Non-Intrusive Audible Quick Response Code for media application

by

Neethling McGrath

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Department of Music
University of Stellenbosch
Private Bag X1, 7602 Matieland, South Africa

Supervisor: Mr. G.W. Roux

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Abstract

Non-Intrusive Audible Quick Response Code for media application

N. McGrath

Department of Music
University of Stellenbosch
Private Bag X1, 7602 Matieland, South Africa

Thesis: MPhil Music Technology

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This study details the design of an *audible QR Code*. The audible QR Code seeks to increase the rate of information exchanged in a human interaction and increase the quality of communication by altering the delivery medium of a QR-Code. The delivery medium is changed from a two dimensional bar code to a audio signal. The design is discussed in detail grounded on telecommunication theory including signal modulation, packet construction and error correction. Three techniques were tested in order to reduce the possible intrusive characteristics of the audible QR Code on media content. The commercial uses of the audible QR Code is discussed as well as possible competitions and the advantages it has over the traditional visual QR Code.

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Uittreksel

Kort Afrikaans Vertaling

N. McGrath

Departement Musiek
Universiteit van Stellenbosch
Privaatsak X1, Matieland, 7602, Suid-Afrika

Tesis: MPhil Musiektegnologie

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Hierdie studie verduidelik die ontwerp van 'n *Hoorbare QR-Kode*. Die Hoorbare QR-Kode se doel is om die hoeveelheid inligting wat verruil word gedurende menslike interaksies te verhoog en ook om die kwaliteit van die kommunikasie te verbeter deur die afleweringsmedium van 'n QR-kode te verander. Die medium word verander van n visuele tweedimensionele streepkode na 'n klanksein. Die ontwerp word in diepte bespreek gegrond op die telekommunikasie teorie wat seinmodulasie, datapakkonstruksie en foutkorreksie insluit. Daar word drie tegnieke getoets wat ontwerp is om die indringende eienskappe van die Hoorbare QR-Kode te verminder. Die kommersiële gebruike van die Hoorbare QR-Kode word bespreek, so ook moontlike kompetisie en die voordele wat die Hoorbare QR-kode besit bo die tradisionele visuele QR-kode.

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Chapter 1

Introduction

There is a number of different mediums used to communicate information between people. However, the way something is said is just as important as what is said. Communication mediums range from something as impersonal as poking someone on *Facebook* to a face to face conversation. Some of the interpersonal nuances is lost in a conversation that is aided by technology such as messaging or even voice calls. These nuances in the interaction are diluted when communicating through technology and has become important as most conversations in modern times is assisted by technology. There is however no denying the efficiency, speed and ease of use technology presents. The audible QR Code provides the ability to use all of the advantages of modern day technology to compliment traditional ways of communicating. It enables people that are communicating verbally to connect to an electronic platform. It also allows people to connect to objects in a more intuitive way.

The visual QR Code has become a popular way of connecting a user to an electronic platform from the real world. A QR Code on the box of a product is used to link to additional services and extra information by linking the user to the products website. The QR Code can also link users to a one-time voucher in order to allow access to a event such as a film or festivals. This can be extended to numerous other uses depending on the requirement. The QR Code format allows the data to be adapted to the users requirement making it very flexible in use.

The audible QR Code transmits a short sound signal that contains a packet of data. The audible QR Code can be seen as a method of transmitting a QR Code audible rather than visually. The medium of differs between the audible QR Code and the visual QR Code in order to function on different working environments. The audible QR Code can use different modulation techniques and frequency bandwidths in order to transmit the data. This will be adapted according to the requirements of the users. The thesis focuses on the design and development of a audible QR Code as well as testing three different techniques in order to reduce the intrusiveness of the code on media content. A prototype will be designed theoretically, combined with a transmitter that will

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be programmed in Matlab. The transmitter generates the audible QR Code that is used in the testing for the non-intrusive techniques. A company called *TargetShout* introduced a product that allows users to record a small clip of audio and link it to friends over social media websites through the use of a QR Code. This product links a user to a recording when a QR Code is scanned. This is not the same as a audible QR Code, however, it does link a user to unconventional content (TargetShout, 2014:np).

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Chapter 2

QR Code

Throughout history civilization has sought for a method of communicating using visual representations. From early cave drawings to Egyptian hieroglyphs, every civilization used symbols to represent something of meaning. A Quick Response Code (QR Code) can compared to these ancient methods of condensing an entire story into one picture. This truly validates the saying "a picture paints a thousand words" (Davies, 1990:1-10).

2.1 The QR Code

QR Codes stem from a long history of technologies that are used to solve one problem, communicating faster. Society needed a way of communicating a large amount of data to people in a way that is efficient, fast and universally accessible. A person is easily able to translate a well drawn picture into a story. So too can a person understand the meaning of road traffic signs and translate it into action. This is however a very small amount of data and also where QR Codes finds its place.

The QR Code found its roots in bar codes. bar codes were developed by General Telephone Electronics to be used to track different steel parts of trains and vehicles in the 1960s. These one-dimensional symbols with black and white strips (fig 2.1) would represent a tracking number and placed on parts to assist in the identification (Neubauer, 2012:np).

Figure 2.1: Bar-Code (Denso, 2011:2)

In the 1960s bar codes were being used in supermarkets to check out items and display the price. A Japanese company named Denso Wave that man-

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ufactured one-dimensional Bar Code scanners started investigating ways of adding more information to this symbol. Denso Wave ended up creating a two-dimensional Bar Code that could contain significantly more data than the original one dimensional Bar-Code (fig 2.2). They called it a Quick Response Code and still owns the naming rights and trademark today (Denso, 2011:1-9).

Denso Wave released the QR Code for public use in 1994. The main focus of the symbol is to be read as quickly as possible while maintaining a large storage capacity. The QR code is patented[1] and trademarked[2] by Denso Wave and is free to use. The QR Code is also defined as an ISO standard (ISO/IEC18004:1-113). The entire QR Code usage license can be found in Appendix A. The automotive industry immediately decided to implement it into the manufacturing lines to track parts just as one dimensional bar codes were used. Denso Wave still possess the patent rights to QR Codes however ask no royalty or usage fee as their main goal was to ensure high usage of the codes.

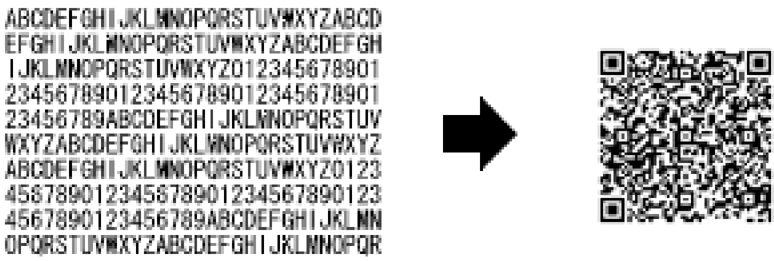

Figure 2.2: QR Code Data Representation (Denso, 2011:6)

In 2002 marketing firms started experimenting with the idea of using QR Codes in advertisements. This required the target audience to respond to the advert rather than having it forced onto them. A QR Code would be placed on an advert and the audience would scan the code in order to interact with the advertisement. This fits the "pull" strategy in marketing, which is considered more effective, rather than the "push" strategy that would force information onto the audience.

The popularity of the QR Code [3] rose sharply after the widespread scanning capabilities of QR Codes by mobile phones were implemented. This gave users

[1]DENSO WAVE has waived the rights to a patent in its possession (Patent No. 2938338) for standardized QR Codes only.

[2]The word QR Code itself is a registered trademark of Denso Wave Incorporated. In the UK, the trademark is registered as E921775, the word "QR Code", with a filing date of 03/09/1998.

[3]There are various different types of symbols that serve the same purpose as QR Codes. This includes Symbol Technologies PDF417, DataMatrix RVSI Acuite CiMatrix and MaxiCode UPS. It is not meaningful to discuss all these different symbols as they all function in very similar ways.

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a tool that could translate the complex array of black and white dots into a meaningful piece of information by using a device that they could be carried around daily.

2.2 Technical

QR Codes are two-dimensional bar code images. Information is encoded into the two-dimensional matrix sequence through a generation algorithm. A scanning device such as a smart phone with a built-in camera is used to read the code. An application on the phone then decodes the information and presents it to the user (Soon, 2008:59-78). Conventional bar codes are only able to store approximately 20 digits. This was a major drawback and the main reason why QR Codes were developed.

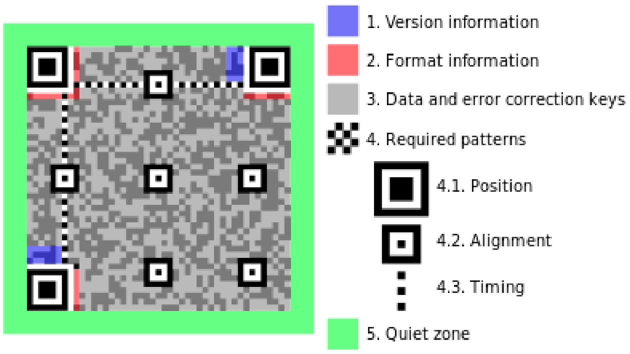

Figure 2.3: QR Code Positioning Data (Denso, 2011:4)

The QR Code is divided into sections, each with a specific function. There are three distinct markers on three of the four edges of the code (fig 2.3). This ensures that the code is correctly oriented by the scanning device and ensures 360° readability (Walsh, 2009:7-8). A quite zone around the code is also required in order to avoid interference. The remaining space in the code is divided into blocks of 8 bits. The first block will contain the level of error correction used. This can either be low, medium or high. The next block contains the length of the data. This is used to ensure that the correct amount of data is read. The data blocks follow the two protocol blocks. These blocks will contain the physical data that will be scanned. The last part of the matrix is filled with the error correction code data blocks. The error correction data blocks ensures that data can be extracted even if there was interference in the scanning process, damage or obscurity of the image (Technology, 2014:np).

Distortion is the main drawback that the QR Code suffers from. bar codes suffer from this to a lesser degree due to it being less complex to read. The code cannot be read if crucial pieces are missing. If the three vital positioning blocks are distorted or obscured then the entire code will be read incorrectly. If the

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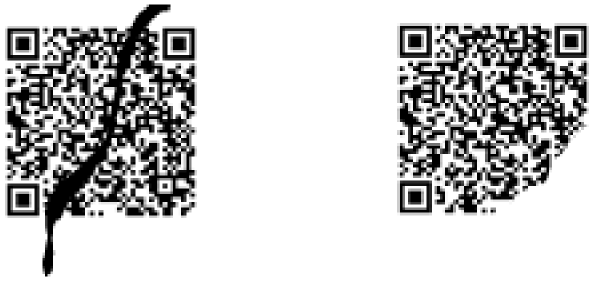

Figure 2.4: Damaged or Obscured QR Code (Denso, 2011:7)

"Quite Zone" (fig 2.5) around the code is not completely void of interference then no data can be extracted. The code can also contain no overlay text or picture as this renders the code unreadable (fig 2.4). QR Codes also suffer from inherent drawbacks. These include the fact that a reader needs close proximity to scan the code. The image quality to interference from noise or bad quality must be sufficient in order to decode the data. It also requires a operator for the reader. It cannot automatically read the code. The combination of these two drawbacks renders the visual QR Code as a method of broadcasting data.

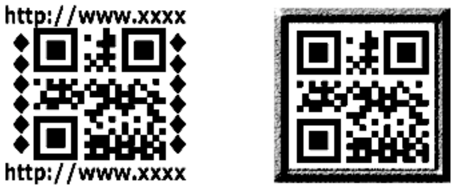

Figure 2.5: Quite Zone Interference (Denso, 2011:7)

QR codes uses Reed-Solomon error detection techniques [4] that utilize an 8 bit codeword (fig 2.6). The codeword is generated as the coefficients of polynomials that represent a check. The codeword is sent together with the message. Once the data is scanned a new codeword is generated in the same manner and checked against the transmitted codeword. If the codewords match the message was sent error-free. This technique has the ability to identify the position of the error if the codewords do not match. Up to 30% of the data can be recovered depending on the error correction code level used (ISO/IEC, 2000:33).

There are 40 different versions of the Denso Wave QR Code. Each ranging in module size from the other versions (fig 2.7). Version 1 is set at 21x21 modules where version 40 is set at 177x177. Currently these are the set of standard QR Code sizes that have to be used under the patent agreement from Denso Wave. These standards ensure that all QR Code scanners are capable of reading any QR Code. Version 40 has maximum capacity of 4,296

[4]Error correction techniques will be explained in further depth in Chapter 3

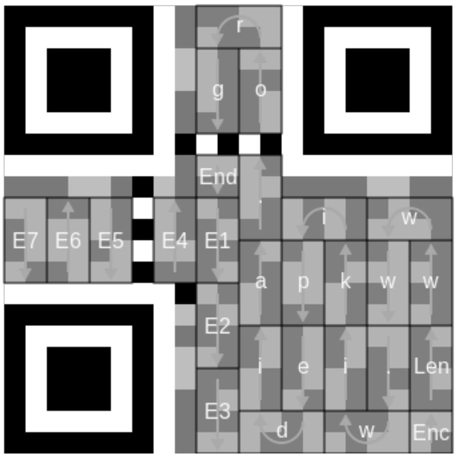

Figure 2.6: QR Code Characteristics (ISO/IEC, 2000:np)

characters (Denso, 2010). There are other matrix codes that use very similar techniques to QR Codes. It is, however, easy to view all the matrix codes under one term as they all achieve the same results by using slightly different techniques (Furht, 2011:341). As the needs of QR Codes evolved the need for more sophisticated types of codes arose. The Micro QR Code version was designed to have a very small footprint. This version would be utilized when space surface area was especially limited.

		QR Code	PDF417	DataMatrix	MaxiCode
Developer		DENSO Wave	Symbol Technologies	RVSI Acuity CiMatrix	UPS
Type		Matrix	Stacked barcode	Matrix	Matrix
Data capacity	Numeric	7,089	2,710	3,116	138
	Alphanumeric	4,296	1,850	2,355	93
	Binary	2,953	1,018	1,556	-
	Japanese, Chinese or Korean characters	1,817	554	778	-
Main features		Large capacity, small size, high-speed scanning	Large capacity	Small size	High-speed scanning
Main applications		All categories	Office automation	Factory automation	Logistics
Standards		AIM, JIS, ISO	AIM, ISO	AIM, ISO	AIM, ISO

Figure 2.7: QR Code and similar codes capacity (Denso, 2011:3)

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2.3 Practicality

Initially bar codes were used to speed up the process of checking out goods at supermarkets and convenience stores (Denso, 2010:np). Bar codes also found uses in tracking parts of supply and assembly lines. Once QR Codes were invented this was a natural substitute to replace Bar-Code usage. This was the case in the early phases of the invention (Soon, 2008:59-78). However marketing firms started realizing the potential of the code and started using it in print advertisements. This led to the most ground breaking use of QR Codes and the reason why it is a feasible technology today.

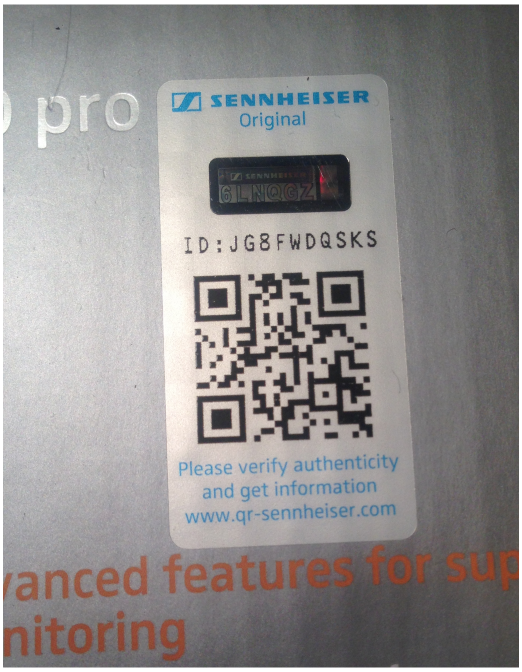

Figure 2.8: QR Code in use

Advertisements must communicate as much information in a small medium as possible due to budget constraints. QR Codes enable marketers to embed a symbol that links a user to more content (fig 2.8) than is being presented on a billboard or newspaper ad. This tool is able to communicate more information in a shorter time than ever before. QR Codes naturally appeal to mobile phone users due to data input method. It is tedious and frustrating entering a large uniform resource locator (URL) into a phone that only contains twelve keys. A user is able to save a significant amount of time by scanning compressed data in the form of an image with a tool one carries around everyday. It is more intuitive to take a picture of a QR Code than to type. Plain text is encoded into the QR Code as binary data. The text used is called special strings that is recognized by the QR Code scanner. The scanner reacts differently to each of the unique tags used. The URL tag usually follows the format of `http://` and

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cause the scanner to open the website. Email address tags are in the format `mailto:` and opens an new email to the address in the code. Telephone number tags are encoded as `tel:` and creates a new contact with the number in the code. There are various other tags including Short Message Service (SMS), MMS, Geographic information, Wi-Fi configuration and more.

The ability to condense information is appealing to marketers and is currently one of the main uses of QR Codes. QR Codes are present in most print ads and billboards. Food Vouchers are sent via Multi Media Messages (MMS) to people on welfare in Japan (Rekimoto *&* Ayatsuka, 2000:1-10). They then scan the code in order to collect their food. This same system is used to validate various tickets bought on the internet. Users will scan a printout or their phone display of the QR Code. Promotion agencies started using QR Codes in interactive competitions. People scan different images and win a prize. QR Codes are also used in amusement parks for directions and links to maps. Sales personal wear a shirt with a QR Code and once it is scanned it links a user to a black email with his email address already entered. This same method is also used to share contact details. McDonald's placed a QR Code on the packaging that link a user to the nutritional value of the meal. This method was also used to show the carbon footprint of certain products. Japan has also started using an encoded version on their passports and the Netherlands designed a coin with a QR Code embedded (Cliffano, 2009:np) (Rekimoto *&* Ayatsuka, 2000:1-10). QR Codes use also extends to security. GOOGLE invested in a company called SLICKLOGIN that uses QR Codes to authenticate users. A QR Code is presented at the login screen and requires the user to scan the symbol with a mobile phone. The mobile phone will open a URL and log the user onto Gmail (Naraine, 2012:np) (SlickLogin, 2014:np).

2.4 Audible QR Code

The term audible QR Code does not exist in modern technology. However, this term describes the combination of two technology fields, Audible Broadcasting and QR Codes. The purpose of the technology remains exactly the same as QR Codes. It is a code that communicates a set of data through a medium. QR Codes use visual symbols as a medium to transfer data. The audible QR Code will use sound waves as a medium to broadcast the data to any reading device in range. A broadcasting device will act as a QR Code generator (fig 2.9). This generator will encode the special character string into a audible code clip. The string will contain various tags just as visual QR Codes. The audible code clip would be analogue to the QR Code symbol. Consequently, the clip would be broadcast over loudspeakers. The audience will be able to receive the code by using a reading device. The device will require a microphone and software in order to decode the clip and extract the special string. This model of communicating the special string through sound rather than visual symbols allows for more flexibility. Firstly, it eliminates the camera requirement of the

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scanning device. No line-of-sight issues or very near proximity requirements. Also a QR Code can now be broadcast to a large audience. It does not require the audience to actively participate in the scanning process. A reading device can be left on standby and push messages to users as they are transmitted from the broadcasting device. The company SlickLogic uses a type of audible QR Code to play a short clip of audio to unlock devices. This company was bought by Google in 2014. TargetShout use a QR Code to link to a short clip of audio that is uploaded after the user records a message (TargetShout, 2014:np). Chirp is an iOS application that plays an audible code from a user's phone. The code contains a link to a photo or a short string of text (AnimalSystems, 2012:np). Acoustic Bar codes use a physical Bar code cut into plastic to transmit a code. The user swipes a mobile phone across the code and acoustic bumps in noise is decoded into a message (Harrison *et al.*, 2012:563-568).

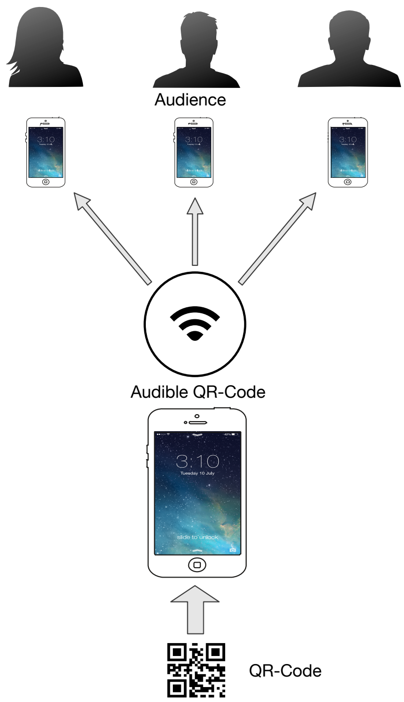

Figure 2.9: audible QR Code Broadcasting Model

Just as the visual QR Code symbol requires key positioning points so too will the audio clip require a protocol in order to ensure faultless transfer of the code. The audible QR Code will require an initial sequence to notify receiving devices that a code will be transmitted soon. The reading devices will recognise the alert signal and start listening for the code. The Audible code will also require protocol data to be transmitted. This protocol data will inform the receiver how to decode the data. This includes the length and error correction code. The audible QR Code is more adapt for certain uses than the visual QR Code. The audible QR Code will be naturally better at

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broadcasting information to an audience. Examples of this could be the score at sporting events, train schedule or air plane boarding times. audible QR Code could also assist in the paying of bills at retail stores.

2.5 QR Codes vs Augmented Reality

An extension of virtual reality (Augmented Reality) can be defined as "a technology that superimposes a computer-generated image on a user's view of the real world, thus providing a composite view" (Oxford, 2014:np).

Augmented Reality superimposes a virtual world over the real world. The virtual world is computer-generated and allows for additional information to be added to a real-world experience (Azuma *et al.*, 2001:1; Azuma, 1993:1). Augmented Reality is meant to assist real world interaction through using computer-generated media (Rekimoto *&* Ayatsuka, 2000:1-10). Linking the virtual world with the real world can be done by using the virtual QR Code on items (Liu *et al.*, 2010:40-43; Kan *et al.*, 2011:339-354). A QR Code can link specific items to virtual information and functions by accessing the resources over the internet. The QR Code, however, is not as immersive as the technologies specifically design for the use in Augmented Reality, such as the Oculus Rift[5]. However it is far easier to implement and has a immediate practical application (Furht, 2011:341).

Augmented Reality has the ability to link users graphically to information in a virtual world. Examples of this includes superimposing a three-dimensional graphic of this virtual world on top of mobile phones camera viewfinder (Azuma *et al.*, 1997:356). This way an interactive environment can be created that extends the natural world. The main problem with Augmented Reality is the ability to recognise items in order to link it through the virtual world to the real world. If a mobile phone is capable of recognising items in the cameras viewfinder it would greatly increase the immersion experience and relevance of information that is superimposed (Chen *et al.*, 2009:181-182). This could mean that users aim their mobile phone over a selection of different foods and have the nutritional content displayed of each item. The hardest part is identifying the item. This is where a QR Code has been able to cross the divide between the real and virtual world in a unique way (Choudary *et al.*, 2009:1023-1024). A QR Code on items can be read and linked to an item in the virtual world. The additional information can be displayed for the specific item without any user interaction. This would complete the Augmented Reality experience. Therefore, if the audible QR Code is included in Augmented Reality, the possibilities are vast and truly impressive.

[5]Additional information on the Oculus Rift can be found at : http://www.oculus.com

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Chapter 3

Communication techniques

The design and background information regarding communication systems is required to follow the development of a prototype in Chapter 6. Modern day communication systems started as wired telegraphy where basic Morse Codes (a sequences of long and short switches of signal) were transmitted over wires. The next phase of innovation came in the form of amplitude modulation (AM) which provided wireless radio services in the 1920s. Frequency modulation (FM) followed in the late 1930s and allowed for more radio channels and smaller receiver radios. The next phase of development was the introduction of mobile phone technology in the 1980s (Benedetto, 1999:1-5). This introduced a communication system that was highly efficient and very versatile. There could be numerous receivers in one cell while maintaining a sufficient signal integrity. This meant that a vast number of devices could communicate at once on the same frequency spectrum in the same area. The technology has since shifted from analogue to digital signals, thereby increasing the capabilities and potential. Current fourth generation mobile communication devices have the potential to receive over 100 Megabits per second and 60 Megabits per second upload speed (Sari *et al.*, 1995:100-109; Stüber, 2011:12). These technologies are achieved through the use of different modulation schemes and error coding systems (Rappaport *et al.*, 1996:255; Saunders & Aragón-Zavala, 2007:90-98).

A basic communication system in its simplest form consists of a transmitter, receiver and a channel (fig 3.1). The channel can represent a wireless communication frequency band or a physical wire whether it is copper or optical fibre.

