Czech Technical University in Prague
Faculty of Electrical Engineering
Department of Measurement

Diploma Thesis

Modern Methods of Layer Sound Transmission Analysis

Bc. Chanh Nguyen Huu

Supervisor: Doc. Ing. Jan Holub, Ph.D.

Study Programme: Biomedical Engineering and Informatics

Field of Study: Biomedical Engineering

January 5, 2015

Acknowledgements

I would like thank to Jan Holub, my supervisor, for his suggestions and constant support during writing of this diploma thesis. Further I would like thank to my family and friends for their supporting during my student life.

Declaration

I hereby declare that I have completed this thesis independently and that I have listed all the literature and publications used.
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In Prague on January 5, 2015 Chanh Nguyen Huu

Czech Technical University in Prague
Faculty of Electrical Engineering

Department of Cybernetics

DIPLOMA THESIS ASSIGNMENT

Student: Bc. Chanh N g u y e n H u u

Study programme: Biomedical Engineering and Informatics

Specialisation: Biomedical Engineering

Title of Diploma Thesis: Modern Methods of Layer Sound Transmission Analysis

Guidelines:

Analyze the methods used for acoustic absorption and transmission measurements. Design and implement in Matlab an acoustic layer(s) transmission estimator based on measurement of reflected acoustic signal. A-priori knowledge of certain layer(s) parameters are expected to be available. Compare the results of your estimation with conventionally measured parameters (delivered by the diploma work supervisor).

Bibliography/Sources:

[1] ASTM Standard C384-04, 2003, Standard Test Method for Impedance and Absorption of Acoustical Materials by the Impedance Tube Method, ASTM International, 2003, DOI : 10.1520/C0384-04R11.
[2] ASTM Standard E1050, 2012, Standard Test Method for Impedance and Absorption of Acoustical Materials Using a Tube, Two Microphones, and a Digital Frequency Analysis System, ASTM International, York, PA USA, 2012, DOI: 10.1520/E1050-12.
[3] ISO 10534, Determination of sound absorption coefficient and impedance in impedance tubes, International Organisation for Standardization, Case postale 56, CH-1211 Genève 20, (1998).
[4] ASTM Standard C423, 2009a, Standard Test Method for Sound Absorption and Sound Absorption Coefficients by the Reverberation Room Method, ASTM International, York, PA USA, 2009, DOI: 10.1520/C0423-09A.
[5] Martin Novotny, Milos Sedlacek: Measurement of RMS values of non-coherently sampled signals, Measurement, Volume 41, issue 3 (April, 2008), p. 236-250. ISSN: 0263-2241 DOI:10.1016/ j.measurement.2006.11.011.

Diploma Thesis Supervisor: doc. Ing. Jan Holub, Ph.D.

Valid until: the end of the summer semester of academic year 2014/2015

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Dean

Prague, January 10, 2014

České vysoké učení technické v Praze
Fakulta elektrotechnická

Katedra kybernetiky

ZADÁNÍ DIPLOMOVÉ PRÁCE

Student: Bc. Chanh N g u y e n H u u

Studijní program: Biomedicínské inženýrství a informatika (magisterský)

Obor: Biomedicínské inženýrství

Název tématu: Moderní metody analýzy akustické propustnosti vrstev

Pokyny pro vypracování:

Proveďte rozbor metod, používaných pro měření akustické absorpce a propustnosti vrstev. Na základě provedeného rozboru navrhněte a v Matlabu implementujte metodu pro odhad akustické propustnosti vrstvy nebo více vrstev na základě měření parametrů odražené vlny a znalosti akustických vlastností materiálu vrstvy (vrstev).
Porovnejte výsledky odhadu s parametry, naměřenými konvenčními metodami (srovnávací výsledky dodá vedoucí DP).

Seznam odborné literatury:

[1] ASTM Standard C384-04, 2003, Standard Test Method for Impedance and Absorption of Acoustical Materials by the Impedance Tube Method, ASTM International, 2003, DOI : 10.1520/C0384-04R11.
[2] ASTM Standard E1050, 2012, Standard Test Method for Impedance and Absorption of Acoustical Materials Using a Tube, Two Microphones, and a Digital Frequency Analysis System, ASTM International, York, PA USA, 2012, DOI: 10.1520/E1050-12.
[3] ISO 10534, Determination of sound absorption coefficient and impedance in impedance tubes, International Organisation for Standardization, Case postale 56, CH-1211 Genève 20, (1998).
[4] ASTM Standard C423, 2009a, Standard Test Method for Sound Absorption and Sound Absorption Coefficients by the Reverberation Room Method, ASTM International, York, PA USA, 2009, DOI: 10.1520/C0423-09A.
[5] Martin Novotny, Milos Sedlacek: Measurement of RMS values of non-coherently sampled signals, Measurement, Volume 41, issue 3 (April, 2008), p. 236-250. ISSN: 0263-2241 DOI:10.1016/ j.measurement.2006.11.011.

Vedoucí diplomové práce: doc. Ing. Jan Holub, Ph.D.

Platnost zadání: do konce letního semestru 2014/2015

L.S.

doc. Dr. Ing. Jan Kybic
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prof. Ing. Pavel Ripka, CSc.
děkan

V Praze dne 10. 1. 2014

Abstract

This diploma thesis deals with analysis of the method for acoustic absorption and transmission measurement, and further deals with design and implementation of these method in the MATLAB for estimate transmission based on the reflected signal. These coefficients are estimated from the reflected and direct sound waves with different frequency.

Keywords: Aborption, Reflection, Direct pulses, Reflected pulses, Pulse detection

Abstrakt

Tato diplomová práce se zabývá analýzou metod používaných pro měření akustické absorpce a propustnosti vrstev, a dále se zabývá návrhem a implementací těchto metod v programu MATLAB pro odhad přenosu na základě akustické odrazivosti signálu. Tyto koeficienty se odhadují z odražených a přímých zvukových vln s různou frekvencí.

Klíčová slova: Absorpce, Odraz , Přímé pulzy, Odražené pulzy, Detekce pulzů

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Chapter 1

Introduction

This thesis deals with analysis and processing of sound data from measurement as well as the creation of a GUI for clearer and more comfortable data analysis. The goal of this thesis was to create such a GUI in the MATLAB that would be able to automatically analyse and process data with minimal input data. The GUI results in the detection of absorption and reflection coefficients of used sample. Beside the automatic analysis, the GUI also contains manual detection for more accuracy and obtaining better results. The GUI description is in Appendix A.

The best possible measured data, in other words data without measurement errors, were crucial for the data analysis and processing. Unfortunately, some random and systematic errors still appear during measurement. These errors, especially random, were eliminated or reduced by minimal criterium of ten measurement which is also important to compute uncertainties of measurement. The errors were also appeared during data analysis, while converting from the analog signal to the digital one. Uncertainties are described in section 5.3.

This thesis includes determining the coefficients like absorption and reflection. The equations with intensity of direct sound waves and intensity of reflected sound waves were used. Where the reflection coefficient is defined as a ratio of reflected and direct intensity, the absorption coefficient is defined as a difference of one and the reflection coefficient. These absorptions are defined in section 3.2. The determination of the properties of measured sample if the sample has rather absorptive or reflective properties.

As tested samples were used three different boards with different absorption and reflection properties. There were used concrete, polystyrene and rubber boards for measurement. Further description is given in section 4.2. It is expected that concrete will have the biggest reflection and other two samples will have relatively the same absorption properties which will be bigger then at used concrete.

This study is divided into 7 chapters starting with Introduction. The second chapter is focused on acoustic theory and basic properties of sound like sound pressure, intensity and spreading. The third chapter describes the definition of the sound spread in the rooms and required coefficients mentioned above. The next fourth chapter deals with the actual measurement, used device and tools and their settings. The fifth chapter is dedicated to principle of data analysis and processing, this chapter also contains figure of basic GUI interface. The sixth chapter contains results. The last seventh section contains the conclusion and outlines future directions.

Chapter 2

Sound Theory

Sound can be viewed as a wave motion in air or other elastic media. Sound is also the center of speech communication among live objects and it is part of everyday sensory experience because it can be viewed as an excitation of the hearing mechanism that results in the perception of sound. Just as humans have eyes for the detection of light and color, so the ears are for the detection of sound.[8] [2]

Sound is characterized by a number of basic phenomena. For example, frequency is an objective property of sound which specifies the number of waveform repetitions per unit of time (usually one second). Frequency can be readily measured on an oscilloscope or a frequency counter. On the other hand, pitch is a subjective property of sound. Perceptually, the ear hears different pitches for soft and loud 100 Hz tones. As intensity increases, the pitch of a low frequency tone goes down, while the pitch of a high frequency tone goes up. Fletcher found that playing pure tones of 168 and 318 Hz at a modest level produces a very discordant sound. At a high intensity, however, the ear hears the pure tones in the 150 to 300 Hz octave relationship as a pleasant sound. We cannot equate frequency and pitch, but they are analogous.[2]

There are another similar dualities between intensity and loudness where the relationship between the two is not linear or the relationship between waveform (or spectrum) and perceived quality (or timbre) is complicated by the function of the hearing mechanism. A complex waveform can be described in terms of a fundamental and a series of harmonics of various amplitudes and phases. But perception of timbre is complicated by the frequency-pitch interaction as well as other factors. The interaction between the physical properties of sound, and our perception of them, poses delicate and complex issues. It is this complexity in audio and acoustics that creates such interesting problems. On one hand, the design of a loudspeaker or a concert hall should be a straightforward and objective engineering process. But in practice,that objective expertise must be carefully tempered with purely subjective experiences and wisdom.[2] [7]

As has often been pointed out, loudspeakers are not designed to play sine waves into calibrated microphones placed in anechoic chambers. Instead, they are designed to play music in listening rooms. In other words, the study of audio and acoustics involves both art and science.[2] The purpose of this chapter will be to describe some basic principles of the sound, especially the physics of sound, its basic properties and description.

2.1 The Sine Wave and Vibration

The simple vybration system can be created by the weight (mass) on the spring in Fig. 2.1 is a vibrating system. If the weight is pulled down to the 5 mark and released, the spring pulls the weight back toward 0.However, the weight will not stop at 0; its inertia will carry it beyond 0 almost to +5. The weight will continue to vibrate, or oscillate, at an amplitude that will slowly decrease due to frictional losses in the spring and the air. In the arrangement of a mass and spring, vibration or oscillation is possible because of the elasticity of the spring and the inertia of the weight.[2]

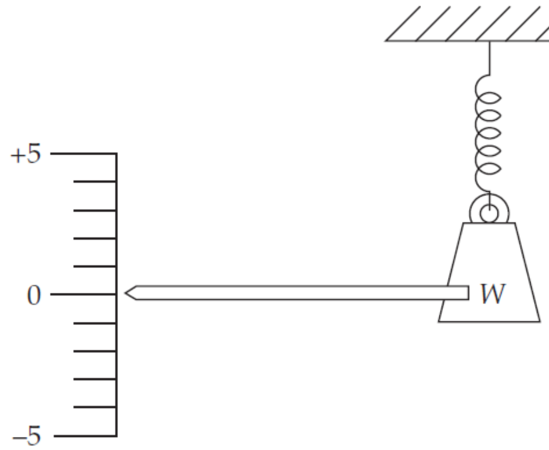

Figure 2.1: A weight on a spring vibrates at its natural frequency because of the elasticity of the spring and the inertia of the weight.[2]

Elasticity and inertia are two things all media must possess to be capable of conducting sound. The weight in Fig. 2.1 moves in what is called simple harmonic motion. If a pen is fastened to the pointer of Fig. 2.2, and a strip of paper is moved past it at a uniform speed, the resulting trace is a sine wave. The sine wave is a pure waveform closely related to simple harmonic motion.[2]

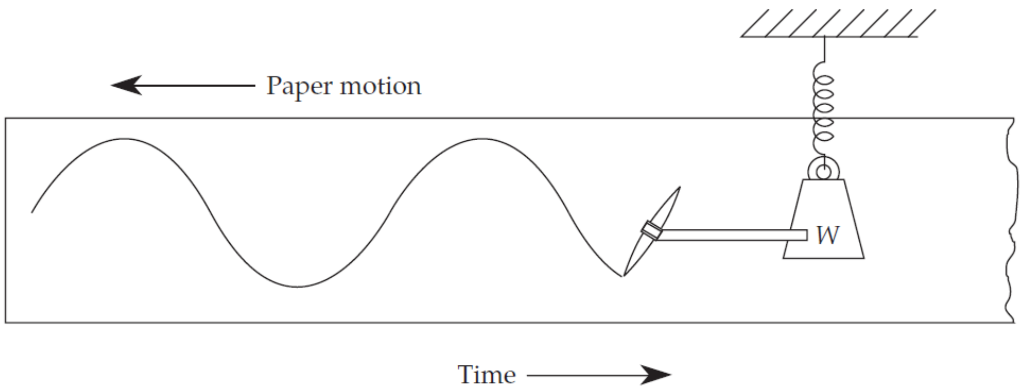

Figure 2.2: A pen fastened to the vibrating weight traces a sine wave on a paper strip moving at uniform speed. This shows the basic relationship between simple harmonic motion and the sine wave.[2]

2.2 Sound in Media

Sound is readily conducted in gases, liquids, and solids such as air, water, steel, concrete, and so on, which are all elastic media. Each substance has other features. Especially speed of sound transmission is the most specific feature which has biggest difference among substances in which it is spread. Imagine a person stationed a distance away, who strikes a railroad rail with a rock. Second person will hears two sounds, one sound coming through the rail and one through the air. The sound through the rail arrives first because the speed of sound in the dense steel is faster than in tenuous air. Similarly, sounds can be detected after traveling thousands of miles through the ocean.[2]

In the other hand without a medium, sound cannot be propagated. For example in Fig. 2.3: In the laboratory, an electric buzzer is suspended in a heavy glass bell jar. As the button is pushed, the sound of the buzzer is readily heard through the glass. As the air is pumped out of the bell jar, the sound becomes fainter and fainter until it is no longer audible. The sound-conducting medium, air, has been removed between the source and the ear. Because air is such a common agent for the conduction of sound, it is easy to forget that other gases as wel as solids and liquids are also conductors of sound. Outer space is an almost perfect vacuum; no sound can be conducted except in the tiny island of atmosphere within a spaceship or a spacesuit.[2]

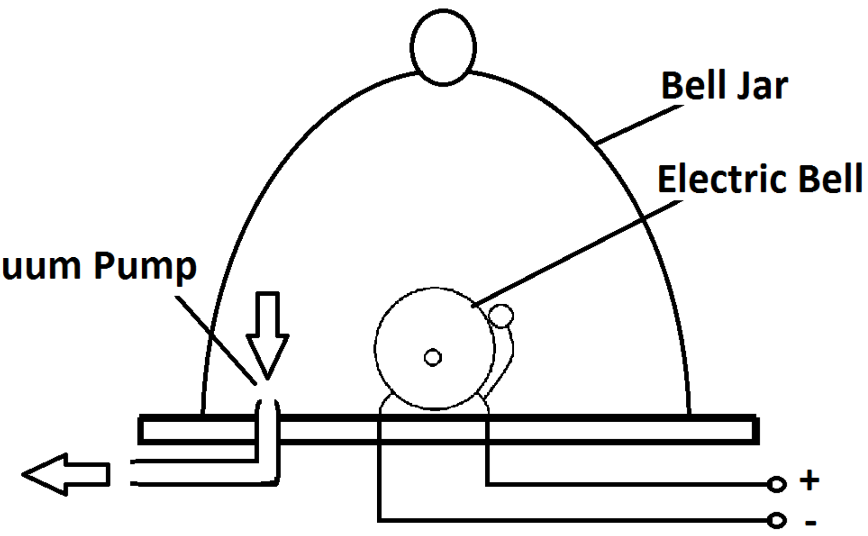

Figure 2.3: Sound transmission during pumping vacuum in a bell jar.

2.2.1 Particle Motion

Particles of air propagating a sound wave do not move far from their undisplaced positions, as shown in Fig. 2.4. The disturbance travels on, but the propagating particles move only in localized regions (perhaps a maximum displacement of a few ten-thousandths of an inch). Note also that the velocity of a particle is maximum at its equilibrium position, and zero at the points of maximum displacement (a pendulum has the same property). The maximum velocity is called the velocity amplitude, and the maximum displacement is called the displacement amplitude. The maximum particle velocity is very small, less than 0.5

in/sec for even a loud sound. As we will see, to lower the level of a sound, we must reduce the particle velocity.[2]

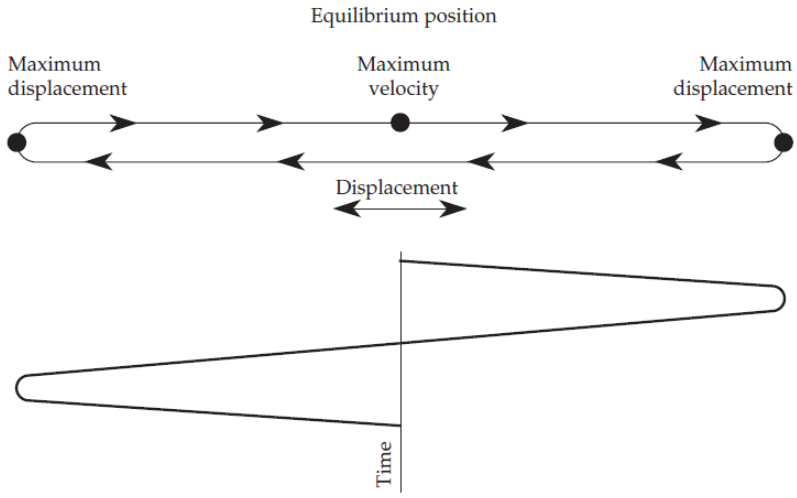

Figure 2.4: An air particle is made to vibrate about its equilibrium position by the energy of a posing sound wave because of the interaction of the elastic forces of the air and the inertia of the air particle. [2]

There are three distinct forms of particle motion. If a stone is dropped on a calm water surface, concentric waves travel out from the point of impact, and the water particles trace circular orbits (for deep water, at least), as shown in Fig. 2.5 A. Another type of wave motion is illustrated by a violin string, as shown in Fig. 2.5 B. The tiny elements of the string move transversely, or at right angles to the direction of travel of the waves along the string. For sound traveling in a gaseous medium such as air, the particles move in the direction the sound is traveling. These are called longitudinal waves, as shown in Fig. 2.5 C.[2]

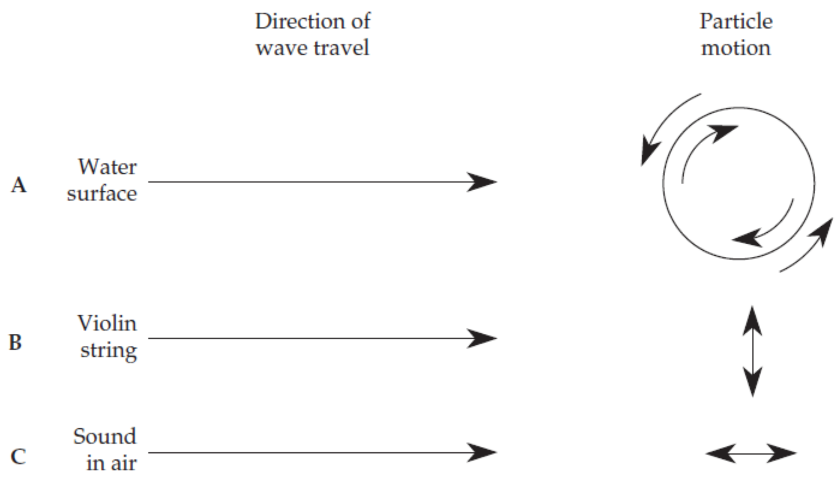

Figure 2.5: Particles involved in the propagation of sound waves can move with (A) circular motion, (B) transverse motion, or (C) longitudinal motion.[2]

2.2.2 Sound Propagation

Sound is propagated by particles which are moving slightly back an forth and this moving is able to carry music form information of a loudspeaker to human ears with a speed of sound. The dots of Fig. 2.6 represent air molecules with different density variations. In reality, there are more than a million molecules in a cubic inch of air. The molecules crowded together represent areas of compression (crests) in which the air pressure is slightly greater than the prevailing atmospheric pressure. The sparse areas represent rarefactions (troughs) in which the pressure is lightly less than atmospheric pressure. The arrows (see Fig. 2.6) indicate that, on average, the molecules are moving to the right of the compression crests and to the left in the rarefaction troughs between the crests. Any given molecule, because of elasticity, after an initial displacement, will return toward its original position. It will move a certain distance to the right and then the same distance to the left of its undisplaced position as the sound wave progresses uniformly to the right. Sound exists because of the transfer of momentum from one particle to another.[2]

Why does the sound wave move to the right in this example? The answer is revealed by a closer look at the arrows (see Fig. 2.6). The molecules tend to bunch up where two arrows are pointing toward each other, and this occurs a bit to the right of each compression region. When the arrows point away from each other, the density of molecules decreases. Thus, the movement of the higher pressure crest and the lower pressure trough accounts for the progression of the sound wave to the right.[2]

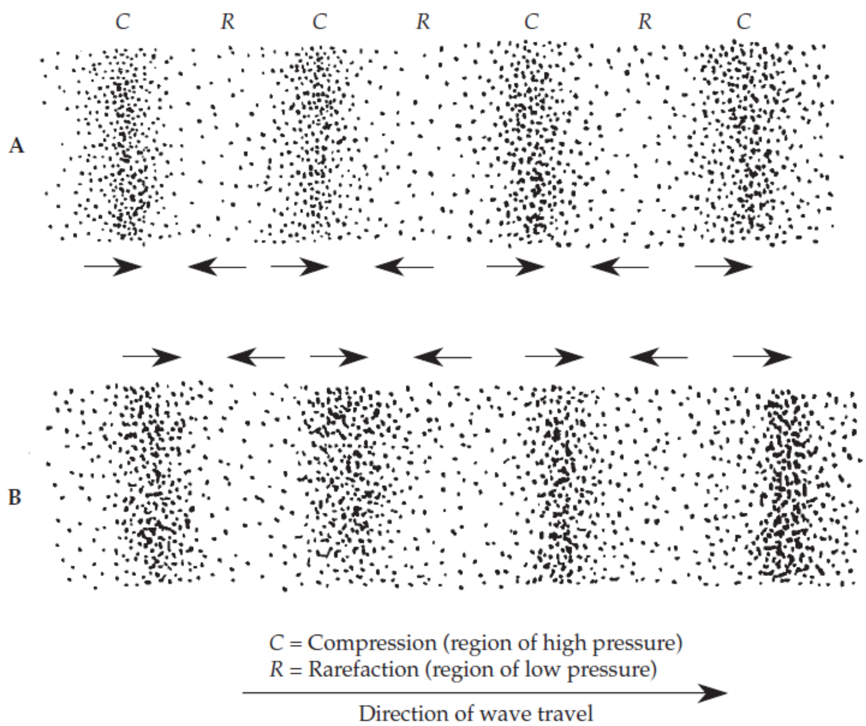

Figure 2.6: Sound waves traveling through a medium change localized air particle density. (A) The sound wave causes the air particles to be pressed together (compression) in some regions and spread out (rarefaction) in others. (B) An instant later the sound wave has moved slightly to the right.[2]

As mentioned previously, the pressure at the crests is higher than the prevailing atmospheric barometric pressure and the troughs lower than the atmospheric pressure, as shown in the sine wave of Fig. 2.7. These fluctuations of pressure are very small indeed.The faintest sound the ear can hear (20 Pa) exists at a pressure some 5,000 million times smaller than atmospheric pressure. Normal speech and music signals are represented by correspondingly small ripples superimposed on the atmospheric pressure.[2]

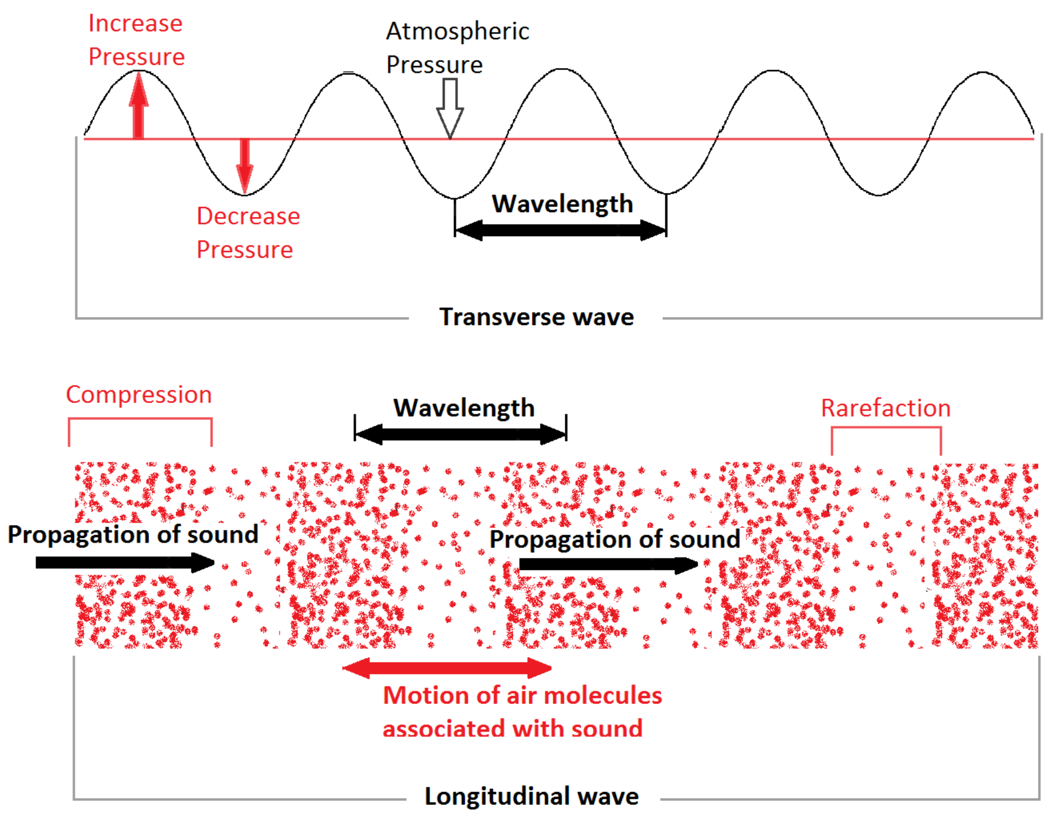

Figure 2.7: Transverse and Longitudinal waves and their description and pressure variations of sound waves are superimposed on prevailing barometric pressure. (Upper) The compressed regions are very slightly above and the rarefied regions very slightly below atmospheric pressure. (Lowe) An instantaneous view of the compressed and rarefied regions of a sound wave in air.

