ABSTRACT

The classical methods of measuring acoustic absorption coefficient using an impedance tube and a reverberation chamber are well established [1, 2]. However, these methods are not suitable for in-situ applications. The two in-situ methods; single channel microphone (P-probe) and dual channel acoustic pressure and particle velocity (Pu-probe) methods based on measurement of impulse response functions of the material surface under test, provide considerable advantage in data acquisition, signal processing, ease and mobility of measurement setup. This paper evaluates the measurement techniques of these two in-situ methods and provides results of acoustic absorption coefficient of a commercial artificial Astroturf, a Dow quash material, and a grass surface. The single channel microphone method uses impulse response calculations from a Maximum Length Sequence (MLS) signal excitation of an electro-acoustic loudspeaker and a Fast Hadamard Transformation (FHT) based cross-correlation algorithm by the de-convolution of recorded single channel microphone signal and the input MLS signal. The dual channel Pu-probe method is based on calculation of the complex impedance of the material surface under test from the frequency response function between the sound pressure and particle velocity time domain signals measured simultaneously at the same position on a material. The complex reflection coefficient calculated from both these methods further provides the acoustic absorption coefficient of the material under test.

INTRODUCTION

As noise pollution continues to attract attention of legislators and noise control engineers worldwide, two novel in-situ techniques are becoming available to provide solutions for measuring the acoustic absorption coefficient of acoustic materials after installation in many applications including automobile and other industries requiring the use of acoustic materials. These in-situ methods are novel and do not require the use of the impedance tube and the reverberation chamber. In addition to computing the acoustic absorption coefficient of the materials accurately and fairly consistently, these methods also provide considerable advantages in ease of measurement setup, low equipment cost, short data acquisition and post processing time, and require no specific and time consuming material preparation as required by the traditional methods.

Presently, the Astroturf surface is used in the estimation of the sound power of machines like the lawn mowers or garden equipments operating on grass surfaces. These machines are operated in a hemi-anechoic chamber on artificial turf surfaces to simulate its operation on grass surface. However it is proposed that the boundary conditions and outdoor environment is different from the simulated indoor environment since the actual grass and soil conditions may not be simulated correctly using artificial surfaces. Hence there is a need to estimate the acoustic absorption coefficient of both grass and artificial turf surfaces.

The absorption property of any material is characterized by its acoustic absorption coefficient defined as the fraction of the incident energy that is dissipated in a material. The acoustic absorption coefficient is a frequency dependent unit-less real number between 0 and 1 while for surfaces and rooms, the acoustic absorption coefficient defined in Sabins (sq. meters) is the product of surface area of the room and the acoustic absorption coefficient of the material covering the surface of the room.

To control the acoustic absorption coefficient over a required frequency range of interest, recently hybrid materials and the development of tailored solutions has become increasingly important with the emergence of multifunctional materials [3]. The commercial acoustic materials used in engineering applications consist mostly of multi-layered absorbers, dense rubber barriers, foam absorbers made of polyurethane and melamine-formaldehyde [4] and the relatively recently developed Astroturf and quash materials.
The phenomenon of sound absorption and transmission in materials is attributed to the dissipation of incident acoustic energy by fluid shearing and heat transfer within the oscillatory viscous and thermal boundary layer in open or closed-cell structures of the materials [5]. However, these theoretical models that estimate the acoustic properties of the natural surfaces are continuously under development, since the changes in the meteorological conditions like temperature, humidity, soil water content and environment are usually hard to track using any single method. Though certain attempts have been made to develop the models for measuring the acoustic properties of material surfaces [6], it is equally hard to estimate the effect of a combination of the environmental changes on the acoustic absorption coefficient of both the natural and artificial surfaces.

Under such circumstances, only the in-situ technique is capable of experimentally characterizing the acoustic properties of material surfaces installed in applications with specific configurations and used under varied or controlled meteorological conditions as in the transport industry or power plant environment. Attempts have been made for using a portable impedance tube and a mobile reverberation chamber scaled down in size for in-situ measurements [7]. The impedance tube is based on plane wave propagation from a sound source, so only the normal incidence acoustic absorption coefficient of a material sample is measured. While the reverberation chamber technique quantifies the random incidence acoustic absorption coefficient of a material surface under test. However the techniques based on the single channel microphone method and the dual channel Pu-probe method measure both random incidence as well as the normal incidence acoustic absorption coefficient. Apart from this, the implementation of these techniques is relatively easy with regards to setup parameters and is less expensive. Also with additional transducers arranged in required orientations, it is proposed that the acoustic absorption coefficient for both normal as well as a complete range of random incidences could be measured in a single acquisition period of less than a minute of time.