The transmitter is usually an antenna in the case of a wireless source or a signal generator in a wired source (Chandler, 1994:np). The signal that is transmitted can be altered in many ways to achieve different transmission characteristics. These behaviours range from adjusting the energy required to transmit the signal, different error and distortion prevention techniques and changing the frequency band the signal occupies (Prasad, 2003:1-5). The receiver consists of a demodulation stage in order to reconstruct the original baseband signal, filters and error correcting modules.

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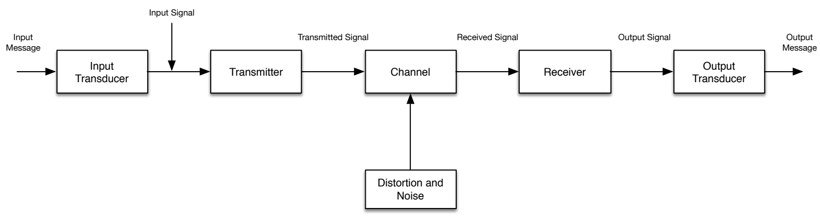

Figure 3.1: Modern day communication system overview

3.1 Modulation techniques

Modulation refers to the process of varying properties of a signal with a modulation signal in order to achieve a desired result. It is possible to transmit a baseband signal directly in its basic form. However, it is usually more effective to modulate it to achieve certain criteria. The modulation process allows the control of the bandwidth in the frequency domain of the signal. This in turn allows separate channels to be allocated to certain uses and control over the radiated signal frequency spectrum. The most general way of modulating a signal involves changing the amplitude, frequency or phase of a sinusoidal linearly (Smith, 2004*a*:363-364). There are various other more complex techniques that exist. Most modern day wireless communication systems use digital modulation techniques; however analogue modulation techniques are far from obsolete. The information contained in this section will provide background to the prototype design and the decision-making process.

Carrier communication (fig 3.2) is known as the use of modulation to shift the frequency content of a message signal. Analogue modulation techniques will vary one parameter of the carrier signal linearly. Another class of modulation is known as pulse modulation. The message signals are baseband digital signals. The original analogue signal is represented in either the pulse width (Pulse Width Modulation) or pulse position (Pulse Position Modulation) (Gibson, 2012:23-90).

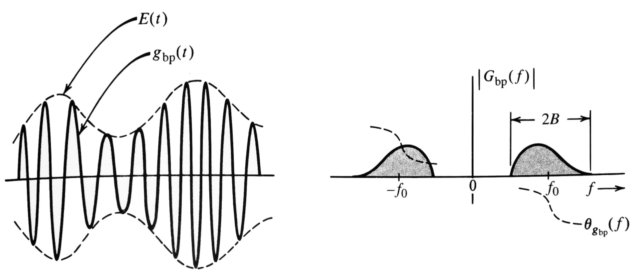

Figure 3.2: Modulation of a signal in the time and frequency domain (Lathi, 1990:143)

Baseband communication refers to the message signal in its original fre-

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quency content form (Personick, 1973:1175-1176). An example of this would be a signal of human speech that would have a frequency spectrum between 500 Hz and 5 KHz. There is no modification or alterations on the baseband signal. Message content restricts the use of wireless baseband communication (Chan & Li, 1990:np). There are large power losses on baseband signal transmissions through air. Therefore the problem of frequency spectrum bandwidth become important. There are many wireless communication transmitters that want to use the same area to transmit their messages. This is not possible if every transmitter uses the same frequency spectrum. There will be too much noise and cross talking between transmissions that none will function correctly. Baseband transmission is however very popular when using dedicated channel cables such as copper or fibre. The problem is solved by assigning specific frequency bands for each channel. This will enable transmission at the same time. Modulation is used to shift the frequency of the baseband signal to the appropriate frequency band (Saunders & Aragón-Zavala, 2007:90-98).

3.1.1 Analogue Modulation Techniques

Analogue signals are continuous without any quantizing or sampling. Quantizing refers to the process of making the voltage level of the signal discrete in order to represent it digitally. Sampling refers to the values of the voltage of the signal in discrete time steps (see chapter 6). These two processes are needed in order to process the signal digitally. This means that the properties that are varied in order to modulate the signal is in the most basic forms the amplitude, phase and frequency (Das, 2010:111).

3.1.1.1 Amplitude Modulation

Amplitude modulation alters the original baseband signal by varying the amplitude of the message signal to the carrier signal (fig 3.3). This is also known as Double Side-band Suppressed Carrier Modulation (DSB-SC) and can easily be understood if the frequency spectrum is analysed. Double side-band refers to the baseband signal being shifted onto the impulse at f_c which is the carrier frequency.The double side-band is due to the baseband signal having a frequency spectrum that is mirrored around the amplitude axis. The carrier is suppressed due to the message signal not having direct current content meaning there is no amplitude in the frequency spectrum at $f = 0$ (Lathi, 1990:140).

$$\phi_{AM} = m * \cos(2\pi f_c t)$$

The baseband signal is shifted onto the carrier frequency f_c. This is achieved by the convolution of the two signals together in time. Thus, the original message content is found in the varying of the amplitude of the carrier. The modulated signal bandwidth remains the same as the message signal

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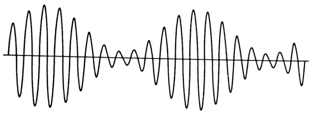

Figure 3.3: Amplitude Modulation Waveform (Lathi, 1990:12)

but is now shifted in the frequency spectrum. There are some restriction on the demodulation due to the nature of the modulation technique used. The phase and frequency of the carrier must be known by the demodulator in order to be able to demodulate a AM signal coherently. There are however two ways of demodulating asynchronously (Bakshi, 2009*b*:16-47). The first method required the signal to be rectified in order to obtain the message data (fig 3.4). The second method uses a diode and envelope detection (fig 3.5).

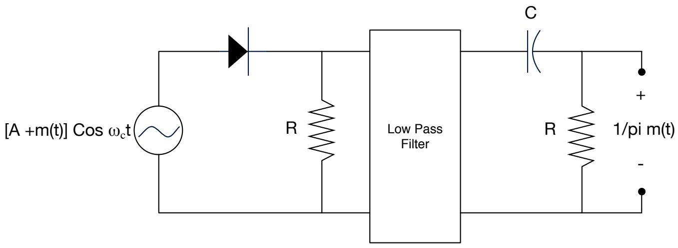

Figure 3.4: AM Demodulation through use of Rectifier

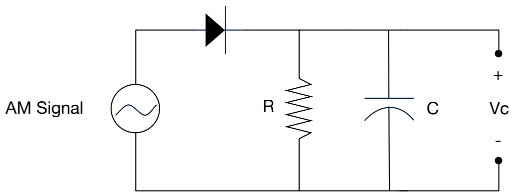

Figure 3.5: AM demodulation using envelope detection

Both asynchronous methods require that the modulation index is above or equal to one (Bakshi, 2009*a*:3-4). The modulation index is defined as m_p (message signal peak amplitude) divided by A (carrier peak amplitude) (Lathi, 1990:153). A modulation index above one is required due to the way the rectifier demodulates the signal. If the modulated signal crosses the time axis in amplitude then there will be an error (fig 3.6). The rectifier is only able to follow the curve of the modulated signal if it remains positive due to the nature of the amplitude modulation. The fact that the original message content can be extracted from the silhouette of the modulated signal supports this observation.

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$$\mu = \frac{m_p}{A}$$

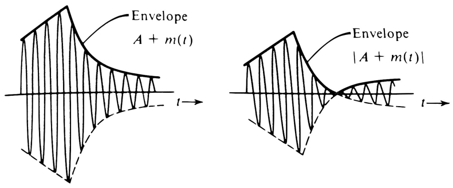

Figure 3.6: Amplitude modulation requirements (Lathi, 1990:157)

The bandwidth of the original message signal is B Hz. The bandwidth of the modulated signal will be $2 \times B$ Hz due to the double side-band nature of the modulation. The modulated signal will require twice the bandwidth to transmit than the baseband signal.

3.1.1.2 Frequency Modulation

Frequency and phase modulation are closely related due to a similar technique being used. Angle modulation encapsulates all frequency modulation and phase modulation techniques. Frequency modulation (FM) varies the frequency of the carrier signal to the amplitude of the message signal (fig 3.7). The message content is embedded into the chance of frequency over time of the modulated signal. An arbitrary constant called the "frequency modulation constant" k_f is used scale the modulation amount (Lathi, 1990:204).

$$\phi_{FM} = A * \cos[\omega_c t + k_f \int m(\alpha) d\alpha]$$

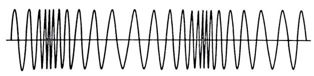

Figure 3.7: Frequency Modulation Waveform (Lathi, 1990:12)

Frequency Modulation can be demodulated incoherently or synchronously. Synchronization between the modulator carrier sinusoidal and the demodulator is required in order to detect a difference in phase so that the original message content can be extracted. This can be done through the application of a Phased-locked loop (PLL). This device synchronizes a slave demodulator to the master modulators carrier frequency and phase. The message content is contained in the frequency of the modulated signal. The demodulating process will isolate the frequency components by differentiating the signal and using an envelope detector on the amplitude (Xiong, 2000:np).

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$$\frac{d}{dt}(A * \cos[\omega_c t + k_f \int m(\alpha)d\alpha])$$

$$= [\omega_c t + k_f m(t)] * \sin[\omega_c t + k_f \int m(\alpha)d\alpha]$$

Frequency modulated signals can also be demodulated synchronously through the use of a PPL. The PPL is used to synchronize the local oscillator to the carrier frequency and phase. Once this is achieved the two modulated signal and local oscillator is mixed together in order to demodulate the signal. The Superheterodyne analogue FM receiver is an example of a system that uses the PLL to demodulate an FM signal (Lathi, 1990:239).

3.1.1.3 Phase Modulation

Phase modulation (PM) modulates the phase of the carrier signal to the message signal amplitude. The difference of the modulated signal phase as a function of time results in the original message content. An arbitrary constant called the "phase modulation constant" k_p is used scale the modulation amount.

$$\phi_{PM} = A * \cos[\omega_c t + k_p m(t)]$$

All the results derived from frequency modulation can be applied directly to phase modulation due to the close relationship they share. Replacing just one part of the FM equation gives us the PM equation (Pozar, 2000:298) (Lathi, 1990:205).

$$m(t) \Leftrightarrow \int m(\alpha)d\alpha$$

In phase modulation the the message signal is directly proportional to $m(t)$. In frequency modulation the modulation signal is directly proportional to the $\int m(\alpha)d\alpha$. Therefore a FM and PM signal are demodulated in the same way. An PM signal is only differentiated after the FM demodulation step (Suksompong, 2008:17-18).

3.1.2 Digital Modulation

Digital signals are signals that have been quantized and sampled. This is required so that computing devices are able to understand the data it represents. An analogue signal requires an infinite amount of data to store every point on the continuous curve. This is not only impossible but impracticable as not all the points are needed. Digital modulation uses digital signals as messages.

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3.1.2.1 Amplitude Shift Keying

In Amplitude Shift Keying (ASK) the carrier signal is turned on or off depending on the baseband digital binary sequence. This technique is also referred to as on-off keying. ASK is very similar to AM and can be demodulated synchronously through a local oscillator but requires the exact phase and frequency as the modulator carrier signal. ASK can also be demodulated asynchronously through the use of an envelope detector due to the envelope varying to binary baseband data (Pozar, 2000:305).

3.1.2.2 Frequency Shift Keying

Frequency Shift Keying (FSK) is achieved by switching the carrier frequency between two or more specific values. The values usually represent the binary baseband symbol representation. Binary only has two symbols so only two values are used, i.e. ω_1,ω_2. These can be chosen on arbitrary frequency values however this usually has to conform to a specific frequency band restriction enforced by government regulation to the application of the technology. FSK can be demodulated coherently through a local oscillator that is synced to the modulation carrier signal. FSK can also be demodulated asynchronously through the use of an envelope detector coupled with band-pass filters on the specific symbol representation band (Pozar, 2000:306-307).

3.1.2.3 Phase Shift Keying

Phase Shift Keying (PSK) alters the phase of the carrier to the baseband binary sequence. If the message signal would be a positive pulse then the phase would be 0° of the modulated signal. If the message signal is a negative pulse, the phase of the modulated signal is ± 180°(Pozar, 2000:307-308).

3.2 Communication Protocol

Data exchange between devices on a network (wireless or wired) is governed by a set of rules and specifications known as communication protocols. These rules and specifications ensure that data is sent and received in a convention that both the transmitter and receiver expect and can interpret. These rules are set up beforehand by being programmed into transmission states. The states are governed by these rules and cannot break them.

These rules and specifications are a set of algorithms that are used to transmit data. A basic system will first establish a connection between two devices. The receiver will indicate to the transmitter that it is ready to receive data. Only when the transmitter received the message that the receiver is ready, will data be transmitted in the sequence that is governed by the protocol known as the data format (Stallings, 1997:12-29).

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The data format is highly customizable depending on the application. The data will be sent in sequence to the receiver. The receiver will transmit an acknowledgement message to the transmitter to indicate that the data has received. The receiver will at this stage check the data according to the specific error correction code of the system.

The receiver will send an error message to the transmitter indicating that an error has occurred on the last transmission and ask the transmitter to send the data block again. If there is no error then the receiver will transmit a message indicating that the data was received and that it is ready to receive more data. Furthermore there are routing rules that govern the route the data packets are allowed to take to the transmitter in complex networks such as the worldwide web. Addressing formats can also be applied to transmission rules the internet protocol is a good example where the address is in the format $x.x.x.x$ where $0 <= x <= 255$ (Fall *&* Stevens, 2011:155-183; O'hara *&* Petrick, 2005:37-40).

This is an example of a basic system however there are many different states and rules that can be applied. There are systems that use packets to ensure that the correct sequence of data blocks are received in vast networks where not all data blocks take the same route to the receiver. Hence, there is also error correction on different stages of the receiving process as well as different handshaking techniques and acknowledgement messages.

Protocol layering is found in any modern communication system. The hardware has a hard-coded protocol that sets the most basic rules physically in circuit design. This cannot be altered once the device has been built. These rules include basic forms of handshaking and parity checks between transmissions and data formatting. Hardware protocols are usually the bottom layer. There are various other layers stacked on top of hardware protocols. Examples of these are software protocols in applications such as *Skype* or *FaceTime* that govern the data once it is received into memory. Another example is Ethernet protocols or Internet protocols that are standards that are used to communicate over a Ethernet or over the worldwide web (Stojmenovic, 2003:602-620).

An easy way to view this is to follow the data path. In a *Skype* conversation the data is received through the Ethernet port of the computer. The physical hardware of the computer has its own protocol that govern the data from memory to the Ethernet module. The data is received over a Ethernet network that has its own protocol that governs communication between different hardware devices. Once the data is received on the device's memory it will be handled by software protocols. Operating systems have protocols to know what data is associated with certain applications. Ports are used to categorize data to different uses. The operating system protocols will assign the data received over the Ethernet connection to the *Skype* application. Once *Skype* receives the data that is specifically formatted according to the rules designed for the application it will process it and interpret the data (Ilyas, 2002:6).

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3.3 Error Correction Code

A real world data transmission channel experiences distortion, noise and external interference that has the ability to render a transmission of data useless. The added noise and interferences can change data symbol values or distort the clock signal causing the listening device to receive data different to what was transmitted. In order to avoid these errors in transmission a redundancy of data needs to be implemented. Redundant data bits need to be transmitted in order to ensure that the original message is received correctly (MacWilliams & Sloane, 1977:1-2).

There are two basic classes of error correcting codes. Block codes and convolution codes. Every block of n bits is encoded into a longer and redundant block of k bits.

$$k > n$$

In convolution codes the codeword is dependant only on the data bits that precede $(N-1)$ bits of the codeword. The encoding is done continuously as data is received as a sequence rather than in blocks. Shannon's work on the capacity of noisy channels yielded the results that the larger the block codeword length the lower the error probability becomes. Also adding that the more complex the codes become the lower the error probability however the longer it takes to decrypt the sequence of data. The goal is set to find a code that has a sufficiently low error probability whilst not increasing the overall data block too significantly. Also keeping the code simple enough to be able to decrypt it quickly (Lathi, 1990:802-803).

One of the most popular error correcting techniques is called cyclic redundancy check (CRC). The technique is designed to identify data errors at the receiver often after basic parity check error correction. Each data block is encoded through the use of CRC in order to be able to check if a data packet is valid. The data packets are converted into CRC blocks of length $n <= 2^m - 1$. m is the number of bits in the data block and common values include 8,12,16 or 32. CRC sets up a syndrome that represents the coefficient values of a polynomial. The coefficient values are determined by the data input block and a long division process. The CRC block is transmitted once the syndrome has been set up. The same process is repeated at the receiver with the received data block. A new syndrome is set up through the long division process explained earlier. The receiver syndrome is compared to the one that was transmitted. If they are not the same then an error has occurred in the transmission. Cyclic codes are very popular due to it being so easy to implement and it can be applied after hardware parity check. CRC are commonly used in mobile communication systems specifically in code division multiple access (CDMA) technologies (Lathi, 1990:813) (Gibson, 2012:91).

Error correction codes are in essence redundant data. Error correcting codes would not be needed in an environment without channel disturbances

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such as distortion and noise. The redundant data is however essential to the transmission because without it then no error could be detected or partially fixed. This could lead to absolute chaos due to the receiver not being able to validate the data received. There is a vast number of techniques that are used to detect errors on data transmissions. All techniques focus on the same two principles outlined by Shannon (Shannon, 1949:10-21).

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Chapter 4

Similar technologies and uses

Audible QR Code's method of delivering data is unique, the application however, is not without competition. The audible QR Code is designed to compete against technologies that are very old such as infra-red and the most recent developments such as Bluetooth and Near-Field Communication. The advantages and drawbacks of these technologies are investigated and compared against the audible QR Code.

4.1 Bluetooth

Bluetooth was the first practical technology that allowed functional data transfer between two mobile devices. Bluetooth succeeded infra-red transmission that was very slow, required line of sight and user intervention. Infra-red was not practical as only a couple of lines of text or very small data packets could be transmitted and could only communicate between two devices. Furthermore, the idea of keeping two phones in line of sight is not very intuitive. Infra-red was used, because there was no better technology at the time and was very inexpensive. Bluetooth is a short-range ad-hoc wireless radio interface that allows mobile devices to connect to a network wirelessly (Prabhu & Reddi, 2004:20-32). Bluetooth uses the 2.4 GHz frequency band (Haartsen, 1998:110-117). Bluetooth eliminated all physical connection requirements not only between mobile phones but also mouses & keyboards, headphones, modems and many other consumer electronics. Bluetooth radio waves are able to propagate through some objects and around corners. It can join in a network of several devices, does not require line of sight or user intervention and has a greater range than infra-red.

4.1.1 Bluetooth technical specifications

Bluetooth was required to be a low cost, energy efficient and small radio transmission device. The system operates in the frequency band between 2.4 GHz

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and 2.45 GHz due to the requirement of a frequency band with no licence fees (Bhagwat, 2001:97). A frequency hopping technique is used to suppress interference due to other devices also operating on the spectrum (Bisdikian *et al.*, 2001:6-7). Devices such as gate openers, two-way radios and baby monitors. Frequency hopping technique will divide the spectrum bandwidth into smaller hopping channels. Each hopping channel is large enough to transmit the entire message. The devices will then hop to different channels in the larger spread spectrum in a pseudo-random order. The interference on the channel will greatly be reduced. It will also require a significant amount of wireless transmission power to totally suppress a Bluetooth communication. If one of the hopping channels are suppressed then error correction on the receiver will still be able to recognise the correct data sent. The hopping order is determined by the master device on the network and the sequence only repeats every 23 hours.

Bluetooth uses Gaussian frequency shift keying (GFSK) using a bandwidth of 1 Mhz. This means the bandwidth of one hopping channel needs to be at least 1 MHz. The hopping channels are statistically equally probable. The pulses are also shaped to occupy a Gaussian spread in the frequency spectrum. A symbol data rate of 1 Mbits/s is achieved (Bray *&* Sturman, 2001:12-13). Bluetooth uses a set packet layout to transmit data. The first 72 bits contain an access code. The access code is unique to the network the device is currently joined to. Every packet sent contains this access code and receiving devices compare this code in order to ensure the packet is meant for them. If it matches the device will interpret the data. If it does not match it will simply ignore it. The access code block is followed by the header block. This 54 bit block contains control data essential to the packet delivery. The packet type, flow control bits, error correction scheme and media access control is contained in the header. The last block is the payload and this is the actual data being transferred. The payload can contain up to 2745 bits (Golmie *et al.*, 2001:32).

Bluetooth uses a combination of authentication and encryption to ensure data security. A hardware implementation of encryption is used that allow mobile devices without significant processing power to decrypt messages. Bluetooth devices are required to pair to one another. This establishes a protected connection between the two devices. This is achieved through the use of three security algorithms (Gehrmann *&* Nyberg, 2001:2-4). A public address which is a pseudo-random code that is transmitted to all devices in range. A private address is generated and only the two communicating devices know. Lastly there is a random number generated periodically and used to authenticate the connection. During transmission challenge responses can be requested from devices to re-authenticate. The session keys for the pairing can also be regenerated at any time. This session key is generated through the private address, public address and random number (Vainio, 2000:5-12).

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4.1.2 Bluetooth Low Energy

The fourth generation of Bluetooth devices is designed to be of low energy consumption (Padgette *et al.*, 2012:5). The design allows that a Bluetooth low energy device can be powered by a coin cell battery (Siekkinen *et al.*, 2012:232-237). Data rates are reduced due to the design being focused on low energy consumption. The device is also less complex and have a lower cost to manufacture. The applications of such devices are vast and range from healthcare and fitness to entertainment systems. These devices will function in a Master-Slave or a broadcasting typography. Gaussian Frequency Shift keying is used to modulate the signal digitally and operates at a bit rate of 1 Mbit/s (Heydon *&* Hunn, 2012:4). The frequency band remains exactly the same as the other generations of Bluetooth at 2.4 GHz. The new Bluetooth generation is specifically designed to excel in advertisement. An advertisement channel will always run parallel to the data channels to ensure the device is always discoverable. This is essential for broadcasting to as many users at the same time as possible (Martelli, 2014:23-27). New devices can join the transmission at any time and no longer have to wait for the other devices to finish data transmission. The same encryption technology is used however these devices are not required to pair as the older generations. There is however extra protocol bits attached to each package that track the device identity through the Media Access Control (MAC) address and random numbers that are generated in order to secure the connection. The connection is as seamless as a Near-Field Communication (NFC) transmission and inherits the same security problems (Gomez *et al.*, 2012:11739-11753). The new generation of Bluetooth devices bridges a gap that previously left consumer electronic designers wanting an energy efficient wireless connection that is secure yet does not require user input to establish a connection. This is essentially NFC but with a range of 50 meters. Doctors are suggesting using these devices for wireless heart rate and blood pressure monitors (Yu *et al.*, 2012:763-767). Athletes could place devices in their shoes to monitor distance travelled during sporting events and training. Entertainment systems want to do away with infra-red thus no longer requiring line of sight on a remote and increasing the response time. The new wave of wearable technology will use these devices to communicate with one another. Your cell phone will alert your watch that you have received a message or notification. All this and more applications are possible but of the most important is Apple's Ibeacons.

4.1.3 IBeacons

Apple was reluctant to invest in NFC while the rest of the major electronic producers adopted it and all its possibilities (Newman, 2014:222-225). Bluetooth was not in a position to compete against NFC due to the pairing requirement of older generations and also power consumption. With the release of Bluetooth 4.0 (Low Energy) Apple announced the product known as IBea-

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cons. An IBeacon is a small wireless device that is placed in specific regions to broadcast data using Bluetooth Low Energy. The data that the beacon broadcasts can be changed and is associated with the beacons location. The Ibeacons data will be displayed on the users phone once the user enters the range of the IBeacon. Proximity is also evaluated meaning more than one beacon can be in range of one another and still be displayed on the users phone. The closest beacon will be prioritized due to its proximity (Gast, 2014:1-9). This is where IBeacons couples with the project. This is essentially a wireless QR Code that is being transmitted through electromagnetic waves to the listener. The beacons started to get popular and widespread during 2014. The uses are focused on advertisement and shopping. Allowing listeners more information about a certain product on a shelf as well as allowing targeted advertisement in large crowds. Another useful application is indoor GPS. GPS is unable to function accurately indoors as the signal from the satellites are too small to determine. This enables users to navigate through a building using the beacons and proximity to them (Aparicio *et al.*, 2008:487-491) (Stojanović *&* Stojanović, 2014:57-72).