2.2.3 Speed of Sound

The compression wave, along with the rarefaction wave that immediately follows it, will be propagated outward at the speed of sound. The speed of sound varies depending on the two factors, the average elasticity and density of the medium in which the sound is propagated. As mentioned above the speed of any wave depends upon the properties of the medium through which the wave is traveling. There are two deficiencies in Fig. 2.6. First, the scale of air contains approximately hundreds billions of molecules. Second, although Fig. 2.6 makes it appear that the air pressure disturbance is propagated in a simple straight line from the vibrating object, it actually travels in all directions from the source.[13] [8]

In general, solids have the strongest interactions between particles, followed by liquids and then gases. For this reason, longitudinal sound waves travel faster in solids than they do in liquids than they do in gases. Even though the inertial factor may favor gases, the elastic factor has a greater influence on the speed of a wave, thus yielding this general pattern Eq. 2.1.[7]

$$c_{solid} > c_{liquids} > c_{gases} \tag{2.1}$$

Where:

c_{solid} ... is sound speed in solid
$c_{liquids}$ ... is sound speed in liquids
c_{gases} ... is sound speed in gases

Sound also travels faster in air as temperature increases. Finally, humidity affects the velocity of sound in air; the more humid the air, the faster the speed. It should be noted that the speed (velocity) of sound is different from the particle velocity. The velocity of sound determines how fast sound energy moves across a medium. Particle velocity is determined by the loudness of the sound.[2]

2.2.4 Sound Speed in Air

The speed of sound in air c is determined by the air itself and is not dependent upon the amplitude, frequency, or wavelength of the sound. For an ideal gas the speed of sound depends only on its temperature and is independent of gas pressure. This dependence also applies to air, in good approximation and can be regarded as an ideal gas. It is a wrong assumption that the speed of sound decreases with altitude above sea level, because the density of air decreases with height. The changing of atmospheric pressure does not change the speed of sound. Only the colder temperature lets decrease the speed of sound at higher altitudes. Speed of sound is dependent nearly only on its temperature.[3]

The expression for the speed of sound c_0 in air is:

$$c_0 = \sqrt{\frac{p_0}{\rho_0}\kappa} \tag{2.2}$$

Where:
c_0 ... is speed of sound in air at 0°C = 331.3 m/s
p_0 ... is atmospheric air pressure 101.325 kPa (standard)
ρ_0 ... is density of air at 0°C: 1.293 kg/m^3 = Z_0/c_0
κ ... is adiabatic index of air at 0°C: 1.402 = c_p/c_v = ratio of the specific warmth

The equation 2.3 described below define with sufficient precision the speed of sound in air in m/s vs. temperature ϑ (theta) in degrees Celsius (centigrade):

$$c_\vartheta = 331.3 + 0.606\vartheta \tag{2.3}$$

With the following formula 2.4 the sound speed is calculated more exactly than in formula 2.3.

$$c_\vartheta = c_0\sqrt{1+\alpha\vartheta} \tag{2.4}$$

Where:

c_ϑ ... is speed of sound depends on temperature in m/s

c_0 ... is speed of sound in air at 0°C

ϑ ... temperature in °C

α ... coefficient of expansion 1/273.15 in °C

Note::

The speed of sound c in air is only dependent on the temperature ϑ. It is completely independent of the air pressure p.[4]

Reason:

Thair pressure and the air density are proportional to each other at the same temperature. This mean in equation 2.2 the ratio p_0/ρ_0 is always constant. The speed of sound in air depends on the density of air and the density of air depends on the temperature. Therefore the speed of sound is the same on a mountain peak as it is at sea level, provided that the temperature is the same.[4]

The sound speed, density of air, specific acoustic impedance vs. temperature are shown in table 2.2. And on the table 2.1 is shown approximate speed of sound in common materials.

Medium	**Sound speed** $[m/s]$	**Sound speed** $[ft/s]$
Air, dry at 20°C	343	1 125
Hydrogen at 0°C	1 280	4 200
Water at 15°C	1 500	4 920
Lead	2 160	7 090
Concrete	3 100	10 200
Wood (soft - along the fibre)	3 800	12 500
Glass	5 500	18 500
Steel	5 800	19 000

Table 2.1: Approximate speed of sound in common materials.[4]

Temperature of air ϑ **[°C]**	**Speed of sound** $c\ [m/s]$	**Time per 1 m** $\delta t\ ms/m$	**Density of air** $\rho\ kg/m^3$	**Impendance of air** $Z\ Ns/m^3$
+40	354.94	2.817	1.1272	400.0
+35	351.96	2.840	1.1455	403.2
+30	349.08	2.864	1.1644	406.5
+25	346.18	2.888	1.1839	409.4
+20	343.26	2.912	1.2041	413.3
+15	340.31	2.937	1.2250	416.9
+10	337.33	2.963	1.2466	420.5
+5	334.33	2.990	1.2690	424.3
0	331.30	3.017	1.2920	428.0
-5	328.24	3.044	1.3163	432.1
-10	325.16	3.073	1.3413	436.1
-15	322.04	3.103	1.3673	440.3
-20	318.89	3.134	1.3943	444.6
-25	315.72	3.165	1.4224	449.1

Table 2.2: The sound speed, density of air and specific acoustic impedance depend on temperature.[4]

2.3 Wavelength and Frequency

A sine wave is shown in Fig. 2.8. The wavelength λ is the distance a wave travels in the time it takes to complete one cycle. A wavelength can be measured between successive peaks or between any two corresponding points on the cycle. This also holds for periodic waves other than the sine wave.[2]

2.3.1 Basic Terminology

In the Table 2.3 are shown main sound units which are important in acoustics.

	Name	**Abbreviation**	**Basic Units**
Wavelength $[\lambda]$	metre	m	m
Frequency $[f]$	Hertz	Hz	s^{-1}
Period $[T]$	second	s	s
Speed of sound $[c]$	metre/sec	m/s	$m.s^{-1}$

Table 2.3: The main units measured in acoustics [1].

The **wavelength** (λ) of a wave is measured in metres.$[m]$ [1]

The **frequency** f is defined as the number of cycles completed in one second. The unit of measurement for frequency is hertz $[Hz]$. Conceptually, frequency is simply the rate of vibration. The most crucial function of the auditory system is to serve as a frequency analyzer - a system that determines how much energy is present at different signal frequencies.[8]

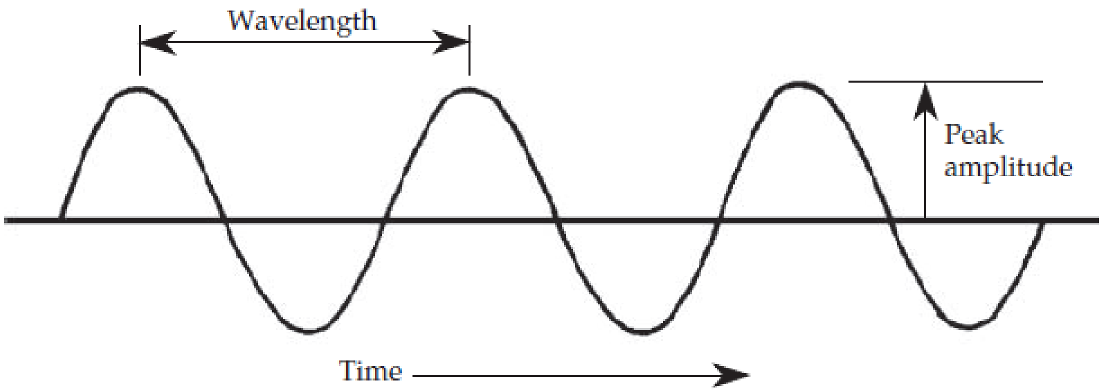

Figure 2.8: Wavelength is the distance a wave travels in the time it takes to complete one cycle. It can also be expressed as the distance from one point on a periodic wave to the corresponding point on the next cycle of the wave.[2]

The **period** T is the time required to complete one cycle of vibration. For example if 20 cycles are completed in 1 second, the period is 1/20th of a second $[s]$ or 0.05 $[s]$. For speech applications the most commonly used unit of measurement for period is the millisecond $[ms]$.[8]

The **speed** or **velocity** of sound c is the number of metres that a wave front can travel in a second. The speed of sound is measured in metres per second $[m.s^{-1}]$.[1]

The period and frequency of a wave are inverse of each other see Eq. 2.5 and Eq. 2.6. Before making calculations be sure that the values are in Hz and seconds. Milliseconds $[ms]$ can be converted to seconds by dividing by 1000. Kilohertz $[kHz]$ can be converted to $[Hz]$ by multiplying by 1000.[1]

$$f = \frac{1}{T} \tag{2.5}$$

$$T = \frac{1}{f} \tag{2.6}$$

It is possible to calculate the frequency of a wave if the wavelength and the speed of sound are known (see formulas in Eq. 2.7). Conversely, it can be calculated the wavelength of a wave from its frequency and the speed of sound (see Eq. 2.8). (Similarly, for period T). It should be clear from this that if the speed of sound changes then there will be a change in the apparent resonant characteristics of a resonator. For example, if the speed of sound increases the frequency of a sound emanating from a resonator will increase.[1]

$$f = \frac{c}{\lambda}, \lambda = \frac{c}{f}, c = \lambda f \tag{2.7}$$

$$\lambda = Tc, T = \frac{\lambda}{c}, c = \frac{\lambda}{T} \tag{2.8}$$

2.3.2 Longitudinal and Transverse waves

There are two basic types of wave motion for mechanical waves:

- **Longitudinal**
- **Transverse**

2.3.2.1 Longitudinal waves

Longitudinal in figure 2.7 waves, also known as "l-waves", are waves in which the displacement of the medium is in the same direction as the direction of travel of the wave or in the opposite direction to the direction of travel of the wave. Mechanical longitudinal waves are also called compression waves, because they produce compression and rarefaction when traveling through a medium. The distance between two compression waves are called wavelength.[14] [9] [12]

2.3.2.2 Transverse waves

Transverse figure 2.7 waves are waves that are oscillating perpendicularly to the direction of propagation. If a transverse wave is moving in the positive x direction, so the oscillation of this transverse wave are moving up and down in directions that lie in the y or z plane. The distance between two amplitudes of the wave is called wavelength. Transverse waves are typically for solid elastic substances like bars, strings stretched between two fixed points. [14] [9] [12]

Transverse waves, also known as shear waves, have the additional property, polarization and are not a characteristic of sound waves.

2.4 Harmonics

A simple sine wave f_1 with amplitude and frequency is shown in Fig. 2.9-A. Another Fig. 2.9-B shows sine wave f_2 with half amplitude and twice the frequency. If combine these two frequencies at each point in time, the waveshape on Fig. 2.9-C is obtained. Next Fig. 2.9-D shows another simple sine wave f_3 with half amplitude and three times its frequency compare with sine wave f_1. The origin sine wave f_1 shown in Fig. 2.9-A has been progressively changed as other sine waves have been added to it. The result is the complex wave form in Fig. 2.9-E. Whether these are acoustic waves or electronic signals, the process can be reversed. The complex waveform can be disassembled, as it were, to the simple f_1, f_2 and f_3 sine components by either acoustical or electronic filters. For example, passing the waveform of Fig. 2.9-E through a filter permitting only f_1 and rejecting f_2 and f_3, the original f_1 sine wave of Fig. 2.9-A emerges in pristine condition.[2]

The sine wave with the lowest frequency f_1 of Fig. 2.9-A is called the fundamental, the sine wave that is twice the frequency f_2 of Fig. 2.9-B is called the second harmonic and the sine wave that is three times the frequency f_3 of Fig. 2.9-D is the third harmonic. The fourth harmonic and the fifth harmonic are four and five times the frequency of the fundamental and so on.[2]

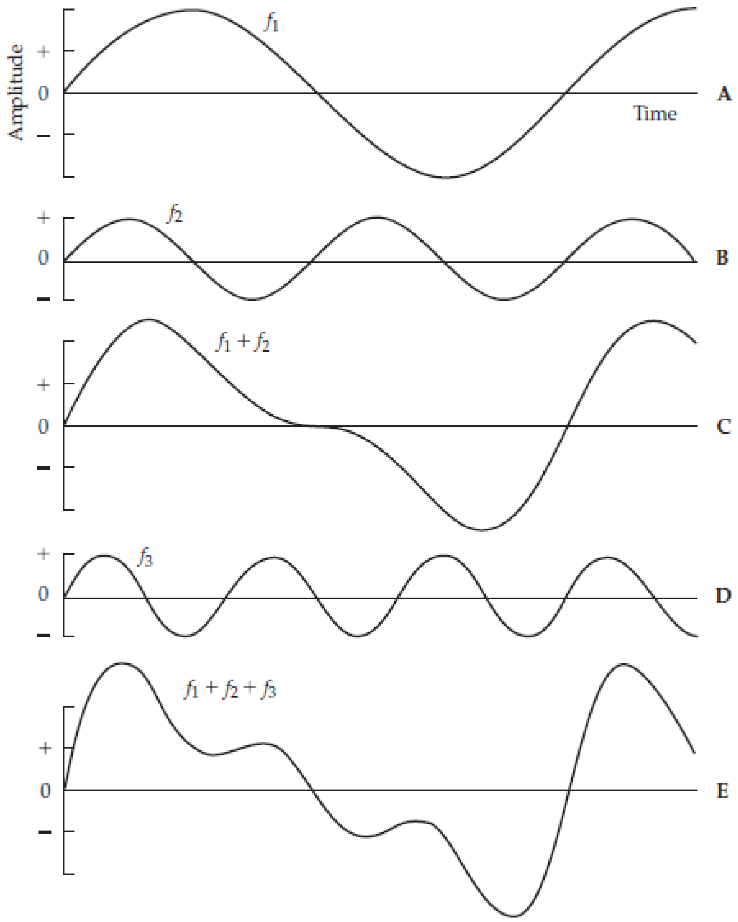

Figure 2.9: A study in the combination of sine waves. (A) The fundamental of frequency f_1. (B) A second harmonic of frequency $f_2 = 2f_1$ and half the amplitude of f_1. (C) The sum of f1 and f_2 obtained by adding ordinates point by point. (D) A third harmonic of frequency $f_3 = 3f_1$ and half the amplitude of f_1. (E) The waveform resulting from the addition of f_1, f_2, and f_3. All three components are in phase, that is, they all start from zero at the same instant.[2]

2.5 Phase

As can see in Fig. 2.9 all three components f_1, f_2 and f_3 start from zero point together which mean that it is called an in-phase conditions. In some cases, the time relationships between harmonics or between harmonics and the fundamental are quite different from this. One complete sine-wave cycle represents 360° of rotation. If another sine wave of identical frequency is delayed 90°, its time relationship to the first one is a quarter wave late (time increasing to the right). A half-wave delay would be 180°, and so on. For the 360° delay, the waveform at the bottom of Fig. 2.10 synchronizes with the top one, reaching positive peaks and negative peaks simultaneously and producing the in-phase condition.[2]

All three components of the complex waveform of Fig. 2.9-E are in phase. That is, the f_1 fundamental, the f_2 second harmonic, and the f_3 third harmonic all start at zero at the same time. What happens if the harmonics are out of phase with the fundamental? A Fig. 2.11-A illustrates this case. The second harmonic f_2 is now advanced 90° and the third harmonic f_3 is retarded 90°. By combining f_1, f_2 and f_3 for each instant of time, with due regard to positive and negative signs, the contorted waveform of Fig. 2.11-E is obtained.[2]

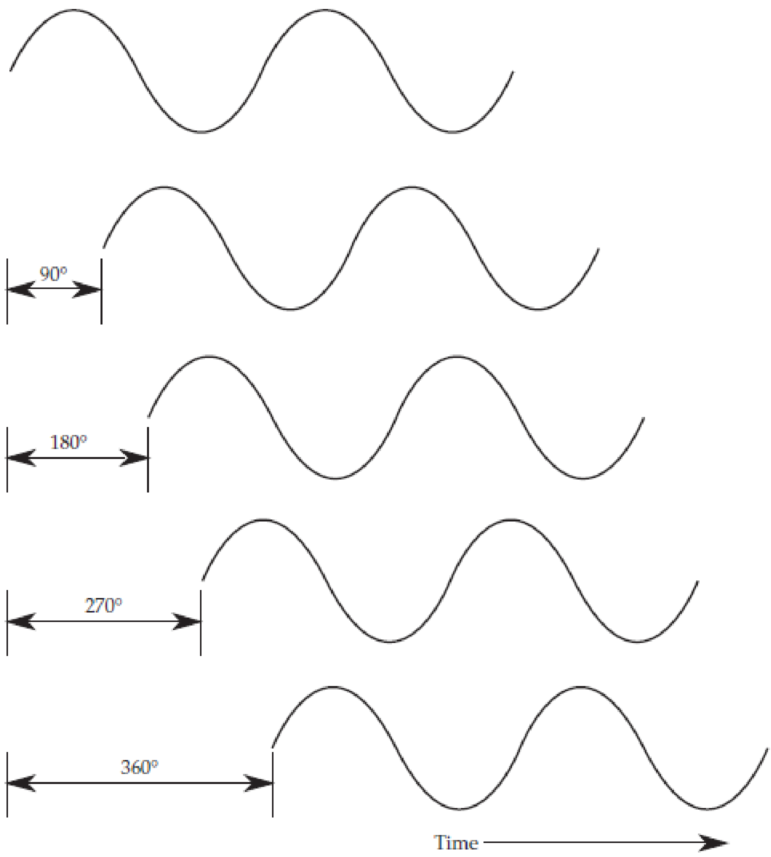

Figure 2.10: Illustration of the phase relationships between waveforms with the same amplitude and frequency. A rotation of 360° is analogous to one complete sine wave cycle.[2]

The only difference between Figs. 2.9-E and 2.11-E is that a phase shift has been introduced between harmonics f_2 and f_3 and the fundamental f_1. That is all that is needed to produce drastic changes in the resulting waveshape. Curiously, even though the shape of the waveform is dramatically changed by shifting the time relationships of the components, the ear is relatively insensitive to such changes. In other words, waveforms E of Figs. 2.9-E and 2.11-Ed very much alike. A common error is confusing polarity with phase. Phase is the time relationship between two signals, while polarity is the +/− or the −/+ relationship of a given pair of signal leads.[2]

Musicians may be inclined to use the term partial instead of harmonic, but it is best that a distinction be made between the two terms because the partials of many musical instruments are not harmonically related to the fundamental. That is, partials might not be exact multiples of the fundamental frequency, yet richness of tone can still be imparted by such deviations from the true harmonic relationship. For example, the partials of bells, chimes and piano tones are often in a nonharmonic relationship to the fundamental.[2]

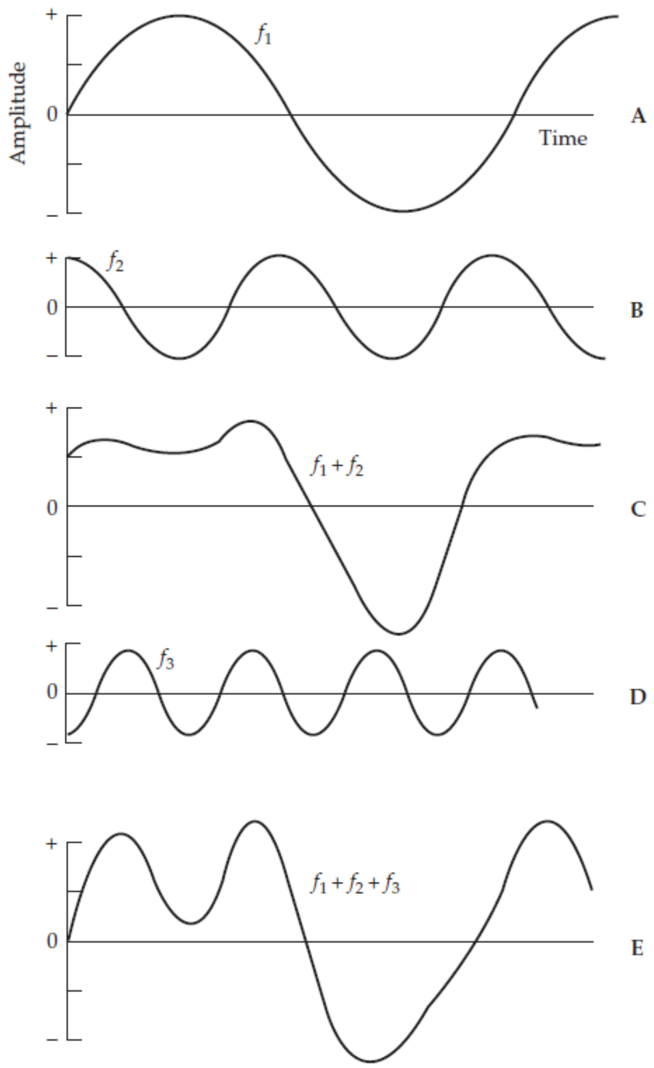

Figure 2.11: A study of the combination of sine waves that are not in phase. (A) The fundamental frequency f_1. (B) The second harmonicf_2 with twice the frequency and half the amplitude of f_1 advanced 90° with respect to f_1. (C) The combination of f_1 and f_2 obtained by adding ordinates point by point. (D) The third harmonic f_3 with phase 90° behind f_1, and with half the amplitude of f_1. (E) The sum of f_1, f_2, and f_3. Compare this resulting waveform with that of Fig. 2.9-E. The difference in waveforms is due entirely to the shifting of the phase of the harmonics with respect to the fundamental.[2]

2.6 Complex Waves

Speech and music waveshapes depart radically from the simple sine wave, and are considered as complex waveforms. However, no matter how complex the waveform, as long as it is periodic, it can be reduced to sine components. The obverse of this states that any complex periodic waveform can be synthesized from sine waves of different frequencies, different amplitudes, and different time relationships (phase). Joseph Fourier was the first to prove these relationships. The idea is simple in concept, but often complicated in its application to specific speech or musical sounds. Let to see how a complex waveform can be reduced to simple sinusoidal components. See Fig. 2.12.[2]

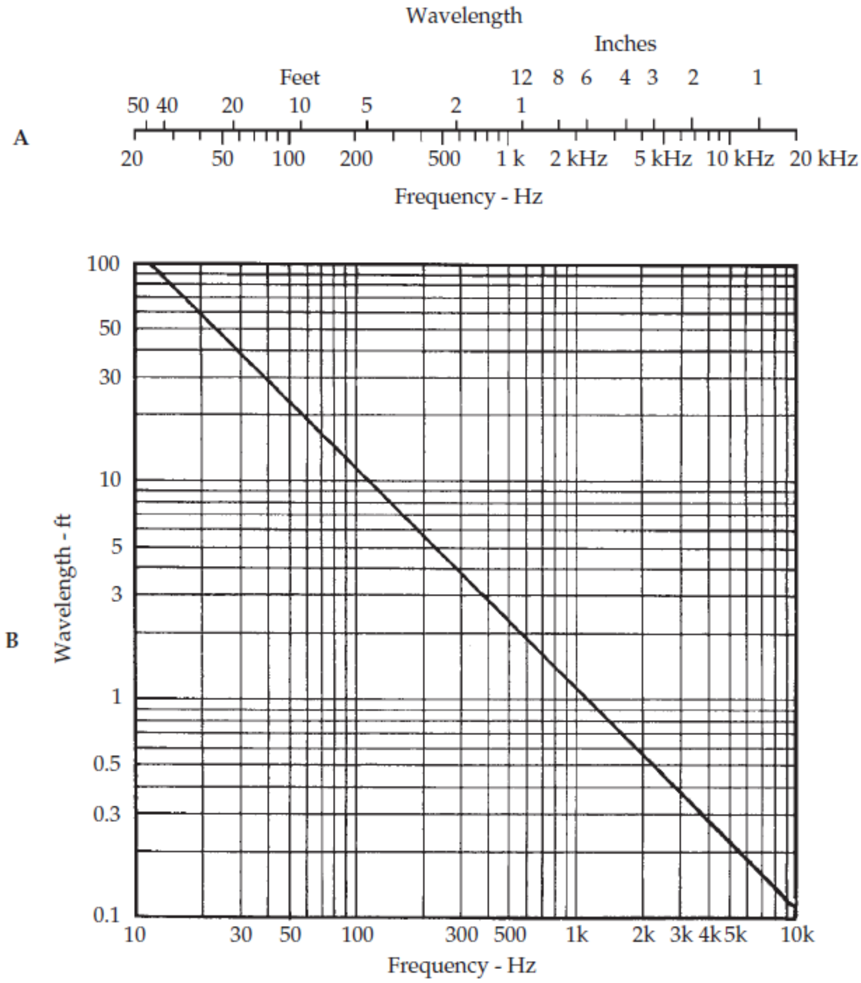

Figure 2.12: Wavelength and frequency are inversely related. (A) Scales for approximately determining wavelength of sound in air from a known frequency, or vice versa. (B) A chart for determining the wavelength in air of sound waves of different frequencies. (Both based on speed of sound of 1,130 ft/sec).[2]

