Investigation of a Sweep Technique for Microphone Placement

by

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March 2015

Declaration

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Date: November 3, 2014

Abstract

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November 2014

Microphone placement is a tedious, time consuming, trial and error based process. Recordists use microphone placement as a method for altering the recorded timbre of an instrument and it is a vital part of the recording process. This thesis investigated a microphone sweep technique and remote microphone positioning as methods for identifying and evaluating microphone placements.

Uittreksel

Ondesoek van 'n Skuiftegniek vir Mikrofoonplasing

("Investigation of a Sweep Technique for Microphone Placement")

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Tesis: MPhil Musiektegnologie

November 2014

Mikrofoonplasing is 'n moeisame en tydrowende proses van eksperimentering. Opname tegnici gebruik mikrofoonplasing as 'n manier om die opgeneemde toonkleur van 'n instrument te verander en dit is 'n noodsaaklike aspek van die opname proses. Hierdie tesis het 'n mikrofoon skuiftegniek en afstandbeheerde mikrofoon posisionering ondersoek as metodes om mikrofoonplasings te identifiseer en evalueer.

Acknowledgements

I would like to express my sincere gratitude to the following people:

- Gerhard Roux, who provided excellent leadership and feedback.
- André Hartshorne for providing equipment, advice and guidance with electronics.
- Angelo Farina for the development of the Aurora plug-ins.
- Ouma Susann for the prayer and *sokkies*.

Dedications

My love Jessica, for all the support, encouragement,
enumerable cups of coffee, delicious food and love.
You are my rock.

Contents

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Nomenclature

Constants

$c =$	$331.26\,\mathrm{m \cdot s}$
$g =$	$9.81\,\mathrm{m \cdot s^2}$
$\epsilon_0 =$	$8.854188 \times 10^{-12}\,\mathrm{F{\cdot}m^{-1}}$

Variables

A	Area	$[\,\mathrm{m^2}\,]$
a	Radius of piston	$[\,\mathrm{m}\,]$
C	Capacitance	$[\,\mathrm{F}\,]$
C	Spring index	[ ]
D	Directivity function	[ ]
D	Mean diameter	$[\,\mathrm{m}\,]$
D_o	Outer diameter	$[\,\mathrm{m}\,]$
D_i	Inner diameter	$[\,\mathrm{mm}\,]$
dB	Decibel	$[\,\mathrm{dB}\,]$
dBV	Decibel voltage	$[\,\mathrm{dB}\,]$
d	Distance	$[\,\mathrm{m}\,]$
d	Depth	$[\,\mathrm{mm}\,]$
d	Wire diameter	$[\,\mathrm{m}\,]$
E	Wave energy	$[\,\mathrm{J}\,]$
e	Error	[ ]
F	Force	$[\,\mathrm{N}\,]$
F_i	Initial force	$[\,\mathrm{N}\,]$
f	Frequency	$[\,\mathrm{Hz}\,]$
f_s	Sampling rate	$[\,\mathrm{Hz}\,]$
G	Shear Modulus	$[\,\mathrm{GPa}\,]$

h	Height	[mm]
I	Current	[A]
I	Intensity	[W·m^{-2}]
k	Number of samples in frequency domain	[ ]
k	Spring rate	[N·m^{-1}]
J_1	Bessel function of the first kind	[ ]
k	Wavenumber	[ ]
l	Length	[m]
l_0	Free length	[m]
N	Number of turns	[ ]
n	Number samples in time domain	[ ]
P	Power	[W]
p	Pressure	[Pa]
Q	Electrical charge	[C]
Q	Directivity factor	[ ]
q	Angle	[°]
R	Resistance	[Ω]
R_{60}	Reverberation time	[s]
r	Radius	[m]
T	Period	[s]
T	Total time	[s]
t	Time	[s]
V	Voltage	[V]
V	Volume	[m^3]
v	Velocity	[m·s^{-1}]
w	Width	[mm]
X	Frequency domain signal	[ ]
x	Time domain signal	[ ]
x	Extension	[m]
x	Coordinate	[m]
y	Coordinate	[m]

Greek Letters

α	Statistical absorption coefficient	[ ]
α_θ	Absorption coefficient	[ ]
$\bar{\alpha}$	Average absorption coefficient	[ ]
ϵ	Dielectric constant	[ ]

θ	Angle	[$^{\circ}$]
λ	Wavelength	[m]
τ	Reverberation Time	[s]
τ_{un}	Uncorrected torsional stress	[MPa]
ϕ_i	Angle of incidence	[$^{\circ}$]
ϕ_r	Angle of reflection	[$^{\circ}$]
ω	Angular velocity	[rad·s^{-1}]

Subscripts

avg	Average
$elec$	Electrical
i	Incident
n	n^{th} component
r	Reflected
rms	Root-mean-squared
$shift$	Shift
un	Uncorrected
0	Reference
1	Measured

Superscripts

$^{-}$(over bar)	Average

Introduction

DURING recording, the recordist experiments with various microphone placements in order to achieve the desired sound (Ballou *et al.*, 2008:593). Although it is mitigated by knowledge and experience, this is a tedious trial and error process.

Microphone placement is a matter of personal taste (Boudreau *et al.*, 2005:1). Bartlett (2012) believes there is no one correct way for placing a microphone. Kefauver (2001:123) describes microphone selection and placement as an art, stating that the recordist is an artist creating a sonic portrait with different techniques and colour selections offered by the microphone palette. By utilizing subtleties that arise from minor adjustments, microphone placement can be used to change the recorded timbre of an instrument and obtain various sounds (Jensenius *et al.*, 2010:209; Ballou *et al.*, 2008:519). According to Huber & Runstein (2010:132), microphone placement is one of the recordist's most valuable tools and can play a very important role in achieving the desired sound.

Recordists make use of short test recordings to evaluate microphone placements (Dyar, 1961:49). They employ past experience, known techniques suggested by the collective experience of the sound engineering community (Kokkinis *et al.*, 2012:1), and rely on aural memory to guide them (Roginska *et al.*, 2012:8). If the sound is not suitable, the placement of the microphone is changed and reevaluated until one is found that exhibits the desired tonal balance and amount of room acoustics (Boudreau *et al.*, 2005:5). In the final stages of this process, very fine adjustments are made, refining microphone placement by millimeters and orientation by a few degrees (Roginska *et al.*, 2012:8).

A new method, namely the *sweep technique*, was investigated. The sweep technique employs a robotic device, controlled by the recordist, to remotely adjust microphone placement, while the recordist monitors the pick-up of the microphone from the studio control desk. This enables the recordist to sweep the microphone through a range of positions, while evaluating the resulting timbre of the sound. Additionally, it allows for fine microphone adjustments to be made.

This study begins with a detailed explanation of the principles of sound, its propagation and behavior within enclosed spaces, after which the signal chain and the equipment of a recording setup is discussed. Current popular microphone techniques are studied, followed by an experiment which was conducted to determine the effect of microphone placement on the recorded timbre of a guitar amplifier. A prototype device was designed with which microphone placement could be controlled and the sweep technique was investigated.

Chapter 1

Sound

Acoustics is a colloquial term that refers to the quality of enclosed spaces and their effect on the perception of speech and music. With regard to the physical sciences, acoustics is the science of sound vibrations in matter (Ingard, 2008:1). Sound not only implies that which can be heard, but also anything else that is governed by analogous physical principles. Therefore disturbances that are too low (*infrasound*) or too high (*ultrasound*) in frequency to be audible are also regarded as sound (Pierce, 1995:1).

1.1 Propagation of Sound

Musical instruments produce sound through mechanical, acoustical or electrical vibrations (Rossing & Fletcher, 2004:1). This creates a physical disturbance in the surrounding medium, the substance through which the disturbance travels (Toole *et al.*, 2002*a*:1.7). The medium in which sound propagates is usually air, but can also be solids or liquids (Conn, 2008:575). A large number of sound sources radiate energy through the action of vibrating solid surfaces upon the surrounding medium, but the effectiveness of radiation varies from source to source (Jacko, 2012:136).

Sound propagates through air by modulating the ambient atmospheric pressure (Brown, 2008:36). As a source vibrates it forces successive molecules out of their positions of rest (Welling *et al.*, 2013:11). The molecules are forced together in a knock on effect (Rumsey & McCormick, 2006:4) and cause an increase in air pressure (Gelfand, 2011:10), known as a *compression* (Toole *et al.*, 2002*a*:1.7).

The molecules have mass and therefore obeys Newton's laws[1] (Tanner &

[1] Sir Isaac Newton (1642-1727) developed a theory for the forces acting on a macroscopic body and the resulting motion of that body (Dale, 1946:1). The laws are collectively known as *Newton's Laws of Motion* and were first published in his famous book *Mathematical Principles of Natural Philosophy*, (Heinz, 2011:256). For more information, please see Benenson & Stöcker (2002:40-43) and Avison (1989:142-144).

Tanner, 2004:19). With each collision, momentum[2] is transferred from one molecule to another (Baehr & Stephan, 2011:286). When the molecules reach their maximum displacement, the elasticity of the medium forces them to return to their neutral position, but because of their momentum, they move past their neutral positions (Tanner & Tanner, 2004:19) and are displaced in the opposite direction to the direction of propagation of the sound. This causes the molecules to be pulled further apart from each other and results in a decrease in air pressure (Hausman *et al.*, 2012:85), known as a *rarefaction* (Rumsey & McCormick, 2006:4).

The molecules continue to oscillate about their neutral positions and create compressions and rarefactions (Toole *et al.*, 2002*a*:1.7). The disturbance is propagated through the medium (Conn, 2008:575) until the elastic forces in the medium exceed the opposing forces of momentum, at which point vibration stops (Tanner & Tanner, 2004:19).

Air is a very elastic medium, which is easily compressed and capable of very rapid expansion (Clark, 2010:81). This elasticity makes air an excellent medium for the transmission of sound energy (Tanner & Tanner, 2004:19).

1.2 Characteristics of Sound Waves

1.2.1 Amplitude

The amplitude is the maximum physical displacement of a particle in the path of a wave (Parekh, 2006:181) and is related to the perceived loudness of a sound (Rumsey & McCormick, 2006:2). Loudness corresponds to the measurable property of *intensity*, I (Biederman & Pattison, 2013:80). Intensity is defined as the power of a sound wave, P, that passes through an area, A, that is perpendicular to the direction of the sound wave (Winn, 2010:19.8), as shown in Equation (1.2.1).

$$I = \frac{P}{A} \tag{1.2.1}$$

The human audible intensity range is from 1 unit of intensity to 10^{15} units. This range is so vast than it is difficult to manage with normal numbering schemes (Baars & Gage, 2010:199). Therefore, intensity is measured in *decibels*[3] (Giordano, 2012:417). The decibel is a dimensionless unit that denotes a logarithmic power ratio (Londerville, 2013:271; Smith, 2013:59), as expressed in Equation (1.2.2), which denotes the ratio of some measured power, P_1, to a

[2] Momentum is the product of an object's mass and velocity. It is a vector quantity and has the same direction as the velocity of the object (Kirkpatrick & Francis, 2006:97).

[3] The decibel was originally used at Bell Telephone Labs to describe the attenuation of a signal in a mile of standard cable and was called a *transmission unit* (America *et al.*, 2010:146).

reference power, P_0. Figure 1.1 shows a graph of this equation which expresses the logarithmic relationship between decibels and the power ratio. As per example, a power ratio of 2 yields a decibel value of 3 dB, whereas a power ratio of 10 yields 10 dB.

$$dB = 10 \log_{10} \frac{P_1}{P_0} \tag{1.2.2}$$

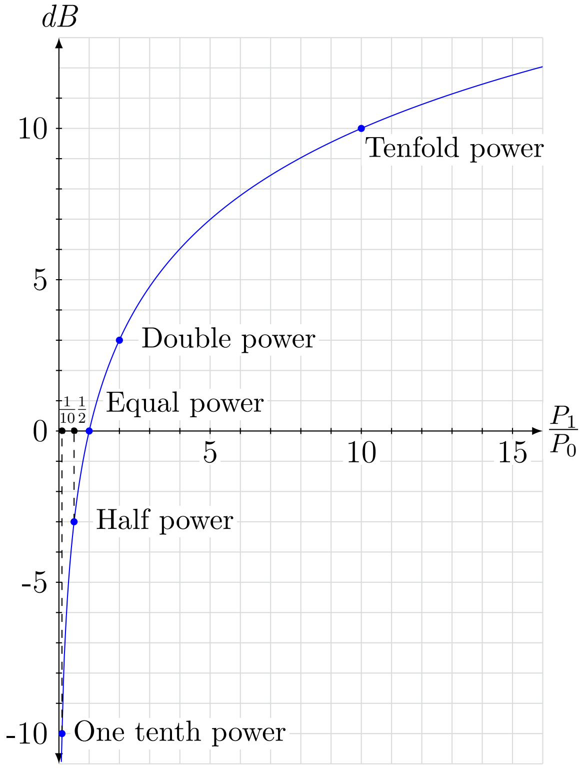


Figure 1.1: Decibel power.

The decibel is always a power ratio, but may be used to express many other variables so long as they are related to power. Electrical power is calculated as shown in Equation (1.2.3), where V and R refer to voltage and resistance, respectively (Brown, 2008:24). Substituting Equation (1.2.3) into Equation (1.2.2) yields decibel change in power as a ratio of voltages, as expressed in Equation (1.2.4) as dBV. If a standard reference value is used, the result is an absolute level and the unit is expressed as *decibel relative the original unit*. For a list of common decibel references used in the audio industry, refer to Table A.1 in Appendix A.

$$P_{elec} = \frac{V^2}{R} \tag{1.2.3}$$

$$\begin{aligned} dBV &= 10\log_{10}\frac{{V_1}^2}{{V_0}^2} \\ &= 20\log_{10}\frac{V_1}{V_0} \end{aligned} \tag{1.2.4}$$

In practice, sound intensities are measured in terms of sound pressures (Baken & Orlikoff, 2000:96). In a similar manner as described above, the decibel *sound pressure level* (SPL) of a signal may be calculated, as shown in Equation (1.2.5) where p refers to *root-mean-square*, or rms pressure and is measured in Pascals (Pa). As the name implies, an rms value of a function is defined as the square root of the mean of the square of the function (Furse *et al.*, 2009:22). Equation (1.2.6) expresses the rms value for pressure.

$$\begin{aligned} dB\ SPL &= 10\log_{10}\frac{p_1^2}{p_0^2} \\ &= 20\log_{10}\frac{p_1}{p_0} \end{aligned} \tag{1.2.5}$$

$$\begin{aligned} p_{rms} &= \sqrt{\frac{p_1^2 + p_2^2 + p_3^2 + \ldots + p_n^2}{n}} \\ &= \left[\frac{1}{n}\int_0^n p^2 dn\right]^{1/2} \end{aligned} \tag{1.2.6}$$

Acoustical power is proportional to the square of sound pressure, and thus a doubling of sound pressure results in a quadrupling of acoustical power (Eargle, 2002*a*:13). This corresponds to a 6 dB increase in SPL, as opposed to the 3 dB increase that would result from a doubling of acoustical power (Baken & Orlikoff, 2000:97).

The International Organization for Standardization (ISO) has defined the reference for SPL as 20 µPa (Probst, 2006:173), which is the lower threshold of human hearing at 1 kHz (Pocock, 2013:215; Dejonckere, 2001:42). The upper threshold is 20 Pa, at which listening becomes painful (Howard, 2009:24). Table A.2 in Appendix A lists a range of SPL and the equivalent sound pressure, power, musical dynamic and corresponding environment in which such a level may be experienced.

1.2.2 Frequency

The frequency of a sound wave, f, describes the rate at which air pressure fluctuates (Davis & Jones, 1989:1). It is also defined as the number of completed wave cycles that pass a fixed point in one second (Behrens & Michlovitz, 2005:60), where a cycle is the combination of one compression and one rarefaction (Furukawa, 2004:69). Frequency is expressed in cycles per second (cps), or Hertz[4] (Hz) (Brown, 2008:28).

Pitch is the psychological perception of frequency (Tanner, 2003:178), where high frequencies result in high-pitched sounds and low frequencies result in low-pitched sounds (Kalat, 2014:113). The perceptible audio spectrum extends from just under 20 Hz to 20 000 Hz (Nathan, 1998:53), a frequency ratio of nearly 10^3. Since every doubling in frequency results in one octave (Norton, 2003:264), humans can hear more than nine octaves (Rossing, 2007:2).

The periodic nature of waves is defined by the time it takes a wave to travel the distance of one wavelength, and is known as the period, T. The period is inversely proportional to the frequency of the signal and can be expressed in Equation (1.2.7) (Rumsey & McCormick, 2006:2).

$$T = \frac{1}{f} \tag{1.2.7}$$

1.2.3 Wavelength

The wavelength, λ, is equal to the speed sound, c, divided by the frequency of the sound, f, as expressed Equation (1.2.8) (Alten, 2012*b*:4). It is the actual distance that a wave travels before it repeats itself (Rozenblit, 1999:42).

$$\lambda = c/f \tag{1.2.8}$$

The wavelength is directly proportional to the speed of sound and inversely proportional to frequency (Thewissen & Nummela, 2008:176). The speed of sound is dependent on the density, elastic properties and temperature of the medium (Furukawa, 2004:69; Kirkpatrick, 2010:331). As a result, sound travels five times faster in water than in air, and more than three times faster in glass than in water (Taylor, 2000:312). The speed of sound also increases with $0.61\,\mathrm{m\,s^{-1}}$ for every one degree of temperature increase (Wagh & Deshpande, 2012:236). Sound waves in air at a room temperature of 22 °C travel at $343\,\mathrm{m\,s^{-1}}$ (Kirkpatrick, 2010:331; Toole *et al.*, 2002*a*:1.9). The speed of sound is independent of variation in ambient pressure and is the same at all frequencies (Newell, 2003:594).

When a wave propagates, the internal friction of the vibrating molecules is transformed into heat, resulting in the energy of the wave diminishing in the direction of propagation (Sohn, 2004:6). The energy is directly proportional to the frequency of the wave and is calculated by multiplying *Planck's constant*[5]

[4] Named after the German physicist Heindrich Rudolph Hertz (Sethares, 2005:12)

with the frequency, as shown in Equation (1.2.9) where E is the wave energy and $h = 6.625 \times 10^{-34}$ J s is Planck's constant (Malainey, 2011:23).

$$E = h\frac{c}{\lambda} = hf \tag{1.2.9}$$

Since wavelength and frequency are inversely proportional, waves with smaller wavelengths have more energy than waves with larger wavelengths. Accordingly, low frequency waves vibrate molecules slower and expend their energy slowly over larger distances, whereas high frequency waves vibrate molecules rapidly and expend their energy quickly over short distances (Behrens & Michlovitz, 2005:60). As a result, low frequency waves can travel further than high frequency waves (Breithaupt, 2000:398).

1.2.4 Phase and Polarity

Phase, ϕ, represents the instantaneous relationship between two signals (Kefauver, 2001:575) and indicates the shift in time between one period relative to another (Eargle, 2005:2.44). It is a notation in which one period of a wave is divided into 360° (Toole *et al.*, 2002*a*:1-11). A *phase shift*, ϕ_{shift}, expresses in degrees the fraction of a period or wavelength by which a wave has been shifted in the time domain. A phase shift of 90° between two identical waves corresponds to a shift of one quarter of the period of the waves, as illustrated in Figure 1.2 (Toole *et al.*, 2002*a*:1-11). A phase shift of 180° represents a special case where the two waves are said to be out of phase and the positive half-cycle of one wave precisely coincides with the negative half-cycle of the other (Rumsey & McCormick, 2006:8).

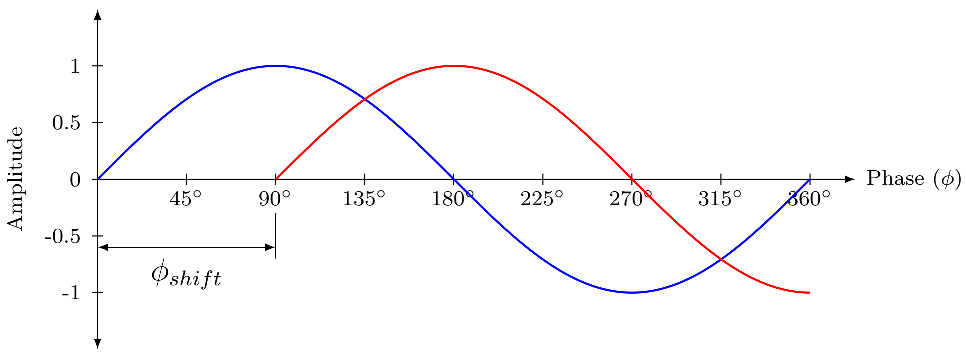


Figure 1.2: The phase shift between two identical sine waves.

Where phase is a matter of degree (White & Louie, 2005:298), polarity defines a condition and is independent of time (White & Louie, 2005:298). It is related

[5] Introduced in 1900 (Myhre, 2004:38) by the German physicist Max Karl Ernst Ludwig Planck (1858 - 1947). Planck was famous for his proposal that energy is transfered in units, called quanta (Rosen, 2008:253).

to the direction of an acoustical, electrical or magnetic force and can only be positive or negative, as illustrated in Figure 1.3 (Alten, 2013:214).

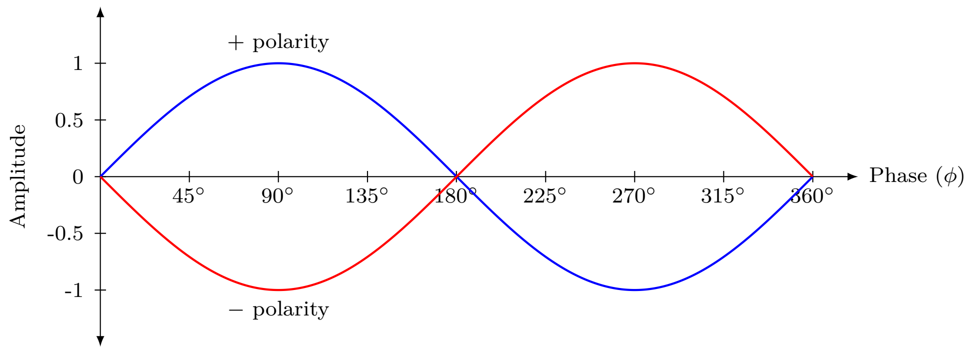


Figure 1.3: The polarity difference of two identical sine waves.

Whenever two waves occupy the same point, interference occurs. Interference is the summation of the amplitude of individual waves (Serway, 2013:448). Constructive interference occurs when waves reinforce each other and sound louder and destructive interference occurs when waves cancel each other and sound softer (Butlin, 2000:133). Interference does not alter the wave forms and both waves separate with the same amplitude and frequency as before they coincided (Sewell, 1999:172).

A recordist may choose to use multiple microphones on a sigle source to obtain a desired sound. In such a case, the microphone channels are combined and interference occurs. Incorrect polarity and phase may cause the onset of destructive interference, therefore it is very important when using multi-microphone setups (Collins, 2003:118). One method of obtaining a good phase relationship between multiple microphones is to perform individual microphone placements one by one, while continually checking the phase relationship between microphones that have been placed and the microphone that is currently being placed.

1.3 Doppler Effect

Whenever there is relative motion between a source of periodic sound waves and an observer, the frequency of sound waves will be changed (Watkinson, 1998:94). When the motion of the source or the observer is such that the distance between them decreases, the pitch of the sound is higher than the actual pitch produced by the source. Similarly, when the distance between the source and observer increases, the pitch is lower (Subrahmanyam & Lal, 1999:181). This phenomenon was first explained by the Austrian physicist Christain Johann Doppler in 1842 is and is known as the *Doppler effect* (Mott, 2014:55).

When the sound source moves in the same direction as the direction of propagation, the sound waves are compressed, as illustrated in Figure 1.4 (Gibbs *et al.*, 2011:74). Therefore, the observer receives sound with shorter wavelengths (Gibbs *et al.*, 2011:74). The wavelength and frequency of a sound wave are inversely proportional and, therefore result in an increase in frequency (Thewissen & Nummela, 2008:176). The perceived pitch of the sound is then higher than the actual pitch of the source (Subrahmanyam & Lal, 1999:181). When the sound source moves in the direction opposite to the direction of propagation, an expansion of the wavelength occurs resulting in a decrease in frequency. Therefore, the perceived pitch is lower than the actual pitch of the source (Subrahmanyam & Lal, 1999:181).

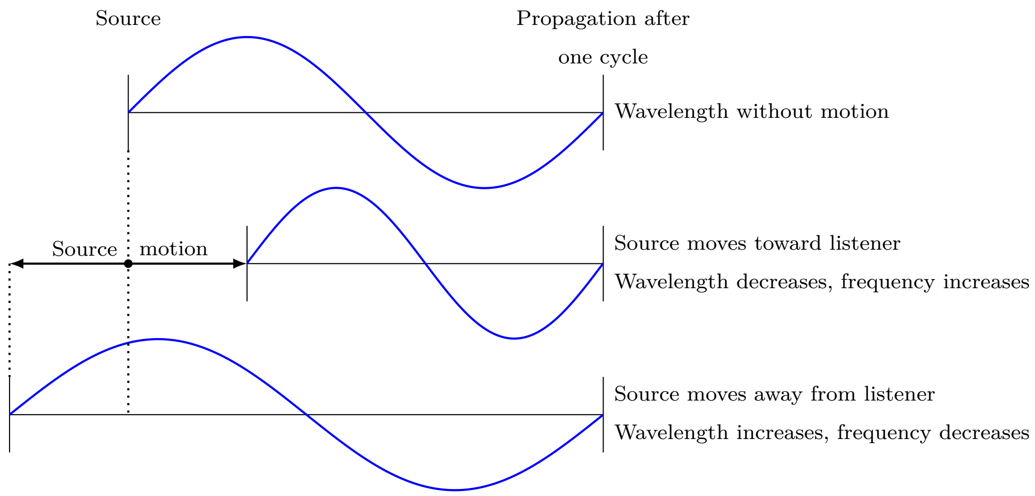


Figure 1.4: The doppler effect (Watkinson, 1998:94).

1.4 Inverse-square Law

The energy from a sound source is distributed uniformly over the surface of a sphere and intensity is calculated as the sound power output of the source divided by the surface area at any radial distance from the source (Toole *et al.*, 2002*a*:1.21). The inverse-square law describes the mathematical relationship between intensity and distance (Eiche, 1990*a*:28) and states that intensity is inversely proportional to the square of the distance between the source and observer (Fisher, 2012:43). This means that sound level decreases substantially as one moves away from the source (Barron, 2009:20), a decrease of approximately 6 dB per doubling of distance from the source (Alten, 2012*a*:117). Since the area of a sphere is $4\pi r^2$ (Bird, 2010:173), the relationship between the intensity measured at some point and the intensity at the reference point is given by Equation (1.4.1), where r is the radius from the sound source (Toole *et al.*, 2002*b*:1-21).

$$\frac{I_1}{I_0} = \frac{r_0^2}{r_1^2} \tag{1.4.1}$$

Substituting Equation (1.4.1) into Equation (1.2.2), gives the decibel intensity ratio relating to the radius from the source, as shown in Equation (1.4.2).

$$\begin{aligned} dB &= 10\log_{10}\frac{I_1}{I_0} \\ &= 10\log_{10}\frac{r_0^2}{r_1^2} \\ &= 20\log_{10}\frac{r_0}{r_1} \end{aligned} \tag{1.4.2}$$

The inverse square law is only valid in free field conditions, where there are no obstructions or reflective surfaces. For a sound source on the floor, energy is only radiated into the upper half of the space. In such a case, the intensity at any given distance from the source is double that given by Equation (1.2.1) on page 4, because the power that is emitted toward the floor is reflected upwards (Hartmann, 1997:36). This directivity of a sound source may be quantified by the *directivity factor*, Q.

The directivity factor is the ratio of the sound intensity in a given direction measured at a certain distance from the source, divided by the average sound intensity over all directions at the same distance. The directivity factor is expressed in Equation (1.4.3), where I_{avg} and p_{avg} are values obtained from an imaginary omni-directional source emitting the same power (Peters *et al.*, 2013:40). Normally, the directivity factor is an increasing function of frequency (Hartmann, 1997:37).

$$\begin{aligned} Q &= \frac{I}{I_{avg}} \\ &= \frac{p^2}{p_{avg}^2} \end{aligned} \tag{1.4.3}$$

In the above-mentioned case, the directivity factor has a value of 2. For a source halfway along the junction of a wall and a floor, the directivity factor is 4 and for a source in the corner of a room, at the junction of three reflecting planes, it has a value of 8 (Peters *et al.*, 2013:42). Incorporating the directivity factor into the inverse-square law yields Equation (1.4.4).

$$I = Q\frac{P}{4\pi r^2} \tag{1.4.4}$$

From Equation (1.4.4) it can be seen that the directivity of the source due to a reflective plane does not violate the inverse-square law. The intensity is still inversely proportional to the distance from the source to listener, however this relationship is different for different directions (Hartmann, 1997:37). The loudspeaker is an example of a sound source that is characterized by its directionality in the forward direction. Another common example is the human mouth (Hartmann, 1997:37).

1.5 Sound Fields

Sound behaves differently in enclosed spaces than outdoors and its propagation can be divided into two regions: the *near field* and the *far field* (Cowan, 2007:339). The far field can further be divided into the *free field*[6] and the *reverberant field* (Sandlin, 2000:324). Where these fields occur are illustrated in Figure 1.5, where r is the distance from the source. The size of these fields depends on the dimensions of the environment, the reflective properties of the surfaces and the frequency of the sound (Cowan, 2007:389).

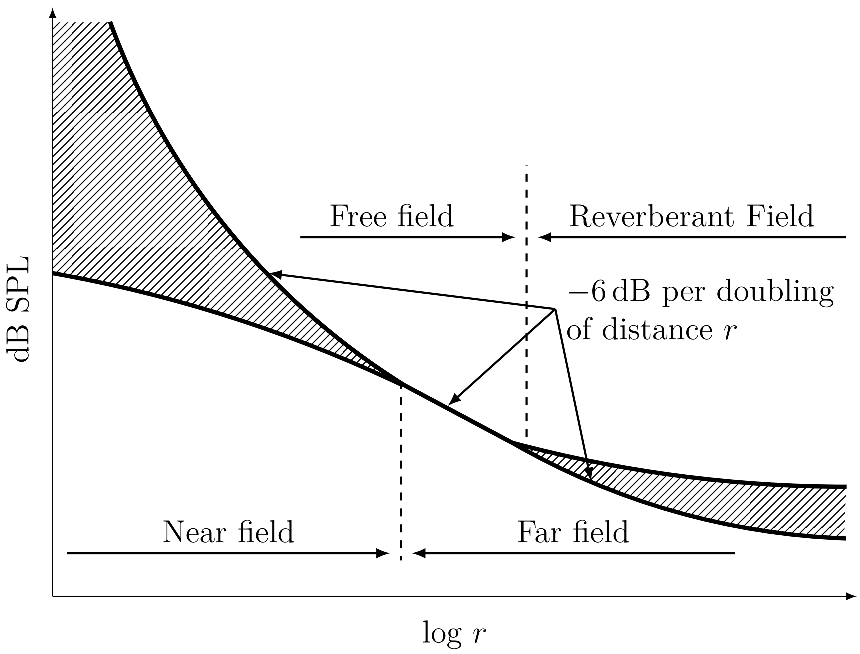


Figure 1.5: Variation of sound pressure level in an enclosed space with increasing distance from the source (Driscoll & Royster, 2003:339).

A sound field may be defined as any area in which sound waves are present (Sandlin, 2000:324). The near field is the region very close to the sound source

[6] Alternatively known at the *direct field* (Sujatha, 2010:387).

(Newell, 2003:597), usually within one quarter of the wavelength of the lowest frequency of interest (Cowan, 2007:390). In this field, sound waves travel directly from the source to the receiver, however, the sound waves radiating from the various parts of the source have not yet combined into a uniform sound field (Driscoll & Royster, 2003:340) and sound pressure levels fluctuate drastically with small changes in distance (Cowan, 1993:53). Therefore, the relationship between the sound pressure level and distance is unpredictable. In Figure 1.5, fluctuations in sound pressure level are illustrated by hatched areas.

In the far field, the sound pressure level becomes uniform and predictable as the individual sound waves combine into a regular pattern of propagation (Cowan, 2007:340). The free field forms part of the far field. A sound field is said to be free field when it is uniform, free from boundaries and undisturbed by other sound sources (Singh:8.12). Under such conditions there are no reflections (Hartmann, 1997:36) and sound radiates equally in all directions according to the Inverse-square law (Pelton *et al.*, 2007:407). Therefore, it experiences a level decrease of 6 dB per doubling of distance from the source (Grondzik *et al.*, 2011:775). Free field conditions exist in large open outdoor spaces or in rooms with highly absorptive surfaces and in which there are no obstructions in the path between the source and the listener (Cowan, 2007:390).

The reverberant field is a sound field which has experienced multiple reflections (Watson & Downey, 2008:72). As the sound pressure levels decay from sources within a room, they drop to a relatively constant level where no one reflection is distinguishable from the rest (Holman, 2010:15). This creates a *diffuse field*. The size of the diffuse field is dependent on the size and reflective characteristics of the room. Within the diffuse field, sound pressure levels remain constant, independent of location (Grondzik *et al.*, 2011:775).

The boundary between the free field and reverberant field is dependent on the reflective characteristics of the surfaces in the room. Therefore, the transition between the two sound fields does not occur at a fixed distance from the source (Driscoll & Royster, 2003:340). Most indoor spaces do not have such a high level of reflection that a diffuse field is created. Instead, there is a near field close to the source, a free field beyond the near field and a reverberant field near the walls (Grondzik *et al.*, 2011:775).

1.6 Direct Sound, Early Reflections and Reverberation

The auditory event perceived by a listener is not only determined by the signal emitted by a sound source, but also by a variety of physical parameters of the environment. These parameters may include the position, orientation and directional characteristics of the sound source and the listener, but also the geometric and acoustic properties of surrounding objects (Lehnert, 1993:1), or spatial attributes. Any sound that reaches a listener can be split into direct sound, early reflections and reverberation. Direct sound travels directly from the source to the listener. Early reflections arrive at the listener after having first traveled to some reflecting surface and reverberation arrives after having undergone several successive reflections (Schissler & Manocha, 2011:1, Davis *et al.*, 2013:184).

The direct sound has very little time to degrade and is, therefore, the purest form of the signal to occur in any sound event (Biederman & Pattison, 2013:70). It has the subjective effect of enhancing the clarity of the sound (Vince & Earnshaw, 2000:116).

Early reflections arrive at the listener very shortly after the direct sound (Peters *et al.*, 2013:128) and consists of a number of discrete reflections of lower amplitude (Dejonckere, 2001:45). Having not been reflected more than once, they retain most of their original sound power (Biederman & Pattison, 2013:71) and provide the listener with auditory cues to the size of the environment (Schissler & Manocha, 2011:1). Reflections that reach the ear within 20 to 30 milliseconds are perceived by the ear-brain system as being part of the direct sound (Peters *et al.*, 2013:128; Cavanaugh *et al.*, 2010:249). Increased early reflections has the same effect as increased direct sound and results in increased intelligibility (Toole, 2008:253). According to Mapp (2008:1394), opinions on how exactly the direct sound and early reflections integrate are somewhat divided, but it is generally agreed upon that early reflections that arrive later than 50 ms after the direct sound are likely to be perceived as disturbing echo's (Gade, 2007:329) and degrade the intelligibility of the sound, with greater effect as the delay increases. Fusing with both the direct sound and reverberation, early reflections serve as a smooth and continuous bridge between the two sonic events (Blesser, 2007:156).

Reverberation includes all the reflections that arrive 80 ms after the direct sound (Beranek, 2004:23). Unlike direct sound and early reflections which reach the listener from distinct directions, the reverberation is fairly diffuse and is not restricted to any specific direction (Sivonen & Ellermeier, 2010:180). The length of the audible decay of the reverberation is subjectively perceived as the reverberation duration (Ahnert & Steffen, 2000:153) and is described by the *reverberation time*, denoted R_{60} or τ (Cowan, 2007:394). The reverberation time is the measure of the time it takes for the sound to decay by 60 dB (Toole, 2008:44). Mathematically, the reverberation time is directly proportional to volume and inversely proportional to the average absorption coefficient of the room, as well as the total surface area of the room (Verma, 2013:507). Wallace Sabine[7] (1868-1919) was the first to discover this relationship and produce

[7] Sabine discovered this relationship when he undertook to improve the intelligibility of speech in the Fogg Art Museum lecture hall at Harvard college. Sound in this room

Equation (1.6.1) (Rothenberg, 2001:488). Here, k is a proportionality constant, whose value depends on the units in which the length is measured (0.05 for feet and 0.162 for meters), V is the volume of room, A is its total surface area and $\bar{\alpha}$ is the average absorption coefficient (Verma, 2013:507). The reverberation time is an extremely useful physical quantity that represents the properties of large spaces in a simplified way (Kim, 2010:312).

$$\tau = \frac{kV}{\bar{\alpha}A} \tag{1.6.1}$$

The *direct-to-reverberant ratio* (DDR) is a ratio of the direct sound measured somewhere in the room and the reverberant sound, which is assumed to be constant throughout the room. A high DDR results in good speech intelligibility (Foreman, 2008:1246).

1.7 Absorption

When sound waves fall on a surface, some of the energy is reflected and some of it is absorbed (Rossing *&* Fletcher, 2004:263). The *absorption coefficient* denotes a material's ability to absorb sound energy (Tocci, 1998:994). The absorption coefficient, α_θ, is the ratio of sound energy absorbed by the surface to sound energy incident on the surface at a given angle, θ. The *statistical absorption coefficient*, α, denotes the ratio of sound energy absorbed to sound energy incident in a perfectly diffuse field (Rossing *&* Fletcher, 2004:263). In either case, the value of the coefficient ranges from 0 to 1.0.

A small absorption coefficient implies that the incident sound is reflected, whereas a large coefficient implies that the sound energy is either dissipated within the material as heat, or transmitted through it (Norton, 2003:284). Materials considered to be sound absorbing have absorption coefficients of 0.5 or greater.

The average absorption coefficient, $\bar{\alpha}$, may be calculated by dividing the sum of the absorption of each surface in a room with the sum of the area of each surface, where absorption is the product of the area of a surface, and its absorption coefficient. This is expressed in Equation (1.7.1), where A_n is the area and α_n is the absorption coefficient of the n^{th} surface of the room, respectively.

---

would persist for over 5 seconds (Daintith *et al.*, 1994:783). Sabine's solution was to place cushions in the lecture hall. When 550 cushions had been arranged on the platform, seats, aisles and rear wall, the reverberation time decreased to about 1 second (Long, 2006:300)

$$\bar{\alpha} = \frac{(A_1\alpha_1 + A_2\alpha_2 + \ldots + A_n\alpha_n)}{(A_1 + A_2 + \ldots + A_n)} = \frac{\sum_{i=1}^{n} A_n\alpha_n}{\sum_{i=1}^{n} A_n} \tag{1.7.1}$$

The performance of sound absorbing materials differ with frequency, therefore the absorption coefficients are usually determined through measurements and presented in octave frequency band (Tocci, 1998:994).

1.8 Reflection

The physical laws regarding reflected sound are analogous to that of optics. Just as light bounces off a mirror, sound waves have equal angles of incidence, θ_i, and reflection, θ_r (Cowan, 2007:387), as illustrated in Figure 1.6, where d is the distance between the source and the wall. The reflected sound acts as though it originated from a virtual sound image. This image is located equidistant from the wall as the original source, but on the opposite side (Everest & Pohlmann, 2009:95). Acoustically reflective surfaces are typically smooth and hard (Cowan, 2007:387).

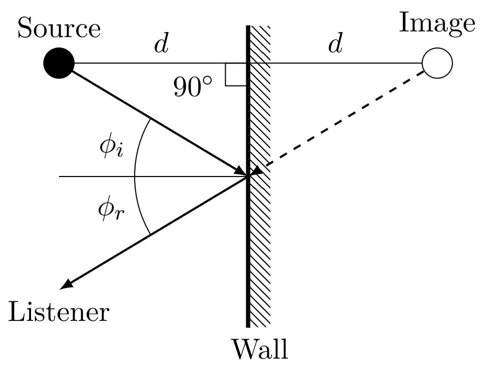


Figure 1.6: Reflection of sound on a hard smooth surface (Everest & Pohlmann, 2009:96).

When two acoustically reflective surfaces are parallel, *standing waves* may occur. If the wavelength of a sound that travels between these surfaces is such that multiples of the half-wavelength correspond exactly with the distance between the reflective surfaces, an acoustic resonance may develop (Talbot-Smith, 2012:29). The reflected waves cancel and build on each other in such a way that a stationary pressure pattern is formed between the two walls, called a standing wave (Cowan, 2007:387). Within a standing wave, a *node* is a point of minimum pressure and an *antinode* is a point of maximum pressure (Talbot-Smith, 2012:30).

Standing waves between two parallel surfaces occur at frequencies, f_n, as shown in Equation (1.8.1), where $n = 1, 2, 3...$, c is the speed of sound, d is the distance between the surfaces (Cowan, 2007:389). Standing waves occur predominantly at low frequencies and can cause coloration of sound quality in small, untreated rooms (Atkinson, 2013:12).

$$f_n = \frac{nc}{2d} \tag{1.8.1}$$

The geometry of a boundary can have a profound effect on the behavior of the incident sound wave (Self, 2009:15). Parallel incident sound waves reflected from a flat surface experience the same angle of reflection, as illustrated in Figure 1.7.

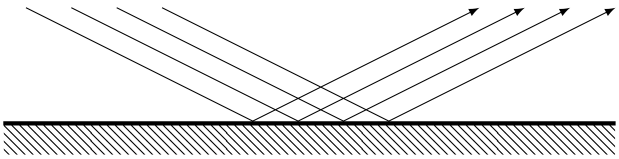


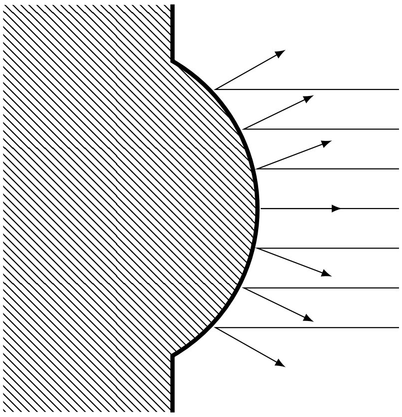
Figure 1.7: Parallel reflections.

Reflection of sound on solid convex surfaces tend to disperse sound energy in many directions, as illustrated in Figure 1.8a (Baukal, 2004:256). In contrast, concave surfaces focus incoming sound energy, as illustrated in Figure 1.8b. Concave reflective surfaces may focus sound in certain areas and defocus sound from others, causing hot spots where sound is concentrated and dead spots where sound cannot be heard (Cowan, 2007:395).

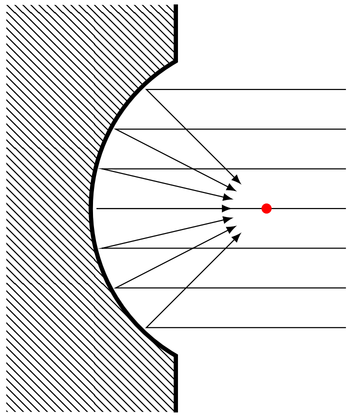
Considering sound as rays is a simplified view. Spherical waves emitted from a point source become plane waves at greater distances from the source. Therefore, incident sound waves on various surfaces may be thought of as plane wavefronts and each ray should be considered as a beam of diverging sound with a spherical wavefront to which the inverse square law applies (Everest & Pohlmann, 2009:97-98).

1.9 Psychoacoustics

Psychoacoustics is the study of the relationship between the experience of hearing and the hearing mechanism (Harris, 1961:3). It covers a multitude of topics concerned with the human perception of sound and includes subjects such as pitch, loudness, localization of sound sources and masking effects (Eargle, 2002*b*:28, Rumsey, 2011:758). These mental impressions are however difficult or even impossible to measure objectively, therefore psychoacoustics is



(a) Convex surface.



