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Design and implementation of a security communication system for a local area: case study the university of bamenda

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***Abstract**:* The expensive, inefficient and unreliable nature of the existing communication systems, makes them inadequate for use in raising security awareness in local areas such as the University of Bamenda campus. With the current sociopolitical crisis in the region, there is a need for a reliable communication system which can be used to improve upon the security of staff and students on campus. This research project proposes a solution which is cost, time and power efficient. This system was analyzed and designed by following an evolutionary prototyping life cycle model which consists of developing a robust prototype in a structured manner which is constantly refined with user feedback to get a better end product. Mathematical analysis of the system components was used to determine the system specifications for the project. This solution has a low power requirement and does not depend on mobile networks. Simulation results showed our system is capable of receiving and transmitting information without any degradation of the signal within a 100m range when supplied with a 9V dc supply.

Keywords: *Communication, Frequency. Modulation, Signal, Distance, Power*

1. INTRODUCTION

In recent years, advances in technology have led to security systems evolving from simple control panels and locks into high-tech gadgets. Today, improvement of security systems is as a result of the technological developments over the past few years. Security and monitoring systems are generally developed to detect intrusion and reduce the rate of perverse practices such as crime, aggression and theft.

Security systems generally work by placing sensors at critical points in the area to be surveyed. These sensors communicate with a central panel located in a monitoring room. Different types of sensors are employed for security: - surveillance cameras are used for video recording of scenes, sound sensors for tracking sound and motion sensors for detecting movements. Some systems use alarms to achieve a greater level of security. The alarm can be triggered by the presence of an intruder.

Establishing a communication network in places that require attention, observation or awareness is a means of tracking an entity, event or process under observation. Audio event detection is a tool which can be used to monitor an area and create awareness. The audio surveillance system presented in this study can be used in security because it monitors life audio information of events in an area. Transfer of audio information over the proposed communication system is a way to address people in different locations.

2. Background

2.1. Communication

The use of technology in communication has become indispensable in today's society. Emerging technologies have given rise to numerous communication tools that have been employed by people to pass on information. In this light, various approaches have been used to address people, provide security, awareness as well as human tracking. Such methods include video surveillance, audio surveillance, motion detection, face recognition and tracking. Technologies adopted and previous works on these methods have been highlighted. They include; mobile and fixed phones, tablets, internet, radio and television sets which are but a few to be stated.

A. Use of Mobile Phones

Mobile phones are being utilized in different sectors as a channel to convey information. Through text messaging, information about vaccination campaigns for infectious diseases can be disseminated (FEANTSA, 2013). For health issues, mobile phones provide a means to contact medical personnel in a timely and efficient way thereby enhancing medical follow up of patients. In companies and enterprises, fixed telephones have been used to organize meetings, to request for services and many other administrative tasks (FEANTSA, 2013). This has the overall effect of speeding up the rate at which tasks are performed plus optimizing the performance of the working personnel. Thus in a nutshell, individuals deploy phones on daily basis to keep in touch with family members, friends, social workers and to get the hereabout from others.

B. Use of Internet

Today, social media serves as the main channel through which information reaches people. This has been made because of the internet. The internet has bridged the geographical gap between people because of the rate and ease at which information flows over the internet (FEANTSA, 2013). As such, Internet services such as emails, VOIP, video calls and other multimedia services facilitate the way through which people are communicating (Ramey, 2013).

C. Use of Radio and Television

From the discovery of radio waves (Haykin, 2007), there have been tremendous advancements in the field of radio communication. The first solution adopted to address the general public was through radio broadcasting (Haykin, 2007). Today, radio sets allow the reception of information broadcasted from different radio stations. Government officials in Cameroon utilize the national radio station to address the population; for instance, during the 8pm news. Radio stations also serve as a platform for diffusing a communique, advertising products, conveying announcements and sensitizing the public on security, health, education and social aspects.

2.2. Security

Traditionally, security systems are based on video and audio surveillance. Today, video surveillance has a vast domain of applications such as in banks, offices, companies and public locations with the aim of ensuring security by the use of video cameras which provide video records of all ongoing activities in these areas. Audio surveillance uses sound from the surrounding to determine the type of event in an area.

A. Video Surveillance

Video surveillance generally involves the use of cameras for recording video scenes within an area. Closed circuit television (CCTV) is a surveillance technology in which video cameras are connected in closed loop and the images collected from these cameras are sent to a control room for further analysis by the operating personnel (Ratcliffe, 2011). Installation of CCTV systems in public areas such as streets, banks, offices is a means to reduce crime, aggression and theft in such areas. The police can make use of recorded videos from CCTV cameras in order to track suspicious individuals who are an offence to the population. CCTV system also optimize security since CCTV operators can contact security services in case people in street are victims of an aggression or any form of accident which necessitate immediate intervention. Thus, this semi-covert CCTV camera has a crime prevention advantage over an overt system because offenders can never be sure in which direction the camera is facing.

Video surveillance systems rely on the detection of activities captured by static cameras placed at key positions and providing a view of all the ongoing activities within the area of interest. A video surveillance system designed to provide security in an institution is presented in (Murungi, 2009). The design is software-based; the proposed system is modelled using video simulation software known as video CAD – Computer Aided Design. Images obtained from simulations showed that the system can effectively display movement of persons within the upper and lower ground floors. However, these results from simulation don't very much guarantee that the system will achieve the same performance when implemented practically. Researched such as (Vandit, Ayesha & Yash, 2018) used Histogram of

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Oriented Gradients features theory of Visual Saliency and the saliency prediction model Deep Multi-Level Network to detect human beings in video sequences. Again, this approach is software based with a detection precision of 83.11%.