As written above, sinusoids are sometimes referred to as simple periodic signals. The term "periodic" means that there are patterns that repeat itself and the term "simple" means that there is only one frequency component present. This is confirmed in the frequency domain representations in Fig. 2.13, which all show a single frequency component in both the amplitude and phase spectra. Complex periodic signals involve the repetition of a nonsinusoidal patterns and in all cases, complex periodic signals consist of more than a single frequency component. All nonsinusoidal periodic signals are considered complex periodic.[8] [6]

All frequencies and amplitudes, merged together in one continuous signal, could be shown by using of the frequency domain. All complex periodic signals shown in Fig. 2.14 have a harmonic spectrum and Fig. 2.14 also shows energy at the fundamental frequency and at whole number multiples of the fundamental frequency.[8], [13]

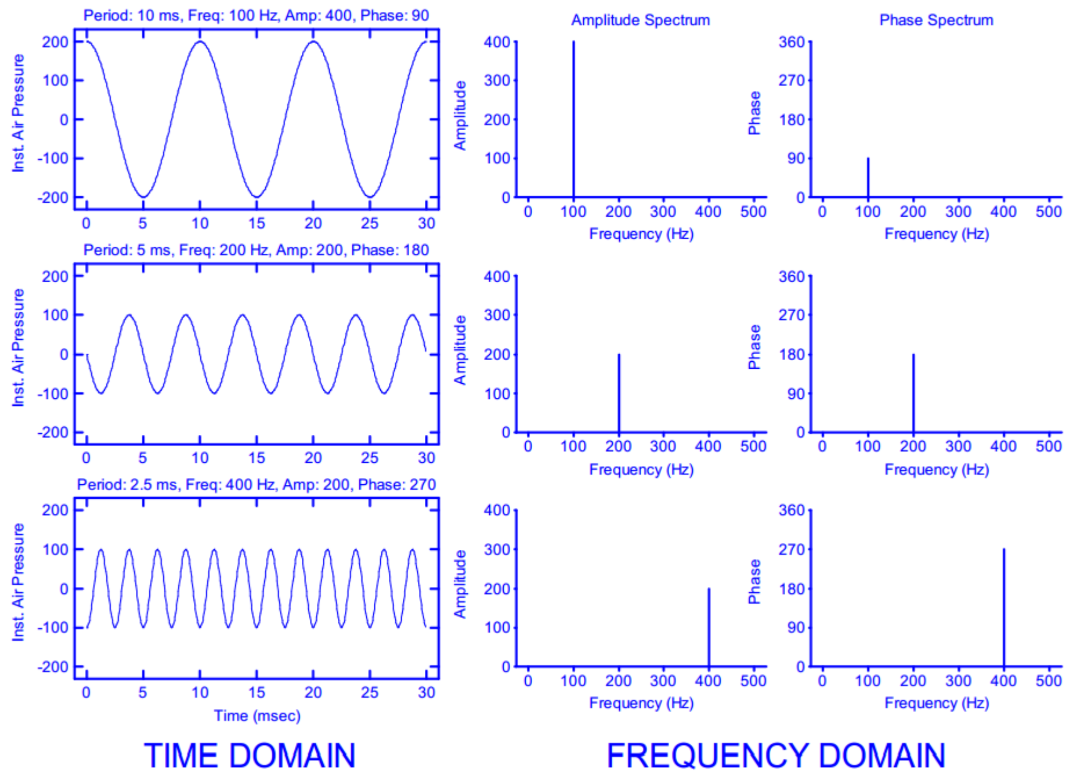

Figure 2.13: Time and frequency domain representations of three sinusoids. The frequency domain consists of two graphs: an amplitude spectrum and a phase spectrum. An amplitude spectrum is a graph showing what frequencies are present with what amplitudes, and a phase spectrum is a graph showing the phases of each frequency component.[8]

2.7 Sound Spectrum

Sound spectrum can be compared like a light spectrum where a range of visible light spectrum exists and can be visible for human eyes. It is the same case as at human ears where human ears can hear audible sound in a range from 20Hz to 20kHz generally defined. The frequency spectrum of a time-domain signal is a representation of that signal in the frequency domain see Fig. 2.13. The frequency spectrum can be generated through a Fourier transform of the signal and output values are usually presented as amplitude or phase. Both are plotted versus frequency see Fig. 2.13.

Fig. 2.15 and 2.16 show two types of waveforms that are typify for infinite number of different waveforms which are commonly encountered in audio. These waveforms have been captured from the screen in Matlab programme with using special visual function ("iPower: Interactive Power Spectrum Demonstrator, Version 2" by Tom O'Haver). There are two different types of signals where in the Fig. 2.15 is sine signal and in the Fig. 2.16 is a triangle signal. Both signals are shown in the time domain. Another two plots represents the spectrum of these particular signals.

For a sine wave, all the energy is concentrated at one frequency. The sine wave produced by this particular signal Matlab generator is not really a pure sine wave. But in real no oscillator is perfect and all have some harmonic content, but in scanning the spectrum of

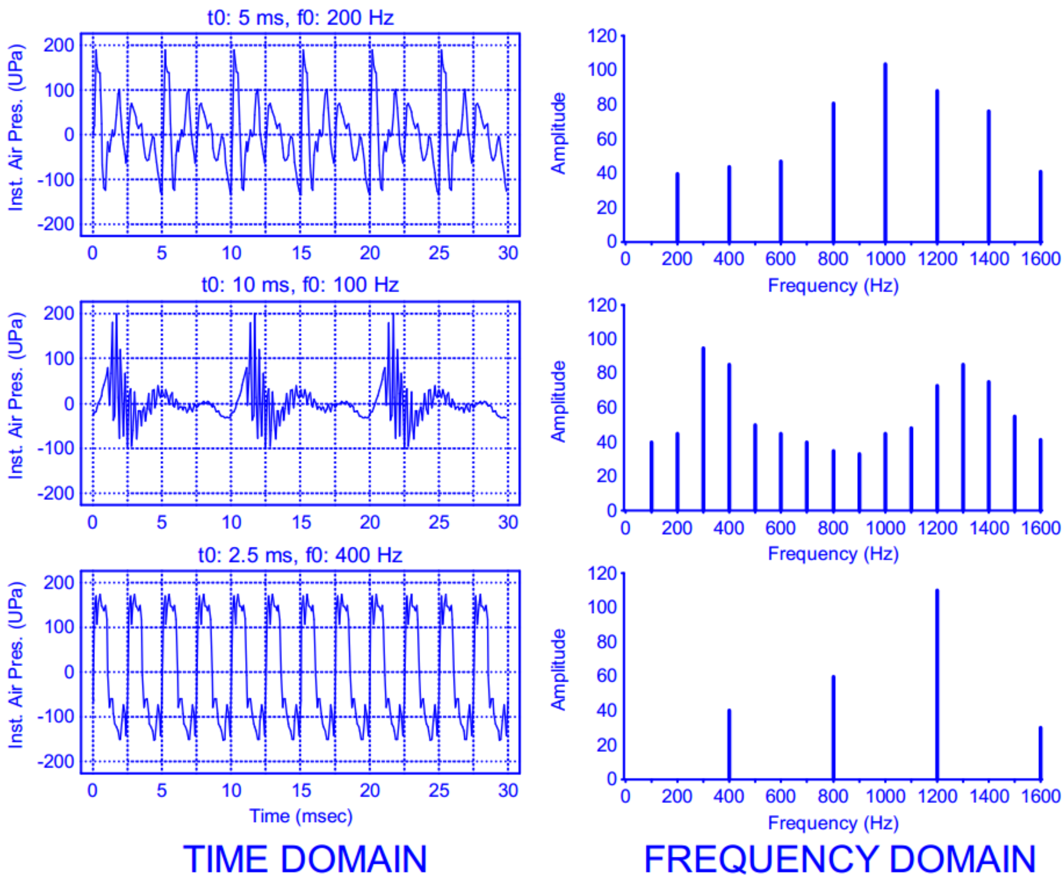

Figure 2.14: Time and frequency domain representations of three complex periodic signals. Complex periodic signals have harmonic spectra, with energy at the fundamental frequency (f_0) and at whole number multiples of f_0 ($2f_0$, $3f_0$, $4f_0$, etc.) For example, the signal in the upper left, with a fundamental frequency of 200 Hz, shows energy at 200 Hz, 400 Hz, 600 Hz, etc. In the spectra on the right, amplitude is measured in arbitrary units. The main point being made in this figure is the distribution of harmonic frequencies at whole number multiples of f_0 for complex periodic signals.[8]

this created sine wave, the harmonics generated were too low to show on the graph scale of Fig. 2.15. Another type of this waveform is triangle which is generated by Matlab where the signal has a major fundamental component of magnitude as shown in Fig. 2.16. As can be seen in this spectrum, there are shown another lower magnitude of fundamental harmonic frequency.[2]

The spectra of sine, triangular, and square waves reveals energy concentrated at harmonic frequencies but nothing between. These are all periodic waveforms which repeat themselves cycle after cycle.[2]

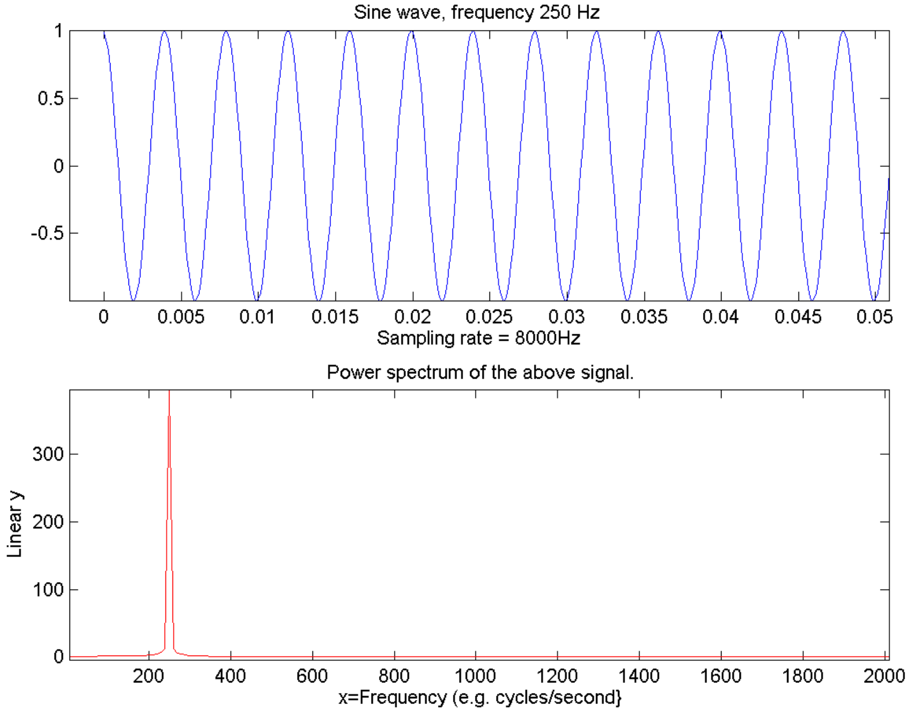

Figure 2.15: Sine waveform signal in the time-domain and spectrum of sine wave signal in frequency-domain.

2.8 Energy Density and Intensity

2.8.1 Sound Energy

Imagine that a sound source generates a sound wave, for instance a musical instrument, where this wave has to deliver some energy to a fluid. This energy is carried away by the sound wave. Accordingly it can characterise the amount of energy contained in one unit volume of the wave by the energy density. As with any kind of mechanical energy one has to distinguish between potential and kinetic energy density: [10]

$$w_{kin} = \frac{\rho_0 v^2}{2} \tag{2.9}$$

$$w_{pot} = \frac{p^2}{2\rho_0 c^2} \tag{2.10}$$

Where:
ρ_0 ... is initial density in kg/m^3
c ... is speed of sound in air [m/s]

The total energy of density is defined as a sum of kinetic and potential energy:

$$w = w_{kin} + w_{pot} \tag{2.11}$$

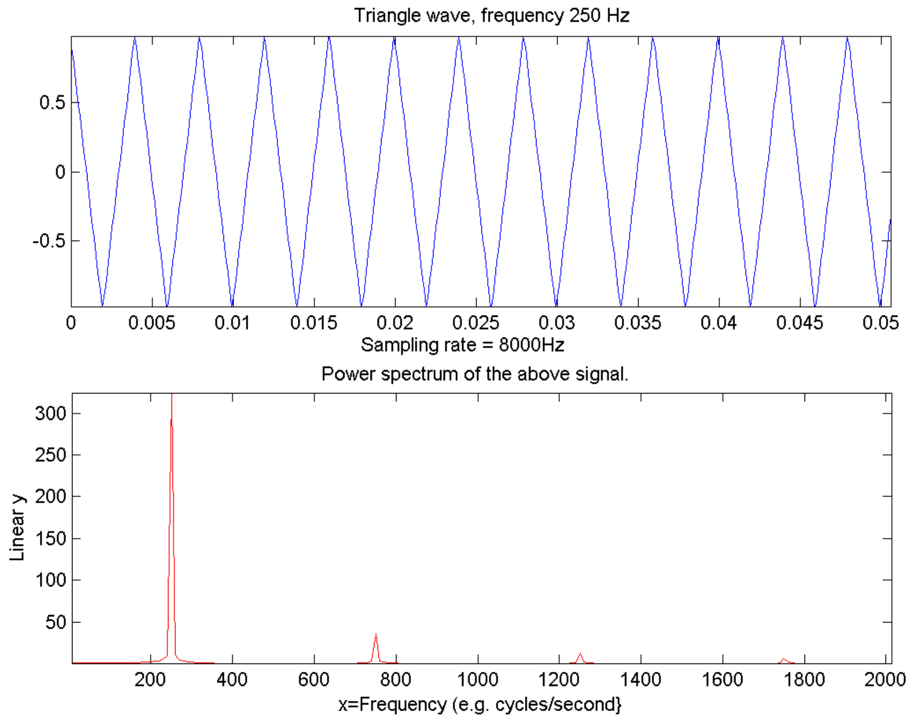

Figure 2.16: Triangle waveform signal in the time-domain and spectrum of triangle wave signal in frequency-domain.

2.8.2 Sound Intensity

Another important quantity is sound intensity (I in $watts/m^2$), which is more important then previous sound energy. The sound intensity is a measure of the energy transported in a sound wave. Imagine a window of $1m^2$ perpendicular to the direction of sound propagation. Then the intensity is the energy per second passing this window. Generally the intensity is a vector parallel to the vector $\mathbf{v}$ of the particle velocity and is given by [10]

$$\mathbf{I} = p\mathbf{v} \tag{2.12}$$

the principle of energy conservation requires

$$\frac{\partial w}{\partial t} + div\mathbf{I} = 0 \tag{2.13}$$

It should be noted that in contrast to the sound pressure and particle velocity these energetic quantities do not simply add if two waves are superimposed on each other. In a plane wave the sound pressure and the longitudinal component of the particle velocity are related by $p = \rho_0 cv$ and the same holds for a spherical wave at a large distance from the centre. Hence it can express the particle velocity in terms of the sound pressure. [10]

Then the energy density and the intensity are

$$w = \frac{p^2}{\rho_0 c^2} \tag{2.14}$$

$$I = \frac{p^2}{\rho_0 c} \tag{2.15}$$

and they are related by

$$I = cw \tag{2.16}$$

2.8.3 Intensity of Harmonic wave

Stationary signals which are not limited in time may be characterised by time averages over a sufficiently long time. Introducing the root-mean-square of the sound pressure by[10]

$$p_{rms} = \left(\frac{1}{t_a} \int_0^{t_a} p^2 \, \mathrm{d}t \right)^{1/2} \tag{2.17}$$

The Eq. 2.14 and 2.15 yield

$$w = \frac{p_{rms}^2}{\rho_0 c^2} \tag{2.18}$$

$$I = \frac{p_{rms}^2}{\rho_0 c} \tag{2.19}$$

Finally, for a harmonic sound wave with the sound pressure amplitude p_{max}, where p_{rms} equals to $p_{max}/\sqrt{2}$, which leads to these equations: [10]

$$w = \frac{p_{max}^2}{2\rho_0 c^2} \tag{2.20}$$

$$I = \frac{p_{max}^2}{2\rho_0 c} \tag{2.21}$$

2.9 Sound Levels and Decibel

A term decibel is one of the most important units of measure in the audio field. Decibels can be applied in many different ways. In particular, decibels are used to measure sound levels in various applications. Sound levels expressed in decibels clearly demonstrate the wide range of sensitivity in human hearing. The threshold of hearing matches the ultimate lower limit of perceptible sound in air, the noise of air molecules against the eardrum. At the other end of the range, the ear can tolerate very high intensities of sound. A level expressed in decibels is a convenient way of handling the billion-fold range of sound pressures to which the ear is sensitive.[2]

2.9.1 Logarithms

Representing 100 as 10^2 simply means that $10 \times 10 = 100$. Similarly, 10^3 means $10 \times 10 \times 10 = 1{,}000$. But how about 267? That is where logarithms are useful. Logarithms are proportional numbers and a logarithmic scale is one that is calibrated proportionally. It is agreed that 100 equals 10^2. By definition we can say that the logarithm of 100 to the base 10 equals 2, commonly written log10 100 = 2, or simply log 100 = 2, because common logarithms are to the base 10. The number 267 can be expressed as 10 to some power between 2 and 3. Avoiding the mathematics, we can use a calculator to enter 267, push the "log" button and 2.4265 appears. Thus, $267 = 10^{2.4265}$ and log 267 = 2.4265.[2]

Logarithms are particularly useful to audio engineers because they can correlate measurements to human hearing, and they also allow large ranges of numbers to be expressed efficiently. Logarithms are the foundation for expressing sound levels in decibels where the level is a logarithm of a ratio. In particular, a level in decibels is 10 times the logarithm to the base 10 of the ratio of two powerlike quantities.[2]

2.9.2 Decibels

The intensities can be expressed as logarithms of the ratios. An intensity I can be expressed in terms of a reference I_{ref} as follows:[2]

$$log_{10}\frac{I}{I_{ref}}\ [bels] \tag{2.22}$$

The intensity measure is dimensionless, but to clarify the value, it will be assigned as the unit of a bel (from Alexander Graham Bell). However, when expressed in bels, the range of values is somewhat small. To make the range easier to use, we usually express values in decibels. The decibel is 1/10 bel. A decibel (dB) is 10 times the logarithm to base 10 of the ratio of two quantities of intensity (or power). Thus, the intensity ratio in decibels becomes:[2]

$$IL = 10\ log_{10}\frac{I}{I_{ref}}\ [decibels] \tag{2.23}$$

This value is called the sound intensity level (IL in decibels) and differs from intensity. Using decibels is a convenience and decibel values are more closely to follow the way of hearing the loudness of sounds.[2]

Sometimes levels are expressed other than intensity in decibels. Equation 2.24 applies equally to acoustic intensity, as well as acoustic power, electric power, or any other kind of power. For example, it can be written as the sound-power level:[2]

$$PWL = 10\ log_{10}\frac{W}{W_{ref}}\ [decibels] \tag{2.24}$$

Where:
PWL ... is sound power level, dB
W ... is sound power, watts
W_{ref} ... is a reference power, 10^{-12}

Sound intensity is difficult to measure. Sound pressure is usually the most accessible parameter to measure in acoustics (just as voltage is for electronic circuits). For this reason, the sound-pressure level (SPL) is often used. SPL is a logarithmic value of the sound pressure, in the same way that the sound intensity level (IL) corresponds to sound intensity. SPL is approximately equal to IL; both are often referred to as the sound level. Acoustic intensity (or power) is proportional to the square of the acoustic pressure p. This slightly alters the defining equation that we use. When the reference pressure is 20μPa, a sound pressure p measured in micropascals has a sound-pressure level (SPL) of:[2]

$$SPL = 10\ log_{10}\frac{p^2}{p_{ref}^2} = 20\ log_{10}\frac{p}{20\mu Pa}\ [decibels] \tag{2.25}$$

Where:

SPL ... is sound pressure level, dB

p ... is acoustic pressure, μPa or other

p_{ref} ... is acoustic reference pressure, μPa or other

2.9.3 Reference Levels

Reference levels are widely used to establish a baseline for measurements. For example, a sound-level meter is used to measure a certain sound-pressure level. If the corresponding sound pressure is expressed in normal pressure units. The sound-pressure reference p_{ref} must be standardized, so that ready comparisons can be made. Several such reference pressures have been used over the years, but for sound in air the standard reference pressure is 20μPa.[2]

This might seem quite different from the reference pressure of 0.0002 microbar or 0.0002 $dyne/cm^2$, but it is the same standard merely written in different units. This is a very small sound pressure (0.0000000035 lb/in^2) and corresponds closely to the threshold of human hearing at 1 kHz. The relationship between sound pressure in pascals, pounds/square inch and sound-pressure level is shown in Fig. 2.17.[2]

2.10 Sound in Free Fields

Sound in a free field travels in straight lines and is unimpeded. Unimpeded sound is not subject to the many influences that we will consider in later chapters. Sound in a free field is unreflected, unabsorbed, undeflected, undiffracted, unrefracted, undiffused and not subjected to resonance effects. In most practical applications, these are all factors that could (and do) affect sound leaving a source. An approximate free field can exist in anechoic chambers, special rooms where all the interior surfaces are covered by sound absorbers. But generally, a free field is a theoretical invention, a free space that allows sound to travel without interference.[2]

The circles in Fig. 2.18 represent spheres having radii in simple multiples. All of the sound power passing through the small square area A1 at radius r also passes through the areas A2, A3 and A4 at radii $2r$, $3r$ and $4r$, respectively. The same sound power flows out through A1, A2, A3 and A4, but an increment of the total sound power traveling in this single direction is spread over increasingly greater areas as the radius is increased. Thus, intensity decreases

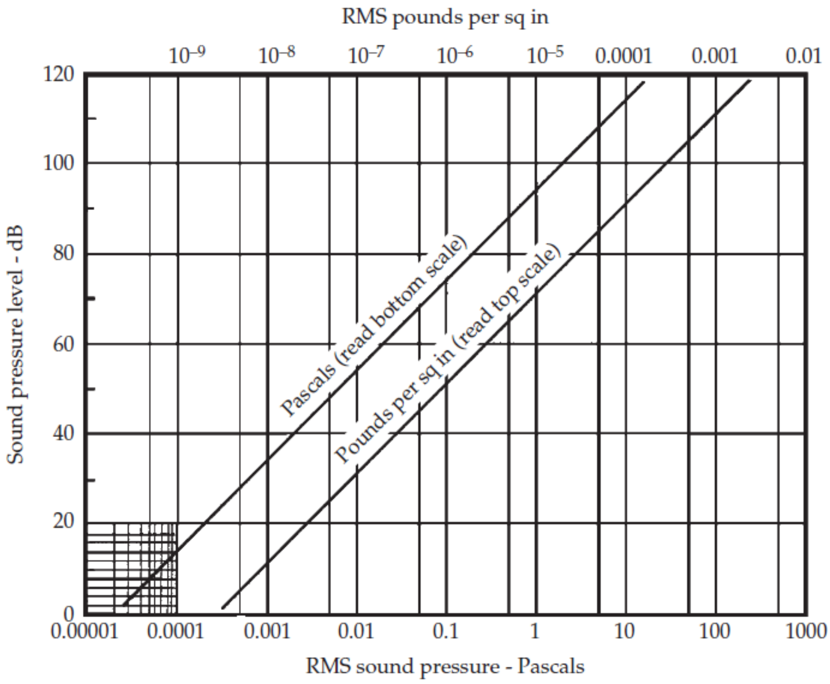

Figure 2.17: This graph shows the relationship between sound pressure in pascals or pounds/square inch and sound-pressure level (referred to 20μPa). This is a graphical approach to the solution of Eq. 2.24.[2]

with distance. This decrease is due to geometric spreading of the sound energy, and is not loss in the strict sense of the word.[2]

2.10.1 Sound Intensity in the Free Field

The area of a sphere is $4\pi r^2$. Therefore, the area of any small segment on the surface of the sphere also varies as the square of the radius. This means that the sound intensity (sound power per unit area) decreases as the square of the radius. This is an inverse square law. The intensity of a point-source sound in a free field is inversely proportional to the square of the distance from the source. In other words, intensity is proportional to $1/r^2$. More specifically:

$$I = \frac{W}{4\pi r^2} \tag{2.26}$$

Where:

I ... is intensity of sound per unit area

W ... is power of source

r ... is distance from source (radius)

In this equation, since W and 4π are constants and doubling the distance from r to $2r$ reduces the intensity I to $I/4$; this is because at twice the distance, the sound passes through an area that is four times the previous area. Likewise, tripling the distance reduces the intensity to $I/9$, and quadrupling the distance reduces intensity to $I/16$. Similarly, halving the distance from $2r$ to r increases the intensity to $4I$.[2]

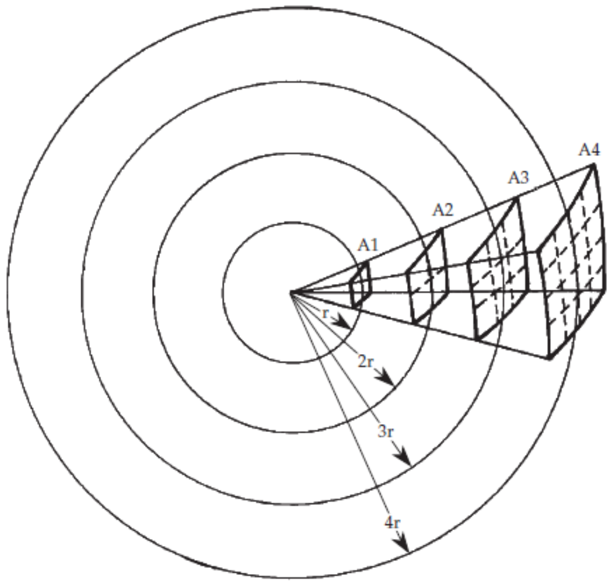

Figure 2.18: In the solid angle shown, the same sound energy is distributed over spherical surfaces of increasing area as radius r is increased. The intensity of sound is inversely proportional to the square of the distance from the point source.[2]

2.10.2 Sound Pressure in the Free Field

The intensity of sound (power per unit area) is a difficult parameter to measure. However, sound pressure is easily measured, for example, by using ordinary microphones. Because the sound intensity is proportional to the square of sound pressure, the inverse square law (for sound intensity) becomes the inverse distance law (for sound pressure). In other words, sound pressure is inversely proportional to distance r.[2]

$$P = \frac{k}{r} \tag{2.27}$$

Where:

P ... is sound pressure

k ... is a constant

r ... is distance from source (radius)

For every doubling of distance r from the sound source, sound pressure will be halved (not quartered). In Fig. 2.19 the sound-pressure level in decibels is plotted against distance. This illustrates the basis for the inverse distance law: when the distance from the source is doubled, the sound-pressure level decreases by 6 dB. This applies only for a free field. This law provides the basis for estimating the sound-pressure level in many practical circumstances.[2]

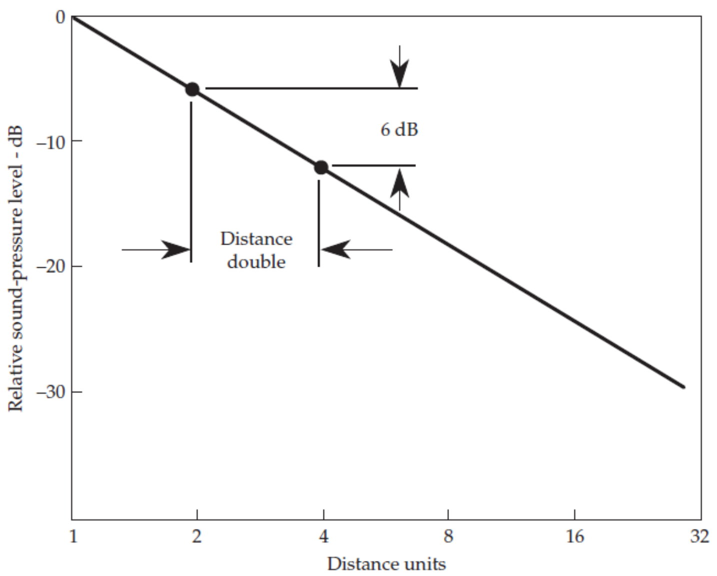

Figure 2.19: The inverse square law for sound intensity equates to the inverse distance law for sound pressure. This means that sound-pressure level is reduced 6 dB for each doubling of the distance.[2]

Chapter 3

Room Acoustics

The previous chapter described the basic properties of the sound that spread especially all around a free space without any obstacles or absorbing objects. It means that sound propagation was unbounded and spread in every directions. This chapter will deal with the acoustics in a standard size rooms with basic properties, ie. propagation of sound in an enclosed space surrounded by standard walls and its acoustics in room in dependence on the geometrical acoustics in the room.