4.2 Near Field Communication

Near-Field Communication (NFC) is a set of standards and protocols for electronics devices to establish a connection using electromagnetic waves while maintaining a proximity constraint of no more than a few centimetres. NFC uses Radio frequency identification (RFId) technology to establish a wireless two-way connection between two devices. NFC devices can operate in two modes. The first mode is called active due to the device supply power to the NFC chip. The other mode is called passive. In passive mode energy is drawn from the electromagnetic waves in order to power the NFC chip. Two-direction communication is only possible when both devices are operating in active mode. This is the large advantage NFC has over older technologies such as contactless smart cards. There has to be at least one active device present for data exchange to be possible (SHARMA *et al.*, 2013:342-345).

4.2.1 Technical specifications

RFId is used to govern the data transactions between the two NFC devices. A maximum distance of ten centimetres between devices is specified by ISO/IEC 14443 (ISO/IEC, 2004). NFC operates in the RFId frequency band of 13.56 Mhz. Amplitude shift keying (ASK) is used during active transmission. Two pulse shapes are used depending on the rate of transmission known as the baud rate. If the baud rate is less than 106 kBaud then Miller coding scheme is used. If it exceeds 106 kBaud the Manchester coding scheme is used. The different coding schemes are based on the pulse shapes. Different shapes possess different characteristics. Some shapes use less energy to transmit whilst being more

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susceptible to interference (Lathi, 1990:337). The mathematics are complex and not in the scope of this chapter (Haselsteiner & Breitfuß, 2006:12-14).

The NFC communication protocol requires that an active device initiate the communication. Device A is called the initiator and transmits the first data that establishes a connection. Device A is required to be active due to a passive device inability to transmit electromagnetic waves without first being powered by an external source. Device B is called the target and replies to device A. Device B can be active and passive. There can also be more than one device B meaning a message can be broadcast from device A (Haselsteiner & Breitfuß, 2006:12-14).

4.2.2 Security issues

NFC is very susceptible to a variety of security exploits (Pasquet *et al.*, 2008:121-126). Listening in or eavesdropping into other devices communication is very easy due to the nature of the wireless protocol (Brown & Diakos, 2011:44-47). An attacker requires no special device or technology in order to achieve this. If the attacker is familiar with the data transaction protocol then a person can extract all data from the transmission. The protection NFC offers to this exploit is built into the design. NFC requires close proximity between the devices in order to work. The electromagnetic fields are not strong enough and also not directed into all directions. This means that a potential attacker would need to be very close and in the right orientation to the NFC antenna (Brown *et al.*, 2013:3525-3528) (Kortvedt & Mjolsnes, 2009:57-66). Data corruption is the easiest of the security flaws to exploit. This essentially cripples all NFC communication in the frequency band. This is achieved by transmitting a large amount of electromagnetic power in the frequency band and essentially corrupting all data being send over it. This is very easy to do if one of the devices is passive due to the low power being emitted by the two devices. This technique is not very harmful as no data is essentially stolen but it can be frustrating as it hampers the functionality of the NFC devices. Data manipulation of the other hand is very harmful. This is where specific bits are manipulated during communication. The transmitting device thinks it sent the correct data. The receiving device trust the data being received as being true and sent from the transmitting device. This technique is very hard to achieve (Van Damme *et al.*, 2009:3). The manipulating device needs complete knowledge of the communication protocol and data being transmitted. A specific bit needs to be targeted and altered without the receiver or sender noticing it. This technique is very unlikely to succeed. The "man in the middle attack" is, however, more likely to succeed. This technique deceives the transmitter into thinking the intruder device is the true receiver. The same happens with the true receiver thinking the intruder device is the true transmitter. All data is relayed through the intruder device and data can easily be manipulated. Completely different data can even be transmitted. Most of these attacks are

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avoided by creating a secure connection between these devices. This requires a public and private key to be set up and transmitted. Once the connection is secure all attacking techniques can be defended against (Haselsteiner & Breitfuß, 2006:12-14; Padgette *et al.*, 2012:24-29). NFC devices do not require users to set up pairing or input passwords to establish a connection. This is all done automatically and is done very fast. This is the inherent advantage NFC has over older Bluetooth versions.

4.2.3 Application

4.2.3.1 Contactless tokens

The first practical application of NFC is contactless tokens. A token can be seen as a key. Essentially it is a sequence of numbers that is stored on a NFC capable device being active or passive. This token is read by an active NFC device in order to provide a specific service. This is the essential function of most basic RDId systems (Dmitrienko *et al.*, 2012:1-17). This is a method of giving input to a electronic device that is more intuitive than typing in a password. The sequence of numbers is usually stored on a passive NFC device such as a key card. Popular functions for this application is access cards, RDId or key cards (Andersson, 2014:4). Another function is where the passive token is placed on a product. Users will then scan the token and be redirected to a url of the item. Only basic protocols can be used due to the passive device not being able to decrypt data and is essentially a one-way communication.

4.2.3.2 Mobile payment

The most popular use of NFC is in the field of mobile payments. The idea that credit and debit cards details can be saved onto mobile devices and recalled once a payment needs to be made is impressive. An active mobile device is required and needs to be presented at a vendor that supports NFC payment (Ondrus & Pigneur, 2007:43). Once the payment needs to be made the shopper simply puts the mobile phone in proximity of the reader and all detail will be transferred and the payment will be made. This is not limited to credit and debit cards. Boarding passes, concert ticketing and loyalty cards can be saved onto the device and recalled at an instant (Chen *et al.*, 2011:1-17). Banks have recently started releasing credit cards with a passive NFC chip built in. Therefore you do not require to save it onto your phone.

4.2.3.3 Pairing

Even though it seems NFC and Bluetooth are in competition this is not the case. Most mobile phones will not do away with Bluetooth in favour of NFC. NFC can, however, be used to circumvent the most tedious part of establishing a Bluetooth connection pairing (Padgette *et al.*, 2012:21). If a user wants to

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pair two Bluetooth devices that also possess NFC chips all the user is required to do is bring the two devices into close proximity of one another. The NFC connection will transmit all data required to pair the two Bluetooth devices and set up a secure connection without requiring the user to enter details and passwords.

4.3 Bluetooth vs NFC vs audible QR Code

Bluetooth and Near-Field Communication technologies are established and successful in practice. There are a vast array of different applications for these technologies and is growing popular with users. Audible QR Code technology will attempt to be applied for the same uses as NFC and Bluetooth. Bluetooth Low Energy will become Apple's standard and possibly the rest of the consumer technology world. NFC is at risk of losing to Bluetooth due to the new generation technology. NFC is no longer required as all the functions can be achieved through the use of Bluetooth and you no longer require close proximity to the field reader. This is a major drawback and concern to NFC devices as it renders the technology useless. The only feature that is unique is that a passive NFC device requires no external power source. This cannot be matched by the new Bluetooth generation. This means that the only use in the future of NFC devices will be tokens. NFC will struggle to be able to beat Bluetooth even though contactless tokens could become a large industry. The audible QR Code will struggle to compete against fourth generation Bluetooth devices. That being said the audible QR Code can still thrive due to the nature of the technology. Audible QR Code technology uses components that are standard on all mobile phones the microphone and loudspeakers. New generation Bluetooth and NFC will need to be added to devices, increasing the cost. Older technologies such as radios, entertainment centres and movie theatres will always be able to broadcast a audible QR Code without the need for Bluetooth. This gives the audible QR Code an advantage over the other technologies.

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Chapter 5

Non-intrusive audible sounds

The audible QR Code is required to use audible sound frequencies in order to transmit the message. This could become intrusive to the audience that is experiencing media content. If a QR code is embedded into a song, audio book, film or advertisement the code must not diminish the experience and remain as subtle as possible. This requires the code to remain non-intrusive and implores techniques to hide the code in the audible frequencies. Three techniques are discussed with experiments to test the viability and efficiency. The first suggests using phasing techniques to hide a code in a stereo mix of media content. The second technique investigated the ear and in particular Presbycusis to use a high frequency band for code transmission. The third technique also suggest using a high frequency band however taking it to an extreme of placing the code in the inaudible range above 20 kHz where a very small amount of humans can hear but some electronic equipment still function.

5.1 The ear

The ear is one of the human body's sensory organs. It contains the smallest and hardest bones in the body as well as the smallest muscles. The ear converts energy in the form of a pressure wave into electrical signals that is sent to the brain for processing. Sound waves are longitudinal waves (fig 5.1) meaning that the motion of air molecules are parallel to the direction of energy.

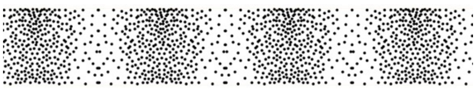

Figure 5.1: Longitudinal Wave

The energy of the wave is contained in the pressure of the air molecules. A high pressure area signals a high energy region while a low energy signals a crest

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and low energy. Sound is thus a pressure wave that propagates mechanical energy in the form of air pressure. The propagation speed of sound differs depending on the propagation medium. Sound moves faster in solid objects such as metal bars. Water also allows for the propagation of sound and it will allow sound to travel faster than in air. Sound propagates at 343 meters per second in air (Bamber, 1986:200-224). The pitch of a sound wave is dependant on the wave frequency. The higher the frequency the higher the perceived pitch (Dallos, 1996:1-43). Middle C on a piano is at 256 cycles per second or Hertz (Brownell, 1997:9). The human ear can sense sound between 20 and 20 kHz. The louder a sound wave the higher its mechanical energy and the higher the air pressure in the wave. The ear consists of three sections called in the outer, middle and inner ear. In all wave theory from electromagnetic to air pressure waves there is always reflection of energy if a wave passes between two different mediums. The outer ear (fig 5.2) is mainly responsible to reduce the amount of energy that is reflected when the air pressure waves pass from the outer to the middle ear. The outer ear is also a crucial part of the hearing process that spreads different frequency components of the sound signal so that the auditory processing units in the brain can distinguish the direction of the sound as well as recognize known frequency patterns of sounds previously heard. This is how our brain is able to recognize different people's voices, sounds of danger and rebuild distorted sounds. The outer ear is also responsible for directing sound into the ear canal to hit the ear drum perpendicular in order to ensure energy efficiency (Hudspeth, 1989:397-404).

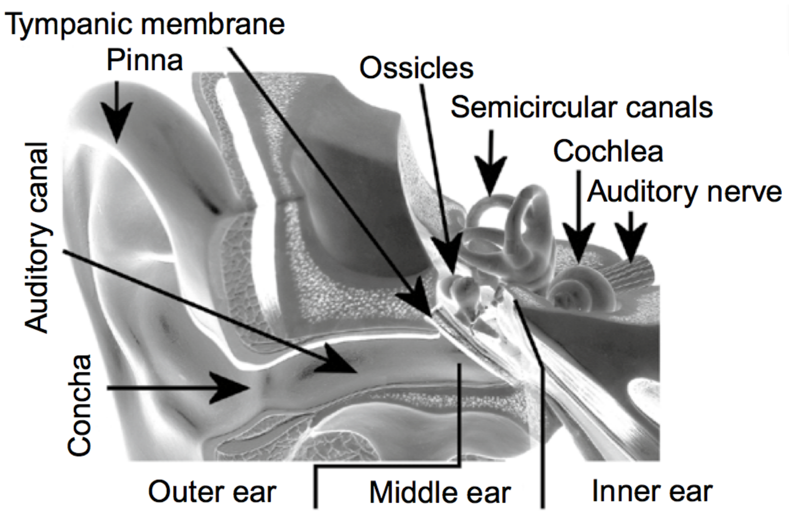

Figure 5.2: The outer ear (Howard & Angus, 2009:79)

The outer ear connects to the middle ear through the ear drum. The middle ear (fig 5.3) is the portion of the ear between the ear drum and the oval window of the inner ear. It is responsible for the transmission of the sound waves from the outer ear to the inner ear. Three *Ossicles* are connected to the ear drum

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and the oval window to transfer the sound waves from the ear canal to the fluid inside the inner ear. The *Ossicles* are three connected bones known as the *Malleus*, *Incus* and *Stapes*. *Ossicles* mean small bones and these three bones are of the smallest in the body. The middle ear is necessary, because air pressure waves reflect almost completely when it moves from air to a liquid. The middle ear uses the *Ossicles* as levers that convert the air pressure into mechanical force. The ear drum vibrates and causes the *Malleus* and *Incus* to move. The *Incus* is connected to the *Stapes* which in turn is connected to the *Fenestra Ovalis* or the inner ear membrane. The membrane is pulled and pushed responding to the movement of the ear drum. This is the only effective way of transferring the energy from the ear canal to the inner ear. The middle ear is connected to the nasal cavity through the *Eustachian* tube that allows the pressure between the two cavities to be normalized. This is essential in order to prevent discomfort and pain in situations such as flying or scuba diving where outside pressure differs from normal.

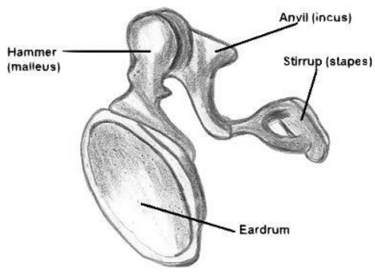

Figure 5.3: The middle ear ossicles (Buzzle, 2014:np)

The final part is called the inner ear. The inner ear is responsible for balance and holds the hearing sensors. The inner ear contains the body labyrinth which consists of the post superior and horizontal canals, vestibule and the Cochlea (fig 5.4). The Cochlea is the heart of the ear and is a spiral shaped bone that turns two and a half times around its axis. The name Cochlea is derived from the Greek word *kokhlos* which means snail. This is because the Cochlea looks like a snail's shell. Waves propagate from the oval window which is connected to the *Stapes* to the centre of the Cochlea in a spiral. The entire Cochlea is filled with a fluid called *Endolymph*. The Cochlea walls is strong bone and the fluid does not compress, so once the oval window moves the fluid inside the Cochlea will also move transferring the mechanical energy from the air pressure wave to the fluid. The fluid requires more energy to move than the original air pressure wave (Dallas, 1992:4575-4585). In order to transfer the correct wave a form of impedance matching is done between the middle and inner ear. The force that the ear drum experiences is equal to the pressure times the area of the drum. The cochlea is filled with fluid that requires more

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energy. The *Ossicles*, acting as levers on the oval window, transfer the force. However, this needs to be amplified. The oval window is smaller than the ear drum thus having a smaller area resulting in the amplification and impedance matching that is required (Zwislocki, 1959:841). Inside is the organ of *Corti* found along the length of the Cochlea. The organ of *Corti* contains hair cells that fire neural triggers according to the amount of degrees that they are bent. These neural signals are sent to the brain for auditory processing. The form of the Cochlea determines the frequency response of the hair cells. Humans are only able to hear between 20 Hz to 20 kHz due to the way that the Cochlea only spirals a set number of times. Different mammals have different hearing ranges due to different forms of the Cochlea (French, 1948:591).

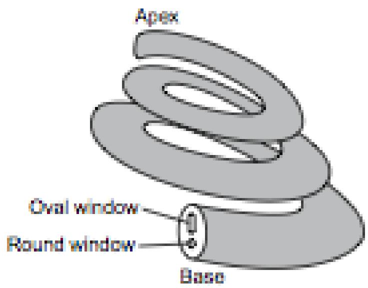

Figure 5.4: Inside the Cochlea (Howard & Angus, 2009:80)

5.2 Hearing loss

Hearing loss is the partial or complete inability to hear. There are various different types of hearing loss caused by a variety of factors, such as excessive exposure to noise, genetics, infections, diseases and physical trauma. In context of this project hearing loss due to ageing and specifically Presbycusis is of interest. Presbycusis is an age-related hearing loss (Pichora-Fuller & MacDonald, 2008:291-300). This effects everyone and steadily deteriorates the ears ability to sense certain frequencies as age increases. Loss of threshold sensitivity in high frequency spectrum of the audible range is the first indication of Presbycusis. The onset of Presbycusis starts as a young adult and becomes more noticeable as time passes. The high frequency threshold will deteriorate as time passes resulting in a cut-off frequency that creeps lower. This pattern is caused by the slowing of the metabolic function of the cochlea (Gates & Mills, 2005:1111-1120; Mazelová *et al.*, 2003:87-94). A study by D.W. Robinson and G.J. Sutton shows a significant inability by older people to experience higher frequency signals ranging upwards of 12 kHz (Robinson & Sutton, 1979:320-334). Presbycusis is likely caused by the ageing of hair cells

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and nerves connected to them that have become damaged or have died. Without the hair cells the ear will not be able to sense certain frequencies. The higher frequencies start to deteriorate much faster than the lower and men suffer more from Presbycusis than women (Stenklev & Laukli, 2004:295-306).

5.3 Phase cancellation

Phase cancellation or wave interference (fig 5.5) is the effect where two waves superimposed onto one another resulting in a different wave with higher or lower amplitudes. Superposition of two waves of the same nature that are propagating in the same space as one another results in interference. The interference is equal to the sum of the individual points of the individual waves displacement. If a positive $sin(x)$ wave is superimposed onto a $-sin(x)$ wave the result would be no wave at all. The crest of the first wave will be interfered with the second wave's nadir superposition, resulting in cancellation. The same is also true for two individual wave crests superimposed, resulting in a wave with double the amplitude as the original two.

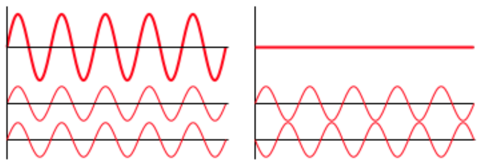

Figure 5.5: Wave interference (Ross, 2014:np)

The phase of a wave is the angular offset relative to the origin. The phase of a sinusoidal wave is easy to express as a $sin(x)$ function is set to complete a cycle in 2π radians or 360^0 degrees.

$$x = A \times sin(2\pi ft + \zeta)$$

In this equation ζ is the phase of the sinusoidal function and adjust the function to move forward or backwards in time relative to the original sinus function (fig 5.6).

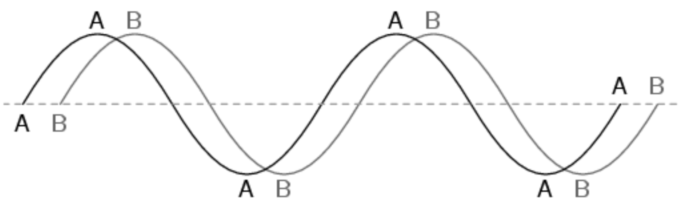

Figure 5.6: Phase shift of sinusoidal (Harvey Lew, 2014:np)

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The difference in time of two similar waves of the same frequency is called phase difference and is relative to the same point in time. If two waves are in phase it means that their crests and nadirs fall together and result in a larger wave during superposition. If two waves are out of phase then one of the wave crests superimpose with a nadir resulting in a smaller wave. Complete cancellation of a wave is possible if a identical wave interferes with it but is π radians,180^0 degrees or just the negative of the original wave. Phase cancellation occurs in sound waves as well and is very noticeable as destructive interference cause some audio frequencies to be accentuated whilst others are completely voided. If one sound source is recorded and two or more microphones are used, phasing issues can become apparent. The microphones must be the same distance from the source in order to capture the sound waves on the same instance during the wave propagation. If the microphones are not the same distance apart then one microphone will capture a crest and another a nadir resulting in a altered wave. The same is true for stereo pair loudspeakers. If one of the speakers polarity is wired wrongly then the negative of the signal will be transmitted resulting in the two waves destroying each other rather than building onto one another. Phase cancellation of sound results in a sound that has frequency content that is removed from the original due to the interference of certain frequencies (fig 5.8). This is called comb filtering and affects a specific group of frequencies. A comb filter is electronically modelled by adding a delayed version of a signal to the original signal (fig 5.7). Just as in phase cancellation this results in destructive interference. The comb filter transfer function is a series of impulses on frequencies with specific ratios to the fundamental frequency. The name comb filter is derived from the look of the transfer function.

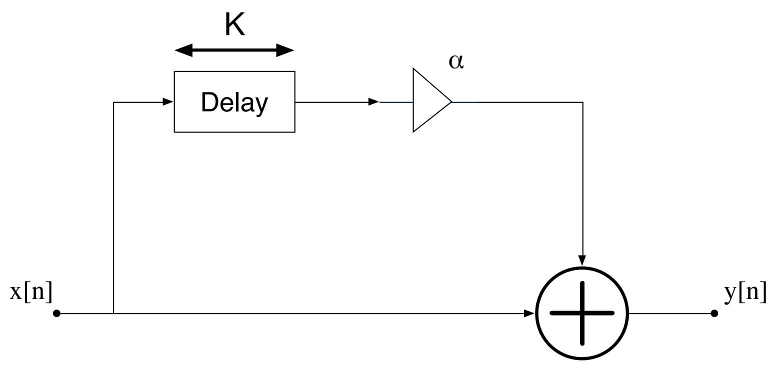

Figure 5.7: Comb filter block diagram

$$y[n] = x[n] + \alpha x[n - K]$$

Different degrees of phase cancellation occurs depending on the amount the two signal's phase differ. If the phase difference θ is between 0^0 to 179^0 the

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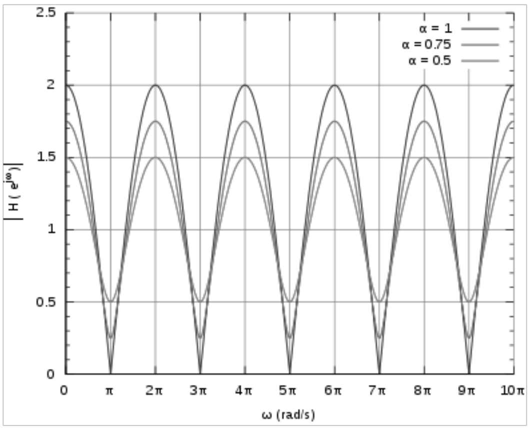

Figure 5.8: Comb filter frequency response (Proakis, 2001:320)

sound will be comb-filtered however if the phase difference is multiples of 180^0 total phase cancellation will occur resulting in no sound being sensed.

5.4 Non-Intrusive QR Code strategy

In order to reduce the possibility of distracting or intruding in the original media content that the audible QR Code will be embedded into adjustments need to be made. The code has to use audible frequencies and requires a minimum amplitude of loudness in order to ensure a low bit error rate during transmission. The only other option is to place the code in the audible frequency range that has less priority over speech for example. The other option is to attempt to use phase cancellation to reduce the loudness at certain positions in the stereo field.

5.4.1 Phase cancellation

Phase cancellation will require a stereo sound image to be transmitted. This means that the media content that the audible QR Code will be placed into will require two or more audio channels. The one channel will transmit the audible QR Code in phase or as it has originally been generated. The second channel will transmit the same QR-Code however it will be π radians out of phase with the original or the inversion. The two channels will transmit the code as well as the original media content into a stereo field towards the audience. The two QR-Code versions will destructively interfere in the centre of the stereo field and partially or completely vanish. The QR-Code can however still be recorded by isolating one channel from the other by physically recording close to the one source. This will record the QR-Code before interference with the inverted version. This technique is somewhat non-intuitive as is does not allow

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the entire stereo image to scan the code. The sensor is physically constrained to the spacial requirements that limits the performance of the code.

5.4.2 High frequency spectrum hiding

The QR Code can be placed in a high frequency band ranging from 15 to 20 kHz. The ear will not sense the code due to Presbycusis in humans and the reduced ability of the aged ear to sense high frequencies.

5.4.3 Ultrasonic frequency bandwidth

The ultrasonic spectrum is define as frequencies above 20 kHz and that is exactly where the QR-Code will be transmitted. Most consumer electronics such as smartphones and computers microphones and loudspeakers do not immediately cut off frequencies at 20 kHz. The frequency response reduces logarithmically starting at 20 kHz resulting in some performance above 20 kHz. This is used to transmit the audible QR Code above audible frequencies while allowing consumer electronics to sense the code with standard microphones and loudspeakers. The extra bandwidth gained is only about 2 kHz and it requires that the electronics analog to digital and digital to analog converters to sample at 48 kHz rather than the standard 44.1 kHz in order to avoid the Nyquist cut-off frequency and the resulting aliasing.

5.5 Experiment

The experiment was done in order to test the phase cancellation and the ultrasonic techniques in a real world scenario with actual microphones and converters. The Presbycusis hiding technique was tested in the same way as the ultrasonic technique however the conversion sampling rate was set to a standard 44.1 kHz because Nyquist will not be a problem. The phase cancellation test was done by playing two versions of the QR-Code using different modulation techniques and carrier frequencies. The one version was played through one of the Genelec 1031 A monitor while an inverted version that is 180 degrees out of phase was played through another Genelec 1031 A monitor. The two monitors played the same QR Code however one was playing an inverted version that in theory would result in no sound in the centre of the monitor pair as destructive interference would take place (fig 5.10). The ultrasonic QR Code test was done by playing the codes at carrier frequencies equal and higher than 20 kHz. The idea was to test the ability of ideal recording equipment to sense the frequency content in the ultrasonic range. The ultrasonic QR Codes were played through the Avid HD I/O with converters sampling at 48 kHz in order to avoid the Nyquist cut-off frequency. B&W 600 series and Genelec 1031 A were used as loudspeakers to transmit the QR Code. An iPhone was used as microphone set to record audio sampling at 48 kHz. Also a Sennheiser

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MHK 8040 (fig 5.11) was used to record the QR-Codes and the Avid HD I/O converters was used to sample the signal also at 48 kHz (fig 5.12).