(b) Concave surface.

Figure 1.8: Reflections from convex and concave surfaces (Self, 2009:15).

said to deal with the subjective responses to objectively measured phenomena (White & Louie, 2005:9).

1.9.1 The Ear

The ear consists of three subdivisions: the *outer*, *middle* and the *inner ear* (Roads, 1996:1058). The outer ear includes the visible part of the ear, the *pinna*, as well as a passage that travels into the ear (Raichel, 2006:213) and terminates the eardrum (Dugan, 2003:15). It acts as a collector and predominantly funnels sound from the front toward the eardrum (Maltby, 2007:211), which vibrates in sympathy with the incoming sound (Atkinson, 2013:17). The middle ear consists of a series of small bones in a lever arrangement that amplify the incoming sound waves and transmit them to the inner ear (Pastorino & Doyle-Portillo, 2005:124). It also acts as an impedance[8] matching transformer (Stach, 2008), between the low impedance of sound propagation in air and the high impedance liquid-filled inner ear (Kollmeier, 2008:150) to ensure optimal energy transfer between the two media (Haines & Ard, 2013:288).

The inner ear consists of the *vestibular apparatus* and the *cochlea* (Chiras & Chiras, 2013:266) of which the vestibular apparatus is responsible for providing a sense of balance and information regarding the motion and position of the head (Haghgooie *et al.*, 2008:551). The cochlea is a coiled, flexible tube that is closed at one end and filled with fluid (Pastorino & Doyle-Portillo, 2005:103). Down its center runs the *basilar membrane*, to which are attached thousands of hair cells (Loy, 2011:153). Vibrations that travel through the fluid in the cochlea stimulate the hair cells, which send corresponding signals to the brain along the auditory nerve (Wright, 2000:82; Starr *et al.*, 2009:498).

[8] In terms of sound propagation, impedance relates to the ease with which sound propagates through a medium (Stach, 2008:60).

The shape of the basilar membrane enables it to perform frequency discrimination and make sense of complex sounds by reducing them to simpler components (Benson, 2003:44). A basic assumption is that the human auditory system does this like a bank of overlapping filters (Chalupper, 2000:2). This hypothesis was originally formulated by Hermann von Helmholtz (1821-1894) and has strongly influenced the thinking and experimental design of psychoacoustics (Moller, 1982:192, Schultz & Schultz, 2011:55). The individual filters act over critical bands, each with a bandwidth of nearly one third an octave (Rabek, 1965:1).

1.9.2 Masking

Masking is the subjective phenomenon wherein the presence of one sound inhibits the ability to hear another sound (White & Louie, 2005:230). The component which causes the masking is known as the *masker*, while the masked sound is known as the *maskee* (Dejonckere, 2001:43). The level by which the maskee needs to be increased to still be heard in the presence of a masker is known as the *masking level*. Simplified, one can regard the masking level as the shift of hearing threshold in the presence of another sound, such as a tone or noise. Masking may occur in both the time and frequency domains (Kleiner, 2011:64).

Individual frequency components of a complex sound may mask other components within the sound. The masking effect is greater for maskees whose frequencies are above the frequency of the masker as well as for maskers of a higher level (Dejonckere, 2001:43). However, masking is greatest when the masker and maskee are close to one another in frequency (Kleiner, 2011:64).

For sounds that are not simultaneous, *forward* and *backward masking* may occur (Howard, 2009:263). Forward masking is encountered when the masked signal remains inaudible for a time after the masker has ended. Backward masking occurs when the masked signal becomes inaudible before the masker begins. Specific manifestations of masking depends on the spectral composition of both the masker and the masked signal, as well as their variations as a function of time. Forward masking may continue for a considerable time after the masker has decayed, but backward masking may only be effective for less than 2 ms to 3 ms before the onset of the masker (Hanzo *et al.*, 2008:473).

1.9.3 Aural memory

Aural memory consists of *echoic memory*, *short term memory* and *long term memory*. After the process in the peripheral auditory system, the first element of the auditory memory is echoic memory (Côté, 2011:10). Current theory suggests that the sensation in the echoic memory are not categorized in any way and persist as raw, continuous sensory data (Snyder, 2000:4). The exact duration of the echoic memory is unclear, but is generally predicted to last 200 ms to 300 ms;

Short term memory refers to the very recent auditory past. It may occasionally last as long as 10 to 12 seconds, but on average lasts only 3 to 5 seconds (Snyder, 2000:50). The short term memory is referred to by psychologists as the *working memory*, because it is the memory system that is used to hold and manipulate information for a brief period of time (Nevid, 2008:212). A metaphor to explain short term memory is that of reverberation. It is a pattern of recirculating electrical energy that reverberates through reentrant loops of neural circuitry in the brain, sustaining the current pattern of activity. Like actual reverberation, if new energy is not introduced into the process by some form of rehearsal, it fades away. The time limit refers to how long the recirculating pattern can be maintained without rehearsal (Snyder, 2000:47).

The long term memory is a more durable memory where elements are stored for periods from a few days up to several decades. Numerous perception processes make use of the long term memory, such as the recognition of musical instruments, identification and localization of sound sources and the identification of spoken words (Côté, 2011:10). Short term memory information is transferred to the long term memory through rehearsal (Nevid, 2008:217).

1.9.4 Localization

Spacial perception is the capacity of the auditory system to interpret or exploit the differential paths by which sounds may reach the head. Through this, the auditory system can determine the location of a sound source or unmask sounds that would otherwise be obscured by noise (Culling & Akeroyd, 2010:123). Exactly how humans are able to localize sound is still not fully comprehended (Bou Saleh *et al.*, 2007:1). The most well known spatial cues used by the human auditory system is the *inter-aural time difference* (ITD), *inter-aural level difference* (ILD) and monaural spectral cues (Courtois *et al.*, 2014:1-2).

Both the ITD and the ILD is dependent on binaural hearing. The ITD is the difference between the arrival time of the sound waves reaching the left and the right ear of the listener (Brian *et al.*, 2005:1). Both ears are not in the same space and therefore, unless the sound source is equidistant form both ears, the path lengths to each ear is different (Braasch, 2005:76). Low frequency sound waves, because of their large wavelengths, are not easily disturbed by the presence of the head and reach the other ear with the same level and intensity, but with a delay, making the ITD an effective cue for locating low frequency sounds (Goldstein, 2013:291).

Inter-aural level differences are primarily brought about by the head casting an acoustic shadow over the far ear, which results in an attenuation of the sound level. The resulting shadow is dependent on the angle of incidence, the size of the head and is proportional to the frequency of the sound (Eddins & III, 2010:137). High frequency sounds are attenuated more (Yost, 2008:442), but the effects are minimal for low frequency sounds for which the wavelength is longer than the head (Palmer, 1995:105).

Both the ITD and the ILD contribute in combination to the localization of sound (Pizzi, 1984), with the ITD providing low frequency location cues and the ILD providing high frequency location cues. The time and level differences provide information for locating sources on the horizontal plane, but provide ambiguous information on the elevation of a source (Goldstein, 2013:293). Elevation information is provided by monaural spectral cues, which arise from the direction dependent filtering functions of the pinna (Eddins & III, 2010:138). These filtering effects modify the signal's frequency spectrum (Bloom, 1977:560) and are a byproduct of the complex delay-and-add network produced by the individual ear-specific convolutions of the pinna. They are also coloured by direction-dependent features of the head, neck and torso (Eddins & III, 2010:138). Psychoacoustic studies suggest that specialized neurons in the central auditory pathways are capable of decoding the directionality of this spectral information (May, 2010:304). Additionally, asymmetry of the ears aids in telling if a sound is coming from the front or the back (Harris, 1961:3).

1.9.5 Timbre

The first research on timbre[9] was performed by the German physicist Hermann Helmholtz in 1863 (Roger *et al.*, 1998:739). Helmholtz devised a way to determine the harmonics in sustained musical notes by using glass resonators. He concluded that the timbre of a periodic tone is determined by the relative amplitudes of its various harmonics (Kammler, 2007:707). Since the mid-nineteenth century, timbre has become increasingly important in the Western sensory experience of classical and popular music (Greene & Porcello, 2005:157).

Timbre is an audio quality distinct from pitch, intensity and perceived loudness (Wessel, 1979:45; Smith, 2009:301), that distinguishes one instrument from another operating at the same pitch and loudness (Kroon, 2010:692). It is an auditory perception and cannot be described by any single acoustic property (Sethares, 2005:11; Klapuri, 2006:8).

A simple sound has a single frequency regardless of amplitude and is sometimes denoted as a *pure tone*. A simple sound is produced by a single vibrating mass that produces an audible sound. However, most vibrating sources are more than just a single mass and produce complex sounds (Hersh & Johnson, 2003:9). A complex sound consists of a *fundamental*, also known as the *first harmonic*, and a series of harmonics at higher frequencies that are integer multiples of the fundamental frequency (Plack, 2014:17). The fundamental determines the pitch that is heard, as well as the period of the sound wave

[9] French for sound, sound colour or bike bell (Bader, 2013:80). Timbre Synonyms include tone, tone colour and texture.

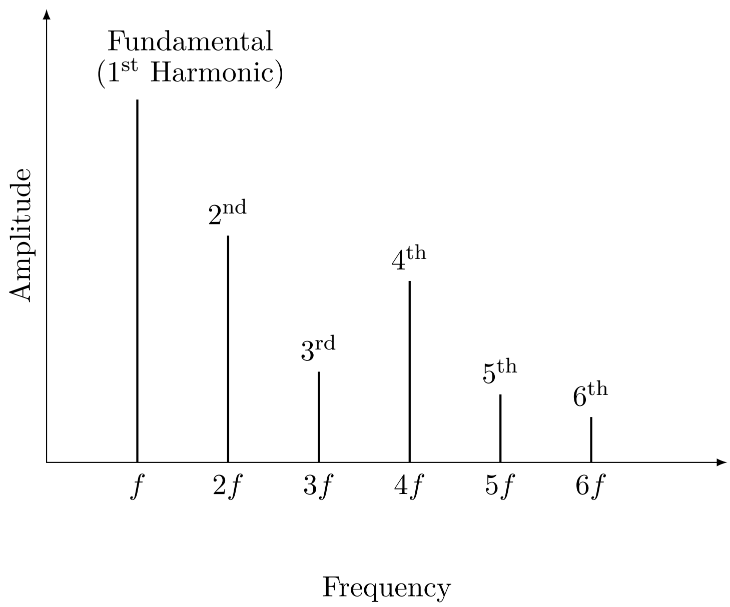


Figure 1.9: Graphical representation of the harmonics of a complex music signal (Thompson, 2005:114).

(Thompson, 2005:114). A graphical representation of a complex music signal is shown in Figure 1.9.

The French mathematician Jean Baptiste Josef Fourier (1768-1830) proved that any periodic signal can be decomposed into an infinite series of sine waves, each at an integer multiple of the fundamental frequency (Montrose & Nakauchi, 2004:13). These frequencies that are whole-number multiples of the fundamental are called harmonics (Huber & Runstein, 2010:52). Just as light may be decomposed into a spectrum of colours, an audio signal contains energy that is distributed over a range of frequencies (Schilling & Harris, 2011:60). The composition of the harmonics is determined during a mathematical operation known as the *Fourier Transform*, in which the signal is transformed from the time-domain to the frequency-domain (Montrose & Nakauchi, 2004:13).

The human ear can perform such a spectral analysis (Yang *et al.*, 2005:63) and can follow changes in a large number of harmonic frequencies (Russ, 2009:41). The timbre of the sound is then determined by the relationship between the level of the fundamental, the levels of the harmonics (Russ, 2009:41) and how the relative amplitudes of those harmonics change over time (Kroon, 2010:692). However, as the frequency increases, the ear's ability to discriminate phase diminishes and the number of harmonics that can be distinguished decreases (Russ, 2009:41). The frequency range of vision is a little less than one octave, but within this range seven million colours may be identified. Given that the audible frequency range is nine times greater, one can imagine how many sound timbres might be possible (Rossing, 2007:2).

Even though timbre cannot be described by any single acoustic property, it depends mainly on the coarse spectral energy distribution of a sound and its evolution over time (Klapuri, 2006:8). The time envelope refers to the overall shape of the sound waveform, representing amplitude change over time (Goldstein, 2010:1003). A short recording of a chord played on piano was made. The waveform of the sound is shown in Figure 1.10, which shows the decrease in amplitude with time.

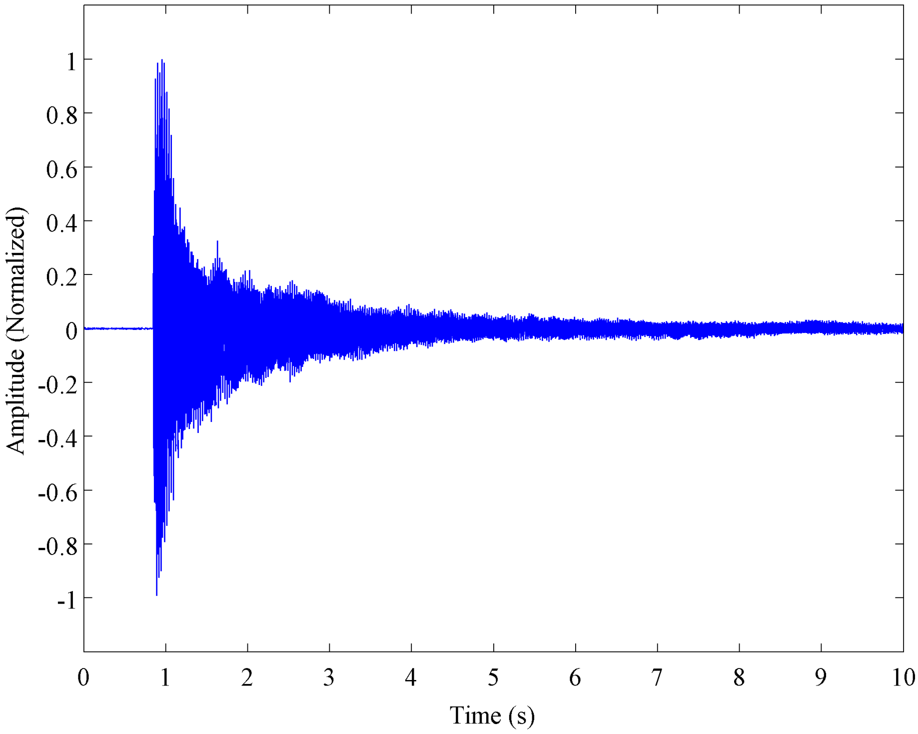


Figure 1.10: Piano chord waveform.

In addition to the envelope, musical sounds contain overtones whose respective strengths change over the duration of the sound. This dynamic spectral variation is known as *spectral flux*. According to Goldstein (2010:1003), both the time envelope and the spectral flux affect the perceived timbre of a sound. To demonstrate spectral flux, a spectrogram of the piano recording is shown in Figure 1.11. A *spectrogram* is a representation that describes the energy content of a signal segment as a function of time and frequency (Dowla & Rogers, 1995:115). Time is represented by the horizontal axis and frequency by the vertical axis (Johnson, 2011:78). The magnitude of a frequency at any given time is represented by the colour of that point (Cai, 2014:87). The colour bar shows the colour range that corresponds to the dynamic range of the signal. In this spectrogram the smallest and greatest magnitudes are represented by the colours blue and brown, respectively. Figure 1.11 shows that the decay is not constant across the frequency range and that some frequencies decay quickly, while others sustain for much longer periods.

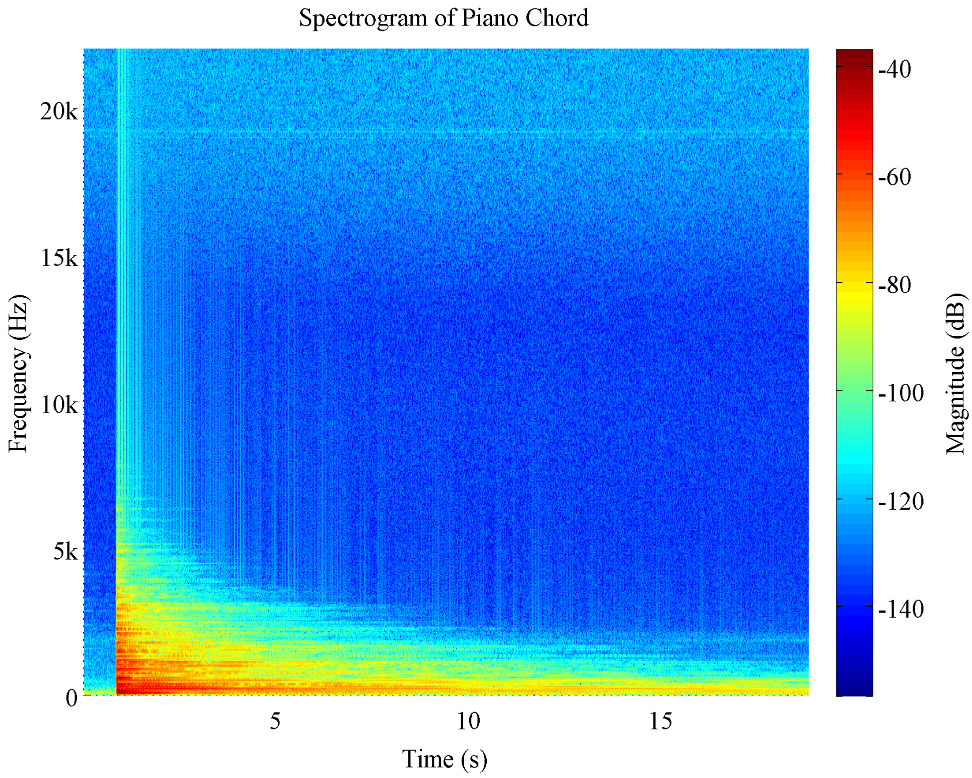


Figure 1.11: Spectrogram of a piano chord.

Due to its subjective nature, Streicher (1990:125) describes timbre as the most difficult attribute of sound to record and reproduce effectively. Most terms used to describe timbre are based on analogies between musical sound and everyday physical and sensory experience. These include terms such as *nasal, dark, mellow, strained, rough, soothing* and *grating*. Timbre is also often described in terms of the material of the sound source, its vocal quality, its resemblance to musical instruments or its sensory or emotional effect (Goldstein, 2010:1002). These terms are highly subjective when applied to music, but are nevertheless helpful in describing timbre (Miller & Shahriari, 2013:18).

Chapter 2

Studio Signal Chain

> *"...there is no substitute for an intimate and detailed knowledge of the characteristics of all the equipment to be used by the recordist. Microphones, for instance, are notoriously unpredictable...you must know how a microphone sounds to be able to use it to best advantage. The same may be said about the rest of the links in the recording chain."* - Robert Fine (White, 1977:1)

In a recording or live sound environment, the sound travels from the source through a range of equipment and processes before it eventually reaches the listener. The entire path that the sound travels is called the signal chain (Evans, 2011:7), because the various pieces of equipment are linked together (Sauls & Stark, 2013:22). As illustrated in Figure 2.1, a signal chain typically consists of an input, preamplifier, outboard equipment, converter, audio interface, computer, storage media and monitors. In between all of these exist various types of lines, cables and connections through which signals are transmitted. Depending on the equipment, the preamplifier, converters and audio interface may be separate and modular or contained in an all-in-one system. Similarly, the on board storage of the computer may be used as the primary storage device, or the computer may be completely omitted and the audio from the audio interface stored directly on some external storage media. In Figure 2.1, the signal chain for a modular system is illustrated in green, while the signal chain for an all-in-one system is shown in blue. Gottlieb (2007:5) believes a shorter signal chain allows for greater signal quality, as it minimizes the risk of distortion and circumvents potential problems before they occur.

The path that the input travels is determined by its level. Microphones and instruments with pick-ups have a very low output, ranging from $-60\,\mathrm{dB\,V}$ to $-20\,\mathrm{dB\,V}$ and $-30\,\mathrm{dBV}$ to $-20\,\mathrm{dBV}$ respectively (Hurtig, 1988:19), and therefore need to be amplified by a preamplifier into line level signals (Zager, 2011:276; Swallow, 2010:131). Line level signals are either $-10\,\mathrm{dB\,V}$ or $+4\,\mathrm{dB\,V}$ (Franz & Lindsay, 2004:12), where $-10\,\mathrm{dB\,V}$ is used for personal

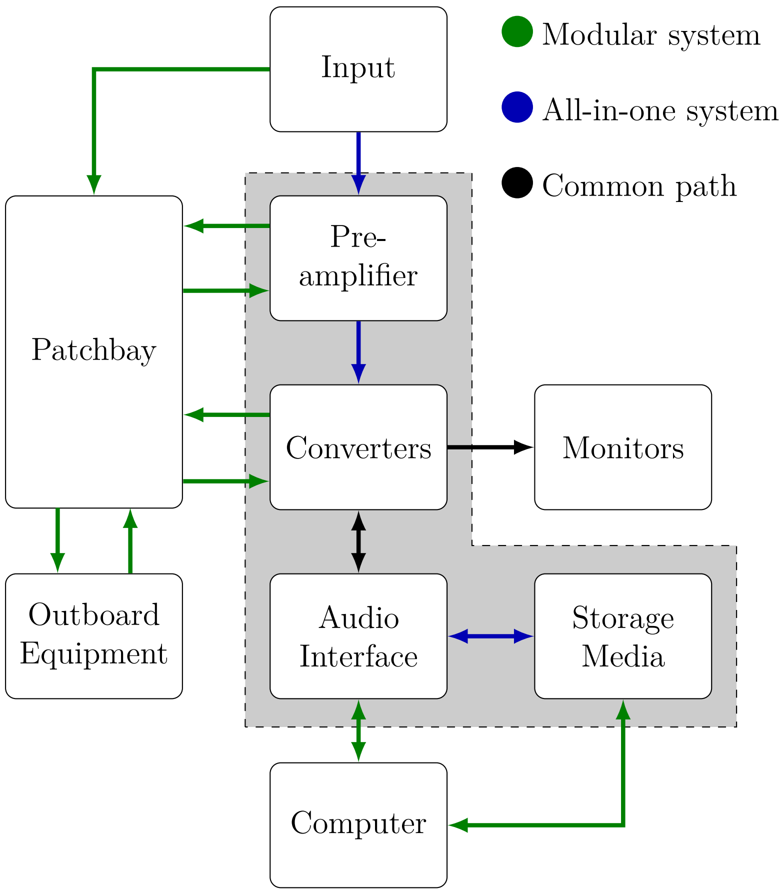


Figure 2.1: Studio signal chain.

recording equipment and home high fidelity (Hi-Fi) gear, where +4 dB V is used for professional equipment (Hurtig, 1988:19).

Equipment for sound origination can conveniently be divided into two categories: primary sources and secondary sources. Primary sources are those which are right at the start of the chain and convert acoustic signals into electrical signals (Talbot-Smith, 2004:397). The microphone is such a source and is the first device that the sound reaches before it is recorded, amplified or broadcast (Josephson, 1997:1). Secondary sources are essentially devices which store the outputs of the primary sources. These include digital and analog devices, such as tape and disc recorders, and sound reproducing equipment (Talbot-Smith, 2004:397-405).

2.1 Microphones

The microphone is the first link in the signal chain. The effectiveness of all other tools and techniques depend upon the quality of the image that the microphone is able to deliver (Zak, 2001:108) For this reason, Hodgson (2010:14) describes microphone selection as one of the most important techniques in all of recording practice. The placement of the microphone is another critial technique for achieving the desired sound. Selection and placement of a microphone depends on an understanding of the microphone characteristics and placements that are suited to the specific situation (Musburger & Kindem, 2012:122) and the required operation and purpose of the microphone during recording (Hegarty *et al.*, 1998:54).

A microphone consists of three main components: a diaphragm, transducer and casing or enclosure (Talbot-Smith, 1999:2.37). The diaphragm moves back and forth in response to changes in the pressure or velocity brought about by a sound wave (Long, 2006:115) and the transducer converts energy from one form into another (Valente, 2002:64). Even though the diaphragm and transducer are separate components, the one may be an integral part of the other or very closely mechanically coupled (Talbot-Smith, 1999:2.38). The casing either partly or fully encloses the transducer and diaphragm. The nature of the casing determines the directivity of the microphone (Talbot-Smith, 1999:2.38).

Microphone types are classified according to their transducers (Sinclair, 2000*a*:215), where the most common types are *moving coil*, *condenser*, *electret*, *piezoelectric* and *ribbon* microphones (Long, 2006:115). Each microphone has its own characteristics, but these do not limit their application. However, for a clean recording free of noise and off-axis coloration, Bartlett (1987:924) suggests the following requirements in a microphone:

- Low self-noise.
- High *headroom* (ability to accept high SPL's without distortion)
- Low pick-up of hum and radio frequency interference
- Low pick-up of mechanical vibration (handling noise)
- Low pick-up of wind noise and *pops* (explosive breath sounds)
- Minimal off-axis coloration
- High sensitivity

2.1.1 Moving Coil

Moving coil, or *dynamic*, microphones have a small self-supporting coil that is fastened to a lightweight plastic diaphragm and suspended in front of a strong permanent magnet (Rumsey & McCormick, 2006:49). When the diaphragm is excited by sound waves, the coil moves to and fro in the presence of the magnetic field of the permanent magnet and a current is induced in its wire that is representative of the sound wave that exited the diaphragm (Chapman, 2005:8). The phenomenon of a current being induced in a conductor when it moves through a magnetic field is known as Faraday's law of electromagnetic induction (Abdo, 2010:271).

This principle requires relative motion between the conductor and the magnetic field (Senty, 2012:3), therefore if the coil is stationary, even if it is displaced from its position of rest, no current is induced (McKenzie, 2013:182). As such, moving coil microphones are only sensitive to rapid changes in pressure and are categorized as pressure microphones (Beranek & Mellow, 2012:200-201).

Moving coil microphones are generally rugged and capable of handling very loud sounds such as drums or guitar amplifiers[1] (Rudolph, 2001:112). They are also relatively inexpensive, resistant to moisture and can potentially handle high gain without feedback, making them microphones ideal for use in live sound (Fraden, 2010:439).

2.1.2 Condenser and Electret

The transducer of a condenser microphone consists of a thin and lightweight metal or metalized plastic diaphragm, located very near and parallel to a perforate backplate. The backplate is electrically conductive and oppositely-charged to the diaphragm (Boré & Peus, 1999:32). Separated by a small amount of air, the diaphragm and backplate form a capacitor[2], also known as a condenser (Davis & Jones, 1989:113). The air acts as an insulating material and is known as a dielectric[3] (Hambley, 2008:92).

The capacitance, C, of the capacitor is proportional to the surface area of the plates and the type of dielectric, but inversely proportional to the distance between the plates as shown in Equation (2.1.1), where A is the area of one plate, d is the distance between the plates and ϵ is the dielectric constant[4] (Hambley, 2008:99).

$$C = \frac{\epsilon A}{d} \tag{2.1.1}$$

The capacitor is incorporated in an electric circuit with a steady DC voltage, V, across its plates, known as the polarizing voltage (Peters *et al.*, 2013:208). This voltage induces a charge, Q, on the plates of the capacitor (Eargle, 2005:26). The distance between the diaphragm and backplate changes in

[1] A guitar amplifier constitutes a system containing an amplifier and one or more loudspeakers. The amplifier and loudspeakers may be a self-contained unit, or contained in separate enclosures, known as the *amplifier head* and *cabinet* (Brosnac, 1987:11). In either case, *amplifier* refers to the whole amplifier-loudspeaker system.

[2] A capacitor is any device that consists of two conducting surfaces separated by an insulating material or dielectric. The capacitance of a capacitor is that property which permits the storage of a charge when a potential difference exits between the conductors and is measured in *Farad* (Bowick, 1982:12).

[3] Other dielectrics used in capacitors include mylar, polyester, polypropylene and mica (Hambley, 2008:92).

[4] The dielectric constant is calculated as $\epsilon = \epsilon_r \epsilon_0$, where ϵ_r is the relative dielectric constant and $\epsilon_0 = 8.854\,188 \times 10^{-12}\,\mathrm{Fm}^{-1}$ is the dielectric constant for a vacuum (Dyer, 2004:359). For air, $\epsilon_r = 1$ (Ballou, 1997:16).

response to pressure variations on the diaphragm, caused by sound waves (Eren, 2003:72), which results in a change in the capacitance (Busch-Vishniac & Hixson, 1998:1412). The charge, Q, remains unchanged, but the voltage, V, changes in accordance with Equation (2.1.2) (Eargle, 2005:26).

$$Q = CV \tag{2.1.2}$$

The output signal of the capacitor is very low and has a very high impedance. Therefore, it requires an internal preamplifier to amplify the signal to a usable level (Alten, 2011:70). Without this preamplifier, the signal would dissipate before it reaches its destination (Lubin, 2010:20). The preamplifier also supplies the polarizing voltage and matches the impedance of the capacitor output to the low-impedance balanced output of the microphone (Ballou, Ciaudelli & Schmitt, 2008:510). As a result, capacitor microphones require external power (Petelin & Petelin, 2005:229).

The perforations in the backplate of the capacitor act as pressure equalizers and ensures that the static pressure on either side of the diaphragm is the same (Peters *et al.*, 2013:208). As a result, condenser microphones are also categorized as pressure microphones (Francis, 2009:146), because like dynamic microphones, it is only sensitive to changes in pressure.

Electret microphones work on the same principle as condenser microphones, but differ in that the polarizing voltage is stored permanently (Ewing, 1997:624). The electret is a dielectric material, capable of permanently sustaining an electric polarization (Lerch, 2008:608). The diaphragm typically consists of a thin polymer film, with one side coated in metal (Busch-Vishniac & Hixson, 1998:1413). The polarizing voltage is usually applied to the diaphragm during manufacture (Atkinson, 2013:52). An electret microphone itself does not require external power to produce a polarizing voltage, however an impedance matching circuit inside the microphone does require some power, which is usually supplied by external power or an internal battery (Rising, 2012:279; Zager, 2011:275).

2.1.3 Piezoelectric

Piezoelectric microphones contain piezoelectric crystals which act as the transducer. The crystal is either attached to a diaphragm or in some cases serves as both the transducer and diaphragm (Busch-Vishniac & Hixson, 1998:1413). When deformed by impinging sound waves, the crystals form a surface charge that is proportional to the force bringing about the deformation (Figliola, 2011:391). This generation of an electric charge by a crystalline material, upon being subjected to stress, is known as the *piezoelectric effect* (Fraden, 2010:86). Modern piezoelectric materials are mostly synthetic, such as barium titanate, which is used in piezoelectric transducers for frequencies of up to several hundred kilohertz (Sinclair, 2000*b*:125). Other piezoelectric materials are quartz and tourmaline (Earshen, 2003:63).

Piezoelectric microphones are also known as crystal or ceramic microphones. Even though they are not known for their sound quality, they can perform well when properly implemented (Davis & Jones, 1989:116). They have a superior ability to handle high sound pressure levels over extremely wide frequency ranges, without distortion. They are rugged and not very susceptible to moisture, but have relatively low sensitivity and a high noise floor (Earshen, 2003:63). The piezoelectric transducers are high-impedance devices and produce substantial output levels (Eiche, 1990*a*:47), such that they can drive preamplifiers directly (Eargle, 2005:49).

Piezoelectric transducers are used in clip-on lavaliere microphones, guitar pick-ups, drum triggers and hearing aids (Brown *et al.*, 1998:501; Dochtermann, 2011:39).

2.1.4 Ribbon

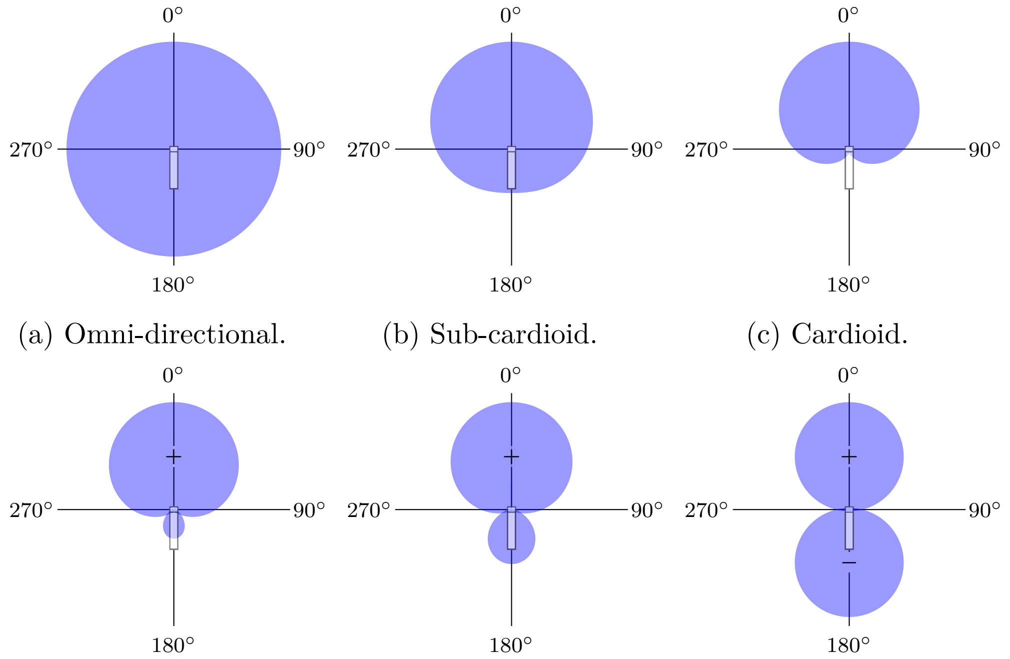
The transducer of a ribbon microphone consists of a thin corrugated ribbon located in a magnetic field, between the poles of a permanent magnet (Talbot-Smith, 1999:2.38). This lightweight ribbon is electrically conductive and doubles as the diaphragm of the microphone (Kleiner, 2011:285; Gupta, 2010:26). The corrugations prevent the ribbon from curling and provide lengthwise flexibility (Kellog, 1967:199).

The ribbon is driven by the difference in pressure, or the pressure gradient, between the front and back of the diaphragm (Eargle, 2005:50). Like moving coil microphones, ribbon microphones operate by electromagnetic induction, therefore its transducer may also be classified as the dynamic type (Fraden, 2010:440; Schroder, 2011:41). When the ribbon is exited, a voltage proportional to the velocity of the ribbon is produced (Olson, 1976:798). Ribbon microphones are categorized as pressure gradient microphones (Gupta, 2010:27). Alternatively, they are known as velocity microphones, since the difference in pressure across the ribbon corresponds to the particle velocity in the sound wave that excites it. Therefore the movement of the ribbon is proportional to the particle velocity in the sound wave (Olson, 2013:332).

Ribbon microphones have a relatively flat frequency response that extends well into the high frequency range (Oswinski, 2009:4). However, ribbon microphones are fragile, 20 to 25 dB less sensitive than condenser microphones, of a non-compact size and expensive to make and repair ((Talbot-Smith, 2004:399); Sujatha, 2010:416). Due to the lower sensitivity, ribbon microphones typically require 10 dB to 30 dB more gain than condenser microphones (Touzeau, 2009:83).

2.1.5 Directivity

The simplest form of a microphone has only one side exposed to the sound field. If such a microphone is sufficiently small, it will respond equally to sound in all directions (Newell & Holland, 2006:383). These microphones have an *omni-directional* polar pattern as illustrated in Figure 2.2a (Sauls & Stark, 2013:67). A polar pattern is a map of the microphone's sensitivity, with regards to the direction from which the sound originates relative to the microphone (Gottlieb & Hennerich, 2009:134). Various common polar patterns are illustrated in Figure 2.2.



(a) Omni-directional. (b) Sub-cardioid. (c) Cardioid.

(d) Super-cardioid. (e) Hyper-cardioid. (f) Bi-directional.

Figure 2.2: Various microphone polar patterns, generated with equations from Eargle (2002*b*:56-58).

Moving coil transducers are inherently omni-directional, but a *cardioid* polar pattern as in Figure 2.2c is achieved by adding an acoustic phase-shifting maze network and ports to the microphone casing (Gottlieb & Hennerich, 2009:135). These cause sound waves impinging from the rear of the microphone to impact both sides of the diaphragm, but with opposite phase, thereby greatly reducing their intensity (Gottlieb, 2007:380). The result is a major reduction of sensitivity for sound approaching from the rear of the microphone, with a minor loss of sensitivity from the sides.

As discussed in the previous section, pressure gradient microphones measure the difference in pressure between the front and rear of the diaphragm (Eargle, 2005:50). When a sound wave impinges from either the front or the rear, it moves the diaphragm in the presence of a magnetic field and a current is induced in it (Holman, 2010:81). However, when a sound wave impinges from the side of the diaphragm, there is no difference in pressure between the front and rear to set the diaphragm in motion. As a result, pressure-gradient microphones have a *figure-eight* or *bi-directional* polar pattern, as illustrated in Figure 2.2f. They are most sensitive to sound coming from the front and rear of the diaphragm and reject sound coming from the sides (Eiche, 1990*a*:50). The front and rear lobes of the figure-eight polar pattern have opposite polarity (Bartlett & Bartlett, 2007:188).

Somewhere between the cardioid and bidirectional polar patterns, there exist a series of intermediate patterns in which the rear lobe of the bi-directional pattern becomes progressively smaller, while the the front lobe takes on a more cardioid shape (Kefauver, 2001:66). These are the *hyper-cardioid* and *super-cardioid* polar patterns shown in Figures 2.2d and 2.2e (Alten, 2012*b*:77). Similar to cardioid polar patterns, these patterns are achieved by introducing an acoustic maze, and result in a very high rejection of off-axis sound (Dejonckere, 2001:171). The front and rear lobes of these polar patterns also differ in polarity.

Some condenser microphones have switchable polar patterns (Bartlett & Bartlett, 1999:66). This is achieved through the use of two identical diaphragms, located on either side of a rigid plate. The plate has perforations, which gives each diaphragm a cardioid polar pattern (Rumsey & McCormick, 2014:59). All the above-mentioned polar patterns may be achieved by electronically manipulating the polarizing voltage on each of the two diaphragms (Atkinson, 2013:42). However, it is up to the manufacturer to decided which polar patterns are implemented in the microphone.

Directional microphones are often used to give the recordist greater control over which sounds are recorded. The main use of directional microphones is to pick up desired sounds and reject unwanted sounds such as reverberation and noise (Olson, 1967:420).

2.1.6 Frequency Response

The frequency response of a microphone is a measure of how consistently it translates a given SPL into an audio signal level at various frequencies (Davis & Jones, 1989:124). According to Rising (2012:284), the ideal frequency response may be ruler-flat, with no peaks or other colourations and a range of 16 Hz to 20 kHz. However, because practical microphones are used in part for their musical or sonic characteristics, they may exhibit certain controlled deviations from a flat response (Davis & Jones, 1989:14). Some manufacturers may deliberately emphasize or cut certain frequencies or enable the user to do so through a switch (Boyce, 2014:43). Frequency response curves tend to taper off toward the upper and lower extremes of the microphone's range, where the response may be quite erratic (Stark, 1996:85). The frequency response of a

Shure SM57 moving coil microphone is given in Figure 2.3. It is common to use a logarithmic scale for frequency and a decibel scale for amplitude as it correlates better with the human hearing mechanism than a linear plot (White & Louie, 2005:166).

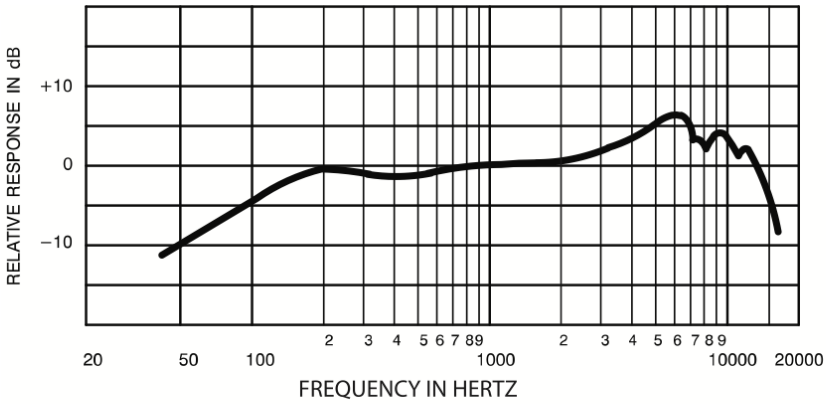


Figure 2.3: Frequency response of Shure SM57 microphone (Shure Incorporated, 2010:12).

2.2 Lines, cables and Connections

"*The proper interconnection of analog audio signals, and an understanding of the principles of balanced and unbalanced lines, is vital to the maintenance of high quality in an audio system, and will remain important for many years notwithstanding the growing usage of digital systems.*" - Rumsey & McCormick (2006:354)

2.2.1 Transformers

A transformer is a device with no moving parts that transfers electric power without any change in frequency, through electromagnetic induction (Pansini, 1999:3; Kulshreshtha, 2009:485). It consists of a pair of windings, primary and secondary, linked by a magnetic circuit or core (Heathcote, 2011:2), as illustrated in Figure 2.4. When an alternating current travels through primary windings, it induces a *magnetic flux* in the core. The magnetic flux refers to the total number of lines of force existing in a particular magnetic field (Bakshi & Bakshi, 2008:1.7). This flux then induces a current in the secondary windings, which delivers power to the load (Hambley, 2008:568).

All values concerning transformers are proportional to the ratio of primary windings to secondary windings, known as the turns ratio (Herman, 2013:320). Mathematically, this law translates into Equations (2.2.1) and (2.2.2), where

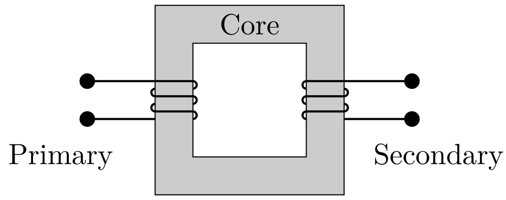


Figure 2.4: Illustration of a transformer (Rumsey & McCormick, 2006:355).

V refers to voltage, I refers to current and N denotes number of turns. The subscripts P and S refer to the primary and secondary windings, respectively.

$$\frac{V_P}{V_S} = \frac{N_P}{N_S} \tag{2.2.1}$$

$$\frac{I_P}{I_S} = \frac{N_S}{N_P} \tag{2.2.2}$$

The transformer is mathematically analogous to a lever, as illustrated in Figure 2.5. Just like the transformer voltages are related to the turns ratio, the velocities at the end of the lever are related to the length ratio, as shown in Equation (2.2.3), where v and l denote velocity and length respectively. The relationship between the forces acting on the lever is similar and is analogous to the relationship between the currents in the primary and secondary windings of the transformer (Hambley, 2008:570). This relationship is expressed in Equation (2.2.4), where F refers to the force acting on the end of the lever.

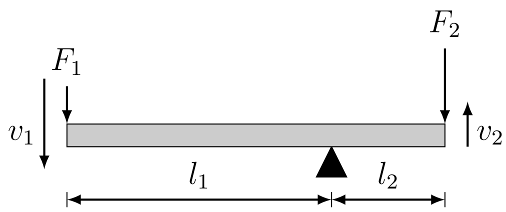


Figure 2.5: Mechanical analog of a transformer (Hambley, 2008:571).

$$\frac{v_2}{v_1} = \frac{l_2}{l_1} \tag{2.2.3}$$

$$\frac{F_2}{F_1} = \frac{l_1}{l_2} \tag{2.2.4}$$

Audio transformers are designed to operate within the 20 Hz to 20 000 Hz audible frequency range (White & Louie, 2005:403). They are used to step voltages up and down, to provide electrical isolation and for impedance matching in microphones (Goldman, 2006:299; Herman & Sparkman, 2009:202). They are also used in microphone splitters, where the secondary windings of a transformer produce a duplicate of the microphone signal (Whitlock, 2008:295), direct inject boxes[5] (Leonard, 2001:51) and to balance lines (White & Louie, 2005:35).