These rooms are free of any objects that could cause the reflectivity of sound propagation from all four walls, floor and ceiling. Another objective of this chapter is to describe these parameters such as reflection, absorption, scattering and wall impedance.

3.1 Energy balance at the impact of sound waves on the obstacle

When the impact of sound waves on an obstacle (eg. a wall surface) the part of the sound waves are reflected and next part are consumed. In addition, the still part of the sound waves pass into the space behind the wall. The energy balance at impact sound waves at a wall is shown in figure 3.2. [17]

The sound power incident on 1 m^2 surface of the wall (ie the intensity of the sound waves hitting the obstacle) I_0 is divided into the following sub-components:

- I_1 - sound intensity of the reflected waves
- I_2 - sound intensity of the absorbed waves
- I_3 - sound intensity of the emitted waves over the wall (for total waves)
- I_4 - sound intensity of the waves passed through a wall by openings and pores
- I_5 - sound intensity of the waves radiated by the wall due to bending vibrations in second half-space
- I_6 - sound intensity of the waves which are led in the form of vibrations to other parts of adjacent structures
- I_7 - sound intensity which is transformed into heat in the wall

All divided intensities described above are shown in Fig. 3.2.

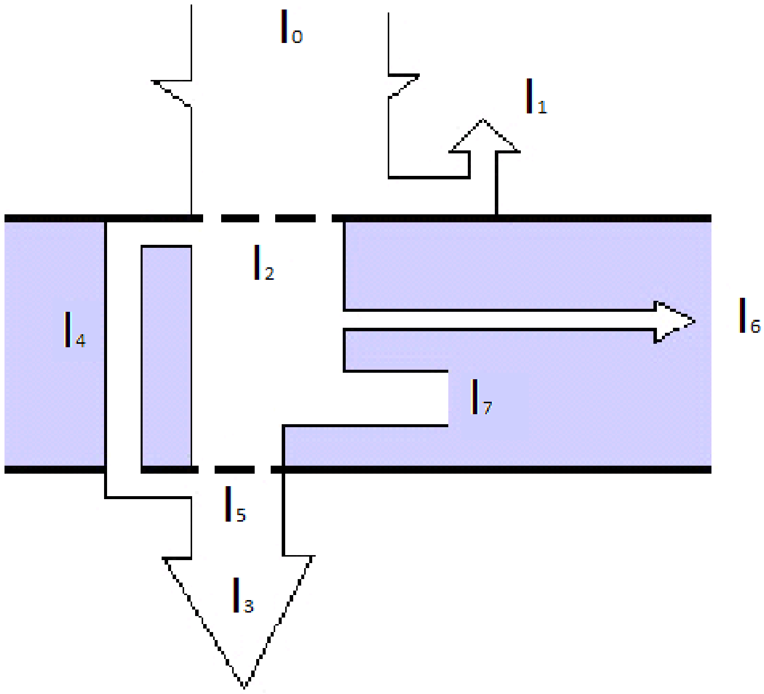

Figure 3.1: Energy balance at the impact of sound waves on the wall. [17]

3.2 Absorption, Reflectiong and Scattering

3.2.1 Sound Absorption

On the basis of the energy balance upon impact sound waves at the wall (see figure 3.2) it can defined a factor of sound. The ability of the body to absorb sound wave is characterized by the absorption coefficient α, which is determined by the ratio of the energy absorbed in a particular area, to the incident direct energy to this area. It can be expressed by the equation: [17]

$$\alpha = \frac{I_2}{I_0} \tag{3.1}$$

From the viewpoint of the law of energy conservation, it is obvious that the sound absorption coefficient $\alpha \in < 0, 1 >$. The wall, in which there is a total engulfment of any incident acoustic energy is characterized by the absorption coefficient $\alpha = 1$. The most suitable materials for sound absorption are recommended especially materials with porous or fibrous structure. Conversely, in the case of the perfect reflection of incident sound waves from the wall surface is then characterized by the absorption factor $\alpha = 0$. [17]

In addition to the type of the concrete material the size of absorption coefficient is dependent on many other factors notably the frequency of the incident sound waves, the material thickness, temperature distribution and the pore size of the material, etc. Values of sound absorption coefficient for some materials and their thickness t depending on the frequency f are given in the table 3.1. [17]

Materials	**t [mm]**	**125 Hz**	**200 Hz**	**500 Hz**	**1000 Hz**	**2000 Hz**	**4000 Hz**
Asbestos	25	0.25	0.6	0.65	0.6	0.6	0.61
Concrete	-	0.01	0.01	0.02	0.02	0.02	0.03
The board acoustic absorption (specially treated)	25	0.22	0.51	0.89	0.98	0.71	0.66
Polished Stone	-	0.1	-	0.01	-	-	0.02
Carpet Weaving (on a concrete base)	9.5	0.09	0.08	0.21	0.26	0.27	0.37
Carpet Weaving (the cardboard thickness 3.10-3 m)	8	0.11	0.14	0.37	0.43	0.27	0.25
Linoleum (on a concrete base)	3	0.02	0.03	0.03	0.04	0.04	0.05
Sand (dry)	100	0.15	0.35	0.4	0.5	0.55	0.8
Felt	25	0.12	0.32	0.51	0.62	0.60	0.56
Wooden Plywood (three-layer)	3	0.2	0.28	0.26	0.09	0.12	0.11
Snow	25	0.15	0.40	0.65	0.75	0.80	0.85
Snow	100	0.45	0.75	0.90	0.95	0.95	0.95
Cinder	280	0.90	0.90	0.75	0.80	-	-
Stucco on Metal Mesh	19	0.04	0.05	0.06	0.08	0.04	0.06
Glass Wool (pressed)	25	0.24	0.30	0.57	0.69	0.70	-
Slag Wool	25	0.26	0.45	0.61	0.72	0.75	-
Resin Bonded Glass Wool	25	0.20	0.41	0.75	0.86	0.86	0.82
Resin Bonded Glass wool	51	0.41	0.60	0.99	0.99	0.84	0.85
Velour Curtains	-	0.05	0.12	0.35	0.45	0.38	0.36
Brick Wall	-	0.02	0.02	0.03	0.04	0.05	0.07

Table 3.1: The values of sound absorption coefficient of some materials. [17]

For other possible calculation of the absorbed intensity I_2 that is absorbed in the wall with a known direct intensity I_0 incident on the wall and the reflected I_1 intensity from the wall after the modify equations 3.2 and 3.3 we can get the another equation for absorption coefficient.

$$I_2 = \alpha I_0 \tag{3.2}$$

$$I_1 = (1 - \alpha) I_0 \tag{3.3}$$

$$\alpha = 1 - \frac{I_1}{I_0} \tag{3.4}$$

3.2.2 Reflection Coefficient

Sound reflectivity of the factor R is the ratio of the intensity of the reflected sound waves from the wall to the intensity of sound waves incident on the wall: [17]

$$R = \frac{I_1}{I_0} \tag{3.5}$$

Like the sound absorption coefficient, the size of the acoustic reflectance is in the interval $R \in < 0, 1 >$. The wall of sound with perfect reflectivity is characterized by reflectance factor $R = 1$. If all of the incident acoustic energy absorbed in the wall, then the factor$R = 0$. In the terms of energy view, the following relationship between sound absorption coefficient and sound reflection coefficient must apply: [17]

$$\alpha + R = 1 \tag{3.6}$$

From the relationship above (equation 3.6) it is apparent that part of the incident acoustic wave energy is absorbed in the wall and the remaining part is reflected from the wall. [17]

In the figure 3.2, it is also apparent that the part of the absorbed sound energy is penetrated over the wall called transmission coefficient, another part in the wall are transformed into thermal energy and the rest of energy is spread in the form of vibration to other parts of adjacent structures. Because this factor the acoustic sound transmission factor τ and the thermal conversion factor ε for which relationship apply: [17]

$$\tau = \frac{I_3}{I_0} = \frac{I_4 + I_5}{I_0} \tag{3.7}$$

$$\varepsilon = \frac{I_7}{I_0} \tag{3.8}$$

On the figure 3.2 there is also shown the sound intensity I_6 propagating by vibration to other parts adjacent structures. In the case of antinoise insulating metal sheet covers its value is negligible. Then, based on the law of conservation of energy can be written: [17]

$$R + \tau + \varepsilon = 1 \tag{3.9}$$

Comparing equations 3.6 and 3.9 can be written:

$$\alpha = \tau + \varepsilon \tag{3.10}$$

3.3 Sound reflection at normal incidence

3.3.1 Acoustic Impedance

The acoustic impedance Z of a material is defined as

$$Z = \rho c \tag{3.11}$$

The SI unit for acoustic impedance is $\mathrm{kg/(m^2 s)}$, with the special name the rayl, where 1 rayl is equal to 1 $\mathrm{kg/(m^2 s)}$

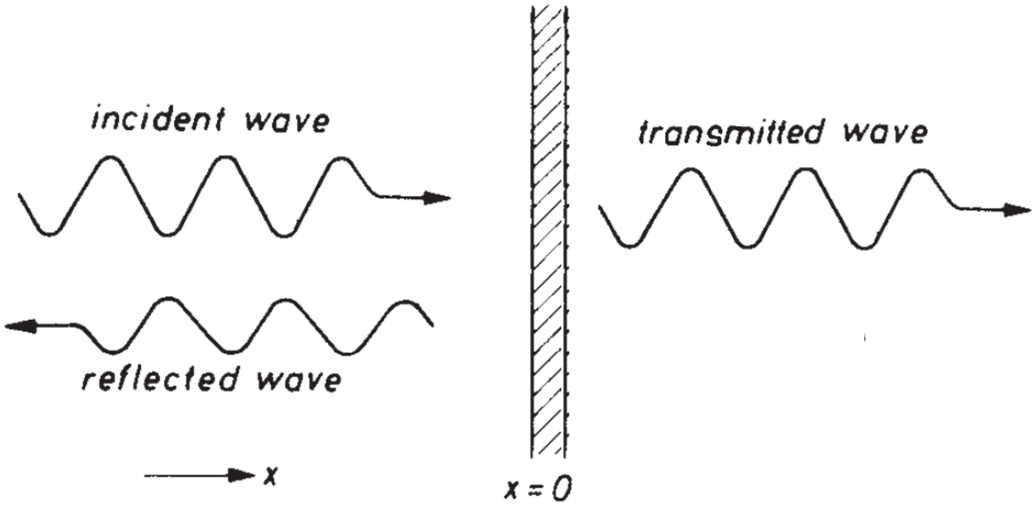

Figure 3.2: Refrection of a normally incident sound wave from a plane surface. [10]

3.3.2 Different Impedance

It is possible to gain equations for reflection and transmission coefficients by using impedance of environment. Where the sound come from one medium with impedance Z_1 (for example impedance of air) to another medium with impedance Z_2 (for example impedance of wall) as shown in Fig. 3.2. By using and expanding of Eq. 3.5 for calculation of reflection coefficient we can get

$$R = \frac{I_1}{I_0} = \left(\frac{Z_1 - Z_2}{Z_1 + Z_2}\right)^2 \tag{3.12}$$

Transmission coefficient τ can be calculated by passing sound wave through an interface between impedances Z1 and Z2 defined as

$$\tau = \frac{I_3}{I_0} = \frac{4Z_1Z_2}{(Z_1 + Z_2)^2} \tag{3.13}$$

Chapter 4

Preparation and Measurement

In this part of diploma thesis are described principles of measuring to get the best or the more possible measuring data to calculate coefficients which are important and are also the basic topic for this thesis. There are many norms which describe principle of measuring. This thesis was inspired by two norms which involve many measuring principles to get the most possible applicable values.

The first one is called 'Road traffic noise reducing devices - Test method for determining the acoustic performance - Part 4: Intrinsic characteristics - In situ values of sound diffraction' [15] and the second one is 'Road traffic noise reducing devices - Test method for determining the acoustic performance - Part 5: Intrinsic characteristics - In situ values of sound reflection and airborne sound insulation' [16] .

This chapter further contains a description of issues during measuring and preparation. Very important was preparation for measuring which try to minimize mistakes and their uncertainty. It was really necessary get the most possible isolated acoustic room where its wall absorbed reflected sounds generated in the room. For measuring were used usual accessories like smart phone with microphone, cheap loudspeaker and mobile application with playing and recording at the same time.

4.1 Preparation

4.1.1 Room parameters

As mentioned above, it was necessary to find isolated room or a room with big dimensions to eliminate reflected sound waves from all walls or from a ceiling. It is one option, another option is free open space without any parasitic noises like traffic noises, common noises, mild or strong wind and objects situated in nearby of measurement. Because of these disadvantage then the isolated room was selected as the fittest option for measuring.

For measuring a small acoustic isolated room was chosen which complies minimum requirements for measuring. Although it seems that it is a small room, so isolation parameters of all walls and ceiling are sufficient to absorb the reflection or parasitic sound waves. Therefore room dimensions can be neglected in this case.

Measurement in a small room without isolation is more difficult regarding measuring distances between microphone and all walls, microphone and ceiling and sometime even distance between microphone and floor in the case when a sample is attached in air above a floor. These conditions or more precisely distances are shown in Fig. 4.1. The conditions regarding distances are described in section 4.2.1. Of course even if a measurement is performed in isolated room, it is necessary to briefly observe these condition to get better results.

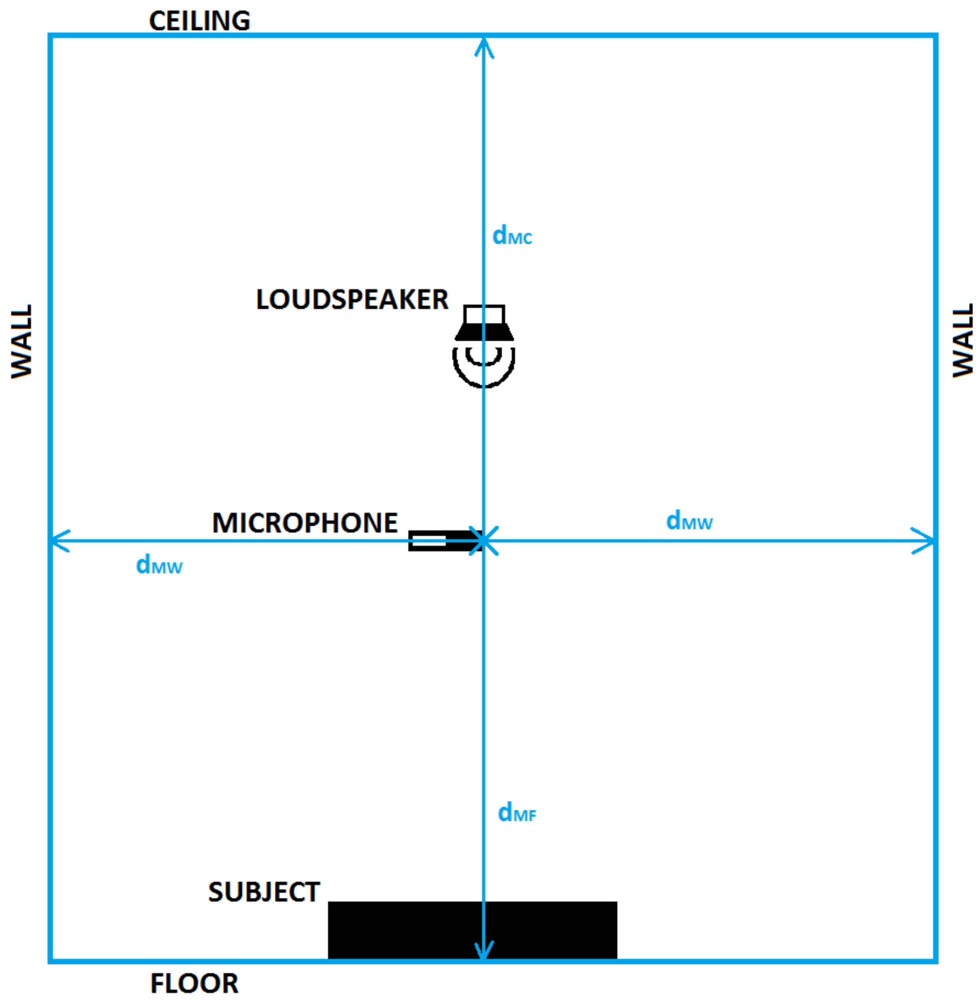

Figure 4.1: Distances from microphone attached in the middle of a room. dMC - distance between microphone and ceiling, dMW - distance between microphone and wall, dMF - distance between microphone and floor

4.1.2 Used Frequencies

At used frequencies, very important was their time duration which was set within 4 millisecond and also selection of concrete frequencies was important as well. There was an issue with membrane of a loudspeaker when some frequencies does not fit into time duration of 4 millisecond. It means that a beginning of used frequency have to start from zero amplitude and end of used frequency have to end either as a whole period or as a half of period. See Fig. 4.2 which shows fifteen frequencies used within time duration of 4 millisecond where redly marked frequencies do not comply condition. At the ends of 4 millisecond windows these frequencies are cutted and forced to zero amplitude.

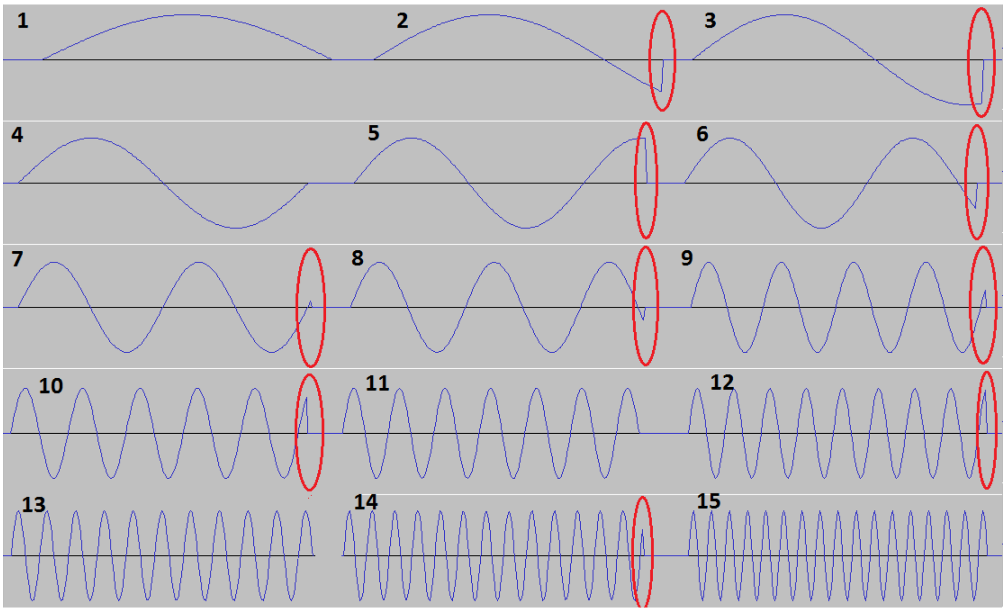

Figure 4.2: Fifteen different frequencies used within 4 millisecond window; 1 - 125 Hz, 2 - 158 Hz, 3 - 197 Hz, 4 - 250 Hz; 5 - 315 Hz; 6 - 393 Hz; 7 - 500 Hz; 8 - 615 Hz; 9 - 774 Hz; 10 - 1000 Hz; 11 - 1200 Hz; 12 - 1500 Hz; 13 - 2000 Hz, 14 - 2400 Hz, 15 - 3000 Hz

An invalid data are given if this condition is not complied see Fig. 4.3 where are used two frequecies. The first one does not comply condition and the result of recorded sample after ending of used frequency are parasitic higher frequencies marked in the red window. The second sample which comply condition does not have an parasitic frequencies which reduce valid measured data.

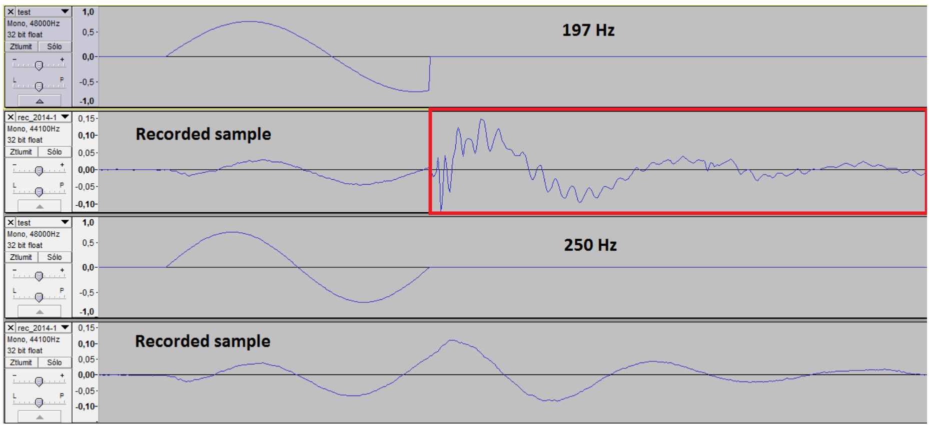

Figure 4.3: Comapre of two frequencies used for measurement; Fist sample with frequency 197 Hz and recorded sample; Second sample with 250 Hz and recorded sample

Frequencies which comply condition described above and which were used for measurement are 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz and 8000 Hz. As shown in Fig. 4.4 all these frequencies nicely comply this condition. Used frequencies will be called as pulses which contains all used frequencies for measurement.

