ELECTROMAGNETIC SOUND CREATION

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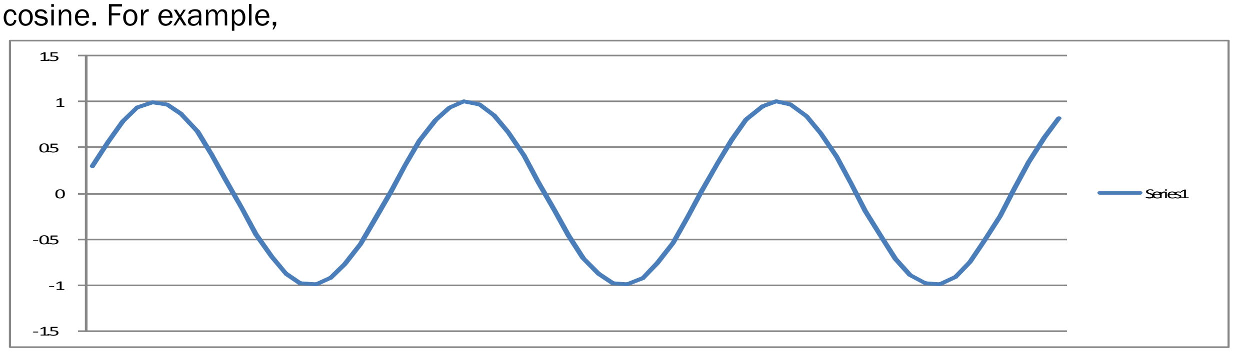
INTRODUCTION

The creation of music and sound are rooted deeply in the principles of physics such as vibration, periodic motion and electromagnetism. Sound waves are mechanical waves[1] (they need a medium to propagate). Even though sound waves are longitudinal waves (The direction of motion of the wave and the direction of motion of the particles of the medium are parallel to each other) when they propagate through air, they act as longitudinal as well as transverse waves[2] (the direction of motion of the wave and the direction of motion of the particles of the medium are perpendicular to each other) when they propagate through solids[2]. For the sake of simplicity, in this project, the sound waves are considered to propagate through solids, and are hence transverse in nature. When sound waves are transverse, they are much easier to study.

The purpose of this project is explaining and exploring the electromagnetic concepts behind music creation. The project includes analysis of the structure of sound waves, steps and principles behind digitalization of sound and exploration on how the sounds of the right frequencies are produced in the speakers from the information in the CD or any other storage device and the effect of current through the electromagnet in the speaker and how it affects the sound produced.

SOUND WAVES

Sound waves are transverse waves that propagate through the oscillations of the particles of the medium about their mean position. The whole sound wave is described by the motion of just one particle on the wave [2]. The wave is represented by one of the two trigonometric functions, sine or cosine. For example,



$y(x,t) = A \sin(\acute{\omega} t + \Phi)$ [3]

Where

$y(x,t)$ is the vertical displacement of the particle from its mean position.

'A' is the maximum displacement of the particle from its mean position

'$\acute{\omega}$' is the

't' is the time taken in seconds,

' Φ', the phase difference, will be considered to be 0, for simplicity.

THE PROCESS OF SOUND DIGITALIZATION

Over the last four decades, sound digitalization has become a very popular method of recording audio such as songs and films. The earliest instances of sound digitalization date back to the 1960s where the first and the least sophisticated techniques were employed to digitalize the sounds from the traditional song, 'A bicycle for two', in 1961 [6]. Towards the end of the 1960s, sound digitalization had become significantly more sophisticated. In 1968, Rissett's 'Computer Suite From Little Boy' was put through a much more advanced digital technique than its predecessor to not only record the sounds, but also to manipulate them[6]. The song itself was split into its component sound signals and each of these signals was worked upon separately. That is the strength, timber, precision and tone of each sound was repeatedly modified and improved until the composer found his desired arrangement. One of the most interesting aspects of 'Computer Suite From Little Boy' was the very unique sound- which started as a very high pitched note and continuously decreased to a low pitched one without a pause. This effect was achieved through a technique called 'analog programming', developed in 1967.[6]