2.2.2 Unbalanced and Balanced Lines

Audio lines are divided into two categories, determined by their configuration, and are termed *unbalanced* or *balanced*. Unbalanced lines have two wires: a center conductor, which carries the signal, and a woven metallic shield which surrounds it and is connected to ground (Utz, 2003:32). Unbalanced lines are prone to picking up noise and interference (Roback, 2004:520) and are typically used in consumer equipment (Wolf & Block, 2013:601).

A common cause of noise in unbalanced lines is the occurrence of *ground loops*. Ground loops occur when a chain of electronic devices are in the vicinity of a power supply that produces an alternating magnetic field. This field can induce minute currents in various cabling between electronic elements, which can cause noise (Eargle, 2002*b*:137).

Balanced lines have two wires twisted together, which are encircled by an outer conductive shield (White, 2014:60). The first wire caries a normal-polarity signal and is termed *positive*. The second wire carries a duplicate of the signal, but with inverted polarity and is termed *negative*[6]. The outer shield serves as a common ground (Osder & Carman, 2007:55). When a signal travels through such a balanced line, both the original signal and its inverted duplicate pick up very nearly the same noise and interference. When these signals reach their destination, the duplicate is inverted again and the two signals are summed together by a differential amplifier (Thompson, 2005:165). The result is that the noise picked up by the original signal and the nearly identical noise picked by the duplicate signal cancel each other out (Self, 2011:466). This process is known as *common-mode-rejection* (Hickman, 2001:69) and makes it possible for balanced lines to transmit signals over much greater distances than unbalanced lines, with very little electrical noise (Utz, 2003:32; Schaefermeyer, 2007:170).

2.2.3 Analog Connectors

Analog audio connections can further be classified by the type of connector (Rudolph, 2001:29). *RCA* and *tip-sleeve* quarter inch plugs fall under the unbalanced category. The RCA plug has a center tip which carries the signal, with an outer sleeve that acts as ground (Perozzo, 1986:292). An RCA plug is shown in Figure 2.6a. The name is derived from the *Radio Corporation*

[5] See Section 2.2.4.

[6] Alternatively, the positive signal is denoted *hot*, while the negative signal is denoted as *cold*.

of America, who introduced the connector in the 1970s as a consumer audio interconnection standard (Ahlzen & Song, 2003:106). While small and inexpensive, they are prone to breakage if frequently unplugged and reconnected (Hurtig, 1988:75) and are mostly used for consumer audio equipment (Winer, 2012:113).

Tip-sleeve (TS) plugs contain an insulating ring that separates the tip from the rest of the plug, known as the sleeve. A TS plug is shown in Figure 2.6b. By convention, the tip is the positive connection and the sleeve is the ground (Biederman & Pattison, 2013:199). The TS quarter inch plug is the most common connection for musical instruments (Trubitt, 1999:165) and is found on all electric guitars[7] and basses, keyboards, synthesizers and mixers (Rudolph, 2001:30).

(a) RCA.

(b) TS.

(c) TRS.

(d) Male XLR.

Figure 2.6: Analog connections.

Tip-ring-sleeve (TRS) quarter inch plugs and *XLR* connectors provide balanced connections. TRS[8] quarter inch plugs are nearly identical to TS quarter inch plugs, with the addition of a second isolating ring, as can be seen in Figure 2.6c. This plug has three electrical contact points: the tip, ring and sleeve

[7] Quarter inch connectors are commonly referred to as guitar plugs, however they were first developed for manual telephone switchboards and may also be called phone plugs (Nelson, 2010:320).

[8] A variant of the TRS quarter inch plug is the *General Post Office* (GPO) plug. Also used in telephone switchboards, GPO plugs follow the same wiring scheme. Its appearance is slightly different in that the ring section is rounder and bulbous, compared to the straight cylindrical ring of the TRS plugs. Many studio's still prefer GPO plugs because they lock more securely in their jacks due to their bulbous shapes.

(Buono, 2008:27). These correspond to the positive, negative and ground connections, respectively (Hurtig, 1988:76). TRS connectors are also available in a 3.5 mm plugs, however these are not balanced and are used as a stereo connection in consumer audio equipment (Ahlzen & Song, 2003:108).

Quarter inch plugs and jacks also come in a more compact version, known as *tiny telephone* (TT) connectors. Studios often prefer TT connectors, because more of them can fit into a given space than standard quarter inch connectors (Gallagher, 2009:216).

XLR connections come with three numbered pins that correspond to a wiring standard as agreed to by the *International Electrotechnical Commission* (IEC). Pin 1 connects to ground, pin 2 to positive and pin 3 to negative (Hurtig, 1988:76). XLR connections are heavy duty, professional grade connectors (Geoghegan & Klass, 2005:69; Bass, 2013:108) and feature a locking latch to prevent them from being accidentally disconnected (Reese *et al.*, 2009:128). A XLR plug is shown in Figure 2.6d. Unless labeled, it is not possible to distinguish inputs from outputs for the the above-mentioned connectors. XLR connectors, however, follow the convention where female connectors are inputs and male connectors are outputs, with the signal following the same direction[9], from female to male[10] (Biederman & Pattison, 2013:199).

All the above mentioned connectors come in male and female variants, where a male connector is called a plug and a female connector is called a jack. However, these terms are often used interchangeably (Sauls & Stark, 2013:127).

2.2.4 Direct-Inject and Re-Amp Boxes

As an alternative using a microphone, signals from a guitar, keyboard or synthesizer may be obtained directly through the use of a *direct-inject* (DI) box (Atkins, 1999:5.75). DI boxes come in two variants, *active* and *passive* (Bregitzer, 2009:30). The simplest DI boxes contain just a transformer (Rumsey & McCormick, 2014:378) and are termed passive, because they require no external power (Biederman & Pattison, 2013:209). The transformer converts the signal from unbalanced to balanced so that it may be sent over greater distances and accepted by equipment that require balanced connections (Grant, 2003:67). It also reduces the impedance and output to levels that are suitable

[9] This convention is because of *phantom power*. An audio input is also a phantom power output. If the phantom power on an input is enabled and the connector is physically open, it presents a shocking hazard and therefore requires the insulated contacts of a female connector (Watkinson, 2001*b*:241). A rule of thumb that the pins of the male connector point in the direction that the signal is traveling. For more on phantom power, refer to Section 2.3

[10] This applies in the context of an individual cable. The opposite applies for the connection from device to cable or cable to device.

for feeding microphone inputs ((Rumsey & McCormick, 2014:378); (Lellis, 2013:37)). An passive DI box is shown in Figure 2.7.

Figure 2.7: RADIAL JDI passive direct inject box.

Active DI boxes boost the signal with an internal preamp (Correll, 2008:228) and uses an electronic circuit to convert the input signal (Lee, 2009:35). Due to the more complex circuitry, active DI boxes always require external power, which is supplied by either the mixing console, preamplifier or an internal battery (Slone, 2002:31).

In addition to an unbalanced quarter-inch jack input and a balanced XLR jack output, it is common for DI boxes to offer a parallel unbalanced quarter-inch jack output connection (Lellis, 2013:37). This allows for direct connection of the musician's instrument to an amplifier, while a duplicate of the signal is simultaneously balanced and fed to another device, such as a recording console. DI boxes also come with a ground lift switch to prevent ground loops (Biederman & Pattison, 2013:209).

A *re-amp box* is a device that converts the output signal from a digital audio workstation (DAW) to the typical output level of a guitar or bass (Watkinson, 2001*a*:95). It is similar to a DI box, but in reverse. Through re-amping, the DAW output signal can be sent through an amplifier with a microphone placed in front of the speaker, allowing it to be recorded again (Huber & Runstein, 2010:147). Whenever re-amping is used, it becomes possible to audition any number of amplifiers, effects or microphone placements until the desired sound has been found (Huber & Runstein, 2013:144-145). A custom built re-amp box, designed by Scott (Dorsey, 2014), is shown in Figure 2.8. For more information, refer to Appendix C.2.

2.2.5 Patch Bay

A *patch bay* is a routing system that acts like a telephone switchboard (Hurtig, 1988:78) and is used to connect various input and output devices, without having to physically manipulate each wire from each device (Zager, 2011:272).

Figure 2.8: Dorsey re-amp box.

It consists of a collection of jacks usually mounted in a rack-mountable frame, each connected to an input or output of an electronic component in the studio (Alten, 2013:133). By connecting the desired outputs and inputs, any piece of equipment can be included into the signal chain. Patch bays have been the industry standard since the early days of broadcasting and are still the easiest and most reliable way to accomplish reconfigurations (Campbell *et al.*, 2007:506). Most patch bays are custom built to accommodate the needs of a particular studio (Zager, 2011:272), but prefabricated patch bays are also available (Huber & Runstein, 2013:454).

Patch bays come in different variants and may contain TS, TRS, TT, GPO or RCA jacks (Huber & Runstein, 2013:454). Patching is accomplished through the use of a patch cord, a short cable that connects points on the patch bay (Gallagher, 2009:152).

Patch bays and *outboard* equipment are housed in supporting structures, known as racks, that allow for easy insertion, removal and configuration of modular component systems (Petersen, 2012:774). Outboard refers to any device that is not physically part of the main mixing console and must be patched into the signal chain (Thompson, 2005:44). The dimensions of outboard equipment are designed to fit these standardized racks and therefore are sometimes also referred to as *rack gear* (Childs, 2011:160). Outboard equipment includes compressors, expanders, noise gates, reverberation units, preamplifiers and equalizers (Gottlieb, 2007:71).

Normally patch bays are wired with outputs at the top and inputs at the bottom (Gottlieb, 2007:74). Several wiring configurations are possible. These include full-normalled, half-normalled, multiples and isolated. In a full-normalled connection, the output one piece of equipment is directly connected to the input of another, without being routed anywhere else. The jacks are connected in such a way that insertion of a patch cord into the output jack interrupts this connection and allows the signal to be routed elsewhere (Franz & Lindsay, 2004:20). This is called *breaking the normal* (Watkinson, 2001*a*:81).

Half-normalled connections work similarly to full-normalled ones, but breaking the normal only occurs when a patch cord is inserted into the input jack (Thompson, 2005:46). Connecting a patch cord to the output jack simply splits the audio between its original path and the path of the patch cord (Huber & Runstein, 2013:455).

A multiple, or *mult*, is a collection of patch jacks that are wired together and serves to split the audio into as many multiples as there are patch points available in the collection (Watkinson, 2001*a*:82). Any jack in a *mult* may be used as an input, because all the jacks are wired in parallel (Savage, 2011:82). The remaining jacks then serve as outputs. Finally, in an isolated configuration, the output jack is completely isolated from the input jack below it and there is no normal (Franz & Lindsay, 2004:22). This configuration is also called *open*.

2.3 Preamplifiers

As mentioned earlier, a preamplifier boosts a microphone's signal to line level (Franz & Lindsay, 2004:10). The required amplification is achieved by increasing the signal voltage or reducing the impedance. The amount of power amplification required varies with the particular application. A general guideline is to provide sufficient amplification to ensure that further signal handling adds minimal signal-to-noise degradation. (Korzekwa & McFadyen, 2004:11.18). At the very least, preamplifiers provide control to adjust the input gain, but in most cases also provide variable input impedances, polarity switching and phantom power[11] (Edstrom, 2010:10).

Phantom power is a DC voltage that is superimposed on the audio signal in a microphone cable, without effecting the signal itself (Leonard, 2001:43). A voltage, supplied via two 6800 Ω resistors, is sent over both the signal lines in a balanced connection, with the same polarity on each line (Davis & Jones, 1989:130). Inside the microphone, the current is transferred from a center tap of a transformer to the microphone electronics and diaphragm. In case of the absence of a transformer, the same routing is accomplished through the use of two identical resistors. The ground shield provides a return path for the current to complete the circuit (Rumsey & McCormick, 2006:67).

For phantom power, 48 V is the most common (Holman & Baum, 2013:122), although phantom powered microphones can operate on a wide range of voltages[12] from 1.5 V or 9 V up to 50 V (Davis & Jones, 1989:130). Phantom power provides a means to remotely power condenser and electret microphones

[11]Most microphone preamplifiers have individual switches on each channel for *phantom power* (Self, 2009:80), although sometimes one switch will control phantom power for several channels.

[12]It is good practice to turn phantom power off when it is not in use (Leonard, 2001:44), although balanced microphones will not be damaged if it is accidentally left on (Self, 2009:114).

(Ford & Silsby, 2007:435). The name comes from the fact that, because of common-mode rejection, phantom power is electrically invisible to the preamplifier (White *et al.*, 2013:278).

Mixing consoles and most audio interfaces have preamplifiers built-in, so a dedicated preamplifier is not always necessary (Edstrom, 2010:10). However, a given preamplifier may have a specific desirable sonic characteristic (Cook, 2013:59) and therefore many recordists use outboard preamplifiers as an alternative to the built-in ones (Eargle, 2002*b*:78).

According to Touzeau (2009:82), good preamplifiers are able to accept a wide range of signal levels and amplify them in a flexible manner so they may be used in a wide range of applications. Self (2010:323) lists the following requirements of a microphone preamplifier:

- Variable gain, ranging from 0 dB to 70 dB, although the range of some designs extend to 80 dB.
- Minimum noise must be produced.
- Input requires a high common mode rejection ratio (CMRR), to reject interference and ground noise.
- Input must have a constant resistice impedance of 1-2 kΩ, which provides appropriate loading for a 200 Ω dynamic microphone, as well as for the internal head amplifiers of capacitor microphones.
- The input must be proofed against the sudden application or removal of 48 V DC phantom power and withstand this for many repeated cycles over the life of the equipment.

2.4 Audio Interfaces and Converters

Digital audio workstation (DAW) is a collective name given to the equipment that make it possible to convert signals from analog to digital and store them (Lellis, 2013:120). A DAW records, edits and plays back like a digital recorder, but has considerably more power due to the software it employs (Alten, 2013:151). DAW's employ digital signal processing, enabling the manipulation of the frequency response, stereo imaging, or dynamics of an audio signal (Reese *et al.*, 2009:143).

In addition to the DAW's own functionality, third party *plug-ins* can be added, extending its flexibility in ways not available in analog signal processing (Eargle, 2002*b*:207). Plug-ins are software modules that provide a form of digital signal processing, running either on the computer's CPU or on a dedicated digital signal processor (DSP) (Rumsey & McCormick, 2006:381). In much the same way as a piece of outboard gear is patched into an analog signal chain, plug-in software allows the digital signal chain to be manipulated so that the user can specify the plug-in's exact location in the signal path.

Essential to a DAW are its analog-to-digital (A/D) converters and digital-to-analog (D/A) converters. To achieve the required resolution, high quality converters are necessary. Therefore, high-end DAW's use dedicated outboard converters that are of very high quality and fidelity (Alten, 2013:153). A/D converters are used at the input stage, while D/A converters are used at the output stage.

Audio converters come in the form of an *audio interface*. An audio interface is a device that allows a computer to interact with external audio devices, such as mixing consoles, microphones or electric instruments (Newhouse, 2004:6), by converting the audio signals from analog to digital (Beauchamp, 2005:107). These come in various configurations, from a dedicated audio converter up to an all-in-one system which provides multiple inputs with onboard preamplifiers, converters, mixer, internal hard drive, CD-writer and effects that enable the user to complete an entire production without the need of a computer or external processors (Edstrom, 2010:15). The grayed area in Figure 2.1 on page 26 illustrates such an all-in-one system.

Audio interfaces may connect to a computer through PCIe[13], USB, Firewire (Pejrolo, 2011:43), and more recently Thunderbolt[14] (Hopgood, 2013:9). Interfaces vary in price, depending on the quality and specifications of their audio converters (Pejrolo, 2011:1.37). Since conversion is the primary purpose of an audio interface, it is important to evaluate the quality of a device's converters (Edstrom, 2010:12).

Figure 2.9 shows the Avid HD Omni, an audio interface that features four channels of analog input with two high quality preamplifiers that can also be used to direct-inject electric instruments, eight channels analog of output, 24-bit converters and samples rates ranging from 44.1 kHz to 196 kHz (Cook, 2013:83, Avid Technology Inc., 2010*c*:1).

Figure 2.9: Avid HD Omni (Avid Technology Inc., 2010*c*:5).

There are two fundamental parameters of digital audio, the *bit depth* and the *sampling rate*. The bit depth is related to the *dynamic range*, which is the span

[13]Peripheral Component Interconnect Express (Shelly & Vermaat, 2009:238).

[14]Since a recordist does not require explicit knowledge of the working principles of these connections in order to use them effectively, the topic of these connection protocols are outside the scope of this text. For more information, please see (Fries & Fries, 2005:39-43).

between the softest and loudest sounds that can be handled by the system. The dynamic range of the human ear is 120 dB whilst the dynamic range of a digital audio system is proportional to the quantization of the system. One bit correlates to 6 dB. Hence the dynamic range of a 16 bit system is limited to 96 dB, whilst a 20 bit system has the same dynamic range as the human hearing mechanism. Most digital mixers provide 24 to 64 bit resolution (Dimpker, 2013:211).

2.5 Monitoring

Monitoring is the act of judging the quality of an audio signal by listening to it (Bartlett & Bartlett, 2007:274). Monitor loudspeakers, or simply *monitors*, are high-quality loudspeakers that are used to monitor the output of the mixing console or DAW (Stephenson *et al.*, 2013:33).

Like microphones, loudspeakers are classified according to the operating principle of their transducers and many types exist (Ahnert & Steffen, 2000:79). These include electrostatic, ribbon and distributed mode loudspeakers. However, the majority of loudspeakers are moving coil devices which employ a cone-shaped diaphragm (Eargle, 2003:9), therefore these shall be the focus of this section[15].

A moving coil loudspeaker consists of a sound radiating diaphragm, a moving coil and a permanent magnet (Ahnert & Steffen, 2000:79). The center of the diaphragm is supported by a *spider*, a movable membrane which allows axial but not radial movement (Watkinson, 2001*c*:45, Raichel, 2006:579). The coil, known as the *voice coil*, is rigidly attached to the center of the diaphragm and lies between the poles of a specially shaped permanent magnet (Rossing & Fletcher, 2004:244). The perimeter of the cone is supported by a flexible rim suspension or surround (Watkinson, 2001*c*:45). To allow the diaphragm to move freely, these fasteners are made from an elastically deformable material that is rigid, yet provides the necessary damping (Ahnert & Steffen, 2000:79). The loudspeaker mechanism as described here is collectively referred to as the transducer or the *driver* (Eargle, 2003:1).

When an alternating current, such as an audio signal, moves through the voice coil, a magnetic field is generated around it. This magnetic field interacts with that of the permanent magnet to exert a force on the voice coil and diaphragm (Kleiner, 2013:301). This force is proportional to the current that passes through the voice coil and makes the diaphragm vibrate, generating pressure variations in the air and resulting in sound waves (Gupta, 1995:47).

Two types of monitors exist. Active monitors have an amplifier circuit built into the loudspeaker enclosure and require external power (Hogan & Fisher, 2009:82) These monitors accept line level inputs in the form of balanced TRS

[15] For more information regarding other types of loudspeakers, please refer to Rumsey & McCormick (2006:81) and Ahnert & Steffen (2000:79)

or XLR connections (Izhaki, 2013:78), as well as unbalanced TS, RCA and sometimes digital inputs (Hawkins, 2002:222). They come equipped with an input gain control and a power switch.

Passive monitors contain no internal amplifiers and therefore require no external power (Alexander & Whitear, 2001:220). The power that drives these loudspeakers comes directly from the audio signal which is amplified by an external amplifier (Izhaki, 2013:77). Passive monitors accept input with a very high gain. To accommodate this, these types of monitors have screw terminals, which offer a greater contact area than conventional unbalanced plugs.

A single driver cannot produce all the frequencies in the audible frequency range, therefore monitors rarely contain only one driver (Alten, 2013:49). Monitors employ two or more drivers to produce the low, midrange and high frequencies separately (Hollembeak, 2011:293). A crossover network filters and distributes the frequencies of the incoming audio signal across the individual drivers in order to efficiently reproduce the signal's frequency range (Alten, 2013:49). In a two-driver system, the signal for the low frequency driver passes through a low-pass filter, which attenuates signals above a certain frequency. The signal for the high frequency driver passes through a high pass filer, which attenuates frequencies below a certain frequency (White & Louie, 2005:91). This certain frequency is called the *crossover frequency* and is the frequency at which both drivers receive the same amount of energy (Hollembeak, 2011:168).

The optimum stereo loudspeaker configuration is such that an equilateral triangle is formed between the loudspeakers and listener, as illustrated in Figure 2.10 (Franz, 2001:15). The listener's head position is known as the *sweet spot* (Holmes, 2013:295). Rumsey & McCormick (2006:477) suggests that the listener should be located just to the rear of the focal point of the loudspeakers, however Izhaki (2013:86) believes the optimum configuration is with the listeners' head just in front of the focal point.

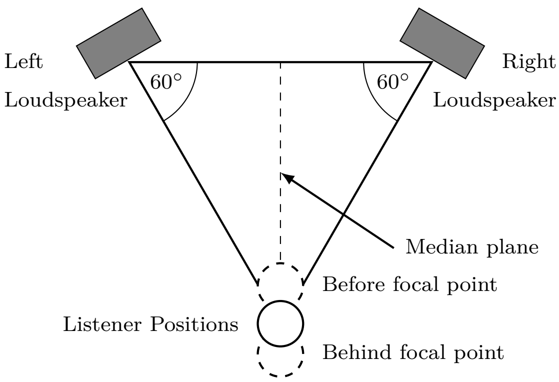


Figure 2.10: Listening position (Franz, 2001:15).

When loudspeakers placed apart at wider angles, the apparent locations of sound sources in between the loudspeakers become less stable, and the system is more susceptible to the effects of head rotation (Rumsey *&* McCormick, 2006:477). In this configuration, sound emitted from both loudspeakers reach the listener at the same time. However, if the listener's head were to move off the median plane, the *precedence effect* may occur. The precedence effect, also known as the Haas effect, occurs when the listener is off the median plane, even by as little as 15 cm, and shifts the virtual image created by the loudspeakers toward the loudspeaker closest to the listener (Franz, 2001:14). The correct loudspeaker configuration is paramount for accurate reproduction of stereo imagery (Viers, 2011:157).

Many sources within a mix are stereophonic. If the recordist wishes to evaluate the timbre of a recorded track within the mix, even if this track is monophonic, it is important that this evaluation is done from the optimum listening position in order to obtain an accurate depiction of the stereo imagery produced by stereo tracks within the mix.

2.6 Equalization

Equalization is used to adjust the frequency response of a system to suit the taste of the recordist and to compensate for environmental problems (Boyce, 2014:132). Successful equalization results in a response that is clear and intelligible (Eargle, 2002*a*:292). Equalizers range from a single band tone control to multi-band outboard equipment. It is the most common type of frequency processor (Winer, 2012:277).

A major role of equalization is the modification of the timbre of both acoustically and electronically generated sounds for artistic purposes. In this context, the ability to amplify or attenuate selected frequency ranges is used to modify a sound's frequency spectrum to achieve a desired effect on its timbre. However, according to (Benade, 1985:232), one cannot use equalization to compensate for poor microphone placement or talent[16] positioning in a room. Equalization relies on spectral modification only and does not modify the envelope or dynamics of an audio signal (Howard, 2009:399).

Equalization is a very powerful tool and small adjustments can have a very noticeable effect. Even the ability to adjust equalization by only ±3 dB over the whole frequency range can result in a wide range of different timbres Newell (2003:367). An equalization curve for a single band is illustrated in Figure 2.11. The curve has a bell shape that is symmetrical about a center frequency. The *bandwidth* is defined as the breadth of the equalizing curve, 3 dB from the maximum amplification or attenuation (Savage, 2011:45).
Two basic types of equalizers exist in audio signal processing, namely *graphic* and *parametric* equalizers (DiPaola *&* DiPaola, 2012:152). A graphic equalizer

[16]The talent refers to the musician being recorded.

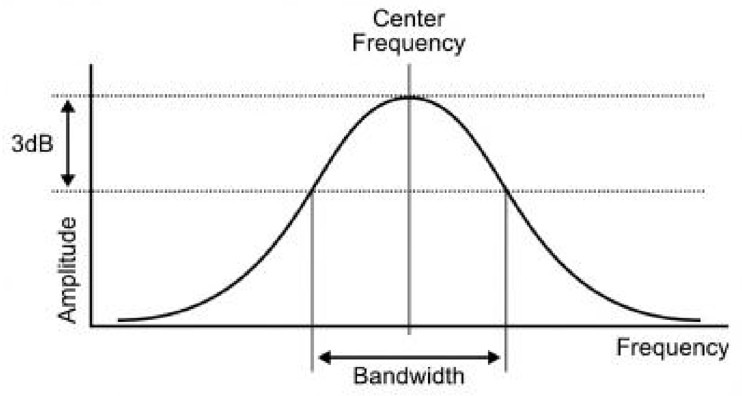


Figure 2.11: Equalization curve (Savage, 2011:45).

has multiple vertical sliders. Each slider increases or decreases the frequency response over a specific frequency range using a variable band filter. As the controls are adjusted, the physical positions of the sliders resemble a curve that approximates the frequency response of the equalizer as a whole (Eargle, 2002a:233). The graphic equalizer normally has fixed frequency and bandwidth values (Hugill, 2012:111). Figure 2.12 shows a graphic equalizer.

Figure 2.12: Moog ten band graphic equalizer

Parametric equalizers have fewer filters than the graphic type, but the parameters of each filter are highly variable (Foreman, 2008:1287). It features variable frequency, amplitude and bandwidth. The variable bandwidth enables the user to select the width of the bell-shaped curve for amplification or attenuation (Kefauver, 2001:210). Due to their flexibility, parametric equalizers with only three or four filter selections can approximate almost any curve needed for equalization and are therefore favoured over graphic equalizers (Foreman, 2008:1287). The ability to choose which frequencies are equalized provides a very powerful timbre control (Hurtig, 1988:45).

Modern systems have equalization functions that are performed through DSP. The equalizer is simply a software module. The DSP devices are controlled by a computer and the settings of the equalizer module are accessible through a graphic user interface that resembles an analog equalizer. With many such devices, the user may choose a graphic or parametric equalizer and control the bandwidth, center frequency, insertion depth and even filter type (Foreman, 2008:1287). The Pro Tools EQ III software equalizer is shown in Figure 2.13.

The EQ III is a five band parametric equalizer with additional high- and low-pass filters. The equalizer is colour coded and the effect of each filter is illustrated as a shaded area of corresponding colour on the frequency grid. The overall equalization curve is shown as a white line.

Figure 2.13: Pro Tools EQ III.

2.7 Transfer Function

A *transfer function* explains the mathematical function of the parameters of a system, acting on an input to produce the required output (Bakshi & Bakshi, 2007:45). Therefore, the transfer function indicates the relationship between the input and output of a system (Singh, 2010:126). A transfer function, $G(s)$, of a system may be represented by the block diagram in Figure 2.14, where $X(s)$ is the input and $Y(s)$ is the output (Fox & Bolton, 2002:217). Mathematically, this transfer function may then be described by Equation (2.7.1) (Bolton, 2002:54).

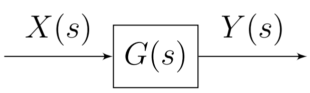
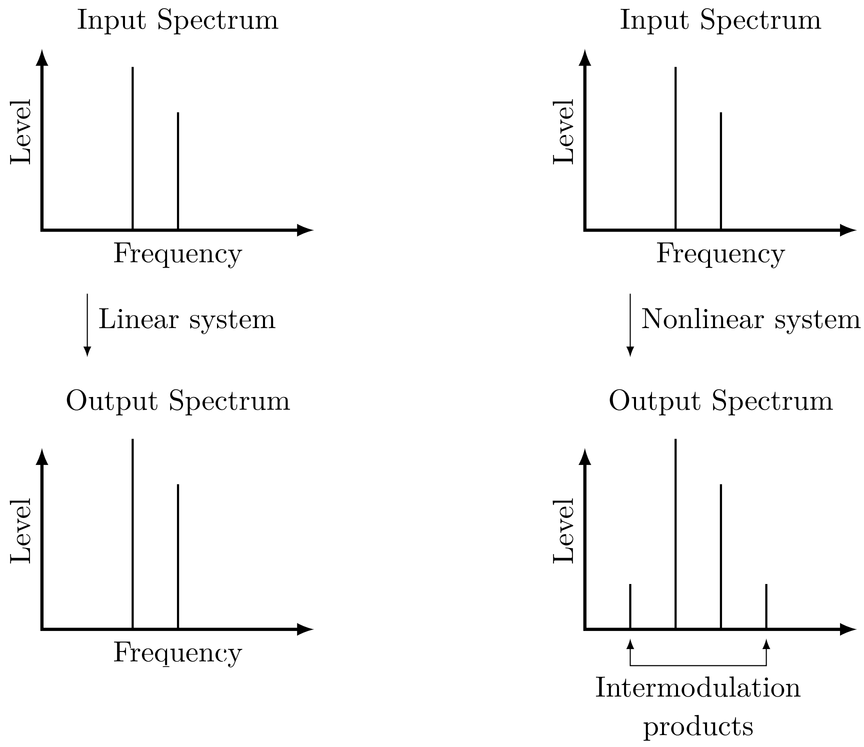


Figure 2.14: A transfer function diagram (Fox & Bolton, 2002:217).



(a) Linear system.

(b) Nonlinear system.

Figure 2.15: Intermodulation (Watkinson, 2009:22).

$$G(s) = \frac{X(s)}{Y(s)} \tag{2.7.1}$$

Every piece of audio equipment has some transfer function that effects the signal in some way (Watkinson, 2001*a*:58). For a linear transfer function, the output of the system is proportional to the input. However, for a nonlinear transfer function the output waveform is distorted. The distortion results in a redistribution of harmonics, changing the timbre of the sound (Watkinson, 2009:21).

Sounds passing through a system with a nonlinear transfer function no longer have independent existence, but interfere with one another, changing one another's timbre and even creating new sounds that did not previously exist. This is known as *intermodulation*. Figure 2.15 illustrate sound passing through a linear and a nonlinear system. In the linear system, the waves experience no interference. However, in the nonlinear system, waves intermodulate to produce sum and difference frequencies, forming intermodulation products (Watkinson, 2009:22).

A transfer function is complex and has real and imaginary components corresponding to magnitude and phase, respectively. Like microphones, each system in the signal chain has a frequency response that describes the way in which the magnitude and phase of the system vary with the frequency of the input. It is a characteristic of the system, not of the signal passing through it (White & Louie, 2005:165-166). For audio equipment, often only the magnitude frequency response is provided.

Figure 2.16 illustrates the effect of a system's frequency response on the spectrum of a signal. The frequency spectrum of the output shows that the harmonic structure has changed, indicating a change in timbre of signal (Watkinson, 2009:21). The timbre of a sound depends on its *frequency spectrum*, the distribution of energy over frequency. In a recording signal chain, the frequency response of every system affects spectrum of the signal that pases through it to some degree.

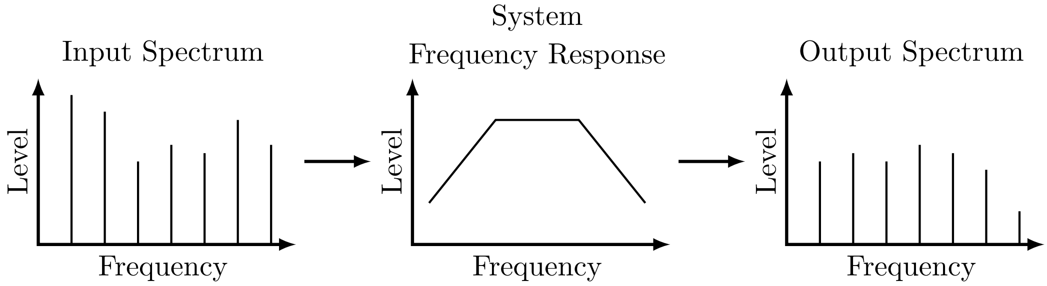


Figure 2.16: Effect of frequency response on timbre (Watkinson, 2009:20).

Prior to the invention of transistors, valves were the main active component in electronics. Triode and pentode valves have nonlinear transfer functions that result in the addition of harmonics to signals that pass through them. The triode valve in particular adds a second harmonic to the signal. When used in preamplifiers, triode valves are considered to provide *warmth* and a soft sound colouration (Zölzer, 2011:117). Valve preamplifiers are still used by recording engineers, many of whom favour them over modern transistor based preamplifiers.

Chapter 3

Microphone Techniques

> "*The correct microphone technique is* [whichever] *one will produce the aural illusions desired by the producers of the recording. With that premise accepted, one must also say that some techniques are better suited than others. So, when choosing a microphone technique, a recording producer must first decide on the goal of his work, and then provide a microphone technique that will produce the sound he wants.*"- Shafer (1981:1)

Microphone technique refers to the selection and placement of microphones. According to Boudreau *et al.* (2005:1), it is largely a matter of personal taste. Whatever method sounds right for the particular sound, instrument, musician and/or song is right as there is no one ideal way to position a microphone. Even though there are no exact rules to be followed, only recommendations Ballou *et al.* (2008:519), it is important to understand how microphones work, their various pick-up patterns, sensitivity and frequency responses (Ballou, 2008:xi). Where a microphone is placed on an instrument is as critical a decision as which microphone the audio professional chooses (Gottlieb & Hennerich, 2009:138).

Kefauver (2001:123-124) states certain parameters within which the recordist must work to be commercially successful and suggests the following precautions:

- Always take off-axis colouration and proximity effect into account.
- Avoid overloading the microphone preamplifier by attenuating the signal.
- Protect the microphone against wind and vibration noises.
- Exercise caution when using acoustic baffles. When possible, move the microphone instead of using a baffle.
- Ensure all microphone lines are properly shielded, balanced and terminated.
- Ensure there are no electrical phase reversals in the signal path, unless required.

- Avoid using two microphones if using only one will produce better results.
- Never use equalization as a substitute for proper microphone selection and placement.

3.1 The Influence of the Microphone

3.1.1 Polar Pattern

Directional microphones are used to pick-up desired sounds and discriminate against unwanted sounds such as reverberation and noise (Olson, 1967:420). A microphone polar pattern is achieved either through a phase-shifting network or an acoustic maze. However, the effectiveness of these devices is related to the wavelength of the incident sound. It is difficult to create such microphone sized devices that are effective at all frequencies (Wickstrom, 1983:7). As a result, directional microphones exhibit increased directionality with increasing frequency (Alten, 2012*b*:77) and many directional microphones effectively become omni-directional at low frequencies (Putnam, 1980:3).

Microphones are also subject to *off-axis colouration.* Off-axis colouration describes how the polar pattern of a microphone changes with the frequency of the sound that it picks up (Lubin, 2010:38). It occurs in microphones where the off-axis sensitivity to high frequencies decreases at a much greater rate than for low frequencies and may result in a response with overly dominant low frequencies (Kefauver, 2001:82).

At a certain distance from the source, a microphone experiences an equal direct sound and reverberant sound. This distance is known as the *critical distance* (Eargle, 2005:15). At any point beyond the critical distance, reverberation is louder than the direct sound (Izhaki, 2013:424) and a directional microphone loses its ability to discriminate against unwanted sources and reflections (Roux, 2011:87).

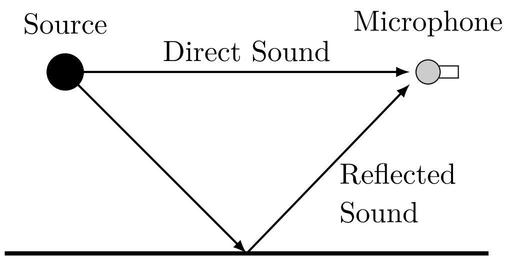
3.1.2 Microphone Dimensions

The presence of the microphone disturbs the sound field by diffracting or reflecting incoming sound waves to some extent (Woszczyk, 1989:1). *Diffraction* is the change in direction of propagation of a wavefront, with no change in velocity, due to the presence of an obstacle (Wright, 1997:347). It occurs when the wavelength of an incident sound is comparable or larger than the dimensions of the obstacle, as with low frequencies (Cowan, 1993:11). When its dimensions are smaller, as with high frequencies, reflection occurs (Sen, 1990:137).

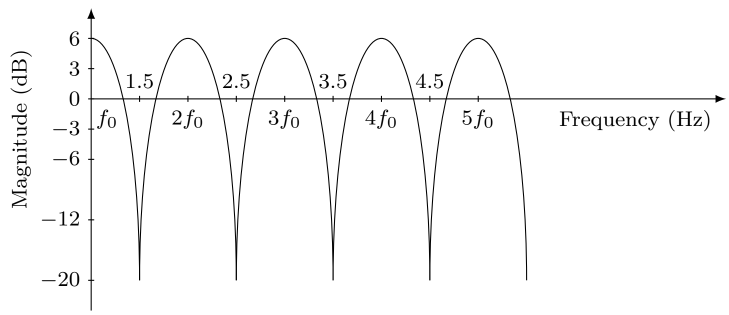
The diffracted and reflected waves are scattered in all directions and interfere with the incoming sound waves. Since the interference occurs in such close proximity to the transducer, it may also significantly alter the transducer's response (Woszczyk, 1989:1).

3.1.3 Distance from Reflective Boundaries

The interference that exists between a sound and a delayed copy of the sound from the same source results in *comb-filtering* (Gervais, 2005:22). When the signals meet, their frequency content is radically modified so that some frequencies are canceled, while others are amplified. The intermediate frequencies experience something in between complete cancellation and doubling (Case, 2007:232). The result is a series of notches and peaks in the frequency response that significantly colour the sound (Gallagher, 2009:35). Figure 3.1a illustrates a microphone that experiences direct sound from a source as well as a delayed copy of the sound, caused by reflections from the floor. At certain frequencies, the longer path length of the reflected sound corresponds to a 180° phase shift, resulting in the frequency response consisting of evenly spaced nulls, as shown in Figure 3.1b. The frequency response shows alternating peaks and troughs, resembling the teeth of a comb (Watkinson, 1998:201). Reinforcements and cancellations continue to occur at multiples of the offending frequency. The amount of signal degradation depends on the relative loudness of the direct and reflected signals (Kefauver, 2001:125).



(a) Reflection from the floor (Ballou *et al.*, 2008:523).



(b) Comb-filtering frequency response (Watkinson, 1998:201).

Figure 3.1: The comb-filtering effect, where f_0 is the frequency that undergoes comb-filtering.

Placing the microphone closer to the primary reflective surface may greatly reduce the effects due to first order reflections (Bullock & Woodard, 1984:1). A boundary layer microphone, such as the Shure Beta 91 shown in Figure 3.2, may also be used. Such a microphone can be placed on a reflective surface, thereby avoiding the comb-filtering effect, because the microphone's diaphragm is effectively in the same plane as the reflective surface (Müller, 1990:2, Talbot-Smith, 1999:2.46).

Figure 3.2: Shure Beta 91 boundary layer microphone

3.1.4 Distance from the Source

Placing a microphone close to a source reduces the pick-up of reverberation and increases the pick-up of direct sound (Holman, 2010:57). It is a primary method for obtaining isolation from noise and other instruments (Moulton, 1990:162). Close microphone placement is a common technique, because it is easier to add reverberation to a signal, than to remove it (Rudolph, 2001:129).

Boudreau *et al.* (2005:15) suggests the need for isolation in the live sound environment as the origin of the close microphone placement technique. Billingsley (1989:1), however, states another possible origin. Early multi-track recording equipment such as microphones, lines, consoles, preamplifiers and signal processors were not as quiet as modern day equipment and introduced noise into the recorded signal. The solution was to record sources at very high levels to achieve a high signal-to-noise ratio, either by increasing gain or decreasing the distance between the microphone and the source. Today, advances in microphone, cable, preamplifier and circuit design have led to excellent signal-to-noise equipment specifications (Billingsley, 1989:5).

By varying the distance from the source between the closest possible position and the critical distance of the microphone, the ratio between direct sound and reverberant sound may be adjusted. Before the existence of recording consoles, microphone distance was the tool with which a pleasing balance was achieved (Billingsley, 1989:2).

3.1.5 Proximity Effect

Directional microphones exhibit a phenomenon where their low frequency response increases when placed closer to the sound source (Franz, 2001:122). This is known as the *proximity effect* and is an inherent characteristic of directional microphones (Eiche, 1990*b*:51). Omni-directional microphones do not exhibit the proximity effect (Gibbon *et al.*, 1997:303).

Directional microphones usually have two acoustic entrances which allow incoming sound waves to act on the front and back of the diaphragm (Milanov & Milanova, 2007:1). Therefore the microphone responds to the difference in sound pressure that acts on either side, or the pressure gradient (Rumsey & McCormick, 2014:54). The pressure gradient is caused by the difference in amplitude and phase of a pressure wave as it arrives on either side of the diaphragm. These amplitude and phase gradients are a function of the sound paths from the source to the front and rear of the diaphragm, respectively. The phase gradient is also frequency dependent (Clifford & Reiss, 2011:2).

For a distant source, the amplitude gradient across the diaphragm is small compared to the phase gradient, because the difference in the sound paths to the front and rear of the diaphragm is small compared to the distance between the microphone and the source. However, as the source is brought closer, the amplitude gradient increases and the phase gradient decreases (Clifford & Reiss, 2011:2). The decrease of the phase gradient occurs especially in the low frequency range because, with the microphone close to the source, the low frequency wavelengths are large compared to the difference in sound paths between the front and rear of the diaphragm. The large magnitude gradient and the small low frequency phase gradient combine to form a large low frequency pressure gradient across the diaphragm, which results in an increased low frequency response known as the proximity effect.

The proximity effect is neither good nor bad and can be used to the recordist's advantage (Edstrom, 2010:72). Too much of it, however, can cause a boomy sound that lacks clarity (Beauchamp, 2005:46). Therefore, the proximity effect must be considered when selecting and placing a microphone (Ciaudelli *et al.*, 2009:175).

3.2 Monophonic Techniques

Monophonic recording is defined as a recording that is done with one or more microphones (or transducers), wherein the case of the latter, the signals are combined into one (Holman, 2010:57). An example of the use of multiple transducers to produce a monophonic signal, is the recording of a bass guitar. This signal often consists of the combination of the signal recorded from the bass amplifier with a microphone and the direct signal, recorded through a DI box. The two signals are either recorded to individual tracks and properly balanced during the mixing process, or combined straight away into one track (Thompson, 2005:157-158).

Monophonic signals introduced in a stereo reproduction system come across as central images, with no spatial information. Through the use of *panning* the image may be positioned within the stereo image (Watkinson, 1998:10). The pan control splits a monophonic signal and controls the proportion of signal fed to the left and right loudspeakers (Rumsey & McCormick, 2014:524). Therefore, the location of the virtual source within the stereo image is controlled solely through level differences between the left and right loudspeakers. According to Dove (2008:829), this is not true stereo, which can only be achieved through coincidentally aligned microphones. However, in almost every case in present day popular music, the stereo image that is created has a monaural origin (Billingsley, 1989:3). Multiple microphones are placed close to individual sound sources, each of which are treated as monophonic elements (Pizzi, 1984:3) and the stereo image is created by panning the various monophonic signals into different locations across the width of the stereo image (Watkinson, 1998:10). According to Pizzi (1984:3), this approach can produce convincing stereo imagery.

Monophonic techniques offer the most flexibility for microphone placement and make it possible to obtain a clean direct sound, with good rejection of unwanted sounds (Woszczyk, 1991:133).

3.3 Leakage

Leakage refers to the amount of signal picked up by microphones other than those intended for a source. It is alternatively known as *spill* or *bleed* (Bartlett, 2009:106). Acoustic barriers may be set up between the microphone and the unwanted sounds to reduce the onset of leakage (Kefauver, 2001:130). Additionally, microphones with tighter polar patterns, which are less susceptible to leakage, may be used (Dittmar, 2013:49). The most effective solution is the use of an isolation booth (Kefauver, 2001:130), a small room within a recording studio wherein the source and microphone are isolated from sounds made by other musicians (White & Louie, 2005:205).