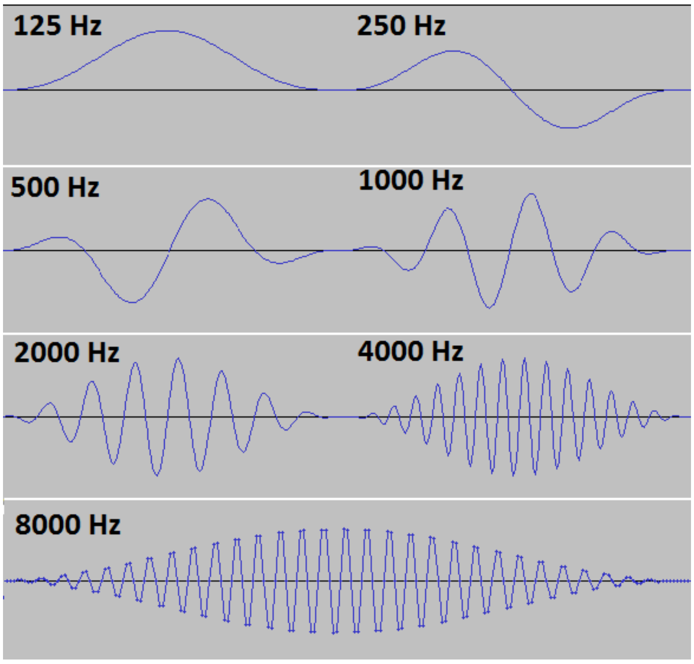

Figure 4.4: Used frequencies for measurement which comply conditions for better results

The Fig. 4.5 shows the whole sound sample which contains all pulses and a start pulse. Start pulse is created from white noise and it informs about beginning of measurement. Other pulses are sorted from the lowest frequency to the highest frequency. The interval between used pulses is around 200 millisecond and this interval seems to be as an ideal for stand alone measurement.

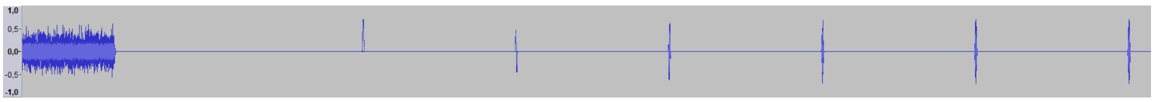

Figure 4.5: Sample of used frequencies prepeared for measurement

4.1.3 Devices and Accessories

For stand alone measurement were used devices like smartphone, loudspeaker, different samples and attached equipments for settings of loudspeaker and smartphone. Smartphone was used for playing and recording during measurement. In Fig.4.6 is smartphone Samsung Galaxy Note 2 with two microphone where all parameters of microphones are described in datasheet [5]. Upper microphone is used for reducinng of parasitic sounds during calling and basic microphone for measurement is lower one which will used for recording during measurement.

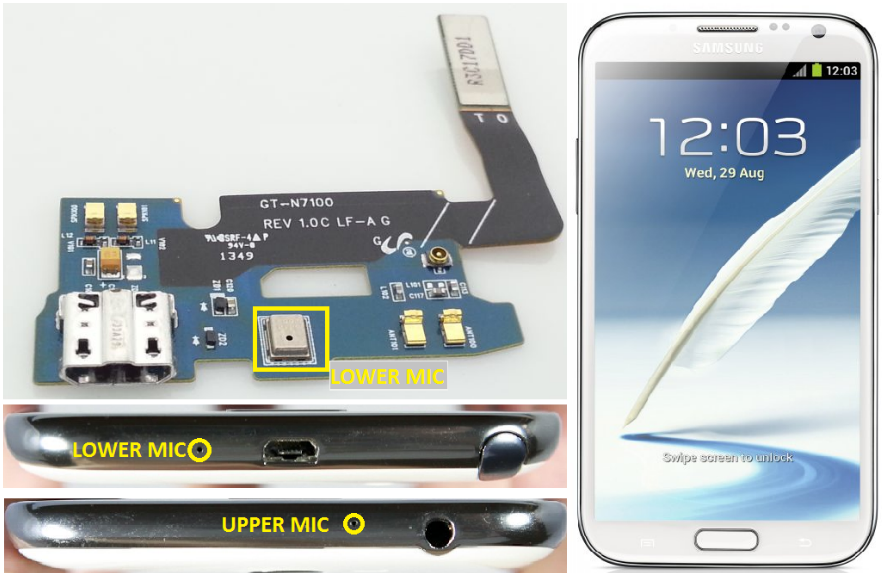

Figure 4.6: Samsung Galaxy Note 2, Microphone BLM15AG121SN1D

Another important device for measurement is small loudspeaker Defender Wild Beat shows in Fig. 4.7 whith basic features. This loudspeaker has basic features without special and additional properties. There was used another bigger loudspeaker (Dynaudio BM5A) with more sophisticated properties but it was not physically possible to measure with it like with smaller loudspeaker.

Figure 4.7: Loudspeaker Defender Wild Beat

4.2 Measurement

Measurement was executed in a small isolated room with three different samples: concrete board, polystyrene board and a last one is a rubber board and devices described in section 4.1.3. All these three boards are shown in Fig. 4.8 and Tab. 4.1 contains dimensions of all measured samples.

Figure 4.8: Samples for measurement; From the left side it is concrete, in middle is polystyrene and on the right side is rubber board

Samples	**Dimension**	**Thickness**
Concrete	50x50 cm	5 cm
Polystyrene	66x50 cm	5 cm
Rubber	49x49 cm	3 cm

Table 4.1: Dimensions and thickness of samples

4.2.1 Conditions of Measurement

For more simpler data analysis it is necessary to take into account all information which were described above and another will be described in this section. In Fig. 4.9 are another distances which describe position among loudspeaker, microphone and sample situated within a room. There are also displayed direction of sound waves from loudspeaker where all these three directions present parasitic reflected sound waves which are recorded into microphone.

Height between microphone and sample have to be so far as it can be possible because the direct wave must be totally recorded when the reflected wave from the sample reaches the microphone. In the fact it means that the duration of the pulses emitted by the loudspeaker must be shorter than the time needed for the wave goes from the microphone to the sample and then to the microphone again. An Eq. 4.1 set the condition for pulse duration and distance between microphone and sample.

$$\tau_p \leq \frac{2h_M}{C_s} \tag{4.1}$$

Where:
τ_p ... is pulse duration [sec]
C_s ... is sound velocity in the air [m/s]

As seen in Eq. 4.1 it could be said that the pulse duration and height of microphone above sample are related together. It means that higher the microphone is placed, that longer duration of the pulse can be. But this measurement has constant pulse duration, and is easier to set height of microphone from the sample than modify pulse duration.

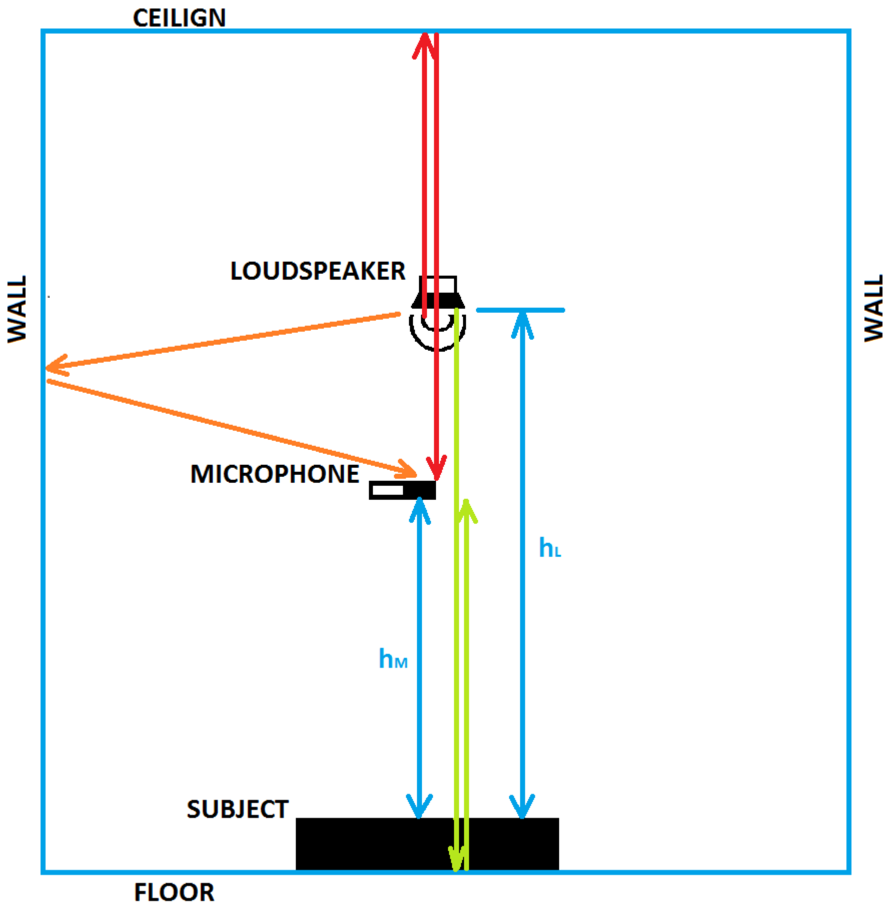

Figure 4.9: Distances and directions of sound waves; hL - height between loudspeaker and microphone, hM - height between loudspeaker and sample; Orange direction - reflection from the wall to microphone, Red direction - reflection from ceiling to microphone, Green direction - reflection from floor to microphone

Using distances defined in Fig. 4.9 and distances in Fig. 4.1 the conditions could be created for getting better data without affecting of parasitic sound reflections shown in Fig. 4.9. Below are created conditions regarding parasitic reflections which will be reduced or eliminated as much as possible. Firstly is important to get and record whole direct impulse which is emitted from the loudspeaker. Considering the time $t = 0$ as the beginning of the pulse emitting from the loudspeaker then the required reflected pulse can be recorded until ending recording of direct impulse:

$$\tau_p + \frac{h_L - h_M}{C_s} \tag{4.2}$$

Before that time, defined in Eq. 4.2, all parasitic reflections should not be recorded by microphone. Another three equations define the minimum of required distances between:

- the microphone and the ceiling (see Eq. 4.6)
- the microphone and the floor (see Eq. 4.7)
- the microphone and the wall (see Eq. 4.8)

All these equations are obtained to correspond conditions of all three distances to avoid parasitic reflections. Firstly is necessary to set equations with conditions:

$$\frac{2\sqrt{\left(\frac{h_L-h_M}{C_s}\right)^2+d_{MW}^2}}{C_s} \geq \tau_p + \frac{h_L-h_M}{C_s} \tag{4.3}$$

$$\frac{h_L-h_M+2d_{MF}}{C_s} \geq \tau_p + \frac{h_L-h_M}{C_s} \tag{4.4}$$

$$\frac{2\left[d_{MC}-(h_L-h_M)\right]+h_L-h_M}{C_s} \geq \tau_p + \frac{h_L-h_M}{C_s} \tag{4.5}$$

Equations 4.3, 4.4 and 4.5 are important to obtain the minimal distances required to guarantee that the parasitic reflections will not distrub the recording.

$$d_{MW} \geq \sqrt{\frac{(h_L-h_M+C_s\,\tau_p)^2}{2}-\left(\frac{h_L-h_M}{2}\right)^2} \tag{4.6}$$

$$d_{MF} \geq \frac{C_s\,\tau_p}{2} \tag{4.7}$$

$$d_{MC} \geq (h_L-h_M)\,\frac{C_s\,\tau_p}{2} \tag{4.8}$$

4.2.2 Process of Measurement

Measurement was provided in a small isolated room where all walls and ceiling were acoustically isolated. Therefore two conditions regarding distances from the microphone to the walls (see Eq. 4.6) and from microphone to the ceiling (see Eq. 4.7) can be neglected. But for surer measurement these two conditions were complied. In the Fig. 4.10 is displayed schematic of measurement, where distance between loudspeaker and microphone is still constant during whole measurement. The only height of the microphone above the sample is changing throughout whole measurement.

For each measured sample was set height of the microphone above sample and for each height of the microphone were recorded ten measurements. This process of ten measurements, was applicated at another heights of the microphone. It means that each sample has seven measurements with different height of the microphone and each height has ten records of measurement. At the end of measurement, each sample will have 70 records.

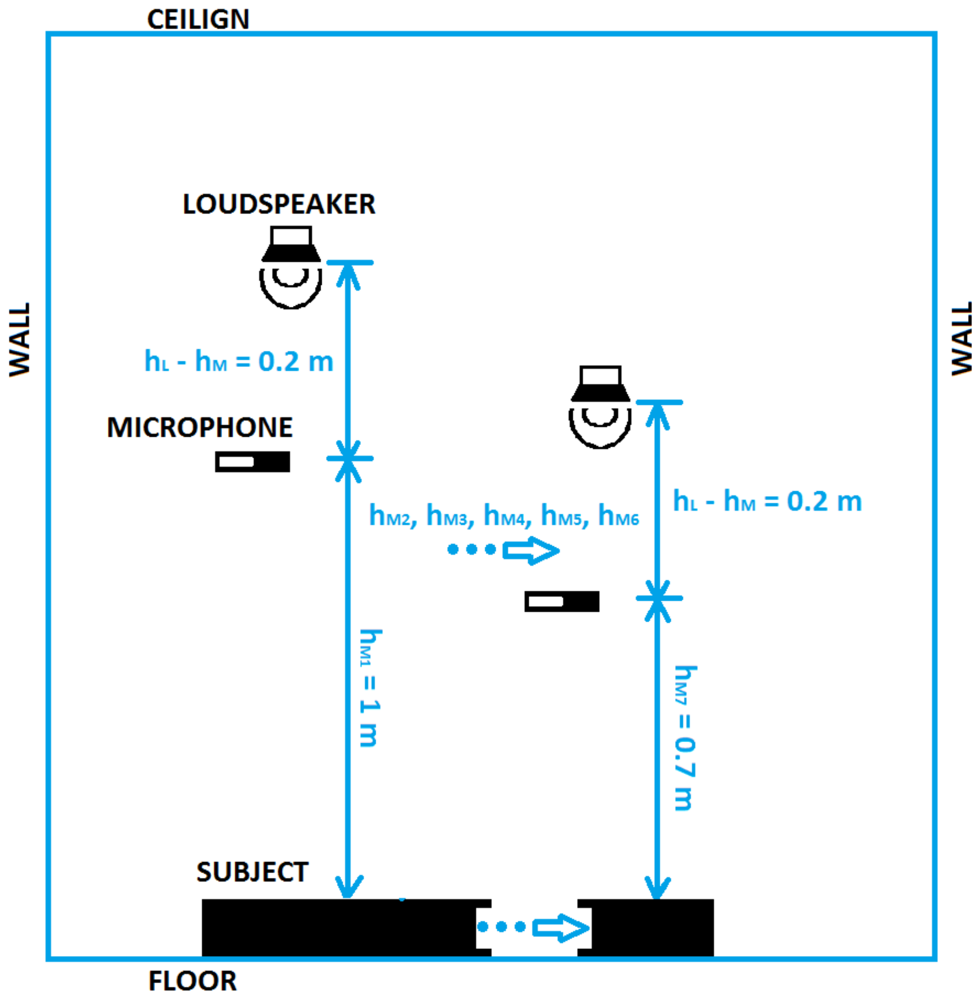

Figure 4.10: Principe of measurement where constant distance between microphone and loudspeaker is 0.2 meters; hM1, ..., hM7 are changing height of microphone in the interval from 1 to 0.7 meters after 0.05 meters

All heights of microphone used for measurement comply conditions for reducing of reflected parasitic sound waves. In Fig. 4.11 are shown ideal waveforms without parasitic waves within direct or parasitic waves within reflected waves. The point 3 in Fig. 4.11 is time delay between end of the direct pulse and start of the reflected pulse. This delay is depend on height of microphone and Eq. 4.9 calculates time delay. All delays of reflected waves are shown in Tab. 4.2 which contains ideal expected time calculated by Eq. 4.9.

$$\tau_{delay} = \frac{2h_M}{C_s} - \left(\tau_p + \frac{h_L - h_M}{C_s}\right) \tag{4.9}$$

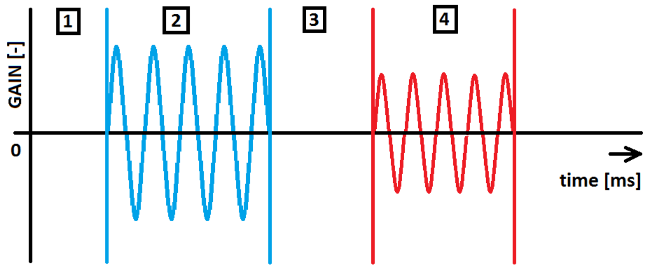

Figure 4.11: The waveform of direct and reflected waves; 1 - time when the microphone detect begenning of the direct puls, 2 - time duration of direct pulse, 3 - delay determined by height of microphone, and the first detection of reflected pulse, 4 - time duration of reflected pulse (time duration should be the same as the duration of direct pulse)

Time Delay [m]	**0.7**	**0.75**	**0.80**	**0.85**	**0.90**	**0.95**	**1.00**
Time Delay [ms]	0.118	0.412	0.706	1	1.294	1.588	1.882

Table 4.2: Time Delay between direct and reflected pulse [velocity of sound was set as 340 m/s]

In Fig. 4.12 are shown principles of measurement and whole fixing system during measurement. For designation of distances was used normal tape measure. Uncertainty of measurement are described in section 5.3, where is important to include setting of fixing system during measurement by using tape measure and rounding according human eye.

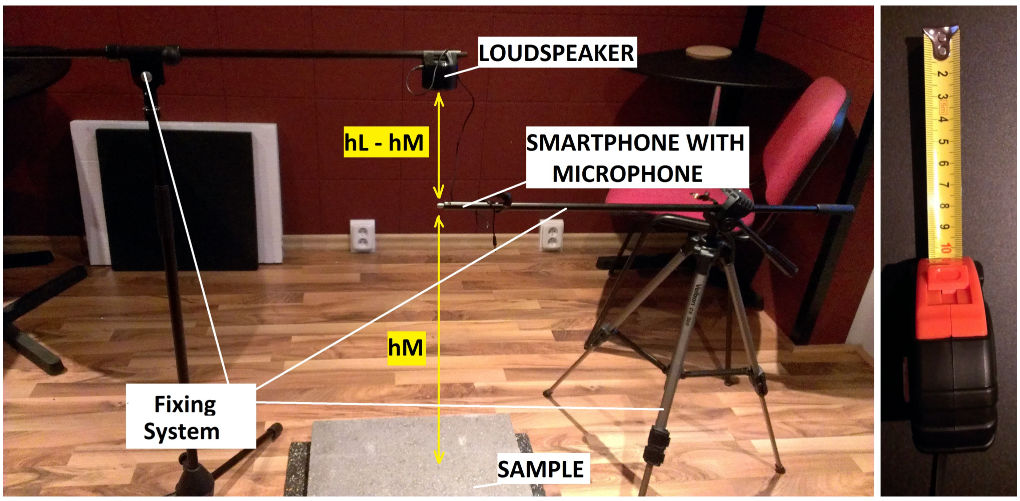

Figure 4.12: On the Left - Used Devices and Fixing System; On the Right - Tape Measure

Chapter 5

Data Analysis

This chapter contains analysis of measured data where important is as accurate as possible detection of beginnings at all used pulses from one recording. One recording should have eight pulses, where first one is the start pulse of whole recording is as shown in Fig. 4.5.

This data analysis is performed also with respect to get pressure of direct and reflected sound pulses. These pressures are essential for another analysing of coefficients what is important as well. Required coefficients are absorption and reflection coefficients where both are defined in section 3.2.

The chapter also contains GUI application which was created for data analysis and detection of direct and reflected pulses. This GUI application is created in MATLAB where two basic functions are set where the first one is automatic detection of beginning of pulses, and the second one is manual detection of the beginning for more accurate detection if automatic detection will be not succesfull.

5.1 Pulse Detection

As was mentioned, the detection of direct pulses is important and even more important is its precise detection of beginning. With precise detection of beginning of direct pulses, the beginning of reflected pulses can be detected as well. Function for detection of pulse beginning was implemented in Matlab, function is called *pulse_detect* (see Appendix B).

This function has got four input variables and one output variable:

- input variables
  - *signal_data* ... Data for processing.
  - *number_pulse* ... Number of the pulses in *signal_data*
  - fs ... Sampling frequency of the *signal_data*
  - Tp ... Pulse duration in seconds.
- output variables
  - *ident_detect_value* ... The coordinates of the individual beginnings of detected pulses in the *signal_data*.

The function *pulse_detect* starts with obtaining a treshold which is set as the maximal found absolute value of *signal_data* divided by 100 and the result is marked as the main treshold. This treshold increases its value each cycle, until the input of *number_pulse* is not equal to a number of detected pulses. The value for increasing of the original treshold was set on 0.2 that has been proved as successfull.

Each value above the treshold is detected as a beginning of pulse, therefore another condition is set. All values that are above the treshold are saved. Fig. 5.1 shows many values and their coordinates which are above treshold. In this case is used Matlab function $diff$ to check the biggest changes between coordinates. If a value of treshold is correctly set then the number of biggest changes should be equal to number of pulses. The Fig. 5.1 contains eight biggest changes which are equal to number of pulses shown in Fig. 5.3. This figure shows recorded pulses where the first one is used as a start pulse and other pulses are required for analysis.

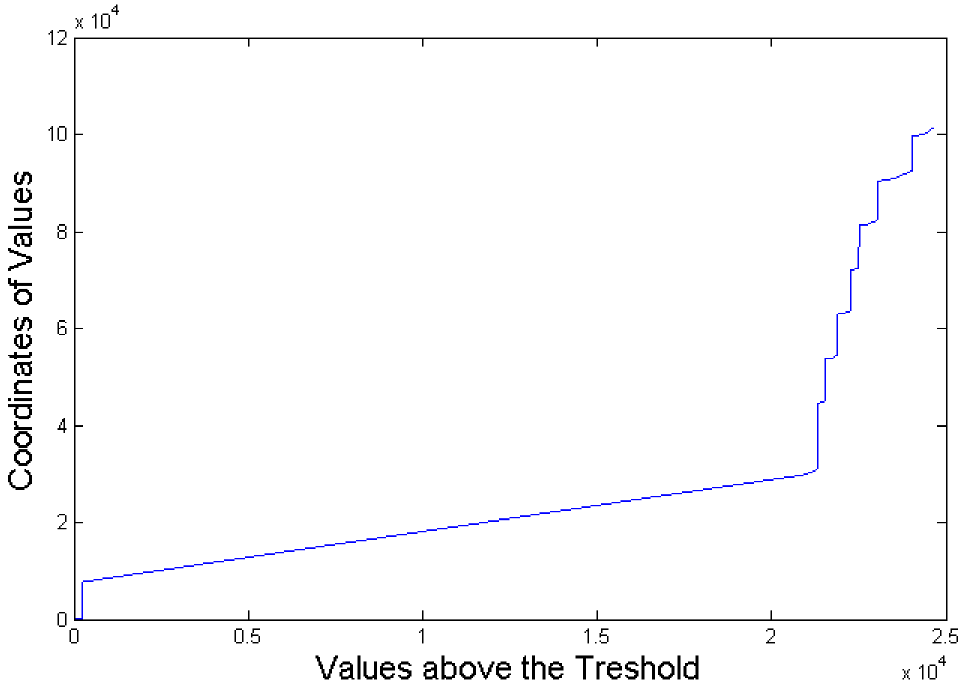

Figure 5.1: Values above the treshold and their coordinates

As output is produced a matrix with recorded coordinates of all pulses which were set at input. This output is further used in another function where the start pulse is eliminated by using the function *get_pulse_data* (see Appendix B).

This function has five input variables and two output variables:

- input variables
  - *signal_data* ... Data for processing.
  - *pulse_detect* ... The coordinates of the individual pulses in the *signal_data*.
  - *w_interval* ... Length of the chosen window with intervals between pulses (in seconds).

  - *w_duration* ... Length of the chosen window with time duration of pulses (in seconds).
  - *fs* ... Sampling frequency of the *signal_data*.

- output variables
  - *data_of_pulses* ... Matrix of the individual windows data. Each column represents one window with pulse data.
  - *ident_StartEnd_pulse* ... The coordinates of the beginning and end of the window. The first line represents start and the second line the end of the window. Each column represents one window.

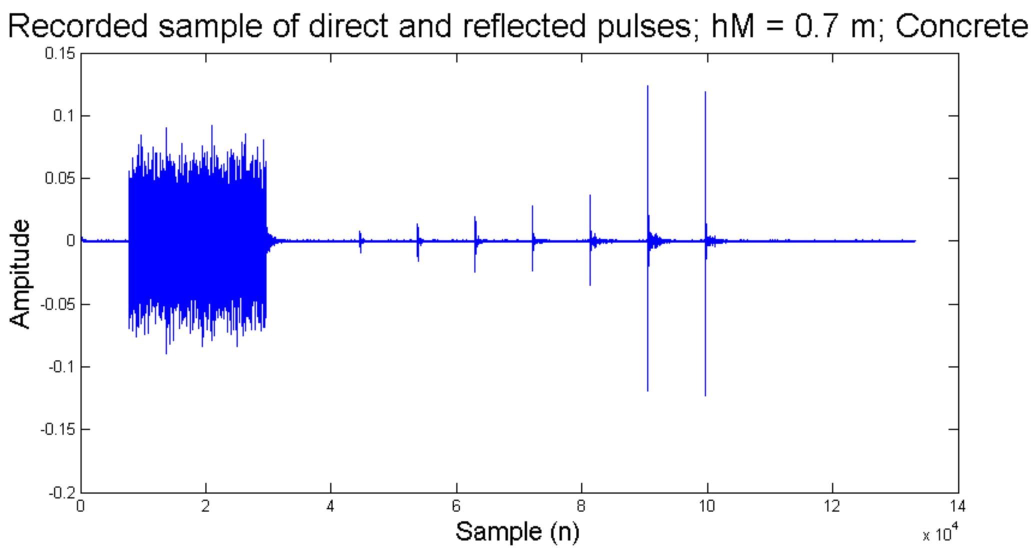

Figure 5.2: Recorded sample of direct and reflected pulses; hM = 0.7 m; Used sample is concrete

The function *get_pulse_data* uses calculated energy to detect a start pulse and eliminates it. Principle of eliminating is, that intervals between pulses is known and the duration of pulses is known as well. As the duration of start pulse is longer than interval between pulses, the energy of start pulse calculated for the same time duration as a time interval between pulses, should be much bigger than energy calculated for other pulses. As shown in Fig. 5.3 the interval between pulses where calculated energy is much lower than energy at start pulse with using the same interval. Energy is computed by this formula:

$$E_X = \int X^2(t)\, dt \tag{5.1}$$

Where $X(t)$ represents the signal data within interval between pulses. This interval also represents limits of the integral.