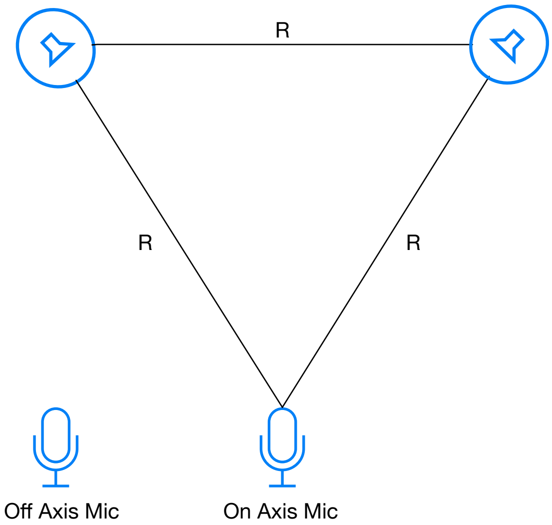

Figure 5.9: Phase Cancellation Experiment

Figure 5.10: Experiment using Sennheiser MKH 8040 and Genelec Loudspeaker

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Figure 5.11: One Mic placed in the centre for phase cancellation and one on the side

Figure 5.12: B&W Loudspeaker and iPhone experiment

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5.5.1 Equipment

5.5.1.1 Converter

The converter houses a analog to digital converter as well as a digital to analog converter. This is the device that will convert the input wave file that is digital into a analog signal that is sent to the loudspeaker. The converter used is part of a larger system called the Avid HD I/O 8x8x8 (fig 5.13). The Avid HD I/O is a multi-channel digital audio interface that couples with the Pro Tools software package. The audio interface houses a ADC and DAC that have a 24-Bit depth and is able to sample at up to 192 kHz. The version that is used is called the 8x8x8 due to it having eight analog inputs, eight analog outputs and eight digital in- and outputs. It is essential that the converters are able to sample at a rate faster than 44.1 kHz which is standard for most audio electronics.

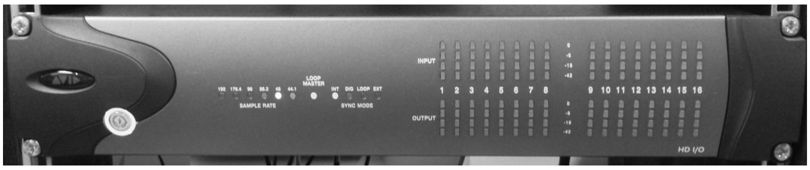

Figure 5.13: Avid HD I/O

5.5.1.2 Microphones

During the testing two microphones were used to record the audible QR Code. The first is a traditional microphone called the Sennheiser MKH 8040. The MKH 840 is a high-end studio condenser cardioid microphone (fig 5.15). This microphone was chosen particularly due to the wide-frequency response. The MKH 8040 has a frequency range of 35 Hz to 50 kHz (fig 5.14). The 8040 requires phantom powering which is a 48 V potential sent to the microphone to charge the capacitor. It requires a balances XLR-3 cable and has a input impedance of 25 Ω. The wide frequency response is of importance in testing the QR-Code in the 20-22 kHz frequency band.

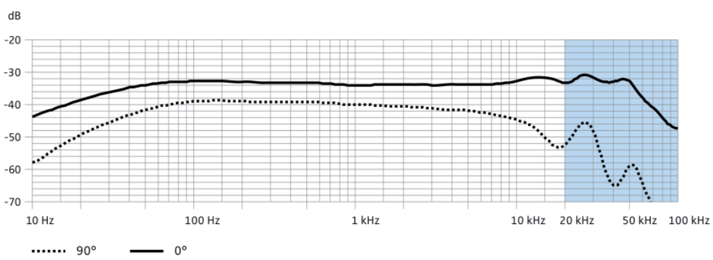

Figure 5.14: Sennheiser MKH 8040 Frequency Response (Sennheiser:9)

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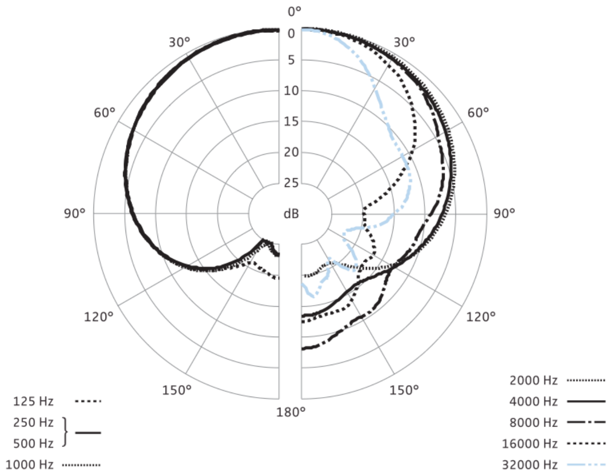

Figure 5.15: Sennheiser MKH 8040 Polar Pattern (Sennheiser:9)

The second microphone used is part of a Iphone smart-phone. The exact frequency response and polar pattern of the microphones inside the phone is not released publicly by Apple. However, the phone is able to sample at 48 kHz allowing for ultrasonic sound sensing enabling the testing of the ultrasonic hiding technique. An approximation of the iPhone frequency response has been generated by B. Faber (fig 5.16).

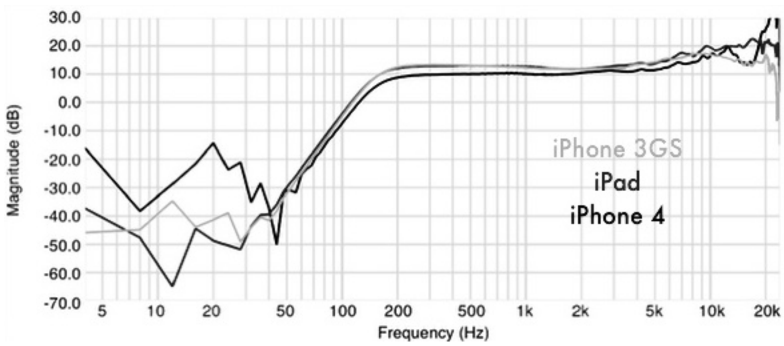

Figure 5.16: Iphone microphone frequency response (Farber, 2010:np)

5.5.1.3 Pre-amplifier

The signal from the Sennheiser MKH 840 will need to be amplified before it is sampled by the converters in the Avid HD I/O audio interface. A amplifier is required that is able to amplify signals above 20 kHz. The Buzz MA2-2 is used to amplify the signal (MA 2.2 User Manual V1:2-8). The Buzz MA2-2 (fig 5.17) has a flat frequency response from 2 Hz to 250 kHz ensuring that it will amplify past 20 kHz as is required. The Buzz will feed an analog signal into the Avid I/O completing the signal chain.

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Figure 5.17: Buzz MA2-2 Amplifier

Min Gain .... +16dB (-4dB with pad in)
Max Gain .... +65dB
Maximum Output Level .... +24dBu unbalanced
Frequency Response .... 2Hz to 250kHz @ 20dB gain (-3dB)
Frequency Response .... 20Hz to 250kHz @ 65dB gain (-3dB)
Harmonic Distortion .... less than 0.008
Slew Rate .... 140 V/uS, @ +20dBu output level
EIN .... -133.5dB A wtg, 150ohm source Z
Signal to Noise Ratio .... -74dBu A wtg, input shorte
CMNR .... 100Hz-80dB, 1kHz -80dB, 10kHz-70dB
Channel Crosstalk .... below noise
Input Impedance .... 3k ohms/1k2 ohms switchable

5.5.1.4 Loudspeakers

Genelec 1031 A monitors (fig 5.18) were used to broadcast the audible QR Code. The Genelec 1031 A is a bi-amplified studio monitor with a wide frequency response and very low colouring to the original source material that is used for studio applications, broadcasting and mastering. This loudspeaker was chosen particularly due to the wide frequency response ranging from 36 Hz to 22 kHz (fig 5.19). This is essential to test the ultrasonic technique to hide the audible QR Code.

Figure 5.18: Genelec 1031 A Monitor

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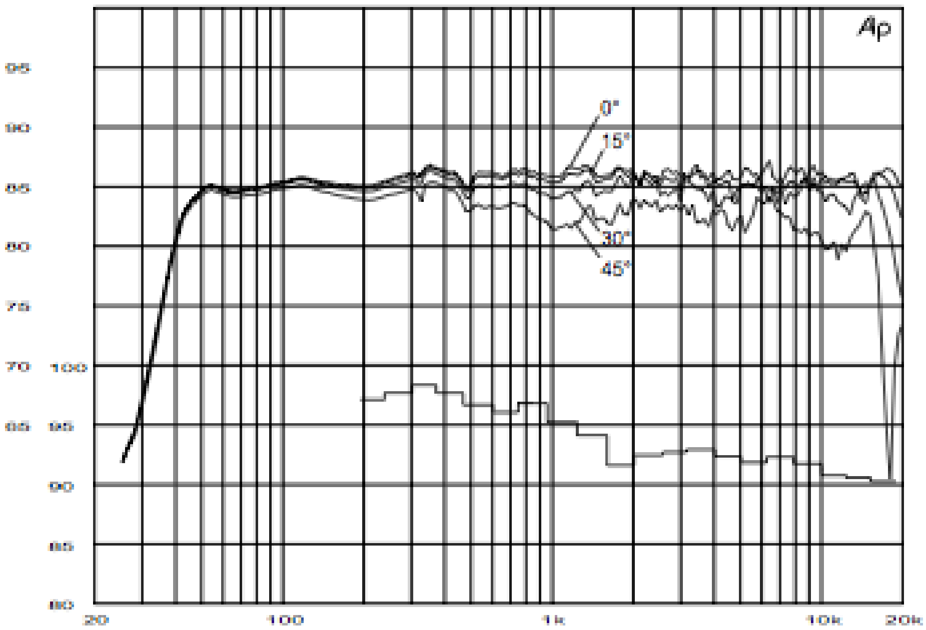

Figure 5.19: Genelec 1031 A Frequency Response (Genelec, 2014:4)

The Bowers and Wilkins 805D loudspeaker was also used during the experiment. The Bowers and Wilkins loudspeaker played back various different modulations and carrier frequency of the audible QR Code to an iPhone in order to verify that the iPhone can sense the code. The Bower and Wilkins loudspeaker has the ability to transmit sound above 20 kHz ensuring accuracy during the experiment.

5.5.1.5 Results

The phase cancellation technique results were not convincing. A very small signal cancellation was noticed on certain nodes in the stereo field however it was very small and not enough to completely cancel the signal. This technique was not effective. The ultrasonic technique results are convincing. The Sennheiser MHK 8040 was able to detect the audible QR Code at carrier frequencies of up to 22 kHz. The iPhone struggled to detect the carrier frequency at 22 kHz however it still functioned correctly up to 21 kHz. The Presbycusis technique functioned as expected. The signal was detected by the iPhone as well as the Sennheiser MHK microphones as the technical specification suggested. There were no external reviews on the loudness of the QR-Code was done as the data on Presbycusis is universal and can be judged from the sources conclusions. Additional results are listed in appendix 5.5.1.5.

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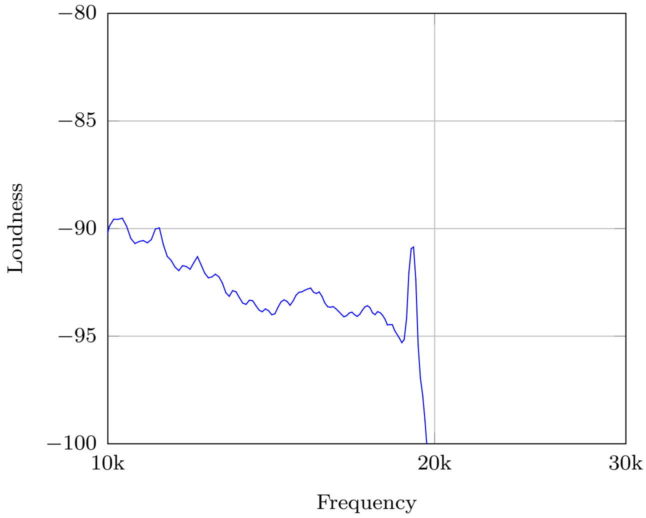

Figure 5.20: ASK 22 kHz signal detected by iPhone

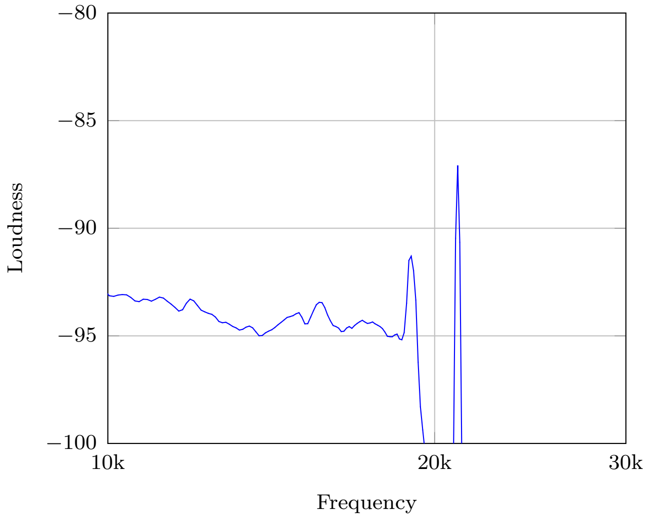

Figure 5.21: ASK 21 kHz signal detected by iPhone

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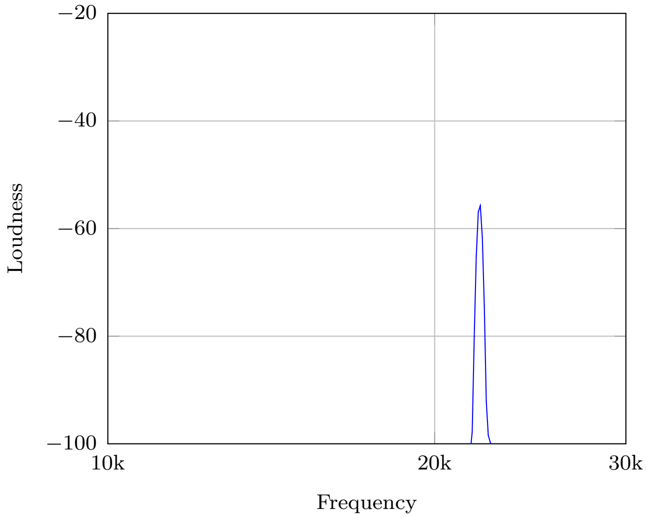

Figure 5.22: ASK 22 kHz signal detected by Sennheiser MKH 8040

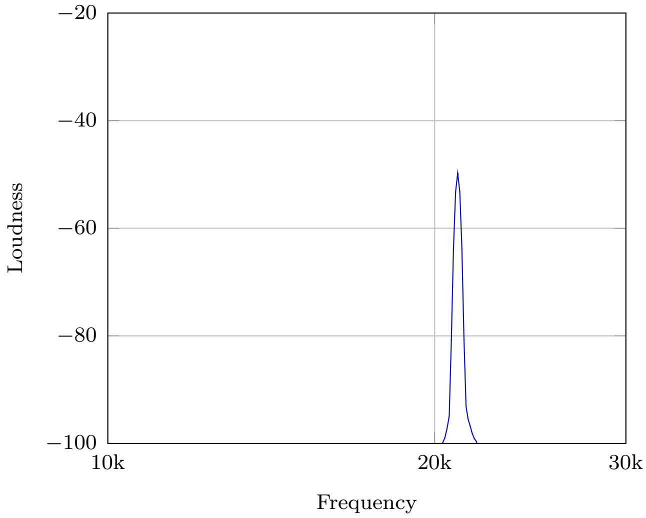

Figure 5.23: ASK 21 kHz signal detected by Sennheiser MKH 8040

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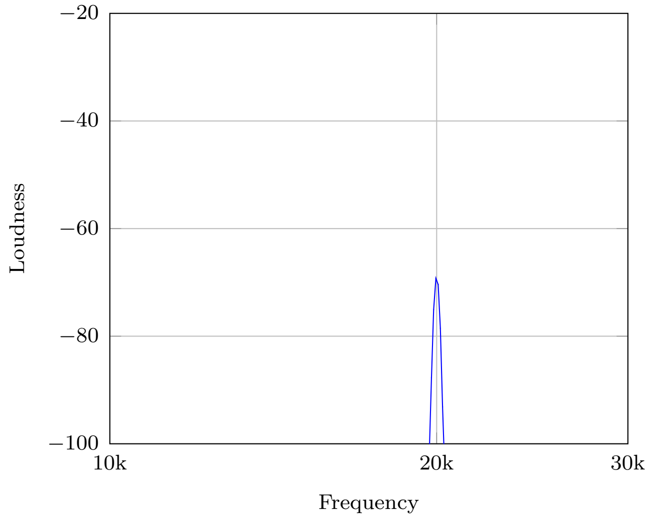

Figure 5.24: ASK 20 kHz signal detected by Sennheiser MKH 8040

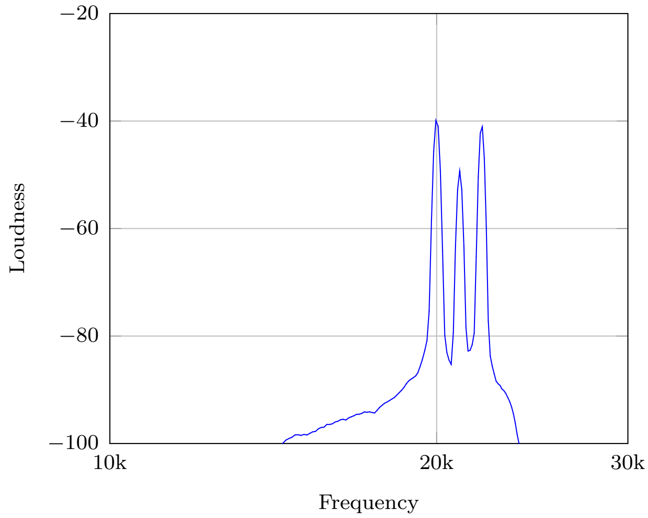

Figure 5.25: MFSK 21 kHz signal detected by Sennheiser MKH 8040

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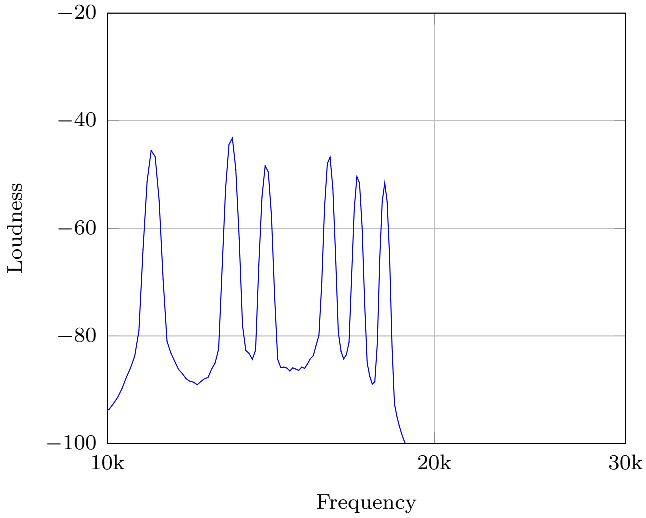

Figure 5.26: MFSK 15 kHz signal detected by Sennheiser MKH 8040

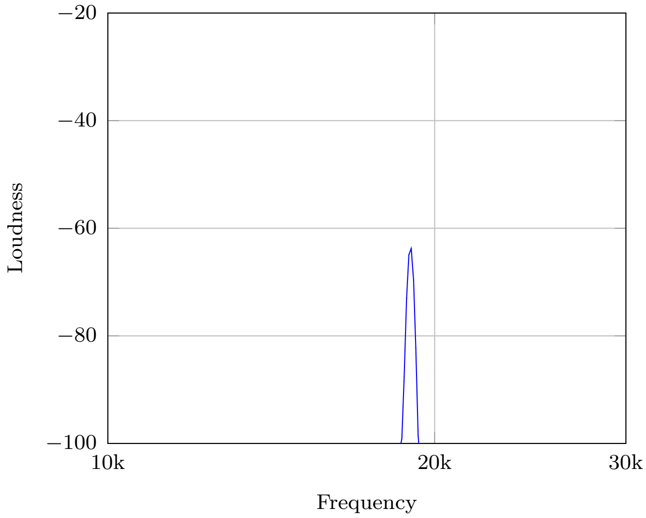

Figure 5.27: ASK 19 kHz signal detected by Sennheiser MKH 8040

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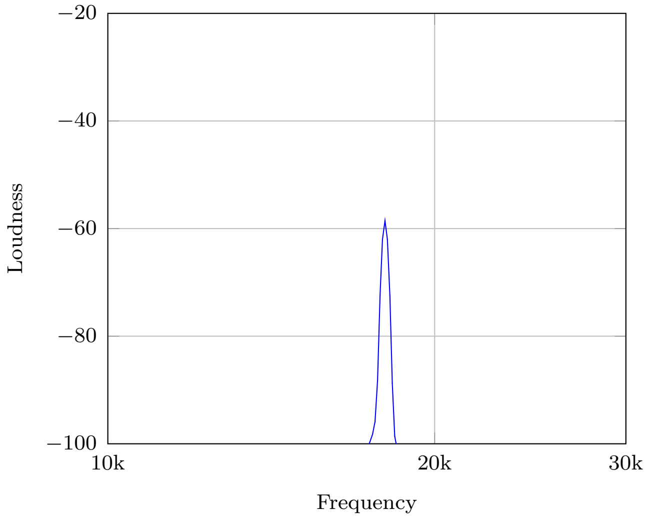

Figure 5.28: ASK 18 kHz signal detected by Sennheiser MKH 8040

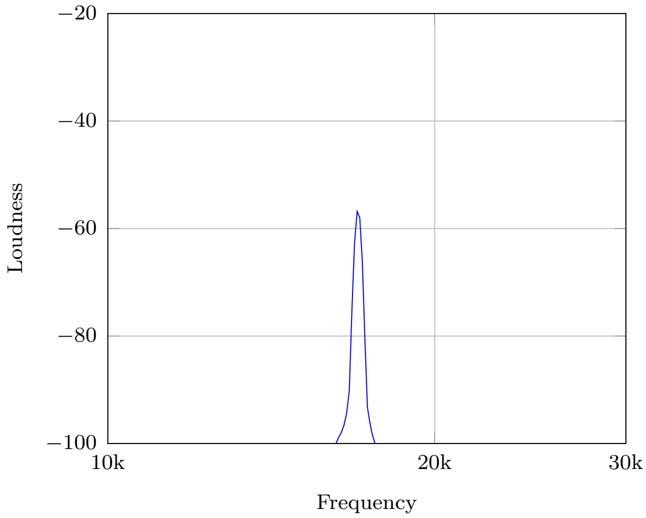

Figure 5.29: ASK 17 kHz signal detected by Sennheiser MKH 8040

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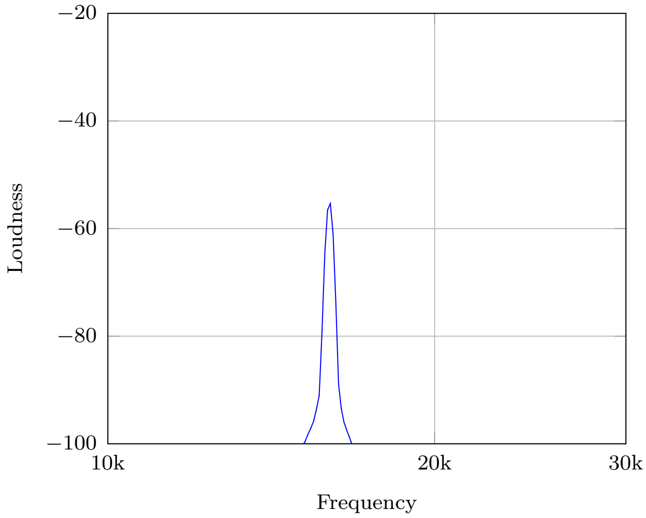