**SINGLE CHANNEL MICROPHONE IN-SITU TECHNIQUE (P- PROBE)**

The in-situ method outlined in International Standards Organization (ISO) 13472-1, 2002 standard [8] has been widely popular for measuring the acoustical absorption coefficient of road surfaces. In this method, only one microphone with a sound source is used to find the acoustic absorption coefficient of a material. This makes it inexpensive as compared to the conventional methods and can be used to find the acoustic absorption of material surfaces installed in an application in actual field conditions or in-situ.

**THEORY**

The in-situ method is based on the excitation of a Linear Time Invariant (LTI) electro-acoustic system by a data acquisition board / system. The system consists of a loudspeaker, a free field microphone, and a material surface to be tested by a broadband, deterministic and stationary signal. The input signal can be concatenated end to end periodically to excite the speaker. The microphone periodically records the sound pressure time domain signal of the complete excited system that is used for calculation of an averaged and periodic impulse response function (IRF) of the LTI system. This IRF is further used for calculation of the acoustic absorption coefficient of the material surface under test. The acoustic absorption coefficient resulting due to slight variations of boundary conditions of the material surface and the atmospheric space within constituents of the LTI system may be averaged to calculate reliable and fairly consistent results.

From the knowledge of digital signal processing, it is known that the circular cross correlation of an input signal, \( \Phi_{xy}(k) \), equals the auto correlation of the input signal, \( \Phi_{xx}(k) \), convolved with the impulse response function of the LTI system, \( h(k) \). A signal called Maximum Length Sequence (MLS) signal has an important property that simplifies this calculation of the impulse response function by a great amount. Its auto correlation function approximates the Dirac delta function, \( \delta(k) \), hence the signal processing for the average IRF of the LTI measurement system reduces to just the calculation of average of the periodic circular cross-correlation between the microphone output signal and the concatenated input MLS signal as shown in Equations (1).

\[
\phi_{xy}(k) = \phi_{xx}(k) \otimes h(k) \\
\phi_{xy}(k) = \delta(k) \otimes h(k) = h(k)
\] (1)

The MLS signal is generated from a set of linear feedback shift registers by choosing specific feedback taps. Equation (2) gives the length of a MLS signal where \( m \) is a positive integer called as the power of the MLS. The length of the MLS signal is required to be greater than the length of the IRF to be measured. More information about the MLS signal can be found from [9 and 10].

\[
L = 2^m - 1
\] (2)

To create a voltage signal from the MLS sequence to be fed to the sound source, the zeros are mapped to ones and the ones are mapped to negative ones. One of the benefits of using the MLS signal is its low crest factor defined as the ratio of the peak voltage to the average RMS voltage of the MLS signal. It is proposed that a low crest factor helps in calculation of accurate results even in the environments with lower signal to noise ratio [12]. However the inherent disadvantage of the MLS signal is slew distortion of the calculated IRF due to continuous time derivative changes during cross correlation of the MLS signal that merely consists of negative and positive
ones. The adverse effect, if any, due to a temporal energy distribution created in the form of slew distorted peak in the IRF is currently unknown, however, limiting the amplitude of this peak places two constraints. First, it requires proper impedance matching between the data acquisition system output and the input of the speaker and next, the maximum amplitude of the input MLS signal provided to the sound source for a better signal to noise ratio needs to be controlled. [10].

ALGORITHM

Figure 1 shows a graphical representation of the entire algorithm of this method. Once the MLS signal is generated it is concatenated end-to-end and applied to the loudspeaker periodically for excitation. Each of the periodic record of the microphone signal is used to calculate the IRF from a mathematical tool called the Fast Hadamard Transform (FHT). Chu [12] provides an elegant example of implementation of the FHT and Gumas [13] provides an excellent algorithm for the application of FHT to any signal. A major advantage of using this transform is that the number of mathematical operations are reduced from N*(N-1) multiplications to N*log10(N) mathematical operations of addition and subtraction, making the calculation time efficient.