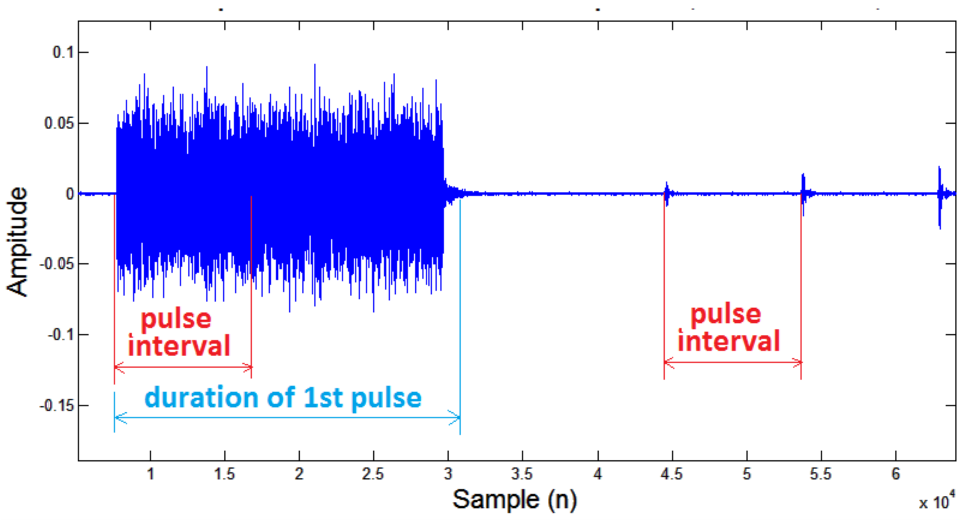

Figure 5.3: Comparing interval between pulses and duration of start pulse

Output of function *get_pulse_data* contains required pulses without start pulse which is not important for next analysis. Output *ident_StartEnd_pulse* with required coordinates of required pulses is further use in function *get_reflected_puslse* (see Appendix B).

This function has six input variables and two output variables:

- input variables
  - *signal_data* ... Data for processing.
  - *pulse_detect* ... The coordinates of the individual pulses in the *signal_data*.
  - *Tpulse* ... Pulse duration (in seconds).
  - *length_ms* ... Distance between microphone and sample (in meters).
  - *sound_velocity* ... Sound velocity (in meters per second).
  - *fs* ... Sampling frequency of the *signal_data*.
- output variables
  - *data_reflect_pulses* ... Matrix of the individual windows data. Each column represents one window with pulse data.
  - *ident_StartEnd_ref_pulse* ... The coordinates of the beginning and end of the window. The first line represents start and the second line is end of the window. Each column represents one window.

This function is used basically for detection of the reflected pulses with variables which are important like beginning of pulse, pulse duration, sound velocity and height of microphone above the sample to detect reflected pulses.

The data points of direct and reflected pulses are obtained by using of all these three functions *pulse_detect*, *get_pulse_data* and *get_reflected_puslse*, where the data of duration pulses are obtained as well. In the Fig. 5.4 are shown detections of beginning and ends of

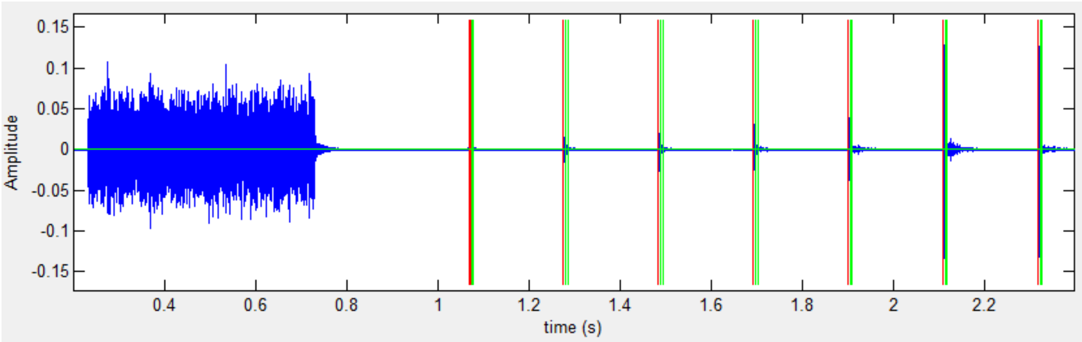

Figure 5.4: Recorded sample with detection of direct and reflected pulses; Red - direct pulses; Green - reflected pulses

direct and reflected pulses. Fig. 5.10 shows more precisely these detection of both pulses.

The detection of pulses using of all three functions described above seem to work pretty nice. All beginnings of all pulses are detected in the range of the acceptability. There are also cases when detection is not so precise as expected. Fig. 5.5 shows example of inaccurate detection of used functions.

The manual GUI detection was implemented because of these inaccurate detection caused by automatic detection (functions described above). With manual detection user can detects direct pulses much better and more precisely. The GUI manual is described in Appendix A and contains both manual and automatic data analysis and also detection of direct pulses.

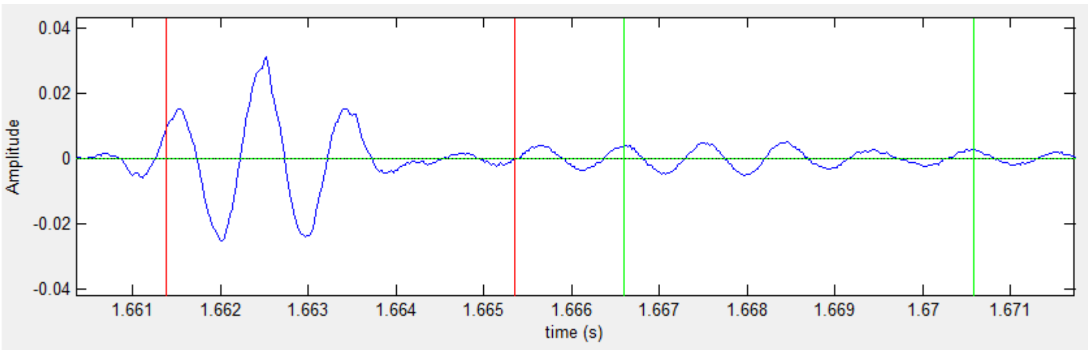

Figure 5.5: Inaccurate detection of direct pulse in recorded sample; Red - direct pulses; Green - reflected pulses; hM = 0.9; Freguency 1000 Hz; Concrete

5.2 Calculation of Coefficients

For calculation of coefficients like reflection or absorption coefficient were used two equations 3.5 and 3.4. As seen these equation are calculated as ratio of two intensities I_1 and I_0. These two intensities could be replaced by two root-means-square of sound pressures p_1 and p_0. Where p_1 is RMS pressure of reflected pulse and p_0 is RMS pressure of direct pulse. Both

RMS pressures are compute as the integral of data which are situated within duration of concrete pulse. It means between the space bounded by red or green sticks.

The Eq.2.17 was used for calculation RMS pressure and below are two new equations 5.2 and 5.3 as the result of modified equations 3.5 and 3.4. The pressure is used due to the fact that values which were recorded are voltage and this voltage is proportional to the sound pressure.

Further from the Eq. 2.12 (or Eq. 2.19) is intensity defined as the product of sound pressure and sound velocity. Because it is expected that sound velocity is in each situation the same, then the sound velocity can be negligible. It means the ration of sound intensities should obtain the similar result as ration of the sound pressure. Therefore the sound pressure is used.

$$R = \kappa \frac{p_1}{p_0} \tag{5.2}$$

$$\alpha = 1 - \kappa \frac{p_1}{p_0} \tag{5.3}$$

The variable κ in equations Eq. 5.2 and 5.3 represents a value which is unknown, meanwhile the thesis was created. Therefore the results of all measurements can give inaccurate values of absorption's and reflection's coefficient.

The κ is depends on many factors like:

- comparing intensities of sounds dependence on the distance in free space and intensities of reflected sounds in the same distance
- dependence of directionality on used microphone
- dependence on spreading of spherical and direct sound waves of loudspeaker

All factors above could have a great effect on the resulting value κ. Of course another factors, which were not mentioned, could be and have an effect too.

5.2.1 Used Window within the Pulse

The violet sticks added in Fig. 5.6 represent the interval of data which are used to compute the RMS presures of both pulses. The interval between violet sticks represent 60% data of the interval of direct or reflected pulses. It means that only 60% of data from the center of pulses are used to compute, and where 20% data from the beginning and 20% data from the end are eliminated.

Using a window within the pulse should eliminate parasitic reflected sound wave which mainly appears at the ends of both pulses and sometimes also at the beginning of these pulses even if conditions in section 4.2.1 are complied. Because there is another issue during

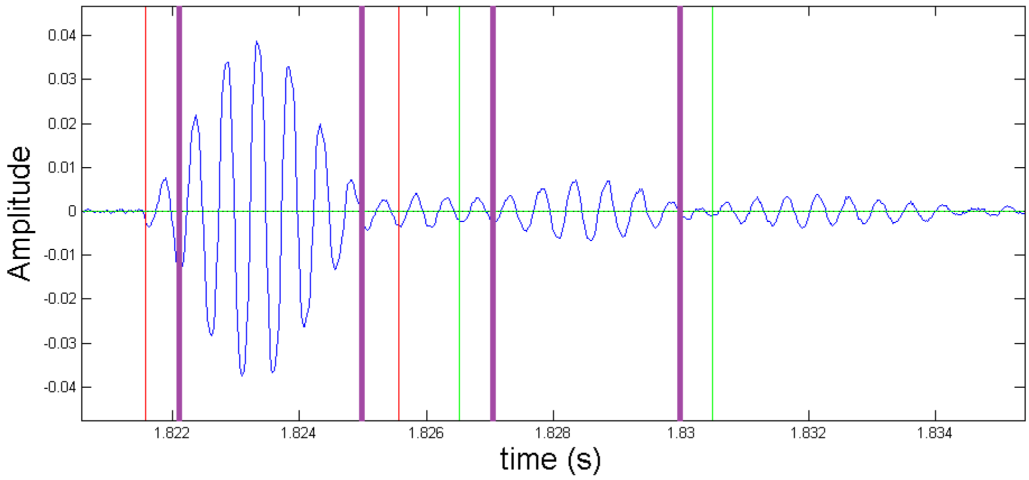

Figure 5.6: Recorded sample; Red - direct pulses; Green - reflected pulses; Violet - interval which represent 60% data of pulse; Frequency = 2000 Hz; hM = 0.85 m; Concrete

measurement and it is the fact that user must stand nearby to perform the recording from the measurement. This issue is eliminated with properties of human body, where the body is expected to be very absorption, therefore the reflected parasitic waves should not be created.

The size of used window can be changed with GUI and the concrete data can be selected by using manual mode of GUI but the length of data within reflected pulse should be the same as the length of data within direct pulse.

5.2.2 Number of Periods

Another task after using percentage window within pulse is determining number of periods which are presented within percentage window. This condition is important for define uncertainty, which are described in literature by Martin Novotny and Milos Sedlacek [11]. There are some parameters of the signal processing affect a lot the result.

The function *num_of_periode* (see Appendix B) was created for this case and has two input variables and two output variables:

- input variables
  - *dataDir* ... Directed waves data.
  - *dataRef* ... Reflected waves data.
- output variables
  - *dataDirOut* ... Contains new window's data with the same number of periode as dataRefOut.
  - *dataRefOut* ... Contains new window's data with the same number of periode as dataDirOut.

This function checks number of zero crossings where two zero crossings are equal to half periode, and three zero crossings are equal to one periode and so on. If there is only one or two zero crossings, then this function uses original input of both windows, otherwise it uses a window corresponding to the number of zero crossings.

5.3 Uncertainties

This thesis contains uncertainties. The uncertainties are caused due to inaccuracy of measurement with used devices, correct and precise designation of distances by using a tape measure, properties of used loudspeaker and microphone, tools used and situated in acoustic isolation room, used phone platform, noise during measurement, where all these items have bad effect on recording of sound recordings. Further are uncertainties caused by the data analysis in the form of imprecise detection beginning of direct pulses in each measurement which is caused by noise, convert analog data to digital data and then analysis of recorded samples, using root-means-square for obtaining RMS value. Some uncertainties can be reduced and some can not, but they are known.

Uncertainties which can be reduced, for example recorded sound samples with many measurements can reduce random errors like noise, therefore each distance has ten repetitions. Where ten measurement at the same distance is the minimum for designation standard uncertainty for a type A (See Eq. 5.4).

$$\mu_A = \sqrt{\frac{\sum_{i=1}^{N}(x_i - \bar{x})^2}{N(N-1)}} \tag{5.4}$$

Where:

μ_A ...is standard uncertainty for a type A

N ...is number of measurements in the set

x_i ...is the result of the i-th measurement

$\bar{x}$ ... is the arithmetic mean of the N results considered

Another uncertainties as mentioned above are caused by processing of data, which were described by Martin Novotny and Milos Sedlacek [11]. This article is focused on measurement of RMS values of non-coherently sampled signal. This thesis computes RMS sound pressure value like equation within article. In order to obtain accurate RMS value, the duration of the signal taken into account is really important and is related to the period of the signal. In the fact, most accurate results are obtained when the signal is computed during integer or integer with half number of periods. That is why the Matlab function *num_of_periode* is searching for number of periodes or number of periodes with half of period.

5.4 GUI for Analysis

For data analysis was created GUI for easier obtaining of absorption and reflection coefficients. The Fig. 5.7 presents first launch of GUI. More information regarding GUI is described in Appendix B.

Figure 5.7: First launch of GUI

Fig. 5.8 shows GUI after opening audio file and then click on the button *START/RESET* for auto analysis of data. The new Fig. 5.9 will appear with results coefficients of absorption and reflection. In Fig. 5.8 is shown item 19 with number of required or useful pulses and item 17 shows graph with detection of the beginnings direct pulses.

Before starting of auto analysis is required to fill all these items like 1, 2, 3, 5, and 7 shown in Fig. 5.7. In Fig. 5.8 all required items are filled and other items are filled as well, where:

1. item is 0.004 seccond
2. item is 0.2 second
3. item is 0.7 meters
4. item is 0.06 meters
5. item is 60 percent
6. item is default temperature 20 Celsius
7. item is 8 pulses

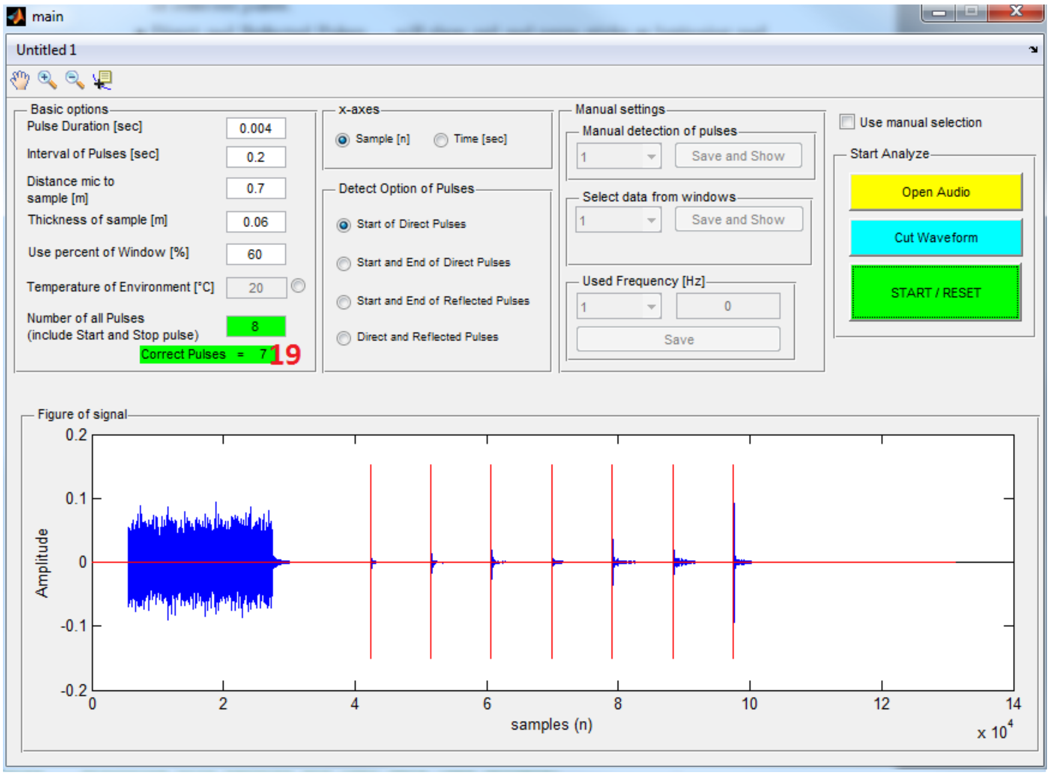

Figure 5.8: GUI after opening audio file and starting auto detection

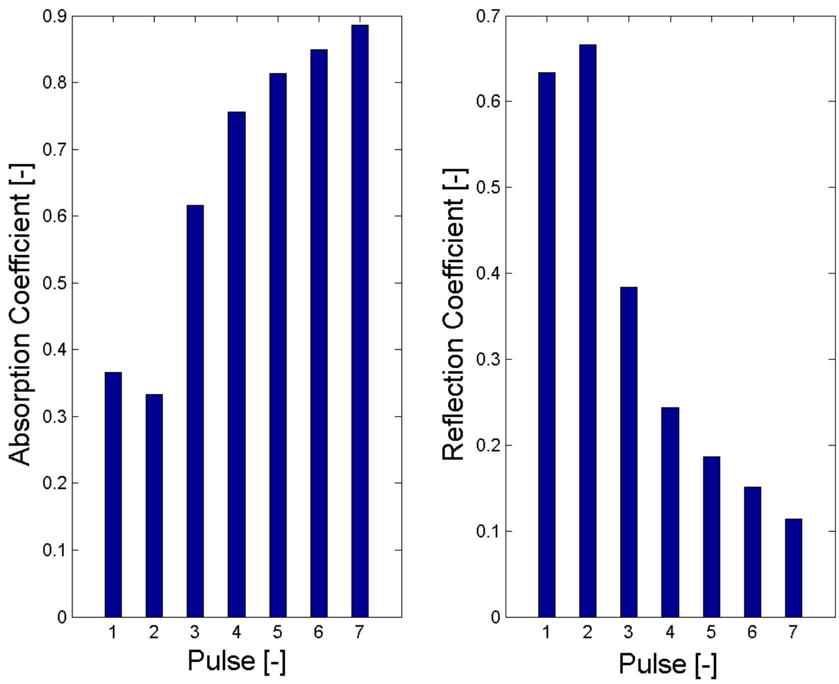

Figure 5.9: Computed coefficients of absorption and reflection; hM = 0.7 m; Concrete

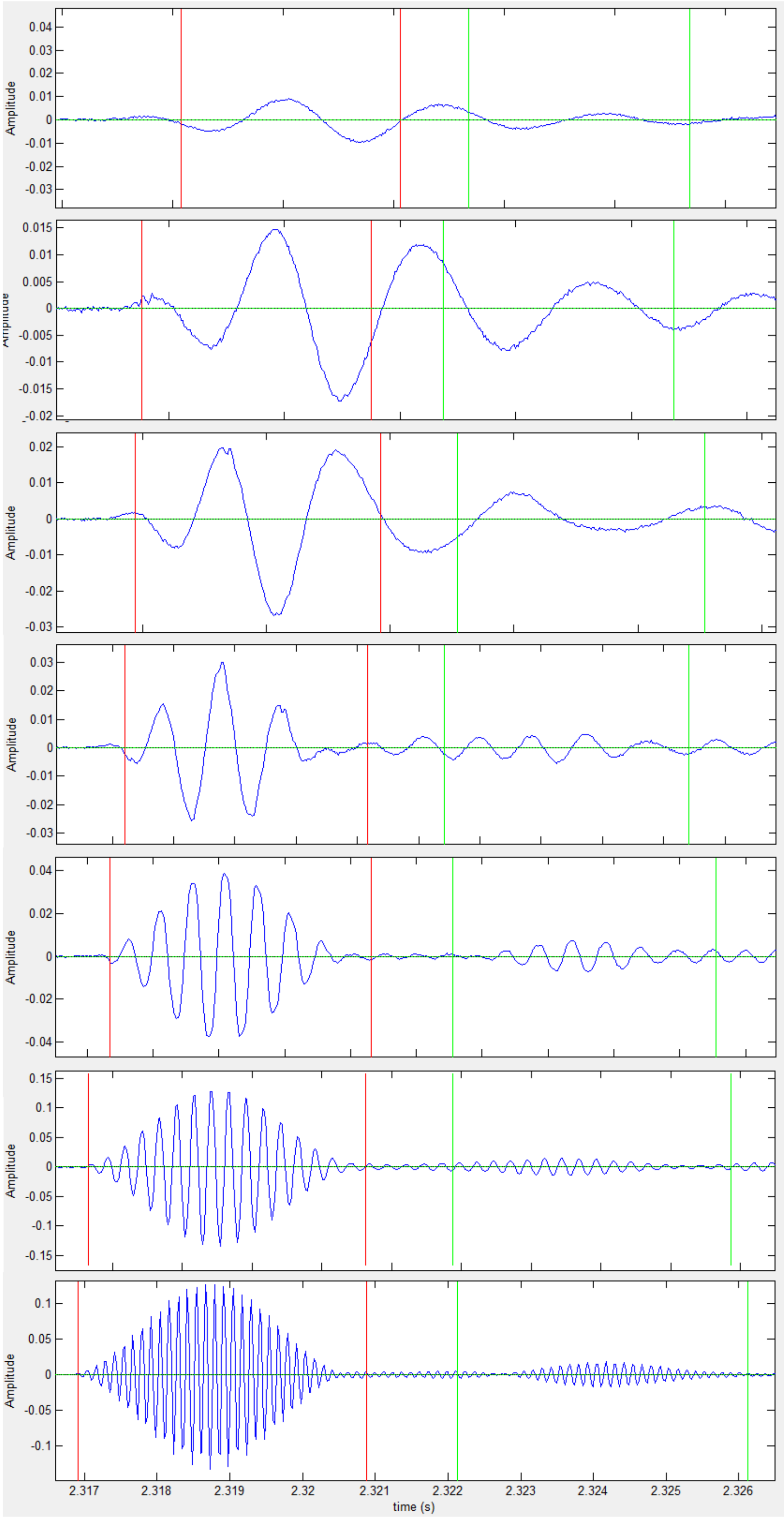

Figure 5.10: Recorded sample of invidual pulses; Red - direct pulses; Green - reflected pulses; From the to to bottom are used these frequency: 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz, 8000 Hz; hM = 0.9 m; Concrete

Chapter 6

Results

This chapter represents results of data analysis which basically was provided through GUI created in MATLAB. It should be note that the results of coefficients are affected by unknown value of κ in equations 5.2 and 5.3. This thesis did not address this unknown value which should be provided and incorporated into this thesis, but it has not happened. Therefore the results below can not be compared with valid data but only as orientational data.

Fig. 6.1 shows all ten measurement with height of microphone $h_M = 0.7$ m of concrete, where green square is average value of all these ten measurement of specific frequency.

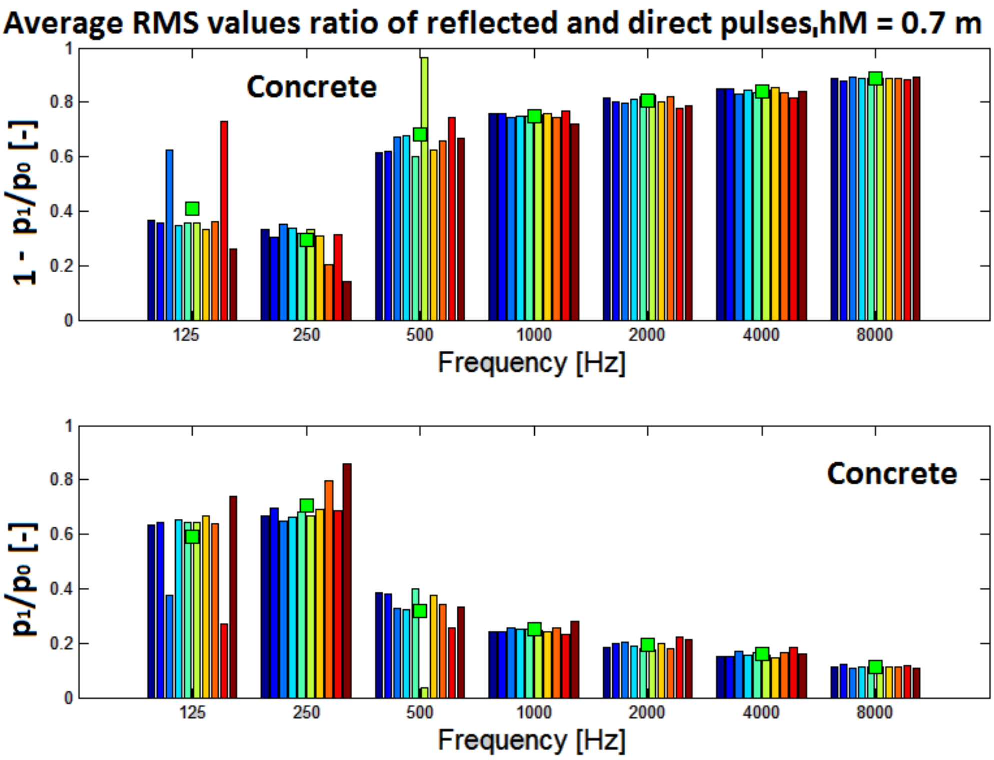

Figure 6.1: Graph of all measurement wit hM = 0.7 m, Concrete; Green square - Average of all measurement of specific frequency

As can be noticed, the low frequencies 125 Hz, 250 Hz and 500 Hz are very unstable because the sample's dimension of concrete is only 50 x 50 cm and these low frequencies need for better results bigger dimensions because these frequencies has large variance. Therefore these low frequencies will be eliminated. Fig. 6.2 shows more detailed further values of used higher frequencies which have lower variance and the dimension of concrete are acceptable for measurement.

