shifted to the correct time points. This may pose a problem if the time delay values do not fall exactly on sampled time values, resulting in a possible maximum $\pm 1/2$ delta-t time delay error. This error becomes pronounced when the signal contains frequency content near the Nyquist frequency, where the phase errors are larger.

To eliminate this problem, a number of approaches may be taken. The time scale axis of the microphone signals may be re-sampled by upsampling using an FIR filter, thereby increasing the number of

time samples and reducing the time delay error. This results in an increase in the amount of data. The time scale axis may also be re-sampled so that the microphone signal time points fall exactly on the time delay value using an FIR filter, without increasing the amount of data. However, filtering the data results in passband ripple affecting the magnitude and filter roll-off affecting the frequency characteristics near the Nyquist frequency. These two approaches process the data completely in the time domain. Another approach is to shift the data in the frequency domain using the shift theorem. The microphone signals are transformed to the frequency domain, multiplied by an $e^{j\omega\tau}$ term to correct the phases exactly, summed for each source point, then inverse transformed back to the time domain. With this approach, the amount of data is not increased and the signals are shifted exactly to the correct time points. Since there are no passband ripple or filter roll-off effects, the data is not altered in magnitude in the passband or at the Nyquist frequency. However, there is a possibility of leakage due to the Fourier transform.

ANALYTICAL EXAMPLES - To evaluate the effects of re-sampling on the accuracy of the direct time domain acoustic array technique, two analytical examples are presented. In order to demonstrate the importance of re-sampling when the signals contain frequency content near the Nyquist frequency, one example is presented of a source near half the Nyquist frequency and one example is presented of a source near the Nyquist frequency. For these examples, the sampling rate was chosen to be 4096 Hz, resulting in a Nyquist frequency of 2048 Hz. The two sources were created by summing sine waves of amplitude 1 with random phase, applying an exponential window, and zero-padding the signals so that they appear as true transients. The first source signal summed sine waves of 800 Hz to 1200 Hz, while the second summed sine waves of 1600 Hz to 2000 Hz. Applying the exponential window and zero-padding the signals added damping, so the resulting signals contained slightly higher frequency content. To avoid any aliasing effects, the source signals were filtered using an FIR filter with a 60 dB drop at a cutoff frequency of 2048 Hz. The two time domain source signals are shown in Figures 8(a) and (b), while the Fourier transforms of the two signals are shown in Figures 8(c) and (d).

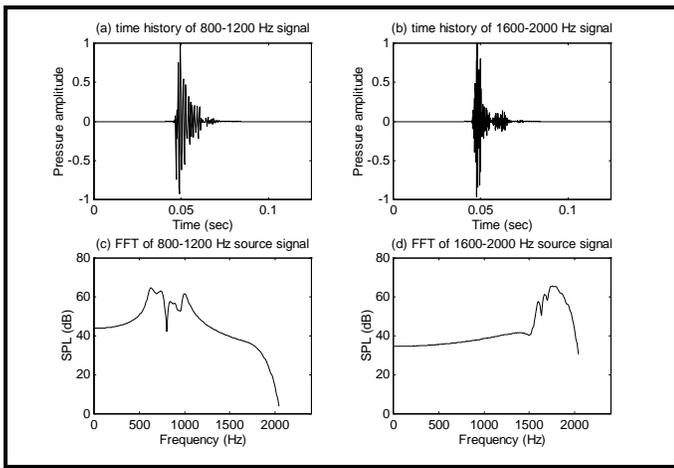


Figure 8: Time histories and FFTs of two source signals.

A 10x10 candidate source point grid was selected with an even spacing of 15.2 cm (6 inches). A 10x10 microphone array, also with an even spacing of 15.2 cm, was placed 15.2 cm from the source point grid. The time delay values were then calculated from the positions and the speed of sound. The two sources were both placed near the center of the source point grid and microphone time histories were simulated by delaying the source signals by the appropriate time delays. This was done using the Fourier transform approach.

The Fourier transform approach of shifting the microphone signals is essentially the same as re-sampling in the time domain to the exact time points using an FIR filter. The difference is that the Fourier transform method is quicker and has no filter effects, but has the potential of leakage. Since the analytical examples presented here are transient events completely observable in the time block, leakage is not an issue. Therefore, only the Fourier transform approach is used here for re-scaling of the time axis. It is compared with the case of not performing any time axis re-scaling.

If the time axis is not re-scaled to account for the fact that the time delays do not fall exactly on a sampled time point, there is a possible maximum $\pm \frac{1}{2} \Delta t$ error in the time delay. This error becomes more critical at frequencies near the Nyquist frequency. For this case, the first candidate source point was selected, the microphone time histories were shifted by the calculated time delays to available time points, summed, and made positive. This procedure was repeated for each candidate source point, resulting in a 10x10 enhanced pressure matrix for each point in time. Figures 9(a) and (b) show the comparisons of the enhanced pressures at the actual source point location and the actual source pressures as a function of time, for each source case. The maximum amplitude error for the lower frequency case is 7% and for the higher frequency case is 34%. The error in the energy, or the area under the curve, for

the lower frequency case is 8%, and for the higher frequency case is 34%. These comparisons are given in the frequency domain in Figures 9(c) and (d). Note the error above about $\frac{1}{2}$ the Nyquist frequency in both the lower and higher frequency source cases. The amplitude errors are approximately 1 dB at $\frac{1}{2}$ the Nyquist frequency and 5 dB at the Nyquist, which was seen in Figure 1. Since the higher frequency source is more affected, it was expected that the error for that case would be higher, which was seen in the 34% error (versus 7 to 8%). Figures 10(a) and (b) show the enhanced pressures as functions of time for all 100 source points, while Figures 10(c) and (d) show the enhanced pressures in the spatial domain at the maximum amplitude time value. While the source locations are clearly discernible, there is amplitude error, especially for the higher frequency source case.