Figure 5.30: ASK 16 kHz signal detected by Sennheiser MKH 8040

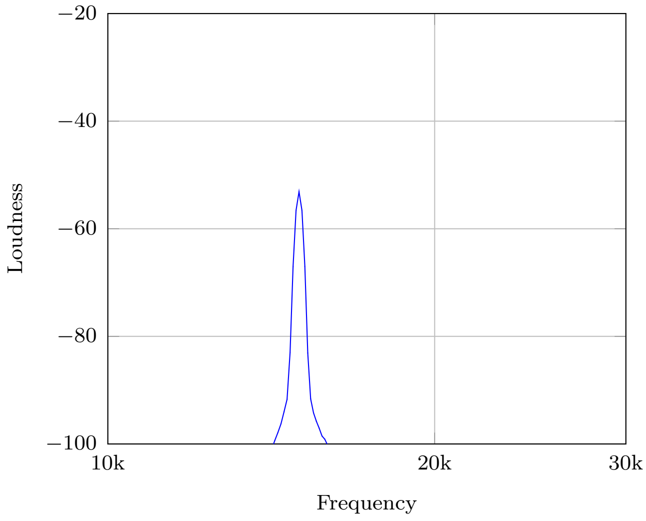

Figure 5.31: ASK 15 kHz signal detected by Sennheiser MKH 8040

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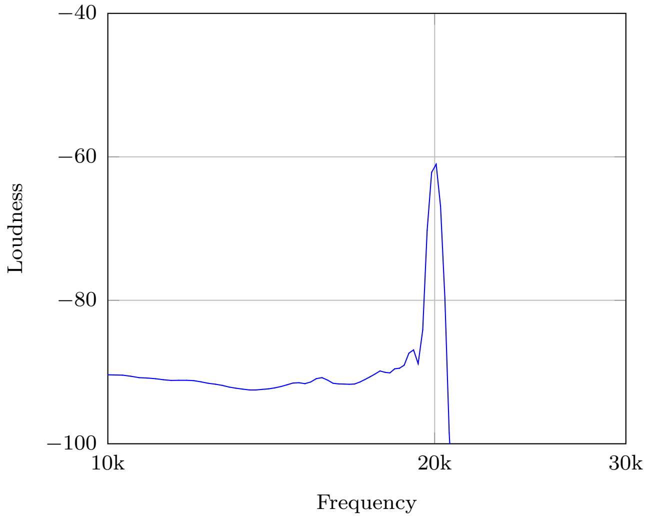

Figure 5.32: ASK 20 kHz signal detected by iPhone

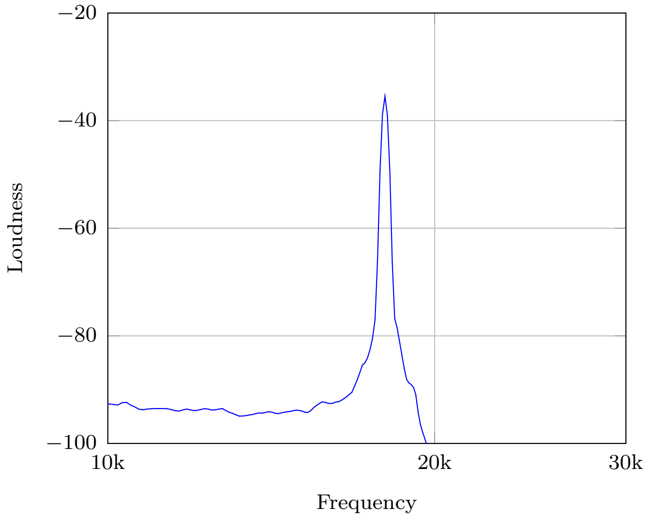

Figure 5.33: ASK 18 kHz signal detected by iPhone

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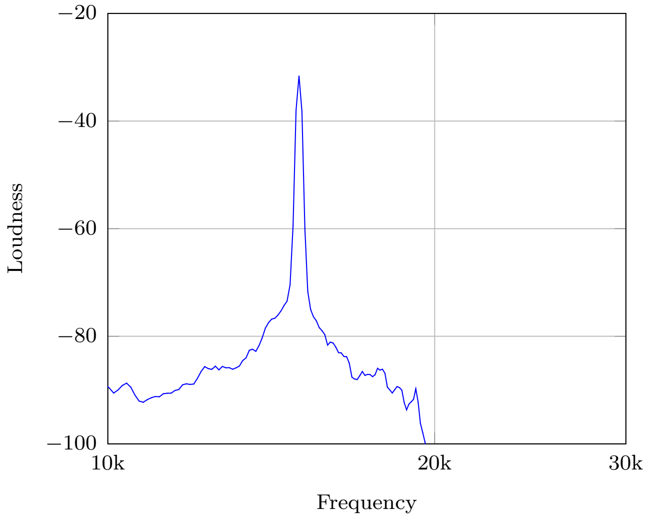

Figure 5.34: ASK 15 kHz signal detected by iPhone

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Chapter 6

Prototype design

The audible QR Code prototype requires a transmitter and a receiver as well as the modulator coupled with the filters. The transmitter (fig 6.1) consists of a modulator that requires two signal inputs. The first is the QR-Code binary data that will be transmitted. The second is the carrier signal that the message signal will be modulated with. The modulated signal will then be transmitted by the loudspeaker or the antenna in the communication system. The receiver (fig 6.2) consists of the reverse of the transmitter. The antenna or microphone in the receiver read the signal from the transmitter. The modulated signal enters the demodulator. The original carrier signal is also used in the demodulator to extract the original message data. The final stage is the filter that is used to reduce the noise and select specific frequency band depending on the modulation technique.

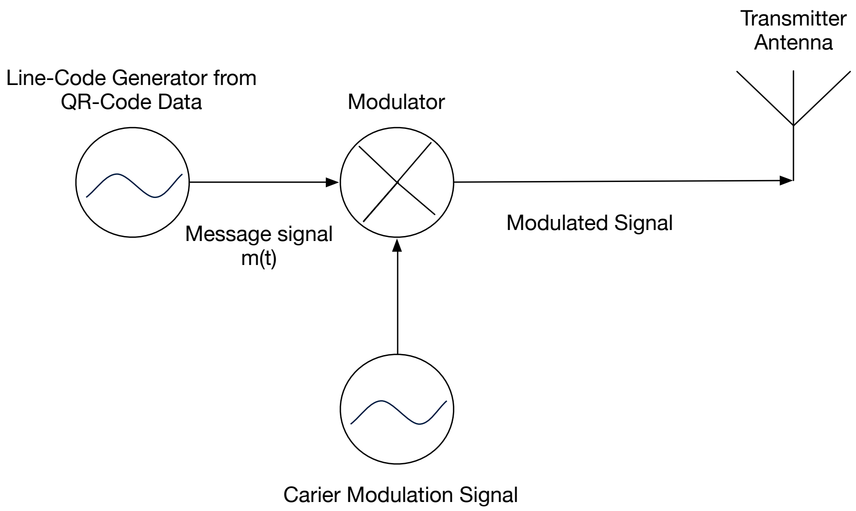

Figure 6.1: Prototype transmitter

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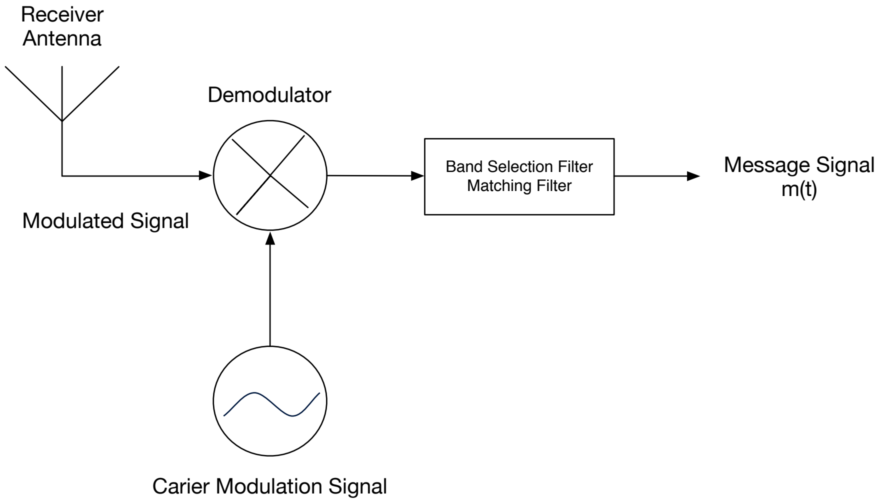

Figure 6.2: Prototype Receiver

6.1 Microphone

The microphone will be responsible for the capturing of audio signals that contain the QR-Code data. The microphone must be adept to specific characteristics to allow the faultless transmission of the audible QR Code. The microphone will form part of the analogue to digital system that will provide a software system with the QR-Code data that is converted from sound waves into binary sequences.

6.1.1 Audio in the analogue and Digital Domains

Computers have made it so easy to manipulate and represent sound that most music systems today are digital (Feldman, 1997:1-6). Sound has to be sampled in order to be saved in the form of bits of binary data. In the past tapes and vinyl records were used to store the sound waveforms. These waves were saved continuously and required no sampling due to the nature of the technology. Vinyl and tapes will trace the exact sound waveform on the storage medium continuously. Reproducing the sound was achieved by simply following the stored waveform. This is not possible in the digital domain. Digital and analogue stand as direct opposites and offer different advantages and disadvantages. The digital domain has become popular due to the ability to process a vast amount of data in a very short time through computers. In order to represent the sound wave data in the digital domain sampling is required. It is not possible to continuously represent the sound wave data because this will require an infinite amount of data points. This is not possible because the system will need to sample at a rate of infinity times per second. The larger problem, however, is the amount of storage space this will require because this process will generate an infinite amount of data as well. In order to repre-

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sent the sound wave as accurately as possible sampling is done. This is also a trade-off between accuracy and amount of data (Watkinson, 2001:1-9). There are two processes to the sound wave that together achieve sampling. Firstly and the most intuitive is the rate at which data is collected this is the amount of data points per second. This is the horizontal resolution and the higher the sample rate the higher the accuracy and larger amount of data generated. The second process is called quantization and is essentially the discretion of the vertical axis of the sound wave data (Vaseghi, 2007:155-171). This is the bit-depth of the sampling and can vary from 1-bit which is one or zero up to very large values. The higher the bit-depth the better the resolution and the higher the amount of data generated (Fries & Fries, 2005:239). This entire process is done by an electric component called a analogue to Digital Converter (ADC) (fig **??**) (Van de Plassche, 2003:2-12).

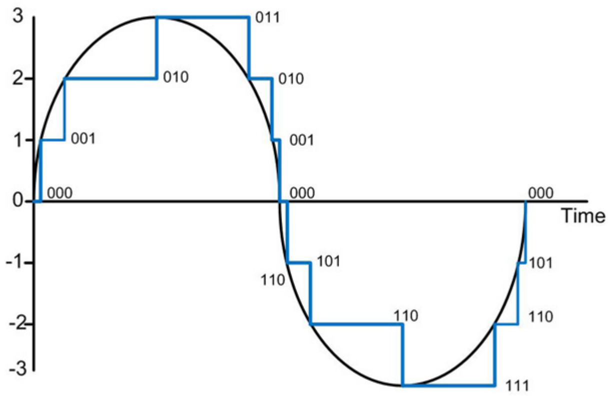

Figure 6.3: analogue vs Digital signal Representation (ScreaminFX, 2014:np)

6.1.1.1 analogue to Digital Converters (ADC)

analogue to Digital converters' sole purpose is to convert a continuous analogue signal such as a voltage or in this projects application a sound wave into the digital representation of the signal amplitude in the form of a block of data binary bits (Pelgrom, 2010:249). The output data will periodically be read by the system and is known as the sample rate. The amplitude of the signal is quantised to a set group of values and will create an error between the original analogue signal and the digital representation of it. This is known as the quantization error. The resolution of the ADC is known as the max value of input signal that maps to the maximum digital amplitude value (Sheingold, 1972:73-88). Essentially limiting the input analogue signal amplitude. If a ADC resolution is $\pm 5V$ any voltage above that will still only be mapped to the max digital amplitude (Griffiths & De Haseth, 2007:66-67). The resolution is the number of discrete values of digital data that is mapped to the amplitude of a analogue signal. The resolution is expressed in the form of bits as it is

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digital. A analogue to digital converter with a resolution of 8-bits can represent 256 different levels of amplitude. This can represent -127 to 128 if the input signal can be negative or 256 if only a positive input signal is expected.

$$2^8 = 256$$

The dynamic range for the input signal is determined by the following formula:

$$\triangle v = 2M_p/L$$

The input signal amplitude can range from $-M_p$ to M_p. This is not necessarily the max amplitude of the input signal however this is the max amplitude that the ADC will recognize. M_p is thus a parameter of the ADC not the input signal and is known as the limiter of the quantizer. L is the amount of discrete levels that represent the input signal amplitude. The amplitude range is divided uniformly(Tan & Jiang, 2013:40). The quantized signal and original signal will not be the same. An error is introduced due to the nature of quantization. The values are not exactly the same amplitude in the digital and analogue representations (Smith, 2003:35-36). $Sinc(2\pi Bt - k\pi)$ is known as the interpolation function[1].

$$m(t) = \sum_k m(KT_s)sinc(2\pi Bt - k\pi)$$

$$\hat{m}(t) = \sum_k \hat{m}(KT_s)sinc(2\pi Bt - k\pi)$$

$$q(t) = \sum_k [\hat{m}(KT_s) - m(KT_s)]sinc(2\pi Bt - k\pi)$$

$m(t)$ represents the analogue signal values. $\hat{m}(t)$ represents the quantized digital values. $q(t)$ represents the error generated by the process and is known as quantization noise (Op't Eynde & Sansen, 1993:93). The quantization noise contributes the most to the integrity of the signal. In order to improve the signal to quantization noise ratio compression is used. Digital compression is essentially non uniform division of levels. This technique becomes powerful where the resolution of the ADC is larger than the input signal amplitude range (fig 6.4) (Pearlman & Said, 2011:2). M_p is larger than the maximum amplitude of the signal. The larger levels in the ADC will not be used frequently and does not require very good resolution. The higher density of levels in the low amplitude range will be used the most and the higher resolution will decrease the quantization error made.

[1]The interpolation function is required to ideally reconstruct a signal from uniform sampling. The signal is represented by K sinc pulses in order to reconstruct the original analogue signal (Lathi, 1990:254)

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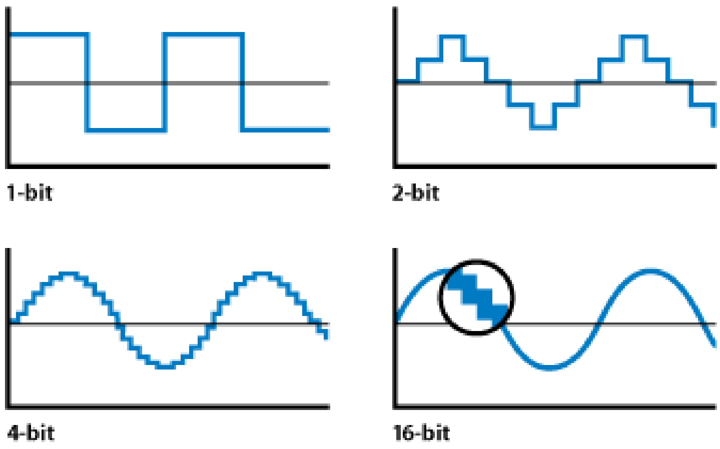

Figure 6.4: Different levels of Bit-Depth during Quantization (ScreaminFX, 2014:np)

Number of Bits	Number of Levels
1	2
2	4
4	16
8	256
16	65536
24	16777216

Table 6.1: Bit-Depth Table

A Compander is used to achieve this process and is essentially a compressor and a expander connected. The input signal will be weighted to a non uniform set of levels. Two specific industry standard curves are used to divide the levels known as the A-Law (Vaseghi, 2008:32) and μ -Law (Chitode, 2009:3-37). The μ-Law is used mainly in America and Japan while the A-Law is used in Europe (Lathi, 1990:274; Smith, 2004*b*:98-106).

$$y = \frac{1}{ln(1+\mu)} ln(1+\frac{\mu m}{m_p}) \quad 0 \leq \frac{m}{m_p} \leq 1$$

$$y = \begin{cases} \frac{A}{1+lnA}(\frac{m}{m_p}) & 0 \leq \frac{m}{m_p} \leq \frac{1}{A} \\ \frac{A}{1+lnA}(1+ln\frac{Am}{m_p}) & \frac{1}{A} \leq \frac{m}{m_p} \leq 1 \end{cases}$$

These two functions define the distribution of the levels of the ADC. The μ-Law function is a continuous curve with parameters μ and m_p, m is the input signal amplitude. μ is known as the compression parameter and determines the degree of compression. An industry standard of $\mu = 255$ is currently being used. m_p is the maximum signal amplitude that the ADC will recognize and will be mapped to the max digital amplitude. The A-Law is a piecewise defined continuous function with compression parameter A that also determine

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the level of compression. The dynamic range of the signal is captured by the quantized amplitude value in binary data. The frequency information of the input analogue signal is dependent on the sample rate. Sampling is the process of converting a continuous time signal into a discrete time signal by taking measurements at a specific time interval. The sample rate is the amount of data points of the amplitude per second. It is measured in Hz or samples per second.

6.1.2 Audio bit-depth

Sampled audio quality is dependant on the bit-depth and the sample rate of the ADC. These two variables directly affect the quantisation noise strength.

$$q(t) = \hat{m}(t) - m(t)$$

The mean square quantisation error known as the power of the quantisation noise is defined as N_q [2].

$$N_q = \widetilde{q^2(t)} = \frac{m_p^2}{3L^2}$$

$$SQNR = \frac{S_o}{N_o} = 3L^2 \frac{\widetilde{m^2(t)}}{m_p^2}$$

The SQNR is known as the signal to quantised noise ratio and is an indication of the quality of a converter (Dunlop & Smith, 1994:90-91). The SQNR has to be high enough to ensure the integrity of the signal (Lathi, 1990:273). L in the equation is the number of levels and is calculated directly from the bit-depth $L = 2^{numberofbits}$. This means that the SQNR is closely dependant on the bit-depth of the system. The higher the bit-depth the higher the SQNR and the higher quality the signal captured through sampling. Another method of increasing the SQNR is through oversampling. By increasing the frequency of samples taken the SQNR will increase due to the spreading of quantisation noise outside of the frequency bandwidth that is used. By sampling for a bandwidth of B_T rather than the original signal bandwidth B the bit-depth increases.

[2]The power of the quantisation noise is calculated using statistics (Vaseghi, 2008:29). The maximum error range lies between ($-\Delta V/2$ and $\Delta V/2$) and assuming that the error is equally likely to be in that range the power of the noise can be calculated as follows:

$$\widetilde{q^2(t)} = \frac{1}{\Delta V} \int_{-\Delta V/2}^{\Delta V/2} q^2 dq$$

$$= \frac{m_p^2}{3L^2}$$

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$$n = log_2 L$$

$$B_T = nB$$

$$\frac{S_o}{N_o} = 2^{2n} \frac{\widetilde{m^2(t)}}{m_p^2}$$

$$\frac{S_o}{N_o} = 2^{2B_T/B} \frac{\widetilde{m^2(t)}}{m_p^2}$$

The term $2B_T/B$ is increased due to the oversampling of the signal and thus increases the SQNR and looks as if it is increasing the actual bit-depth. By using this process a ADC that has a bit-depth of only x-bits can achieve a bit-depth of y-bits by sampling faster than the signal bandwidth if $y > x$ (Hauser, 1991:3-26). Audio is usually sampled at 44.1 KHz and is used in audio CDs and m-peg. The standard was developed by Sony and is derived from the Nyquist-Shannon criterion. Human hearing frequency response stretches from 20 Hz to 20 kHz (Lieberman, 2011:389). There are no filters that can filter a band steep enough to cut exactly at a specific frequency. There is always a slope connected to the filtering frequency response. Due to this a frequency bandwidth is chosen that is above the 20 kHz mark. The maximum frequency of the signal recorded will be at 22.05 kHz in using the 44.1 kHz sampling frequency. Other popular standards include 48 kHz, 88.2 kHz, 96 kHz and 192 kHz (Rumsey & McCormick, 2006:207-208). The significance of these numbers are usually associated by either a multiple of 44.1 kHz or of 48 kHz. Even though sampling at 44.1 kHz is essentially sufficient to reproduce the signal a increase in sampling frequency will increase the SQNR. The difference in practical quality is subjective even though a difference is numerically objective. Various bit-rates are used for audio sampling. 8-bit is usually not suited for media applications other than older generation video game consoles. The most popular is 16-bit. The 16-bit-depth standard was pioneered by the use of it in Compact Disks (CD) (Souvignier:116). $2^{16} = 65536$ levels meaning that a single sample can have 65536 different values attached to it. 24-bit is also popular in multimedia applications during the recording of content in order to have the highest quality samples before converting to other formats. A CD has a bit-depth of 16-bits and is sampled at 44.1 kHz and two channels resulting in a data rate of 1,411,200 bits/second (Mandal, 2003:344). This is significantly high and it becomes obvious why a very high bit-depth and sampling frequency is not realisable. Not only does the physical limitations of the electronic equipment limit the bit-depth and sampling frequency but also the economic constraints due to massive storage and processing equipment to manage the large amount of data. Software codec formats were developed to code and decode the raw data generated by the hardware into a more manageable data size but requires significant processing power. This is the main reason why CDs were designed to play raw data.

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6.1.3 Aliasing during sampling

The maximum rate of information is two pieces of information per second per Hertz (Lathi, 1990:262). This is due to Nyquist-Shannon sample theory. This theorem states that a band-limited continuous time signal with a highest frequency component at B Hertz can uniquely be recovered from its samples $x(n) = x(nT)$ only if the rate of sampling $F_s \geq 2B$ samples per second (Proakis, 2001:259). $2B = F$ is known as the Nyquist frequency and is the minimum sampling frequency from which a continuous time signal can be reconstructed from its samples. A signal $a_c(t)$ is shown in the time and $A_c(\omega)$ frequency domain (fig 6.5).

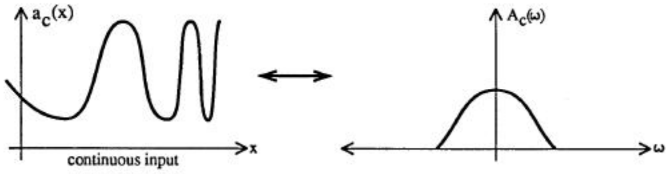

Figure 6.5: Signal $a_c(t)$ and $A_c(\omega)$ (Lathi, 1990:143)

$a_c(t)$ is a band limited continuous time signal. In order for the frequency spectrum to be band limited the time signal will need to be finite.This is due to the Fourier transform that is used on the time and frequency domain signals. This means that for a signal to be truly band limited the time signal will have to be infinite. Theoretically this is easy to achieve but in practice a infinite time signal is not realisable. $a_c(t)$ is sampled at a sample frequency $F_s = T$. $A_c(\omega)$ is the Fourier transform of $a_c(t)$ (fig 6.6) and (fig 6.7) (Sneddon, 1995:1-54).

$$A_c(\omega) = \int_{-\infty}^{\infty} a_c(t)e^{-j\omega t}dt$$

$$a(n) = a_c(t) \times \Delta_T = a_c(t) \times i(t)$$

$$a(n) = a_c(nT)$$

$$t = nT = \frac{n}{F_s} = \frac{n2\pi}{\omega}$$

$$A(\omega) = \Sigma_{n=-\infty}^{\infty} a(n)e^{-j\omega n}$$

Aliasing is the distortion of frequency information due to overlapping or error during sampling or modulation (fig 6.8). The sampled signal $a(n)$ results in no aliasing due to the sampling frequency being high enough and no frequency interference of distortion occurs. $a(t)$ can be recovered perfectly through the use of ideal interpolation (fig 6.9) (Zuch, 1979:13-18).