A rule of thumb used by recordists is the *3:1 distance rule*. To reduce leakage and maintain phase integrity, this rule states that for every unit of distance between a microphone and its source, nearby microphones should be placed at least three times that distance away. The 3:1 distance rule is illustrated in Figure 3.3, where d is the distance between the microphone and its source. Some recordists err on the side of caution by employing a greater distance ratio of 5:1 (Huber & Runstein, 2010:136).

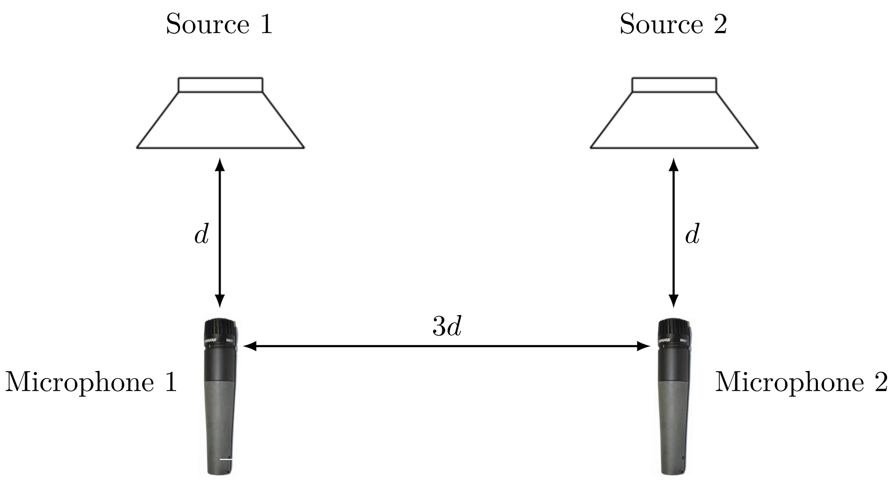


Figure 3.3: The 3:1 distance rule (Huber & Runstein, 2010:137).

3.4 Stereophonic Techniques

Two channel stereophony involves the use of two loudspeakers that receive non-identical signals to produce a *stereo image*. A stereo image is the illusion of localization of phantom sound sources from loudspeakers (Corey, 2012:59). Through ILD and ITD cues, the brain is capable of localizing phantom sound sources on a line between the two loudspeakers, adding an element of realism to the audio. This is the most popular spatial reproduction method (Pulkki, 2002:1). According to Watkinson (1998:194), such a reproduction system, having some spatial realism, will generally be preferred to a technically superior monophonic system.

Accurate stereo imaging is the foundation for the art of stereo recording (Streicher & Dooley, 1984) and several stereo microphone configurations and systems exist that enable stereophonic reproduction (Plewa & Pyda, 2010:1). For improved spatial accuracy, Watkinson (1998:216) suggests using well matched microphones which are identical in output level, phase and frequency response. Additionally, microphones used in crossed coincident pairs must have good off-axis response to prevent colouration of the central image.

With any stereo technique, there is a trade off between the width of the stereo image and localization of the sound source (Fisher, 2012:81). It is up to the recordist to select a technique that is suitable to the situation.

3.4.1 Monophonic Compatibility

Phase problems may arise when summing stereo signals, canceling various frequencies, disrupting the relative loudness of individual elements and reducing the overall quality (Bartlett & Bartlett, 1999:217). Monophonic compatibility refers to the amount of success with which all elements in a stereo signal are preserved when the signal is summed (Roux, 2011:107).

The most important criterion for whether or not stereo signals are mono compatible is how the audio sounds in mono (Pizzi, 1984:4). Ideally, it would sound as if it were recorded with only one microphone (White & Louie, 2005:243).

3.4.2 Coincident Techniques

Coincident techniques are achieved with a pair of directional microphones, aligned on a common axis, but set at an angle to each other. Due to their close proximity, sound reaches both diaphragms simultaneously (Ballou *et al.*, 2008:542) and no phase problems arise (Huber & Runstein, 2010:142). Signals from the two coincident directional microphones are each assigned to one loudspeaker, thus producing a stereo image between the loudspeaker pair. Localization depends fully on level differences between the two loudspeaker signals, producing ILD at the listener's ears (Schneider, 2012:1).

An advantage of the coincident pair is that the angular accuracy of the stereo imaging is unaffected by the distance of the microphone pair from the sound source. A disadvantage is that without the time difference common to some other microphone techniques, the stereo image sometimes seems lacking a sense of *space* (Streicher & Dooley, 1984:3).

3.4.2.1 XY

In the XY technique, the microphone pair is set at an angle of 60° to 120°, where this angle determines the width of the stereo image (Streicher & Dooley, 1984:3). The XY technique is shown in Figure 3.4. Microphones with cardioid or hyper-cardioid polar patterns are used (Pulkki, 2002:3), which result in a less reverberant sound because of reduced pick-up from the rear (White & Louie, 2005:200).

The midpoint between the two microphones is aimed at the source (Huber & Runstein, 2010:142), meaning most of the sound arrives off axis and is subject to off-axis colouration, especially when configured at larger angles

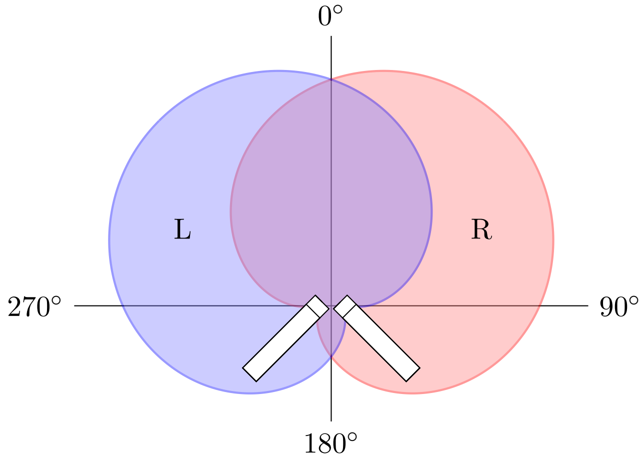
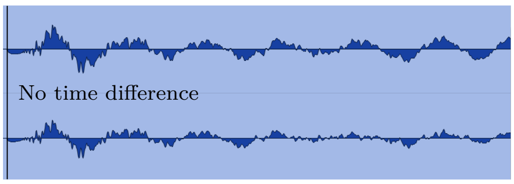


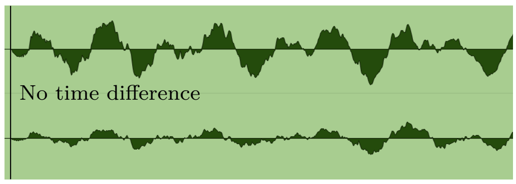
Figure 3.4: XY microphone technique.

(Hibbing, 1989:7). The XY exhibits good directionality and mono compatibility (Pulkki, 2008:756; Holman, 2008:83).

An experiment was conducted to illustrate the level and time differences that occur between the left and right channels in stereo microphone techniques. Details for this experiment may be found in Appendix B.2. Figure 3.5a shows the signals produced by an XY configuration for an on-axis source. The signals are the same level and there is no time difference between them. Figure 3.5b shows the resulting signals for an 90° off axis-source. Once again there is no time difference between the signals, however there is a substantial difference in level, as indicate by the sizes of the waveforms.



(a) XY Front.



(b) XY Side.

Figure 3.5: Phase and level differences of XY technique.

3.4.2.2 Blumlein Pair

The Blumlein pair is a special variation of the XY technique that employs bi-directional microphones, set perpendicular to each other (Pulkki, 2002:3), as shown in Figure 3.6. It was named after Alan Blumlein who first proposed the technique in 1934 (Streicher & Dooley, 2002:18).

This configuration may be worked from either the front or the back with equal effect, however the stereo channels from the rear are reversed and must be

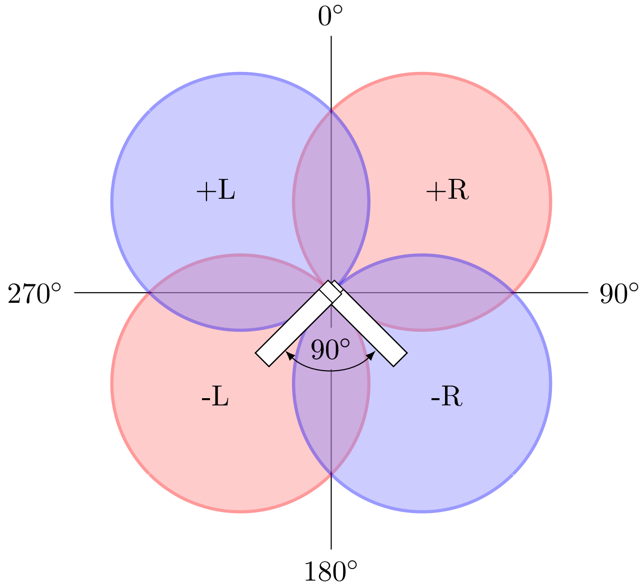


Figure 3.6: Blumlein pair microphone technique.

accounted for when arranging the stereo image (Streicher & Dooley, 2002:19). Either side of the Blumlein pair represent out-of-phase regions, where sounds are picked up by lobes with opposite polarity (Rumsey, 1999:2.82). Sound picked up in these regions will suffer cancellation and must be avoided, as it may become vague and difficult to localize, or cancel out entirely when summed to mono (Streicher & Dooley, 2002:19)

The Blumlein pair gives a very accurate correlation between the actual angle of the source and the apparent position of the stereo image when reproduced on loudspeakers (Rumsey, 1999:2.82). It often yields an excellent pick-up of the overall ambiance of a studio or concert hall (Huber & Runstein, 2010:142).

3.4.2.3 MS

The mid/side (MS) technique employs one microphone (middle component) aimed directly at the centerline of the sound source, and a bi-directional microphone (side component) orientated laterally (Streicher & Dooley, 1984:5), as illustrated in Figure 3.7. Weighted sums performed on the signals, create two virtual microphones with which a stereo image is generated (Pulkki, 2008:757).

Consider a bidirectional microphone with its 0° axis orientated to the audience left direction and its perpendicular axis aimed at a source. Sound picked up from either lobe of its polar pattern is denoted $+L$ and $-R$, where the sign indicates the polarity. The signal received from this microphone may therefore be represented as $S = L - R$. The signal from the center microphone of the MS configuration is simply $M = L + R$, since the polarity of its polar pattern is uniform (Boyce, 2014:269).

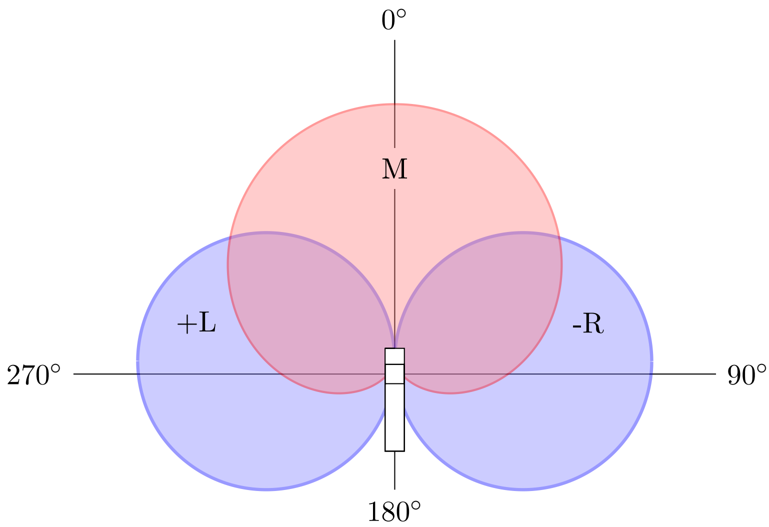


Figure 3.7: MS microphone technique.

To extract the separate left and right channels, the middle and side components are summed as shown in Equations (3.4.1) and (3.4.2). It is possible to perform this summation using three channels, or alternatively by using a dedicated sum-difference matrix (Rumsey, 2001:166, Eargle, 2005:173).

$$\begin{aligned} M + S &= L + R + L - R \\ &= 2L \end{aligned} \tag{3.4.1}$$

$$\begin{aligned} M - S &= L + R - (L - R) \\ &= 2R \end{aligned} \tag{3.4.2}$$

A great advantage of the MS technique that results from the summing procedure, is that the stereo width may be controlled by changing the relative levels of the mid and side components. This may even be done post-recording (Bartlett & Bartlett, 2007:236).

The MS technique was first defined by Alan Blumlein in the 1930s and subsequently developed by Holger Lauridsen in the 1950s for Danish radio. It has been known to offer an extremely accurate and high quality stereo image as well as unsurpassed monophonic compatibility (Streicher, 2002:3). Although the side microphone must always be bi-directional, the center microphone may be of any front orientated polar pattern. The output results will, however, vary accordingly (Pizzi, 1984:11).

3.4.3 Near-Coincident Techniques

Near-coincident configurations refer to stereo microphone configurations where the microphones are close enough together to be essentially coincident at low frequencies, yet far enough apart to appreciate the time delay between channels for sources to the far left or right. As a result, these techniques develop stereo imaging from level and time differences (Streicher & Dooley, 1984:6). The distance allows for some phase difference between the two signals, but is not enough to lose mono compatibility (White & Louie, 2005:256)

3.4.3.1 ORTF

The ORTF consists of two cardioid microphones, spaced 17 cm apart, with a base angle of 110°, as shown in Figure 3.8 (Pulkki, 2002:4). Due to the greater angle, the ORTF creates a wider stereo image while maintaining good mono compatibility (Savage, 2011:26), but it loses some directionality of sounds (Fisher, 2012:81). The acronym stands for *Office de Radiodiffusion Têlêvision Fraiņçaise*, where the technique was developed in the 1960's (Dochtermann, 2011:87).

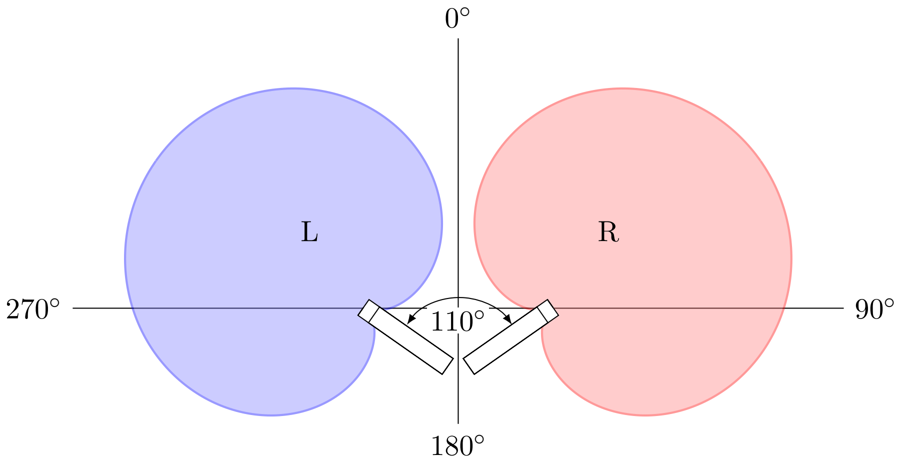
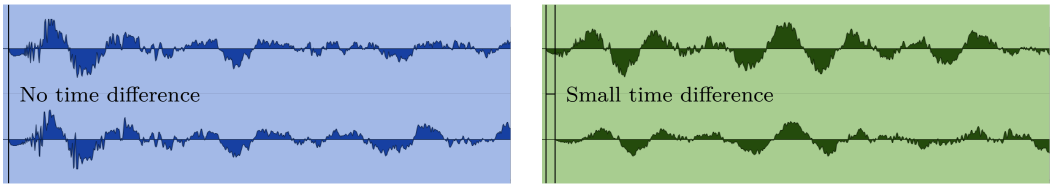


Figure 3.8: ORTF microphone technique.

A variation of the ORTF is the NOS (Nederlandse Omroep Stichting), developed by the Dutch Broadcasting System (Kefauver, 2001:112). The microphone pair is spaced 30 cm apart, but with an included angle of 90° between microphone axes (Ceoen, 1971:4). When summed to mono, this configuration yields an attenuated low frequency response due to phase differences induced by the greater distance between the microphones (Franz & Lindsay, 2004:108).

The experiment previously mentioned was continued for the ORTF configuration. For the on-axis source, Figure 3.9 shows no level or time differences. However, for the off-axis source, Figure 3.9b shows a level difference as well as a small time difference between the left and right signals of the ORTF configuration. Details of the experiment are given in Appendix B.2.



(a) ORTF Front.

(b) ORTF Side.

Figure 3.9: Phase and level differences of ORTF technique.

3.4.3.2 Faulkner Phased-Array System

Developed by the British recordist Tony Faulkner (Streicher & Dooley, 1984:551), this technique uses two bi-directional microphones, facing the sound source with parallel axis and spaced 20 cm apart (Bartlett & Bartlett, 1999:135). It relies entirely on time cues to convey the localization of the sound sources, except very close to sources, where parallax[1] comes into play (Eargle, 1994:166). According to Faulkner, the array is not mono compatible in theory, but in practice it has presented no problems (Bartlett & Bartlett, 2007:240).

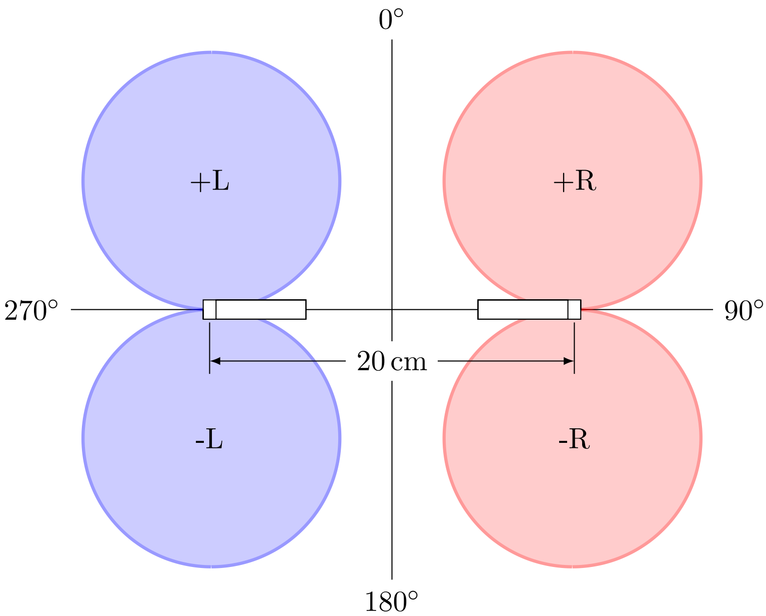


Figure 3.10: Faulkner phased-array microphone technique.

[1] The difference in apparent direction of an object as seen from two different view points (Heilbron, 2005:243).

3.4.4 Spaced Pair

A spaced pair consists of two identical microphones, placed on parallel axes and separated by a distance ranging from 20 cm up to a few meters (Pulkki, 2002:3, Oswinski, 2009:92). It was the first technique used to relate a stereo image (Streicher & Dooley, 1984:8).

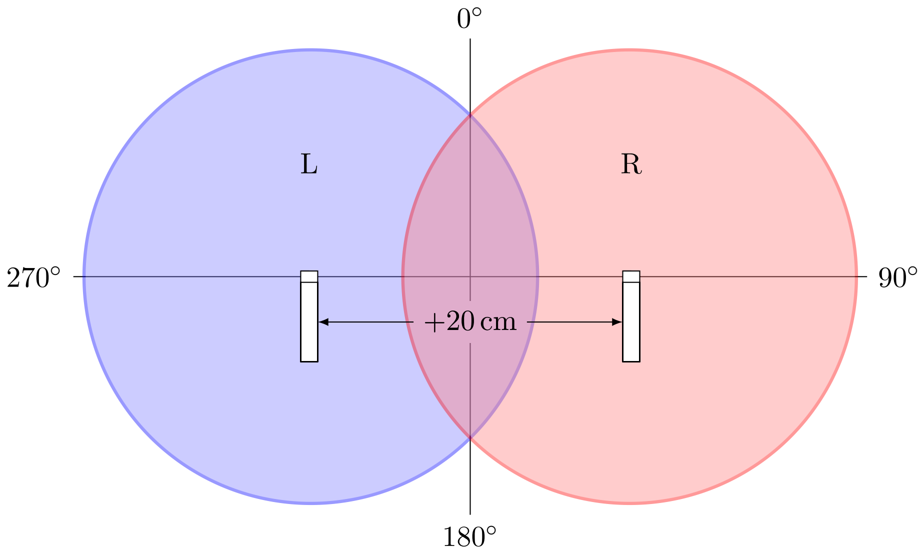
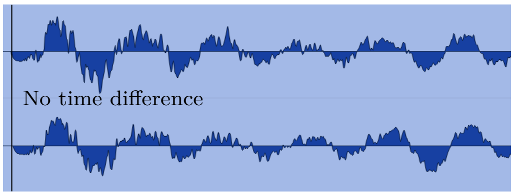


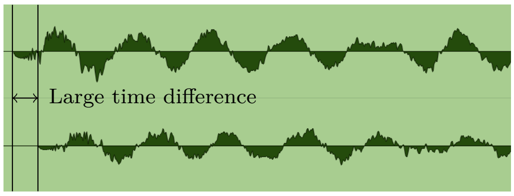
Figure 3.11: Spaced pair microphone technique.

The microphones may have any polar pattern, but omni-directional are the most popular choice (Oswinski, 2009:64). Localization is determined by level as well as time differences between the two signals (Sharma, 2003:587). The greater distance between the microphones create large phase differences and as a result, this technique is characterized by increased spaciousness, vague stereo imaging and poor localization (Tagg, 2012). Due to the large phase differences, this technique is not mono compatible (Izhaki, 2013:193, Bartlett & Bartlett, 1999:123).

The experiment to illustrate the time and level differences of stereo microphone configurations was concluded with the spaced pair technique. Figure 3.12a and Figure 3.12b show the time and level differences that occurred between signals for an on-axis and a 90° off-axis source, respectively. Figure 3.12b shows very large time and level differences, illustrating why this technique is not considered mono compatible. For spaced pair configurations with greater distances between microphones, these time and level differences will be even greater. Details of the experiment are given in Appendix B.2.



(a) Spaced pair front.



(b) Spaced pair Side.

Figure 3.12: Phase and level differences of spaced pair technique.

3.4.5 Accent Microphone Placement

Accent microphone placement is used to supplement stereo and surround sound microphone techniques. It involves the use of a microphone to capture only a single sound source, or small group of sources, within the total ensemble that is being recorded (Moylan, 2007:294). Accent microphone placement is alternatively known as *spot miking*.

It is a relatively close microphone placement technique in that the placement must be close enough to emphasize the source, but still distant enough to capture a natural sound (Alten, 2013:407). A good accent microphone should add presence to the source, but not be identifiable as a separate pick-up within the mix (Huber & Runstein, 2010:140)

Chapter **4**

The Effect of Microphone Placement on Timbre

Instruments are very complex radiators that project sound energy multidirectionally and in different and constantly changing proportions of spectral density (Woszczyk, 1979:2). Their frequency spectra vary with distance and radiation angle. Therefore, the frequency spectrum picked up by a microphone varies with its placement (Bartlett, 1981:2). The placement of a microphone plays an important role in creating the desired sound and is one of the recordist's most valuable tools (Huber & Runstein, 2010:132). Even tiny adjustments of angle and distance can dramatically alter the timbre of the sound (Gore, 2014). When considerable attention is paid to the exact microphone placement relative to the instrument, it can be used as a primary determinant of the timbre of the recorded sound (Moulton, 1990:162).

Woszczyk (1979:3) believes it is most frequently the microphone placement that allows the recordist to become an interpreter of the sound produced by an instrument. Wuttke (1999:4) believes placing a microphone appropriately with regard to the radiation pattern of the instrument to be recorded, is substantially more important than the type of microphone that is used.

The possibility of timbre control in recording exists through the control of several variables. These variables include the directivity of the sound radiation from the instrument, the position of the instrument relative to the microphone and its surroundings, the directional characteristics of the microphone, the distance between the microphone and the instrument, and the acoustic characteristics of the room (Woszczyk, 1979:7).

The sounds radiating from various parts of the instrument combine into a complete audio picture at some distance from the instrument (Boudreau *et al.*, 2005:27), because most musical instruments are designed to sound best at a distance, for instance, where the audience sits (Bartlett, 1981:1). Therefore, a close microphone position may not accurately reproduce the sound of the source, and equalization may be required to achieve a sound similar to the natural sound of the source (Ballou *et al.*, 2008:582).

To further understand how timbre changes with microphone placement, experiments were conducted. There are various factors that influence the timbre of a sound. It may be affected by the properties of the transmission medium, the reflective and absorptive characteristics of the environment and the operation of the sound generating elements of the source itself. The placement of the microphone relative to the source and the environment has a major influence in the recorded timbre as well. This presents a vast number of variables that make a general analysis of the effect of microphone placement of timbre very difficult.

A test signal, played through an electric guitar amplifier, was recorded with various microphone configurations and a frequency analysis was conducted to determine the frequency spectrum for each microphone placement. Such an experiment has been conducted by Case (2010), however, since his article does not describe the room or surroundings that the tests were conducted in, a new experiment was conducted for the sake of thoroughness.

4.1 Equipment

4.1.1 DAW and Interface

Avid Pro Tools 10.3.2 was used in conjunction with an Avid HD I/O audio interface. The Avid HD I/O features 24-bit digital-to-analog and analog-to-digital converters and supports sample rates of 44.1 kHz, 48 kHz, 96 kHz, 88.2 kHz, 96 kHz, 176.4 kHz and 192 kHz (Cook, 2013:83; Avid Technology Inc., 2010*b*:5). Complete specifications for the Avid HD I/O may be found in Appendix C.1.

4.1.2 Re-amp Box

A custom built re-amp box, designed by Scott Dorsey (2014), was used to unbalance the Pro Tools output and match the impedance of the guitar amplifier input. The re-amp box is shown in Figure 2.8 on page 39. This unit features a potentiometer that adjusts the output volume and a switch that toggles between a 1 MΩ purely resistive source and a 10 kΩ source with 0.1 H inductance. These circuits form very simplified models of guitar pickups (Dorsey, 2014). Full specifications for the re-amp box are given in Appendix C.2.

4.1.3 Guitar Amplifier

A Fender Pro Junior III guitar amplifier was used. It features a 15 W, class A valve amplifier and a 25.4 cm Fender Special Design loudspeaker

(Fender Musical Instruments Corporation, 2010:7). Valve amplifiers add harmonic content to the signal it receives and are characterized by a *warm* timbre (Denyer *et al.*, 1992:201). Full specifications are listed in Appendix C.3.

4.1.4 Microphone

A Heil PR 20 moving coil microphone was used for the experiment recordings. Moving coil microphones are robust and capable of handling high SPL (Talbot-Smith, 1999:2.38; Thompson, 2005:15). They are the most popular microphone choice for use with electric guitar amplifiers (Slone, 2002:28). The Shure SM57 is very common choice for this application, but for this experiment the Heil PR 20 was favoured for its greater frequency range. The frequency response of the Heil PR 20 extends from 50 Hz to 18 000 Hz, where the Shure SM57 is limited to a range of 50 Hz to 15 000 Hz (Heil Sound Ltd, 2010;Shure Incorporated, 2010:11). Specifications for the Heil PR 20 are listed in Appendix C.4.

4.1.5 Preamplifier

A Buzz Audio MA2.2 preamplifier was used to amplify the microphone signal to line level. The Buzz Audio MA2.2 features 65 dB gain with a signal-to-noise ratio of −74 dBu. The full specifications are listed in Appendix C.5.

4.1.6 Sound Level Meter

A Radio Shack Digital Sound Level Meter was used to measure the output of the guitar amplifier. This particular model has a range of 50 dB to 126 dB, with and accuracy of ±2 dB at 114 dB SPL. The sound level meter is shown in Figure 4.1. Holman (2008:70) praises the RadioShack sound level meter as a standard for the film industry. Full specifications are listed in Appendix C.6.

4.2 Experiment Layout

The experiment was conducted at Stellenbosch University Studios, in recording room known as the *Submarine*. The Submarine is an asymmetrical, quadrilateral room with a surface area of 39.5 m^2 and no parallel surfaces. The entire room is isolated from the rest of the building by a rubber shell. The interior consists of hard walls treated with with movable sound absorbers, a hard odd shaped ceiling and a soft tiled floor. A large double window connects the studio to the main control room. The combination of hard surfaces, its asymmetrical shape and well-spaced absorbers gives the submarine has a nat-

Figure 4.1: RADIOSHACK DIGITAL SOUND LEVEL METER.

ural and compact sound with a controlled reverberation time of roughly 0.6 seconds (Roux, 2012).

A signal chain diagram of the experiment is shown in Figure 4.2. To ensure repeatability, a guitar amplifier was chosen as the sound source. A musician may, to some extent, control the timbre of an instrument or may move the instrument while playing (Davies, 2011:161). Whether intentional or not, either case was undesirable as it may have introduced anomalies into the recorded audio. Through re-amp technology, a guitar amplifier was fed with a test signal, thereby acting as a sound source that could produce the exact same performance on demand. A photograph of the setup is shown in Figure 4.3. The guitar amplifier was placed on the floor, 1 m from the rear wall, equidistant from the side walls and facing down the length of the room.

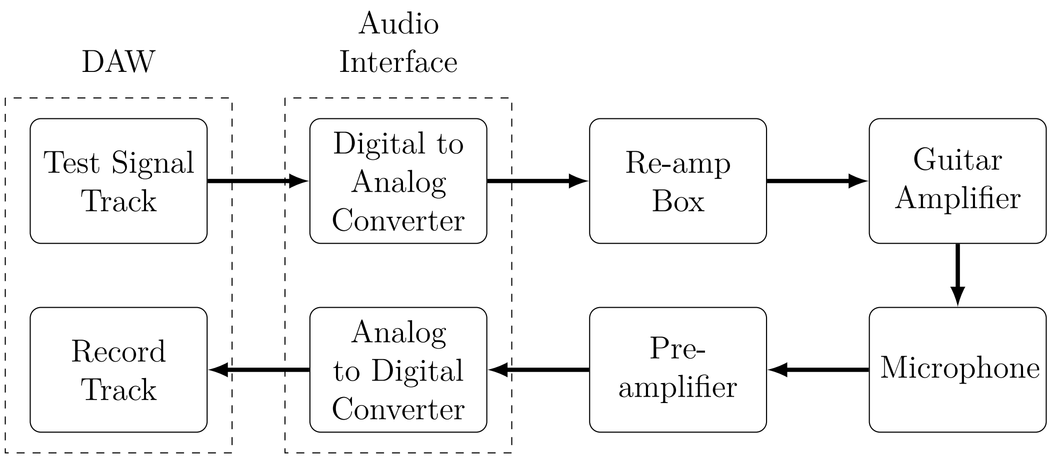


Figure 4.2: Experiment setup.

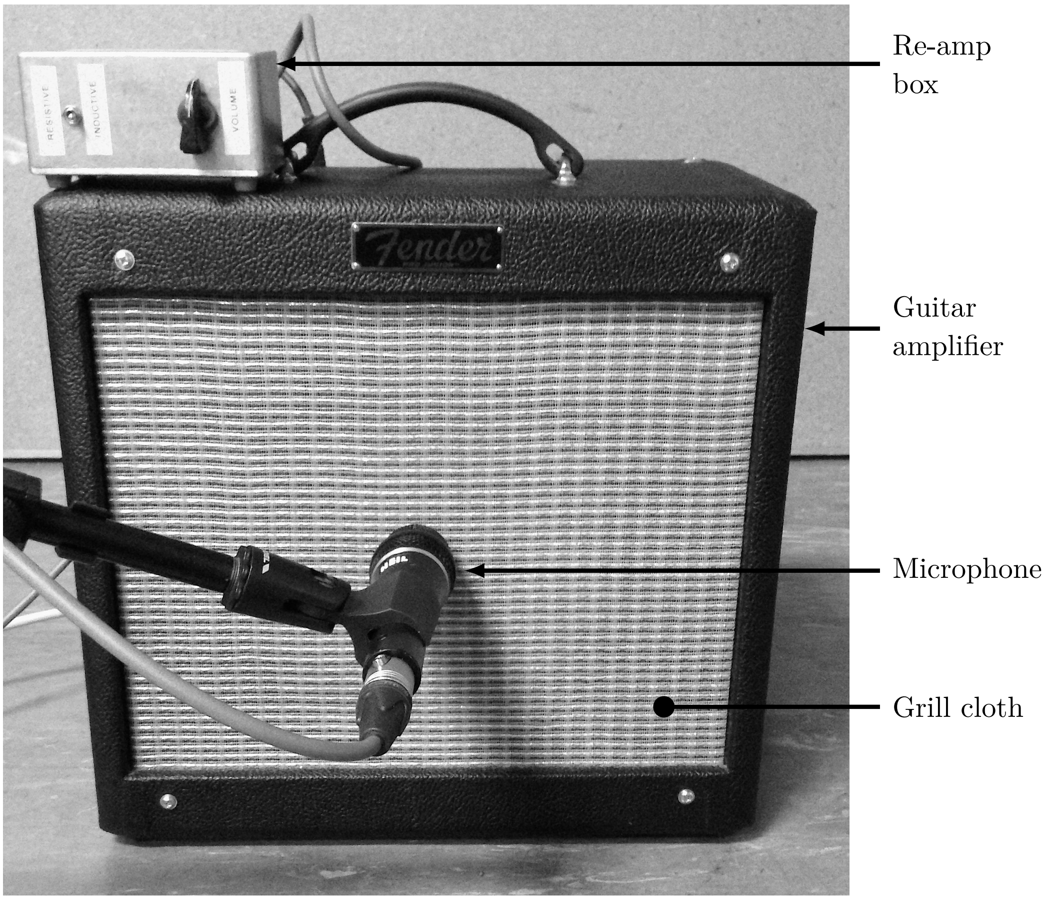


Figure 4.3: Photograph of experiment setup.

Low frequencies radiating from a loudspeaker are effectively omni-directional, because the wavelength of the sound is large compared with the dimensions of the loudspeaker and its enclosure. This results in efficient diffraction of sound around the speaker enclosure (Rumsey & McCormick, 2006:97). However, as the frequency increases, the radiation of sound from the loudspeaker becomes more directional, effectively being beamed tighter on the axis of the loudspeaker (Holland, 2001:15).

A useful technique for the analysis of the radiation of sound from a loudspeaker diaphragm, is to replace the vibrating diaphragm with a distribution of equivalent point monopole sources (Holland, 2001:11-12). Assuming that all of the point sources vibrate in phase, the model may be simplified further to a planar source, such as a rigid vibrating disk (Beranek & Mellow, 2012:553). This is the most commonly used loudspeaker model (Kärkkäinen & Mellow, 2005:1) and is known as the *baffled piston* model. The *directivity function* D for such a rigid circular piston mounted in an infinite baffle, is given by Equation (4.2.1), where k is the wavenumber[1] and a is the radius of

[1] The wavenumber is defined as the number of alternate positive and negative cycles that

the piston (Beranek & Mellow, 2012:556). The product of the wave number and the piston radius ka is a convenient dimensionless value that represents frequency. Figure 4.4 shows the normalized directivity function $20\log_{10}|D(\theta)|$ for ka values ranging from 0.5 to 20 and illustrates the increased directivity of a loudspeaker with increasing frequency.

$$D(\theta) = \frac{2J_1(ka\sin\theta)}{ka\sin\theta} \tag{4.2.1}$$

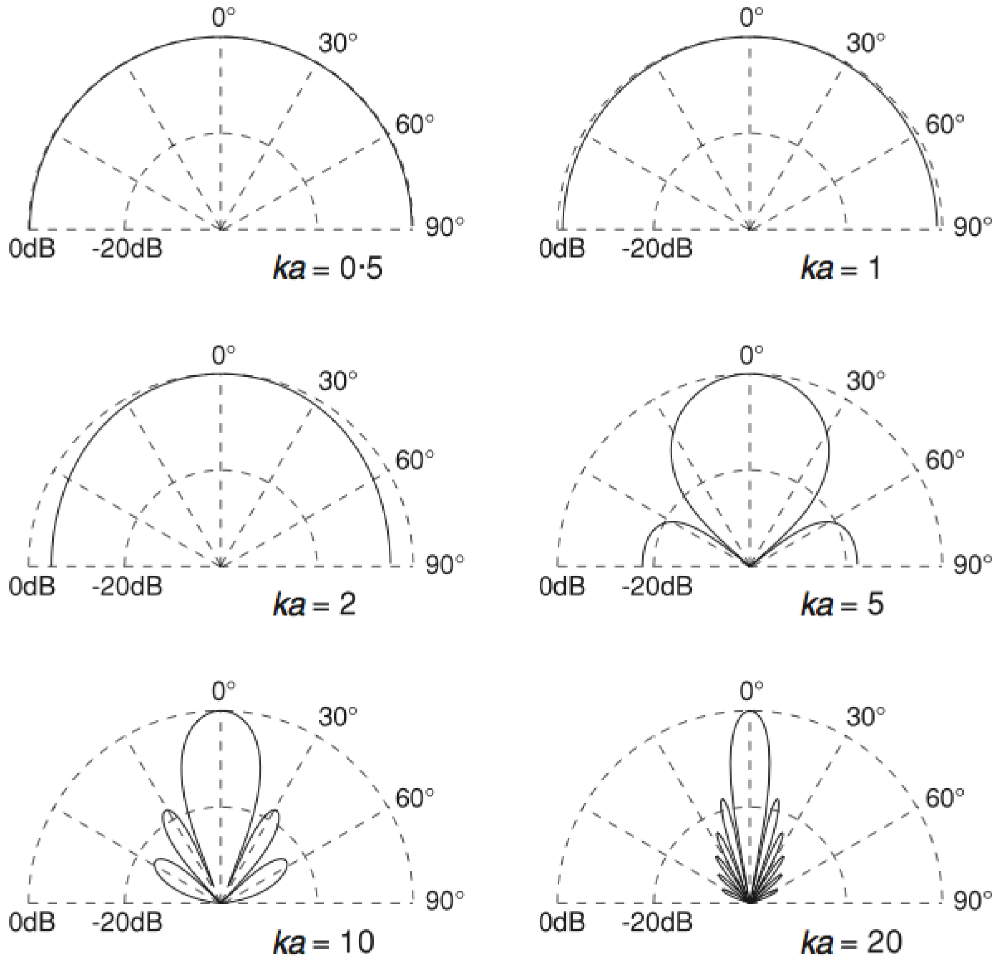


Figure 4.4: Directivity of a rigid piston mounted in an infinite baffle, for ka values of 0.5, 1, 2, 5, 10 and 20 (Holland, 2001:14).

occur in a given distance; it has the units of radians per meter and usually has the symbol k. It is directly related to the speed of sound through the equation $k = c/\omega$, where c is the speed of sound (Holland, 2001:11).

When ka becomes greater than 3, the piston becomes highly directional (Beranek & Mellow, 2012:555). The directional characteristics of a loudspeaker is akin to the multi-directional radiation of real musical instruments. The high degree of control and repeatability of a re-amped guitar amplifier, in combination with its directional characteristics, made this configuration ideal for measuring frequency spectra at various microphone positions.

The output of the guitar amplifier was calibrated to 110 dB SPL using RadioShack Digital Sound Level Meter. The sound level meter settings for this calibration are summarized in Table 4.1. The test signal for the experiment consisted of a logarithmic swept sine wave, ranging from 20 Hz to 20 000 Hz, over a period of twenty seconds. This type of signal approximates the frequency response of pink noise, but has the great advantage that the total power of signal is devoted to a single frequency at any given time. For more information on noise and swept sine wave as test signals, refer to Appendix D.

Table 4.1: Guitar amplifier calibration.

Parameters	
Input signal	Pink noise
Weighting	C
Response	Slow
Range	100 dB to 110 dB
Measurement type	Continuous average
Measured value	110 dB

4.3 Procedure

Three experiments were conducted to investigate the effect that the distance from the guitar amplifier, the radial distance from the guitar amplifier axis and the angle of the microphone has on the frequency spectrum of the recorded sound. The experiments and the variables are summarized in Table 4.2. The variables of interest are also illustrated in Figure 4.5.

The microphone starting position was chosen to be right up against, but not touching, the grill cloth[2] of the amplifier, on the loudspeaker axis and with the microphone perpendicular to the amplifier. This corresponds to the origin of the axis in Figure 4.5. The first variable in question was the distance between the microphone and the loudspeaker, x. From the starting position,

[2] The grill cloth is a thin material mesh than covers the front of a guitar amplifier or loudspeaker cabinet.

Table 4.2: Experiments

Experiment	**Variable of interest**	**Symbol**
Distance	Distance between amplifier loudspeaker and microphone.	x
Radius	Radial distance between amplifier loudspeaker axis and microphone.	r
Angle	Microphone angle relative to loudspeaker axis.	θ

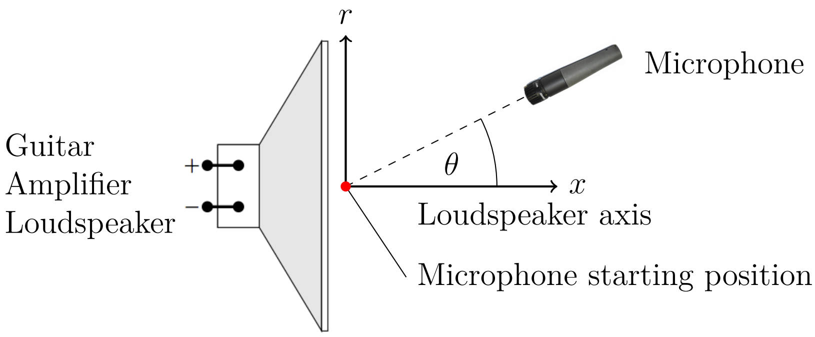


Figure 4.5: Variables of interest.

test recordings were made at incremental distances from the guitar amplifier. The microphone was kept on-axis and perpendicular to the guitar amplifier, only the distance from the amplifier was varied. Returning to the starting position, the process was repeated for the variable, r. The microphone was kept perpendicular to the guitar amplifier and at a constant distance from the grill cloth, while the radial distance from the center of the loudspeaker was incremented.

Finally, the microphone was kept at the center of the loudspeaker and a fixed distance from the grill cloth while recordings were made with the microphone at incremental angles relative to the axis of the loudspeaker. A summary of the experiment recordings is listed in Table 4.3.

Table 4.3: Summary of experiment recordings.

Variable	x	r	θ
Initial value	0 cm	0 cm	0°
Increment	8 cm	2 cm	15°
Final value	56 cm	14 cm	90°
Number of recordings	8	8	7

4.4 Frequency Analysis

The recordings were subjected to frequency analyses in order to determine the frequency spectrum at each microphone position. Frequency analysis, also referred to as spectrum analysis, shows the energy distribution of a signal over a range of frequencies (Fries & Fries, 2005:245; Holman, 2010:7). The background for frequency analysis is the *Fourier transform* (Bandyopadhyay, 2005:374), a technique which converts signals from the continuous time-domain to the corresponding frequency-domain and is applicable to both periodic as well as aperiodic signals (Gurung, 2009:130).