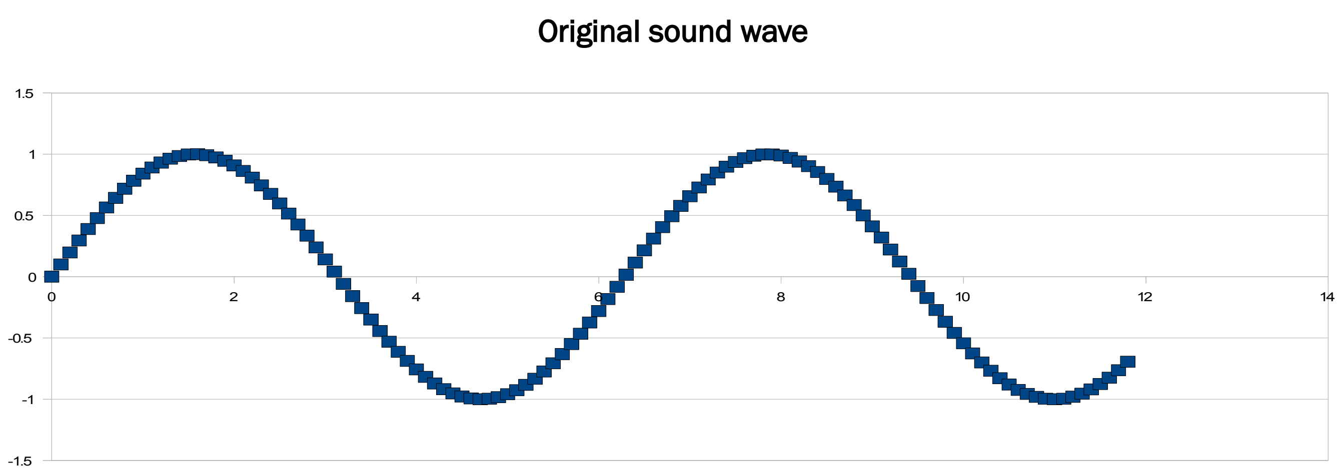
Sound waves or are continuous displacements of air pressure through time. The sound digitalization process goes like this:

First the sound waves created by a person or a musical instrument are converted to electrical signals after being captured by the microphone. Then, the electrical signals are converted to digital signals through a process called 'Sampling and Quantization'. Then the converted digital signals are passed through a digital processor, where it is filtered and additional effects are added. Then, the sound file is stored in a hard drive.[10]

In the process of playing the stored file, the stored digital signals are converted back to electric voltage signals. When these electric voltages are passed through a speaker or an earphone, the signals are converted back to sound, which we hear.

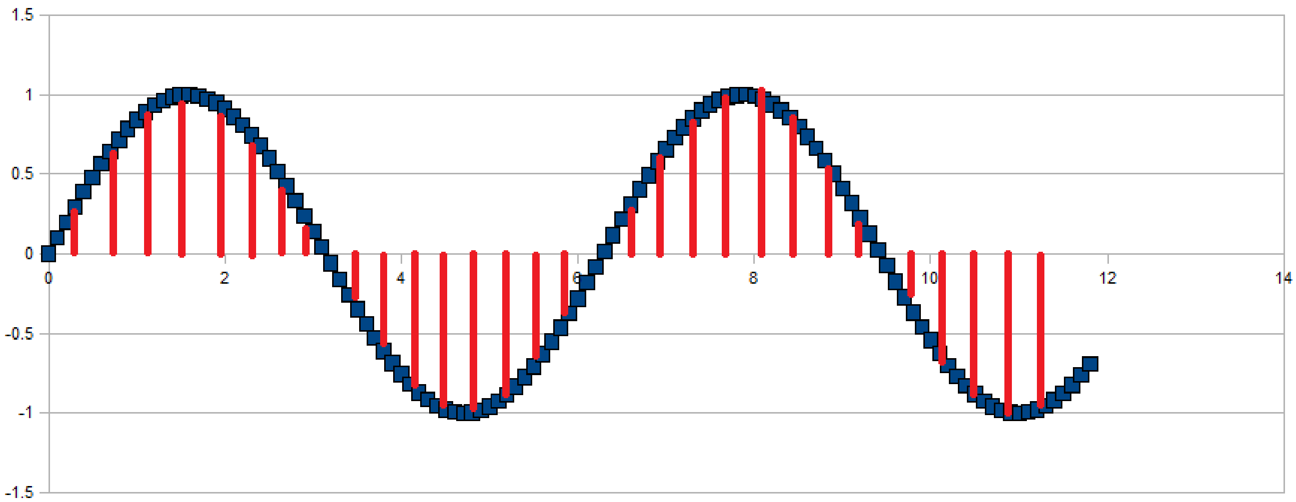
SOUND SAMPLING AND QUANTISATION PROCESS

Sampling of sound, by definition, is the reduction of a continuous signal to a discrete signal.[10] The continuous signal has, as the name suggests, continuous values over time. These are also called 'analog signals'. When a sound signal is sampled, specific values of the wave function are obtained regularly and periodically, over a time period 'T', called the 'Sampling interval'. The number of times the sampling is made over this time period is called the 'sampling frequency'[10]. A typical musical sound wave looks like:



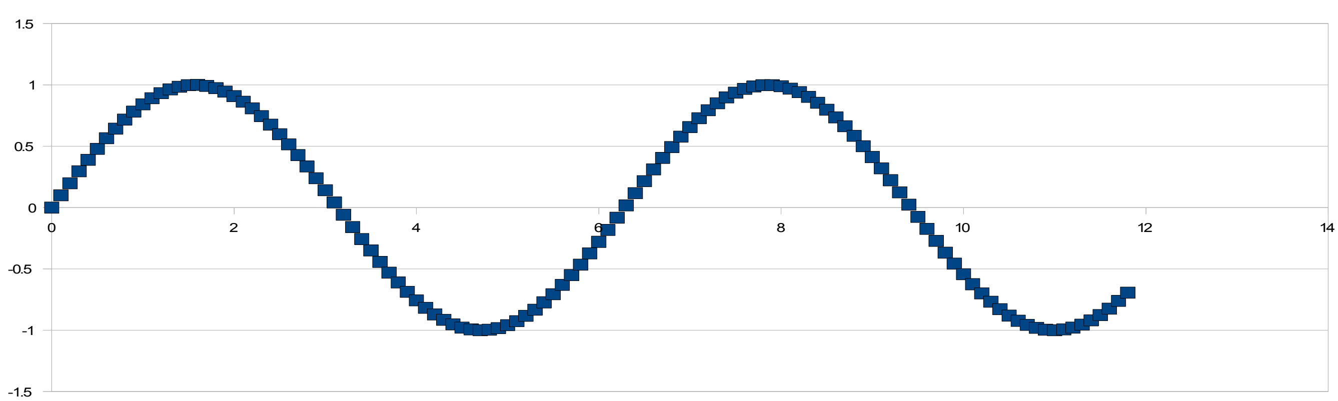
The same sound wave, sampled over time period seconds, once every seconds looks like

Sampled sound wave

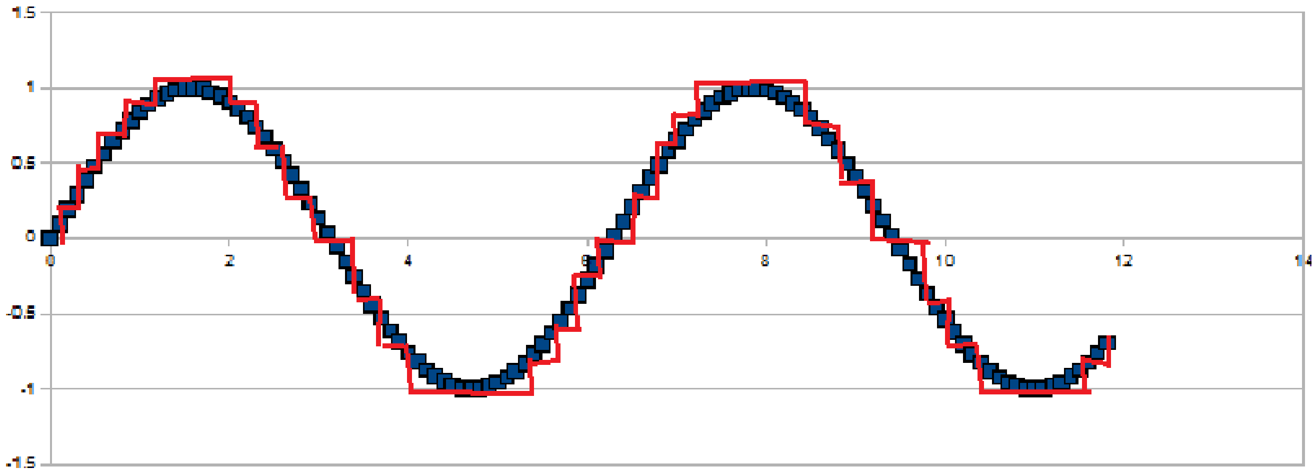


The second process, called 'Quantization', is defined as 'the process of mapping a large set of input values to a smaller set – such as rounding values to some unit of precision[10]. The rounding off of the values is done to eliminate 'unwanted precision'. The quantization process is very similar to the sampling process, and the notable difference is that in quantization, the values at specific points in the interval are shared by multiple time periods. Since the separate or continuous values of the original analog signal of the sound wave is lost as the wave is quantized, perfect reconstruction of the original analog signal is impossible[10]. A typical quantization process looks like this:

Original sound wave



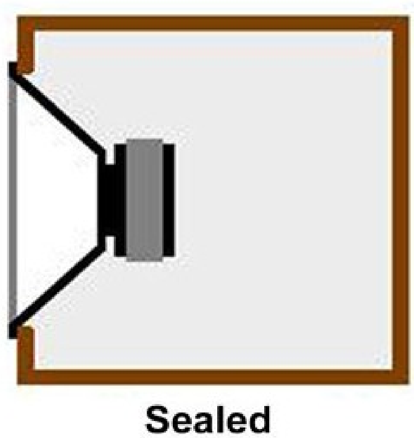
Quantized sound wave



SOUND PRODUCTION IN SPEAKERS

As stated above, the analog signals (electrical voltages) are passed through a Analog- Digital convertor where they are converted back into digital signals. These digital signals are sent to the speaker where they are converted into sound waves.

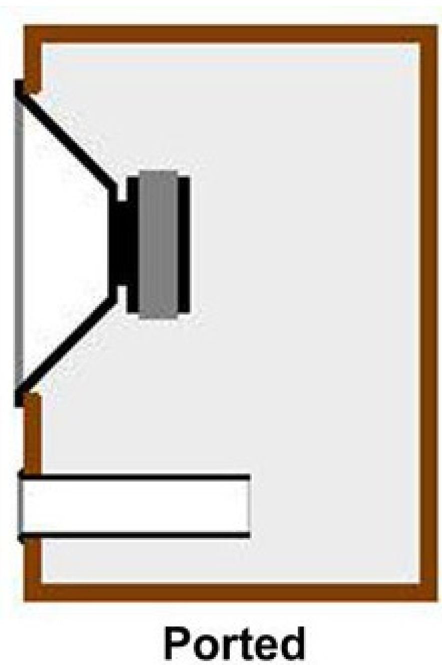
A traditional speaker consists of an enclosed space where all the units/ parts of the speaker are contained. The enclosed container is made up of a material that effectively absorbs the vibrations produced in the driver. The enclosed container is of many types, the most common being the acoustic suspension enclosure, which looks like:



(http://www.dummies.com/how-to/content/types-of-speaker-enclosures-sealed-and-ported.html)

The acoustic enclosure container is completely sealed and no air can escape out or into the container. So, the internal pressure in the enclosure is constantly changing, due to the vibration of the diaphragm. This varying pressure produces waves in the air that reach our ears as sound. This is the least efficient design of speakers as the speaker has to do a lot of work to overcome the air pressure and hence is not very energy efficient.[9]

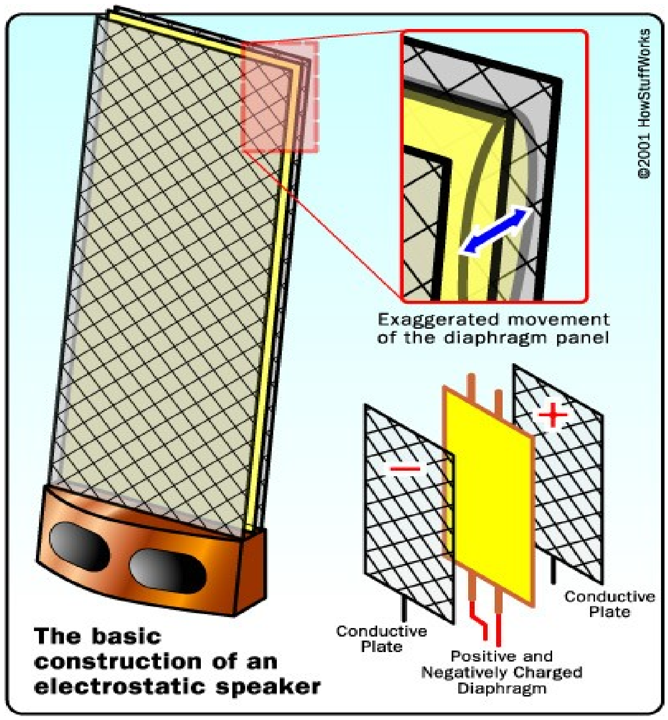
The other type of enclosure, the bass reflex- type, looks like:



(http://www.dummies.com/how-to/content/types-of-speaker-enclosures-sealed-and-ported.html)