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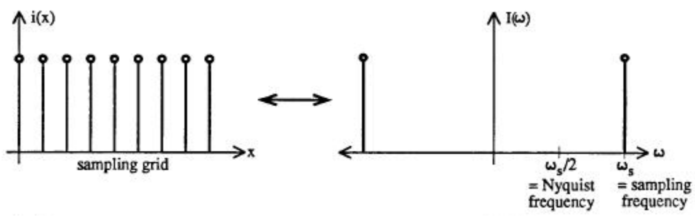

Figure 6.6: The sampling function also known as the Dirac Delta (Proakis, 2001:366)

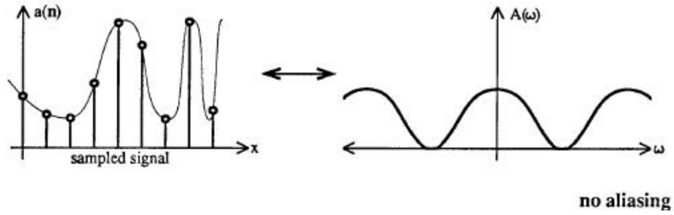

Figure 6.7: a(n) sampled signal (Proakis, 2001:366)

$$a_c(t) = \frac{1}{F_s}\int_{-\frac{F_s}{2}}^{\frac{F_s}{2}} [\Sigma_{-\infty}^{\infty} a(n)^{-j\omega n/F_s}] e^{j\omega t} dF$$

$$= \Sigma_{-\infty}^{\infty} a_c(nT) \frac{sin(\pi/T)(t-nT)}{(\pi/T)(t-nT)}$$

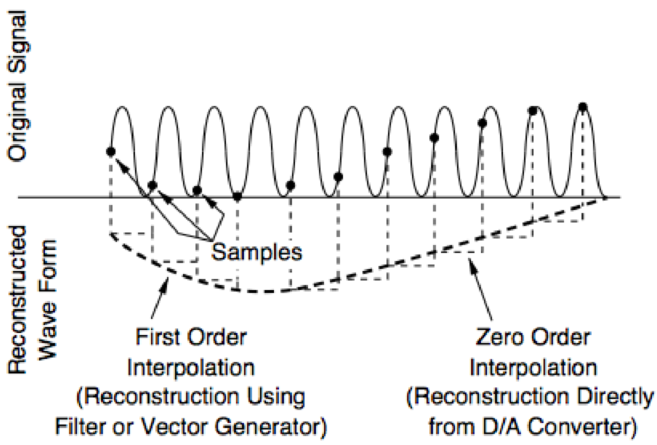

Figure 6.8: Reconstruction error due to aliasing

Adding weighted and time shifted instances of the ideal interpolation function will result in the reconstruction of the original continuous time signal (fig 6.10).

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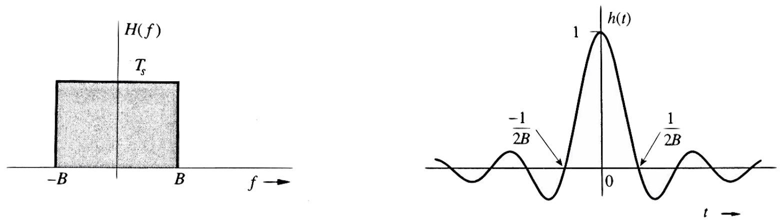

Figure 6.9: Interpolation function used to reconstruct signals (Lathi, 1990:254)

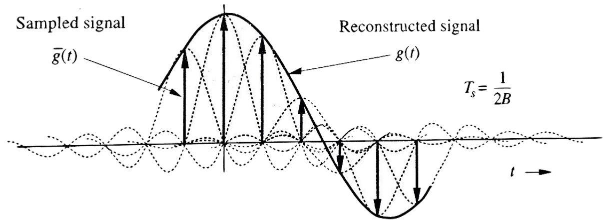

Figure 6.10: Signal reconstructed using interpolation (Lathi, 1990:254)

If the sampling rate is lower than the Nyquist frequency then aliasing will occur and frequency components will overlap and distort (fig 6.11).

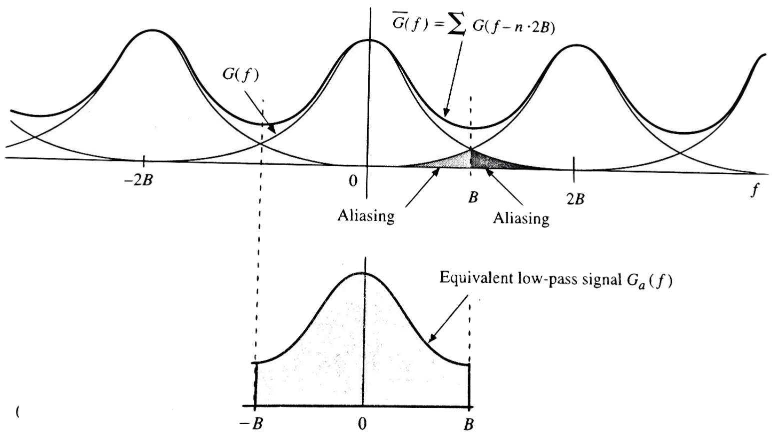

Figure 6.11: Aliasing caused by low sample rate (Lathi, 1990:263)

If the time signal is not band limited then aliasing will occur due to frequency components that are higher than $2B$. Nyquist states that in order to perfectly reproduce a time signal the sampling rate will need to be twice the frequency of this sampling frequencies' highest frequency component. If the sample frequency bandwidth is B then the F_s or sampling frequency will need to be $2B$ (Luo *et al.*, 2010:94). In order to perfectly reproduce the time signal in a error free, noiseless and distortion environment Nyquist needs to be achieved.

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6.1.4 Microphone hardware

6.1.4.1 Types of microphones

The purpose of a microphone is to convert audible sound or air pressure waves into an electric signal. This can be achieved through a number of different techniques each with advantages and disadvantages. The microphone will provide a voltage signal to the ADC so that the analogue sound signal can be converted into binary data. The first type is called the condenser microphone and is called this due to the use of a capacitor to convert the air pressure into signal. Capacitors used to be called condensers in the past and is where the name is derived from. The condenser microphone has a capacitor attached to a membrane (Owsinski, 2005:6). The membrane will move as air pressure is detected. The membrane will cause the distance between the two plates of the capacitor to vary following the air pressure signal. The following equation governs the capacitor:

$$C = \frac{Q}{V}$$

In the equation V is voltage over the capacitor. Q is the charge in Coulomb and C is the capacitors capacitance in Farad. The capacitance of the capacitor will change as the distance between the plates change (Sauls & Stark, 2013:66). The capacitance is governed by the following formula:

$$C = \frac{\varepsilon A}{d}$$

ε is a constant of the dielectric used to isolate the two plates. A is the area of each plate and d is the distance between the plates. This formula assumes square plates that are perpendicular to one another. The membranes movement will alter d and thus cause V to change. This is the voltage signal that is fed into an amplifier due to the low level of the signal then into the ADC. A charge is required to first initialize the capacitor. Phantom power is used to generate this charge and is a extra 48V which is sent to the microphone.

The next type of microphone is called a dynamic microphone. Dynamic microphone converts sound waves into an electric signal through the use of electromagnetic induction. A magnetic coil is attached to the membrane and placed in a magnetic field that is usually generated by a permanent magnet. The coil moves inside the magnetic field crossing magnetic field lines. According to Faraday's law of induction current will be induces in the coil. Faraday's law is defined in the following mathematical equation:

$$\mathcal{E} = -\frac{d\Phi_B}{dt}$$

$\mathcal{E}$ is known as the Electromotive force and Φ_B is the magnetic flux. The EMF is produced due to the membrane that moves in the magnetic field. This EMF will cause a change in the magnetic flux. This in turn will induce a

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current in the coil. The current is used to generate a voltage signal over a resistor. This voltage signal will be sent to the ADC as input. The dynamic microphone is very robust and has a low tendency to generate feedback noise. The frequency response is now however equal over the entire audible frequency spectrum and requires more than one membrane.

The last popular microphone type is known as a ribbon microphone. It is called the ribbon microphone due to a thin metal ribbon that is used as a membrane. The ribbon is placed inside a electromagnetic field. The ribbon extends and contracts due to air pressure. The change in the electromagnetic field generates a voltage. This voltage is used as output of the microphone. It also uses electromagnetic induction just as the dynamic microphone and follow the same mathematical and electrical laws. It does not however use a coil but rather a ribbon.

6.1.4.2 Microphone polar patterns

The microphone polar pattern is the microphones sensitivity to sound waves in each direction from its central axis. An omnidirectional polar pattern has the same amplitude response in all three axis (Musburger & Kindem, 2009:182). It does not matter where about the sound source is generated from the microphone. Microphones that are non-directional usually have a very flat response in the frequency domain and generally does not colour the sound (Ballou, 2008:493-497). A cardioid microphone response (fig 6.12) so named due to the cardioid graph the polar pattern generates. The cardioid rejects sound from the rear and allows sound from all other directions. The pattern is ideally suited to reject sound waves from the rear of the microphone. Sound waves from the front will be detected.

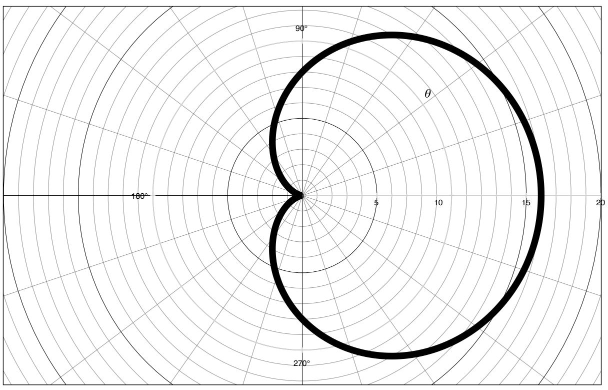

Figure 6.12: Cardoid Polar Pattern

A bi-directional or figure-8 microphone has a polar pattern (fig 6.13) that looks like a figure-8 (Atkinson, 2013:38). Sound waves from both sides are

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rejected. The sound waves from the front and back are detected.

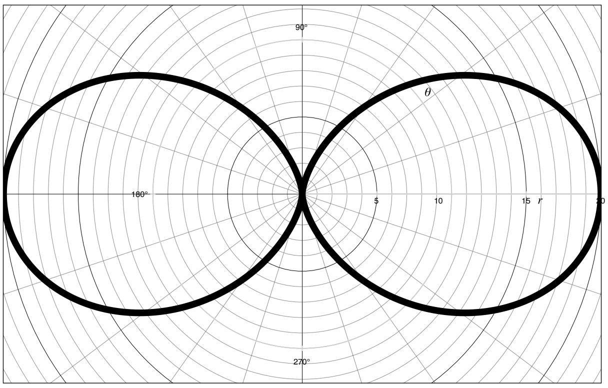

Figure 6.13: Figure-8 Polar Pattern

The last microphone polar pattern is called a parabolic microphone or shotgun microphone. The polar pattern has very sharp parabolas (fig 6.14). The microphone is extremely direction sensitive. It has a very narrow polar pattern that detect sound. The microphone will detect sound from the front very well. It also detects sound from the sides and back but at a lower level (Davis & Davis, 1989:119). All other directions are suppressed.

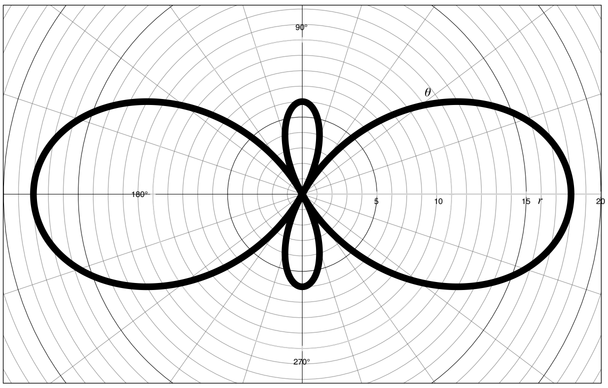

Figure 6.14: Parabolic Polar Pattern

6.1.4.3 Microphone frequency response

The frequency response of a system is the measure of output of a system due to a certain input signal over a range of frequencies. The frequency response of a system characterizes the effect that the system has on the input signal making the output more predictable. A bode plot is used to graph the frequency

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response of a system due to a impulse input signal. An impulse input signal is used because it has a uniform frequency characteristics. The mathematical impulse signal is not realizable in reality because it requires an infinite amplitude signal for a infinity small time interval. The impulse is known in math as the Dirac delta function or δ (Allen & Mills, 2004:186). This function is seen as a infinitely thin and infinitely high spike on the origin with an unity area.

$$\delta(x) = \begin{cases} \infty & x = 0 \\ 0 & x \neq 0 \end{cases}$$

$$\int_{-\infty}^{\infty} \delta(x)dx = 1$$

$$\int_{-\infty}^{\infty} \delta(x)A \cdot e^{-2\pi i x \xi}\, dx = A \cdot e^{-2\pi i x \xi}$$

The Fourier transform of the Dirac delta is a unity amplitude over all frequencies. This characteristic is used to map the frequency response of systems. An impulse input signal is sent into the system and the output will produce the frequency response. The $e^{-2\pi i x \xi}$ term is the resulting phase response (Davis *et al.*, 2001:64-65). This is the delay the system will affect to input signals at the specific frequency. A mathematical Dirac delta is not realizable in reality. A impulse is synthesised that does not have a infinity small time period. Due to the signal not being perfect mathematically it will not have infinite frequency components. It requires that a specific frequency band be analysed. Another method is iterating a sinusoidal input signal at a specific frequency at a specific amplitude through a system. The sinusoidal has a unity frequency response at one specific frequency. Scanning through all the frequencies and adding up the data will also produce a frequency response of the system.

All microphones have specific frequency response. Some frequencies are amplified while some are rejected over the audible spectrum of sound. There is some angle distortion that is accompanied by the frequency alterations of the microphone. It is essential to know the frequency response of a microphone in order to effectively sense information at a specific frequency bandwidth. If sound is transmitted at a frequency that is rejected by the microphone then the system will require to amplify the signal and will increase the noise in the system as this will also be amplified in the process. Also if the sound is transmitted in a frequency band outside of the microphones frequency response then the system will not detect the input signal at all (Holman, 2010:88).

In order for a speaker to reproduce the sound wave that the data represent extrapolation of data points is required. Digital to analogue converters are electric circuits that convert bits of data into voltages that are continuous.

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6.2 Loudspeaker

The loudspeaker will be responsible for the transmission of the QR-Code data. The binary data sequences will be converted into voltage signals through the use of a digital to analogue converter. After the conversion the voltage signal will be modulated. The modulated analogue voltage signal will then be converted to an audible air pressure signal through the loudspeaker.

6.2.1 Digital to analogue conversion (DAC)

Digital to analogue converter (DAC) is the reverse process of analogue to digital converter discussed in section 6.1.1.1. A DAC converts binary data that is discrete into analogue continuous signals of either current or voltages (Dillon, 2001:36). The DAC is essential in order to convert the binary data gathered from a sampler back into values of that can be used in analogue circuits (Saha & Manna, 2007:336-341). The application of DAC are vast-ranging from mobile music players to convert the data to the earphones to military radar systems. This component is essential to the digital revolution and works in tandem with the ADC to convert back and forth from the analogue to the digital domain and back. The DAC converts binary data that represents a fixed point value in a sequence of values to a analogue signal. This means that in a binary sequence there is a set number of blocks of binary data that represent numerical values. The binary block bit-depth is set for a system. Meaning this is the resolution of the DAC and will couple the highest value in the binary block to the highest value of the analogue signal after conversion. This is exactly the same as the M_p for the ADC. A 8-bit DAC maximum digital value will be $2^8 = 256$. The systems maximum digital value is thus 256 meaning the analogue signal that is generated by the conversion maximum value will correspond to that. The resolution of the DAC is thus 256 levels and can be divided equally or following some compression system such as A-law or μ-Law. The binary block values are fed into the DAC and analogue values in the form of either a current or voltage signal is generated. The binary data has no time element associated with it as it is only a sequence and can be generated as slowly or quickly as the data is read however there is usually a set sampling frequency associated to the application. The data was sampled at F_s when it was stored so to reproduce it the same sampling frequency will need to be used. Impulses will be generated at the specific instance that corresponds to the sampling at the amplitude that is represented by the binary block data. Due to quantizing and the error that is generated by the process the signal will not match the original one perfectly. Interpolation is used to generated data that is most likely for the gaps between samples in order to smooth out the analogue signal rather than having constant values between samples. If the signal was sampled at or above Nyquist frequency then the signal can be reconstructed with a high degree of precision the only change would be the noise generated by the quantization. If no interpolation is used a zero-hold system is utilized (Nassar, 2001:67-68).

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The zero-hold will simply keep the output the same until a new sample is read into the DAC. This results in a graph that does not closely resemble the original. However, after filtering out the redundant frequency elements that are usually above the original signal's highest usable frequency and Nyquest frequency, an analogue signal that resembles the original is generated. The output is held constant for T seconds which relates to the sampling frequency. In order to reduce the noise generated by the quantizing process oversampling can be used. Oversampling will sample faster than is required by the Nyquist theorem resulting in more data points than is required to reconstruct the time signal. The noise is however reduced due to the increase density of samples resulting in less quantization error.

6.2.2 Practical realisation of a DAC

In practice there are a few methods to achieve digital to analogue conversion. One method is achieved through the use of a DAC connected to a sample and hold circuit (fig 6.15). The final phase is a low-pass filter that cuts out all frequency content above the Nyquist criteria of the original signal.

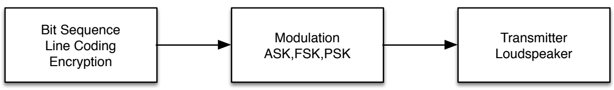

Figure 6.15: Sample and hold diagram

The resolution of the DAC is contained by a $\pm V_{max}$ that is expected by the device connected to the output. The digital binary block data is arranged so that each binary sequence is associated to a level on the DAC and corresponds to a analogue value (fig 6.16).

The DAC is characterized by the settle time that defines the time for the sample and hold block (fig 6.17) to reach the appropriate value (Kularatna, 2003:98-100). If two consecutive binary blocks represent a large change in output signal amplitude a spike in the amplitude will be noticed. The spike is a result of circuit elements not being able to instantly change value. Inductors and capacitors require time to change voltage and current and result in large spikes if sudden changes are required. A DAC must hold the amplitude value constant until the sample and hold circuit has settled to the new value in order to avoid such spikes. The sample and hold will feed data on a clock cycle that represents the sampling frequency (Austerlitz, 2002:58). This is the time interval in which the value stays constant and is known as T. Two sequential values will be minimum T seconds apart.

In section 6.1.1.1 the interpolation function was briefly discussed. For the DAC the interpolation function is no longer just a sequence of impulses but

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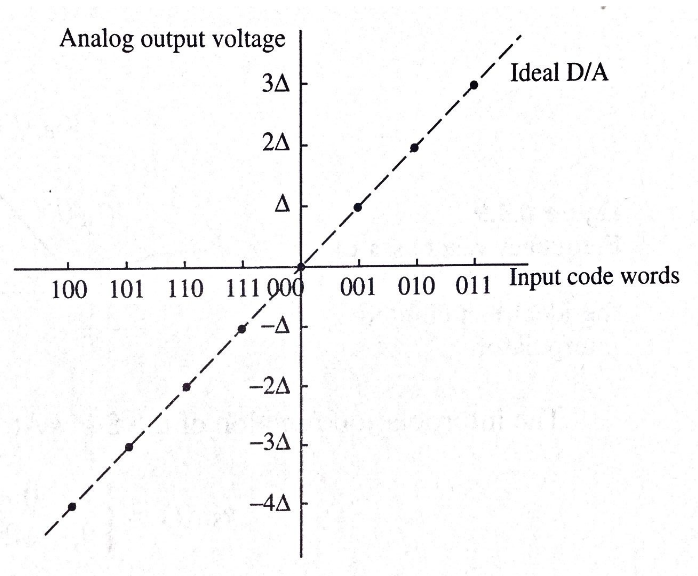

Figure 6.16: Sample and old voltage to binary conversion (Proakis, 2001:379)

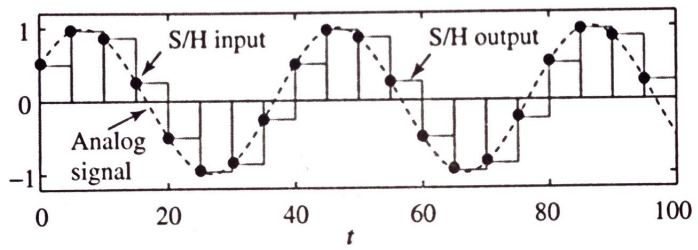

Figure 6.17: Sample and hold reconstruction (Proakis, 2001:379)

rather blocks that is generated by the sample and hold circuit (fig 6.18). The sample and hold generates square pulses define as follows:

$$g_{sh}(t) = \begin{cases} 1 & 0 \leq t \leq T \\ 0 & otherwise \end{cases}$$

The frequency-domain characteristics obtained through Fourier transforms results in the following:

$$G_{sh}(F) = \int_{-\infty}^{\infty} g_{sh}(t) e^{-j2\pi Ft}\, dt = T \frac{sin\pi FT}{\pi FT} e^{-2\pi F(T/2)}$$

The ideal sample and hold frequency response would have no components above the Nyquist criteria. However due to the response being limited it means the frequency response if not limited in frequency. This means there are frequency components up to ∞ and down to $-\infty$. This is due to the sharp edges of the time signal and Fourier theory. The resulting sample and hold will allow frequency components past the Nyquist frequency that is unwanted. A

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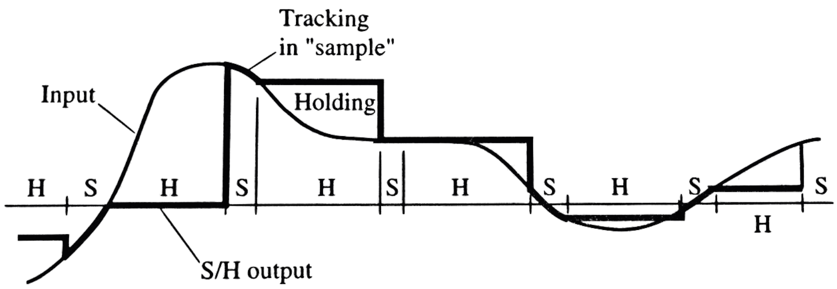

Figure 6.18: Reconstruction using a sample and hold DAC (Proakis, 2001:372)

low-pass filter is used to exclude these extra frequency components (Engelberg, 2008:1-9).

A pulse width modulator (PWM) circuit is also a popular design to achieve digital to analogue conversion. A pulse width modulator adjusts the duration of a pulse in order to modulate data into the signal (fig 6.19). The binary data is passed into a circuit that determines the duration of the pulse. This will correspond with the largest digital value being the largest analogue voltage value and the largest pulse width value. The pulse is then integrated and the value is presented as output in the form of voltage or current. The higher the pulse width the higher the output voltage due to the area of the pulse. The output has to be filtered in order to remove high frequency content due to the sharp edges of the pulses (Mercer, 2008:13; Hiorns *et al.*, 1991:142-147).

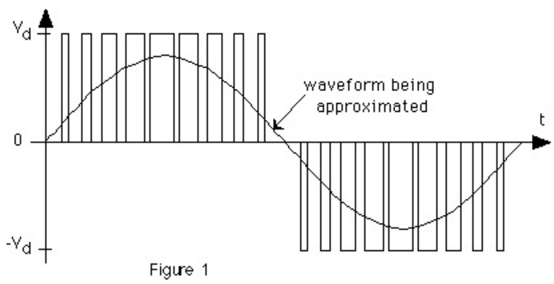

Figure 6.19: PWM DAC Reconstruction of a digital signal (Mercer, 2008:12)

A resistor ladder or (R-2R) ladder (fig 6.20) is another method that is used to produce a analogue output from digital data (Kester, 2005:155). The R-2R ladder scheme is a electronic circuit with repeating resistors of values $R\Omega$ and $2R\Omega$ in order to scale the output. The individual ladder act as current dividers and thus adjust the output voltage depending on the digital value that is presented to the circuit (Naylor, 1999:np ; Wang *et al.*, 2001:1026-1031).

$$V_{out} = V_{ref} \times BinaryInput/2^N$$

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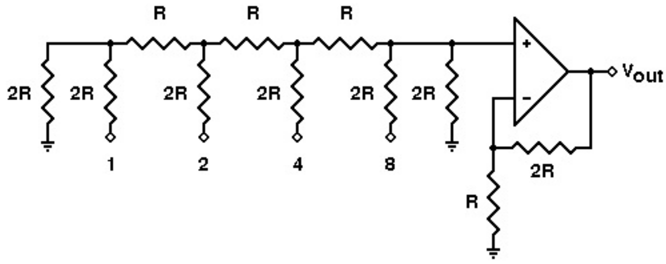

Figure 6.20: R-2R Ladder (Mercer, 2008:14)

The output voltage V_{ref} is a function of one variable. V_{ref} can be chosen and is used to scale the entire systems output range. A industry standard of $3.3V$ is used for CMOS circuits. N is the DACs binary resolution and for the example $N = 4$. The $BinaryInput$ is the variable that adjust the V_{out}. $BinaryInput = \frac{DecimalValue}{Decimalresolution}$. $BinaryResolution = 2^4 = 16$. For example a binary value of 1000 would result in a V_{out}:

$$V_{out} = 3.3 \times \frac{8}{16} = 3.3 \times 0.5 = 1.65V$$

6.3 Transmission

Digital data transmission is the most effective method of transmitting data. The only analogue modulation application left in modern times is radio broadcasting. All modern communication systems use digital modulation there are several reasons why. Digital communication resists channel disturbances such as noise, distortion and line faults better than analogue communication (Smith, 2004*b*:15-16). A digital message is not rendered void if there is some alteration to the signal however analogue signals will not be usable. The other core advantage digital signals have over analogue signals is the ability to repetitively regenerate the signal. Signals, analogue and digital, diminish in amplitude due to resistance during transmission. The signals amplitude reduce however channel noise remains constant over the entire length of the channel. The receiver will be required to be close enough to the transmitter to be able to distinguish between the noise and the signal. Amplifying the signal does not solve this problem because during the amplification process the signals as well as the noise will be amplified and the signal to noise ratio (SNR) will remain the same (Simon & Alouini, 2005:4; Martin, 1976:264). Analogue signals transmission distance is is limited to the initial transmission signal power. In order to transmit the analogue signal further more power is required. Digital signals experience the same effect. The receiver requires a high SNR in order to extract the data. Digital signals can however be regenerated and retransmitted extending the distance of the transmission and essentially making the transmission distance unlimited. The digital signal is transmitted to a regenerator

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that receives the signal with a good SNR. The regenerator will then extract the digital data and regenerate the digital signal and transmit it to the next regenerator (Freeman, 2004:261). The bit error rate (BER) can be adjusted by reducing the distance between regenerators or increased to reduce cost and the number of regenerators required (Barry *et al.*, 2004:1-7). The greatest advantage of digital is the use of microprocessors that can analyse and process data faster than humans. Digital line coding schemes allows the altering of transmission bandwidth and power requirements as well as reducing BER and the implementation of security such as encryption. Digital circuitry is able to multiplex a number of digital signals together more efficiently than analogue signals. Storage and reproduction of digital data is more cost effective than analogue alternatives. The cost of digital system are significantly less than analogue and is reducing as time passes as most technological advances occur in the digital field (Lathi, 1990:326).