B. Audio Surveillance

Audio surveillance is based on monitoring an area by collecting and analyzing acoustic information in other to determine the type of events taking place in that area. This section presents related works conducted in the domain of audio surveillance. It explores the different approaches, methods and models that are conceived for the design of an audio event detection system. There are two approaches; the first approach involves the detection (capturing) and classification of sound events that are indicative of the occurrence of abnormal situations indicating threat for example screaming, explosion, guns hot and so on (Uzkent & Barkana, 2011). This approach is software-based characterized by modeling and simulation using appropriate software, MATLAB (Gupta, Bansal & Choudhary, 2017, p. 197). The second approach is the continuous monitoring and tracking audio information in an area (Oguche, Agber &Tarkaa, 2017, p. 24). The second approach usually implicates the design and implementation of a system capable of providing acoustic information of a locality under observation.

Efficient signal processing techniques and algorithms have been developed over the years to detect abnormal events such as gunshots, screams, glass break and explosion (Ntalampiras, Pottammitis & Fakotakis, 2009). Classical audio event detection is based on two fundamental operations. The features extraction and classification/pattern recognition (Uzkent & Barkana, 2011). Feature extraction also called parametrization of the audio signal, involves transforming the waveform of the audio signal into a series of vectors parameters or acoustic features. These features characterize properties associated with a particular type of sound which are used in pattern recognition to identify the audio signal. Examples of audio feature set include MFCC, PSD and LPCC (Uzkent & Barkana, 2011). More details about audio features set can be found in (Gupta, Bansal & Choudhary, 2017, p. 197). The aim of pattern recognition is to classify (identify) a particular class of sound using the features set previously discussed. Several pattern recognition methods exist, for example we have HMM, GMM, NN, and SVM.

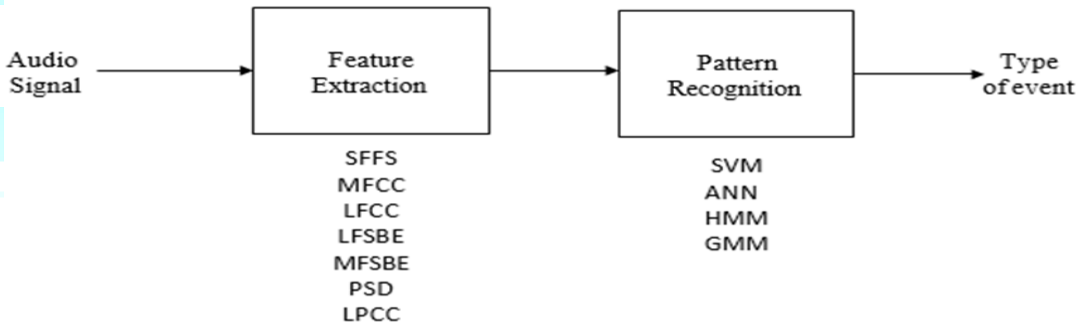

Figure 1: Overview of an Audio Surveillance System (Uzkent & Barkana, 2011)

A system capable of detecting four classes of alarming events (human voice, gunshot, chainsaw or tractors) in a wild environment is discussed in (Oltean, Grama et al. 2015). The system has a modular structure with two modules: -The first is the feature extraction block/module and event classification module. The feature extraction block/module computes mel-frequency coefficients (mcc) and energy of discrete wavelength transform. The second is an event detection module comprised of two artificial neural networks to detect the occurrence of an alarming event and to determine the type of abnormal event (classification). The system has high computational complexity because of the large number of audio features set used and still achieves a high correct recognition rate of 99.5%.

An abnormal sound recognition system for monitoring indoor sounds is presented in (Chang & Chang, 2013). It uses sequential floating forward selection (SFFS) to select high discriminative features. Support vector machine (SVM) was finally used to classify the sounds into six categories (screaming, infants crying, coughing, glass breaking, laughing, doorbell ringing) (Chang & Chang, 2013). Experimental results showed

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that the methods adopted yielded an accuracy rate of 86% compared to 99.5% obtained in (Oltean, Grama et al. 2015).

Another study used a different extraction technique called pitch-range feature extraction along with SVM method to detect three class of abnormal events (glass breaking, gunshot and dog barking) (Uzkent & Barkana, 2011). Correct recognition rates were obtained in the range 79 - 92%.

In (Ntalampiras, Pottammitis & Fakotakis, 2009), mel-frequency cepstral coefficients(MFFC) features and Hidden Markov Model (HMM) were used to detect and classify three classes of abnormal events such as explosion, gunshot and screams in a noisy environment. The results obtained a correct classification rate of 93.3%.

A previous study deployed LPC, MFCC, energy and perceptual linear prediction coefficient (PLPC) as audio features parameters while Gaussian Mixture Models (GMM) and Support Vector Machine (SVM) methods were executed to achieve a correct recognition rate of 75 to 98% for shout detection and non shout detection in public transport vehicle (Rouas, Louradour & Ambellouis, 2006).

Work such as (Grezl & Cernocky, 2009, p. 671-675), provides an efficient framework for analyzing general audio surveillance of public transportation area such as a metro system. This project involves live recordings of audio signals captured by microphones and classifying the audio events based on the supervised training approach. The results obtained for detecting normal events from unexpected sounds shows that the system can be used for several applications.

2.3. People Tracking

A lot of advancements in the field of population tracking has been made in the past years. Most of these approaches make use of algorithms which extract features from the face of persons to identify them.

2.3.1. Face detection and recognition

Human identification can be done by extracting and classifying the biometric features such as face, fingerprints, ear, iris, palm, gait or speech and all of these biometric features are either used separately or combined together depending on the security application (Binu, Jacob et al. 2011). The following are research studies on tracking people through face recognition and detection.

A multistage approach comprised of HOG classifier with a K-mean clustering technique which are used for face detection and tracking of people is proposed in (Etienne & Francois, 2006).