In this type, as shown, the enclosure is made with a small outlet or hole towards the bottom of the speaker. This is done so that the when the diaphragm moves backwards (that is, into the speaker's enclosure), the resulting air pressure is pushed out through the outlet in the speaker and hence one gets two sound waves instead of one, which boosts the overall sound level of the speaker. However, there is a disadvantage associated with this kind of speaker. The pressure from outside that pushes the diaphragm back into the enclosure is reduced and hence the vibration of the diaphragm is somewhat reduced, which affects the sound precision.[9]

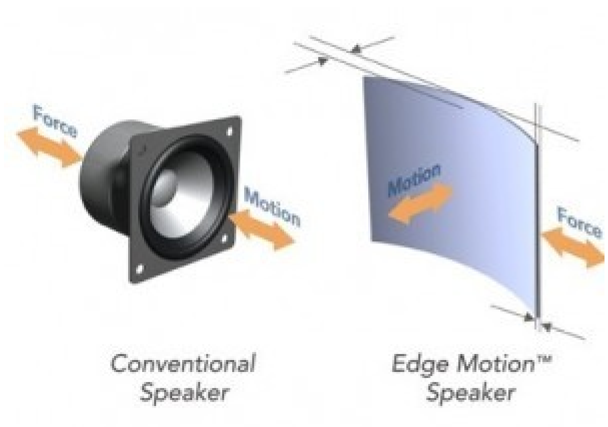
The more advanced and sophisticated type of speaker is the electrostatic speaker. In this kind of speaker, there is no driver or an enclosed container. Instead, the speaker consists of a very thin diaphragm that is made of a strong material, which is suspended between two conducting panels that are connected to the electric supply. The electric supply, which is constantly varying, produces varying charges on the conducting panels, which creates a time varying electric field between them. The diaphragm, which is between these panels is moved very fast or vibrated by this time varying field, and hence, sets the air around it in motion, which creates the sound waves[9]. The electrostatic speaker looks like:



(http://electronics.howstuffworks.com/question713.html)

LATEST TECHNOLOGIES FOR SOUND PRODUCTION

The diaphragm with the electromagnet has been the traditional design for the speaker for a long time, now (Over 40 years, to be exact). But, in 2011, development of a new speaker technology began to take root. This new technology was developed by a company called Emo Labs[9, 10]. This new speaker incorporates a new technology called the Edge Motion. This technology, like its predecessors, also uses a vibrating membrane/ diaphragm, but instead of being attached to an enclosure or a separate speaker system, the diaphragm is attached to a standard television screen. The difference between this model and its predecessors is that the electric panels responsible for creating the electric field that vibrates the diaphragm is on either side of the diaphragm, and not on the front and/or in the rear. So, the diaphragm, which is stretched across the screen, cannot move sideways, even though to field pushes it to. Therefore, it flexes or folds to and fro into and out of the television screen, which creates the vibrations, which in turn create the sound waves[9, 10]. The new speaker looks like:



(http://www.wired.com/gadgetlab/2009/05/0501piezo/)

DRIVER VIBRATION

The driver, which is the part of the speaker that vibrates to create the sound, in connected to an electromagnet, which is placed close to a permanent magnet. The electromagnet is connected to a device called the 'amplifier', which rapidly switches the flow of current through the electromagnet. That is, the orientation of the poles are changed rapidly. This causes rapid changes in the direction of current flow through the electromagnet, which causes rapid shifts in the orientation of its poles. The permanent magnet, which is in close proximity to the electromagnet, naturally exerts a force on it. Due to the rapid change in configuration of the orientation of the electromagnet's poles, the direction of the force also rapidly reverses. This causes the electromagnet to move back and forth rapidly, or vibrate, which in turn vibrates the driver, which creates the sound. [8, 9, 10.]

SOUND AMPLITUDE CALCULATION

A typical loudspeaker's permanent magnet produces a magnetic field in the range of 1T to 2.4T[9].

For calculation, lets consider a speaker with a permanent magnet that creates a field of magnitude 1.5T. The number of turns of the coil in the speaker varies with the size and purpose, and type of the speaker, but for calculation, let's consider a voice coil with 500 turns. Also, the voice coil is 2cm long. The voice coil, along with the iron core and the diaphragm weighs approximately 2kg. The typical power rating of a commercial speaker is around 1.5W. And since the domestic voltage in USA is 110 V, the current rating of a typical speaker is[9, 8]:

$$I = P/v$$

$$I = 1.5/110 \quad = \quad 0.01364A$$

From this information, the maximum magnitude of field generated in the speaker's electromagnet can be calculated. The permeability of a typical soft iron core is 80 mH.[8]

$$B_{max} = \mu ni$$

$$B_{max} = .08 \times 25000 \times .01364$$

$$B_{max} = 27.28 \text{ T}$$

This is the maximum magnitude of the magnetic field that can be generated in the voice coil and this corresponds to the maximum volume of the sound out of the speaker, which also corresponds to the maximum amplitude of vibration of the diaphragm.