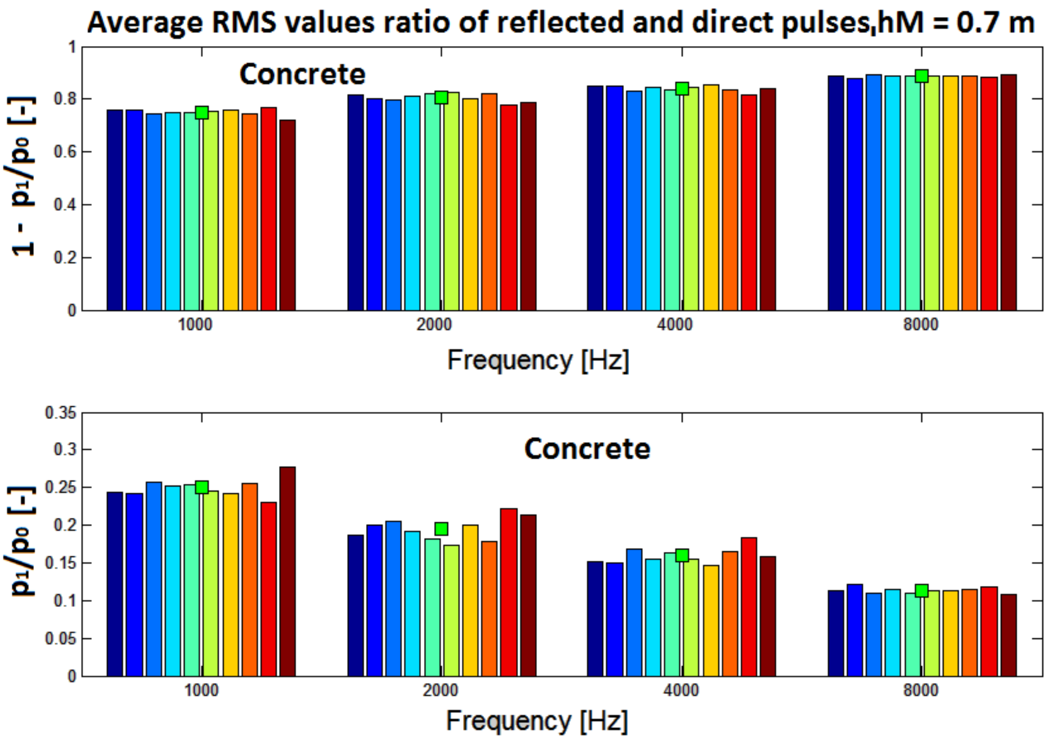

Figure 6.2: Graph of all measurement wit hM = 0.7 m, Concrete; Green square - Average of all measurement of specific frequency

Fig. 6.3 shows relevant average values of pressure ratio which are dependent on height of microphone. At frequencies like 4000 Hz and 8000 Hz can find some dependence on distance, but it must not be forgotten that these values are computed without unknow value as described above.

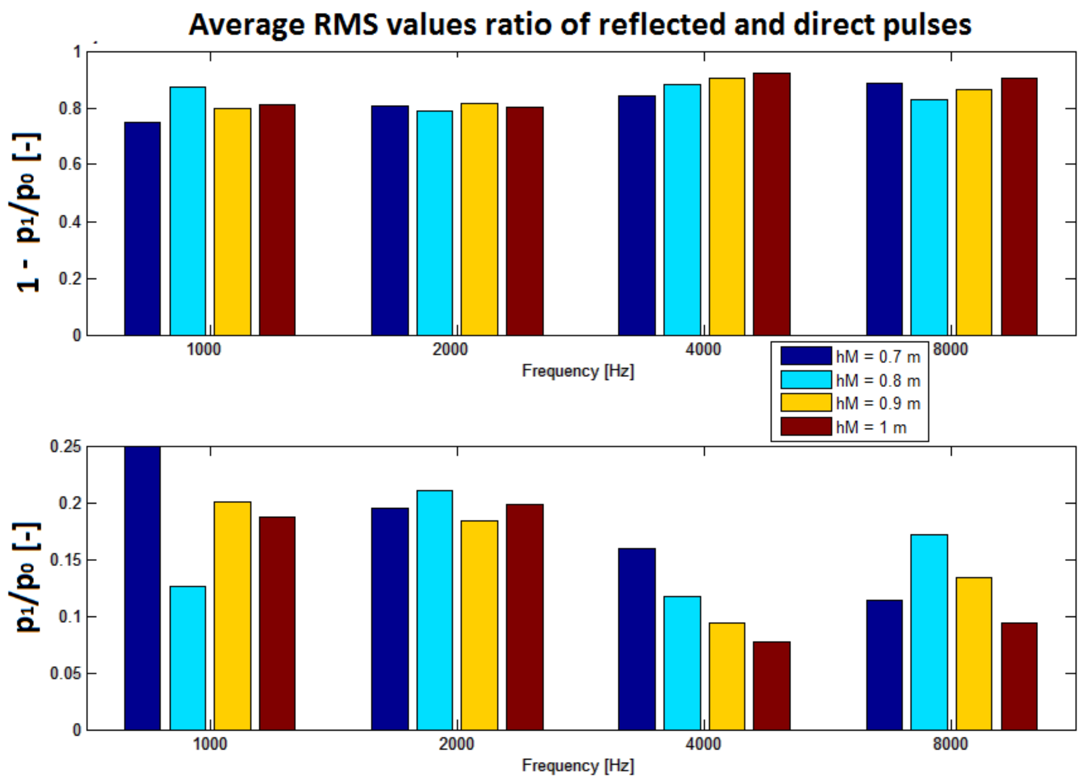

Figure 6.3: Graph shows average values of ratio depence on height of microphone (hM), Concrete

Chapter 7

Conclusion

The task of this diploma thesis was analyze the method for acoustic absorption and transmission measurement, further design and implementation of these method in the MATLAB for estimate transmission based on reflected signal.

The important fact in this thesis was introduction with the properties of sound, behavior of sound in different environment and its parameters. The next assumptions was knowledge of used materials so it is expected that the concrete will be more reflective than foam and foam will be more absorptive than concrete, is expected. With this information and knowledge is predicted, that sound intensities of reflected sound waves should have lower intensities than intensity of direct sound waves.

During measurement of used samples (concrete, polystyrene and rubber boards) , the devices were set into suitable positions to comply of conditions described in section 4.2.1 even if the measurement was performed inside acoustic isolated room. The measurement can never be accurate because there are other factors which can raise the error of measurement then it is important to know about them like properties of microphone and loudspeaker especially if are used usual average device as smartphone.

For analyzing data and creating new functions was very important to precisely detect beginnings of directed pulses during whole measurement due to additional information. The beginnings are important as well to detection of reflected pulses in the recorded signal sample. Additional information represent pulse duration, time iterval between duration, number of pulses and temperature of environment. Temperature is used for estimate of sound velocity (see Eq. 2.3).

As the result of the detection of both pulses are data windows with data of pulses which are used for computing RMS sound pressure and then intensity of reflected and direct pulses. RMS pressure can be used instead of intensity (described in section 5.2) then ratio of reflected and direct pressures can be computed. This ratio is define as reflective coefficient (see Eq. 5.2) and one minus reflective coefficient gives absorptive coefficient (see Eq. 5.3) where all measurement and devices are ideal. Then the value of κ is not used in equations. The issue is that all measured device have their error properties then all these errors are hide in the value κ. Therefore results of coefficient can not be called as reflection and absorption but only ratio of the pressures (or intensities).

This thesis should not deal with this κ value, which should be supplied for finishing of created methods and comparing of results. This value has many factors which affects correct results (see section 5.2). This study attempted to investigate all possibilities and combinations to obtain this unknow value but unsuccessfully.

Thesis can be improved with using better devices like loudspeaker and microphone used in really acoustic isolated room or into empty room with acoustic isolated walls and ceiling. Mainly properties of microphone and loudspeaker should be known better and get laboratory measured data which can be used better to define as unknown value as better detection of pulses beginning. The results then can be compared with another measurement with normal or usual devices and find out differencies and maybe estimate differential constants for specific devices.

Bibliography

[1] S. ACOUSTICS. Speed of sound - temperature matters, not air pressure. http://clas.mq.edu.au/speech/acoustics/basic_acoustics/basic_acoustics.html#t01, Accessed on April 28, 2014.

[2] F. Alton Everest and K. C. Pohlman. Master handbook of acoustic. Fifth Edition, 2009.

[3] S. Eberhard. Calculation of the speed of sound in air and the effective temperature. http://www.sengpielaudio.com/calculator-speedsound.htm, Accessed on April 28, 2014.

[4] S. Eberhard. Speed of sound - temperature matters, not air pressure. http://www.sengpielaudio.com/SpeedOfSoundPressure.pdf, Accessed on April 28, 2014.

[5] C. EMIFIL. Data sheet - inductor type blm15a-sn series 0402 size. BLM15AG121SN1D.

[6] G. N. Gibson. Complex tones, fourier analysis and the missing fundamental. http://www.phys.uconn.edu/~gibson/Notes/Section7_4/Sec7_4.htm, Accessed on April 30, 2014.

[7] T. Henderson. Sound waves and music. http://www.physicsclassroom.com/class/sound, Accessed on September 2, 2014.

[8] J. M. Hillenbrand. The physics of sound. http://homepages.wmich.edu/~hillenbr/206/ac.pdf, Accessed on April 27, 2014.

[9] HyperPhysics. Sound propagation. http://hyperphysics.phy-astr.gsu.edu/hbase/sound/tralon.html, Accessed on April 28, 2014.

[10] H. Kuttruff. Room acoustic, fifth edition. Institut fur Technische Akustik, Technische Hochschule Aachen, Aachen, Germany. Fifth edition published 2009 by Spon Press.

[11] M. Novotny and M. Sedlacek. Measurement of rms values of non-coherently sampled signals. Measurement, Volume 41, issue 3, April, 2008, p. 236-250.

[12] U. of Salford. Wave type. http://www.acoustics.salford.ac.uk/feschools/waves/wavetypes.php#introd, Accessed on April 28, 2014.

[13] J. Reichl and M. Vseticka. Mechanicke kmitani a vlneni. http://fyzika.jreichl.com/main.article/view/150-mechanicke-kmitani-a-vlneni, Accessed on April 27, 2014.

[14] D. Russell. Longitudinal and transverse wave motion.
`http://www.acs.psu.edu/drussell/demos/waves/wavemotion.html`,
Accessed on April 28, 2014.

[15] E. STANDARD. Road traffic noise reducing devices - test method for determining the acoustic performance - part 4: Intrinsic characteristics - in situ values of sound difraction. CNET/TS 1793-4, December 2003.

[16] E. STANDARD. Road traffic noise reducing devices - test method for determining the acoustic performance - part 5: Intrinsic characteristics - in situ values of sound reflection and airborner sound insulation. CNET/TS 1793-5, March 2003.

[17] M. Vasina. Evironmentalni fyzika, univerzita tomase bati - ustav fyziky a materialoveho inzenyrstvi.
`http://ufmi.ft.utb.cz/texty/env_fyzika/EF_04.pdf`,
Accessed on May 2, 2014.

Appendix A

Manual of GUI

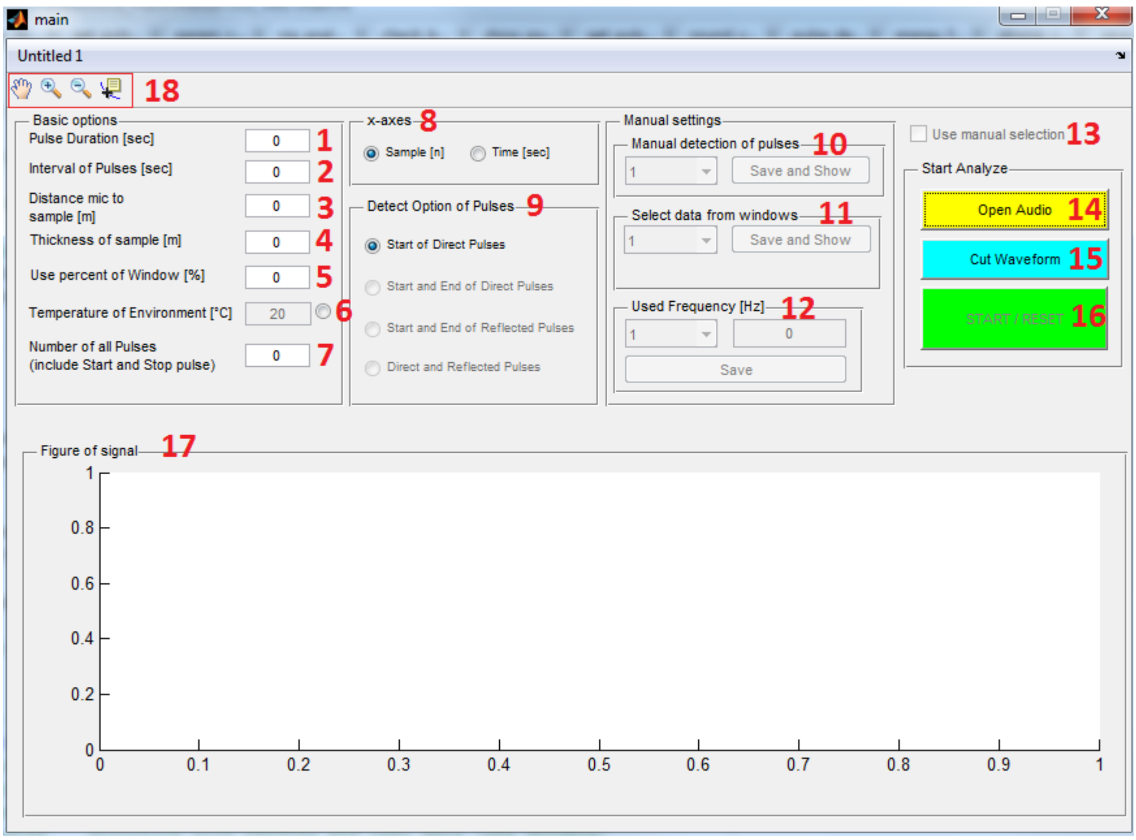

Figure A.1: First launch of GUI

Description of Fig.5.7:

1. Pulse Duration ... fill time duration of one required pulse in seconds.
2. Interval of Pulses ... fill interval between required pulses in seconds.
3. Distance mic to sample ... fill used distance between microphone and sample in meters.
4. Thickness of sample ... fill thickness of used sample in meters.

5. Use percent of Window ... fill percent of used window within interval of pulse for processing in percent.
6. Temperature of Environment ... fill temperature of environment in celsius; Default value is 20C; Check radio button for editing.
7. Number of all Pulses ... fill number of all pulses include start pulse.
8. x-axes ... select one radio button.
   - Sample[n] ... will show x-axis as sample-axis in graph 17 if radio button is selected.
   - Time[sec] ... will show x-axis as time-axis in graph 17 if radio button is selected.
9. Detect Option of Pulses ... select one radio button .
   - Start of Direct Pulses ... will show red sticks as beginning of direct pulses.
   - Start and End of Direct Pulses ... will show red sticks as beginning and ends of direct pulses.
   - Start and End of Reflected Pulses ... will show green sticks as beginning and ends of reflected pulses.
   - Direct and Reflected Pulses ... will show red and green sticks as beginning and ends of direct and reflected pulses.
10. Manual detection of pulses ... select concrete pulse and manually detect beginning of direct pulse in graph 17 and save coordinates; Coefficients are saved in the folder $ManualDetection$.
11. Select data from windows ... will show two graphs in 17: first with direct pulse and second with reflected pulse; select concrete pulse and manually detect window with data within direct and reflected pulses; Coefficietns are saved in th folder $WindowDetection$.
12. Used Frequencz [Hz] ... select concrete pulse and fill known frequency used in pulse and save.
13. Used manual selection ... select checkbox for availability of Manual settings box; 10, 11 and 12 will be available.
14. Open Audio ... press button to select audio file (MP3, WAV).
15. Cut Waveform ... select stretch of data for eliminating in graph 17.
16. START/RESET ... button for start auto analysis and reset data if manual mode was used; New graph will appeared with coefficients of absorption and reflection; Coefficients are saved in the folder *AutoDetection*.
17. Figure of signal ... shows main graph of signal and two graphs of direct and reflected pulses if 11 is used.
18. Tools for working with figures - shift, zoom in, zoom out, get data point
19. Correct Pulses - show the number of required pulses
20. Get Direct Points - get interval of data points - user must select two data points
21. Get Reflect Points - get interval of data points - user might or might not select one data point

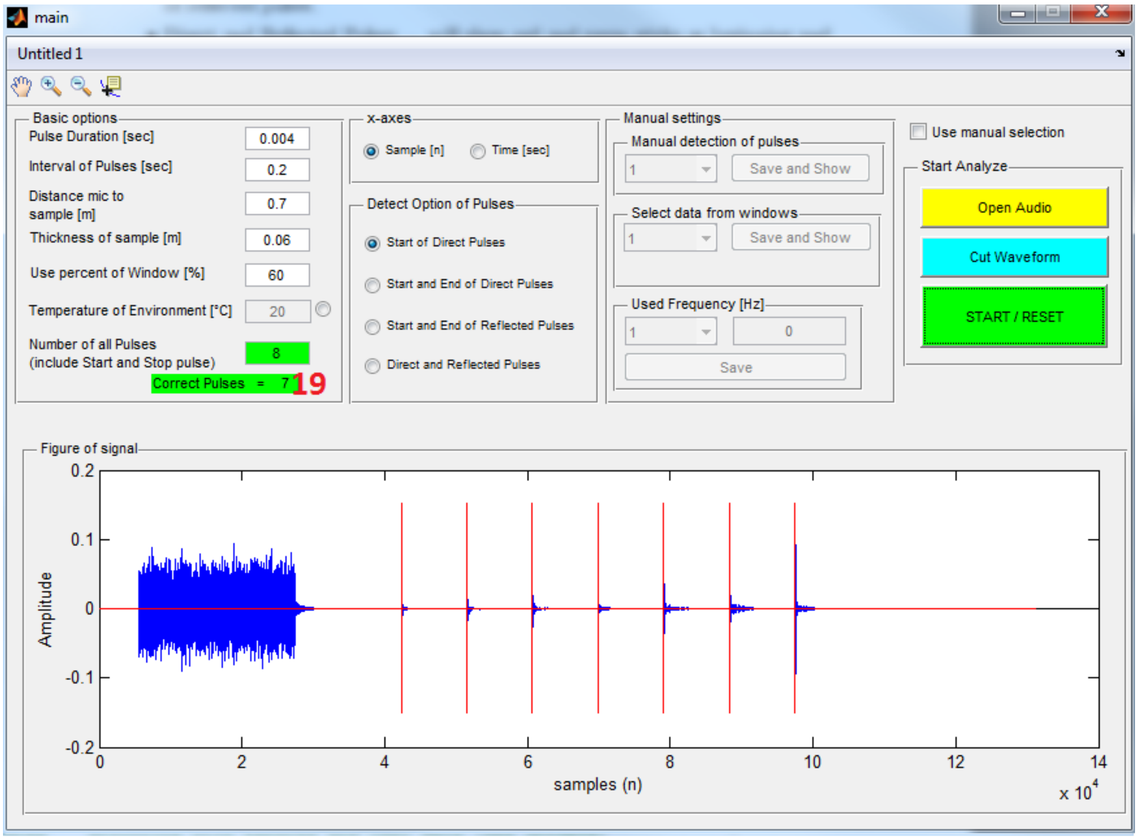

Figure A.2: After the auto detection, the user can select another options in the field 9 - *DetectOptionofPulses* and then click again on *START/RESET* button to show selected option.

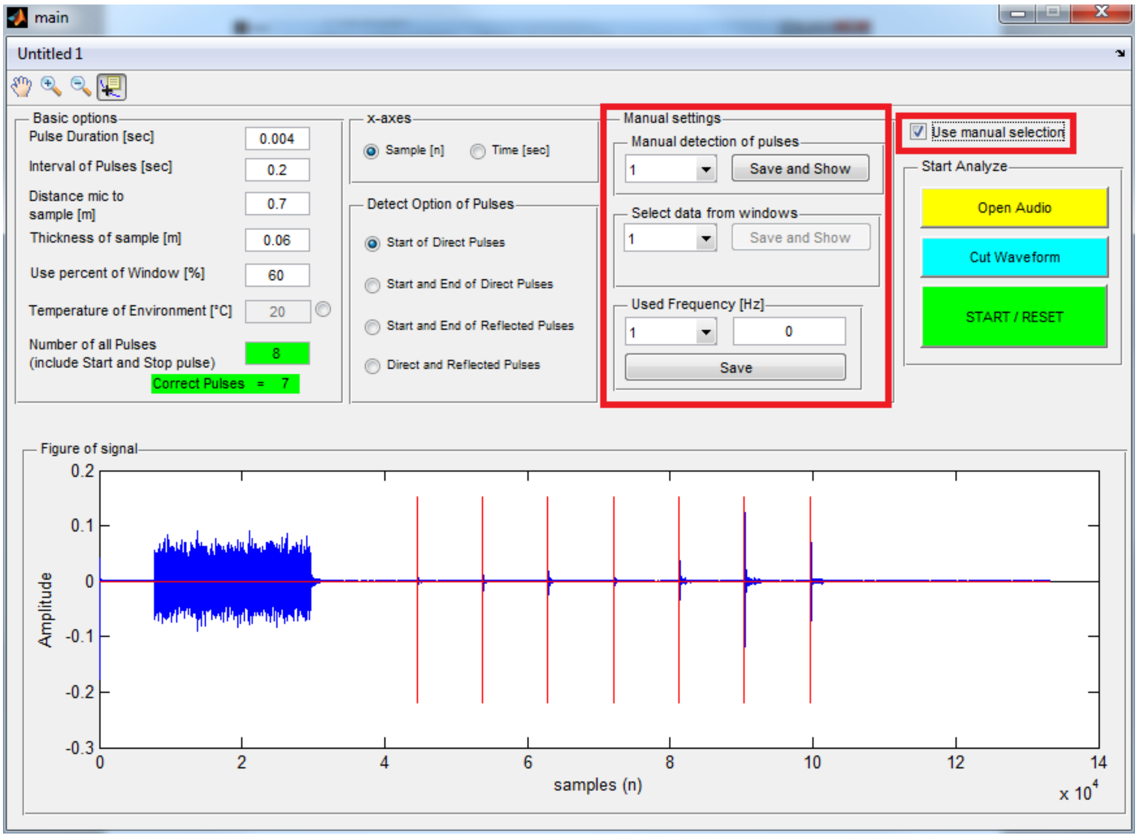

Figure A.3: User can also select checkbox 13 - $Usermanualselection$. If user select this checkbox then fields 10 - $Manualdetectionofpulses$, 11 - $Selectdatafromwindows$ and 12 - $UsedFrequency[Hz]$ will be available.

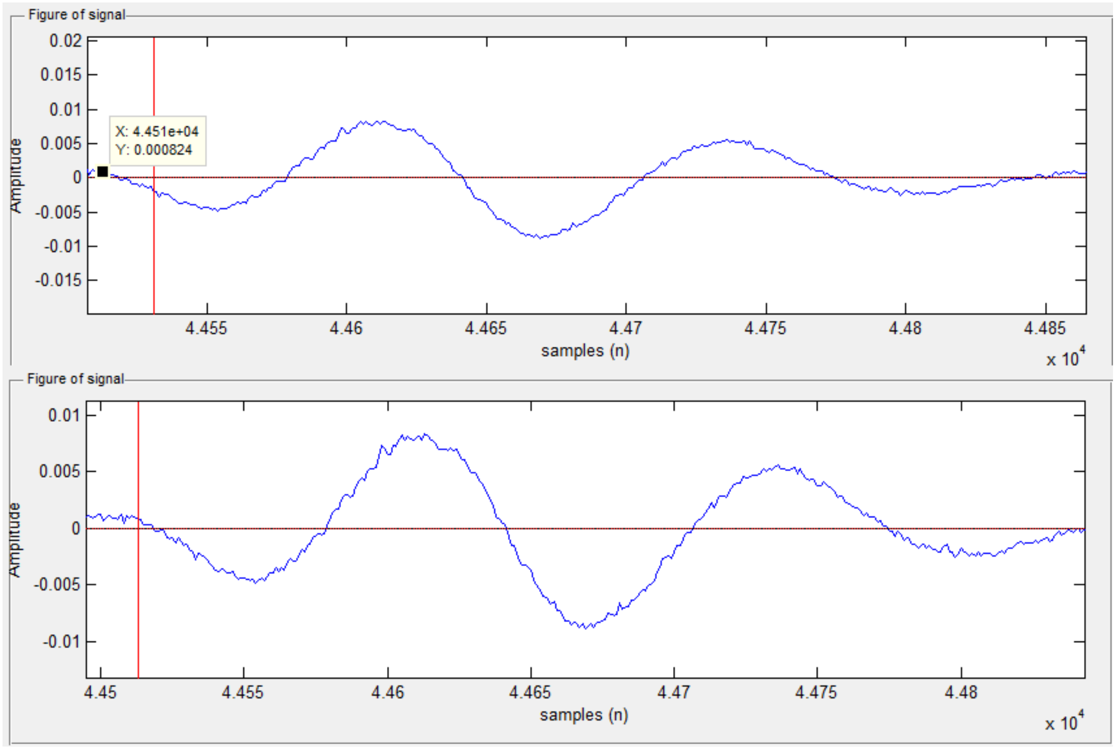

Figure A.4: Upper graph shows manual detection of the first pulse with its beginning, where is used data point to select the new value of pulse beginning. Lower is the result of the previous detection after clicking on *SaveandShow* button in the field 10 to save new data of pulse and show figure with coefficients.