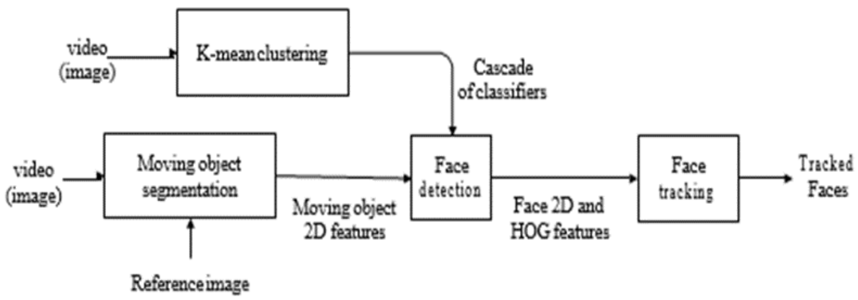

Figure 2: Overview of the face tracking algorithm structure (Etienne & Francois, 2006).

The proposed face detection algorithm when evaluated with standard algorithms show similar performance with limited cost. Such an algorithm is implemented to identify people by determining the distinct face of each individual.

Work as (Mykhalo, Stefan & Bernt, 2008), was previously conducted to detect and track multi people in clustered scenes. The proposed method first compares the detection performance of the single- frame person detector with the tracklet-based detector and then evaluate the tracks produced. Results from this system outperforms the state-of-the-art of on single frame detection.

Figure 3: Examples of detection and tracking of specific persons in image sequences of crowded street scenes (Mykhalo, Stefan & Bernt, 2008)

An approach to detect and recognize human faces was conducted in (Bhavani et al, 2017, p. 1562-1565). The method consists of performing a frontal image alignment process followed by classification employing sparse representation techniques. Such a system can be used on school campus for surveillance. It can be used to track and find the location of a person on campus. The approach can be further enhanced for an automated attendance tracking.

In (Michalis, 2013), a framework to track and identify multiple people in a crowded scene captured by cameras has been designed. A people detector is first employed to estimate the position of individuals. These positon estimates are used by face detector to search face locations and identify people in the scene. Models of the face based on Local Binary Histograms of various people are captured offline and stored in a database. An SVM classifier is built from those models and is able to recognize query faces with great variation in facial expression, pose and illumination.

2.3.2. Motion Detection

People can also be tracked by putting in place systems having the capability of detecting human motion. Much work has been done on the development of tracking algorithms. Some systems use a single camera to detect human motion and tracking overtime (Pavithra et al., 2016). The video camera is placed at fixed places where security is needed. By the time the human movement is captured by the camera, it is immediately detected and the object is tracked by background suppression method. Whenever the human object is detected, information is processed and this makes the system to awake and produce an alert sound. MATLAB 7.14 (R2012A) is the main software tool used for simulation test and results.

In (Nizar, Ziad & Rada, 2008), a real time system for human detection, tracking and motion analysis is developed. The system detects and monitors people in both indoor and outdoor environments. The detection and tracking is performed in two stages; the background model is used to get background pixels by using the background subtraction method, noise cleaning and object detection are applied followed by human modelling to recognize and monitor human activity such as human walking and running. The system was designed using C# Microsoft.Net software tool. For indoor scenes, the percentage of tracking is 96% for one, two and three persons. For outdoor scenes, the percentage of tracking is 87% for one, two and three persons in the scene.

Pyroelectric infrared sensors (PIR) perceive the heat from the human body to determine the presence of humans in an environment. In (Jaeseok & Sang, 2014), a study on human movement detection and identification using PIR sensors is presented. Three PIR-sensor modules were placed in a hallway for monitoring people; one module on the ceiling and the other two modules on opposite walls facing each other. Results obtained showed that raw data set captured from a PIR sensor of each module achieved more than 94%

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accuracy was achieved. This efficiency was recorded in classifying the direction, speed and distance in identifying subjects using reduced future set extraction from two pairs of PIR sensors of each of the three modules.

From our literature survey, it is observed that most previous research works in the context (security, awareness, tracking of population) of this study are based on developing algorithms, modeling and designing framework using software tools. The experimental results obtained are software based and still have to be implemented in applications where surveillance is necessary. Little has been done on implementing monitoring systems capable of tracking real life events of an area.

A burglar alarm system (Ahmed, Mohammed & Agbo, 2006, p. 97-102) is activated (sends a siren) whenever the sound from the sensor is above the threshold intensity. This system can easily be triggered by any type of sound whose intensity is above threshold, thereby making the system vulnerable.

A surveillance system (Oguche,Agber &Tarkaa, 2017, p. 24) designed to transmit and retrieve recorded information by frequency modulation using bipolar junction transistors at the transmitting and receiving end. Because of the low transmitting power (0.3watts), the operating range (distance) of this system is limited to 12m. As a result of this, the system can't be deployed in a vast environment like public places and school campus.

A wireless communication system is designed and implemented for the transmission of video and voice information over wired and wireless means (Saha et al., 2016. p. 161-172). The proposed system utilizes LAN, Wi-Fi and GSM network for the transmission of both video and audio data. Although the system provides secured and reliable wireless communication, the transmission and reception modules employed as hardware components made the designed system costly. Also, the cost of running the system through GSM network is expensive, thereby increasing the OPEX.

3. Problem Statement

Different communication means exist to connect people within an area. Therefore, the motivation of this project is the need to provide a cost efficient solution which is a daunting challenge due to the problems associated with the means available to transmit information. The available means thus far require mobile network, constant power supply and internet access for their operability.

Unpredicted circumstances such as theft, aggression, crime occur quite often and precautionary measures need to be taken in order to raise awareness of the occurrence of such events followed by taking appropriate action to put an end to them. Conventionally, surveillance systems usually operate based on video recording of the scene that occurred in the area of interest. Nevertheless, video surveillance has some inherent limitations. In situations of obscured vision (for example a foggy weather) taking video might be impossible (Baiying & Man-Wai, 2015) since video based surveillance makes use of fixed camera with limited coverage area which limits the area under survey (Baiying & Man-Wai, 2015). Thus, it is challenging to deploy video cameras in areas where accessibility is limited or restricted. Audio surveillance on the other hand is suitable for cloudy weather scenarios since it needs just the acoustic information conveyed by soundwaves, sound sensors can capture audio information that may be difficult or impossible to obtain by any other means (Ntalampiras, 2013, p.191-205) and these sensors can be installed effortlessly in any position thereby increasing the monitoring area.