The Fourier transform, $\mathcal{F}$, of a signal, $x(t)$, is expressed in Equation (4.4.1), where $\omega = 2\pi f$ is the angular frequency. (Chen & Ling, 2001:26). The *Inverse Fourier Transform*, $\mathcal{F}^{-1}$, performs the opposite function as the Fourier transform and allows the corresponding time-domain signal to be determined from a given frequency response (Kester, 2005:322). The inverse Fourier transform is shown in Equation (4.4.2)

$$X(\omega) = \mathcal{F}\{x(t)\} = \int_{-\infty}^{\infty} f(t)e^{-j\omega t}dt \tag{4.4.1}$$

$$x(t) = \mathcal{F}^{-1}\{X(\omega)\} = \frac{1}{2\pi}\int_{-\infty}^{\infty} X(\omega)e^{j\omega t}d\omega \tag{4.4.2}$$

For non-continuous time signals that are characterized by a series of points extracted at equal time intervals, such as digital audio, the analysis is performed using the *Discrete Fourier Transform* (DFT) (Bandyopadhyay, 2005:374). The DFT may be understood as a numerical approximation of the Fourier transform (Smith, 2007:xi). For a discrete sampled function $f(k)$, the DFT and Inverse DFT are given in Equations (4.4.3) and (4.4.4), where k refers to the samples in the time domain and n refers to the samples in the frequency domain (Cios *et al.*, 2007:181). The output of the DFT $X(k)$ is the sampled spectrum (Loy, 2011:115).

$$X(n) = \sum_{k=0}^{N-1} x(k)e^{-j2\pi nk/N} \quad \text{for } n = 0, 1, 2, ..., N-1 \tag{4.4.3}$$

$$x(k) = \frac{1}{N}\sum_{n=0}^{N-1} X(n)e^{-j2\pi nk/N} \quad \text{for } k = 0, 1, 2, ..., N-1 \tag{4.4.4}$$

There are several methods for calculating the DFT. The *Fast Fourier Transform* (FFT) is one such method. While it produces the same results as other methods, it is incredibly more efficient and computes the DFT much more rapidly (Smith, 2003:225; Bandyopadhyay, 2005:374). The FFT is used extensively in digital signal processing applications, including spectrum analysis, high-speed convolution, filter banks, signal detection and estimation, system identification and audio compression (Smith, 2007:xi).

For a discrete time-domain signal, the time between samples is known as the *sample period* and is given by the inverse of the *sampling rate* or *sampling frequency*, $f_s = 1/\Delta t$. (Dimpker, 2013:211). The sampling rate is expressed in Hertz (Hz) (Oja & Parsons, 2009:198). The time of the whole signal is given by $T = (N-1)\Delta t$. The FFT of a discrete time-domain signal is illustrated in Figure 4.6.

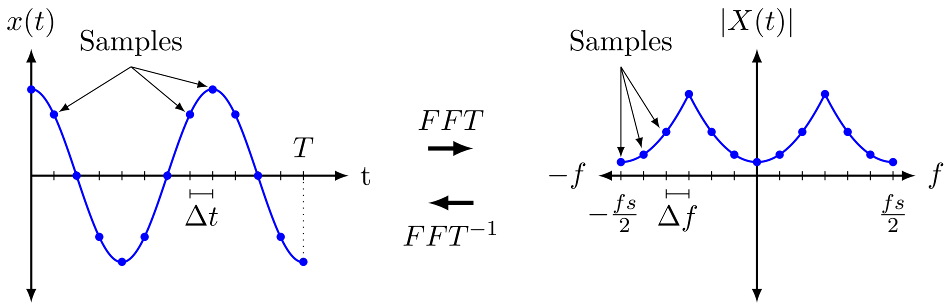


Figure 4.6: Illustration of the transformation between a discrete time signal and a discrete magnitude frequency spectrum, using the FFT.

The output of the FFT is a set of complex numbers $X(n)$ that retain information on the magnitude and phase of all the frequencies within the spectrum. The frequency values in which the FFT is calculated and the distance between adjacent bins is equal to $\Delta f = f_s/N$ (Gatti & Ferrari, 1999:769). The magnitude is obtained by taking the absolute value of the spectrum, $|X(n)|$. Any sampling is limited in the bandwidth of the signals it can represent. The highest frequency that may be represented accurately for a given sampling rate is $fs/2$ and is known as the *Nyquist frequency*. The bandwidth from 0 Hz to $fs/2$ is the Nyquist bandwidth (de Silva *et al.*, 2005:15.77).

When an analog signal is not sampled at the minimum Nyquist frequency, aliasing occurs and the sampled data does not accurately represent the true signal (Oshana, 2006:66). Aliasing is the phenomenon where high frequency signals inject energy into lower frequencies, causing distortion of the actual signal (Havskov & Alguacil, 2010:104).

As illustrated in Figure 4.6, the FFT computes a symmetrical two-sided spectrum containing negative and positive frequencies. However, the negative frequency information is redundant and is therefore discarded (Norton, 2009:512). Plotting the magnitude of the spectrum $|X(n)|$ against frequency yields the magnitude frequency spectrum. As per example, the frequency spectra of a logarithmic and a linear swept sine wave was calculated. The resulting plot is shown in Figure D.4. The MATLAB code used to calculate the frequency spectra is given in Appendix E.3.1.

4.5 Experiment Results

4.5.1 Microphone Distance

The results of the distance experiment are shown in Figure 4.7. With each increment, a decrease in signal level is noticeable which is especially prominent in the low frequency range. The overall reduction in level is due to the loss of energy as the sound propagates. However, the greater level reduction in the low frequency range may be attributed to a loss of proximity effect, which decreases as the microphone is moved further from the source. In order to more clearly illustrate the change in frequency spectrum with microphone placement, the data was normalized relative to that of the microphone starting position. The resulting graph is shown in Figure 4.8.

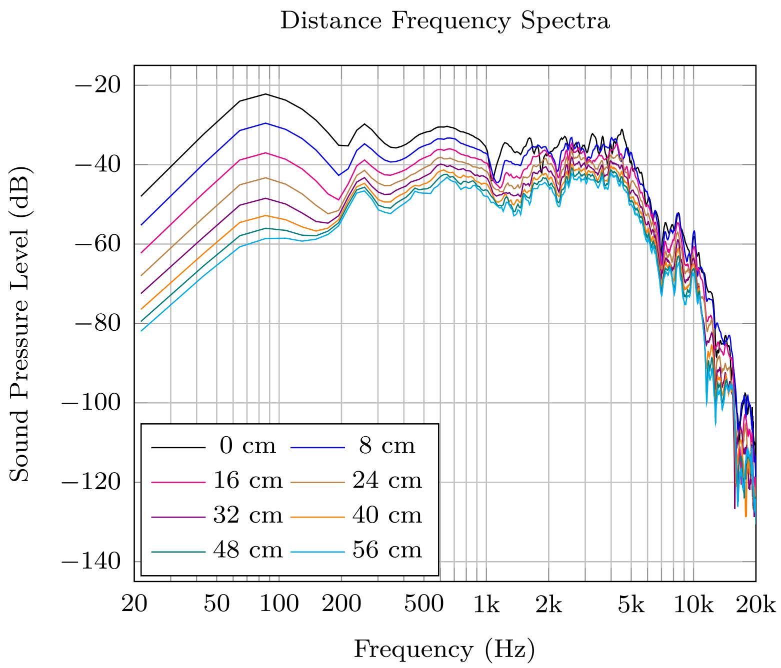


Figure 4.7: Frequency spectra at incremental distances from guitar amplifier.

The frequency spectrum becomes more erratic as frequency increases, with localized boosts and attenuations starting just above 1000 Hz. As the microphone is moved further from the source, the effects of the recording environment become noticeable. Reverberation as a result of late reflections increases the average level of signal picked up by the microphone (Alten, 2012*b*:217). Additionally, comb-filtering starts to occur. Comb-filtering is the result of reflections from the wall and floor that interfere with direct sound approaching the microphone.

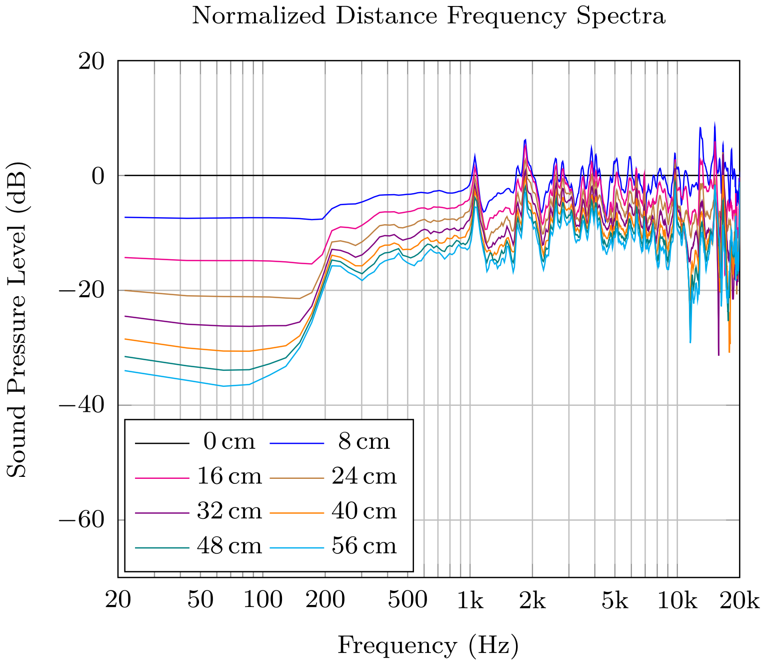


Figure 4.8: Normalized frequency spectra at incremental distances from guitar amplifier.

4.5.2 Radial Microphone Distance

The results of the radius experiment are shown in Figure 4.9. The normalized frequency spectra are shown in Figure 4.10. For microphone positions under 10 cm from the center of the loudspeaker, there was very little difference in level in the low and midrange frequencies. However, at 12 cm and 14 cm where the microphone was just on or beyond the edge of the loudspeaker, 8 dB and 10 dB attenuation occurred respectively in the low frequency range. In the midrange, the attenuation was less prominent at 4 dB and 8 dB. The loudspeaker of this amplifier was contained within a wooden baffle. At distances of 12 cm and 14 cm from the center of the loudspeaker, the microphone was on the edge or completely in front of this baffle, which resulted in a reduction in level. Upwards of 1000 Hz the frequency spectrum becomes even more erratic than in the distance experiment. This may be attributed to the formation of modes within the loudspeaker diaphragm.

A loudspeaker diaphragm is actuated by a moving coil. At low frequencies, the period of the signal is long compared to the speed of propagation through the diaphragm and the entire diaphragm essentially moves in the same phase. However, the propagation speed of vibrations through the loudspeaker diaphragm is finite. As frequency increases, the finite propagation speed results in phase shifts between different parts of the diaphragm (Watkinson, 2001*c*:49). These phase shifts result in the formation of modal patterns in the diaphragm that are related to frequency. Figure 4.11 illustrates a loudspeaker experienc-

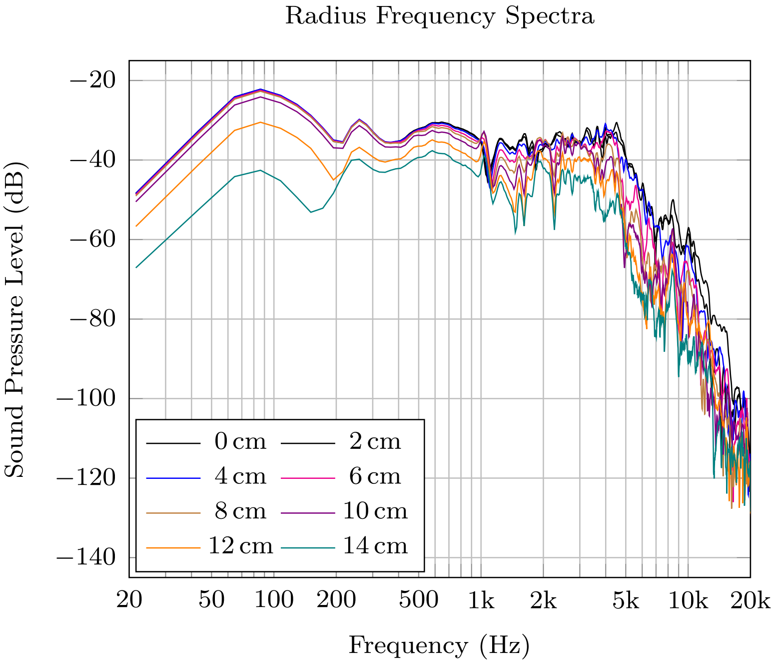


Figure 4.9: Frequency spectra at incremental radial distances from guitar amplifier.

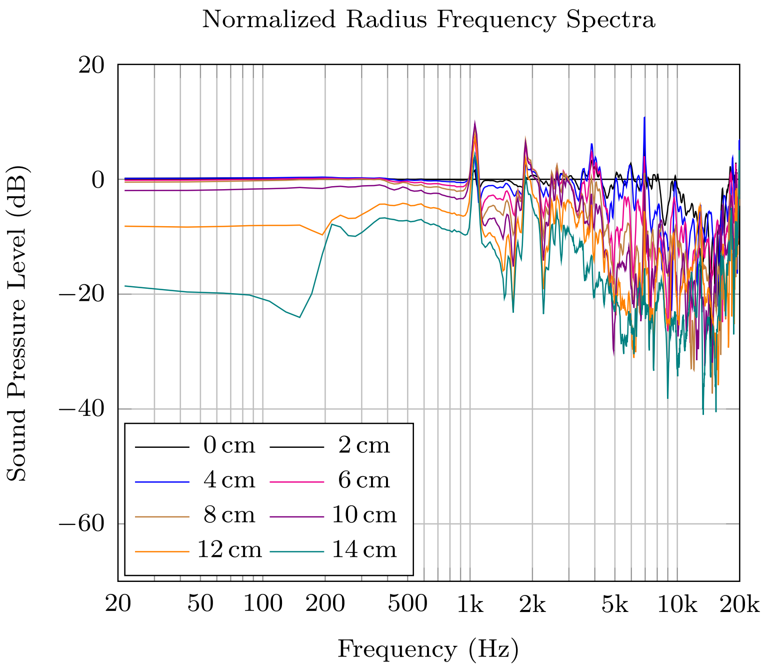


Figure 4.10: Normalized frequency spectra at incremental radial distances from guitar amplifier.

ing modal patterns at various frequencies. These illustrations are snapshots of the loudspeaker diaphragm as viewed from the front, with the positive and negative signs indicating outward and inward motion, respectively (Eargle, 2003:28).

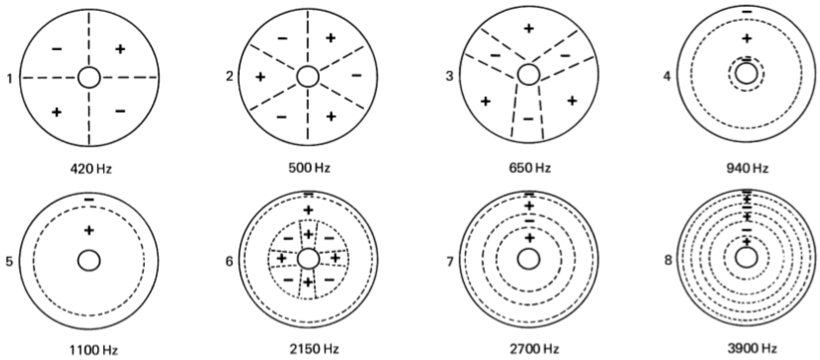
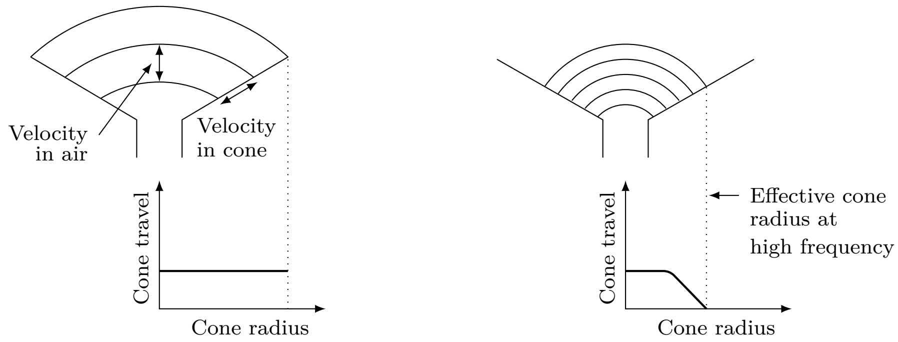


Figure 4.11: Example of nodal patterns of a loudspeaker diaphragm (Watkinson, 2001*c*:51).

At low frequencies, the modes are pie-shaped and are known as *radial* modes. As frequency increases, the modes begin to form rings around the center of the loudspeaker, known as *concentric* modes. The modes of the loudspeaker emit sound waves of differing phase. Where these waves coincide, destructive or constructive interference occurs, resulting in boosts and attenuations in the frequency response. Case (2010:81) describes these as "*pockets of attenuation*," although pockets of interference is more appropriate as both constructive and destructive interference occurs.

Figure 4.10 shows a high frequency attenuation in the range of 2 dB to 4 dB with each radial increment, resulting in a total decrease between the first and last positions of roughly 20 dB. As frequency increases, the modal patterns caused by phase shifts between different parts of the diaphragm eventually result in concentric modes. These concentric modes travel outward along the radius of the diaphragm, but some are reflected back and result in complex constructive and destructive interference patterns. These interference patterns are called *cone break-up* patterns and cause a ragged frequency response in loudspeakers (Gottinger, 2007:221).

The number of concentric modes increase with frequency and reduce the distance with which the outer edge of the cone travels, as illustrated in Figure 4.12. As a result, the effective radiating area of the loudspeaker diaphragm decreases so that high frequencies radiate only from the center (Gottinger, 2007:221; Fahy & Gardonio, 2007:145). The directivity of the loudspeaker is wide at low frequencies, but as frequency increases it narrows considerably, forming a beam on the axis of the diaphragm (Evans *et al.*, 2009:2). The reduction in the high frequency content in Figure 4.10 may be attributed to the increased directivity of the loudspeaker.



(a) Low frequency. (b) High frequency.

Figure 4.12: The effective loudspeaker diameter as a function of frequency (Watkinson, 2001*c*:53).

4.5.3 Microphone Angle

The frequency spectra for incremental microphone angles are shown in Figure 4.13. The normalized frequency spectra are shown in Figure 4.14. A small level difference occurs with each increment, which is especially apparent in the low frequency range due to loss of proximity effect. For a cardioid microphone set 90° off-axis, the diaphragm is perpendicular to the source and sound reaches either side of it simultaneously. Therefore, there is no gradient across the diaphragm and no proximity effect occurs (Eargle, 2002*b*:67). At 90°, a cardioid microphone behaves like an omni-directional microphone (Josephson, 1999:6). The trend of erratic high frequency spectra continued and an attenuation of roughly 2 dB to 5 dB per increment occurred.

The 90° position of the microphone corresponds to the *hanging method*, shown in Figure 4.15, which involves hanging the microphone by its cable from the handle of the guitar amplifier or cabinet. This technique may be employed because of a limited number of microphone stands, but there are valid reasons that warrant its use. The off-axis response of a cardioid microphone is only 3 dB to 6 dB less than the on axis response, therefore a high level is still achievable (White, 2014:195). The effects of off-axis coloration are most prominent in this position and may result in a reduction of high frequency content. However, this may be an ideal method for achieving a more balanced response from an overly bright guitar amplifier.

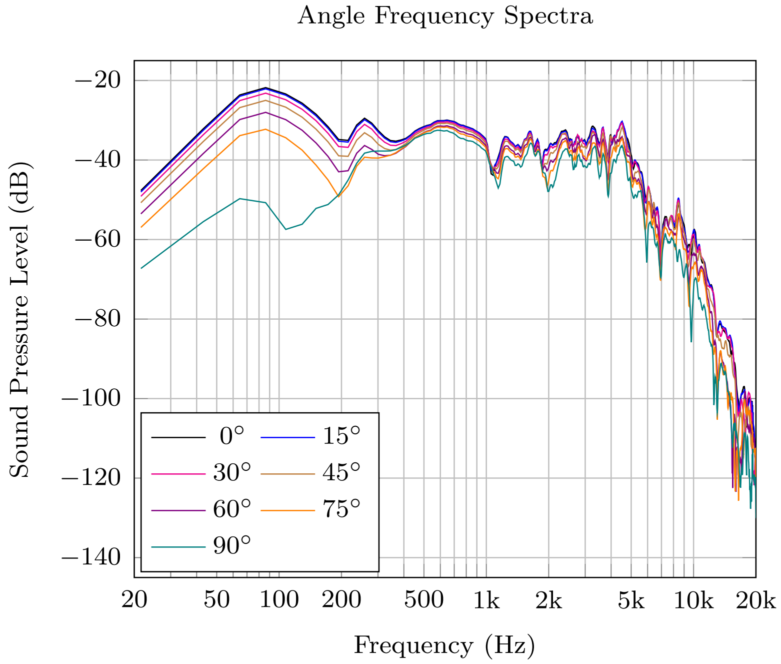


Figure 4.13: Frequency spectra at incremental microphone angles.

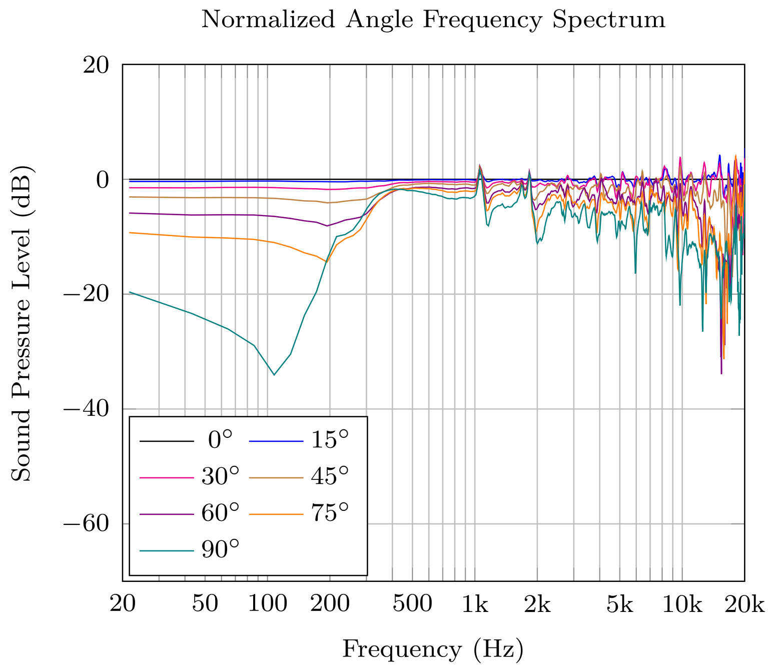


Figure 4.14: Normalized frequency spectra at incremental microphone angles

Figure 4.15: Hanging microphone technique.

4.6 Discussion of Results

The distribution of energy over the frequency range is one of the major determinants of timbre. However, timbre depends upon more than just the frequency spectrum of a sound (Moore, 2012:469). Any change in a waveform causes a change in the frequency content of the sound and therefore also change in its timbre (Truax, 2001:147). Therefore, the changes in recorded frequency spectra in Figures 4.7 to 4.10, 4.13 and 4.14 indicate that a change in timbre has occurred.

Increasing the distance between the microphone and the guitar amplifier substantially decreased the level of the recorded signal and introduced comb-filtering effects. There was a large reduction in low frequency content due to the loss of proximity effect. Off-axis placements tended to result in attenuated high frequencies due to the directional characteristics of loudspeakers, while increasing the angle of the microphone resulted in only a slightly more erratic response, which Case (2010:83) describes as *choppiness*. In both these cases, a loss of proximity effect also occurred.

Many factors explain the changes in frequency spectrum as a result of microphone placement. These may include comb-filtering, proximity effect, nearfield anomalies of the loudspeaker and asymmetric coupling to the front and rear of the microphone capsule. However, the recordist need only assess the impact of these effects on their production (Case, 2010:83). The colouring effects of the microphone itself along with those of its placement must be appropriately matched with both the source material and the stylistic expressive intensions of the project, a task requiring aesthetic judgment and technical expertise (Zak, 2001:109).

4.7 Evaluating Microphone Placement

To evaluate microphone placements, a short test recording may be made and listened to through either loudspeakers or a close-fitting pair of headphones (Dyar, 1961:49). Boudreau *et al.* (2005:5) suggests placing the microphone at various distances and positions until a placement is found that has the desired tonal balance and room acoustics when listened to over the studio monitors. If the sound is not desirable, the recordist must experiment with different positions, microphones, isolation of the instrument or different sounds of the instrument itself. As the recordist Bruce Bartlett states:

> *"... it pays to experiment with all sorts of microphone positions until a suitable sound is found."* - Bartlett (2012:2)

Microphone placement is, to a large degree, dependent on the technical understanding of acoustics, yet is a skill that can only be acquired through practice and trial and error, leading to the development of tacit knowledge (Watson, 2014:76). In addition to possessing technical and theoretical expertise, successful recordists possess the capacity to differentiate timbral, dynamic and technical details of sound and can translate their aural impressions into the appropriate technical judgments and alterations (Corey, 2012:ix). The capacity of the recordist to make appropriate choices at all stages of a project relies directly on the ability to listen critically to the audio material (Thompson *et al.*, 2013:1). *Critical listening* involves the evaluation of sound quality to define what is physically present, the identification of characteristic qualities of the sound being evaluated and the identification of any undesirable sounds or characteristics that influence the integrity of the audio signal (Moylan, 2014:158). Critical listening is performed from the optimum listening position, as described in Section 2.5 on page 43, from where the recordist experiences the correct stereophonic sound reproduction.

Recordists cannot rely on one set of recording procedures as each recording project has its own set of requirements. Instead, they must rely on a combination of technical knowledge and critical listening to guide their work (Corey, 2012:ix). In the final stages of microphone placement, the recordist makes fine adjustments, refining microphone position by a few millimeters and orientation by just a few degrees. The recordist employs past experience and known techniques by others, but inevitably relies on aural memory to guide him or her in finding suitable placements (Roginska *et al.*, 2012:8).

A popular technique for comparing microphone placements is the *A/B test*. A/B testing refers to comparing two versions of the same material by switching between them (Kroon, 2010:5). It is done between audio components as a way of determining if one is superior to the other (Holmes, 2013:1). In the recording environment, A/B testing is performed by switching between either recorded tracks or between live microphones, while monitoring the difference. It is vital that the switch be made quickly as aural memory is very short, lasting only a few seconds (Winer, 2012:90; Cowan, 1984:365). If even a small amount of time is required to switch to a new setup, detecting a difference between the components is very difficult, since, by that time, the reference contained in the aural memory would have already deteriorated (Winer, 2012:90). It would be advantageous to perform finer microphone adjustments directly from the listening position, where critical listening may identify very subtle changes in timbre. In order to do so a method is required for controlling the position of the microphone relative to the source.

Hence, this study proposed an alternative method for comparing and evaluating microphone placements, known as the *sweep technique*. The sweep technique involves remotely moving the microphone through a range of positions relative to the source, while monitoring the pick-up of the microphone through the studio monitors. A technique which may seem simlar, known as *Shavering*, applies to electric guitar recording. Once the musician has found a suitable guitar sound with the guitar amplifier that is to be recorded, the instrument is unplugged from the amplifier and the gain and volume controls of the amplifier are turned fully open so that a strong *hiss* is produced. The recordist then moves the microphone in front of the amplifier loudspeaker, while monitoring the response through a pair of headphones[3] (Gallagher, 2010). According to Gallagher (2010), there are different opinions among recordists as to where to place the microphone. Some search for the position where they hear the brightest hiss, while others aim to find the position which yields the darkest. Still other recordists look try to find a position yielding a hiss that sounds the same over the headphones as it does when listening to the amplifier in the room. Finally, some recordists look for a *balanced* hiss with a neutral tonality (Gallagher, 2010). Once the microphone has been placed, the gain and volume controls of the amplifier are returned to the settings previously chosen by the musician.

The difference between the sweep technique and shavering is that shavering is performed without input into the guitar amplifier, whereas the latter is done while the musician or a pre-recorded track[4] playing through the amplifier. A requirement for the sweep technique was a means of actuating the microphone remotely, which is the scope of the next chapter.

[3] The instrument is unplugged to prevent any signal from accidentally entering the amplifier. With the microphone placed close to the amplifier, any sound that is played through the amplifier and picked up by the microphone will come across extremely loudly in the recordist's headphones.

[4] This is achieved through re-amping.

CHAPTER 5

Conceptualization, Design and Manufacture of a Robotic Microphone Stand

THIS chapter details the design and manufacture of a prototype robotic microphone stand, which was used to implement the sweep technique.

5.1 Design Strategy

A design strategy, or methodology, describes the general plan of action for a design project and the sequence of particular activities which the designer expects to undertake to carry through the plan (Cross, 2008:193). This project adopted a prescriptive design strategy, which encourages improved ways of working and offers a more algorithmic, systematic procedure to follow, as opposed to descriptive design models which rely on the use of previous experience, general guidelines and *rules of thumb* to lead designers in what they hope to be the right direction (Cross, 2008:34). With prescriptive models, the intention is to ensure that the design problem is fully understood, that no important elements are overlooked and that the *real* problem is identified. The strategy used in this project, presented by Cross (2008:56), employs the most relevant and widely used methods and covers the whole design process. It consists of the following eight stages:

1. **Identify Opportunities:** Identify and define an opportunity for a new or improved product.
2. **Clarify Objectives:** Clarify design objectives and sub-objectives, and the relationships between them.
3. **Establish Functions:** Establish the functions required, and the system boundary of a new design.

4. **Setting Requirements:** Make an accurate specification of the performance required of a design solution.

5. **Determining Characteristics:** Set targets to be achieved for the engineering characteristics of a product, such that they satisfy customer requirements.

6. **Generating Alternatives:** Generate the complete range of alternative design solutions for a product, and hence widen the search for potential new solutions

7. **Evaluating Alternatives:** Compare the utility values of alternative design proposals, on the basis of performance against differentially weighted objectives.

8. **Improving Details:** Increase or maintain the value of a product to its purchaser whilst reducing its cost to its producer.

Figure 5.1 illustrates the eight stages of the design process positioned within a symmetrical solution model. This figure suggests how the eight stages relate to each other. As per example, the figure shows that Clarifying Objectives is appropriate for both understanding the problem-solution relationship as well as for decontructing the overall problem into subproblems (Cross, 2008:57).

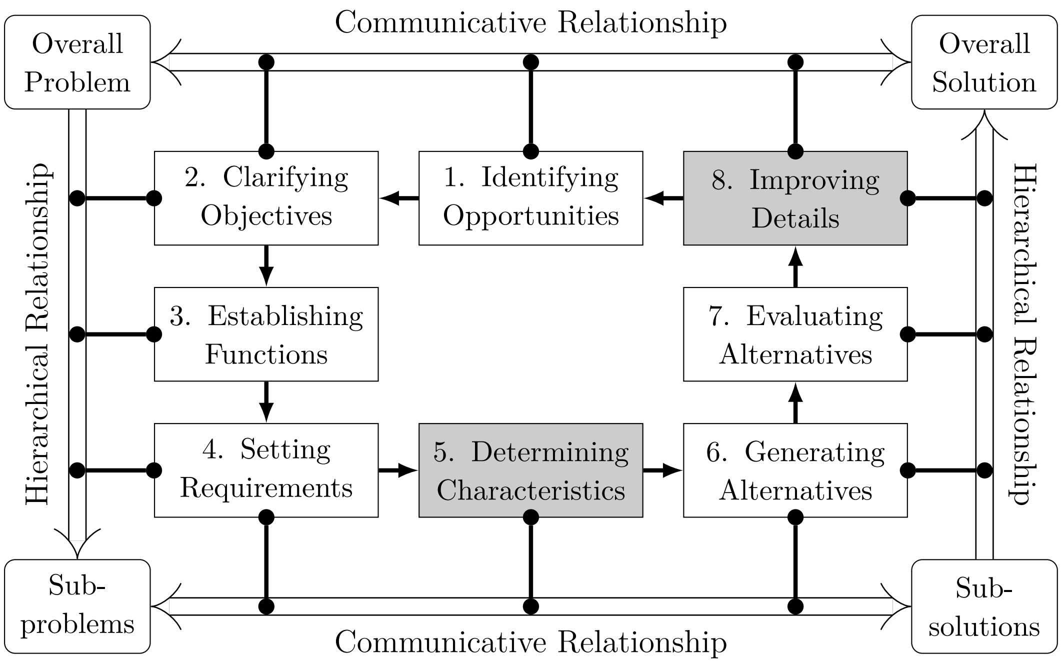


Figure 5.1: Cross design model (Cross, 2008:57).

This design model integrates the procedural aspects of design with the structural aspects of design problems. The anti-clockwise sequence of steps, from

Identifying Opportunities through to Improving Details, represent the procedural aspects. Structural aspects are represented by the arrows illustrating the communicative relationship between the problem and the solution and the hierarchical relationship between the problem and sub-problems and the solution and sub-solutions, respectively (Cross, 2008:58).

The goal of this project, however, was not to create a marketable product for the recording industry, but simply a prototype for implementing the sweep technique. Determining Characteristics utilizes the *Quality Function Deployment Method* and is used to translate customer needs, wants and values into technical requirements (Hurst, 1999:133). Improving Details aims to increase or maintain the value of a product to its purchaser (Cross, 2008:179). As the device was not intended for consumer use, these steps were omitted.

5.1.1 Identifying Opportunities

The proposed sweep technique involves moving a microphone relative to a source, whilst listening to its response over the studio monitors. As explained in the previous chapters, microphone placement is a crucial element of recording and can greatly influence the recorded sound. Studio monitors also have an ideal listening position, called the *sweet spot*, which is defined as the apex of an equilateral triangle formed by the loudspeakers and the listener's head. The sweep technique enables the recordist to adjust the placement of a microphone and immediately listen to its response from the optimum listening, thereby allowing for critical listening. The need for a way to implement the sweep technique created the opportunity to design a device with which a microphone's position may be remotely controlled.

For the sake of simplicity, the intended application of the prototype was restricted to the Fender Pro Junior III guitar amplifier used in the previous chapter.

5.1.2 Clarifying Objectives

The start of a problem, and therefore the design solution, is generally vague. The objective is not very specific and requires clarification. During this process, the initial objective may change considerably. This is advantageous as these changes reflect a better understanding of the problem and eventually lead to a more fitting design solution (Tooley, 2009:29). A popular technique for organizing design requirements is the *objectives tree*, which allows for the clear and concise representation of the project requirements (Haik & Shahin, 2010:120). The main objective is broken down into sub-objectives, which may consists of higher- or lower-level objectives and the means of achieving them (Cross, 2008:79). The objectives tree for this design is shown in Figure 5.2.

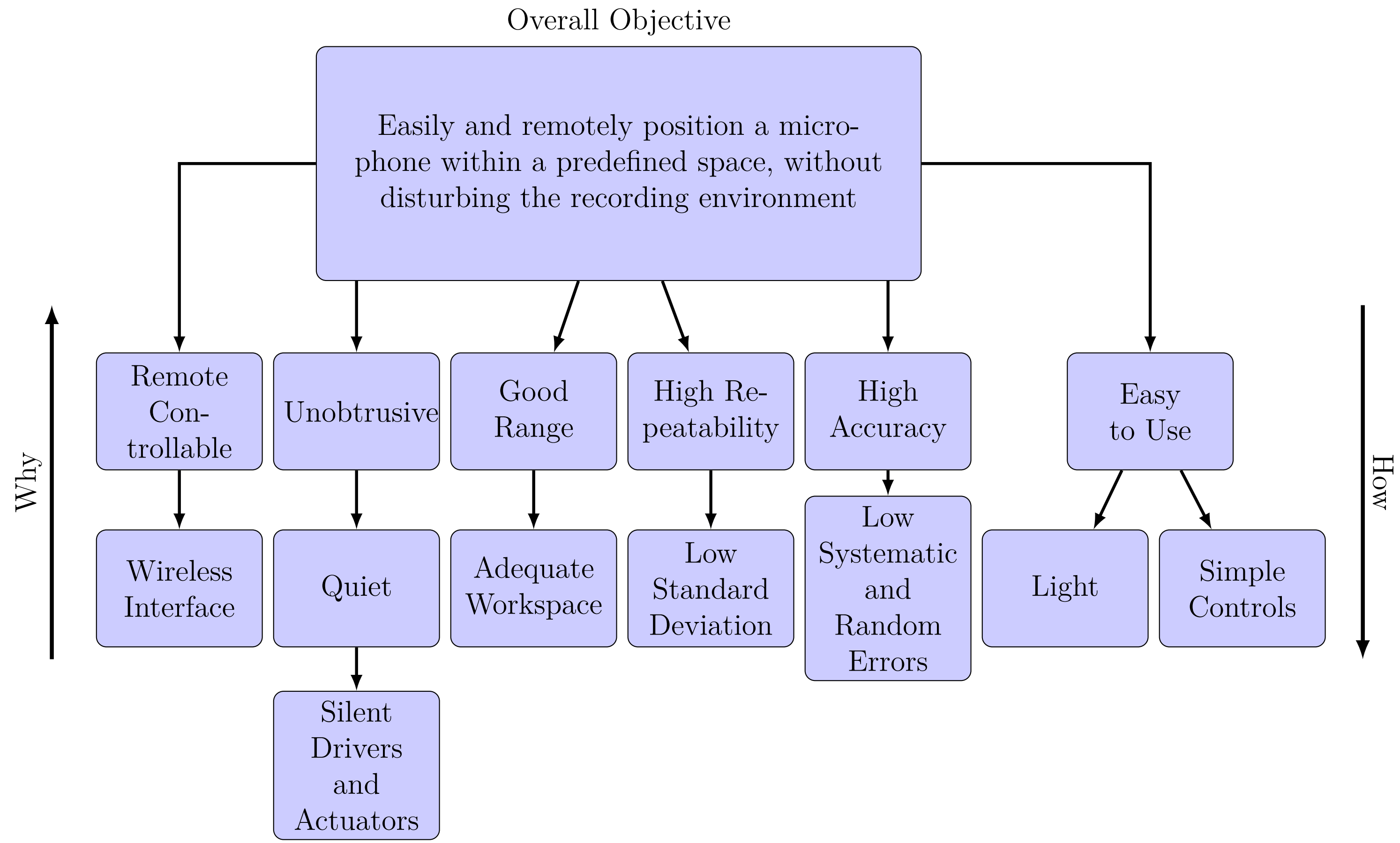


Figure 5.2: Objectives tree.

The overall objective for this device was to easily and remotely control the position of a microphone, without disturbing the recording environment. As shown in Figure 5.2, this overall objective was broken down to produce several lower-level objectives. Working down the objectives tree indicates *how* a higher-level objective is achieved, while working up the tree indicates *why* a lower level objective is included (Cross, 2008:81). The objectives tree method is not a design solution, but a a clear way of translating and clarifying what the project goals are, down to a level that is necessary for the next steps of the design process (Dym, 1994:133).

5.1.3 Establishing Functions

Functional analysis concentrates on *what* must be achieved by identifying and listing the inputs and outputs of the designed device in an organized way (Dym & Brown, 2012:118). According to (Cross, 2008:94), the simplest way of expressing this is to represent the device to be designed as a *black box*, which converts certain inputs into desired outputs. This expresses the relationship between the inputs and outputs, regardless of the solution within the box (Haik & Shahin, 2010:135). Figure 5.3 shows the black box system model for the device to be designed.

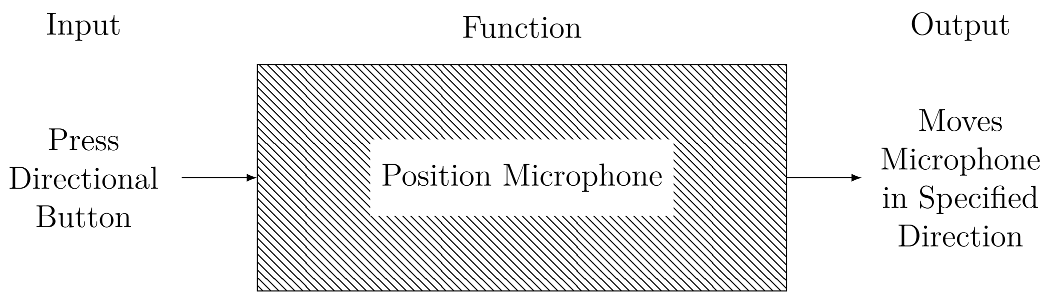


Figure 5.3: Black box system model.

In the next step of the functional analysis, the black box was replaced with a *transparent box*, wherein the overall function is decomposed into a block diagram of sub-functions that combine to achieve the overall function (Dym & Brown, 2012:119). This project's transparent box is shown in Figure 5.4.
Inputs and outputs cross the system boundary, which surrounds the sub-functions of the device. As Cross (2008:94) explains, the system boundary is a conceptual threshold that is used to define the function of the device. By resizing the boundary to include or exclude inputs and outputs, the function of the device is changed. With the sub-functions adequately defined at an appropriate level, it is possible to identify a suitable component for each sub-function (Cross, 2008:96). However, many possibilities exist and it is necessary to establish a limit based on the required performance.

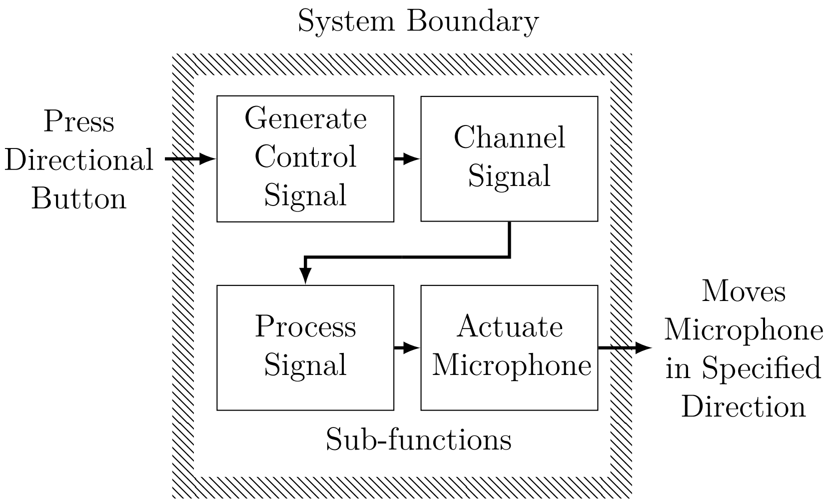


Figure 5.4: Transparent box system model.

5.1.4 Setting Requirements

The objectives and functions that state what a design must do are typically vague, with no indication to the limit of performance. Setting the requirements entails relating these objectives and functions to a list of performance specifications, that detail the exact performance that is to be achieved.

According to (Cross, 2008:106), three levels of generality should be considered during design, known as *alternatives*, *types* and *features*. These are described in Table 5.1. The level of generality determines the amount of freedom that the designer has to produce design solutions. The higher the level of generality, the greater the freedom the designer has to make major design decisions (Walker *et al.*, 1991:106). There were no design restrictions in this project, resulting in the highest level of generality and therefore any conceivable device for remote controlling the position of a microphone was acceptable.

Table 5.1: Levels of design generality.

Generality	Classification	Description
High	Alternatives	Any alternative design solution that performs the required function is acceptable.
Intermediate	Types	Designs are restricted to different types of a set overall solution.
Low	Features	Design is constrained to considering different features within a particular type of product.

In robotics, a workspace defines the collection of points that a robot can reach

(Niku, 2010:15). For the device in question, the workspace specifications were derived from the diameter of the loudspeaker of the guitar amplifier. The loudspeaker was a 254 mm 8 Ω Fender Special Design speaker. The workspace was chosen so that its height and depth were equal to the diameter of the loudspeaker, with the width equal to half of the diameter, as illustrated in Figure 5.5.

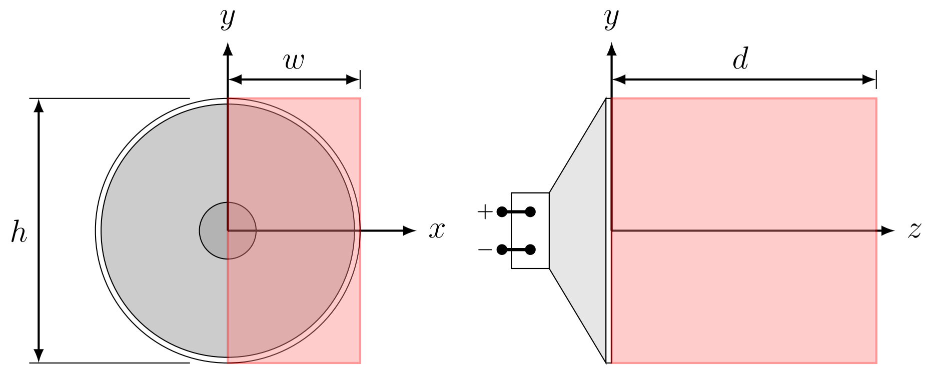


Figure 5.5: Device workspace, where w, h and d respectively indicate the width, height and depth of the volume.

