A Simple Method for Evaluating Audible Signals for Acoustic Measurements

by

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Abstract: Audible acoustic energy is attractive for use in positioning systems because of its ease of implementation with simple hardware and the ability for humans to setup and troubleshoot the system with the help of their own ears. Some signals however that historically have been used in acoustic positioning applications can be quite annoying to humans in the immediate area of the system. Such signals include chirps and pseudorandom noise. This annoyance can be reduced if the sounds employed are those of music passages or other similarly familiar sounds. The sounds however must possess appropriate autocorrelation and cross-correlation properties for them to be suitable for use in acoustic measurements. The sounds used in a positioning system can be produced and measured with common computer hardware and appropriate control and analysis software such as LabVIEW. This paper discusses the use of simple computer hardware together with elegant software to perform audible acoustic measurements such as position of the receiver and/or transmitters and determination of the speed of sound. Desirable qualities of sounds for use in an acoustic positioning system are discussed. Signals that have historically been used in such applications are discussed and analyzed. An algorithm for evaluating candidate music and/or sound signals is presented. An example of an acoustic measurement system employing a human-friendly sound clip is presented and discussed. Applications in the field and as an instructional tool are also presented.

I. Introduction

Acoustic measurements using signals in the ultrasonic range (>20 kHz) are very common in commercial and military applications [1, 2]. Ultrasonic signals are desirable because of their directionality and transparency to human detection. Typical ultrasonic positioning and measurement systems use short bursts produced by one or more acoustic transmitters. These bursts are received by tuned ultrasonic receivers. Using the speed of sound and the measured transit time of the burst from the transmitter to receiver, the resulting distance traveled by the burst (Time of Flight (TOF)) can be determined. Ultrasonic measurement systems typically use transmitters and receivers that are highly resonant. This resonant property of the transducers makes them highly frequency selective such that other acoustic noises in the area are rejected. The transmitted signals however are limited to bursts of a single frequency. The receiver is then simply looking for the presence or absence of an acoustic signal in its resonant frequency band.

Detection of the received ultrasonic signal is typically performed by a specially designed circuit [3, 4].

Audible acoustic signals (20 Hz to 20 kHz) can also be used in acoustic measurement systems [5-8]. As with ultrasonic systems, audible acoustic measurement systems primarily involve determining the time of flight of the acoustic signal. The methods of determining the time of flight are quite varied. There are several advantages, however, to working with audible signals.

One major advantage of using audible signals is that equipment for performing audible acoustic measurements is very inexpensive and readily available. Modern consumer electronics devices such as computers, personal digital assistants (PDAs) and home theater systems have built-in microphones and speakers and/or have ports for connection of external equipment. Another advantage is that systems employing audible signals can easily be setup and debugged with the help of human ears. Finally, audible acoustic energy is very prevalent in our environment. Although there exists a great deal of noise (which can be useful or harmful in acoustic measurements), there is a plethora of intentional acoustic energy such as tones, voices and music. This is the type of acoustic energy that is the focus of the work presented here.

II. Acoustic Measurements: Time-of-Flight

Most acoustic ranging and position measurements rely on the determination of the transit time of the signal from one point to another. This transit time is often referred to as the Time-of-Flight (TOF). The distance between the two points, d, and the TOF are related by the speed of sound, ν, as shown in Equation (1).

$$d = \nu \cdot \mathrm{TOF} \tag{1}$$

Figure 1 shows a simple transmitter-receiver setup. The acoustic energy is emitted by the transmitter (speaker) and is received by the receiver (microphone) at a distance, d, after a time delay equal to the TOF. If the speed of sound, ν, is known, the distance, d, can be calculated from the measured TOF. Conversely, if the distance, d, is known, the speed of sound, ν, can then be determined from the measured TOF.

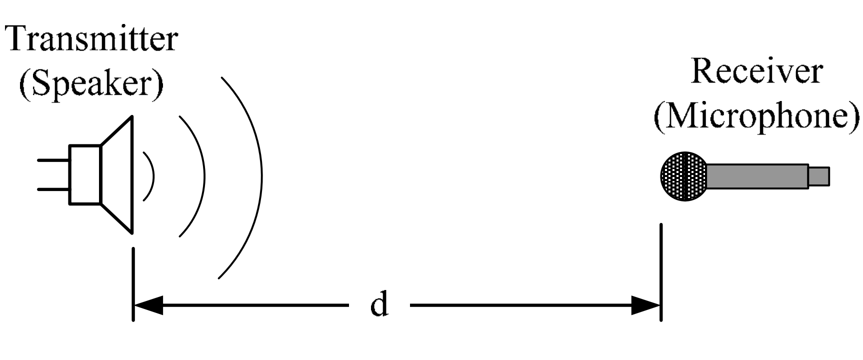


Figure 1. Simple Acoustic Measurement Setup

To perform an acoustic measurement as described above, the time at which the signal is transmitted and the time at which the signal is received must be known such that the TOF can be calculated. Figure 2 shows a measurement setup in which only the *difference* in the time of flight from each transmitter to the receiver need be measured to determine the receiver position along the constrained path between the transmitters.[7] As before, the setup of Figure 2 can also be used to determine the speed of sound if the transmitter-to-receiver distances, d1 and d2, are known.[6]