Calculating the Amplitude of vibration:

For most speakers, a frequency of 1kHz produces an amplitude of vibration of 1cm, for maximum current. Modern speaker have the ability to play sounds even beyond 20,000Hz, the hearing range of humans. But, for calculation purpose, let's consider a good quality speaker that can produce sounds from 20Hz to 20,000kHz, the human hearing range[8]. Therefore, the amplitude of vibration of the speaker's diaphragm is described by the equation:

$$A(t) = 1cm \sin(\omega t)$$

$$A(t) = 1cm \sin(2\pi f * t)$$

Where 'f' is the frequency of the sound wave or the musical note being played and 'ω' is the angular frequency.

Now, consider the popular rhyme "Mary had a little lamb". The song includes the following notes: D, E, F# and A of the second octave (from a traditional piano or keyboard)[5]:

NOTE	FREQUENCY (Hz)
D	1244.51
E	1396.91
F#	1567.98
A	1864.65

Now, the for a given current, the magnetic field created in the voice coil, the force of repulsion/ attraction between the coil and the permanent magnet and the amplitude of vibration for each note is calculated using the above equations. Let the given current be: 0.00682A. The force on a magnetic moment (due to the voice coil, here) is given by:

$$F = B\, dm/ds$$ [8]

Where B is the magnetic field due to the permanent magnet, 'm' is the magnetic moment of the voice coil. Since

$$F = Ma, \quad \text{(where M is the mass of the elcetromagnet)}$$

$$B\, dm/ds = Ma$$

or

$$a = (B/M)\, dm/ds \text{-----------------}(1)$$

From Newton's equation of motion, $s = ut + 0.5at^2$. Here, $u = 0$, $t = 1/f$, acceleration is found from equation (1), and s= the amplitude of vibration of the diaphragm. Therefore,

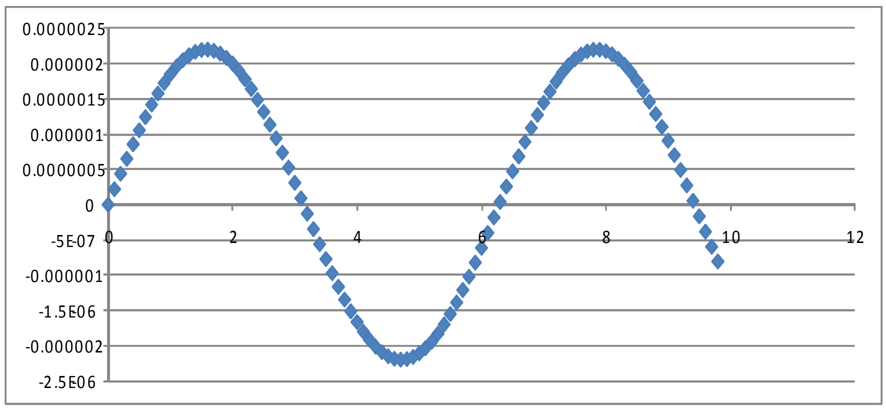
$$A = 0.5 \times [(B/M)\ (dm/ds)] \times (1/t)^2$$

S, with this equation, the specific amplitudes of vibrations for each note is calculated.

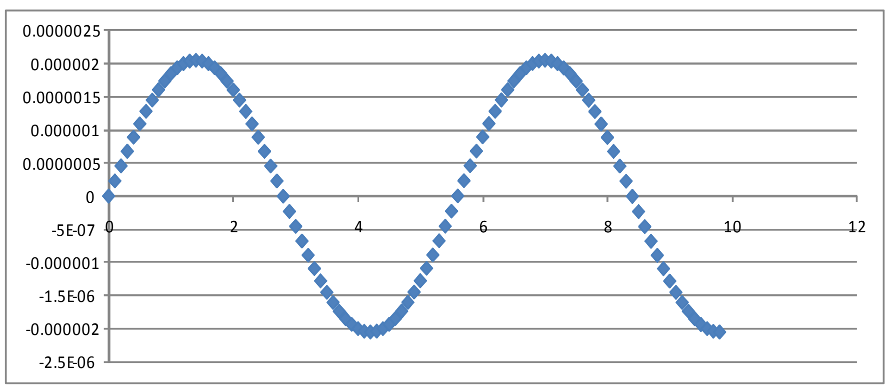
Current	Note	Amplitude
0.00682A	D	2.19×10^{-6} m
0.00682A	E	2.06×10^{-6} m
0.00682A	F#	1.64×10^{-6} m
0.00682A	A	1.16×10^{-6} m