The width of the workspace was chosen as such because a loudspeaker is symmetrical about the yz-plane that cuts through the center of the loudspeaker. However, the same simplification was not performed about the horizontal plane. For microphone positions that are further from the amplifier, comb-filtering may occur due to reflections from the floor. Comb-filtering is dependent on the distance between the microphone and the floor, therefore the microphone response above and below the xz-plane is not identical.

The payload refers to the weight that the device can carry, while remaining within its other specifications. Two microphones were selected as possible payloads, the Heil PR20 and the Karma Silver Bullet. Dynamic microphone are commonly used with guitar amplifiers, therefore the Heil PR20 was the primary choice. Because of its light weight and small form factor, the Karma Silver Bullet was selected as a secondary choice. The specifications of the Heil PR20 and Karma Silver Bullet are given in Appendix C.4 and Appendix C.7, respectively. The specifications of the design were determined from the previously determined objectives, functions, required workspace and payload. These are listed in Table 5.2.

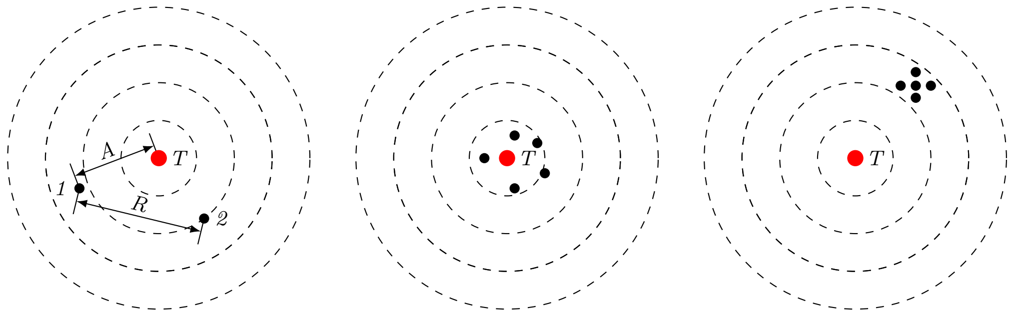
Accuracy and repeatability specifications require additional clarification, as these terms are often confused (Morris & Langari, 2012:18). The accuracy specification is a tolerance that defines the acceptable deviation of the actual location of a device's end-effector from its specified location. It is a function of the resolution of a device's actuators and feedback components, as well as the inaccuracies of its mechanical components (Niku, 2010:15). These

Table 5.2: Design specifications.

Attribute	**Description**	**Specification**
Payload	Weight the device can carry, while remaining within its other specifications	0.390 kg
Accuracy	How accurately a specified point can be reached.	10 mm
Repeatability	How accurately the same position can be reached if the motion is repeated several times.	20 mm
Speed	The speed with which the end point of the robot can travel.	$1\,\mathrm{cms^{-1}}$
Mass	The total mass of the device.	<5 kg
Workspace	The volume containing all the points the microphone can reach.	Height 254 mm Width 127 mm Depth 254 mm
Noise	The level of noise when the device is operated.	< 40 dB
Range	Maximum distance at which device can be operated remotely.	10 m

inaccuracies may include backlash in gears, loose linkages and effects such as deflection caused by the payload. The inaccuracies that generate the largest positional error, establishing the worst condition, are used to determine a realistic spatial resolution, from which the accuracy specification is derived (Hunt, 1983:36). Accuracies for industrial manipulators range from ±10 mm to ±0.01 mm, where the latter is for manipulators with accurate kinematic models and solutions and precisely manufactured and measured kinematic elements (Scheinman & McCarthy, 2008:83).

Repeatability is also affected by actuator resolution and component inaccuracies, but unlike accuracy which deals with arbitrary target positioning, repeatability is only concerned with the ability of a machine to return to a previously programmed position (Hunt, 1983:38). It is a statistical term associated with several achieved positions for the same target, and is expressed as the positional deviation from the average of these displacements (Kundra, 1993:316). Repeatability specifications range from 1 mm to 2 mm for equipment such as large spot-welding robots, to as little as 0.0005 mm (5 µm) for precise micro-positioning manipulators (Scheinman & McCarthy, 2008:83). The concepts of accuracy and repeatability are illustrated in Figure 5.6.



(a) A is a measure of accuracy and R is a measure of repeatability.

(b) High accuracy and high repeatability.

(c) Very high repeatability, but low accuracy.

Figure 5.6: An illustration of accuracy and repeatability. The marks indicate the actual positions achieved by a machine programmed to move to the target position, T, after several runs.

Performance specifications define fairly precisely how a satisfactory design must perform, but not yet what form it must take (Walker *et al.*, 1991:110). By choosing specifications in such a way, no possible design solutions are excluded.

5.1.5 Generating and Evaluating Alternatives

This section deals with selecting the appropriate methods for achieving the overall design solution. This is achieved by listing and evaluating alternative methods for achieving each individual sub-function, as determined in the functional analysis performed in Section 5.1.3. The process is known as *morphological analysis*[1], a systematic approach for analyzing the structure or form of an idea, object, device, system, product or process (Yan, 1998:69).

5.1.5.1 Generate Control Signal

Potential methods for generating a control signal were through the use of a game controller, touch screen device of a command line interface. Game controllers are input devices that direct the movements and actions of on-screen objects in video games (Shelly & Vermaat, 2008:244). Standard controllers have a directional pad, four primary buttons, two shoulder buttons and a pause button. Extended controllers include all the above-mentioned features, as well as two additional shoulder buttons and two directional thumb sticks (Richter, 2013:268). Many controllers are wireless and easily connect to a device through

[1] Morphology refers to the study of form (Sanie *et al.*, 1994:103).

Bluetooth[2]. For a game controller to be recognized by a computer, all that is required is a Bluetooth interface and some widely available open source software (Gams & Mudry, 2008:2).

Due to the vast array of inputs available on modern game controllers, as well as connectivity capabilities through standard Bluetooth technology, game controllers make excellent control devices. Additionally, controllers, such as the NINTENDO WIIMOTE and SONY SIXAXIS, are mass produced, making them inexpensive and widely available (Gams & Mudry, 2008:1).

Unlike game controllers that use physical buttons, triggers or joysticks as input devices, touch screens[3] produce input signals in response to a touch or movement of the finger on the display, the latter known as a *gesture*[4] (Greenstein, 1997:1318). Buttons, sliders and xy-pads can all be virtually represented on the screen of the device. A great advantage of touch interfaces is that the input device is also an output device that can provide visual feedback to the user. Therefore, there is direct hand-eye coordination (Greenstein, 1997:1321). Furthermore, the possible inputs are completely customizable, limited only to what can be displayed on the screen.

Touch screen technology can provide interfaces for multiple applications and is easier to use, more intuitive, and more customizable than interfaces based on keyboards and mice (Nichols, 2007:12). It is increasingly being used with computers and mobile devices, such as tablets and cellphones (Morley, 2008:60). Software such as HEXLER TOUCHOSC[5], allows the user to create custom templates that map buttons and encoders displayed on the screen to functions in the media server software (Claiborne, 2014:162). TOUCHOSC connects through either OSC (Open Sound Control) or MIDI (Musical Instrument Digital Interface) (McGuire, 2013:244, Barron & Orwig, 1997:97).

The final option for generating a control signal was a *command line interface*. In a command line interface, the user responds to visual prompts by typing in commands on a specified line. A well known command line interface is MS DOS (Microsoft Disk Operating System) (Balagurusamy, 2008:18). Command line interfaces may be difficult to use because they require exact spelling, grammer and punctuation. Even a minor error, such as not entering a period, will generate an error message (Shelly & Vermaat, 2008:402).

[2] Bluetooth is an open standard that enables communication among devices with a standard short-range wireless radio connection (Prabhu & Reddi, 2004:20)

[3] Touch tablets contain touch-sensitive surfaces not used for feedback display. Touch screens contain touch-sensitive surfaces overlaid by a display (Aghajan *et al.*, 2009:8).

[4] Harbour (2012:55) defines a gesture as any nonverbal communication using one's hands. In terms of gestures as touch screen inputs, a gesture might be to flick an object across the screen rather than dragging it to the exact location where the user wants it to be.

[5] See `http://hexler.net/software/touchosc` for more information.

5.1.5.2 Channel Signal

Game controllers and touch screen devices, such as tablets, and cellphones have Bluetooth connectivity, where the latter can also connect to other devices through Wi-Fi[6]. In addition, they can also be connected to a computer through USB (Universal Serial Bus) (Mueller, 2003:1036, Shelly & Vermaat, 2010:234), therefore there are various methods for channeling a signal from either a game controller or touch screen device to a computer or other device.

OSC is a protocol for communicating between computers, synthesizers and multimedia devices and is optimized for modern networking technology (Scheible & Tuulos, 2008:273). It was developed in 1997 at Berkeley's Center for New Music and Audio Technologies at the University of California (Doornbusch, 2009:68). OSC is a high-resolution content format with a dynamic, URL-style naming convention for versatile control. A great advantage of OSC is the ability to enact real-time sound and media control over local area and wide area networks (LAN/WAN) (Hopgood, 2013:16.11). OSC and MIDI are very similar, but the main difference is that the latter uses predetermined messages which dictate what the recipient should look like (d'Escrivan, 2012:170). This form of standardization allows for universal continuity (McGuire, 2013:244). However, the protocol was developed with the keyboard paradigm in mind and a limited resolution and ability to handle continuous parameter changes (Miranda & Wanderley, 2006:160). In comparison, OSC has no restrictions in terms of messages or data types that it can carry, making it very flexible (d'Escrivan, 2012:170).

5.1.5.3 Process Signal

Processing the signal refers to interpreting it and generating a corresponding output that drives the actuators of the device. Options for achieving this consisted of either a microcontroller or a computer.

Bangia (2010:354) defines a microcontroller as a special purpose, single chip computer designed and built to perform a particular, narrowly defined task. A microcontroller contains all the key components of a computer, such as microprocessor, memory, input and output interfacing circuits and peripheral devices such as A/D converters and timers (Godse & Godse, 2009:14.2), within a single integrated circuit (Barrett & Pack, 2011:3). Microcontrollers execute programs loaded into their memories. Under control of these programs, data is received as inputs, manipulated and sent to external devices as outputs (Ibrahim, 2014:2). Microcontrollers are powerful tools that enable designers to create sophisticated input and output manipulation (Ibrahim, 2001:1). In addition, they are cheap and consume very little power (Kurt, 2006:297).

[6] Wi-Fi, short for *wireless fidelity*, is a protocol used for wireless communication (Davis, 2004:8).

The Arduino Uno microcontroller is shown in Figure 5.7a. The Arduino is an open-source microcontroller that is very popular among prototypers, do-it-yourself enthusiasts, interaction designers and educators (Faludi, 2010:57). Since its debut in 2005, over 500 000 units have been sold world wide and it is estimated that the number of Arduino clone boards that have been sold is even greater (McRoberts, 2013:1). The Arduino Uno is one of various models in the Arduino range[7].

(a) Arduino Uno.   (b) Raspberry Pi Model B+.

Figure 5.7: A microcontroller and a single-board computer.

Unlike a microcontroller, a computer is a multipurpose device (Parsons & Oja, 2010:3). Computers must be flexible enough to handle huge quantities of data quickly and require far more computing power than a microcontroller (Frenzel, 1999:210). As a result, they are far more expensive.

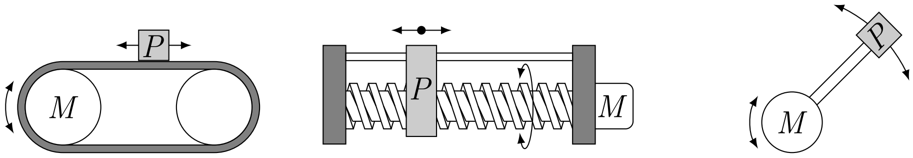
Alongside the traditional concept of a computer, a new trend has appeared in the form of the credit-card sized computer. Surely the most popular, is the Raspberry Pi, shown in Figure 5.7b. The Raspberry Pi is an open-source, single-board computer that was created for educational purposes. However, due to its size, low cost and low power requirements, it has been used as a key component in embedded systems and implementations by hobbyists worldwide, much like the Arduino (Peña-rois *et al.*, 2013:196). The Raspberry Pi differs from the Arduino in that it is still a computer, capable of connecting to a keyboard, mouse and monitor and running an operating system. Launched in 2012, the Raspberry Pi is capable of running Linux and its applications, and can even handle playback of high definition video (Goodwin, 2013:275).

5.1.5.4 Actuate Microphone

To actuate the microphone, conveyors, robot arms and lead screws were considered. Conveyors consist of equipment capable of moving material in a continuous or intermittent fashion along a fixed path, between two or more points.

[7] See `http://arduino.cc/` for more information.

The movement may be horizontal, vertical, inclined or any combination of the three (Fayed & Skocir, 1996:1). Fixing a microphone to a conveyor allows its position along the travel path to be controlled through the rotation of the motors that drive it. The conveyor principle is illustrated in Figure 5.8a.



(a) Conveyor belt. (b) Lead screw. (c) Rotational link.

Figure 5.8: Three methods of actuation. M indicates a motor and P indicates the payload.

Lead screws, also known as power or jack screws, are machine elements that convert rotary motion into translational[8] motion along the axis of rotation (Collins *et al.*, 2010:462). It consists of a screw (or threaded rod), a nut and a part to hold either the screw or the nut in place (Bhandari, 2010:184). The conversion occurs at the meshing of the lead screw and nut thread, where the contacting heads slide against each other, creating a friction force opposing the direction of motion (Vahid-Araghi & Golnaraghi, 2010:85). Depending on the configuration, either the nut is held stationary and the screw moves axially as it rotates, or the nut is kept from rotating, but is free to translate as the screw turns in a set of bearings. The latter is illustrated in Figure 5.8b. One of the main applications of lead screws is to provide accurate motion in machining operations, such as a lathe (Bhandari, 2013:283). At least one lead screw or conveyor is required per axis to achieve three dimensional motion.

The final method for actuating the microphone that was considered, was a rotating link, as illustrated in Figure 5.8c. This is a very simply mechanism where the angular position of the payload at the end of the linkage is controlled through the rotation of a motor. In contrast to conveyors and lead screws, such a mechanism generates arc motion. To achieve motion that occurs in a three dimensional volume, three of these actuators are required.

A servo motor is a device that consists of a direct current (DC) motor, a gear train, an internal potentiometer and some electronics. The gear train drives the output shaft, but also drives the potentiometer, which feeds back the shaft position to the internal electronics that control the DC motor (Hellebuyck, 2003:173). Servos are typically limited to 130° to 180° of rotation (Branwyn, 2003:90). Various types of DC servo motors exist, but the operation of all DC servo motors are similar (Firoozian, 2008:59). The servo motor is controlled by a pulse-width-modulated (PWM) signal (Hellebuyck, 2003:173)

[8] Translational motion is characterized by movement along a linear path between two points (Myers, 2006:31).

and is used where precise angular displacement is required (Barrett & Pack, 2012:136).

The rotor of a stepper motor rotates in discrete angular steps in response to a programmed sequence of excitation (Desai, 2008:133). The principle operation is the magnetic attraction and repulsion of like and unlike poles respectively. Each revolution of the stepper motor's shaft consists of a series of discrete individual steps, where one step is the angular rotation of the output shaft in response to a step pulse. Since the motor shaft only moves a certain number of degrees when the pulse is delivered, the positioning and speed of the motor can be controlled by controlling the number of pulses sent to the motor (Jain, 2004:372). Therefore, a stepper motor can provide precise positioning and speed control without the use of a feedback sensor (Kant, 2007:544). This form of operation is known as *open-loop* control and is attractive because it is simple and inexpensive as additional components, such as position and speed sensors, are not required (Bartelt, 2010:592). Although economically advantageous, open loop control for stepper motors has its limitations. The motor's response to a given input may become oscillatory or even unstable in some speed ranges. Therefore, the control of the motor is not very fast. Additionally, the motor may fail to follow the pulse if the frequency is too high or the inertia of the load is very large (Subrahmanyam, 2011:700).

A morphological chart is shown in Figure 5.9. It consists of a leading column listing the desirable functions, and various concepts for each function listed in corresponding rows (Lieu & Sorby, 2008:5.7). The morphological chart acts as a summary of the morphological analysis (Cross, 2008:138). Selecting any one solution for each sub-function generates an overall solution that fulfills the design requirements.

A game controller was readily available and was used to generate the control signal. Using the controller, the signel could be channel either through Bluetooth or USB. Bluetooth was selected to eliminate the need of USB cabling between the device and its controller. A Arduino microcontroller was used to process the input from the game controller and control the motors. Even though a Raspberry Pi microcomputer could also be used, it would have required additional components to control the motors. The Arduino was less expensive and comes with all the required components built in. Finally, it was decided to perform the required actuation through rotational links in the form of an anthropomorphic arm. Though more complex than either conveyor belts or lead screws, the anthropomorphic arm was a more compact option.

5.2 Anthropomorphic Arm

An anthropomorphic arm consists of three rigid links connected by rotational joints. It is also known as an *RRR* model, because all three joints are of the

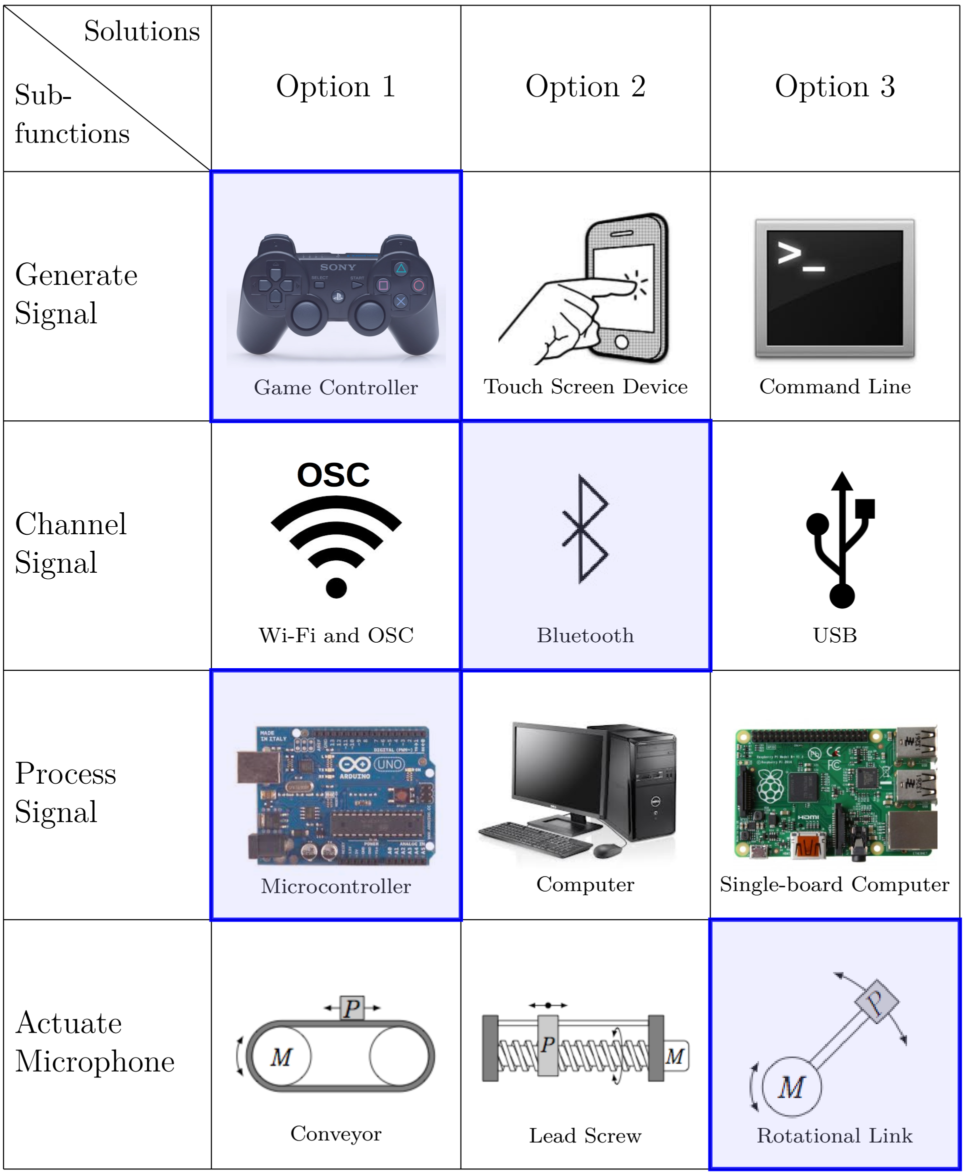


Figure 5.9: Morphological chart.

rotational type. To a large extent, this type of robot resembles a human arm (Bajd *et al.*, 2013:6).

An anthropomorphic arm is illustrated in Figure 5.10. The first joint axis provides rotation in the horizontal plane. The second and third joint axes are parallel to each other, but perpendicular to the first and provide rotation in the vertical plane. The combination of these joints facilitates motion of the payload, or end-effector, in three-dimensional space. The workspace, encompassing all the points that can be reached by the end point of the arm, has a spherical shape (Bajd, 2010:5). This workspace is dependent on the configuration and size of the links and joints used in the arm's design (Niku, 2010:15).

Positional control is performed through the control of motors at each of the joints. Linear motion is achieved by actuating two or all three of the joints in conjunction.

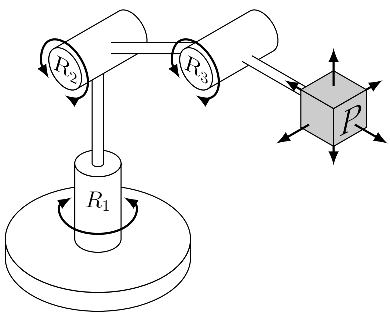


Figure 5.10: An illustration of an anthropomorphic arm.

5.2.1 Spring-balance Mechanism

A spring balance mechanism was implemented in the design. It was inspired by the two-degree-of-freedom design of the anglepoise desk lamp, invented by the engineer George Carwardine in 1932 (Schenk, 2010:1). A two-degree-of-freedom model is shown in Figure 5.11.

The springs exactly counteract the forces caused by gravity, for any configuration of the lamp's linkages. Therefore, it is in static equilibrium and remains in whatever configuration it is moved (Deepak & Ananthasuresh, 2012:1). Incorporating a balancing arrangement permits the use of weaker, and therefore cheaper, motors since they do not have to support the dead weight of the device. The actuators only need to be powerful enough to overcome the inertia and momentum of the payload, i.e. to accelerate and decelerate it (Schenk, 2010:5). Spring balancing can be seen in other applications, such as the ingenious STEADICAM[9], a camera stabilizing system that allows the operator to run, jump and climb stairs whilst providing smooth controlled footage, and the BALANCEBOX[10], a system that aids the vertical adjustment of large and heavy wall mounted equipment such as interactive whiteboards and touch-screen panels (Millerson & Owens, 2012:135).

The force-extension relationship for a normal helical coil spring is given in Equation (5.2.1), where k is the *spring rate*, F is force in the spring and F_i is initial force when there is no extension (Budynas & Nisbett, 2008:527). The spring rate is defined as the force required to produce one unit of displacement (Humar, 2012:26). The extension is given by $x = (l - l_0)$, where l is the total length of the spring and l_0 is *free length* or unstretched length of the spring.

[9] See `http://www.steadicam.com/` for more information.
[10] See `http://www.balancebox.eu` for more information.

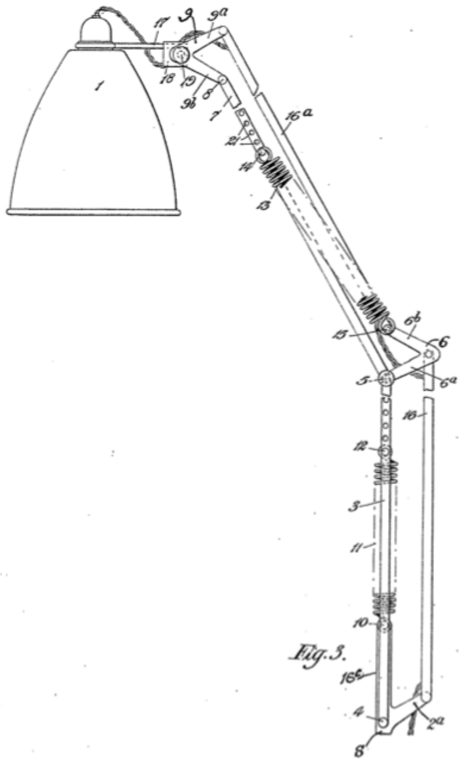


Figure 5.11: Original anglepoise design by George Carwardine (Carwardine, 1934:12).

$$F = F_i + kx \tag{5.2.1}$$

An important prerequisite for this spring balance mechanism was the implementation of *zero free length* springs. For a spring suspended between two bodies with relative motion, the spring force is a function of its two anchor points. In a zero free length spring, this function is linear, but for a normal spring it is nonlinear, despite both springs having a linear force-deflection relationship (Deepak & Ananthasuresh, 2012:1). The result is that the force in the spring is dependent on the length of the entire spring, not its extension, as shown in Appendix F.1.

Helical coil springs are typically manufactured with an initial force so that the coils remain in contact when no external force is present. These springs cannot actually have a zero free length, because the coils cannot lie in the same plane (LaCoste, 1988:20). If the spring is manufactured with an initial force of $F_i = kl_0$, the force relation is as expressed in Equation (5.2.2) and the force in the spring is proportional to the length of the entire spring, not just its extension.

$$\begin{aligned} F &= k(l - l_0) + kl_0 \\ &= kl \end{aligned} \tag{5.2.2}$$

Figure 5.12 illustrates the force-extension relationship of a real spring that was manufactured with an initial force of kl_0. The *spring rate*, k, represents the gradient of this curve (Humar, 2012:26). At length L_1, the coils of the spring are in contact and the spring cannot contract further. When the extrapolated force-extension data of the spring passes through the origin, it is a zero free length spring (LaCoste, 1988:21). If it is manufactured with an initial force of kl_0, a real spring can behave like a zero free length spring (Banala, 2008:39-40). The spring properties were calculated and optimized using SOLVER within MICROSOFT EXCEL, as described in Appendix F.2.

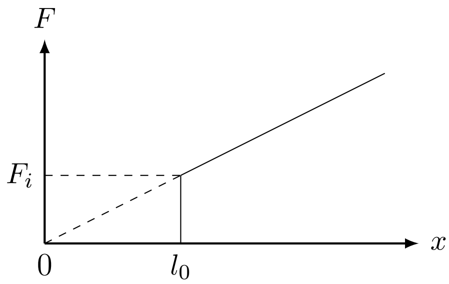


Figure 5.12: Force-length graphs of helical coil springs.

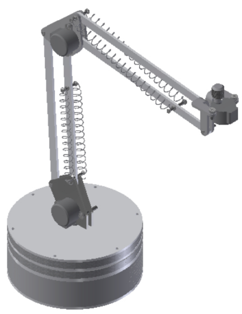
Like the original anglepoise design of Figure 5.11, each set of links were designed to form a parallelogram. Since both pairs of opposite sides of the parallelogram remain congruent, the part to which the links attach experiences no rotation when moved in the vertical plane (Wheater, 2007:172).

5.2.2 Prototypes

Manufacturing costs were kept to a minimum by designing parts that could be manufactured cheaply. Components consisted of laser cut aluminum, with the exception of a shaft and threaded attachment that were manufactured by lathe. Laser cutting uses a high-intensity infrared laser beam focused on the surface of a work piece. The laser heats up the material and creates a localized melt throughout the depth of the piece. The molten material is then removed by a gas jet. Laser cutting benefits from high cutting speeds, a narrow cut width and reduced material waste (Maini, 2013:418). Springs for the spring balance mechanism were designed and manufactured, as detailed in Appendix F.2.

Figure 5.13a shows the computer-aided design (CAD) model of the first prototype, with the manufactured prototype shown in Figure 5.13b. Zero free length springs are difficult to produce and expensive (Sacks *et al.*, 2003:163).

Unfortunately, due to incorrect spring manufacturing, this prototype was unsuccessful. The springs only exhibited the correct force-extension relationship over a very short range, resulting in restricted motion of the anthropomorphic arm. Time and monetary constraints prevented further development of this prototype.



(a) *Prototype 1* CAD model.

(b) *Prototype 1.*

Figure 5.13: The first prototype.

As a compromise, the Heil PR20 was replaced with the secondary choice microphone, the Karma Silver Bullet. The spring balance mechanism was abandoned in favor of two powerful servo motors, capable of supporting the payload. The stepper motors responsible for rotation at the base and at the end-effector were sufficient and remained unchanged. This led to the second anthropomorphic arm prototype, as shown in Figure 5.14. Although the Karma Silver Bullet is an omni-directional microphone, its angle relative to the source still has a significant effect on the recorded timbre, due to off-axis colouration. Some parts had to be re-designed to fit the servo motors, but the the overall design remained the same. Specifications for the Karma Silver Bullet are listed in Appendix C.7.

5.2.3 End-effector

The *end-effector* is the device that is connected to the end of the anthropomorphic arm. End-effectors are devices through which a robot interacts with the environment around it. There are several variants, each with a specific operation such as gripping, cutting, manipulating or picking up parts. The purpose of the end-effector for this device was to provide attachment for the test microphone and to counteract yaw[11] when the arm is rotated through the horizontal plane by the motor in its base. The end-effector consisted of two housing plates and a motor. In addition, the end-effector facilitated the

[11] Rotation about the vertical axis.

(a) *Prototype* 2 CAD model.

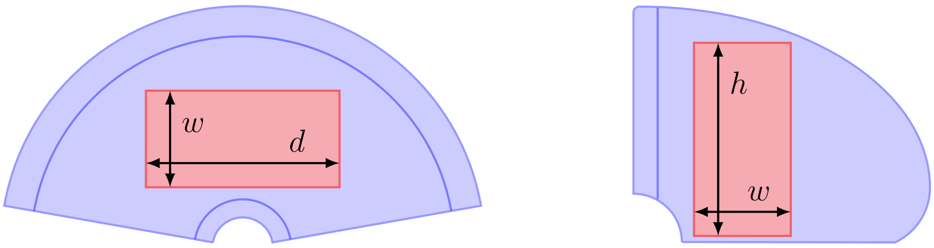
(b) *Prototype* 2.

Figure 5.14: The second prototype.

independent rotation of the microphone for setting its angle relative to the source.

5.2.4 Workspace

The workspace of an anthropomorphic arm is spherical (Duchemin *et al.*, 2001:309). The dimensions of the device were chosen so that its workspace encompassed the required workspace, as illustrated in Figure 5.15, where w, h and d are respectively the width, height and depth defined in Section 5.1.4.



(a) Viewed from above.

(b) Viewed from the side.

Figure 5.15: Device workspace. Blue illustrates actual workspace, while red illustrates required workspace.

The workspace may be found mathematically by determining equations that define the robot's links and joints, including their limitations, such as the range of motion for each joint. Alternatively, the workspace may be determined empirically by virtually moving each joint through its full range of motion, combining all the space it can reach and subtracting the space it cannot (Niku, 2010:15).

5.2.5 Kinematics

Kinematics is a branch of science that analyzes motion, although with no attention to what causes the motion (Jazar, 2010:14). In robotics, these take the form of *forward* and *inverse* kinematics. When data concerning the joints of a manipulator is known, forward kinematics is used to calculate the position of the manipulator at any instant (Niku, 2010:59). In inverse kinematics, the location of the end-effector is known and the problem is to find the joint variables necessary to bring the end-effector to the desired location (Tsai, 1999:54). Inverse kinematics is highly nonlinear and usually a much more difficult problem than forward kinematics (Jazar, 2010:14).

Consider the model of a two-degree-of-freedom manipulator illustrated in Figure 5.16. Line B is the imaginary line between the origin and point (x, y). If the angles of the joints, θ_1 and θ_2, as well as the lengths of the links, L_1 and L_2, are known, the end position (x, y) may be calculated with Equations (5.2.3) and (5.2.4), respectively.

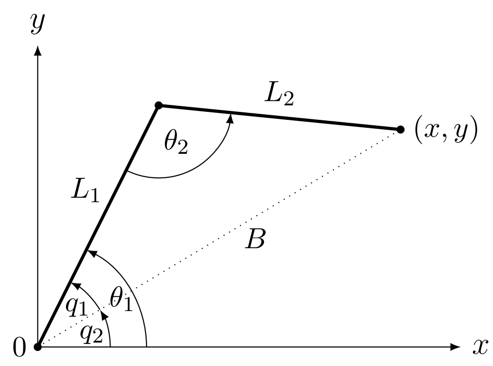


Figure 5.16: Illustration of a two-degree-of-freedom robot.

$$x = L_1 \cos(\theta_1) + L_2 \cos(\theta_1 + \theta_2 - 180°) \tag{5.2.3}$$

$$y = L_1 \sin(\theta_1) + L_2 \sin(\theta_1 + \theta_2 - 180°) \tag{5.2.4}$$

For the same model, the inverse kinematics may be calculated with Equations (5.2.5) to (5.2.9). It is important that the *arctan*[12] in Equation (5.2.6) be calculated for the correct mathematical quadrant. Many programming languages and scientific calculators offer the *atan2* function (Davidson & Hunt, 2004:18), a computer function that requires two signed inputs and returns the *arctan* of the appropriate mathematical quadrant (Agarwal & Pershad, 1998:148).

[12]The *arctan* function is the inverse of the *tan* function (Schinazi, 2011:81).

$$B = \sqrt{x^2 + y^2} \tag{5.2.5}$$

$$q_1 = \arccos\left(\frac{L_1^2 + B^2 - L_2^2}{2L_1B}\right) \tag{5.2.6}$$

$$q_2 = \arccos\left(\frac{y}{x}\right) \tag{5.2.7}$$

$$\theta_1 = q_1 + q_2 \tag{5.2.8}$$

$$\theta_2 = \arccos\left(\frac{L_1^2 + L_2^2 - B^2}{2L_1L_2}\right) \tag{5.2.9}$$

The existence of solutions is guaranteed only for configurations that belong to the workspace of the manipulator (Siciliano *et al.*, 2009:91). One of the difficulties of inverse kinematics is the possibility of multiple solutions, as illustrated in Figure 5.17 (Siciliano *et al.*, 2009:91).

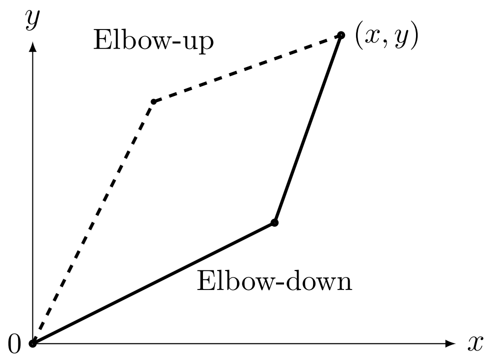


Figure 5.17: Multiple solutions for inverse kinematics (Ceccarelli, 2004:122).

In this figure two manipulator configurations exist that result in the desired target position (x, y). The two configurations are termed *elbow-up* and *elbow-down*. The availability of multiple solutions permits the elimination of solutions that may possess undesirable features. This is achieved by imposing workspace constraints involving either the configuration of the manipulator, or both the configuration and velocity of the manipulator (Vepa, 2009:94). Forward and inverse kinematics were incorporated in the Arduino software.

5.2.6 Electronic Components

A schematic of the electronic components used in *prototype 2* is shown in Figure 5.18. The servo and stepper motors were controlled from the digital output pins of the Arduino Uno microcontroller and powered from by Huntkey

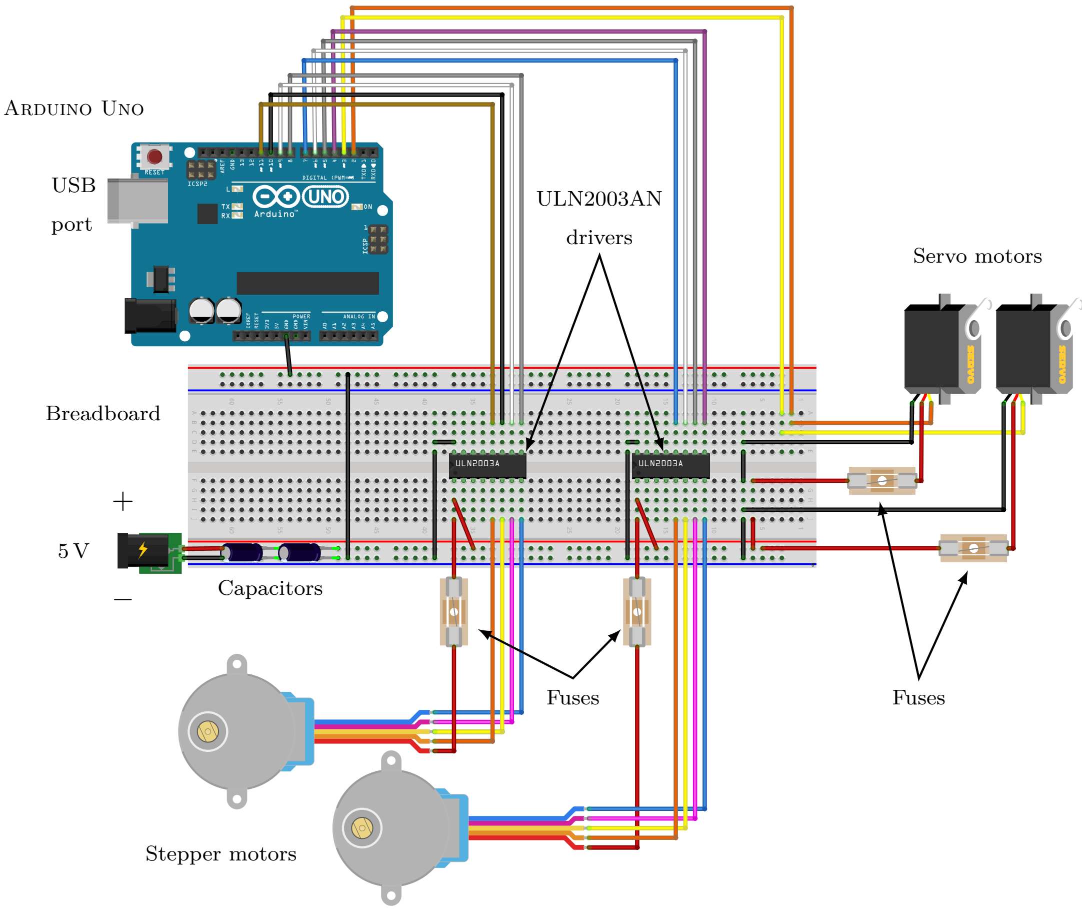


Figure 5.18: *Prototype 2* Fritzing schematic.

CP-350 computer power supply. The Arduino Uno was powered from the USB serial bus of a computer.

To ensure a steady supply of peak current, two 2200 µF capacitors were placed in parallel with the power supply to the motors. In addition, 600 mA and 1.5 mA fuses were placed in the positive power lines of the stepper and servo motors, respectively. These were to prevent damage to the motors in case too much current was drawn under load.

Each stepper motor were driven using a ULN2003AN based driver board. The ULN2003AN is an integrated circuit that contains an array of Darlington[13] transistor pairs (EFY Enterprises, 2009:215). A parts list of the components may be found in Appendix C.8.

A Sony Dualshock 3 game controller was connected to a computer running Processing. A simple program was written in Processing to relay the input messages from the game controller, through the USB serial connection, to the Arduino Uno. These messages were interpreted by software writ-

[13] A Darlington configuration is a transistor circuit used to greatly amplify an input current (EFY Enterprises, 2009:215).

ten for the ARDUINO UNO, which configured the motors to move accordingly. The PROCESSING and ARDUINO source code is available in Appendices E.6 and E.8.

5.3 Implementation

With a payload capacity of 0.250 kg, *Protoype 2* easily supported the KARMA SILVER BULLET microphone. Speed was limited to 1 cms^{-1}, although greater speeds were achieved. The device proved to be very quiet, emitting a maximum noise level of 23 dB SPL. The main source of noise was the internal reduction gearbox of the servo motors. The gearbox contained spur gears, which are characteristically noisy at high speeds (Darbyshire, 2010:197). Since the guitar amplifier was calibrated to an output 110 dB SPL, a very good signal to noise ratio of 87 dB SPL was achieved. The device weighed 3 kg and had a wireless range of 12 m.

The servo motors exhibited large nonlinear positioning errors which resulted in an offset of 6° to 20° from their target positions. The nonlinearity of this error proved difficult to account for and resulted in a diminished accuracy of 50 mm. As this error was systematic, it did not affect the repeatability of *Prototype 2* and the specification of 20 mm was still achieved (Zeller *&* Carmines, 1980:77).

A test was conducted to illustrate the pick-up of a microphone while a sweep is performed. The setup for this test was similar to that of Figure 4.2 on page 68. *Prototype 2* was set up in front of the FENDER PRO JUNIOR III guitar amplifier, which was fed with a pink noise test signal. A photograph of the setup is shown in Figure 5.19.
Unlike a swept sine signal, the average properties of pink noise do not change over time[14] (Woodgate, 2001:701). *Prototype 2* was equipped with the KARMA SILVER BULLET condenser microphone. From the control desk in the adjacent room, the microphone placement was adjusted through a range of positions relative to the guitar amplifier. The resulting spectrogram shown in Figure 5.20 shows the continuous change in coarse spectral distribution of the sound picked up by the microphone.

Implementation of *Prototype 2* was successful. The controls proved to be intuitive and made it easy sweep through microphone placements. However, visual feedback was required to ensure that the device did not collide with the guitar amplifier. Additionally, once a microphone had been placed, it was impossible to move the microphone to a microphone stand without disrupting its exact placement. The consequence is that the device can not be used to perform microphone placements successively. Recording of each individual source has to be completed before *Prototype 2* can be used for microphone placement with the next source.

[14] Such signals are described as being *stationary* (Woodgate, 2001:701).

Figure 5.19: Implementation of *Prototype 2*.

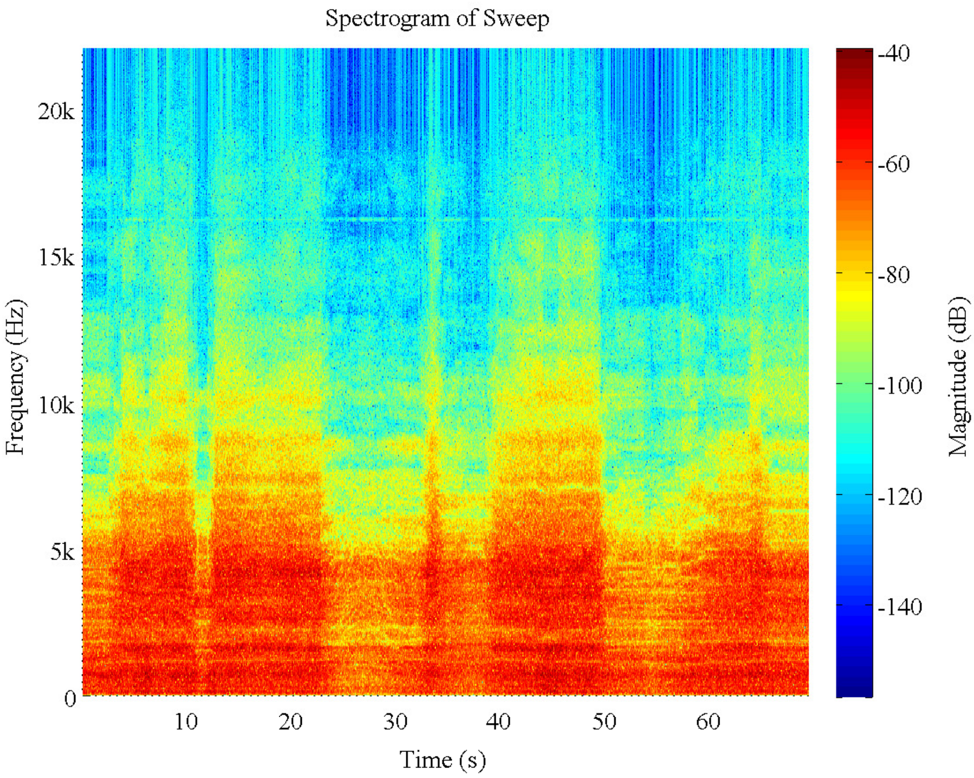


Figure 5.20: Spectrogram of microphone sweep.