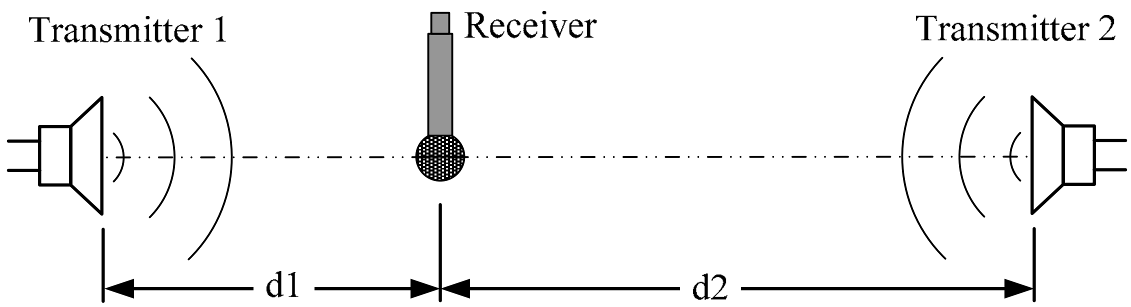


Figure 2. Simple Acoustic Measurement Setup Using Difference of Arrival Times

The setup shown in Figure 2 can be extended to two and three dimensions by including additional pairs of transmitters in various geometric configurations. Such a discussion is beyond the focus and scope of this paper.

III. Time-of-Flight Measurement

The previous section discussed how the time of flight of the acoustic energy can be used to determine the transmitter-receiver distance or the speed of sound. The problem now becomes that of *measuring* the time of flight when audible acoustic signals are used.

Unlike resonant ultrasonic transducers, typical audio range microphones have a wide bandwidth of sensitivity. This wide bandwidth permits the microphone to receive both the desired and undesired sounds present in the measurement system area. The detection of the intentionally transmitted sound(s) in the received sound is accomplished by further signal processing of the received signal. By performing a cross correlation of the transmitted and received signals, it may be possible to detect the transmitted signal if it possesses enough uniquely identifiable properties.

A way to quantify the uniqueness of a signal is to evaluate its *autocorrelation* function. The autocorrelation function of a signal with a high level of “Uniqueness” will exhibit a distinct central peak with quickly diminishing side lobe tails. Certain pseudorandom noise sequences exhibit excellent autocorrelation qualities. Historically, RADAR systems have emulated the echolocation techniques used by bats by utilizing simple *chirp* signals which also exhibit superior autocorrelation qualities [7, 9].

A chirp is a signal of short duration over which the frequency changes monotonically. Such a signal when created with audible frequencies produces a sound similar to that made by a bird chirping. A chirp with an increasing frequency is referred to as an *up-chirp* while a *down-chirp* signal exhibits a decreasing frequency. Figure 3 shows the autocorrelation of an up-chirp signal with unity amplitude, zero mean, initial frequency of 770Hz, final frequency of 1477Hz and duration of 50ms. This signal was sampled at 11.025 kHz with 8 bits of resolution.

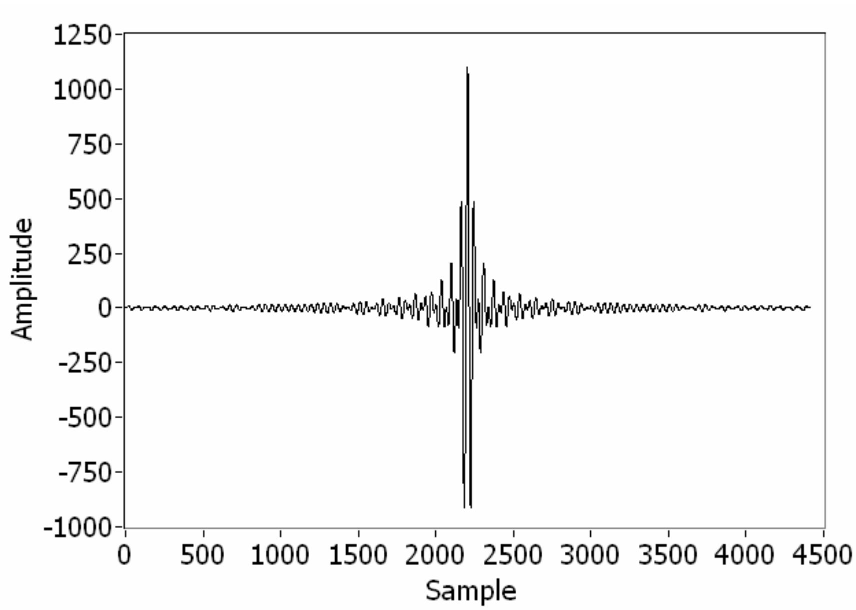


Figure 3. Up-Chirp Autocorrelation

If a system as shown in Figure 2 is used, the difference in arrival times of the signals from each transmitter must be measured. If the two signals are sent simultaneously and the receiver is located at the midpoint between transmitters, obviously, the signals will arrive at the same time. For this case it is therefore also crucial that the *cross-correlation* of the two transmitted signals exhibit no distinct peaks with magnitudes near that of the individual signal cross-correlations with the received signal. This requirement is necessary to allow the individual signals to be extracted from the composite received signal. Sending an up-chirp from one transmitter and a down-chirp from the other transmitter would satisfy this additional requirement.

Figures 4a and 4b show the cross-correlations of the received signal with an up-chirp and down-chirp respectively for distances of d1=d2=3 ft as shown in Figure 2. The time difference between the peaks of each cross-correlation is zero which is the expected time of flight difference for this configuration.