Some other hindrances to people accessing the current communication systems in place include:

- *Lack of communication credit:* Meanwhile the issue of communication credit has been dealt with in many developed countries, it is still a major issue in developing countries. This is due to the fact that many people are living in deplorable conditions and cannot spare money for communication credit. This becomes a big impediment to communication as it is impossible to communicate without communication airtime. Also, credit is needed to purchase internet bundles.
- *Epileptic power supply:* The power supply in Cameroon is far from being stable especially in the Bambili area where the University of Bamenda is located. Some of these power outages are so lengthy that even the backup power supply to the telecommunications base stations runs out and communication becomes

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totally impossible. At such a time, if there is any security problem, it is impossible to alert anyone to come to the rescue.

- Signal strength: The signal strength, otherwise known as reception or service refers to the transmitter power output as received by a reference antenna at a distance from the transmitting antenna. Ideally, for communication this strength should be greater than -58dBm. Less than that, signal gets poorer and at -96dBm, there is no signal at all. In Bambili, the signal strength constantly fluctuates between -65dBm to -90dBm which is not a good signal strength for effective communication.

4. Aims and Objectives

The main aim of this research project is to design a security communication system which is more cost and power efficient than the existing system, which can serve as a communication system for security purposes based on the internet of things technology (IoT). So, taking into cognizance the fact that world is more or less a global village as the internet has brought people very close to each other virtually despite physical distance while communication systems are becoming more sophisticated and more efficient with technological growth.

As such, the objectives of this work are to:

- Identify the requirements for the communication system
- Identify components that can be used to build this system
- Analyze the various components to know the parameters that each requires to obtain optimum efficiency of the system
- Run a simulation of the proposed system to predict the efficiency of the prototype before building
- Build and install a hardware prototype

5. Research Questions

Therefore, in order to effectively carry out our research, the following major questions have been used.

- How can these technologies be used in developing a cost efficient communication system to improve on the security of persons on the University of Bamenda campus?
- How can we create a communication system that can be used within a local area like the university of Bamenda campus?
- How will this system mitigate the problems cited in the problem statement?
- How accessible is the system?
- How time efficient is the system?
- How will feedback be gotten from the system?

6. DESIGN AND IMPLEMENTATION OF PROPOSED AUDIO SURVEILLANCE SYSTEM

6.1. General Overview of the System

The proposed system provides audio surveillance of a monitored area. A couple of transmitter/receiver blocks had been installed in different locations to capture human voice discussions at any outer position under control by a sound transmitter. This system can be used to provide home security. The transmitter/receiver blocks can be installed or kept mobile within the locality. Some key particularities of this system are no communication cost, low energy consumption and standalone supply source which implies that it cannot be cut off as a result of electricity shortage generally being experienced in developing countries. It is worth noting that the free communication aspect is crucial in developing countries where the standard of living does not warrant majority of the population to have in several situations a communication airtime to call or alert. Also, in developing countries, national security scarcely finds its ways due to poor or lack of street localization

6.1.1. Choice of Life Cycle Model

A life cycle model is an abstract representation of a structured set of activities required for software development (François, 2009). A wide range of them exist, differing in the size of the team involved in the project, the needs of the client, the time allocated and the allocated budget.

However, all are structured to allow the production of quality software consistent with the original specifications. Between specification, design, implementation, validation, improvement or maintenance, the process models aim to increase the product development team's efficiency. Therefore, success of a project depends on various factors such as the composition of the team involved in the development, the time allocated, the budget delivery constraints and technical skills are crucial in determining the choice of the process model to adopt. So for the success of this project, we have deployed the *evolutionary prototyping methodology for its numerous advantages and also due to the fact that at any stage, feedback from the users of the system could be used to enhance the system until we have a final system with all the necessary features*

6.1.2. Description of Life Cycle Model

Prototype is a working model of software with some limited functionality. Evolutionary prototyping consists of building a prototype that is an incomplete version of the software program being developed which is then refined based on the regular feedback from the potential users. With this methodology, the system is not build in isolation but with the participation of the potential users of the platform (Ince and Hekmatpour, 1987). Here, maintainability of the code, style, design patterns or testing count from the beginning, which makes it possible to evolve the prototype into a fully featured, enterprise-grade product. It is illustrated in Figure 4 below.

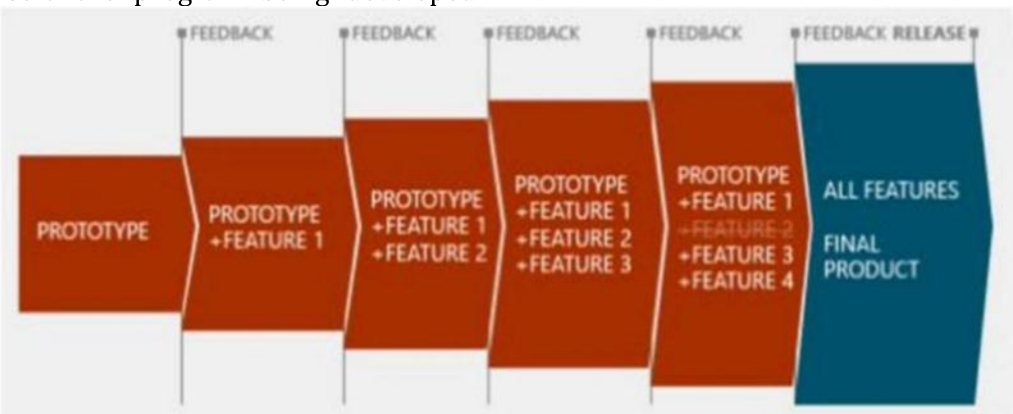

Figure 4: Features aggregated to the prototype to build the final product (O'Connor, 2017)

6.1.3. 6.1.3. Life Cycle Model Steps

Step 1: Identification of Basic Requirements

This step involves understanding the basic product requirements especially in terms of user interface. The more intricate details of the internal design and external aspects like performance and security can be ignored at this stage.