6.3.1 Line Coding

Digital data is converted into electrical square waves to be transmitted. The square waves represent the digital data in different ways and is called line coding. There are different line-coding schemes that inherent different bandwidth, power and transmission speed characteristics (Anderson, 2006:16). The goal of a line coding scheme is to minimize the transmission bandwidth to as small as possible. The coding scheme must be power effective while maintaining a specified BER and bandwidth. Error detection and correction is desirable and must be included if possible. The power density spectra which is essentially the power in different frequencies must have no direct current components and avoid the lower frequency bands. Synchronization and a direct current offset is experienced with bad coding schemes during long sequences of a particular digital value. This sequence must not affect the coding scheme and must remain transparent. The timing detail must be accurately portrayed as this is essential to the entire system (Lathi, 1990:330). On-off switching is the most intuitive line coding scheme. A digital one is represented by a pulse and a digital zero is represented by no pulse (fig 6.21) (Helfrick, 2010:315-316).

$$p(t) = \sqcap(\frac{t - T_b/2}{T_b})$$

$$On - off(t) = \begin{cases} p(t) & Binary \Rightarrow 1 \\ 0 & Binary \Rightarrow 0 \end{cases}$$

Polar switching is very similar to on-off switching. A digital one is still represented by a pulse but a digital zero is represented by the negative of the pulse (fig 6.22) (Lathi, 1990:374).

$$bipolar(t) = \begin{cases} p(t) & Binary \Rightarrow 1 \\ -p(t) & Binary \Rightarrow 0 \end{cases}$$

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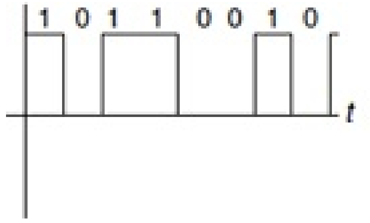

Figure 6.21: On-Off Line Coding (Lathi, 1990:329)

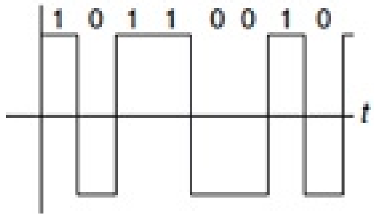

Figure 6.22: Polar Coding Scheme

Polar Coding Scheme (Lathi, 1990:374)

The bipolar coding scheme is derived from the polar coding scheme and is very similar. Digital zero is always represented by a zero, however, digital one is represented by a positive or negative pulse. The sign of the pulse is dependent of the previous digital one in the line code. If the previous one was a positive pulse the next digital one will be a negative pulse (Viswanathan, 1992:169-175). The digital one pulse representation thus alternates between positive and negative (fig 6.23).

$$bipolar(t) = \begin{cases} \pm p(t) & Binary \Rightarrow 1 \\ 0 & Binary \Rightarrow 0 \end{cases}$$

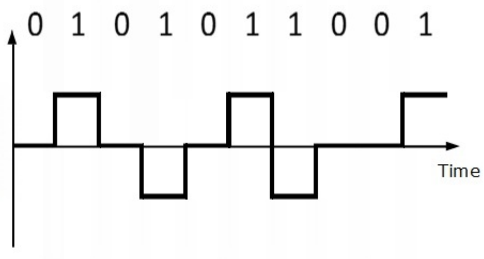

Figure 6.23: Bipolar Coding Scheme (Lathi, 1990:329)

6.3.2 M-Ary Digital Modulation

Binary digital modulation ASK, FSK and PSK all transmit one bit of data every T_b seconds. Otherwise stated it transmits at a bit rate of $R_b = \frac{1}{T_b}$

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bits/s. Greater transmission rates can be achieved by increasing R_b and thus decreasing T_b. However, there is another way, called M-ary signalling. M-ary signalling increases the number of bits transmitted per second by increasing the number of bits one pulse represents (Singal, 2010:205-206). By reducing T_b more bandwidth will be required for the transmission. This is not always the option as bandwidth is limited. M-ary signalling does not increase the transmission bandwidth as T_b stays the same however more power is required to transmit different pulses that represent more bits. M-ary signalling can be applied to ASK,FSK and PSK.

6.3.3 M-Ary ASK

M-Ary ASK will transmit different amplitudes of the carrier signal to represent different bit word values (fig 6.24). An ASK system will now transmit log_2M bits per second by modulating the carrier amplitude to 2^M different amplitude values.

$$\psi(t)_M ASK = 0, Acos(\omega_c t), 2Acos(\omega_c t), \cdots, (M-1)Acos(\omega_c t)$$

The M-ASK signal is still regarded as a AM signal that can be demodulated non-coherently through envelope detection and coherently through carrier convolution. The M-ASK transmission bandwidth remains the same as the binary ASK signal bandwidth, consequently the power required increases exponentially at a rate of M^2 (Lathi, 1990:380).

6.3.4 M-Ary FSK

M-ary FSK as M-ary ASK is very similar to the binary equivalent. In the binary case there was only two frequencies used to transmit the digital data. M-ary uses 2^M different carrier frequencies in order to transmit the signal. The carrier frequencies are selected from a the set $\{Acos(2\pi f_i t)$ where i $= 1, \ldots, M\}$. The transmission bandwidth does increase substantially due to the increase in the number of carriers at various different frequencies. The frequency set i must thus be chosen to reduce the overall bandwidth of the signal but also large enough so that the receiver is able to distinguish between the various frequencies. The minimum frequency deviation δf is calculated through the use of orthogonal sets of base functions that are used as carrier signals (Lathi, 1990:380-382).

$$\int_0^{T_b} Acos(2\pi f_m t)Acos(2\pi f_n t)dt = 0 \quad m \neq n$$

$$sin[2\pi(m-n)\delta f T_b] = 0 \quad m \neq n$$

The requirements thus state that the two functions will be orthogonal if δf meets the following requirement:

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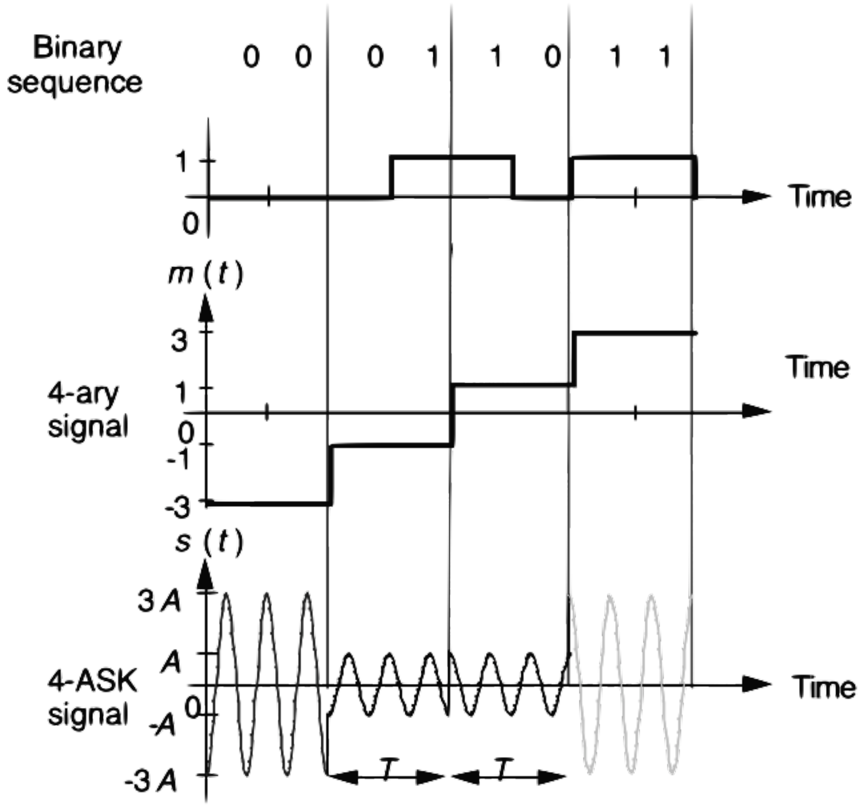

Figure 6.24: MASK Modulation (Lee, 2002:np)

$$\delta f = \frac{1}{2T_b}$$

The transmission bandwidth will be $M\times$ binary transmission bandwidth and the transmission remains exactly the same.

6.3.5 M-Ary PSK

2-Ary PSK uses two unique phase offsets to the carrier in order to signal the two different binary values. M-Ary PSK extends the amount of phase offsets to 2^M. Rather than transmitting a cosine or sinusoidal signal depending on the signal as in the binary case M-Ary will adjust the phase in multiples that is normalized over 2π or 360°(fig 6.25) (Stranneby, 2004:153).

$$\psi(t)_{MPSK} = A \times sin(2\pi f_c t + \theta_m)$$

$$\theta_m = \theta_0 + \frac{2\pi}{M}(m-1)$$

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The difference in phase will be unique depending on the Ary scheme used. The individual messages are divided up equally in the phase space.

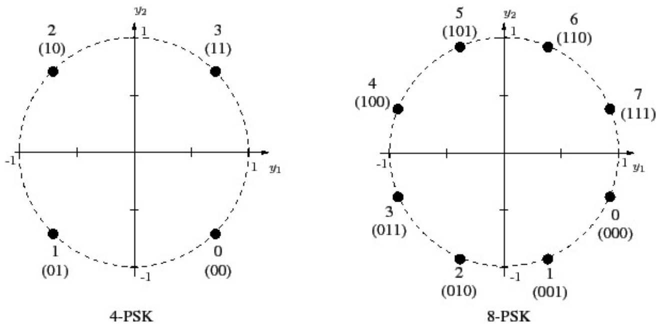

Figure 6.25: M-Ary PSK Phase Shift Distribution (Lathi, 1990:383)

M-Ary PSK is the most popular technique in modern communication systems. It has the ability to transmit more information over a smaller transmission bandwidth than FSK. It also requires less transmission power than ASK and similar transmission power to FSK. M-Ary PSK just as FSK requires coherent demodulation and cannot be demodulated with envelope detectors or filters such as ASK (Lathi, 1990:385).

6.3.6 Pulse shaping and line code modulation

As stated in section 6.3.1 the main goal of line coding is to produce a signal of data that has a specific power density spectra as well as a small message bandwidth. This is essential because perfect square waves does not posses these characteristics and need to be shaped. The different coding schemes express different power density spectra. The power density spectra is known as a PSD or $S_y(f)$. PSD is a function of frequency and indicates the frequencies where power is present in the signal. This is useful as this will be the power that needs to be transmitted for the receiver to demodulate the signal. The PSD also shows the message signal bandwidth. This is the lowest frequency to the highest frequency with significant power (Lathi, 1990:343-347). The three line coding schemes mentioned in section 6.3.1 have different PSD characteristics. A polar line code scheme that has a full width pulse has a PSD that has Direct current and low frequency content (fig 6.26). It also has a bandwidth that is double the bit rate. Polar signalling does not have a favourable PSD and the bandwidth is double the theoretical minimum making polar not a very desirable option. Polar is however very transparent meaning a clock signal can easily be extracted from it.

On-off signalling has very little desirable characteristics. On-off is more susceptible to bit errors and interference for the same amount of power compared to polar and bipolar. On-off is also not transparent and a long sequence

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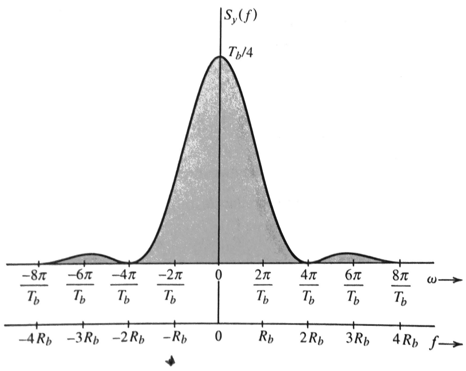

Figure 6.26: Power Density Spectra for Polar Line Coding (Lathi, 1990:336)

of zeros will make the clock extraction hard. It also has a DC power offset as well as large transmission bandwidth (fig 6.27).

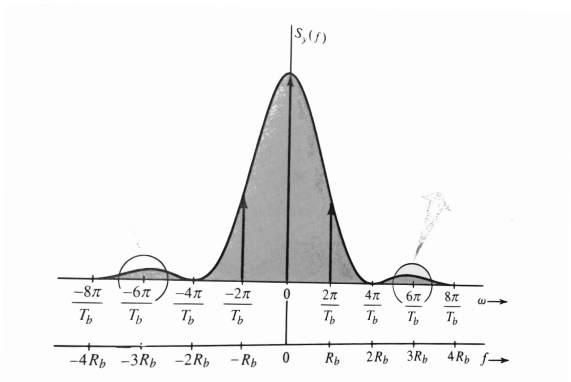

Figure 6.27: Power Density Spectra for On-Off Coding (Lathi, 1990:339)

Bipolar offers the best characteristics of the three coding schemes. Bipolar has no power presence in DC. It also has the lowers transmission bandwidth of the three whilst maintaining transparency. Bipolar is also just as resistive to channel disturbances and bit errors as polar. It however requires twice as much power as polar line coding (fig 6.28).

The PSD can now be shaped through the use of different pulses instead of just coding schemes. This means that the bandwidth and power can be adjusted even more to ensure resistance to channel interference and to improve transparency. Nyquist theory states that a maximum of R_b bits/s can be transmitted over a minimum theoretical transmission bandwidth of $R_b/2$ Hz (Lathi, 1990:345). This is achieved by transmitting a signal known as a sinc (fig 6.29).

$$p(t) = sinc(\pi R_b t)$$

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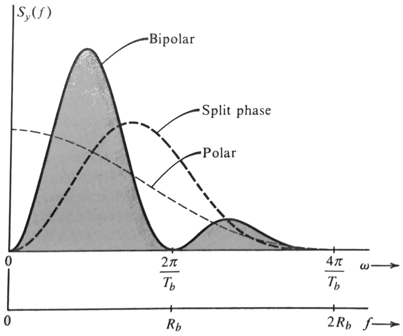

Figure 6.28: Power Density Spectra for Bipolar line coding (Lathi, 1990:341)

$$sinc(\pi R_b t) = \begin{cases} 1 & t = 0 \\ 0 & t = \pm nT_b \end{cases}$$

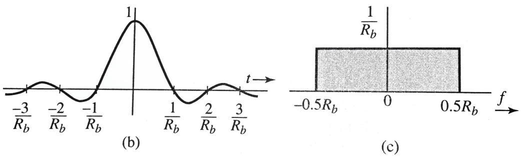

Figure 6.29: The Nyquist Pulse (Lathi, 1990:344)

This pulse is perfect in the theoretical world but not realizable in the real world. It requires $-\infty$ time and also decays too slow in time. This is however used to create a pulse that is realizable in the real world. A frequency roll of is required to ensure that the pulse decays faster in order to ensure that timing detail is not distorted in channel too much due to jitter. This is done by rolling of the frequency spectra of the pulse (fig 6.30).

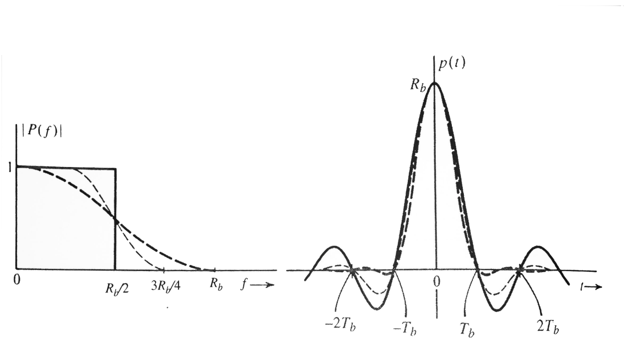

Figure 6.30: Perfect pulse altered through a roll off (Lathi, 1990:346)

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This pulse can be approximated by a filter with the impulse response of $p(t)$. The bandwidth will vary between $R_b/2; R_b$ Hz. Another method of altering the pulse is by affect the DC offset caused by polar and on-off switching. There will be a DC offset unless the overall amount of zeros and ones are equal in a line code of data. This can be avoided by changing the type of pulse used to transmit ones and zeros. If the pulse used can be summed to zero over a time period for zero symbols and ones then there will be no DC offset. This scheme is known as Manchester coding (fig 6.31) (Das, 2010:91-92).

$$\int_{-\infty}^{\infty} p(t)dt = 0$$

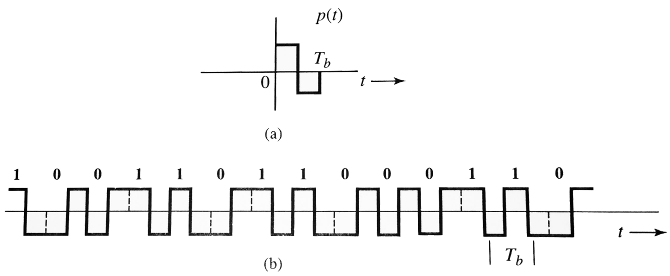

Figure 6.31: Manchester Coding (Lathi, 1990:337)

6.4 Security

The security of the audible QR Code is essential if it is to be used in mobile payments, voucher systems and transmission on sensitive data. The audible QR Code is designed to be broadcast to as many receivers that are in range. This presents the problem that no private transmissions is possible unless encryption is used. Eavesdropping on audible QR Code transmissions is easy, as all you need is a receiver that can record the audible message. However if encryption is used then only the devices with the ability to decrypt the messages will be able to understand and interpret the information sent. Other communication protocols will use a pseudo random key as public key and transmit a private key at the start of the transmission. This allows an eavesdropper to read the key and gives it the ability to decrypt the message. The audible QR Code is also susceptible to this security flaw. The audible QR Code is however able to use different mediums to transmit the private key as the reader will have access to internet through wi-fi or normal GSM networks. The private key can thus be sent in a different medium as the information transaction medium increasing the security significantly. A private key can for instance be sent via a SMS (Short Message Service) to a user allowing a secure transaction. Mobile

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payments can be done securely without the risk of a intruder listening in on the exchange of data. Data alteration and extraction is a great concern of the audible QR Code. While using the audible medium to transmit data is convenient, this allows intruders to alter and read data in transmission with the use of only a speaker and microphone. The intruder will have to know the exact sequence of data transmission as well as recognise the packed of data being sent. Changing an individual bit is very hard and for all intents insignificant as the receiver will not recognise the message received expecting an error has occurred. The receiver will request that transmitter to resend the data in such a case. If the intruder knows intricate details of the transmission such as bit sequences and package data then the transmission is at risk. This can easily be countered by encoding as in eavesdropping. Transmission corruption is also a concern however this is easily noticed. The jamming device will have to be in audible range and can easily be located and removed. The only adverse affects of corruption is that transmission will not be possible. The harm that can be achieved is quite insignificant and can easily be solved. The man in the middle attack mentioned in chapter 4.2.2 is a risk but not practical for the audible QR Code. A man in the middle attack required a device to fool the receiver. The receiver must think that the message is being transmitted from the original transmitter. Through the use of encryption,a private key and periodic authentication check this can be completely eliminated. The audible QR Code can be just as secure as Bluetooth and NFC because these technologies do not rely on the transmission medium to secure the connection. No pairing is required unless a secure connection is required an in such a case pairing can be done without user interaction just as NFC and Bluetooth Low Energy. This is however not very relevant to the audible QR Code because although it has the ability to match other technologies in security it is designed to broadcast messages to many people. These messages must not be encrypted, they must be easy to view and effective at communicating a large piece of information quickly and in a way that is understandable. The QR-Code can adapt to a function an could be used for encrypted as secure connections if so required but it is not the field in which it is designed for.

6.5 Prototype in Matlab

A prototype transmitter was created in Matlab[3] that generates the audible QR Code (fig 6.32). The prototype has the ability to create the audible QR Code using various different modulation techniques and carrier frequencies. The packet data can also be changed depending on what data wants to be transmitted. The transmitter consists of three parts the first is the data input and encryption. Data input is the payload that will be delivered to the receiver. The data will be formatted into packets with each having a identifier, checksum

[3]The full Matlab code can be found in appendix B.

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and error correction redundancies for reconstruction at the receiver. This part is usually done by a microprocessor as it involves large computations. In this model the payload is already formatted and is used as a sequence of binary data.