After the IRF h(k) is calculated, there is still the problem of separating the impulse responses within the IRF of the system. For each IRF that is measured against a material surface under test, there will be two impulses followed by parasitic reflections after some time. The first of these two impulses is an incident impulse due to the sound wave incident on the microphone from the loudspeaker. The second impulse represents the sound wave reflected from the material surface to the microphone after some time. Since the microphone is placed close to the surface, these two impulses overlap. Hence a simple time domain separation of these two impulses is not possible. However, a measurement in free field is often necessary with the loudspeaker and microphone dimensions and the digital signal processing parameters kept same. After application of the FHT to this recorded microphone signal, the resulting IRF will only contain a single impulse corresponding to the incident sound wave. The incident impulse can be subtracted from the IRF of the system under test to obtain only the reflected impulse from the material surface. One of the requirements of the time domain signal subtraction is that the data acquisition system should have high time domain resolution i.e. a very high sampling rate of the order of 50 KHz so that the shape of the impulses is properly captured for signal subtraction. Also, it must be ensured that the distance of the microphone from the sound source in both configurations remains strictly constant.

After the incident and the reflected impulses are separated using time domain signal subtraction, each impulse is time windowed to eliminate the slew distorted peaks and the parasitic reflections. A half Blackman-Harris time window is suggested in this step [14]. The reciprocal of the length of the time window determines the low frequency limit of the acoustic absorption coefficient. Also, the length of the time widow, Tw, the distance of the speaker from the test surface, ds, and the distance of the microphone from the test surface, dm, determines the maximum radius of the sampled area, Rmax given by Equation (3). The speed of sound in air is denoted by c.

\[
R_{\text{max}} = \frac{1}{d_s + d_m + cT_w} \left( \frac{d_s + cT_w}{2} \right),
\]

After the time windowed IRFs of the incident and the reflected waves are found, the FRFs of the system can easily be found by applying the Fast Fourier Transform (FFT) to the IRFs. These FRFs are further used to calculate the acoustic absorption coefficient using Equations (4) and (5) where Kr is called the geometrical spreading factor that accounts for the difference in path lengths between the incident and the reflected waves. Qw(f) is the sound power reflection coefficient, a frequency dependent function and the H_i(f) and H_r(f) are the FRFs of the incident and the reflected IRFs.

\[
K_r = \frac{d_s - d_m}{d_s + d_m},
\]

\[
\alpha(f) = 1 - Q_w(f) = 1 - \frac{1}{K_r^2} \left| \frac{H_i(f)}{H_r(f)} \right|^2
\]

The acoustic absorption coefficient can further be expressed on a one-third octave band using Equation (6) where,

\[
\alpha(f) \text{ is the acoustic absorption coefficient of the material surface under test.}
\]
\( \alpha_i(f) \) is the acoustic absorption coefficient in each one-third octave band,

\( f_i \) is the starting frequency of the one-third octave bands

\( \Delta f_i \) is the bandwidth of the one-third octave bands, and

\( R(f) \) is the sound power reflection coefficient compensated with the geometrical spreading factor.

\[ \alpha_i(f) = 1 - \frac{1}{\Delta f_i} \int_{f_i}^{f_i+\Delta f_i} |R(f)|^2 df \]  

(6)

**MEASUREMENT SETUP AND DSP PARAMETERS**

The in-situ method was implemented using a Zircon unit consisting of a LS14 loudspeaker unit, a CA12 loudspeaker cable assembly, a DPA 4060-B prepolarized miniature condenser microphone with a microdot connector, a commercial power amplifier, and a Zircon F-stand for floor measurements [15].

![Figure 2: Single microphone in-situ measurement setup](image)

Figure 2: Single microphone in-situ measurement setup

Figure (2) shows measurement chain used to measure the acoustic absorption coefficient using this method. The Zircon speaker has a flat FRF that does not negatively affect the FRF of the complete test system. The dimensions of \( d_s=1.25 \text{ meter} \), \( d_m=0.25 \text{ meter} \) result in a maximum radius of \( R_{\text{max}}=1.3 \text{ meter} \) if a time window of 5 milliseconds is used in the signal processing. These dimensions were kept strictly constant throughout all measurements. A laptop sound card based data acquisition system was used to generate the input MLS signal and the Mathworks MATLAB environment was used to record the microphone voltage signal.