Step 2: Developing the Initial Prototype

The initial Prototype is developed in this stage, where the very basic requirements are showcased and user interfaces are provided. These features may not exactly work in the same manner internally in the actual system developed.

Step 3: Review of the Prototype

The prototype developed is then presented to users and supervisors. The feedback is collected in an organized manner and used for further enhancements of the system under development.

Step 4: Revise and Enhance the Prototype

The feedback and the review comments are discussed during this stage. The changes accepted are again incorporated in the new prototype developed and the cycle repeats until the customers' expectations are met.

The earlier steps of the prototype contain only the core part of the future product, and then, features are added progressively. It is illustrated in Figures 5 and 6 respectively below

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Figure 5: Evolutionary prototyping detailed steps

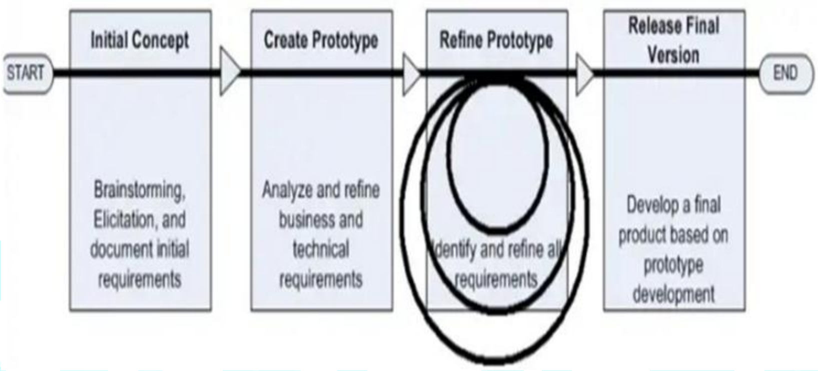

Figure 6: Evolutional prototyping model steps (O'Connor, 2017)

The main goal when using evolutionary prototyping is to build a very robust prototype in a structured manner and constantly refine it. Another reason for this methodology is that the evolutionary prototype, when built, forms the heart of the new system, and the improvements and further requirements will then be built. One of the main advantages of this methodology is that it helps to develop parts of the system that we understand instead of working on developing the whole system.

6.1.4. System Analysis

The term system is derived from the Greek word system, which means an organized relationship among functioning units or components. The term analysis refers to breaking a whole into its parts with the intent of understanding the parts' nature, function, and interrelationships. Analysis focuses on capturing the business requirements for the system by identifying the "what" of the system, and it leads directly into the design phase, during which the "how" of the system is determined. The system design consists mainly on the design and test of PLL based Audio FM communication system and the design and test of the low power audio amplifier for the end user listening, the use of the PLL FM communication system in local area network implementation as well as in addressing planification is also projected.

The mathematical analysis is required for the choice of discrete components needed to realize the transmitter and receiver circuits.

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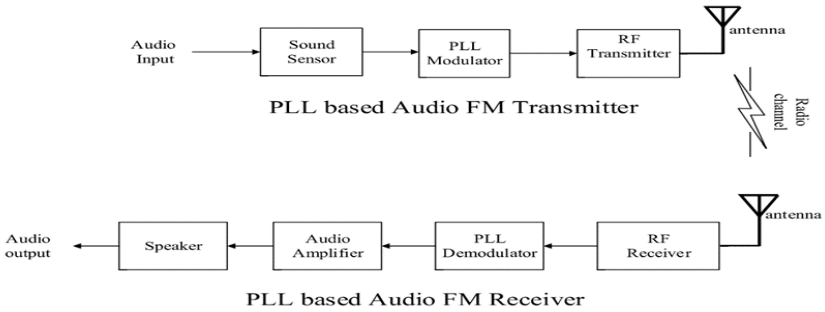

Figure 7: RF communication system block diagram

At the transmitting end, the sound sensor captures and converts sound into electrical signals. The PLL modulator is used to modulate the audio signal into a PLL square wave carrier signal. The RF Transmitter takes care of the second modulation and transmission of the PLL digital signal by RF propagation through the antenna. The RF Receiver picks up the RF transmitted signal, converts it into a modulated PLL square wave signal at the receiving end. The PLL demodulator removes the PLL square wave signal to produce the original sound signal. The audio Amplifier is used for sound amplification and provides the necessary output power to drive the speaker. The speaker finally converts back the electrical sound signal into audio for the end users.

This system can be used as an alternative to cell phones for communication; firstly, because it does not use network to function since transmission is by RF. Secondly, it is impossible to make phone calls during mobile network disturbances, in such circumstances, the proposed system is appropriate since it does not need any airtime to operate.

6.2. Description of materials

A. Sound Sensor

The sound sensor module detects the presence of sound and produces an output voltage which is proportional to the intensity (level) of sound perceived. The sound sensor is used to detect sound from the surrounding and send it into the system in the form of electrical signal.

B. PLL 4046 IC

The PLL 4046 IC is used to modulate and demodulate the audio signal at the transmitter and receiver respectively. The internal structure of PLL4046 IC consists of two phase comparators, VCO and source follower (Morgan, 2003). PLL technique is applied to generate the carrier frequency at the transmitting and receiving ends.

Figure 8: FM Modulation by PLL

A voltage message signal, m(t) is applied to the control voltage of the VCO and the output FM signal, $X_{FM}(t)$ is a constant amplitude sinusoidal carrier wave whose frequency is a linear function of its control voltage. The audio signal here is represented by the message signal, m(t).