The sound waves of these notes look like this:

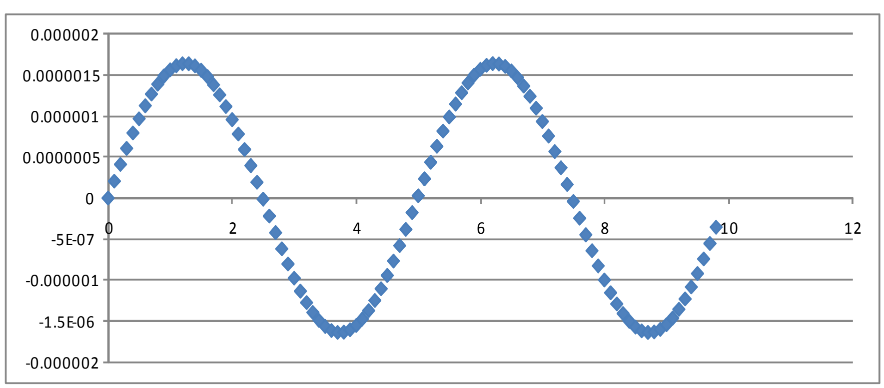
NOTE 'D'



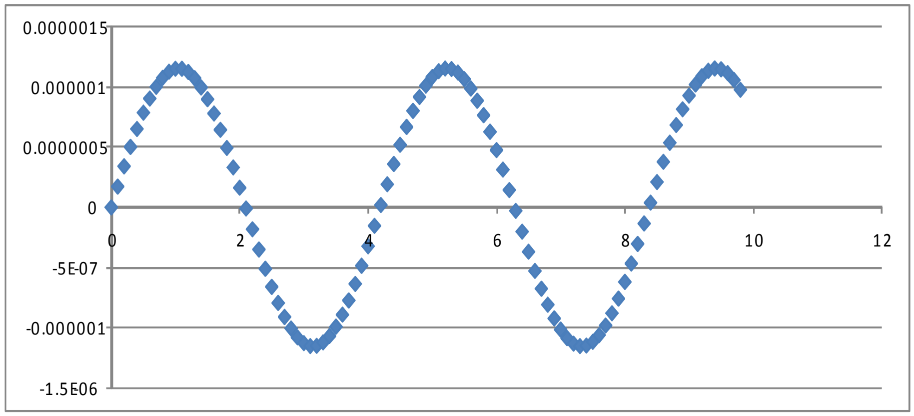
NOTE 'E'



NOTE 'F#'



NOTE 'A'

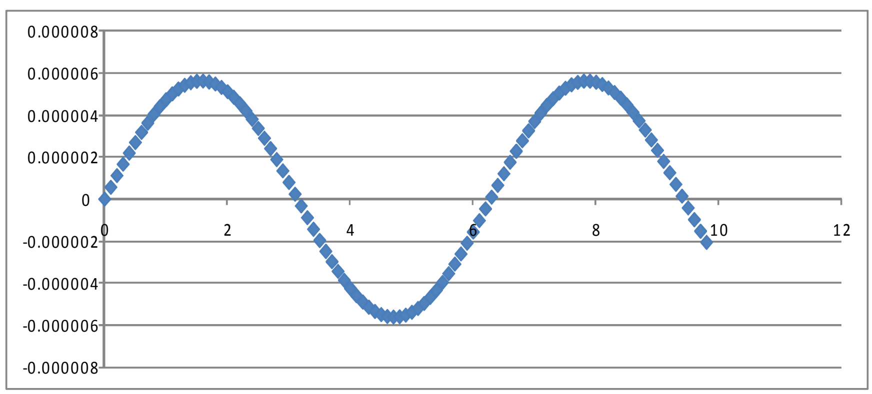


Now, lets take the maximum current that can flow through the wire, 0.01364 A, and calculate the amplitudes of vibrations for each notes. Note: This current corresponds to full volume.