A MLS signal of size \( L=2047 \) sampled points obtained from power \( m=11 \), was used as input to the system with a sampling frequency of 44.1 KHz of the sound card. This produced an input signal of duration 46.395 milliseconds and was considered long enough to record the entire IRF of the test system that was found to be lower than 5 milliseconds. Also the sampling frequency of 44.1 KHz of the sound card provided enough temporal resolution to capture the shape of the IRF for proper signal subtraction in the signal processing. After signal subtraction of the two IRFs, the incident and the reflected impulses were time windowed using a 5 milliseconds half Blackman-Harris time window resulting in a low frequency limit of 200 Hz.

![Figure 3: Single microphone in-situ method GUI](image)

Figure 3: Single microphone in-situ method GUI

A graphic user interface shown in Figure (3) was custom programmed utilizing MATLAB environment. This software has the following functions:

1. Choice of a configuration of measurement setup from a set of 3 measurement setups developed from laboratory and commercial equipments.
2. Creation and storage of MLS signal of required power, length and sampling rate for output through the laptop sound card.
3. Measurement of free field and global IRFs, separation of the reflected impulse from the global IRF using signal subtraction technique and application of time window.
4. Computation of the incident and reflected FRFs by performing FFT on the time windowed IRFs.
5. Numerical and graphical computation and display of the sound absorption coefficient based on the computed FRFs in one-third octave bands.
6. Options of report generation using a single click operation that results in a detailed report in the hypertext markup language file for future records.
7. Functions for storage of the dimensions of the test setup and atmospheric conditions along with location, date and time of each measurement as required by the ISO 13472 standard.
8. An option of offline signal processing for previously recorded time domain measurement signals.
Another technique for measuring the acoustical absorption coefficient, in-situ, is based on the calculation of the complex impedance of a material surface under test. Measuring the particle velocity and sound pressure level simultaneously at the same place on the material by the exciting a sound source does this. This is called the Pu-probe method that uses an integrated particle velocity sensor and a miniaturized pressure microphone in a single compact size probe [16]. The Pu-probe offers a full bandwidth sweep up in the range from as low as 5 Hz to as high as 20 KHz.

THEORY

The Pu-probe based in-situ technique is similar to the single microphone type in-situ method discussed previously except that the Pu-probe replaces the single microphone. Also, any broadband, deterministic and stationary signal can be used for sound source excitation. Hence the system is not confined for use with only the MLS signal. This method is based on calculation of the specific acoustic impedance of the material surface from the acoustic excitation of a LTI electro-acoustic system consisting of a simple speaker and a Pu-probe.

The particle velocity transducer of the Pu-probe is composed of two micro-machined hot wire anemometers in which the resistance of the hot wire depends on the temperature distribution over it. A particle velocity signal due to acoustic excitation from a sound source in the perpendicular direction changes the temperature distribution of the hot wires of the transducer because the upstream wire is cooled more by airflow than the downstream wire. An electronic bridge circuit senses the resulting resistance difference in the two wires that provides a signal proportional to the particle velocity of the acoustic flow over the Pu-probe.

The frequency response function between the sound pressure and particle velocity is calculated for a material surface under test that is further calibrated to specific acoustic impedance of the material surface under test in ambient air conditions. This calibrated acoustic impedance is further used to calculate the acoustic reflection coefficient from which the acoustic absorption coefficient of the material can be calculated [16,17].

ALGORITHM

Four different signal types of signals; Logarithmic Sine Sweep, MLS, White Noise and Random signal, have been utilized for this test. However, experimental results with only the logarithmic sine sweep signal are presented. In this measurement setup, a concatenation of the input excitation signal is not required. Instead, a single length of the input signal greater than the length of the impulse response to be measured is used. The sensitivity of the particle velocity probe in addition to the noise external to the LTI system dictates the signal to noise ratio of the measurements.

The Logarithmic Sine Sweep signal was generated using a graphic user interface programmed in Matlab, which also had capabilities of generating other signals like MLS, Random and White noise. Once the signal is generated, it is applied to the sound source for excitation of the LTI system consisting of the speaker, Pu-probe and the material surface under test. The signal from the Pu-probe consisting of a sound pressure and the particle velocity time domain voltage signals are recorded using two channels stereo of a laptop sound card. The frequency response function between these two signals is used to calculate the reflection coefficient of the material surface under test that is further used to calculate the acoustic absorption coefficient of the material. Lanoye [16] and De Bree [17] describe the complete theory of these calculations from the first principles and the modeling of the Pu-probe measurement system.