Chapter 6

Discussion of the Sweep Technique

Microphone placement is a trial and error process, mitigated by knowledge and experience. Recordists rely on critical listening and aural memory to compare the changes in timbre caused by different microphone placements, however these methods are not necessarily the most scientific approach due to the difficulty of identifying optimum configurations.

6.1 The Sweep Technique

The sweep technique entails moving a microphone through a range of microphone placements, while monitoring the change in timbre from within the sweet spot of the studio monitors. It is performed on a live source and is achieved through an auxiliary device. *Prototype 2* represents one of many devices that may be used to accomplish microphone positioning.

Conducting microphone placements in this way enables the recordist to evaluate the recorded timbre of the sound source through critical listening from the optimum listening position in real time and adjust the the microphone position accordingly. Therefore, the sweep technique may be expressed as *closed loop* control system, as illustrated in Figure 6.1.

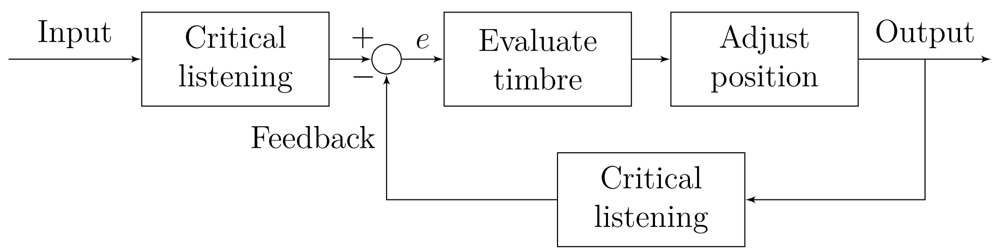


Figure 6.1: Sweep technique feedback loop.

In electronics, a closed loop system is one that includes *feedback*, where the output of the system is measured by a sensor, fed through a controller and back into the input of the system (Onwubolu, 2005:6). The feedback is compared with a reference value to form an error, e, representing the difference between the actual output and the desired output (Perdikaris, 1991:4). In response to this error, the controller performs an action, which is proportional to the error, and serves to reduce the error to zero, thereby achieving the desired output.

Implementation of the sweep technique creates a biomechatronic system consisting of the recordist, recording equipment and a robotic microphone stand, such as *Prototype 2*. The input of the system is the original timbre picked up by the microphone and the output is the timbre resulting from adjusting the microphone placement with *Prototype 2*. The recordist acts as both controller and sensor and serves to evaluate the difference between the actual timbre and the desired timbre. The desired timbre may be in the mind of the recordist, or a recording which the recordist compares with the output of the system through A/B testing[1].

By performing a sweep, trends in timbre can be identified. As per example, the experiment results discussed in Section 4.5.2 showed abundant high frequency content in the center of the loudspeaker, which decreased as the microphone was moved to the edge of the loudspeaker. This corresponds to a *bright* timbre in the center, which becomes *warmer* and more *mellow* towards the edge of the loudspeaker (Huber & Runstein, 2010:150; Anderton, 2011:53). Utilizing the sweep technique, the recordist may quickly identify such trends, which may be used as guides when performing microphone adjustments.

Furthermore, the sweep technique may be performed in conjunction with any stereo or multi-microphone technique, provided the entire microphone configuration is moved as a whole. This may be achieved through the use of a stereo bar or similar component capable of supporting more than one microphone.

6.2 Applications of the Sweep Technique

The sweep technique functions much like an *equalization sweep*, a common technique used by recordists to identify frequencies at which to apply equalization (Case, 2007:108). An equalization sweep involves increasing the gain of a movable band filter and sweeping through a range of different frequencies, as shown in Figure 6.2 (Middleton & Gurevitz, 2008:348). The filter is configured with a high Q-value, which corresponds to a very narrow bandwidth (McGuire & Pritts, 2013:128). As the filter is moved through the adjustable frequency range, a small band of frequencies are amplified so that they become easily detectable. The recordist may identify the frequency band from which unde-

[1] Instead of switching between two recorded tracks, the recordists uses A/B testing to switch between a recorded track and the live output of the microphone with which the sweep technique is being implemented. As with conventional A/B testing, the switch must be made quickly to make use of the recordist's limited aural memory.

sired sounds originate and apply equalization as required. In the same way that the recordist may use an equalization sweep to identify undesired sounds, the sweep technique may be used to identify suitable microphone placements by sweeping the microphone through a range of positions relative to the source.

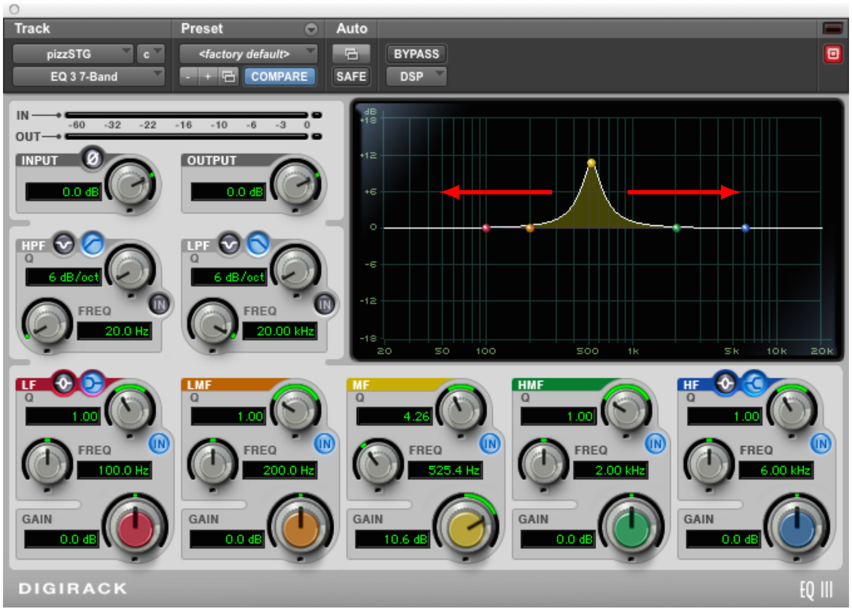


Figure 6.2: Equalization sweep.

The *mixdown* is the process in which all the separately recorded audio tracks are processed, positioned within the stereo image and combined into monophonic, stereophonic or surround sound, depending on the requirements of the final product (Alten, 2013:517; Huber & Runstein, 2010:429). It is the combination of mixing and *bouncing*, which consists of recording several prerecorded tracks into one track or a pair of tracks (Thompson, 2005:80). During mixing, the audio is repeatedly played while adjustments in level, panning, equalization and effects are made for each track or group of tracks (Huber & Runstein, 2010:429). It is a real time process in which the recordist adjusts the overall dynamic shape of the music (Shepherd, 2003:224).

Mixing is a correlative process. The adjustment of a parameter on one track may have a significant effect on another track, group of tracks and/or the overall mix. As such, it benefits from an iterative coarse-to-fine approach, as illustrated by Figure 6.3 (Izhaki, 2013:40-41). Initially, adjustments to mixing parameters are rough, but become finer as the mix progresses. Previous mixing decisions are continually refined until the final mix is achieved. The most attention is paid to the late mixing stages, where the subtlest changes are made (Izhaki, 2009:119). The mixdown process is complete when a single version of the mix has been approved and bounced. This recording is then ready to be mastered[2] to its intended medium (Huber & Runstein, 2010:429).

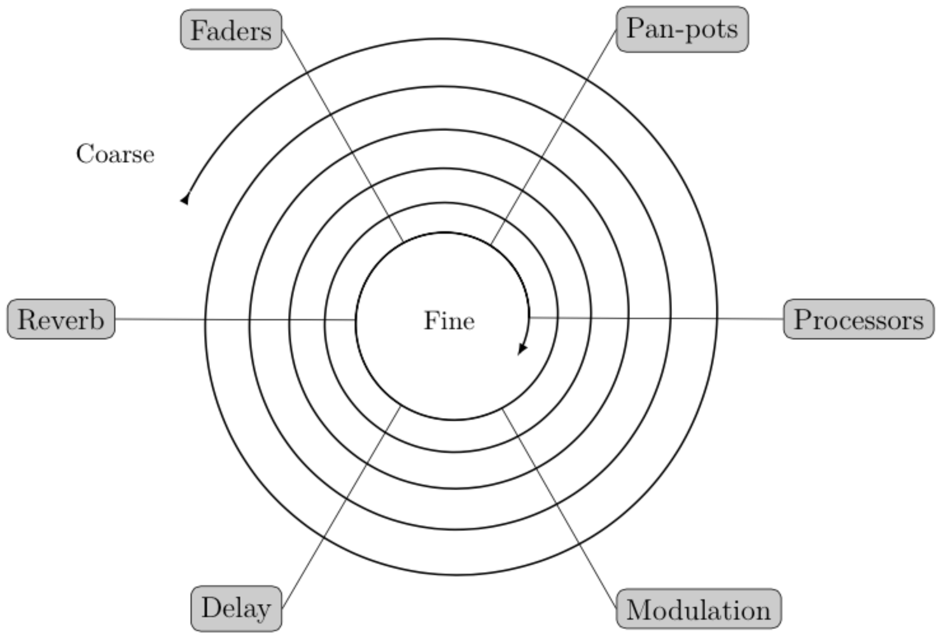


Figure 6.3: Iterative coarse-to-fine mixing approach (Izhaki, 2013:41).

Often in pop, rock and jazz recordings, each instrument is recorded onto one track (Brice, 2001:340). During each of these recordings, decisions are made about the expected level of each track, its location within the stereo image, added effects and its timbre. Therefore, the mixing process begins even before all the tracks have been recorded. During mixing, these decisions are finalized as the audio tracks and signal processing are combined into the final recording (Case, 2007:64). The sweep technique can aid the making of these decisions, by allowing the audition and adjustment of microphone placements for new recordings while the recordist evaluates the resulting timbre within the mix of instruments and tracks that have already been recorded. In this way, the recordist may use the sweep technique to search for placements that result in timbres that are suitable to the mix of instruments that have been recorded so far.

When multiple sounds play simultaneously in a mix, masking occurs and modifies the overall sound. Louder sounds mask quieter sounds (Izhaki, 2013:13).

[2] Mastering is the last stage in the production process. The main purpose of mastering is to treat the final mixes so that the frequency spectrum of the recordings work well on all types of playback systems, the overall volume level is competitive with other mastered recordings and, in the case of an album, that the overall equalization and volume of each song creates a cohesive final product (Franz & Lindsay, 2004:222).

Masking also occurs between sources that are placed in the same location within the stereo image. Due to level adjustments and limited space within the stereo image between the loudspeakers, masking may present a major problem during mixing (Gibson, 2005:31). When performing the sweep technique, the effects of masking are incorporated when the recordist evaluates the microphone placement. Therefore, the recordist may use the sweep technique to adjust the timbre in such a way so as to reduce masking.

6.3 Sweeping with Digital Simulation Software

Digital guitar amplifier simulators have become incredibly popular. *System identification* is used to determine the normal output of an electric guitar and the output of a microphone placed in front of a guitar amplifier. System identification is the subject of identifying filter coefficients, given certain measurements of input and output signals. The guitar is played in a variety of ways to create a collection of input and output data and used to identify the sound of the amplifier (Smith, 2008:340). From this data, digital simulators may be designed. These simulators take the sound of a direct-recorded guitar and modify it to sound as if it is being played through an actual guitar amplifier (Bartlett, 2009:179). Simulators from companies such as Line 6 and Avid are very popular. Line 6 provides various amplifier models in their POD range of guitar effects pedals. Avid provides guitar models in the Amp Farm plug-ins for Pro Tools (Schonbrun, 2009:139). Many of these plug-ins have a graphical user interface that looks like the original equipment being modeled[3].

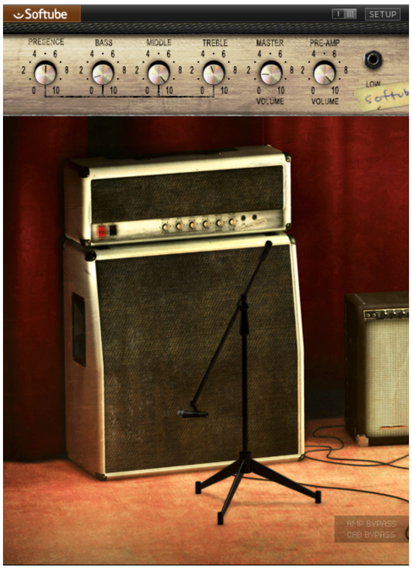
Many simulators not only model the sound of the instrument and amplifier, but also the effects of microphone placement. The Softube Vintage Amp Room[4] plug-in provides the user with three amplifier models and controls for equalization, master volume and preamplifier gain, as shown in Figure 6.4. The graphical user interface of the plug-in mimics that of a real recording setup, as shown in Figure 6.5a. The microphone position may be adjusted along a fixed path, as illustrated in Figure 6.5b. At position a, the microphone is on the edge and angled to the center of the loudspeaker. Moving towards position b, the angle of the microphone decreases until it is on-axis to the loudspeaker. From here, the microphone remains on-axis as it moves further from the amplifier toward position c.

The loudspeaker colouration of a signal is considered a primary element in the sound of an electric guitar (Chappell, 1999:50). Therefore, many simulators

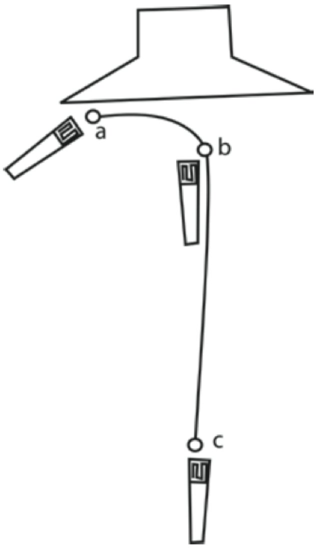
[3] See `http://line6.com/pod/` and `http://www.avid.com/US/products/Amp-Farm` for more information.

[4] See `http://www.softube.com/index.php?id=var` for more information.

Figure 6.4: Controls of the virtual guitar amplifier.



(a) One of three amplifier models.



(b) Microphone path (Softube, 2013:9).

Figure 6.5: Softube Vintage Amp Room

provide the option of selecting models of different sized loudspeakers or cabinets. Other features include effects such as delay, compression, distortion and reverberation.

As with the sweep technique, the microphone position in these simulators may be adjusted while the user listens to the response. Thereby, the user may identify virtual microphone placements that yield a suitable timbre. The benefit of using software simulations is that no auxiliary device is required to perform the sweep. However, the range through which the virtual microphone may be moved is limited to the capabilities of the software.

6.4 Limitations of Sweep Technique

The greatest limitation of the sweep technique is that equipment is required to control the position of the microphone. Whether an existing product is used or a device is designed and manufactured, it represents an additional expense. Furthermore, the mass of the microphones that the device can support, the workspace within which the microphone can be moved, as well as the range from which it can be operated is dependent on the capabilities of the device.

The speed at which the microphone can be moved is limited by wind noise and the Doppler effect. Noise is always undesirable, but moving the microphone too quickly may result in the occurrence of the Doppler effect. In such a case, the sound picked up by the microphone is not representative of the actual sound of the source. Similarly, mechanical vibrations due to the motion of the device or the vibration of its actuators negatively influence the pick-up of the microphone and cause noise. However, once a suitable microphone placement is found, the microphone and the device keeping it in place remain static, thereby preventing any further wind or mechanical noise.

6.5 Recommended use for the Sweep Technique

The sweep technique is not intended to enable a recordist to perform any conceivable microphone placement from the comfort of the mixing console, but instead it is intended to aid the making of fine adjustments in the final stages of the process. Therefore, the recordist must still rely on experience and knowledge of acoustics when determining a suitable microphone placement space within a recording environment. Only then should the recordist implement the sweep technique to perform finer adjustments to microphone placements, until a suitable sound has been found.

Conclusion

Instruments are very complex radiators that project sound multi-directionally, in different and constantly changing proportions (Woszczyk, 1979:2). Several different modes of vibration with different patterns of radiation are excited at the same time (Fletcher, 1998:256). The different spectra produced by the instrument or source are integrated into a pleasing composite only at some distance (Bartlett, 1981:726). Since the spectra varies with distance and radiation angle, the spectrum picked up by a microphone varies with its placement, thus microphone placement relative to an instrument greatly effects its recorded tone quality (Bartlett, 1981:726).

According to Wuttke (1999:4), placing the microphone appropriately with regard to the radiation pattern of the instrument is substantially more important than the type of microphone that is used. When considerable attention is paid to the exact microphone placement relative to the instrument, it can be used as a primary determinant of the timbre of the recorded sound (Moulton, 1990:162). There are several variables that effect the recorded timbre of a sound. These variables include the directivity of the sound radiation from the instrument, the position of the instrument relative to the microphone and its surroundings, the directional characteristics of the microphone, the distance between the microphone and the instrument and the acoustical characteristics of the room (Woszczyk, 1979:7). However, the process of microphone placement is correlative and changing one variable can have a significant effect on some or all of the other variables. Still, it is beneficial to experiment with all sorts of microphone positions until a suitable balance is found, as a recordist cannot compensate for poor microphone placement through equalization during the later stages of the project (Bartlett, 2012; Benade, 1985:232).

A proper understanding of the recording equipment, the operating principles of microphones and their polar patterns, sound propagation and acoustics, mitigates the need for excessive experimentation to find a placement that is suitable for a particular recording. However, it still remains a trial and error process.

The sweep technique, introduced in this study, further alleviates the need for excessive and time consuming experimentation by allowing the recordist to hear and evaluate several microphone placements in quick succession. This is achieved by sweeping the microphone through a range of positions, while the recordist monitors the pick-up of the microphone from the sweet spot of the studio loudspeakers. From here, the recordist is capable of performing critical listening and quickly make decisions with regards to the placement of the microphone. In addition, the recordist is capable of quickly making fine adjustments to a microphone's placement and immediately hear the effect on its timbre. To facilitate remote microphone positioning, the robotic anthropomorphic arm *Prototype 2* was developed.

The major drawback of *Prototype 2* was that, once the sweep technique had been performed and a suitable placement had been found, the functionality of *Prototype 2* was reduced to that of a microphone stand. There was no way to move the microphone to a conventional microphone stand and still retain its exact positioning. Therefore, *Prototype 2* could not be used for other sources until recording of the current source was completed. This strongly conflicted with the purpose of the sweep technique, which was to speed up and simplify the microphone placement process.

Future work will consist of designing a device that can facilitate remote microphone placement by attaching to a conventional microphone stand and actuating its components. The objective here will be to design the device so that, once microphone placement had been completed, it can be removed and attached to another microphone stand, without disrupting the position or orientation of the microphone. Furthermore, as the lack of visual feedback with regards to the position of *Prototype 2* proved problematic, the next prototype will include a visual feedback system.

Appendices

Appendix A

Decibel

Table A.1 lists commonly used reference values for power calculations in the audio industry.

Table A.1: Standard decibel reference values (Brown, 2008:25)

Electrical Power		
dBW	1	Watt
dBm	0.001	Watt
Acoustical Power		
dB-PWL or L_W	10^{-12}	Watt
Electrical Voltage		
dBV	1	Volt
dBu	0.775	Volt
Acoustical Pressure		
dB SPL or L_p	0.00002	Pascals

Table A.2 lists a sound pressure level range of 0 dB SPL to 130 dB SPL with the corresponding sound pressure. To illustrate how loud or soft these values are, an equivalently noisy or quiet environment that corresponds to each value is also listed. Finally, the equivalent musical dynamic is listed for the range 40 dB SPL to 100 dB SPL.

Table A.2: Decibel levels (Russ, 2009:42).

dB SPL (dB)	**Sound Pressure** (µbar)	**Power** ($\mathrm{W\,m^{-2}}$)	**Equivalent Environment**	**Musical dynamic**
130	632	10	Pain threshold	
120	200	1	Aircraft taking off	
110	63	10^{-1}	Loud amplified music	
100	20	10^{-2}	Circular saw	*fff*
90	6	10^{-3}	Train	*ff*
80	2	10^{-4}	Motorway	*f*
70	0.6	10^{-5}	Factory workshop	*mf*/*mp*
60	0.2	10^{-6}	Street noise	*p*
50	0.06	10^{-7}	Noisy office	*pp*
40	0.02	10^{-8}	Conversation	*ppp*
30	0.006	10^{-9}	Quiet room	
20	0.002	10^{-10}	Library	
10	0.0006	10^{-11}	Leaves rustling	
0	0.0002	10^{-11}	Threshold of hearing	

Appendix B

Experiments

B.1 Comb-filtering

The effect of comb-filtering on the frequency spectrum picked up by a microphone was studied. A microphone was set up relative to a guitar amplifier according to Figure 3.1a. The microphone was placed at a height of Y and a distance of X from the amplifier. Pink noise was recorded. A frequency analysis of the recorded signal yielded the frequency response shown in Figure B.1.

The frequency spectrum is shown in Figure B.1. In this frequency spectrum, the comb-like teeth, that is characteristic of comb filtering, is visible between 6 kHz to 10 kHz.

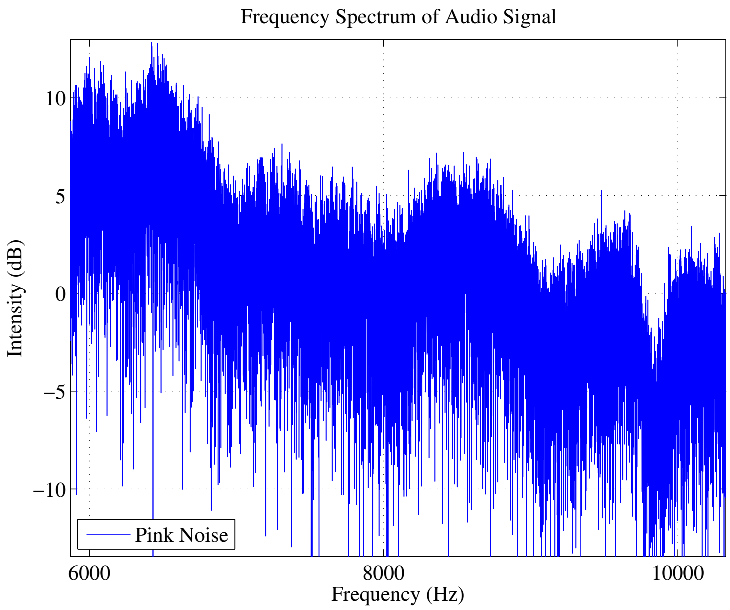


Figure B.1: Comb-filtering experiment

B.2 Level and Time Differences in Stereo Microphone Techniques

The level- and time differences that occur in the XY, ORTF and spaced pair microphone techniques were studied. Microphone configurations were used as described in Chapter 3, with spaced pair microphones set 60 cm apart. For each configuration, two recordings of a hit on a floor tom were made. First the floor tom was positioned in the center and on-axis of the microphone configuration. Then for the second recording, the floor tom was positioned 90° off-axis of the microphone configuration. Figure B.2 illustrates the experiment setup for the XY technique.

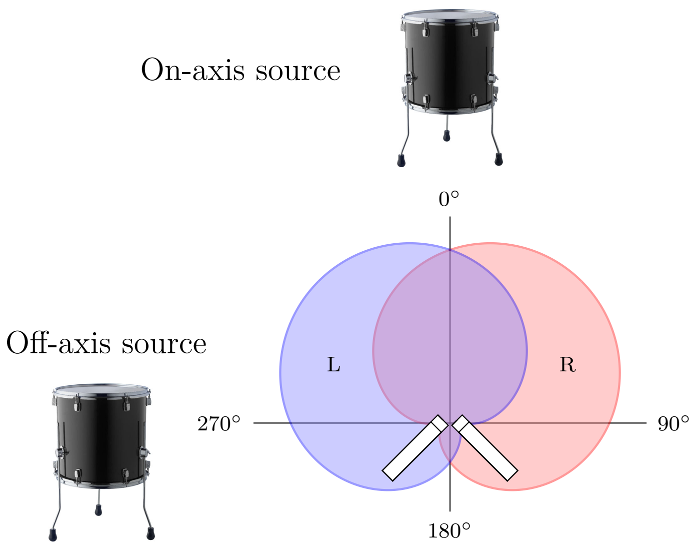


Figure B.2: Configuration for stereo microphone technique experiment.

The resulting waveforms are shown in Figures 3.5, 3.9 and 3.12 on page 58, on page 62 and on page 64. With the floor tom on-axis, the source was equidistant from either capsule and the sound wave that radiated from it approached each microphone from the same relative angle. Therefore, all three microphone techniques yielded no time or level differences.

With the floor tom off-axis, the XY technique still yielded no time and level differences, as shown in Figure 3.5b. In the XY technique, the capsules of the microphones are so close together that an approaching sound wave reaches them at essentially the same time. There is a time difference, however it is so small that it is negligible. The XY technique did yield a substantial level difference between the channels, because the microphone pair in an XY configuration are somewhat aimed in opposite directions.

For the same reason, the ORTF technique also experienced level difference between channels, as shown in Figure 3.9b. However, this level difference was more substantial due to the increased distance between microphone capsules. There was also a small time difference between channels, however this difference was still small. In comparison, the spaced pair yielded both large level and time difference between channels, as shown in Figures 3.12a and 3.12b. The spaced pair technique may be used with a distance of as much as a few meters between microphones. Such a configuration would yield even greater time and level difference between channels.

Appendix C

Equipment Specification

C.1 Avid HD I/O

The Avid HD I/O is a dedicated Pro Tools audio interface and digital signal processor. It features eight analog inputs and outputs, as well as additional digital inputs and outputs through AES/EBU, S/PDIF and ADAT. The analog-to-digital converters and the digital-to-analog converters both feature dynamic ranges greater than 120 dB, with minimal total harmonic distortion plus noise (THD+N). The specifications are as follows (Avid Technology Inc., 2010*a*:2):

Analog inputs	8
Analog outputs	8
AES/EBU I/O	8 + 2
S/PDIF I/O	1 in + 1 out (coax or optical)
ADAT I/O	8
S/MUX	S/MUX II and IV
Word Clock I/O	1 in + 1 out
Loop Sync I/O	1 in + 1 out
Digital interface	DigiLink Mini
Dynamic range ADC	122 dB
Dynamic range DAC	125 dB
THD+N ADC	−144 dB (0.0002 %)
THD+N DAC	−110 dB (0.000 32 %)
Frequency response ADC	20 Hz to 20 000 Hz (± 0.03 dB)
Frequency response DAC	20 Hz to 20 000 Hz (± 0.15 dB)

C.2 Dorsey Re-amp Box

The Dorsey re-amp box contains an inexpensive 600:600 Ω transformer. It outputs to a 680 Ω load resistor, which absorbs most of the signal entering it and ensures proper loading of the transformer. A small amount of signal is siphoned from the resistor and sent through a volume potentiometer and either a purely resistive load or a resistive and complex load which simulate guitar pickup source impedances. The routing is controlled through a toggle switch (Dorsey, 2014). The parts required for the re-amp box are listed in Table C.1, while the schematic is shown in Figure C.1.

Table C.1: Parts list of the Dorsey (2014) re-amp box.

Quantity	Name	Part Number
1	Female panel XLR	RSC1009-ND
1	10 k audio taper potentiometer	51AAD-B28-D15-ND
1	Enclosure	377-1101-ND
1	SC1104-ND	Quarter-inch guitar jack
1	Tamura transformer	MT4152-ND
1	0.1 H	M-7104-ND
1	SPDT locking switch	CKN1215-ND
1	Knob	226-4033-ND
1	681 Ω resistor	681XBK-ND
1	1 MΩ resistor	1.00MXBK-ND
1	10 kΩ resistor	10.0KXBK-ND

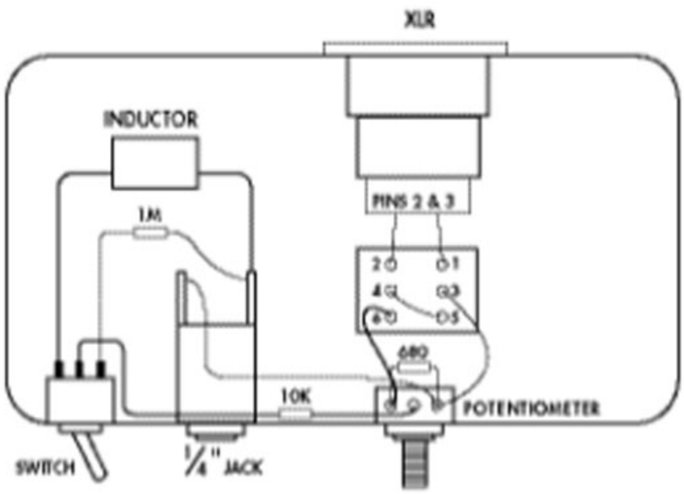


Figure C.1: Schematic of the Dorsey re-amp box

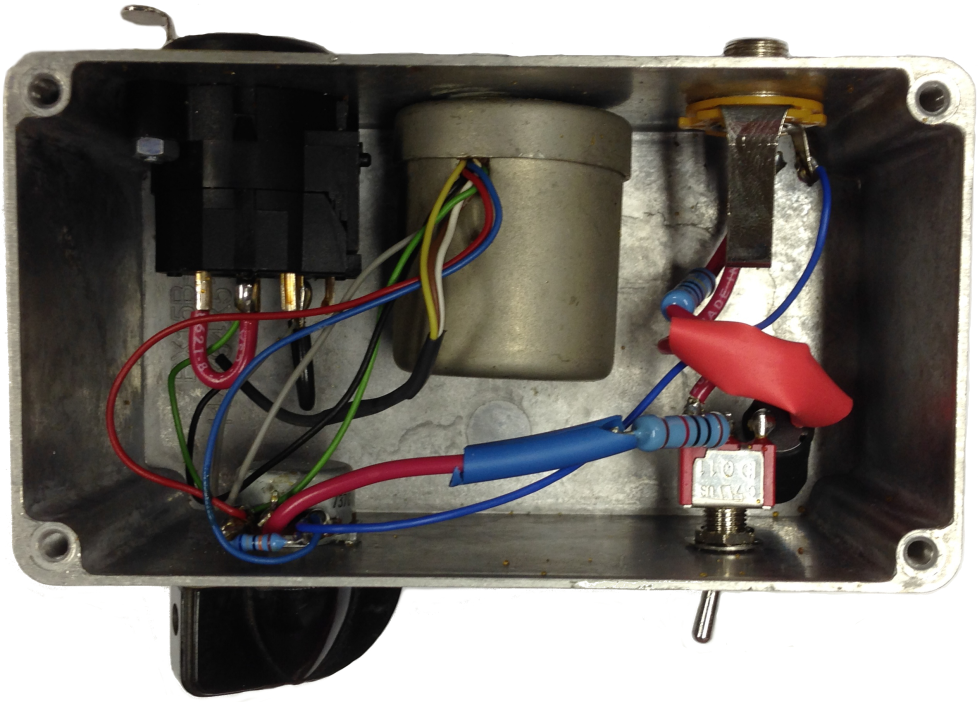


Figure C.2: Internal view of Dorsey re-amp box.

C.3 Fender Pro Junior III Guitar Amplifier

The Fender Pro Junior III is a small class A valve amplifier. Valve amplifiers are favoured by many guitarists for their warm timbre, great dynamic capabilities and ability to distort (Clement, 2004:38). The specifications listed in the owners manual are listed as follows (Fender Musical Instruments Corporation, 2010:7):

Type	PR 257
Power requirement	70 W
Power output	15 W RMS minimum into 8 Ω at 5 % THD, 1 kHz
Input impedance	1 MΩ
Speaker	1x 10” 8 Ω Fender Special Design speaker
Height	35.6 cm
Width	38.1 cm
Depth	22.2 cm
Weight	9.1 kg

C.4 Heil Sound PR 20 Dynamic Microphone

The Heil Sound PR 20 is a high quality dynamic microphone, designed for commercial broadcast, recording and live sound applications. Its specifications, as listed in the owners manual, are listed below (Heil Sound Ltd, 2010). In addition, the on-axis and 180° off-axis frequency response of the Heil Sound PR 20 is given in Figure C.3.

Output connection	3 pin XLR
Element type	Dynamic
Frequency response	50 Hz to 18 000 Hz
Polar pattern	Cardioid
Rear rejection at 180° off axis	−30 dB
Impedance	600 Ω balanced
Output level	−55 dB at 1 kHz
Weight	397 g
Maximum SPL	145 dB

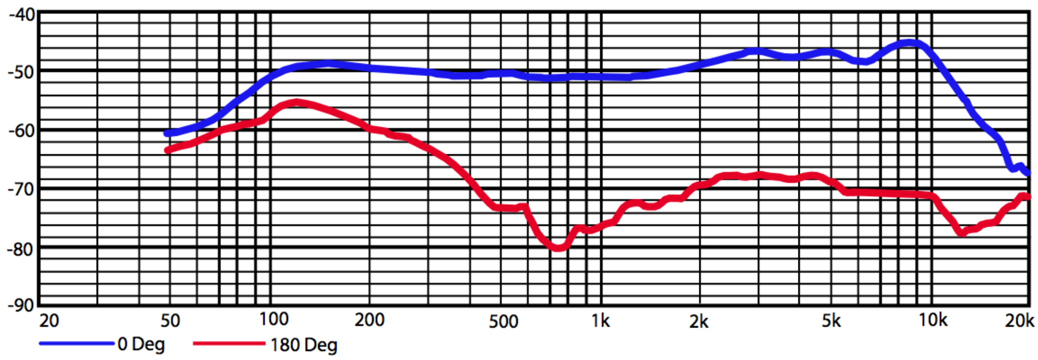


Figure C.3: Heil Sound PR 20 frequency response (Heil Sound Ltd, 2010).

C.5 BuzzAudio MA2.2 Preamplifier

The BuzzAudio MA2.2 is a Class A stereo preamplifier. The owners manual lists the following specifications (Buzz Audio, 2000:9):

Min Gain	+16dB (-4dB with pad in)
Max Gain	+65dB
Maximum Output Level	+24dBu unbalanced.
Frequency Response	2Hz to 250kHz @ 20dB gain (-3dB).

. . . . 20Hz to 250kHz @ 65dB gain (-3dB).
Harmonic Distortion . . . . less than 0.008% 100Hz to 10kHz.
Slew Rate . . . . 140 V/uS, @ +20dBu output level.
EIN . . . . -133.5dB A wtg, 150ohm source Z.
Signal to Noise Ratio . . . . -74dBu A wtg, input shorted.
CMNR . . . . 100Hz-80dB, 1kHz -80dB, 10kHz-70dB.
Channel Crosstalk . . . . below noise.
Input Impedance . . . . 3k ohms/1k2 ohms switchable

C.6 RadioShack Digital Sound Level Meter

The RadioShack Digital Sound Level Meter is a hand-held device with a sound measuring range of 50 dB to 126 dB. The device can measure averaged, maximum and minimum sound levels over a preset time. In addition, the weighting characteristics may be set to either A- or C-weighting. When set to A-weighting, the device responds to frequencies in the 500 Hz to 10 000 Hz range. For C-weighting, the device responds to frequencies ranging from 32 Hz to 10 000 Hz (Janardhan, 2008:83-84). When set to *fast*, the device updates the displayed bar graph every 0.2 s. When set to *slow*, the bar graph is updated every 0.5 s. The specifications according to the user manual are listed below (RadioShack Corporation, 2011:12):

Model 33-2055
Battery . . . . 9 V Alkaline
Microphone . . . . Electret Condenser
Range . . . . 50 dB to 126 dB
Accuracy . . . . ±2 dB at 114 dB SPL
Reference . . . . 0 dB = 0.0002 Micro Bar
Weighting . . . . A and C
Display Response . . . . Fast and Slow
Voltage . . . . 1 Volt Peak-Peak Min.
. . . . (Open Circuit, Full Scale at 1 kHz)
Impedance . . . . 10 kΩ Min. load
Distortion . . . . Less than 2 % at 1 kHz, 0.5 V p-p Output
. . . . (Input: Microphone output, Output: 10 Ω)
Operating Temperature . . . . 0 °C to 50 °C
Storage Temperature . . . . −40 °C to 65 °C
Dimensions . . . . 159 x 64 x 44 mm
Weight (including battery) . . . . 190 g

C.7 Karma Silver Bullet

The Karma Silver Bullet is a small diaphragm condenser microphone with an omni-directional polar pattern. A photograph of the microphone is shown in Figure C.4. It measures a mere 48 mm in length. At its base, the diameter measures 18 mm, tapering off to 16 mm at its tip. As shown in Figure C.5, the Karma Silver Bullet has an overall flat frequency response, with −4 dB dip at 2 kHz and a 6 dB boost just before 6 kHz. Specifications from the Karma website are listed below (Karma Mics, 2014).

Figure C.4: Karma Silver Bullet condenser microphone.

Model	Silver Bullet
Transducer	Condenser
Polar pattern	Omni-directional
Frequency response	20 Hz to 20 000 Hz
Sensitivity	−32(2) dB
Maximum SPL for 0.5 THD	135 dB
Mass	0.024 kg
Length	48 mm
Diameter (base)	18 mm
Diameter (tip)	16 mm

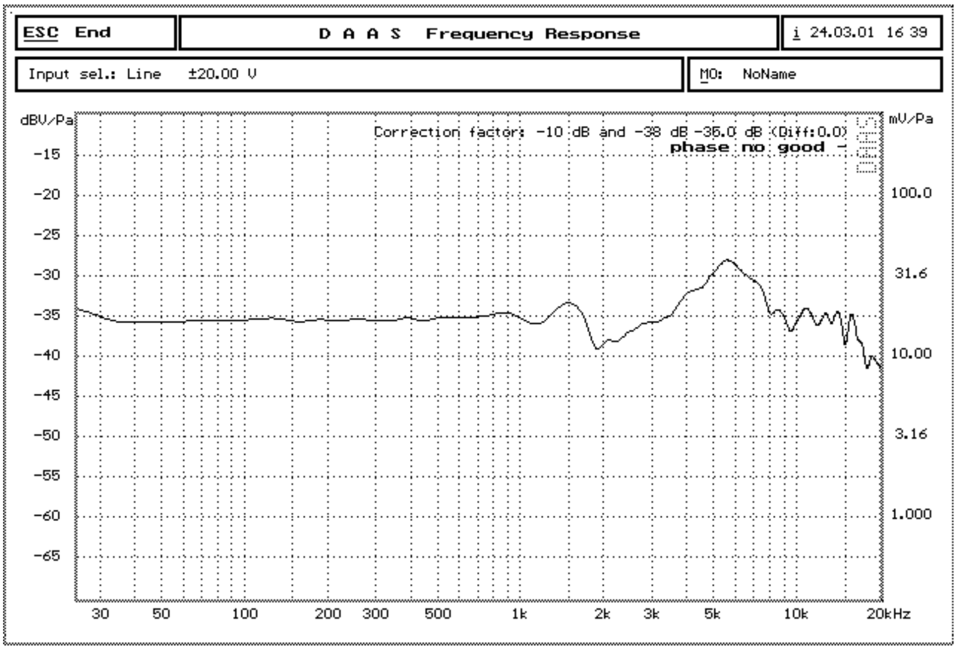


Figure C.5: Karma Silver Bullet frequency response (Karma Mics, 2014)

C.8 Electronic Components

Table C.2 lists a table of all the electronic components used in this study.

Table C.2: Parts list of electronic components.

Component	Quantity	Manufacturer/Supplier	Model
Microcontroller	1	Arduino	Uno
Fuse	2	Yebo Electronics	1.5 A 250 V
Fuse	2	Yebo Electronics	600 mA 250 V
Capacitor	2	Hitano	2200 µF 63 V
Stepper motor	2	Prototronics	28byj-48
Stepper motor driver	2	Prototronics	ULN2003AN
Servo motor	2	Communica	RA001B-S06NF
Controller	1	Sony	Dualshock 3
Power supply	1	HuntKey	CP-350

Appendix D

Noise as Test Signals

Noise is a random, unpredictable signal that is added to the audio signal during processing, either analog or digital. Sources of noise, as described by Davis (2007:756), include the following:

Acoustical: Random fluctuations of air molecules against the microphone diaphragm.

Electrical: Fluctuations in electrical charge in analog electrical circuits from discrete electrons.

Mechanical: Surface irregularities in grooved media.

Magnetic: Irregularities of magnetic strength from discrete magnetic particles.

Artimetic: Random errors from digital signal processing quantization of numerical signals to discrete values.

Noise is typically undesirable in a recording, however, it has other specific uses (Dabbs, 2004:160). White and pink noise, shown in Figure D.1, are electronically generated noises that are often used as test signals (Everest & Pohlmann, 2009:85). White noise contains a constant power per frequency bandwidth, as illustrated in Figure D.2. Therefore, the power between 100 Hz and 200 Hz is equal to the power between 1100 Hz and 1200 Hz (Self, 2010:9). Theoretically, it contains contributions from all frequencies throughout an infinite bandwidth, but in practice this bandwidth is merely broad (Mims, 2000:18). Audible white noise contains equal contributions from all frequencies perceptible to the human ear. It is analogous to white light, which comprises all wavelengths of light, and therefore colours, perceivable by the human eye (Kefauver, 2001:246). Each higher octave contains twice as many 1 Hz bands as the previous octave, therefore, white noise energy doubles with each increasing octave (Everest & Pohlmann, 2009:85).

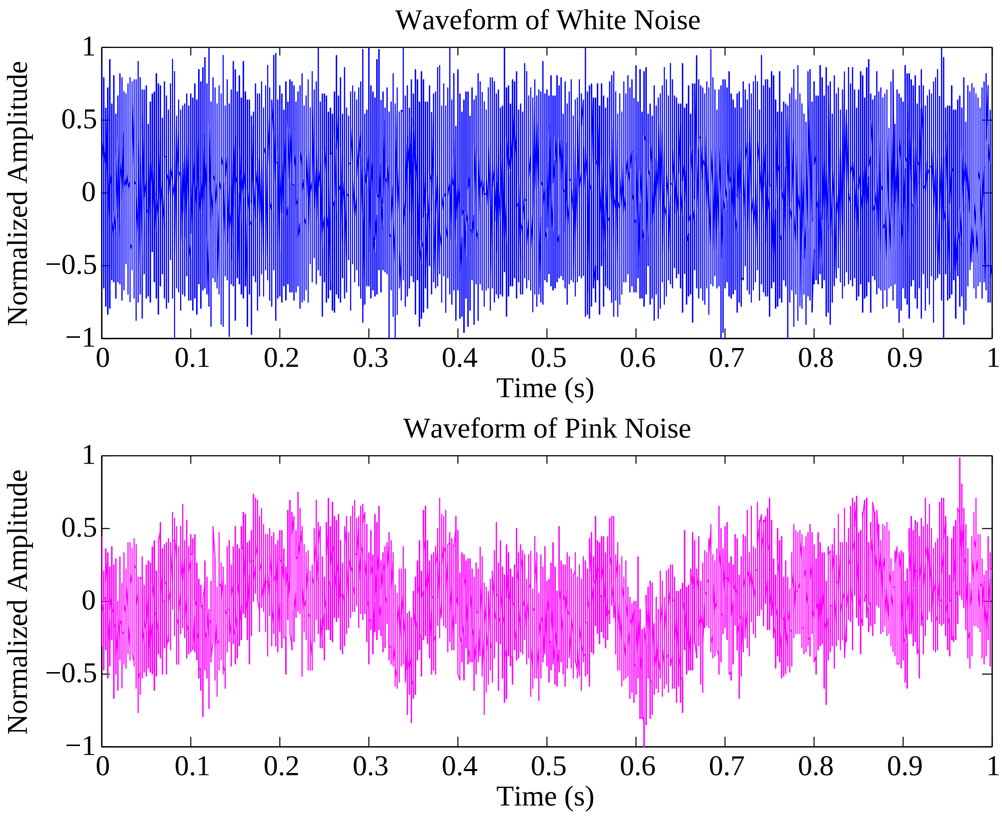


Figure D.1: Waveforms of white and pink noise.