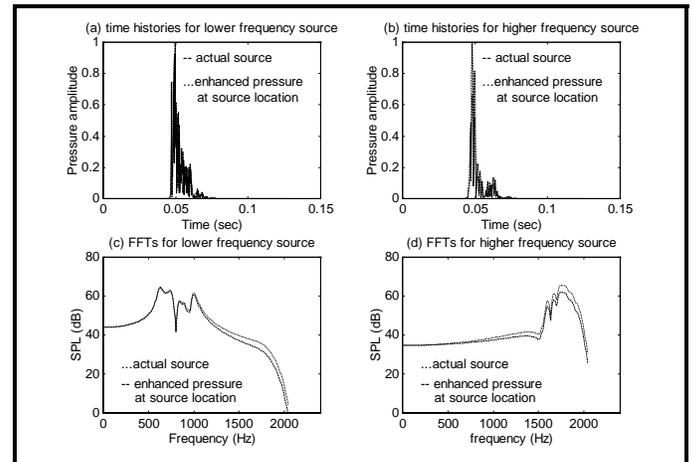


Figure 9: Time histories and FFTs of absolute value of actual sources and enhanced pressures at actual source locations. Original microphone time histories.

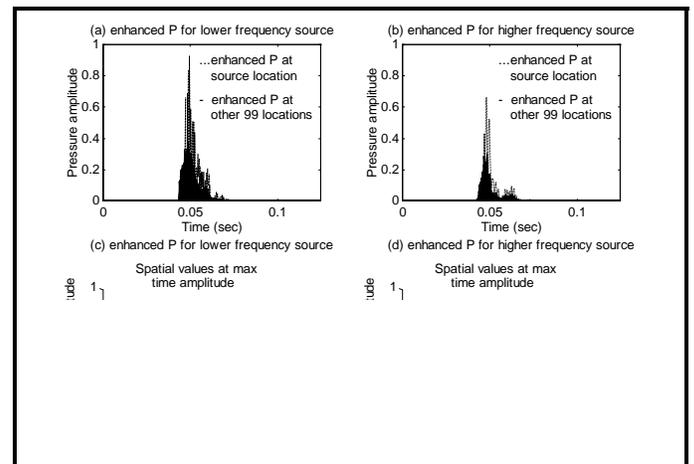


Figure 10: Enhanced pressures as functions of time and source image maps at maximum time amplitude. Original microphone time histories.

Using the Fourier transform technique to re-scale the time axis, the maximum amplitude errors are .0008% and .02% for the lower and higher frequency cases, respectively, as seen in Figures 11(a) and (b). The resulting energy errors of the enhanced pressures are .02% and .4% for the lower frequency and higher frequency sources, respectively. In viewing the frequency domain data, given in Figures 11(c) and (d), it is seen that the error near the Nyquist frequency has been significantly reduced.

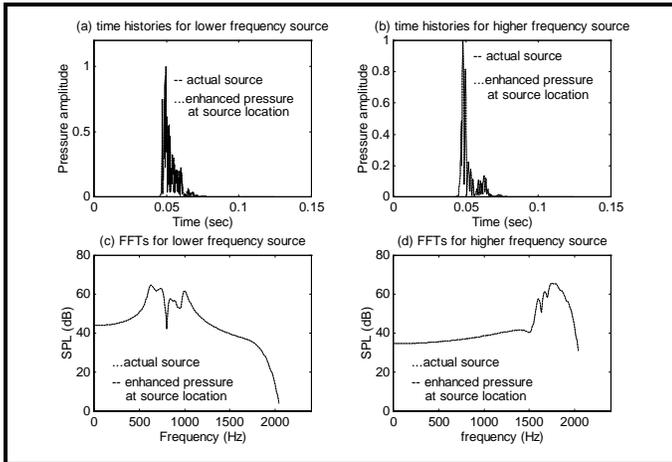


Figure 11: Time histories and FFTs of absolute value of actual sources and enhanced pressures at actual source locations. Re-sampled microphone time histories.

CONCLUSION

The details associated with averaging transient signals were presented. Several efficient data acquisition strategies were discussed for acquiring transient data and the necessary trigger signals for averaging.

A new data acquisition system was also proposed which eliminates many of the limitations and disadvantages current data acquisition hardware possesses. This new data acquisition system is based on time stamping the exact instant in time when a trigger event is detected. This acquisition system also allows data acquired on multiple separate acquisition systems to be processed as though it were acquired on the same acquisition system.

The major error encountered when averaging transient signals was shown to be the suppression of frequency information near the Nyquist frequency due to not synchronizing the acquisition of the responses with the start of the transient event. Several methods were presented which effectively re-sample the data to align data samples at specified instants in time which correspond to a trigger event or a time delay.

Examples of averaging both non-transient and transient data were presented which showed the

improvements achieved by time scale re-sampling. An example of averaging with sinusoidal noise added to the transient signal of interest was also presented. This last example clearly showed the effectiveness of averaging transient events before performing a time/frequency analysis when a significant amount of noise is present.

Finally, time scale re-sampling was applied to an acoustic array technique for analyzing transient noise sources, in order to improve the accuracy in the synchronization of the delayed time signals. The improvements in time scale re-sampling in this process were shown with analytical cases containing sources near $\frac{1}{2}$ the Nyquist frequency and the full Nyquist frequency. Estimation errors were reduced, from 7-8% in the lower frequency case and 34% in the higher frequency case, to near-zero values for both cases.

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