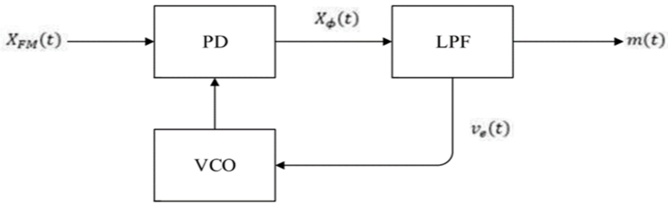

Figure 9: FM demodulation by PLL

The phase detector (PD) measures the phase difference, $X_{\Phi}(t)$, between the incoming FM signal and the frequency (f_{vco}) generated by the voltage control oscillator (VCO). The low pass filter (LPF) generates an error voltage signal $v_e(t)$ that is proportional to, $X_{\Phi}(t)$. The $v_e(t)$ signal is used to adjust (tune) f_{vco} close to $X_{FM}(t)$. This process continuous until the loop attains the lock condition, that is when the two input signals of the PD are in phase. The demodulated signal m(t) is then obtained. The LPF is obtained by connecting external components to the PLL 4046 IC.

C. RF TX-RX Module

The RF transmitter XD-FST module (ACOPTEX, 2017) is used for RF transmission at 433MHz. It operates in the 3V-12V supply, making it ideal for low battery powered applications. The communication range varies and depends on how much voltage is supplied to the transmitter module, RF noise in the environment and the type of antenna used. The RF receiver XD-RF module (ACOPTEX, 2017) has an

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operating frequency range 315/330/433MHz with a 5V supply. Quality of signal received depends on the type and size of antenna employed.

D. LM386 low power audio amplifier

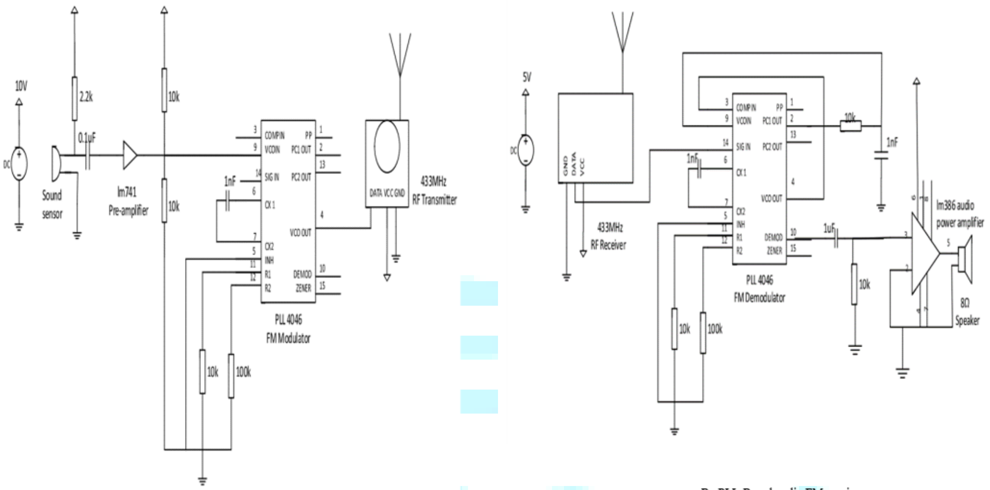
The LM386 is a power amplifier (Manoha, 2018). It is an operational amplifier with a default gain of 20, but addition of external resistor and capacitor between pins 1 and 8 increases the gain to a range of 20 to 200. It requires few components to achieve the desired gain. The LM386 is used for amplifying the audio signal at the receiver end.

E. Speaker

The speaker is used to convert the signal from the amplifier into sound. A speaker with an impedance of 8ohm is chosen in order to obtain a desired signal output power of 0.7Watts (Manoha, 2018).

6.2.1. Design Proposal

The proposed system architecture is shown below. The proposed RF communication system comprises of two parts; PLL based audio transmitter and receiver.



A. PLL Based audio FM transmitter

B. PLL Based audio FM receiver

Figure 10: RF Communication system architecture

6.2.2. Mathematical Analysis

The design is based on evaluating the value of system components through calculations.

A. Design of PLL based Audio FM Transmitter

PLL 4046 IC design parameters such as frequency offset, carrier frequency, free running frequency need to be determined to configure the PLL4046 IC to function as an FM modulator.

I. Frequency offset fmin

In radio engineering, a frequency offset is an intentional slight shift of broadcast radio frequency to reduce interference with other transmitters, so it is necessary to include an offset frequency in this design. For the PLL CD4046 IC, the offset frequency is equal to the minimum frequency f_{min} that the VCO can generate. The minimum frequency f_{min} is determined by the value of capacitor C_1 and resistor R_2, expressed as shown below.

$$f_{min} = \frac{1}{R_2 C} \quad 3.1$$

For $f_{min} = 10kHz$, we choose $R_2 = 100k\Omega$ and $C_1 = 1nF$

II. Center frequency fc

The output frequency of VCO, f_{vco} is a function of the input control voltage v_{in} and external resistor R_1 and capacitor C_1. The mathematical expression relating VCO input voltage v_{in} and VCO output frequency f_{vco} is given by:

$$f_{vco}(v_{in}) = \frac{0.16\ x\ v_{in}}{R_1 x C_1} \quad 3.2$$

From (Morgan, 2003), VCO generates the center frequency f_c when its input voltage v_{in} is biased at half the supply voltage V_{DD}.

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That is, $f_{vco} = f_c$ at $v_{in} = {}^1/_2 V_{DD}$.

$$f_c = f_{vco}({}^1/_2 V_{DD}) = \frac{0.16 \times {}^1/_2 V_{DD}}{R_1 \times C_1} \qquad 3.3$$

with $V_{DD} = 10V, R_1 = 10K \text{ and } C_1 = 1nF$,

$$f_c = \frac{0.16 \times {}^1/_2 (10V)}{10K \times 1nF}$$

$$f_c = 80KHz$$

III. Free running frequency f0

The free running frequency, f_0 is the frequency generated by the VCO in the free running state, that is when there is no input voltage applied to the VCO. The free running frequency can be expressed as follows

$$f_0 = f_c + f_{min} \qquad 3.4$$

$$f_0 = (80 + 10)kHz$$

$$f_0 = 90kHz$$

So, a carrier signal of 90KHz will modulate the audio signal (20Hz-20KHz) from the sound sensor.