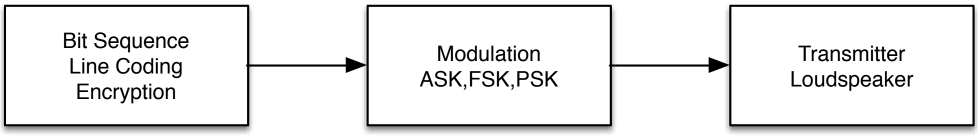

Figure 6.32: Matlab Transmitter Model

```
binarysequance= [1 0 1 1 1 0 0 1 1 0 1 1 0 0 1 0 1] ;
```

The next part is the modulation of the payload. The model uses three different modulation schemes. Amplitude, frequency and phase shift keying. It also uses different version of the modulation schemes to transmit more data known as M-ary scheming. The binary sequence is converted into a line coding scheme. There are many different schemes as discussed in section 6.3.1, in the model a basic on-off scheme is used. The bit rate is set to by changing R_b and the sampling frequency by changing F_s. F is the carrier frequency, this can be change depending on the frequency band the message must be transmitted in.

```
fs = 48000 ; % sample freqs
f = 22000 ; % Message freq
Tb = 1/1 ; % Set Data Rate bits/s
Rb= 1/Tb ;

for i = 1:n-1
     for j = 1:(fs/Rb)
    linecode(1,1+j+(i*(fs/Rb))) = binarysequance(i);
    end
end
sinsampled = sin(2*pi*(f)*t);
ASK_signal = linecode.*sinsampled;
```

The different modulation techniques have different frequency characteristics and power requirements. The transmitter will format the payload into a modulated signal. F_{ASK} is just the frequency domain of the ASK signal and is converted by using a FFT (Fast Fourier Transform) (Madisetti, 2010:1-21).

```
f_ASK = abs(fft(ASK_signal)) ;
```

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The last part is the transmission of the signal. The model uses a speaker to transmit the code through air pressure waves. The waves will reach the receiver and will be demodulated and decoded.

```
sound(ASK_signal,fs);
wavwrite(ASK_signal,fs,'ASK_Test');
```

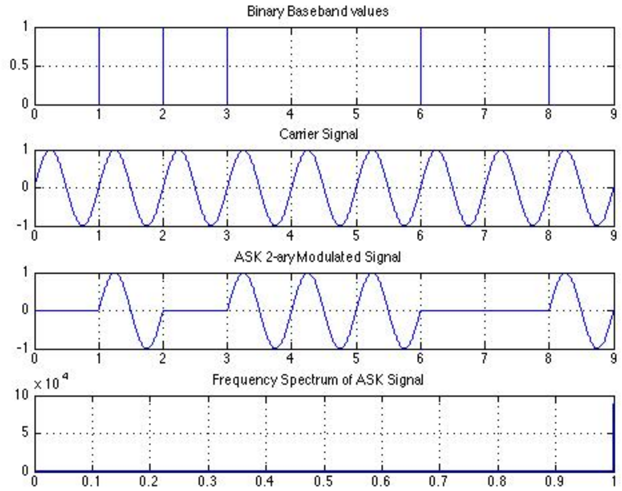

Figure 6.33: ASK Line Coding, Carrier Signal and Frequency content

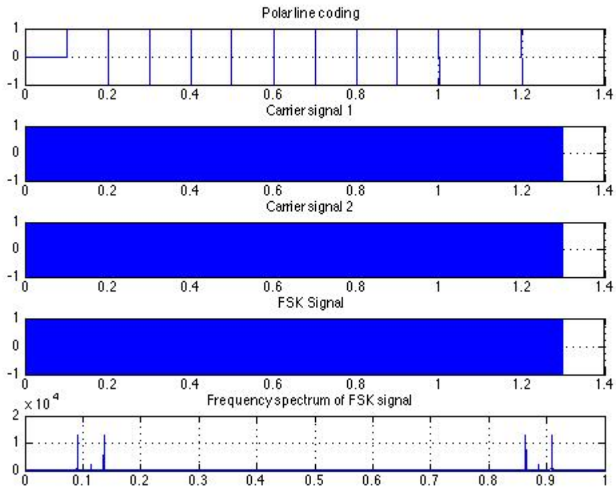

Figure 6.34: FSK Line Coding, Carrier signal and frequency spectrum

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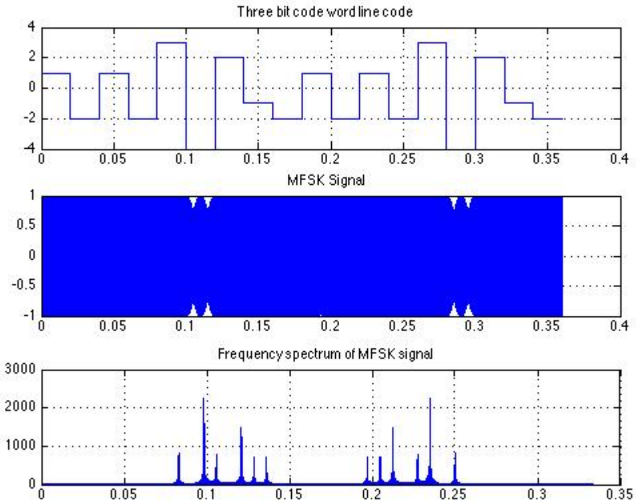

Figure 6.35: M-FSK Line coding, Carrier Signals and Frequency Spectrum

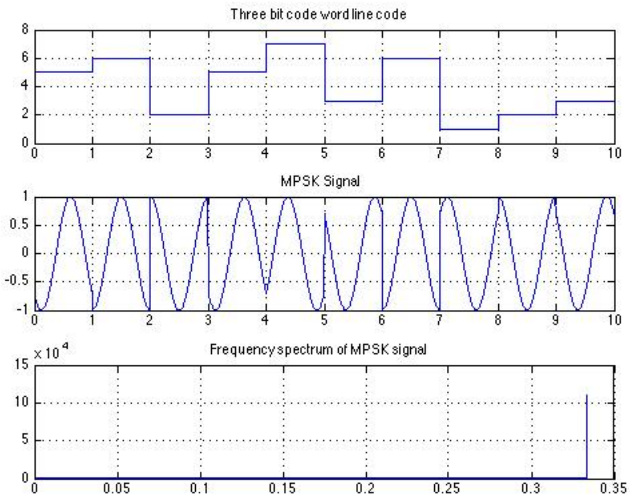

Figure 6.36: M-PSK Line coding,Carrier signal and frequency spectrum

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Chapter 7

Practical uses of the audible QR Code

The audible QR Code at the core remains a method of transferring information accurately, faster and more dense than traditional methods such as language and symbols. The inability of humans to understand the audible QR Code or decipher it is where the mobile phone becomes essential (Winter, 2011:48). The audible QR Code will never be viable unless mobile phones remains viable or an equivalent tool that can act as a receiver. The receiver allows us to draw information out of a range of acoustic frequencies. The receiver will present the information in a form that humans can understand and use to better the experience and effectiveness of modern day communications in a wide range of fields. This chapter will provide ideas for the use of the audible QR Code, but it is not limited to only these fields as uses can be found in more fields. This is the characteristic that makes the audible QR Code quite special as it will evolve as the need arises.

7.1 Film & music

Film and music media is perhaps the hardest medium for the audible QR Code to excel in. The nature of the code is that it is audible. This could perhaps distract viewers and listeners from the content or annoy them into not liking the product. The techniques used to hide the audible QR Code and make it less intrusive are invaluable to the functionally and practicality of the code in this format. Royalty collection and tracking is currently done through the use of companies that track the frequency of film and music plays on radio stations and other mediums. The artist will receive a set amount of money for each play of the content they own. It is essential that the tracking is done correctly and accurately so that neither the artist nor the company playing the media feel hard done. Traditional methods to track play counts was done through cue sheets that the broadcaster would send to a royalty collection society. The

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cue sheet would list all the films or music that would be broadcast on the day. The royalty collection society would then calculate the play count and reward the artist accordingly. This model does not work any more as most media content is distributed through digital mediums such as Youtube and iTunes. It is not possible to use a cue sheet to track the play count in these medium (Kobalt, 2014). Embedding a audible QR Code into media content would allow companies to track play counts on radio stations and other mediums very accurately and reduce the amount of processing required as modern systems use an auto-correlation analysis that is very processing intensive. The audible QR Code can be placed in a song recording to broadcast the artist details, the song and the current time in the song. This can be useful to users that are listening to a song over the radio or a song is playing in a film. The user will be able to identify the song and be linked to iTunes, for instance to buy the song. This can also be extended into films to track the current position in time of the film through audio as well as allowing users to identify the film through the use of the audible QR Code.

7.2 Radio

The audible QR Code excels especially in broadcasting mediums because it is able to provide a large amount of additional data in a short amount of time. The audible QR Code can be broadcast over FM and AM radio stations to enhance the listeners' experience and provide more content. The current song ID can be embedded in order to inform a listener to what is playing. Contact details of guests including cellphone number, websites and email addresses can be transmitted to users avoiding the need to repeat numbers and providing a listener with a fast way of digitally saving the data and reducing the distraction in case a listener is driving. Advertisement's can also use the audible QR Code to provide additional information to listeners about the product that would otherwise take too much time during the broadcast. The duration of the advertisement is connected to the cost and any way of reducing the time while increasing the amount of information transferred can only be beneficial to the company, radio station and the listeners. The advertisements website, product information, pricing and location can be embedded into the advertisement through the audible QR Code distribution. Emergency Alert Systems allows the government to issue warnings ranging from weather to traffic accidents. The alert systems can utilize the audible QR Code to provide a detailed description of the warning being issued. The location, severity and cause of the warning could be included while transferring information fast and effectively in order to avoid panic.

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7.3 Mobile phones

Mobile phones will essentially be the audible QR Code reader for most applications due to its mobility and access to connections such as Wi-Fi and GSM networks that links the reader to the internet. There is, however, a couple of uses that is mobile phone specific and intuitive and that allows the sharing of information between two mobile phone users.

7.3.1 Contact detail sharing

The sharing of contact details is usually an awkward interaction where cellphone numbers have to be repeated, names and surnames spelled out and email addresses deciphered. A more streamlined and natural way is handing over a business card that contains all these details. However, this is still on a piece of paper and will need to be entered into the cellphones directory, but can be done using an audible QR Code. An audible QR Code will be generated to contain the users' cellphone number, name and surname, email and all information that could be relevant. Once the user wants to share contact details all that is required is to transmit the audible QR Code and the receiver will have all the required information saved on the users phone in a way that is very quick and intuitive (fig 7.1).

Figure 7.1: Auduble QR-Code contact detail sharing

7.3.2 Broadcasting of links

Sharing of media through mobile phones is quite easy making use of various methods such as Multimedia Messaging Services (MMS), Mxit, Whatsapp and social media websites such as Facebook and Twitter. The audible QR Code is able to share links that point to media content such as videos, photos, documents and recordings from various websites such as Youtube, Soundcloud and Instagram. The audible QR Code is generated to contain the URL of the media to be shared. The QR-Code is transmitted and the receiver will read the URL and link the user to the media (fig 7.2). This can be extended to any website or any content on the internet limited to the readers' capability.

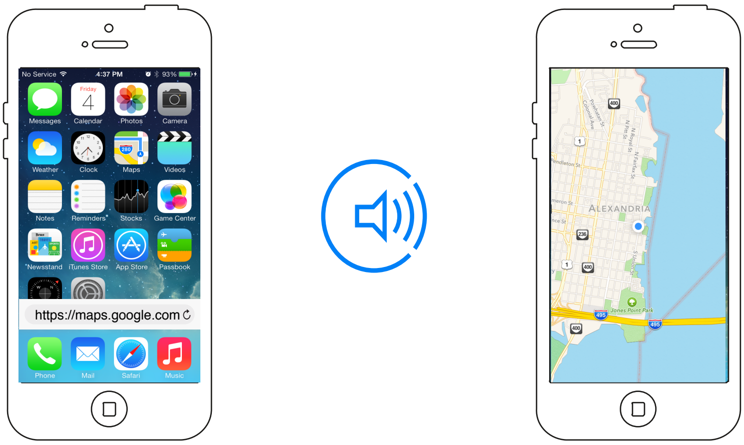

Figure 7.2: Sharing of URL through audible QR Code

7.4 Public broadcasting system

Broadcasting systems are very effective in large areas such as airports, train stations or malls. Informing large groups of people changes such as gates of flights, sales at certain shops and emergencies is essential. The audible QR Code can be transmitted over existing broadcasting systems. Additional information can be embedded into the audible QR Code to assist in the broadcast. The more dense information ensure faster transfer of information over the broadcasting systems as well as avoiding confusion due to mistaken words. An emergency broadcasting system is incorporated into all modern day media such as radios, TVs and internet connections. In the case of radio broadcasting a attention signal is transmitted at a combination of sinusoidal frequencies in order to change the channel of the radio to the emergency frequency.

7.5 Retail

The audible QR Code can be used in the same manner as the visual QR codes in a retail environment. Everything from food packaging to ticketing uses visual QR codes. However, it requires the user to engage with the code by scanning it. The audible QR Code can broadcast a certain item to audience inside a store causing their phones to alert them of the item. This causes the

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item to be pushed to the user's interest rather than asking the user to engage in the item. A store can broadcast items that are on sale, newly available, low on stock, as well as promotions to customers. The phone will scan the code and inform the user of the item, linking to additional information, the price and the ability to alert a sales clerk of you interest. This can be extended to more than just retail stores. The audible QR Code can accommodate outdoor markets promoting items, duty free shopping on air planes and restaurants with slights alterations. The main advantage that the audible QR Code has over visual QR Codes and similar methods is the ability to broadcast and push information to a customer that does not necessarily have an interest in the product. This allows retail stores to interact on a different level with their customers and will increase the quantity of sales and the quality of their service.

7.6 Indoor GPS

In chapter 4.1.2 the idea of using Bluetooth Low Enegy as a indoor positioning system is discussed. Indoor positioning is required due to the inability of Global Positioning Systems (GPS) signals to penetrate indoor environments. In large buildings such as airports, train stations and malls it helps to have a system that can guide you through the structure. Although Bluetooth Low Energy is very well-suited for this task so is the audible QR Code. Broadcasting points can be set up as landmarks indicating a reference point (van Diggelen, 2002:337-347; Van Diggelen & Abraham, 2001:1-10). Software on a smart phone will be used to interpret the different reference points and guide the user to the destination location. Throughout the journey more reference points will be broadcast to the receiver allowing the software to update the current position and calculate the direction the user must move to reach the destination (fig 7.3). It will function exactly as outdoor GPS systems however the reference point will now be calculated from a audible QR Code broadcasting loudspeaker rather than a satellite that is orbiting the earth.

7.7 Cross device input

Technology has reach the point where humans are constantly switching between devices in order to function. A email is received on a smart phone, read and replied to by composing on a computer rather than the small touch keyboard of the phone. Electronic device functions are no longer unique and uses overlap increasing productivity, but also creating a problem. User input to devices differ depending on the platform that is used. Smart phones generally use a touch keyboard while computers use a QWERTY keyboard. Text data that is contained in a message on a smart phone can only be used as input on the computer by typing the entire message on the keyboard. This is very restricting as a user would more intuitively copy a large text message and paste

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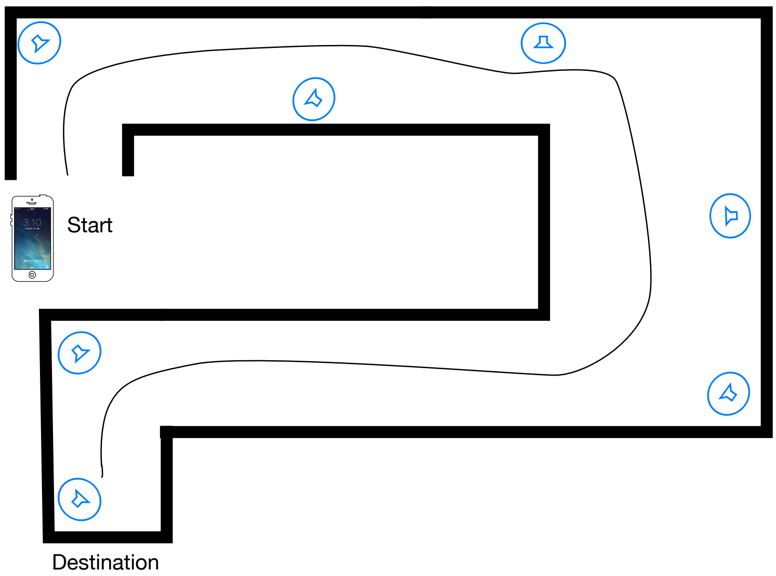

Figure 7.3: Indoor GPS application of audible QR Code

it rather than type it out. This is especially useful when a user receives a text message link to a website, but the smart phone does not support the website or requires a computer to access the link. The user is required to type out the entire message as a url in a browser on the computer. This can be time consuming and is open to errors. An audible QR Code can be used to transmit the text between the smart phone and the computer essentially copying the text and pasting it on the other device (fig 7.4). This is not restricted to computers and smart phones (Ballagas *et al.*, 2006:70-76; Johanson *et al.*, 2002:1-5). A security access point can receive an audible QR Code. This code can be generated by a token from the security system that is sent to the user's phone allowing a one-time access key that can be very long and traceable. This idea can be extended into any application that requires cross device communication of basic text at a fast and efficient rate allowing the audible QR Code to meet needs that it was not even designed for.

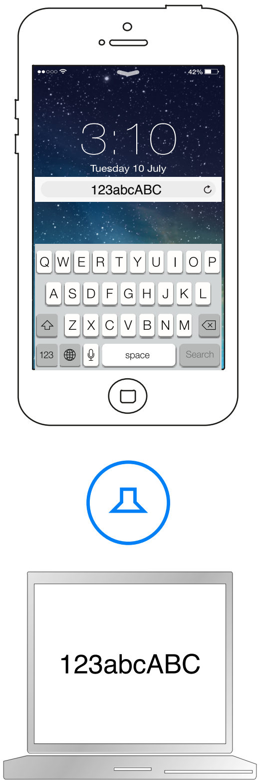

Figure 7.4: Sharing large amounts of text through the use of the audible QR Code

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Chapter 8

Conclusion

The audible QR Code will not replace the visual QR-Code, it is not intended to do so. However, the audible QR Code is able to thrive in an environment where the visual QR-Code would restrict users and frustrate them. The audible QR Code is able to broadcast data to an audience of people at a significant range enabling it to function in many more roles. The audible QR Code can function in many media applications and is able to assist users and extend the experience further than is currently possible. It is also able to compete against other technologies in fields of retail, advertisement, positioning and many more. The use of modern day communication techniques, error correction and security ensures that the code is equally safe, practical and efficient as Bluetooth Low Energy and NFC. Audible QR Code furthermore has the advantage of not requiring additional hardware other than a microphone to receive and a loudspeaker to transmit. All modern smart phones are delivered with these components as defaults. The audible QR Code uses the audible frequency bandwidth of humans to transmit the data. For some applications such as films and music this can be intrusive. The use of ultrasonic frequency bands as well as using the upper limits of the human hearing frequencies allows the code's presence to be cancelled. The test conducted shows that common smart phones such as the iPhone is able to sense the code even up to a frequency of 21 kHz. With a higher quality microphone from Sennheiser the code was detected up too 22 kHz. This is limited by the maximum sampling frequency of the iPhone's analogue to digital converters of 48 kHz. Another technique suggested the cancelling of the code by transmitting two versions out of phase in a stereo field. This was, however, unsuccessful and did not affect the code's intrusiveness. The audible QR Code is an alternative means of communicating a large amount of information fast and affective and is adaptive enough to be coupled with any other audible communication method. The audible QR Code is not intended to transmit photos, videos and audio clips it is designed to transmit the needed information to access the media content through higher performance connections such as GSM and Wi-fi just as the traditional visual QR-Code would. The audible QR Code has the potential to redefine close-

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quarter communication and if it does not it will show the potential there is to increase human interaction better than ever before.

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Appendices

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Appendix A

QR Code Patent

QR Code usage licence:

> *The use of QR codes is free of any license. The QR code is clearly defined and published as an ISO standard. Denso Wave owns the patent rights on QR codes, but has chosen not to exercise them. In the USA, the granted QR code patent is US 5726435, and in Japan JP 2938338. The European Patent Office granted patent "EPO 0672994". to Denso Wave, which was then validated into French, UK, and German patents, all of which are still in force as of November 2011. The word QR code itself is a registered trademark of Denso Wave Incorporated. In UK, the trademark is registered as E921775, the word "QR Code", with a filing date of 03/09/1998. The UK version of the trademark is based on the Kabushiki Kaisha Denso (DENSO CORPORATION) trademark, filed as Trademark 000921775, the word"QR Code", on 03/09/1998 and registered on 6/12/1999 with the European Union OHIM (Office for Harmonization in the Internal Market). The U.S. Trademark for the word "QR Code" is Trademark 2435991 and was filed on 29 September 1998 with an amended registration date of 13 March 2001, assigned to Denso Corporation.*

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Appendix B

MatLab Code

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%Neethling McGrath 15707059
%%ASK -Modulation Using perfect line code
%%MPhil Music Technology
%%2014
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%



binarysequance= [1 0 1 1 1 0 0 1 1 0 1 1 0 0 1 0 1] ;
n = length(binarysequance);

fs = 48000 ; % sample freqs
f = 22000 ; % Message freq
Tb = 1/1 ; % Set Data Rate bits/s
Rb= 1/Tb ;

t = 0:1/fs:n*Tb;

%linecode = zeros(1,n*Tb*fs+1) ;

for i = 1:n-1


    for j = 1:(fs/Rb)

     linecode(1,1+j+(i*(fs/Rb))) = binarysequance(i);

    end

end
```

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```
sinsampled = sin(2*pi*(f)*t);
ASK_signal = linecode.*sinsampled;

f_ASK = abs(fft(ASK_signal)) ;

subplot(4,1,1)
plot(t,linecode)
grid on ;
title('Binary Baseband values')  % title
subplot(4,1,2)
plot(t,sinsampled)
grid on ;
title('Carrier Signal')  % title
subplot(4,1,3)
plot(t,ASK_signal)
grid on ;
title('ASK 2-ary Modulated Signal')  % title
subplot(4,1,4)
plot(t/(n*Tb),f_ASK);
grid on ;
title('Frequency Spectrum of ASK Signal')  % title

sound(ASK_signal,fs);
wavwrite(ASK_signal,fs,'ASK_Test');


%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%Neethling McGrath 15707059
%%Binary-FSK -Modulation Using perfect line code
%%MPhil Music Technology
%%2014
%%M = 2
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%



binarysequance= [1 0 1 0 1 0 1 0 1 0 1 0 0] ;
n = length(binarysequance);

fs = 48000 ; % sample freqs
fc = 21000 ; % Centre Freq
fw = 1000; % frequency spread
Tb = 1/1 ; % Set Data Rate bits/s
Rb= 1/Tb ;

t = 0:1/fs:n*Tb;
```

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```
for x = 1:n

    if (binarysequance(x) == 0) binarysequance_polar(x) = -1 ;
    else binarysequance_polar(x) = 1 ;
end
end



for i = 1:n-1


    for j = 1:(fs/Rb)

     linecode_polar(1,1+j+(i*(fs/Rb))) = binarysequance_polar(i);

    end

end

f1 = fc - fw ;
f2 = fc + fw ;



sinsampled1 = sin(2*pi*(f1)*t);
sinsampled2 = sin(2*pi*(f2)*t);

FSK_signal = sin((2*pi*t*fc)+(2*pi*t*fw.*linecode_polar));
F_FSK = abs(fft(FSK_signal)) ;

subplot(5,1,1)
plot(t,linecode_polar)
grid on ;
title('Polar line coding')  % title
subplot(5,1,2)
plot(t,sinsampled1)
grid on ;
title('Carrier signal 1')  % title
subplot(5,1,3)
plot(t,sinsampled2)
grid on ;
title('Carrier signal 2')  % title
subplot(5,1,4)
plot(t,FSK_signal)
grid on ;
title('FSK Signal')  % title
```

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```
subplot(5,1,5)
plot(t/(Tb*n),F_FSK)
grid on ;
title('Frequency spectrum of FSK signal')  % title

sound(FSK_signal,fs);
wavwrite(FSK_signal,fs,'FSK_Test');
```

```
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%Neethling McGrath 15707059
%%Binary-FSK -Modulation Using perfect line code
%%MPhil Music Technology
%%2014
%%M = 8 meaning three bit representation
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%



binarysequance= [1 0 1 0 1 0 1 0 1 0 1 0 1 1 1 0 0 0 1 1 0 0] ;
n = length(binarysequance);

for y=1:n/3


   three_arry(y) = binarysequance(y*3)*1 ;
   three_arry(y) = three_arry(y) + binarysequance(y*3-1)*2 ;
   three_arry(y) = three_arry(y) + binarysequance(y*3-2)*4 ;

   end


fs = 44100 ; % sample freqs
fc = 100 ; % Centre Freq
fw = 10 ; % frequency spread
Tb = (1/50) ; % Set Data Rate bits/s
Rb= 1/Tb ;

t = 0:1/fs:n*Tb/3;

for x = 1:n/3

    three_arry(x) = three_arry(x) -4  ;

end
```

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```
for i = 1:(n/3)


    for j = 1:(fs/Rb)

     linecode_mary(1,1+j+((i-1)*(fs/Rb))) = three_arry(i);

    end

end

%f1 = fc - fw ;
%f2 = fc + fw ;



%sinsampled1 = sin(2*pi*(f1)*t);
%sinsampled2 = sin(2*pi*(f2)*t);

MFSK_signal = sin((2*pi*t*fc)+(2*pi*t*fw.*linecode_mary));
F_MFSK = abs(fft(MFSK_signal)) ;
subplot(3,1,1)
plot(t,linecode_mary)
grid on ;
title('Three bit code word line code')  % title
subplot(3,1,2)
plot(t,MFSK_signal)
grid on ;
title('MFSK Signal')  % title
subplot(3,1,3)
plot(t/(n*Tb),F_MFSK)
grid on ;
title('Frequency spectrum of MFSK signal')  % title

%sound(MFSK_signal,fs);
%wavwrite(MFSK_signal,fs,'MFSK_Test');

%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
%%Neethling McGrath 15707059
%%Binary-FSK -Modulation Using perfect line code
%%MPhil Music Technology
%%2014
%%M = 8 meaning three bit representation
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%



binarysequance= [1 0 1 1 1 0 0 1 0 1 0 1 1 1 1 0 1 1 1 1 0] ;
```

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```
n = length(binarysequance);

for y=1:n/3


   three_arry(y) = binarysequance(y*3)*1 ;
   three_arry(y) = three_arry(y) + binarysequance(y*3-1)*2 ;
   three_arry(y) = three_arry(y) + binarysequance(y*3-2)*4 ;

   end


fs = 48000 ; % sample freqs
fc = 22000 ; % Centre Freq
fw = 1 ; % frequency spread
Tb = (1/1) ; % Set Data Rate bits/s
Rb= 1/Tb ;

t = 0:1/fs:n*Tb/3;

for x = 1:n/3

    three_arry(x) = three_arry(x)   ;

end




for i = 1:(n/3)


    for j = 1:(fs/Rb)

     linecode_mary(1,1+j+((i-1)*(fs/Rb))) = three_arry(i);

    end

end

%f1 = fc - fw ;
%f2 = fc + fw ;



sinsampled1 = sin(2*pi*(fc)*t);
%sinsampled2 = sin(2*pi*(f2)*t);
```

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```
PFSK_signal = sin((2*pi*t*fc)+((2/8)*pi.*linecode_mary));
F_MPSK = abs(fft(PFSK_signal)) ;
subplot(3,1,1)
plot(t,linecode_mary)
grid on ;
title('Three bit code word line code')  % title
subplot(3,1,2)
plot(t,PFSK_signal)
grid on ;
title('MPSK Signal')  % title
subplot(3,1,3)
plot(t/(n*Tb),F_MPSK)
grid on ;
title('Frequency spectrum of MPSK signal')  % title

sound(PFSK_signal,fs);
wavwrite(PFSK_signal,fs,'MPSK_Test');
```

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