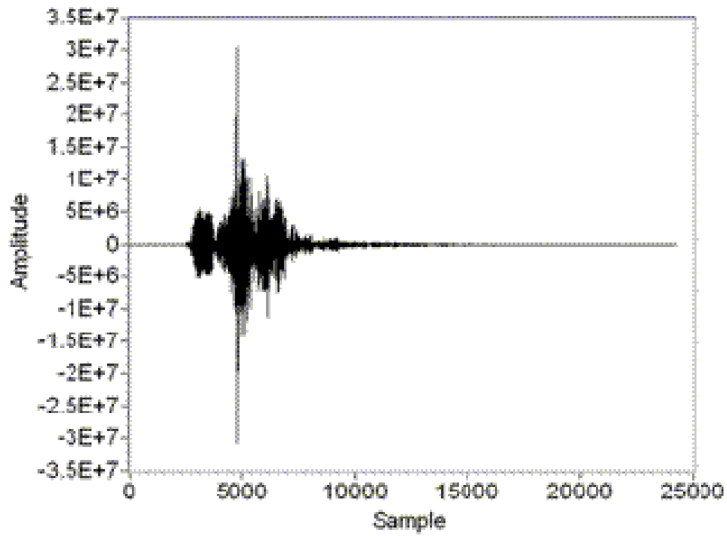


Figure 4a. Cross-Correlation of Received Signal and Up-Chirp

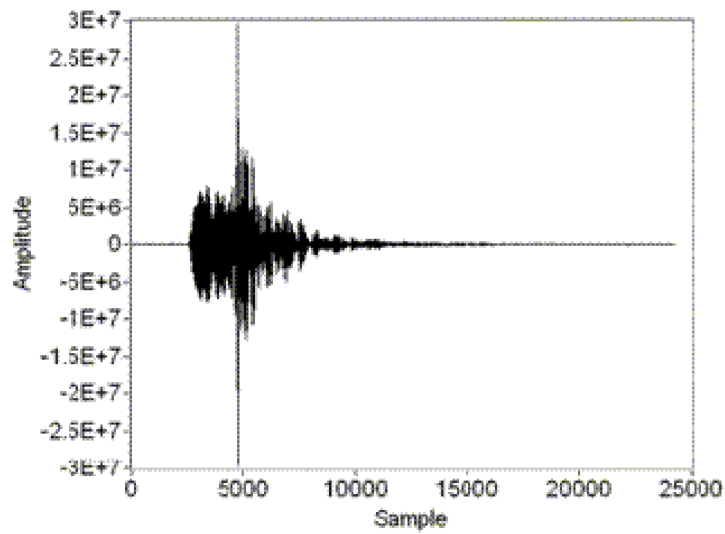


Figure 4b. Cross-Correlation of Received Signal and Down-Chirp

IV. Useful Qualities of Musical Sounds

The chirp signals used in the previous section produce excellent measurement results but their sound can be quite bothersome and fatiguing to humans in the measurement area for long periods of time. This situation can be improved if audible sounds can be found that possess appropriate correlation properties and are more pleasant sounding to humans. Some music passages have been found to make suitable acoustic measurement signals.

Music is very prevalent in digital formats such as .WAV and .MP3. These files can easily be played on the same computer equipment that can also contain powerful analysis software. Combining these functions allows music to be used as a source of acoustic energy for positioning and measurement systems.

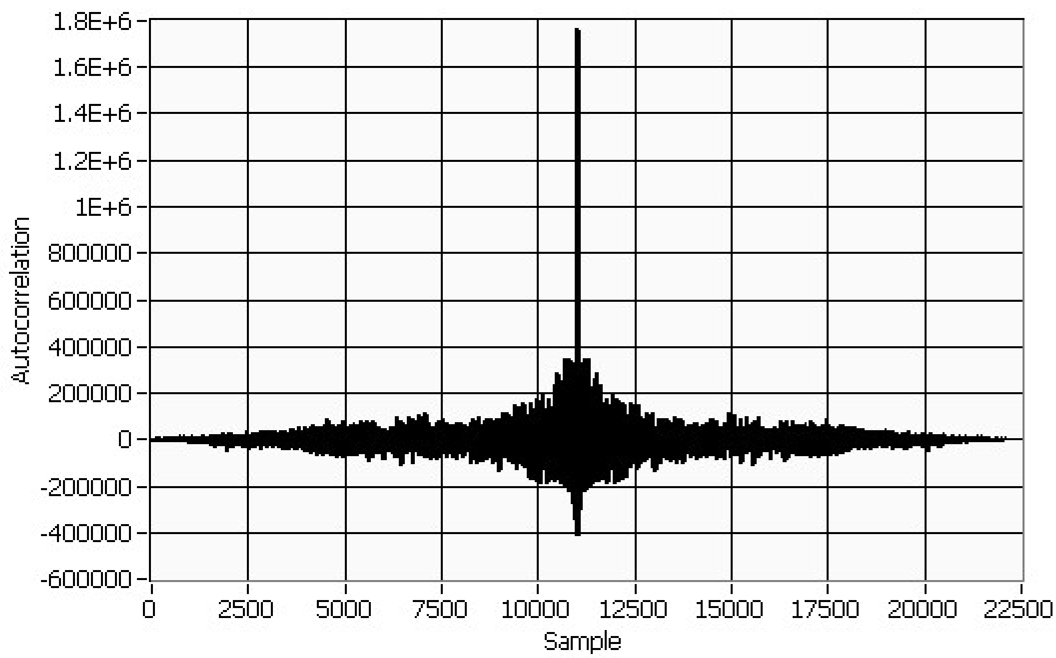


Figure 5. Autocorrelation of 1s Burst of "Margaritaville" Song

As an example, Figure 5 shows the autocorrelation of a 1.0 second excerpt (burst) of the song entitled, "Margaritaville" by Jimmy Buffet. Notice that this signal exhibits a very distinct central peak with quickly diminishing side lobe skirts.