Two measurements, similar to the single microphone in-situ technique; one in free field and other with the LTI system facing the material surface, are acquired for calculation of its acoustic absorption coefficient. After the frequency response function (FRF) between the sound pressure and the particle velocity signals from the Pu-probe signals for free field is calculated, it is inverse Fourier transformed to find the IRF of the LTI system. This IRF is time windowed to prevent the effect of any spurious reflections and is further Fourier transformed to the frequency domain to obtain the windowed FRF. The resulting function is the ratio of Cross Spectrum function of the sound pressure and the particle velocity to the Autospectrum function of the particle velocity. In essence, this is the complex impedance of the LTI system in the free field as shown in Equation (7).

\[
Z_{pu}(f) = \frac{S_{pu}(f)}{S_{uu}(f)}
\]  

(7)

Here,

- \(Z_{pu}(f)\) is complex impedance of the LTI system,
- \(S_{pu}(f)\) is cross spectrum of the particle velocity
- \(S_{uu}(f)\) is auto spectrum function of sound pressure level and particle velocity

This free field complex impedance of the LTI system from Pu-probe method is now calibrated by dividing it with the impedance of air as is calculated from Equation (8) to find the calibrated or the specific complex impedance of the LTI system.
Here,

\[ Z_{\text{freefield}}(f) = \frac{Z_{pu}(f)}{\rho_c} \quad (8) \]

\[ Z_{\text{freefield}}(f) \] is the Specific Impedance of the LTI system in the frequency domain,
\[ \rho_c \] is the density of ambient air, and \( c = \) speed of sound in air.

Another measurement is taken with the Pu-probe kept as close as possible to the material under test. Again, the FRF between the sound pressure and particle velocity time domain voltage signals of the system in this setup is found which is inverse Fourier transformed, time windowed and further Fourier transformed to obtain the complex impedance of the system, \( Z_{\text{reflec}}(f) \).

Now the impedance of the material surface \( Z_{\text{material}}(f) \) under test is found by dividing the complex impedance of the reflected surface with the calibrated free field impedance of the Pu-probe as given in Equation (9).

\[ Z(f) = Z_{\text{reflec}}(f) / Z_{\text{freefield}}(f) \quad (9) \]

The reflection coefficient of the material is calculated from Equation (10). The acoustic absorption coefficient of the material can be further deduced from Equation (11) and converted to third-octave acoustic absorption coefficient using Equation (6).

\[ R(\omega, \theta) = \frac{Z_c \cos \theta - \rho_c}{Z_c \cos \theta - \rho_c} \quad (10) \]

\[ \alpha(\omega, \theta) = 1 - |R(\omega, \theta)|^2 \quad (11) \]

MEASUREMENT SETUP AND DSP PARAMETERS

The Pu-probe in-situ method was implemented using the same Zircon unit consisting of a LS14 loudspeaker unit, a CA12 loudspeaker cable assembly, a commercial power amplifier, and a Zircon F-stand for floor measurements. However in this case, the Pu-probe replaces the microphone. An additional bridge electronic circuit developed by Microflown Technologies is used in series between the Pu-probe and the stereo laptop sound card.

The logarithmic sine sweep signal is processed from 200 Hz to 16 KHz. Since the upper frequency limit of the 8 KHz is valid with the ISO 13472-1 standard and the Pu-probe has valid frequency range of 5 Hz - 20 KHz, a signal with higher frequency beyond 8 KHz is chosen for the frequency sweep. With the sampling frequency of 44.1 KHz an input sweep signal takes about 3 seconds. The IRF is well within 5 milliseconds of the Hanning time window with which the sound pressure to particle velocity calibration FRF and material FRF is calculated.

To calibrate the Pu-probe, it is held at a distance of 20 centimeters from the sound source in its near field and logarithmic sine sweep signal is played to measure the time domain sound pressure and particle velocity signals [17]. The FFT of these measurements leads to their FRF from which the IRF is calculated using IFFT. The IRF is time windowed before calculating the free field complex impedance or calibration value of the Pu-probe. Care must be taken to overlap and align the IRF before calculating the FRFs so that the proper delay due to distance of the sound source and material from the Pu-probe is incorporated in the FRF calculation from the two time domain signals. The impedance and the acoustic absorption coefficient of the material are calculated using Equations (9) and (11).
A graphic user interface shown in Figure (5) was custom programmed in MATLAB. This software provided the many of the similar functions described earlier with the single channel P-probe method. In addition, the GUI,

1. Allowed choosing a type of signal to be generated and played from lower frequency to higher frequency limit,
2. Allowed choice of block length, sampling rate of sound card and other digital signal processing parameters and saves them for processing, and
3. Enabled the measurement of free field and material field sound pressure and particle velocity time domain voltage signals.