IV. Maximum frequency fmax

VCO generates maximum frequency f_{max} when VCO applied voltage v_{in} is equal to the supply voltage V_{DD}.

V. Frequency Sensitivity gain KVCO:

From (Morgan, 2003), the
That is $f_{vco} = f_{max}$ at $v_{in} = V_{DD}$

$$f_{max} = f_{vco}(V_{DD}) = \frac{0.16 \times V_{DD}}{R_1 \times C_1} \qquad 3.5$$

with $V_{DD} = 10V, R_1 = 10K \text{ and } C_1 = 1nF$,

$$f_{max} = \frac{0.16 \times 10V}{10K \times 1nF}$$

$$f_{max} = 160KHz$$

VCO gain is expressed as:

$$K_{vco} = \frac{\Delta f}{\Delta V} = \frac{f_{max} - f_{min}}{V_{DD} - V_{min}} \qquad 3.6$$

$$K_{vco} = \frac{(160 - 10)KHz}{(10 - 0)V}$$

$$K_{vco} = 15\,KHz/V$$

$K_{VCO} = 15KHz/V$ means the VCO has been designed with a tuning gain of 15kHz per Volt.

B. Design of PLL based Audio FM Receiver

I. VCO design

To demodulate the FM signal, the carrier frequencies of both modulator and demodulator should be the same. We use the same R_1, R_2 and C_1 for the second PLL demodulator IC so that the carrier frequency remains the same as that of the modulator IC. Therefore, $R_1 = 10K$, $R_2 = 100K$ and $C_1 = 1nF$

II. Low pass filter design

One of the key considerations when designing a PLL demodulator system is the low pass filter. The output of PC1 is filtered by the LPF. The purpose of the LPF is to pass the dc and low-frequency portions of v_{in} and to attenuate high frequency ac components. The LPF components are connected externally through pins 2,13 and 9.

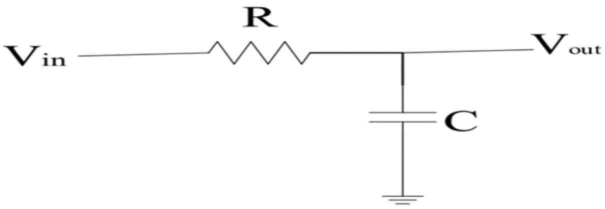

Figure 11: LPF of PLL demodulator

The capture range and cutoff frequency of the loop filter must be determined to effectively demodulate the audio signal.

A. Capture Range (CR)

The loop filter should be designed with a large capture range (CR) that it is able to follow the anticipated variations of the FM signal. The time constant of the filter should meet the range of frequencies at which the message signal is expected. In order to achieve this, the response time ($\tau = RC$) of the filter should be made sufficiently short by choosing the appropriate capacitor (C) and resistor (R) values.
The loop filter is designed with a short time response of 0.01ms ($\tau = 0.01ms$). For $\tau = RC$ = 0.01ns, we choose $R = 10k$ and $C = 1nF$
From (Morgan, 2003), the capture frequency f_c of the loop filter is given by;

$$f_c = \pm \frac{1}{2\pi}\sqrt{\frac{2\pi f_0}{\tau}} \qquad 3.7$$

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$$= \pm\frac{1}{2\pi}\sqrt{\frac{2\pi f_0}{RC}} = \frac{1}{2\pi}\sqrt{\frac{2\pi\times 90kHz}{10k\times 1nF}} = \pm 37.9kHz$$

$$\rightarrow f_c \cong 38kHz$$

f_c = ±38kHz means the FM demodulator circuit can detect any frequency with a frequency deviation of 40kHz above or below the center frequency of an incoming signal.

The Capture Range (CR) of the loop filter is therefore

$$\text{CR: } f_O - f_C < f < f_O + f_C$$

$$\text{CR: } 100KHz - 38KHz < f < 100KHz + 38KHz$$

$$\text{CR: } 62KHz < f < 138KHz$$

The FM demodulator circuit is designed with a bandwidth or capture range of CR:

$$62kHz < f < 138kHz$$

This means that the FM demodulator can successfully detect or demodulate any incoming frequency signal within this frequency interval.

B. Cut-off frequency fp

Another design issue to be taken into consideration is the cut-off frequency K of the LPF which attenuates dc components and high frequency components of phase comparator 1 (PC1) output. The cut-off frequency f_p is given by (Morgan, 2003)

$$f_p = \frac{1}{2\pi R_P C_P} \qquad 3.8$$

We use the same resistor and capacitor values used to compute the capture range (CR).

$$f_P = \frac{1}{2\pi \times 10K \times 1nF}$$

$$f_P = 15.92KHz$$

$$f_P \cong 16KHz$$

The loop filter attenuates high frequencies above 16KHz.

C. Low Power Audio Amplifier design.

The audio amplifier is designed with a gain of 20. From (Manoha, 2018), the default gain is 20 with no capacitor connected between pins 1 and 8. The 1uF capacitor and 1k resistor form a filter at the input (pin 3) to filter out high frequency components (noise) because the lm386 amplifier will amplify the input audio signal together with noise. The power amplifier will provide necessary power to draw the speaker thereby producing a loud audible sound.