After the autocorrelation of the sound bursts is evaluated, it is also important to determine the uniqueness of each burst in the temporal vicinity surround it. This quality of the burst is important in determining the aperture size used by the receiver in the acoustic measurement system. To determine the uniqueness of each burst in its neighborhood, the cross-correlation function is calculated.

V. Testing Candidate Musical Sounds

Candidate music must be tested to determine if it contains any passages with the appropriate signal properties. Figure 6 shows a block diagram of an algorithm to test the autocorrelation of sequential sections of a digital music file. Only monaural signals are discussed here for simplicity. The candidate music file is first placed into an array which is then sliced into short sections (bursts). The autocorrelation function of each burst is then calculated.

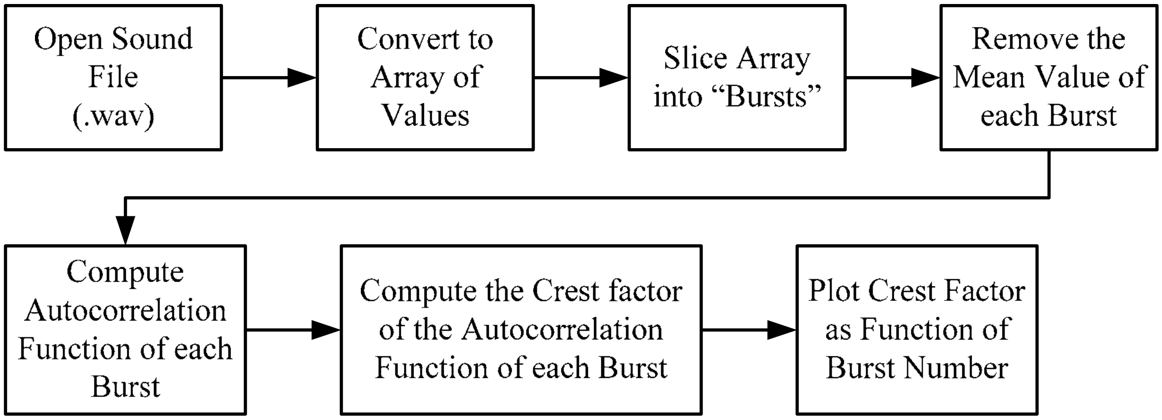


Figure 6. Block Diagram of Burst Autocorrelation Evaluation Process

The autocorrelation function produces an array of values as shown previously in Figure 5. In order to compare the autocorrelation functions of the bursts, the *crest factor* for each burst is calculated. Recall that the crest factor is defined as the ratio of a signal's peak value to its root-mean-square (RMS) value. The crest factor provides a value that is an indicative of the "Peakedness" of the autocorrelation function. The crest factor then provides a figure of merit for comparing the autocorrelation functions of the bursts and therefore their suitability for acoustic measurement use.

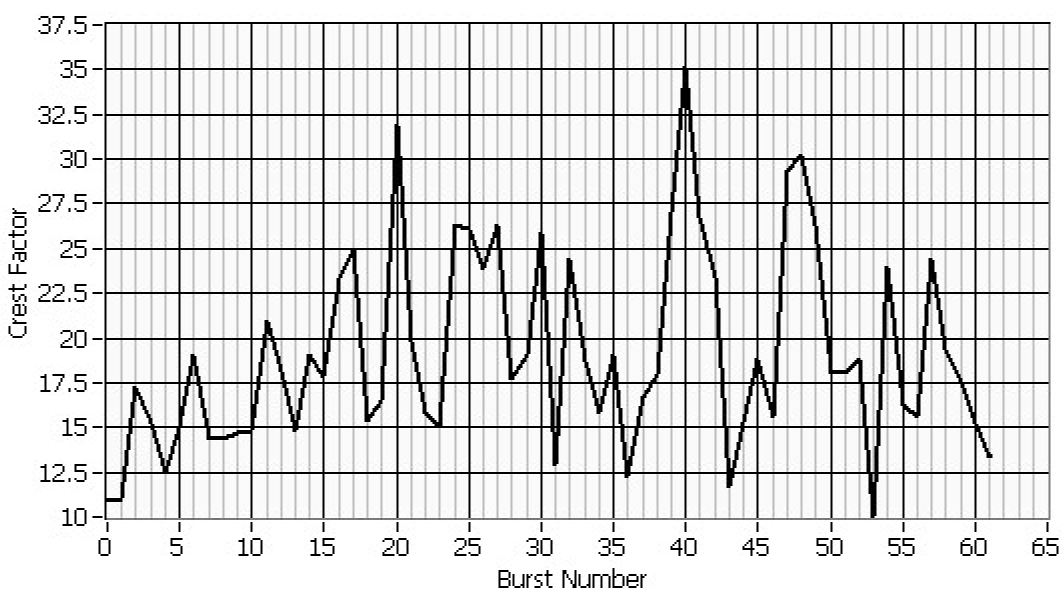


Figure 7. Autocorrelation Crest Factor Plot

Figure 7 shows a plot of the autocorrelation crest factors for the first 15 seconds of the song "Margaritaville" by Jimmy Buffet using 1.0 second bursts. The song clip was sampled at 11.025

kHz using 8 bits of resolution. Notice the wide range of autocorrelation crest factor values. The bursts with higher crest factors would be more unique and therefore may be better suited for acoustic measurement work. Figure 8a shows the autocorrelation function of the 40th burst of Figure 7 while Figure 8b shows the 53rd burst. Here the difference in the uniqueness of the bursts becomes quite apparent.