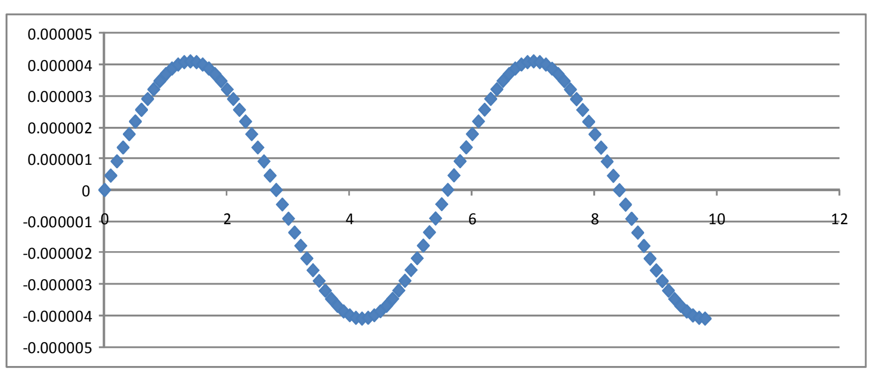
Current	Note	Amplitude
0.01364 A	D	5.65×10^{-6} m
0.01364 A	E	4.12×10^{-6} m
0.01364 A	F#	3.56×10^{-6} m
0.01364 A	A	2.31×10^{-6} m

The sound waves of these notes look like:

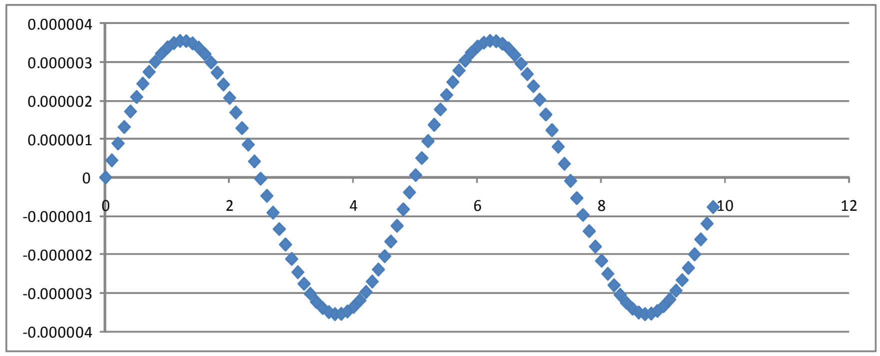
NOTE 'D'



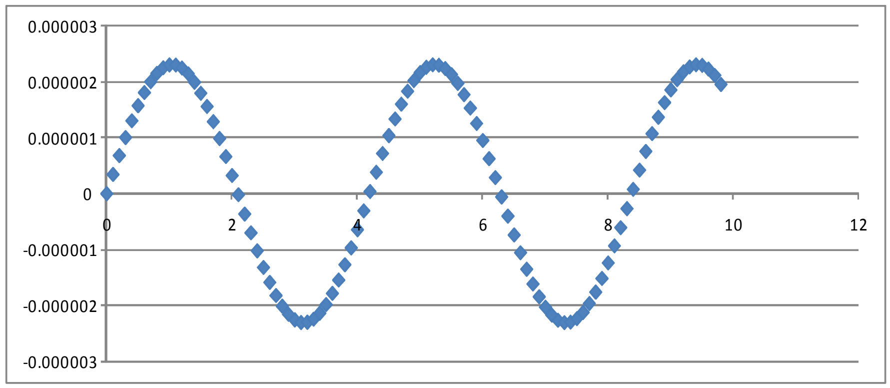
NOTE 'E'



NOTE 'F#'



NOTE 'A'



Note that the frequencies of the waves in both Table 1 and 2 remain the same. This is why one hears the 'same sound' , irrespective of the amplitude of vibration of the sound wave. The pitch and the tone of the sound depends only on the frequency of the sound wave and not on the amplitude of the vibration. So what does the amplitude of the vibration determine? It determines the volume of the sound produced. In other words, when the information from the CD or any storage device is relayed into the speaker, the amplifier in the speaker rapidly switches the orientation of the electric signal in the electromagnet. Hence, it also fluctuates the orientation of the oles and the direction of the force with the same frequency, that is, the frequency of the corresponding sound wave. The amount of current in the electromagnet determines the amplitude of the vibrations and hence the volume, and not the pitch of the sound itself.

CONCLISION

1. Sound is digitalized through a process called 'Sampling and Quantization'. Where the continuous analog signal is converted to discrete signals, that get rid of unnecessary precision.
2. Sound is produced in the speaker by the vibration of driver/ diaphragm due to a fluctuating field. The frequency of the fluctuation is equal to the frequency of the sound produced.
3. The rate of change of this fluctuation determines the frequency of the sound. The volume is determined by the amplitude of the vibrations, which is determined by the magnitude of current in the electromagnet.
4. A new technology of sound production is being developed, where the conducting panels are on either side of the vibrating diaphragm and not on the front and/or the rear.

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