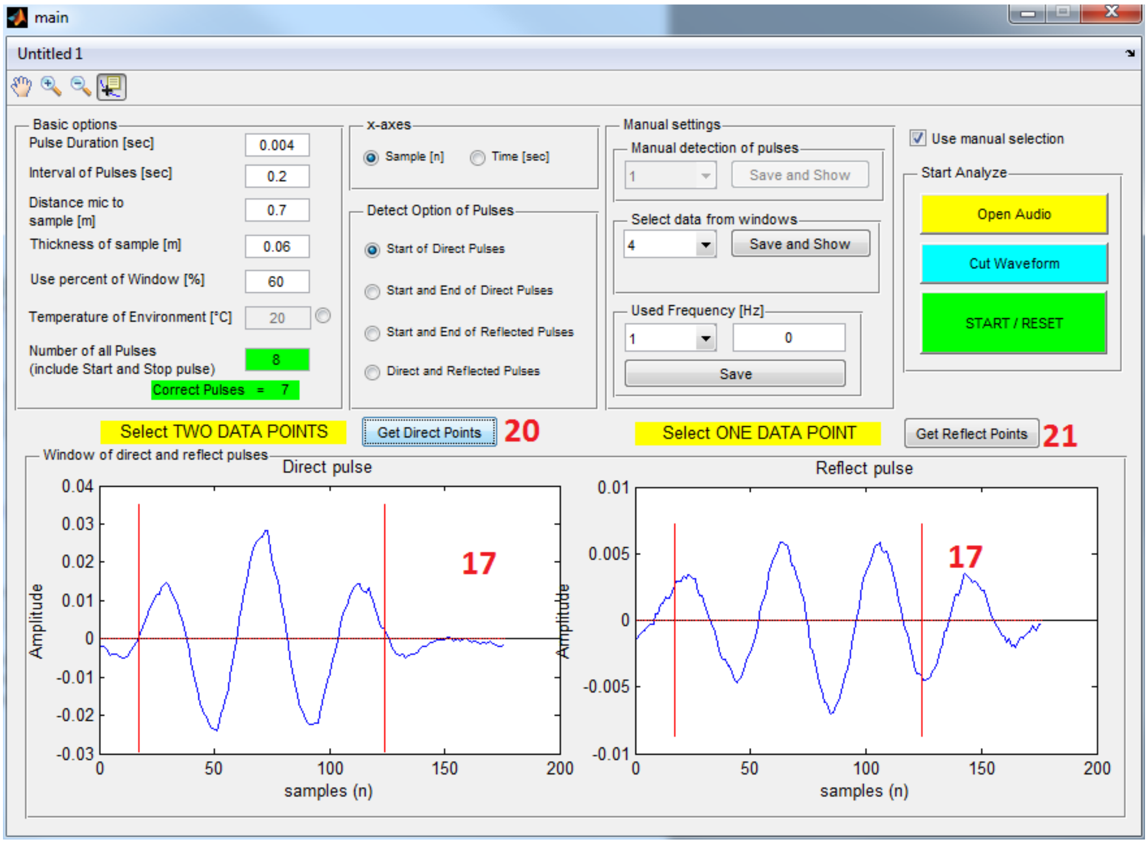

Figure A.5: If a user clicks on and select the pulse in the field 11 then two graphs will be appeared in graph field 17. The left graph shows direct data and the right shows reflected data of pulses. In the left graph, a user must select the interval of direct data then clicks on the button 20. The button 21 will be available after clicking on the button 20. In the right graph, a user might or might not select one data point. Then user clicks on the button *SaveandShow* in the field 11 to save new data of both graph and show figure with coefficients.

Appendix B

Created Matlab Functions

```
function [data_of_pulses, ident_StartEnd_pulse] = ...
    get_pulse_data(signal_data, pulse_detect, w_interval, w_duration, fs)
%GET_PULSE_DATA
%   Gets the length of the individual pulses. Start pulse with long
%   response will be removed.
%
%   signal_data ... Data for processing.
%
%   pulse_detect ... The coordinates of the individual pulses in
%   the signal_data.
%
%   w_interval ... Length of the chosen window with intervals between ...
    pulses (in seconds).
%
%   w_duration ... Length of the chosen window with time duration of ...
    pulses (in seconds).
%
%   fs ... Sampling frequency of the signal_data.
%
%   data_of_pulses ... Matrix of the individual windows data. Each column
%   represents one window with pulse data.
%
%   ident_StartEnd_pulse ... The coordinates of the beginning and end
%   of the window. The first line represents start and the second line is
%   end of the window. Each column represents one window.

y = abs(signal_data);

% Conversion the time to samples
window_time = w_interval;
window_pulse = w_duration;
window_length_time = round( window_time * fs);
window_length_pulse = round( window_pulse * fs);

% Pulses with start/stop pulse in the signal
Energy = zeros(length(pulse_detect),1);
data_pulses = zeros(window_length_time,length(pulse_detect));

for i = 1 : length(pulse_detect)
    % Windows of invidual intervals
    data_pulses(:,i) = y(pulse_detect(i) : (pulse_detect(i) + ...
        window_length_time)-1);
    % Calculating the energy of each window
```

```
    Energy(i,1) = sum(abs(data_pulses(:,i)).^2);
end

% Eliminate if start pulse starts in zero
if Energy(1,1) < min(Energy(Energy>min(Energy)))
    Energy(1,1) = max(Energy * 10);
end

% Chose only start/stop pulse of the signal.
SS_pulse_check = find( Energy(:) > (max(Energy) - ...
    abs(sum(Energy)/length(Energy))) );

% Pulses without start/stop pulse in the signal
% Remove start/stop pulses in de signal
new_pulse_detect = pulse_detect;
new_pulse_detect(SS_pulse_check) = [];
data_of_pulses = zeros(window_length_pulse,length(new_pulse_detect));
ident_StartEnd_pulse = zeros(length(new_pulse_detect),2);

% The start of the pulses
start_pulses = new_pulse_detect ;
% The end of the pulses
end_pulses = start_pulses + window_length_pulse - 1;

% Cordination of the start and end of the reflected pulses
ident_StartEnd_pulse(:,1) = start_pulses;
ident_StartEnd_pulse(:,2) = end_pulses;

for j = 1 : length(new_pulse_detect)
    % Windows with time duration of the pulses
    data_of_pulses(:,j) = signal_data(new_pulse_detect(j) : ...
        (new_pulse_detect(j) + window_length_pulse)-1);
    % Start of the window
    ident_StartEnd_pulse(j,1) = new_pulse_detect(j);
    % End of the window
    ident_StartEnd_pulse(j,2) = (new_pulse_detect(j) + ...
        window_length_pulse) - 1;
end

end
```

```
function [data_reflect_pulses, ident_StartEnd_ref_pulse] = ...
    get_reflected_puslse(signal_data, pulse_detect, Tpulse, length_ms, ...
    sound_velocity, fs)
%GET_REFLECTED_PUSLSE
%   Gets the length of the reflected pulses.
%
%   signal_data ... Data for processing.
%
%   pulse_detect ... The coordinates of the individual pulses in
%   the signal_data.
%
%   Tpulse ... Pulse duration (in seconds).
%
%   length_ms ... Distance between microphone and sample (in meters).
%
%   sound_velocity ... Sound velocity (in meters per second).
%
%   fs ... Sampling frequency of the signal_data.
%
%   data_reflect_pulses ... Matrix of the individual windows. Each column
%   represents one window with reflected pulse.
%
%   ident_StartEnd_ref_pulse ... The coordinates of the beginning and end
%   of the window. The first line represents start and the second line is
%   end of the window. Each column represents one window.

y = signal_data;
sv = sound_velocity;

% Distance of direct pulse
l1 = Tpulse * sv;
% Distance from the microphone to sample and back
l2 = 2 * length_ms;

% Check the condition
if (l1 / 2) > (l2 / 2)
    error('The distance between the microphone and the sample must be ...
        greater: Tpulse*sound_velocity ≤ length_ms');
end

% The length of the directed and the reflected pulses
w_reflect = round(Tpulse * fs);
w_direct = round(Tpulse * fs);

% The length of the delay of the reflected pulses
dl = l2 - l1;
t_delay = dl / sv;
w_delay = round(t_delay * fs);

% The start of the reflected pulses
start_ref_pulses = pulse_detect + w_direct + w_delay - 1;
% The end of the reflected pulses
end_ref_pulses = start_ref_pulses + w_reflect;

% Cordination of the start and end of the reflected pulses
ident_StartEnd_ref_pulse(:,1) = start_ref_pulses;
ident_StartEnd_ref_pulse(:,2) = end_ref_pulses;

data_reflect_pulses = zeros(w_reflect,length(pulse_detect));
for j = 1 : length(pulse_detect)
```

```
    % Windows with data of the pulses
    % Window length is equal to the pulse duration
    data_reflect_pulses(:,j) = y(start_ref_pulses(j) : ...
        (start_ref_pulses(j) + w_reflect)-1);
end

end
```

```
function check_buttons(data_audio, pulse_det, tag1, tag2, fs)

switch tag1
    case 'start_direct'
        switch  tag2
            case 'radio_sample'
            show_pulse_detect(data_audio, pulse_det, 0);
            case 'radio_time'
            show_pulse_detect(data_audio, pulse_det, 0, fs);
        end
     case 'start_end_direct'
        switch  tag2
            case 'radio_sample'
            show_pulse_detect(data_audio, pulse_det, 1);
            case 'radio_time'
            show_pulse_detect(data_audio, pulse_det, 1, fs);
        end
     case 'start_end_ref'
        switch  tag2
            case 'radio_sample'
            show_pulse_detect(data_audio, pulse_det, 2);
            case 'radio_time'
            show_pulse_detect(data_audio, pulse_det, 2, fs);
        end
     case 'direct_ref'
        switch  tag2
            case 'radio_sample'
            show_pulse_detect(data_audio, pulse_det, 3);
            case 'radio_time'
            show_pulse_detect(data_audio, pulse_det, 3, fs);
        end
end
end
```

```
function [dataDirOut, dataRefOut] = num_of_periode(dataDir, dataRef)
%NUM_OF_PERIODE
%   This function find out number of periode in each data windows by using
%   zero crossing and set the min number of periode from both data windows.
%
%   dataDir ... Directed waves data.
%   dataRef ... Reflected waves data.
%
%   OUTPUTS:
%   dataDirOut ... Contains new window's data with the same number of ... periode as dataRefOut.
%   dataRefOut ... Contains new window's data with the same number of ... periode as dataDirOut.

%Find Zero Cross
% Create cell array with zero crossing index
zeroIndexDir{1,length(dataDir(1,:))} = 0;
zeroIndexRef{1,length(dataRef(1,:))} = 0;

% Number of zero crosing
checkZerosDir = zeros(1,length(dataDir(1,:)));
checkZerosRef = zeros(1,length(dataRef(1,:)));

% Min zero crossing of directed and reflected pulses
minZeroNum = zeros(1,length(dataDir(1,:)));

% Fill index of zero crossing and number of them
for kk = 1:length(dataDir(1,:))
    % Find zero crossing and get index
    zeroIndexDir{1,kk} = find(diff(sign(dataDir(:,kk))));
    zeroIndexRef{1,kk} = find(diff(sign(dataRef(:,kk))));
    % 2 zero crossing means that in the window is 0.5 periode.
    % 3 zero crossing means 1 periode
    % 4 zero corssing = 1.5 periode ... ans so on.

    % Find out number of zero crossing
    checkZerosDir(1,kk) = length(zeroIndexDir{1,kk});
    checkZerosRef(1,kk) = length(zeroIndexRef{1,kk});

    % Condition - if num of zero crossing in direct and reflected window is
    % not equal then get the lowes number from one of them and save
    if ((checkZerosDir(1,kk)) ≥ (checkZerosRef(1,kk))) || ...
       ((checkZerosDir(1,kk)) ≤ (checkZerosRef(1,kk)))
            minZeroNum(1,kk) = min([(checkZerosDir(1,kk)) ...
                (checkZerosRef(1,kk))]);
    end

    % Get index of the first zero crossing
    firstZeroIndexDir = zeroIndexDir{1,kk};
    firstZeroIndexRef = zeroIndexRef{1,kk};

    % If there is only 1 or 2 zero crossing then use whole window
    if minZeroNum(1,kk) == (1) || minZeroNum(1,kk) == (2)
       dataDirr{1,kk} = dataDir(:,kk);
       dataReff{1,kk} = dataRef(:,kk);
    else
       dataDirr{1,kk} = dataDir( firstZeroIndexDir(1,1) : ...
           firstZeroIndexDir(minZeroNum(1,kk),1), kk);
       dataReff{1,kk} = dataRef( firstZeroIndexRef(1,1) : ...
           firstZeroIndexRef(minZeroNum(1,kk),1), kk);
```

```
    end

end

    dataDirOut = dataDirr;
    dataRefOut = dataReff;

end
```

```
function param_check(hM, hL, Tpulse, Cs, dMW, dMC, dMF)
%PARAM_CHECK
%   Checks main parameters and their condition.
%   If the parameters are suitable, the message dialog box will appear.
%   hM ... Height between microphone and test sample (in meters).
%   hL ... Height between loudspeaker and test sample (in meters).
%   Tpulse ... Pulse duration (in seconds).
%   Cs ... Celerity of sound (meters per second).
%   dMW ...Distance between microphone and wall (in meters).
%   dMC ...Distance between microphone and ceiling (in meters).
%   dMF ...Distance between microphone and floor (in meters).
%
%THE CONDITIONS
%   Tpulse ≤ (2 * hM) / Cs
%   dMW ≥ sqrt((((hL-hM) + (Cs*Tpulse))/2)^2 - ((hL-hM)/Cs)^2)
%   dMC ≥ (hL-hM) + ((Cs*Tpulse)/2)
%   dMF ≥ (Cs * Tpulse) / 2

%% Algorithm
% Check the parameter Tpulse
if Tpulse > (2 * hM) / Cs
    error('Tpulse does not satisfy the condition: Tpulse ≤ (2 * hM) / Cs');
end

% Check the parameter dMW
if dMW < sqrt((((hL-hM) + (Cs*Tpulse))/2)^2 - ((hL-hM)/Cs)^2)
    error('Distance between the microphone and the wall does not satisfy ...
        the condition: dMW ≥ sqrt((((hL-hM) + (Cs*Tpulse))/2)^2 - ...
        ((hL-hM)/Cs)^2)');
end

% Check the parameter dMC
if dMC < (hL-hM) + ((Cs*Tpulse)/2)
    error('Distance between the microphone and the ceiling does not ...
        satisfy the condition: dMC ≥ (hL-hM) + ((Cs*Tpulse)/2)');
end

% Check the parameter dMF
if dMF < (Cs * Tpulse) / 2
    error('Distance between the microphone and the floor does not satisfy ...
        the condition: dMF ≥ (Cs * Tpulse) / 2');
end

% Message Dialog Box
msgbox('All parameters are acceptable');

end
```

```matlab
function [ident_detect_value] = pulse_detect(signal_data, number_pulse, ...
    fs, Tp)
%PULSE_DETECT
%   Detects the number of pulses including start/stop pulse.
%
%   signal_data ... Data for processing.
%
%   number_pulse ... Number of the pulses in signal_data.
%
%   fs ... Sampling frequency of the signal_data.
%
%   Tp ... Pulse duration in seconds.
%
%   ident_detect_value ... The coordinates of the individual beginnings of
%   detected pulses in the signal_data.

% Transform signal data as absolute value
y = abs(signal_data);

% Get max value of 'y'
maxy = max(abs(y));

% Set treshold
org_threshold = maxy / 100;
threshold_data = org_threshold;
shiftTreshod = .2;  % Rise value of threshold.

pulse = 0;

% Until 'pulse' and 'number_pulse' are unequal then repeat cycle
while pulse ≠ number_pulse

    % Find the data with higher threshold
    detect_value_high = find(y(:) ≥ abs(threshold_data));

    % Detect the biggest changes
    diffDetect_high = [min(detect_value_high); diff(detect_value_high)];

    % Find the threshold
    sort_high = sort(diffDetect_high,'descend');
    threshold_high = min(sort_high(1:number_pulse,1))/2;
    threshold = max(threshold_high);

    % Pulse detection
    ident_diffDetect_high = diffDetect_high(:) ≥ threshold;

    % Find the coordinates of the pulses
    ident_detect_value_high = detect_value_high(ident_diffDetect_high);

    % Sorts of high and low values which represent the beginnings of pulses
    ident_value = sort(ident_detect_value_high);
    ident_detect_value = ident_value;
    for i = 2 : length(ident_value)
        if ident_value(i) - ident_value(i-1) ≤ fs*Tp
            ident_detect_value(i) = 0;
        end
    end

    % Gain the coordinates of the pulses
    coordinate_ident = ident_detect_value == 0;
```

```
    ident_detect_value(coordinate_ident) = [];
    pulse = length(ident_detect_value);

    threshold_data = threshold_data + (org_threshold * shiftTreshod);
end
end
```

```
function show_pulse_detect(signal_data, pulse_detect, stop_pulse, fs)
%SHOW_PULSE_DETECT
%   Shows the plot with the beginnings of the pulses.
%
%   signal_data ... Data for processing.
%
%   pulse_detect ... The coordinates of the individual pulses in
%   the signal_data.
%
%   stop_pulse ... If the stop_pulse == 0, the beginning of the direct
%                  pulses are shown.
%
%                  If the stop_pulse == 1, the beginning and end of
%                  the direct pulses are shown.
%
%                  If the stop_pulse == 2, the beginning and end of
%                  the reflected pulses are shown.
%
%                  If the stop_pulse == 3, the beginning and end of
%                  the direct and the reflected pulses are shown.
%
%   fs ... Sampling frequency of the signal_data. If the 'fs' is used,
%   the plot is displayed in the time domain


show_plot = zeros(length(signal_data),1);

for i = 1 : length(pulse_detect(:,1))
    if stop_pulse == 0
        show_plot(pulse_detect(i,1),1) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,1),2) = min(signal_data) + ...
            min(signal_data)/4;
    elseif stop_pulse == 1
        show_plot(pulse_detect(i,1),1) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,1),2) = min(signal_data) + ...
            min(signal_data)/4;
        show_plot(pulse_detect(i,2),3) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,2),4) = min(signal_data) + ...
            min(signal_data)/4;
    elseif stop_pulse == 2
        show_plot(pulse_detect(i,3),1) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,3),2) = min(signal_data) + ...
            min(signal_data)/4;
        show_plot(pulse_detect(i,4),3) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,4),4) = min(signal_data) + ...
            min(signal_data)/4;
    elseif stop_pulse == 3
```

```
        show_plot(pulse_detect(i,1),1) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,1),2) = min(signal_data) + ...
            min(signal_data)/4;
        show_plot(pulse_detect(i,2),3) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,2),4) = min(signal_data) + ...
            min(signal_data)/4;
        show_plot(pulse_detect(i,3),5) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,3),6) = min(signal_data) + ...
            min(signal_data)/4;
        show_plot(pulse_detect(i,4),7) = max(signal_data) + ...
            max(signal_data)/4;
        show_plot(pulse_detect(i,4),8) = min(signal_data) + ...
            min(signal_data)/4;
    end
end

if nargin == 4
    % Shows the time domain
    N = length(signal_data);
    ts = 1/fs;
    tmax = (N - 1) * ts;
    t = 0:ts:tmax;

    plot(t,signal_data)
    hold on

    if stop_pulse == 0
        stem(t,show_plot(:,1),'r','Marker','none')
        stem(t,show_plot(:,2),'r','Marker','none')
    elseif stop_pulse == 1
        stem(t,show_plot(:,1),'r','Marker','none')
        stem(t,show_plot(:,2),'r','Marker','none')
        stem(t,show_plot(:,3),'r','Marker','none')
        stem(t,show_plot(:,4),'r','Marker','none')
    elseif stop_pulse == 2
        stem(t,show_plot(:,1),'g','Marker','none')
        stem(t,show_plot(:,2),'g','Marker','none')
        stem(t,show_plot(:,3),'g','Marker','none')
        stem(t,show_plot(:,4),'g','Marker','none')
    elseif stop_pulse == 3
        stem(t,show_plot(:,1),'r','Marker','none')
        stem(t,show_plot(:,2),'r','Marker','none')
        stem(t,show_plot(:,3),'r','Marker','none')
        stem(t,show_plot(:,4),'r','Marker','none')
        stem(t,show_plot(:,5),'g','Marker','none')
        stem(t,show_plot(:,6),'g','Marker','none')
        stem(t,show_plot(:,7),'g','Marker','none')
        stem(t,show_plot(:,8),'g','Marker','none')
    end

    hold off
    xlabel('time (s)')
    ylabel('Amplitude')
else
    % Shows the samples of the signal
    plot(signal_data)
    hold on

```

```
    if stop_pulse == 0
        stem(show_plot(:,1),'r','Marker','none')
        stem(show_plot(:,2),'r','Marker','none')
    elseif stop_pulse == 1
        stem(show_plot(:,1),'r','Marker','none')
        stem(show_plot(:,2),'r','Marker','none')
        stem(show_plot(:,3),'r','Marker','none')
        stem(show_plot(:,4),'r','Marker','none')
    elseif stop_pulse == 2
        stem(show_plot(:,1),'g','Marker','none')
        stem(show_plot(:,2),'g','Marker','none')
        stem(show_plot(:,3),'g','Marker','none')
        stem(show_plot(:,4),'g','Marker','none')
    elseif stop_pulse == 3
        stem(show_plot(:,1),'r','Marker','none')
        stem(show_plot(:,2),'r','Marker','none')
        stem(show_plot(:,3),'r','Marker','none')
        stem(show_plot(:,4),'r','Marker','none')
        stem(show_plot(:,5),'g','Marker','none')
        stem(show_plot(:,6),'g','Marker','none')
        stem(show_plot(:,7),'g','Marker','none')
        stem(show_plot(:,8),'g','Marker','none')
    end

    hold off
    xlabel('samples (n)')
    ylabel('Amplitude')
end

end
```

```
function show_pulse_window(signal_data, fs)
%SHOW_PULSE_WINDOW
%   Shows the window with the pulse.
%
%   signal_data ... Data for processing.
%
%   fs ... Sampling frequency of the signal_data. If the 'fs' is used,
%   the plot is displayed in the time domain.

if nargin == 2
    % Shows the time domain
    N = length(signal_data);
    ts = 1/fs;
    tmax = (N - 1) * ts;
    t = 0:ts:tmax;

    plot(t,signal_data)
    xlabel('time (s)')
    ylabel('Amplitude')

else
    % Shows the samples of the signal

    plot(signal_data)
    xlabel('samples (n)')
    ylabel('Amplitude')

end
end
```

```
function [Cs] = sound_speed(temperature)

% Equation for the speed sound [meters per second]
 Cs = 331.3 + 0.606*temperature;
```

```
function [dataDir, dataRef] = used_win_interval(usedPercent, data_dir, ...
    data_ref)
%USED_WIN_INTERVAL
%   This function uses required percent of the present window.
%   Interval is in the middle of used window.
%
%   usedPercent ... Percents from window, which will be used for analyse.
%
%   data_dir ... data of directed windows.
%
%   data_ref ... data of reflected windows.

usedInterval = round((length(data_dir) / 100) * usedPercent);
border = round( ((length(data_dir) - usedInterval) / 2));

dataDir = data_dir(border : (border +  usedInterval),:);
dataRef = data_ref(border : (border +  usedInterval),:);

end
```

```
function [pressDir, pressRef] = wave_rmsPressure(dataDir, dataRef)
%WAVE_RMSPRESSURE
%   This function couts an RMS pressure of direct and reflect waves.
%   Interval is in the middle of used window.
%
%   dataDir ... Directed waves.
%   dataRef ... Reflected waves.
%
%   OUTPUTS:
%   pressDir ... RMS pressures of directed waves.
%   pressRef ... RMS pressures of reflected waves.

% EnDir = sum(abs(dataDir(:,:)).^2)';
% EnRef = sum(abs(dataRef(:,:)).^2)';

% RMS pressure

if iscell(dataDir)
    for i = 1:length(dataDir)

        pressDir(1,i) = sqrt(sum(dataDir{1,i}.^2,1)/length(dataDir{1,i}));
        pressRef(1,i) = sqrt(sum(dataRef{1,i}.^2,1)/length(dataRef{1,i}));

    end
else
    pressDir(1,:) = sqrt(sum(dataDir(:,:).^2,1)/length(dataDir(1,:)));
    pressRef(1,:) = sqrt(sum(dataRef(:,:).^2,1)/length(dataRef(1,:)));
end

pressDir = pressDir';
pressRef = pressRef';

end
```