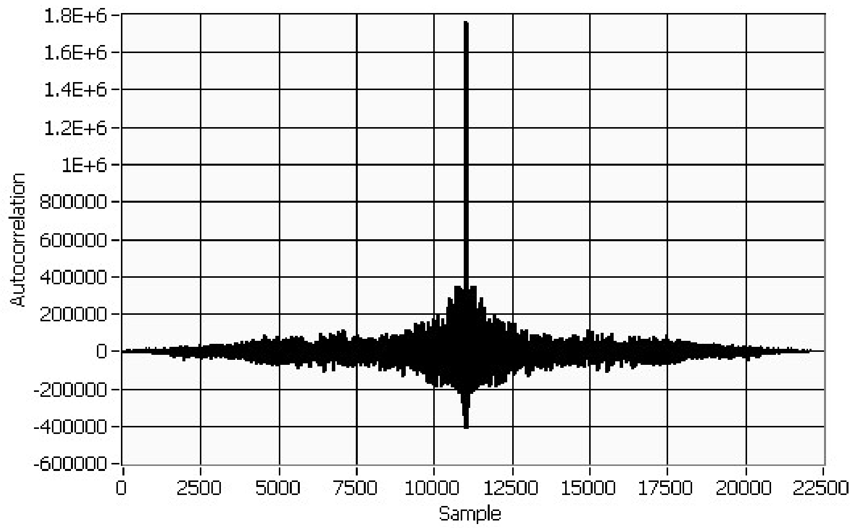
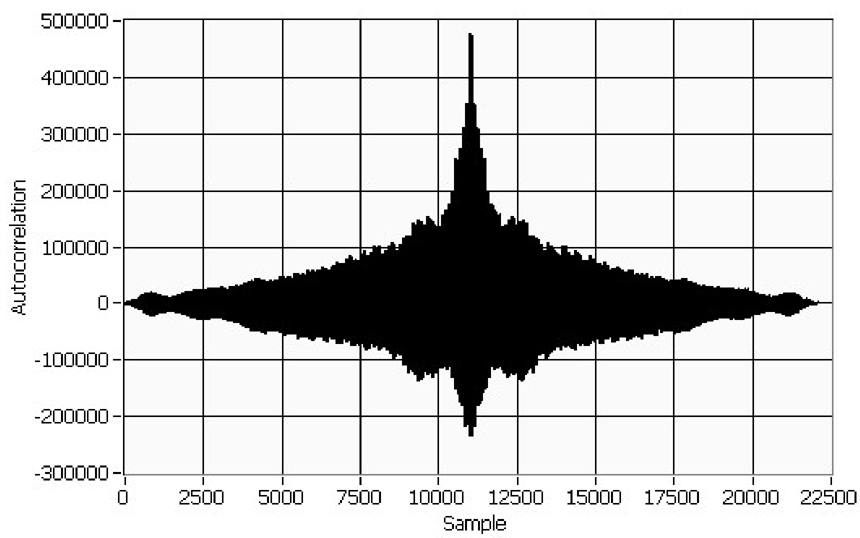


Figure 8a. Autocorrelation of 40th 1s Burst



Figures 8b. Autocorrelation of 53rd 1s Burst

The algorithm of Figure 6 can be applied to any candidate digital sound file. The profile of the crest factor plot can then be analyzed and the bursts that produce the relative peaks can be further investigated. The duration of the bursts can be adjusted to determine its effect on the autocorrelation function.

There is a tradeoff to be made regarding the length of the musical bursts. As the burst duration is increased, the autocorrelation function becomes more peaked as the uniqueness of the sound increases. Longer bursts however will require more memory and computation time in the acoustic measurement application. The LabVIEW front panel and block diagram of the LabVIEW virtual instrument used to conduct these tests are shown in Appendix A1 and Appendix A2 respectively.

VI. A Simple Acoustic Measurement System

Many configurations of transmitter-receiver geometry and signal timing can be implemented in an acoustic measurement system. A simple system for analysis purposes is that shown previously in Figure 2. In this system, the receiver (computer microphone) is constrained to motion along the straight line connecting the two transmitters (external computer speakers).

To estimate the position of the receiver in Figure 2, the difference in arrival time of the transmitted bursts must be determined. If a monaural signal is used, enough delay must exist between the arrival times such that the peaks of the cross-correlation function can be discerned. This delay can be either natural due to sufficient transmitter spacing or one channel can be delayed in software to produce the same effect.

Notice however that if a natural delay is employed, the region near the midline point must be avoided as this is where the difference of arrival times will be small. This situation can be avoided by introducing a delay in one channel via software that is sufficient to maintain the required minimum time of arrival difference even when the receiver is near one of the transmitters. The required delay is typically very small (~50 ms) and is in no way bothersome to humans and actually enhances the sound of monaural signals by producing a false sense of depth.