EXPERIMENTAL RESULTS

Figures (6) through (8) show some sample results obtained from both of the methods. The acoustic absorption coefficients of a porous quash surface and a grass-like artificial Astroturf surface are presented.

The acoustic absorption coefficients of the 1-inch thick quash surface from both P-Probe and P-u probe methods is shown in Figure 6. It is seen that both methods produce similar results in the frequency range of 400 Hz – 8 KHz. However, there is less correlation in the acoustic absorption coefficient in the low frequency range. One reason for this deviation could be due to the fact that in the single microphone based in-situ technique the calculation of acoustic absorption coefficient was not corrected with a reference surface measurement. This is supposed to increase the accuracy of the results as described in the ISO standard [8]. Though slight discrepancies in acoustic absorption coefficient exist in each one-third-octave band, an overall trend in the values is evident even with the highly porous and absorptive acoustic material.

The Astroturf surface results from Figure 7 show overall lower acoustic absorption coefficient in the frequency range of 400 Hz – 8 KHz as compared to the quash surface. Quantitatively, this is expected for a grass-like artificial acoustic material about 0.25-inch thick having thin nylon fibers of 0.5-inch length on its top. Figure (8) shows the acoustic absorption coefficient of the actual outdoor grass surface from the single microphone in-situ technique. The acoustic absorption coefficient for grass surface from the Pu-probe in-situ technique is not shown since the measurement data for grass was very sensitive to the wind flow over the probe in outdoor-conditions.

If grass is dry it is expected to show acoustic absorption coefficient as high as 0.7 even in the low frequencies and steadily rising to higher absorption coefficient in the higher range of frequencies. This is so for dry grass because the matted surface forms a dense combination of interleaved grass blades and the deep pore layer of the dry soil prevents the reflection of the trapped incident acoustic energy thus increasing the acoustic absorption. In contrast, the wet grass should show lower acoustic absorption coefficient in the low frequency range, rising steadily in the higher frequencies. This trend in the acoustic absorption coefficient of grass surface is found from measurements with the single microphone in-situ technique. The dual channel Pu-probe in-situ technique however suffers from sensitive particle velocity transducer in low wind conditions. The acoustic absorption coefficient of grass surface from the Pu-probe in-situ technique using a windscreen should be analyzed and is the future area of study.
CONCLUSIONS

The in-situ technique based on ISO 13472-1 using a single channel microphone has been utilized with an addition of many features ranging from signal processing to studying various effects like slew distortion previously unexplained. Another in-situ technique based on the relatively newly developed Pu-probe has also been used. The implementation of both these techniques have been made robust, quick, inexpensive and easy to setup with a laptop based data acquisition system and a custom programmed Matlab GUI.

The acoustic absorption coefficients of the Dow quash and Astroturf surfaces calculated from both these in-situ techniques agree fairly well, though slight discrepancies exist in the low frequency region. The acoustic absorption coefficient calculation of grass surface from the Pu-probe method was sensitive to wind flow over the Pu-probe. Also any change in the sound pressure level of the sound source changes the amplitude of the particle velocity signal of the Pu-probe. The signal processing of this technique based on normalization of the particle velocity signal to prevent this effect or the use of windscreen while taking outdoor measurements should be studied further.

ACKNOWLEDGMENTS

We acknowledge Mr. Loren De Vries and Mr. Ryan Schott from John Deere Co. for providing the artificial turf surfaces and partial funding for this project. Also many thanks to Dr. Hans de Bree of Microflown Technologies for providing us the P-u probe used in this study.

REFERENCES

7. Down-sized reverberation chamber & Mobile Impedance Tube references
12. W. T. Chu, "Impulse-response and reverberation-decay measurement made by using a periodic


CONTACT

For additional information, contact the authors at:

Niranjan Londhe
Phone: (906)-370-3694
Email: nrlondhe@mtu.edu

Dr. Mohan Rao
Phone: (906)-487-2892
Email: mrao@mtu.edu

Dr. Jason Blough
Phone: (906)-487-1020
Email: jrblough@mtu.edu

Michigan Technological University
Mechanical Engineering – Engineering Mechanics
1400 Townsend Dr.
Houghton, MI 49931