Pink noise consists of bandlimited white noise, with characteristcs similar to that of white noise that has been passed through a −3 dB per octave filter (Ott, 2011:9.51). As a result, pink noise decreases at a rate of 3 dB per octave, as shown in Figure D.3. Pink noise has equal power bandwidth ratio. Therefore, the bandwidth from 100 Hz to 200 Hz contains the same power as the bandwith ranging from 200 Hz to 400 Hz (Self, 2010:9). Pink noise is used for testing amplifiers and loudspeakers as it presents a constant level of noise accross the human hearing range (Swallow, 2010:145).

Other coloured noises include *red*, *blue*, *violet* and *grey* noise. Coloured noise refers to any broadband noise with a non-white spectrum (Vaseghi, 2008:39). The energy of red noise decreases at a rate of 6 dB per octave. Red noise is alternatively known as *brown* noise. Blue noise energy increases with frequency at 3 dB per octave, while that of violet noise increases at 6 dB per octave. Grey noise is actually pink noise modified by an equal loudness curve to give the perception of equal loudness at all frequencies (Self, 2010:9).

As an alternative to white and pink noise, *swept sine waves* are also used as test signals. A swept sine wave[1] consists of a sine wave of which the frequency is smoothly varied, either increasing or decreasing. The frequency may be

[1] Alternatively known as a *chirp* (Su, 2006:631).

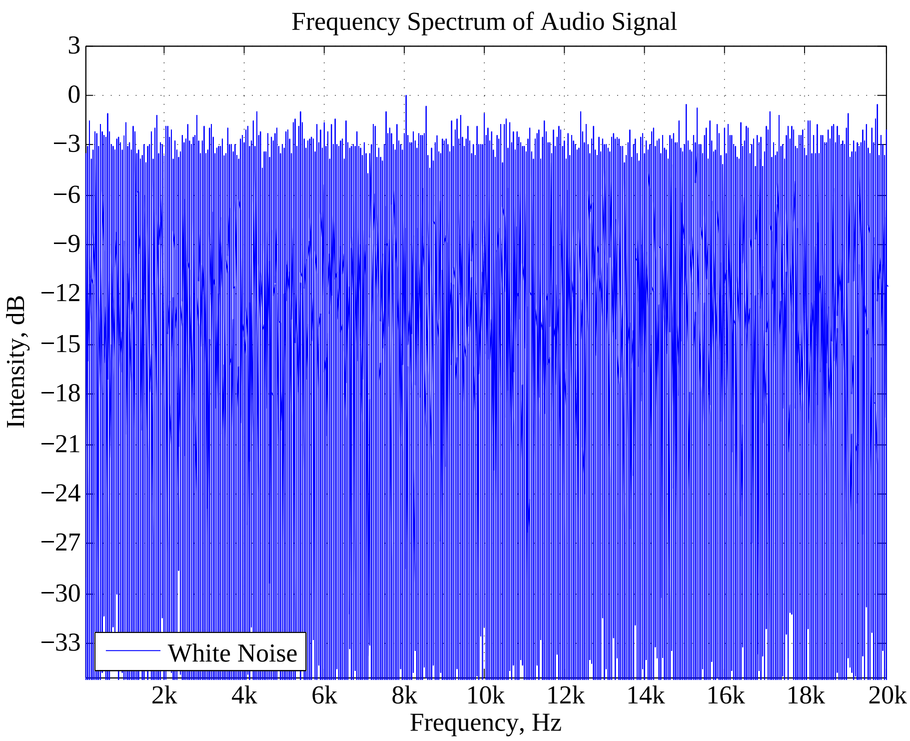


Figure D.2: Frequency spectrum of white noise.

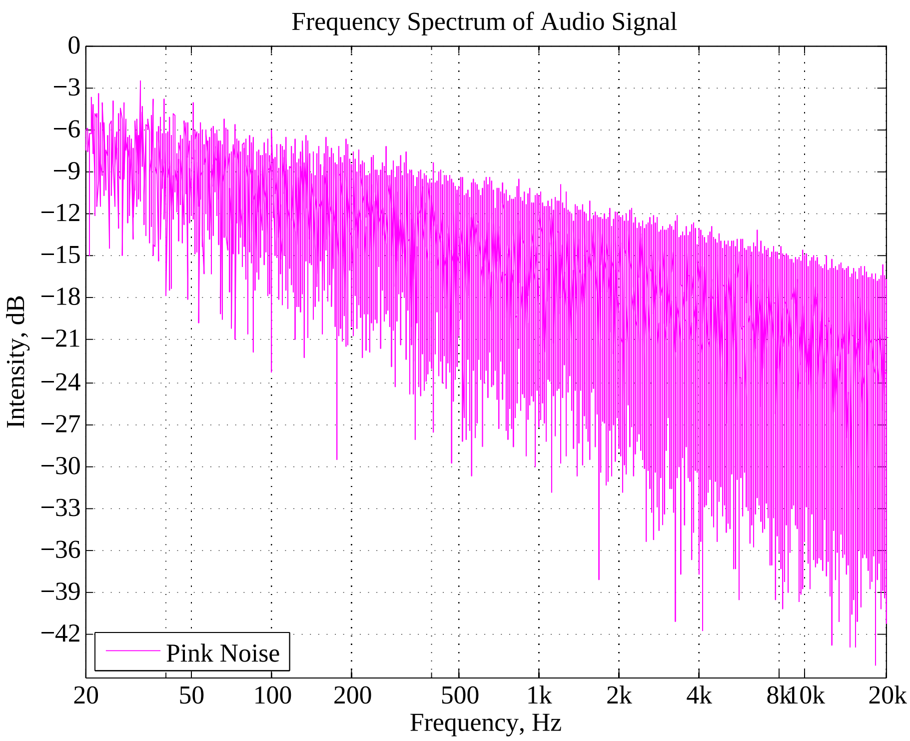


Figure D.3: Frequency spectrum of pink noise.

varied linearly or logarithmically. A linearly swept sine wave produces a near uniform frequency spectrum, as shown in Figure D.4, which is similar to white noise. A logarithmically swept sine wave produces frequency spectrum that decreases at a rate of 3 dB per octave (Kuttruff, 2009:261) and corresponds with the frequency spectrum of pink noise.

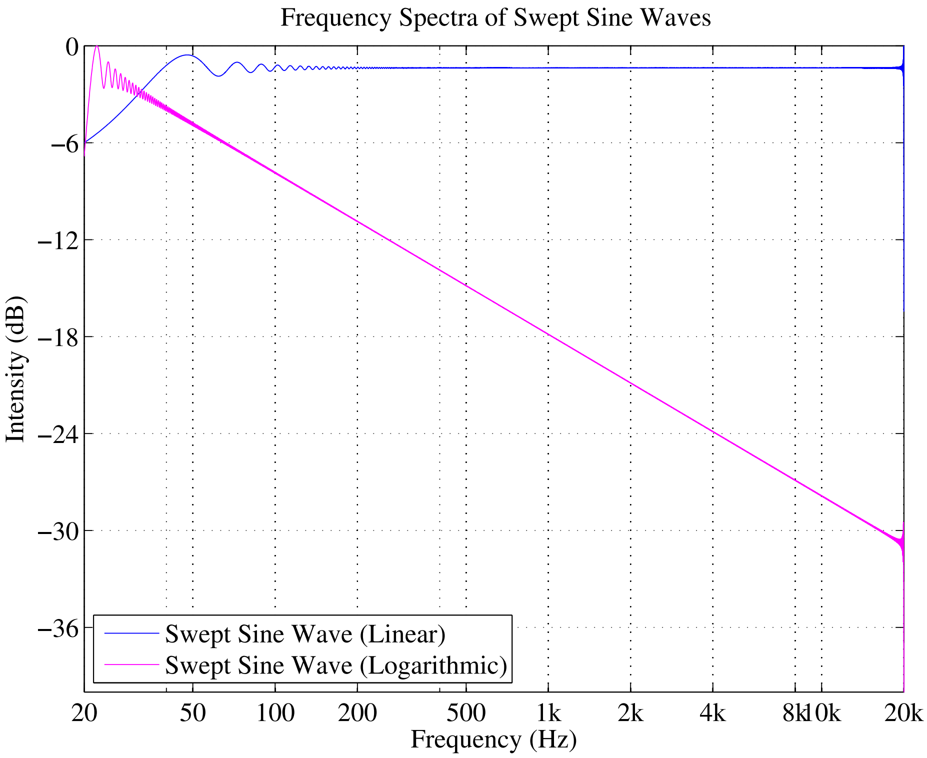


Figure D.4: Frequency spectra of linearly and logarithmically swept sine waves.

The advantages of a swept sine testing over either white or pink noise testing is that at any particular time, the power of the signal is dedicated to a single frequency. This is shown in the spectrograms of linearly and logarithmically swept sine signals, shown in Figures D.5 and D.6. A distadvantage is that swept sine tests take longer than noise tests (de Silva, 2007:1.107).

Equation (D.0.1) shows the equation for a logarithmic swept sine wave, where ω_1 is the initial angular velocity in radians, ω_2 is the final angular velocity and T is the duration of the sweep in seconds (Farina, 2007:3).

$$x(t) = \sin\left[\frac{w_1 t}{\ln(\frac{w_2}{w_1})}(e^{\frac{t}{T}\ln(\frac{w_2}{w_1})} - 1)\right] \tag{D.0.1}$$

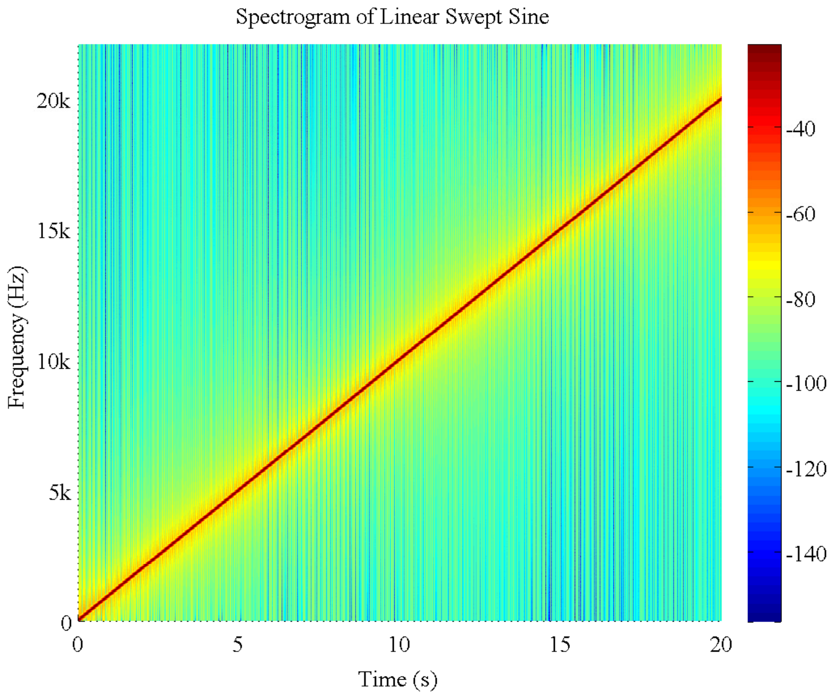


Figure D.5: Spectogram of a linearly swept sine wave.

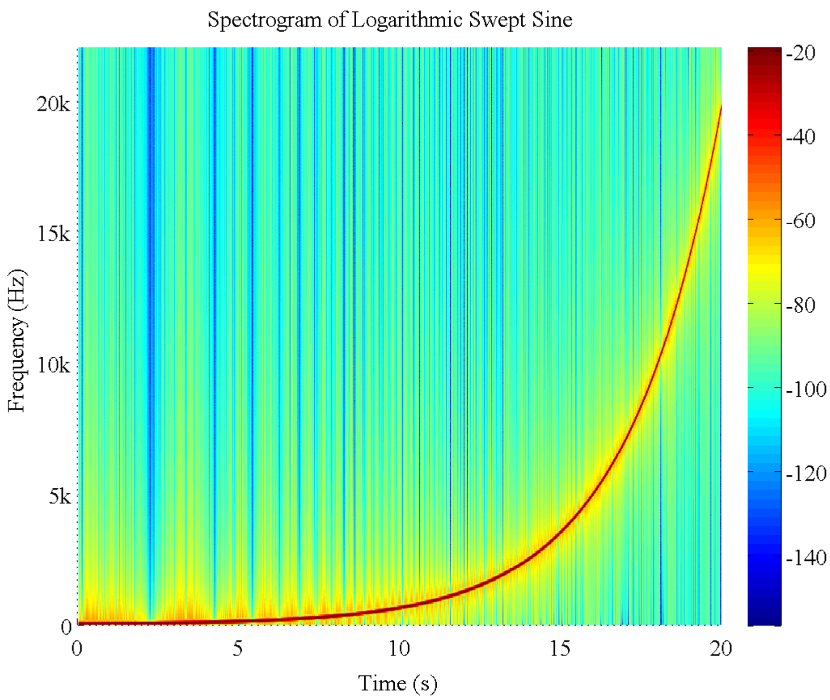


Figure D.6: Spectogram of a logarithmically swept sine wave.

Appendix E

Software

E.1 Audacity

As mentioned on its website[1], Audacity is a free, open source, cross-platform software for recording and editing sounds. Audacity is designed for music-orientated audio applications, supports WAV, AIFF, AU, FLAC and other major audio formats and is fully customizable (Li *et al.*, 2006:1; Schroder, 2011:xx). The most important feature of Audacity for this study was its sound analysis capabilities. These features include spectrogram view modes for visualizing frequencies and the plot spectrum command for detailed frequency analysis (Cai, 2014:82). Screen-shots of these features are shown in Figures E.1 and E.2.

In addition, Aurora software modules were installed to add additional features to Audacity. Specifically of interest was the *Chirp* function, which enables the user to generate a swept sine wave. The Aurora software modules were created by Angelo Farina[2] and are freely available[3]. Audacity 2.0.5 was used throughout this study.

E.2 Microsoft Excel

Microsoft Excel 2011 was used in conjunction with the Solver add-on. Solver is a numerical computational tool that can calculate maximum, minimum or target values for a specified target cell by to changing the values of specified linked cells within a set of constraints. In a typical problem, the constraints and the target cell are functions of the changing cells. Solver employs iterative numerical methods, which involves plugging in trial values for changing cells and observing the results calculated by the constraint cells

[1] http://audacity.sourceforge.net/
[2] http://pcfarina.eng.unipr.it/
[3] http://pcfarina.eng.unipr.it/Public/Aurora-for-Audacity/

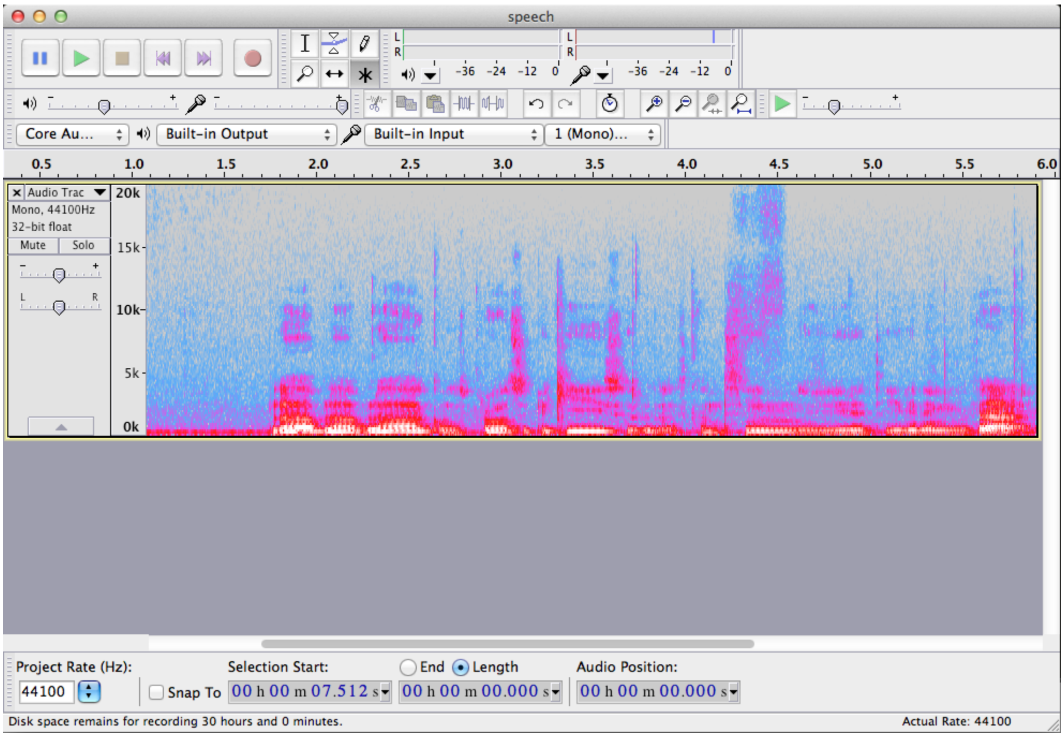


Figure E.1: AUDACITY spectrogram.

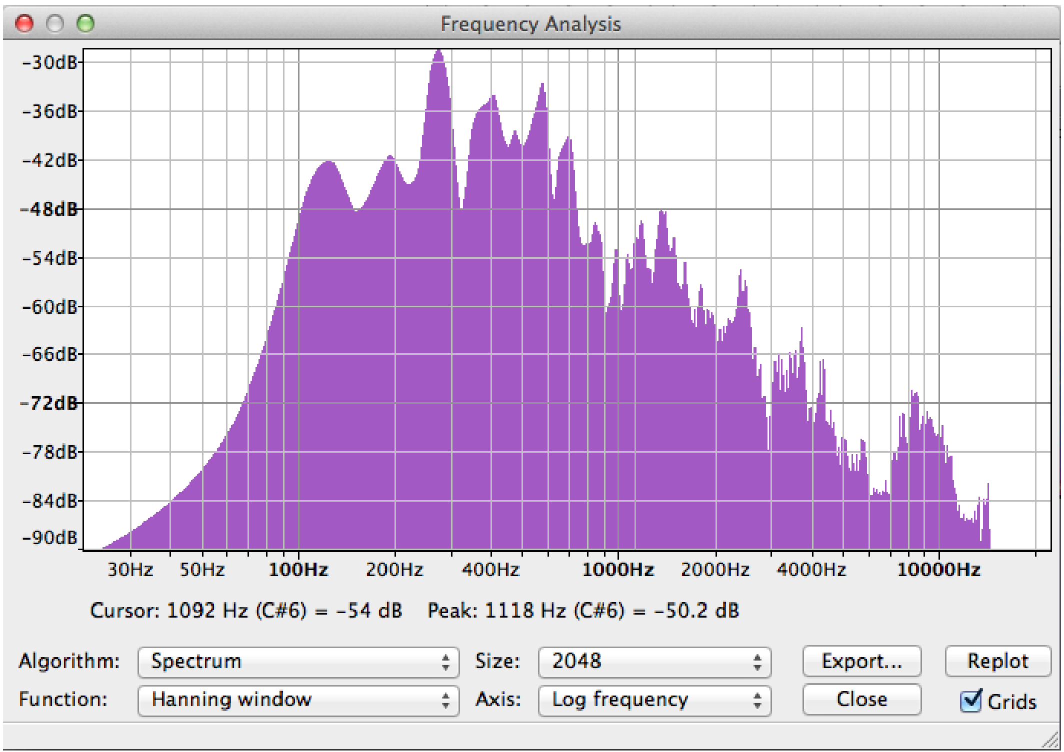


Figure E.2: AUDACITY frequency spectrum.

and the target cell. Each trial is called an iteration. Since a pure trial and error approach would be extremely time-consuming, SOLVER performs extensive analysis on the observed outputs, and the rates of change as inputs are varied, to guide the selection of new trial values (Liou, 2007:488). Once the output values of SOLVER have converged, a solution is found.

SOLVER was developed primarily for solving optimization, i.e. finding maximum and minimums, however it may also be used to solve equations (Liengme, 2008:211). SOLVER was used to determine the spring properties that fulfilled the design requirements, but produced the least uncorrected torsional stress within the spring. The parameters for the optimization are listed in Table F.2, with the resulting spring properties listed in Table F.3.

E.3 MATLAB

Designed for scientific and engineering use, MATLAB (MATrix LABoratory) is a powerful interactive system for matrix based computation. It is useful for many forms of numeric computation and visualization. The MATLAB language is a high-level matrix/array language which includes control flow statements, functions, data structures, input/output, and object-orientated programming features (Kharab & Guenther, 2011:1).

MATLAB was used to analyze audio files and plot sound waveforms, spectrograms and frequency responses.

E.3.1 Frequency Spectrum Code

```
[language=Matlab]

% Frequency Spectrum

char colour;
for i = 0:1:1

switch i    % Read audio files
case 0
[y,fs] = audioread('swept_sine_lin_test_signal.wav');
colour = 'b';
case 1
[y,fs] = audioread('swept_sine_log_test_signal.wav');
colour = 'm';
end

NFFT = length(y)                  % Determine length
y = fft(y,NFFT);                  % Calculate FFT
f = ((0:1/NFFT:1-1/NFFT)*fs).'; % Frequencies
```

```
magY = abs(y)/NFFT;              % Calculate absolute value of FFT
max_value = max(magY);           % Calculate max value for scaling
magY = magY/max_value;           % Scale FFT to max value

% Plot figure
semilogx(f,10*log10(magY.^2),colour);   % Plot graph
hold on

% Graph limits
xlim([20 20000]);                % x limits

% Graph ticks
set (gca, 'XTick',[20,50,100,200,500,1000,
2000,4000,8000,10000,20000])
set(gca,'XTickLabel','20|50|100|200|500|1k|
2k|4k|8k|10k|20k')
set (gca, 'YTick',[-42,-39,-36,-33,-30,-27,
-24,-21,-18,-15,-12,-9,-6,-3,0])
set(gca,'YTickLabel','-42|-39|-36|-33|-30|
-27|-24|-21|-18|-15|-12|-9|-6|-3|0')
hold on

% Graph labels
set(gca, 'FontName', 'Times New Roman', 'FontSize', 14)
title('Frequency Spectrum of Audio Signal')
xlabel('Frequency, Hz')
ylabel('Intensity, dB')

% Graph legend
leg = legend('Swept Sine Wave (Linear)',
'Swept Sine Wave (Logarithmic)');
set(leg,'Location','SouthWest');
grid on
end
hold off
```

E.3.2 Spectrogram Code

```
[language=Matlab]
% Spectograms

% Read audio file
[x,fs] = audioread('piano.wav');
% Plot spectrogram
spectrogram(x, 1024, 3/4*1024, [], fs, 'yaxis');
% Add colour bar
```

```
h = colorbar;
% Export plot
print('-depsc','-tiff','-r300','piano_spectrogram')
```

E.4 AUTODESK Inventor Professional

AUTODESK Inventor Professional is computer aided design and drafting (CADD) software. It is a 3-D solid modeling and 2-D drafting program generally for CADD in the manufacturing industry. Autodesk Inventor Professional provides a comprehensive and flexible set of software for 3-D mechanical design, simulation, design vizualiazation and communication, tooling creation and 2-D documentation (Madsen, 2011:73). This software was used to design the robotic prototypes for this project.

E.5 LaTeX, Sublime Text and BibDesk

This document was produced in LaTeX[4]. LaTeX is a markup language designed and implemented by Leslie Lamport and based on Donald E. Knuth's typesetting language TeX (Gratzer, 1999:xiv). Essentially it is a programming language which dictates the conversion of text and LaTeX commands into a printable document. The output of LaTeX is of extremely high quality, comparable with good quality books and journals (Wilkins, 1995:2). LaTeX is especially popular in the scientific community, because it contains features that make it very easy to write complex mathematics and formulae.

LaTeX is one of a number of 'dialects' of TeX. It has features for automatic numbering of chapters, sections, theorems, equations etc., and extensive cross-referencing capabilities, which make it particularly suited for the production of long articles, books, theses and dissertations (Wilkins, 1995:2).

Sublime Text 2.0.2[5] is a cross-platform programming editor that was used for writing this LaTeX document. Sublime Text is a glorious editor with rich features, all designed to increase efficiency and enhance the user's experience. Features include multiple cursors, an extensive *Command Palette* with *fuzzy logic* string matching, support for multiple programming languages and a vast array of plug-ins. Arguably the most powerful feature of Sublime Text is that nearly every aspect of the software is completely customizable.

BibDesk[6] is a citation management software that was used to store and categorize references. BibDesk not only keeps track of bibliographic information, but organizes and renames associated files according to the user's specification and is specifically well suited for LaTeX users.

[4] http://www.latex-project.org/
[5] http://www.sublimetext.com/
[6] http://bibdesk.sourceforge.net/

E.6 Arduino

ARDUINO not only refers to the microcontroller, but to an open-source development platform. The ARDUINO programming language is based on the C programming language (Wheat, 2011:26). The ARDUINO platform was used to calculate forward and inverse kinematics and control the various motors of the robot prototype designed in this project.

E.7 Arduino code

```
//------------------- Robot Arduino Code -------------------------

 #include <Stepper.h>
 #include <Servo.h>

 // Define constants
 const float L_1 = 200.0;   //mm Length of link 1
 const float L_2 = 200.0;   //mm Length of link 2

 int motor1Pin1 = 8;        // Blue   - 28BYJ48 pin 1
 int motor1Pin2 = 9;        // Pink   - 28BYJ48 pin 2
 int motor1Pin3 = 10;       // Yellow - 28BYJ48 pin 3
 int motor1Pin4 = 11;       // Orange - 28BYJ48 pin 4

 int motor2Pin1 = 4;        // Blue   - 28BYJ48 pin 1
 int motor2Pin2 = 5;        // Pink   - 28BYJ48 pin 2
 int motor2Pin3 = 6;        // Yellow - 28BYJ48 pin 3
 int motor2Pin4 = 7;        // Orange - 28BYJ48 pin 4

 Stepper motor1(512,motor1Pin1,motor1Pin2,motor1Pin3,motor1Pin4);
 Stepper motor2(512,motor2Pin1,motor2Pin2,motor2Pin3,motor2Pin4);

 Servo servo1;
 Servo servo2;

 float theta1;
 float theta2;

 float inc = 5.0;          // servo incremental value
 int steps = 4;            // stepper incremetal value

 int message = 0;          // Holds one byte of Serial message

 float x;                  //cm Horizontal position
```

```
float y;                    //cm Vertical position


void setup()
{
  Serial.begin(9600);  //set serial to 9600 baud rate
  pinMode(motor1Pin1, OUTPUT);
  pinMode(motor1Pin2, OUTPUT);
  pinMode(motor1Pin3, OUTPUT);
  pinMode(motor1Pin4, OUTPUT);
  pinMode(motor2Pin1, OUTPUT);
  pinMode(motor2Pin2, OUTPUT);
  pinMode(motor2Pin3, OUTPUT);
  pinMode(motor2Pin4, OUTPUT);

  motor1.setSpeed(25);
  motor2.setSpeed(25);
  servo1.attach(2);
  servo2.attach(3);

  x = 100;
  y = 100;
  inverseKinematics(x-30.0,y-10.0); // Initial position
  moveServo(theta1,theta2);         // including offset
  forwardKinematics();
}

void loop()
{
  if (Serial.available() > 0)   // Check if serial message available
  {
    message = Serial.read();     // Read serial message

    switch(message)             // Perform action based on input
    {
      case 'L':
        moveStepper(-steps,steps);
      break;

      case 'R':
        moveStepper(steps,-steps);
      break;

      case 'l':
        moveStepper(0,-steps);
      break;
```

```
case 'r':
  moveStepper(0,steps);
break;

case 'U':
  if(inRange(x,y+inc) == true)
  {
     //Calculate theta1 and theta2 for new (x,y) position
     inverseKinematics(x,y+inc);
     moveServo(theta1,theta2);
  }
  else
  Serial.println("Out of range");
break;

case 'D':
  if(inRange(x,y-inc) == true)
  {
     //Calculate theta1 and theta2 for new (x,y) position
     inverseKinematics(x,y-inc);
     moveServo(theta1,theta2);
  }
  else
  Serial.println("Out of range");
break;

case 'F':
if(inRange(x+inc,y) == true)
  {
     //Calculate theta1 and theta2 for new (x,y) position
     inverseKinematics(x+inc,y);
     moveServo(theta1,theta2);
  }
  else
  Serial.println("Out of range");
break;

case 'B':
  if(inRange(x-inc,y) == true)
  {
     //Calculate theta1 and theta2 for new (x,y) position
     inverseKinematics(x-inc,y);
     moveServo(theta1,theta2);
  }
  else
```

```
        Serial.println("Out of range");
      break;
    }
  }
}

void forwardKinematics()
{
  theta1 = servo1.read();  // reads servo position
  theta2 = servo2.read();
  x = L_1*cos(deg2rad(180.0-theta1)) + L_2*cos(deg2rad(90.0-theta2));
  y = L_1*sin(deg2rad(180.0-theta1)) + L_2*sin(deg2rad(90.0-theta2));
}

// Calculate theta1 and theta2 from (x,y)
void inverseKinematics(float kin_x, float kin_y)
{
  int oldTheta1 = theta1;
  int oldTheta2 = theta2;
  float B = sqrt(sq(kin_x) + sq(kin_y));
  float q_1 = atan2(kin_y, kin_x);
  float q_2 = acos((sq(B) + sq(L_1) - sq(L_2))/(2.0*B*L_1));
  float theta1_rad = PI - (q_1 + q_2);
  float theta2_rad = PI/2 + theta1_rad - acos((sq(L_1)
      + sq(L_2) - sq(B))/(2.0*L_1*L_2));
  theta1 = rad2deg(theta1_rad);
  theta2 = rad2deg(theta2_rad);

  if (isnan(theta1) == 1 || isnan(theta2) == 1)
  {
  theta1 = oldTheta1;
  theta2 = oldTheta2;
  Serial.println("Nan error");
  }
  else if (theta1 < 0 || 180 < theta1 || theta2 < 0 || 180 < theta2)
  {
  theta1 = oldTheta1;
  theta2 = oldTheta2;
  Serial.println("range error");
  }
}

float rad2deg(float rad_value) // Convert radians to degress
{
  float deg_value = rad_value*180.0/PI;
  return deg_value;
```

```
}

float deg2rad(float deg_value) // Convert degress to radians
{
  float rad_value = deg_value*PI/180.0;
  return rad_value;
}

boolean inRange(float check_x,float check_y)
{
  if(0.0 <= check_x && check_x <= 400.0 && 0.0 <= check_y
      && check_y <= 400.0 && sqrt(sq(check_x) + sq(check_y)) <= 400.0)
    {
      Serial.println("number is in range");
      x = check_x;
      y = check_y;
      return true;
    }
  else
  {
    return false;
    Serial.println("number is out of range");
  }
}

void moveStepper(int steps1, int steps2)
{
    for (int i = 1; i < 4; i++)
    {
      motor1.step(steps1);
      motor2.step(steps2);
      steps1=steps1/2;
      steps2=steps2/2;
    }
}

void moveServo(float theta1, float theta2)
{
    servo1.write(theta1+10);
    servo2.write(theta2+10);
}
```

E.8 Processing

PROCESSING is based on the JAVA programming language, but the programming environment is very similar to that of ARDUINO. Processing was used to receive and relay input messages from a BLUETOOTH connected controller, through the serial USB connection, to the ARDUINO.

```
// ps3_arduino_control

import org.gamecontrolplus.gui.*;
import org.gamecontrolplus.*;
import net.java.games.input.*;
import processing.serial.*;

ControlIO control;
ControlDevice stick;

Serial arduinoPort;
String arduinoMessage;  //Message to be sent to Arduino

public void setup()
{
  size(568,320);
  frameRate(25);

  control = ControlIO.getInstance(this);
  // Identify suitable controller
  stick = control.getMatchedDevice("robot");
  if (stick == null)
  {
    println("No suitable device configured");
    System.exit(-1);
  }
  // Serial port
  arduinoPort = new Serial(this,Serial.list()[2],9600);
}

public void draw()
{
    background(0);
    //Left D-pad
    fill(0,50,0);         // Fill color
    stroke(0, 225, 0);    // green line
    rect(89,25,81,81);    // UP
    rect(8,106,81,81);    // LEFT
    rect(89,187,81,81);   // DOWN
    rect(170,106,81,81); // RIGHT
```

```
  //Right D-pad
  rect(359,25,81,81);  // FORWARD
  rect(359,187,81,81); // BACKWARD
  stroke(200, 200, 0); // yellow line
  fill(50,50,0);       // Fill color
  rect(278,106,81,81); // ANGLE LEFT
  rect(440,106,81,81); // ANGLE RIGHT

\\ If certain button pressed, send corresponding message
if(stick.getButton(" UP").pressed())
{
    arduinoPort.write("U");
    color_green();
    rect(89,25,81,81);
}
else if (stick.getButton("RIGHT").pressed())
{
    arduinoPort.write("R");
    color_green();
    rect(170,106,81,81);
}
else if (stick.getButton("DOWN").pressed())
{
    arduinoPort.write("D");
    color_green();
    rect(89,187,81,81);
}
else if (stick.getButton("LEFT").pressed())
{
    arduinoPort.write("L");
    color_green();
    rect(8,106,81,81);
}
else if (stick.getButton(" FORWARD").pressed())
{
    arduinoPort.write("F");
    color_green();
    rect(359,25,81,81);
}
else if (stick.getButton("BACKWARD").pressed())
{
    arduinoPort.write("B");
    color_green();
    rect(359,187,81,81);
}
```

```
  else if (stick.getButton(" ALEFT").pressed())
  {
      arduinoPort.write("l");
      color_yellow();
      rect(278,106,81,81);
  }
  else if (stick.getButton(" ARIGHT").pressed())
  {
      arduinoPort.write("r");
      color_yellow();
      rect(440,106,81,81);
  }
}

void color_green()
{
  fill(0, 255, 0);
  stroke(0,255,0);
}

void color_yellow()
{
  fill(200,200,0);
  stroke(200,200,0);
}
```

E.9 Fritzing

FRITZING is a cross-platform and open-source program that lets the user convert a circuit design into a direct graphical representation on the computer. The main benefit is that circuits designed in FRITZING look just like the real thing. It provides intuitive, direct and immediately recognizable representaion of a solderless breadboard and the components attached to it (Allan & Bradford, 2013:54). FRITZING was used to create the schematic shown in Figure 5.18 on page 106.

Appendix F

Springs

F.1 Zero free length springs

A spring-balance mechanism, supporting a mass m, is illustrated in Figure F.1a. The mass of link AZ and the spring BC is considered negligible. The area of the triangle ABC is expressed in Equation (F.1.1).

$$\Delta\mathrm{ABC} = \frac{1}{2}a \times (\mathrm{DA} \perp \mathrm{BC}) \tag{F.1.1}$$

Therefore, the perpendicular distance from A to the line BC is DA may be as shown in Equation (F.1.2).

$$(\mathrm{DA} \perp \mathrm{BC}) = \frac{2\Delta\mathrm{ABC}}{a} \tag{F.1.2}$$

However, the area of triangle ABC calculated as in Equation (F.1.3).

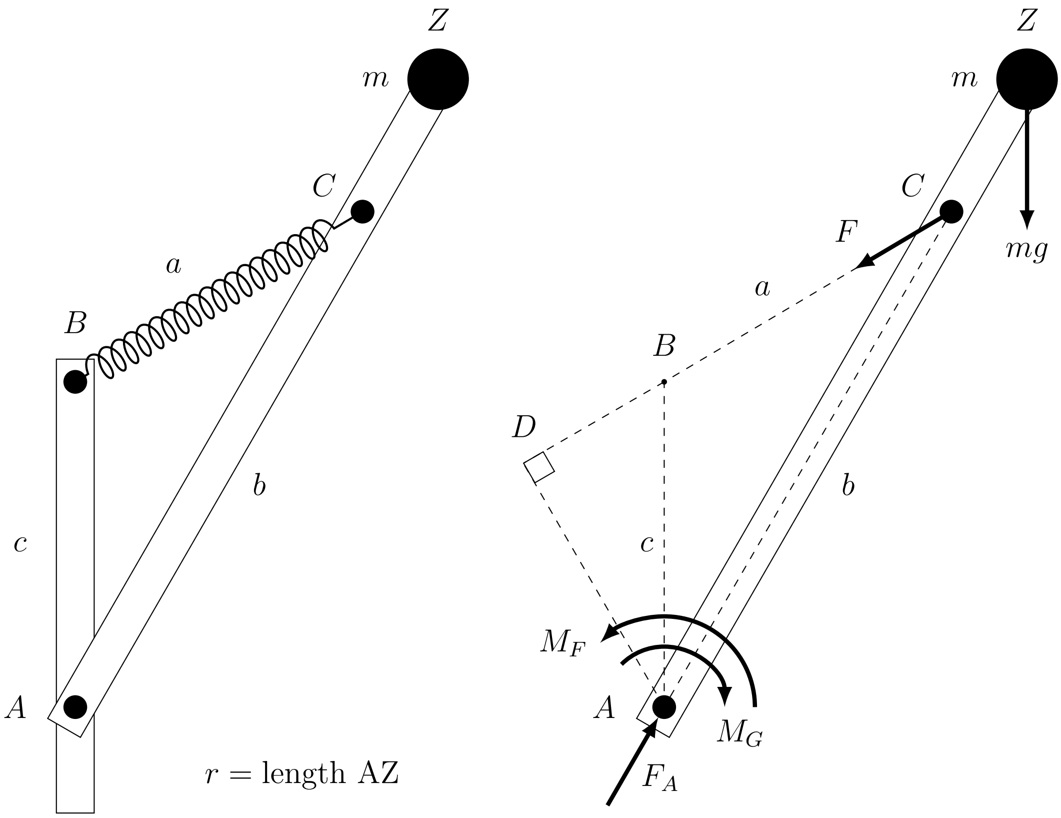
$$\Delta\mathrm{ABC} = \frac{1}{2}bc\sin\theta \tag{F.1.3}$$

Thus, the perpendicular distance (DA $\perp$ BC) may be calculated using Equation (F.1.4).

$$(\mathrm{DA} \perp \mathrm{BC}) = \frac{bc\sin\theta}{a} \tag{F.1.4}$$

Then the moment, M_F of the spring force F about point A is expressed in Equation (F.1.5).

$$M_F = F \times \frac{bc\sin\theta}{a} \tag{F.1.5}$$



(a) Spring-balance mechanism (Carwardine, 1934:503).

(b) Force diagram of spring-balance mechanism.

Figure F.1: Analysis of spring force mechanism.

The moment due to the force of gravity, M_G, is expressed in Equation (F.1.6), where g is the *gravitational acceleration* cause by the gravitational force of the earth.

$$M_G = mgr\sin\theta \tag{F.1.6}$$

To achieve equilibrium, these forces must be equal, as shown in Equation (F.1.7).

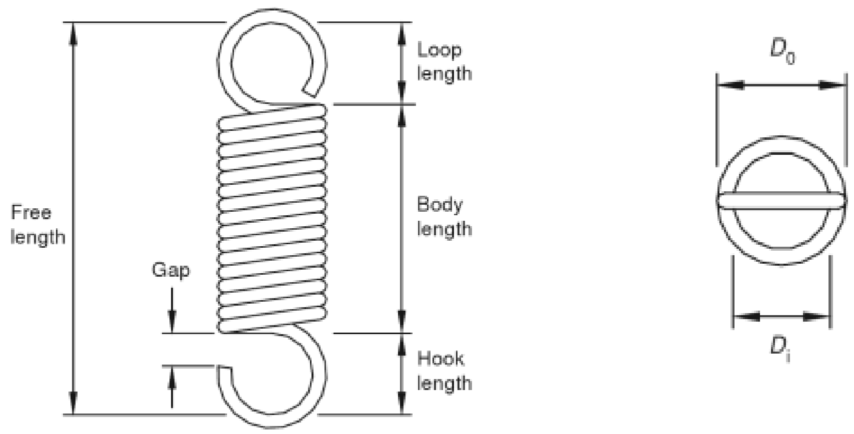
$$\begin{aligned} F \times \frac{bc\sin\theta}{a} &= mgr\sin\theta \\ F &= \frac{mgr}{bc}a \end{aligned} \tag{F.1.7}$$

In Equation (F.1.7), the value $\frac{mgr}{bc}$ is constant and a is a variable that corresponds to the entire length of the spring. Therefore, the force in the spring, F, is proportional to the length of the spring, a (Carwardine, 1934:503). The spring rate, k, is expressed in Equation (F.1.8). To achieve equilibrium in this spring-balance mechanism, a zero length spring[1] is required.

$$k = \frac{mgr}{bc} \tag{F.1.8}$$

F.2 Spring design

The principal dimensions of a helical coil spring are shown in Figure F.2. Helical coil springs are easy to manufacture, cheaper than other springs, are highly reliable and have linear force-extension relationships. Due to these advantages, helical coil springs are popular and extensively used in a number of applications (Bhandari, 2010:394).



(a) Side view. (b) Top view.

Figure F.2: Principal dimensions for a helical extension spring with hook and loop end configurations (Childs, 2003:239).

Helical extension springs are loaded in tension and exert a pulling force. Hooks or loops are incorporated in the spring to facilitate attachment (Childs, 2003:239). These springs are manufactured with an initial force, F_i, that needs to be overcome by an external force, F, before the coils of the spring start to separate. The spring properties, and their equations, for helical extension springs are listed in Table F.1, where N_b refers to the number of active body coils and G is a property of the spring material[2], known as the *Shear Modulus*.

Solver was used to calculate the spring properties that result in the smallest uncorrected torsional stress. Solver uses iterative numerical methods to calculate target values, within constraints defined by the user. The variables

[1] For more information, see LaCoste (1988),Harding & Shepherd (2011) and (Carwardine, 1934).

[2] Refer to Bansal (2010:6) for more information.

and constraints used in SOLVER are listed in Table F.2. For more information, refer to Appendix E.2. The resulting spring properties are listed in Table F.3.

Table F.1: Equations for spring calculation (Childs, 2003:239).

Property	**Symbol**	**Equation**
Mean diameter	D	$\frac{D_0+D_i}{2}$
Wire diameter	d	$\frac{D_0-D_i}{2}$
Outer diameter	D_0	$D+d$
Inner diameter	D_i	$D-d$
Free length	l_0	$(2C-1-N_b)d$
Spring rate	k	$\frac{Gd^4}{8N_bD^3}$
Extension	x	$k(F-F_i)$
Uncorrected tortional stress	τ_{uncorr}	$\frac{8F_iD}{\pi d^3}$

Table F.2: Solver variables parameters for spring optimization.

Variable	**Symbol**	**Constraint**
Uncorrected torsional stress	τ_{uncorr}	Minimize
Spring index	C	$4 \leq C \leq 12$
Free length	l_0	$l_0 < r - c$
Mean diameter	D	
Turns	N_b	
Initial force	F_i	

Table F.3: Spring properties.

Spring Property	**Symbol**	**Value**
Spring rate	k	83.311 N m^{-1}
Mean diameter	D	10.8 mm
Wire diameter	d	0.9 mm
Spring index	C	12
Number of coils	N	64.5
Free length	L_0	78.7 mm
Initial force	F_i	6.5 N
Uncorrected torsional stress	τ	244.4 MPa

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