Figure 9 shows a block diagram for the process involved in a simple acoustic measurement system. A monaural music file that has been previously evaluated and the location of suitable bursts identified is used to create a stereo pair of signals in which one channel is just the delayed version of the other channel. These signals are sent to the computer’s sound card and played through the speakers.

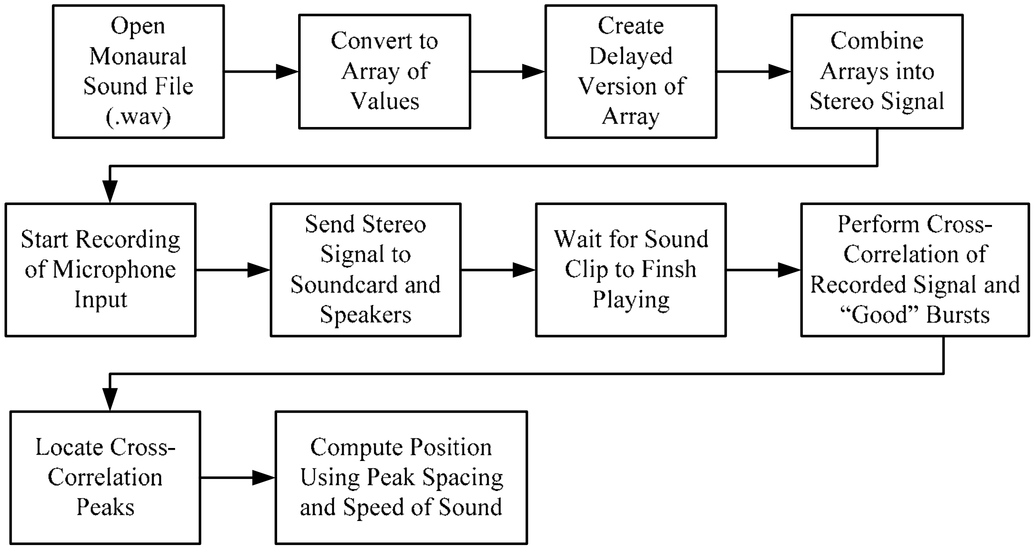


Figure 9. Block Diagram of Simple Positioning System Signal Processing

As the stereo signal is played, the microphone signal is recorded and stored in memory. At the completion of the sound clip, the recorded data is analyzed using the cross-correlation of the recorded signal with the known burst waveform to extract the relative time of arrival of each of the desired bursts; both the original and the delayed version. The time of arrival difference can then be determined and therefore an estimate of the receiver location can be calculated.

Figure 10 shows the actual cross-correlation of the received signal (the entire sound clip) and burst number 40 of "Margaritaville" for a system as shown in Figure 2 with d1 = d2 = 10 ft (the microphone is at the midline) and an intentional delay of 550 samples (~50ms) introduced into one channel. The difference between the two obvious peaks (550 samples) is extracted from this data and used to determine the position of the microphone. In this example, the right channel music signal (Margaritaville) was delayed by 550 samples with respect to the left channel which corresponds well with the measured cross-correlation peak spacing for the microphone at the midpoint. As the microphone is moved toward either speaker, the peak spacing changes accordingly.

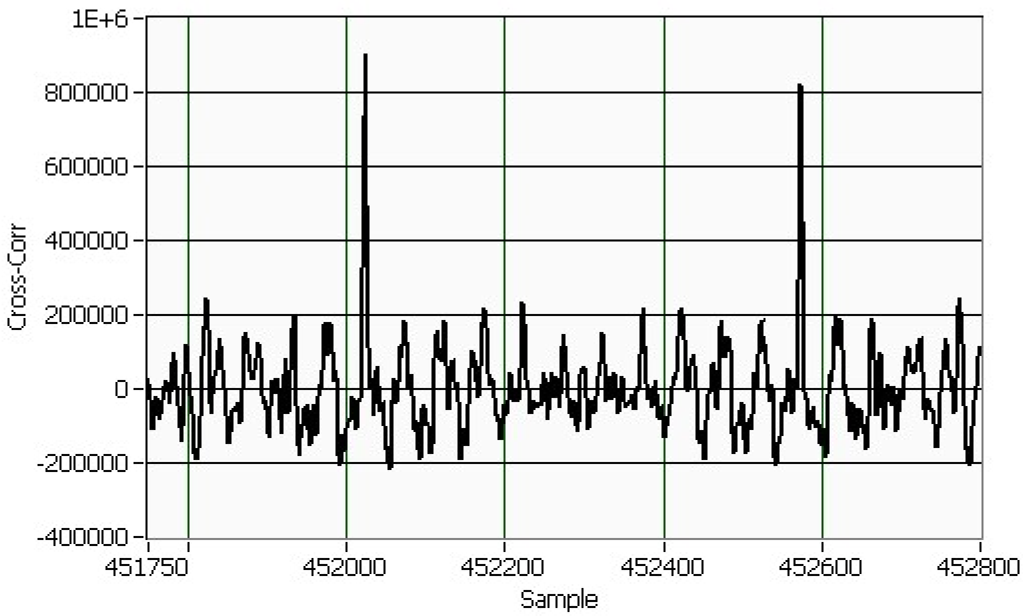


Figure 10. Cross-Correlation of Burst #40 with Recorded Signal

The actual temporal window width (aperture time) surrounding each burst determines the portion of the recorded signal that is used in the cross-correlation computations. Depending on the characteristics of the sound clip, the aperture could be the entire length of the recorded signal or just a small portion near the burst. As before, a large aperture, like a long burst will result in more memory usage and longer computation time.

VII. An Example Measurement

The simple system described in the previous section was implemented using a computer external microphone and speakers that are typically included with modern systems. An extension cable was used to allow the speaker separation to reach 20 feet. LabVIEW software was used to control the hardware, manipulate the sound file and perform the signal processing.

The monaural "Margaritaville" sound clip was modified to produce a stereo signal in which the right channel was delayed by 550 samples (at 11.025 kHz) with respect to the left channel. This stereo sound clip was played through the speakers. Simultaneously, the sound was recorded by the microphone as it was slowly moved along the line between the speakers.

Analysis was performed by calculating the cross-correlation function of the recorded signal with various 1 second bursts. Bursts with a 1 second autocorrelation crest factor greater than 20 were found to give acceptable results. The LabVIEW software located the two peaks corresponding to the left and right channel signals. The difference between these peaks, minus the intentional delay of 550 samples, gives the difference of arrival times of the signals. From this and the speed of sound (~340 m/s), the position of the microphone can be estimated. Table 1 shows the results of the measurements.

Burst Number	Estimated d1-d2	Known d1-d2
11	-8.1 ft	-8 ft
17	-6.1 ft	-6 ft
20	-3.9 ft	-4 ft
24	-2.0 ft	-2 ft
27	0.0 ft	0 ft
30	1.9 ft	2 ft
40	4.0 ft	4 ft
47	5.9 ft	6 ft
57	7.9 ft	8 ft

Table 1. Example System Test Results

VIII. Applications

The data presented, indicates that excellent positioning results can be obtained with simple hardware and familiar musical sounds. The system environment can be expanded by implementing a wireless microphone as the receiving device allowing the computer to be remotely located from the measurement area. The computer sound output can also be connected to a stereo public address system or similar sound system to increase the channel separation distance. Such a system would then allow for locating the position of the object/person carrying the microphone within a building environment such as a museum or commercial store while background music is playing as is typically done. By knowing the location of a user, specific information or services can then be personalized and delivered to that location.

There are also numerous applications of the positioning system described here in educational environments. The simple system used in the example can be used to demonstrate how the speed of sound in air can be measured using common computer hardware. The concepts of the Global Positioning System (GPS) can also be demonstrated with this simple system. Also, signal processing concepts and techniques including auto and cross correlation can be demonstrated using the hardware, software and sounds presented and described here.

IX. Conclusions

A method for screening sound clips for possible use as the signals for acoustic measurements has been presented and discussed. Musical sound clips have been chosen for use in audible acoustic measurements to allow the use of simple equipment while still keeping the annoyance to humans at a minimum. The uniqueness of a candidate sound can be evaluated by performing computing the autocorrelation of small sections (bursts) of the sound clip. The autocorrelation functions of the bursts can then be compared quantitatively by computing the crest factor of each function. The user can then set a threshold of burst crest factor values above which suitable results can be obtained. The data from a simple example system shows reasonable accuracy for applications in room-sized environments.

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Biographies

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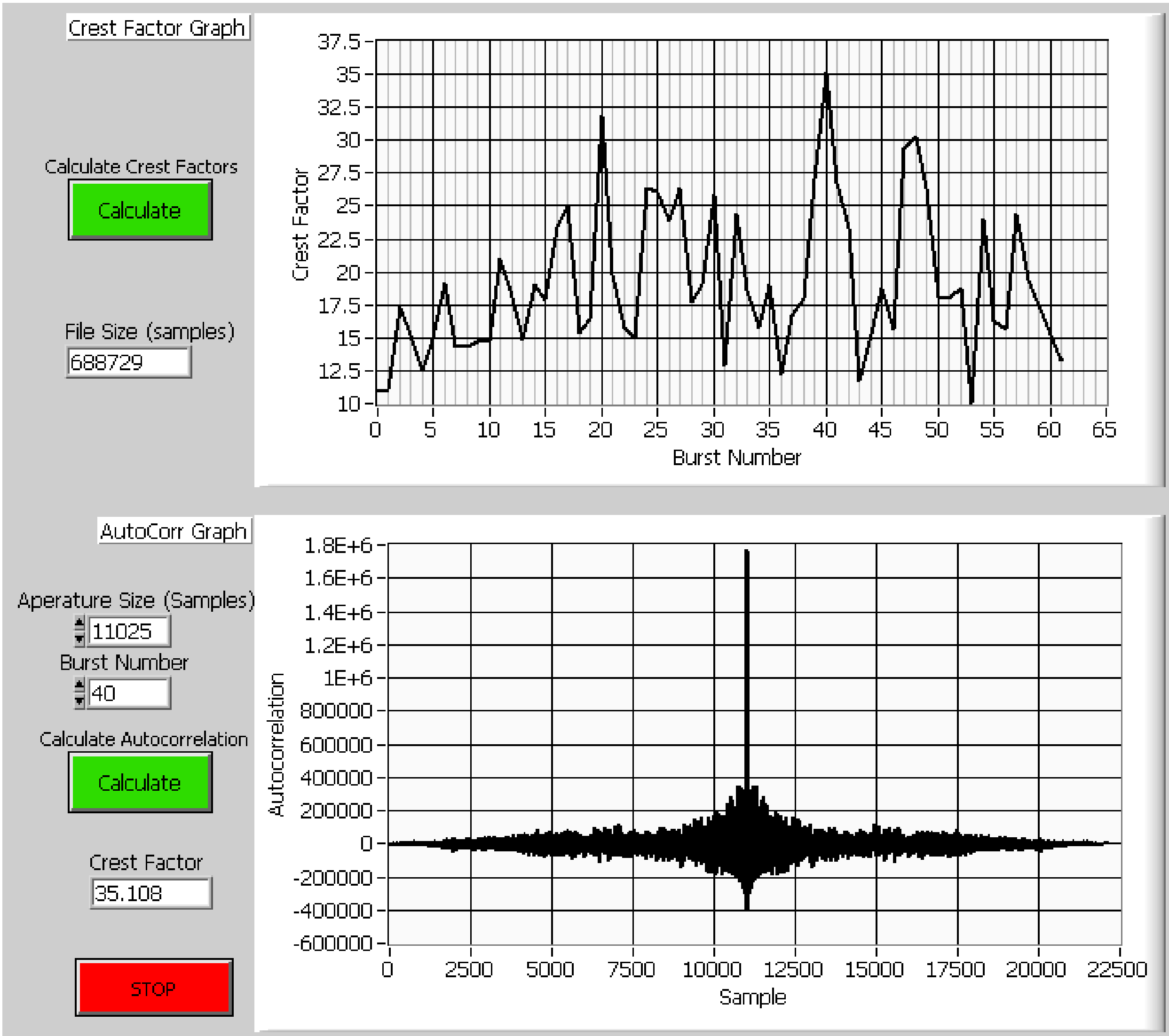
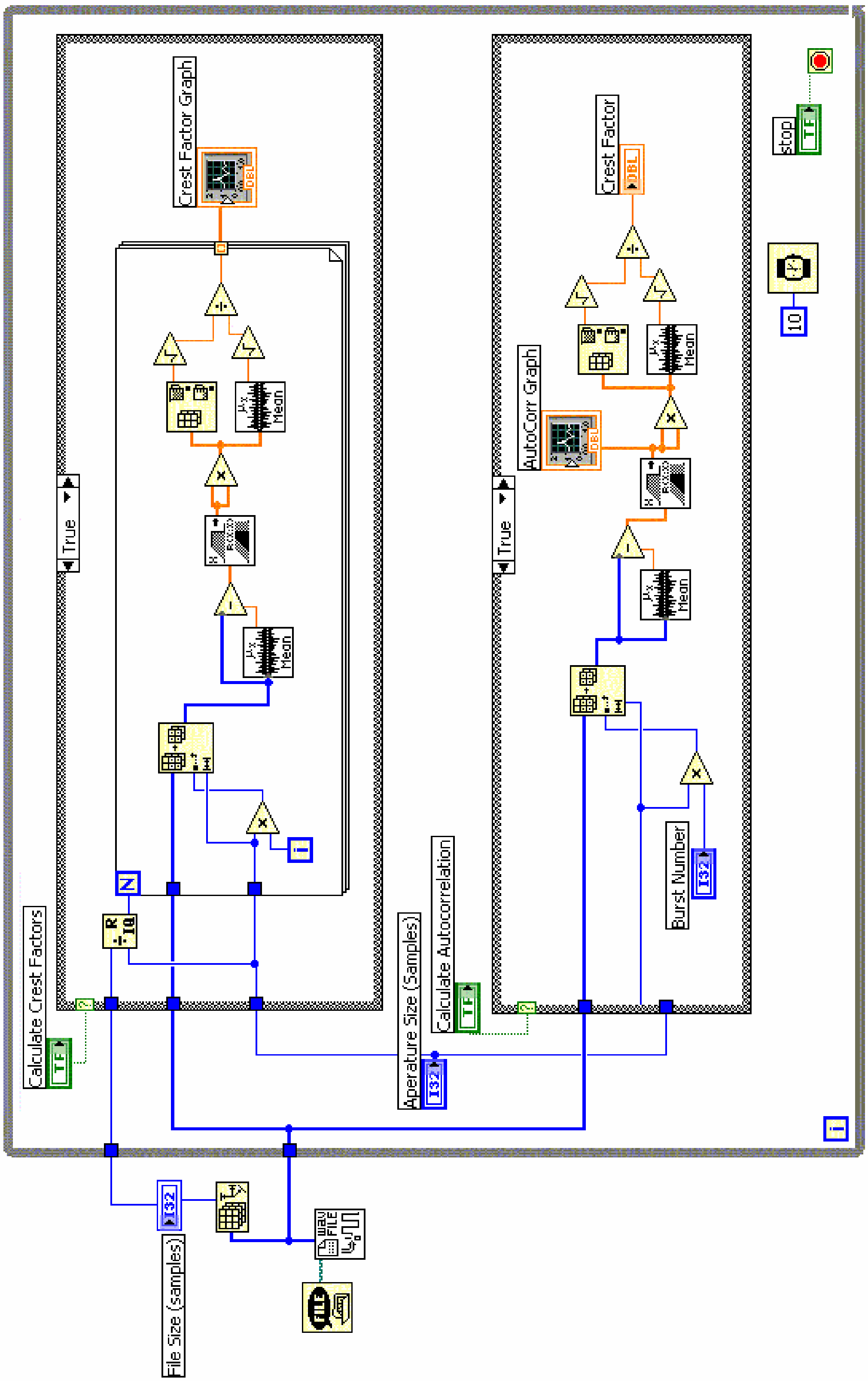


Appendix A1. LabVIEW Front Panel for Music Signal Evaluation VI.



Appendix A2. LabVIEW Block Diagram for Music Signal Evaluation VI.