6.3. Circuit and system analysis

6.3.1. Various circuit of the system

A. Modulator and demodulator circuits

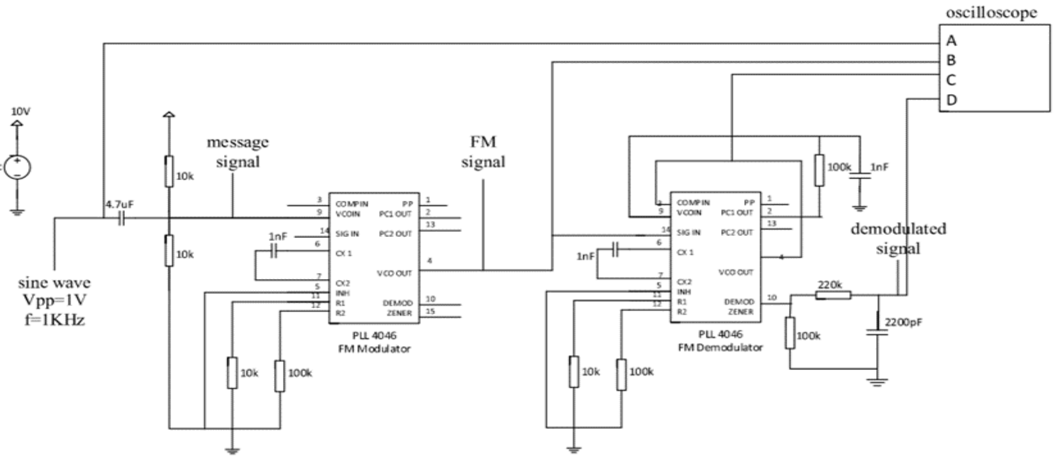

Figure 12: PLL modulator and demodulator circuits

A voltage divider network of two resistors with $R_1 = R_4 = 10k\Omega$ is used for clamping the VCO input control voltage (pin 9) at ${}^{1}/{}_{2}\,V_{DD} = 5V$. We send a message signal of frequency 1kHz and 1V peak to peak amplitude at input control voltage (pin 9) through a capacitor C_7. The VCO output (pin 4) is the FM modulated wave. Phase comparator 1, PC1 compares the phase difference between an incoming square wave signal (pin 14) of frequency 95KHz and the carrier wave signal (pin 4 to pin3) and produces an output voltage signal (pin 2). This voltage signal is then passed through the external low pass filter (pin

2). It can be noticed that the frequency of the incoming signal is 95KHz which is within the capture range CR*: 60KH z< f < 138KHz* of the loop filter. So we expect the FM Demodulator to successfully capture or synchronize with the incoming signal.

B. Audio Amplifier Circuit

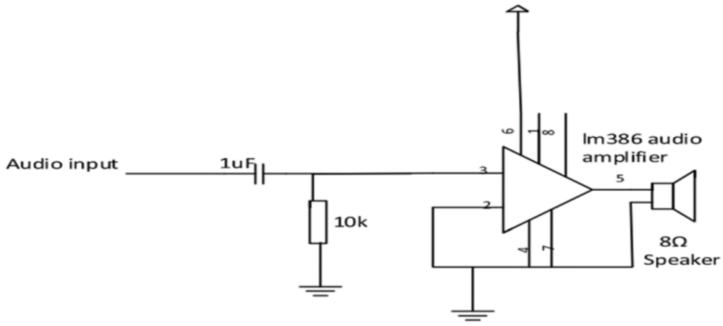

Figure 13: Audio amplifier circuit

The R, C filter at pin 3 filters noise and high order frequency components from the audio signal. The amplifier gain is 20 since an 8Ω speaker is used (Manoha, 2018).

6.3.2. System Simulation

The proposed system is simulated using Proteus 8 and the screenshots are shown below.

A. Simulation of Modulator and Demodulator Circuits

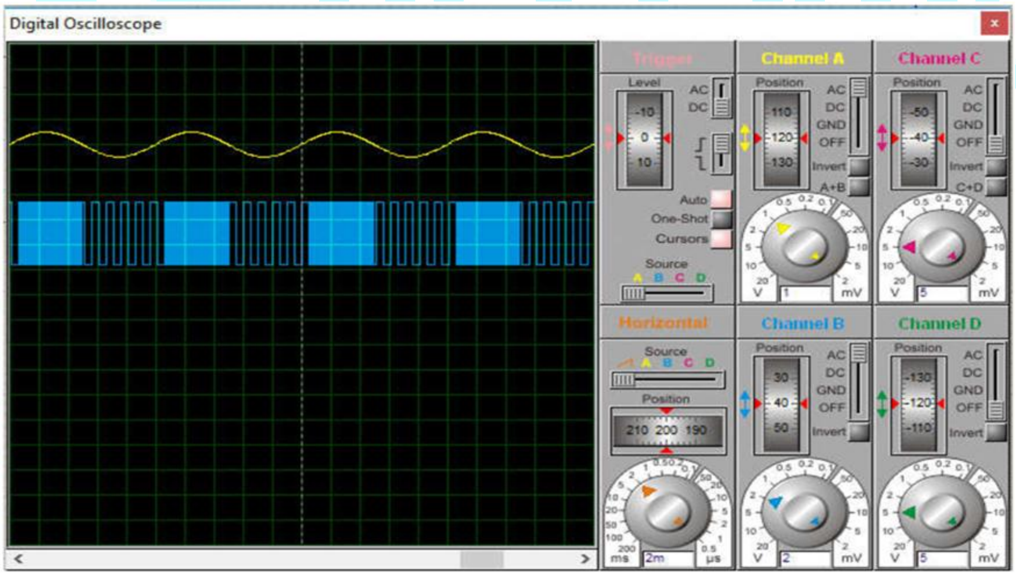

Figure 14: Output of PLL FM Modulator circuit

Channel A: Sine wave (message signal) in yellow.
Channel B: FM modulated wave in blue.

Analysis

The amplitude of the FM modulated signal is constant at ${}^{1}\!/_{2}\, V_{DD} = 5V$. The frequency of the FM modulated signal keeps on changing with time. The FM modulated wave shows variations in carrier frequency; regions of compression and depression indicate increase and decrease in amplitude of message signal respectively. The output from the PLL FM Modulator is an FM square wave signal which is going to the input of the RF Transmitter as shown on the proposed architecture.

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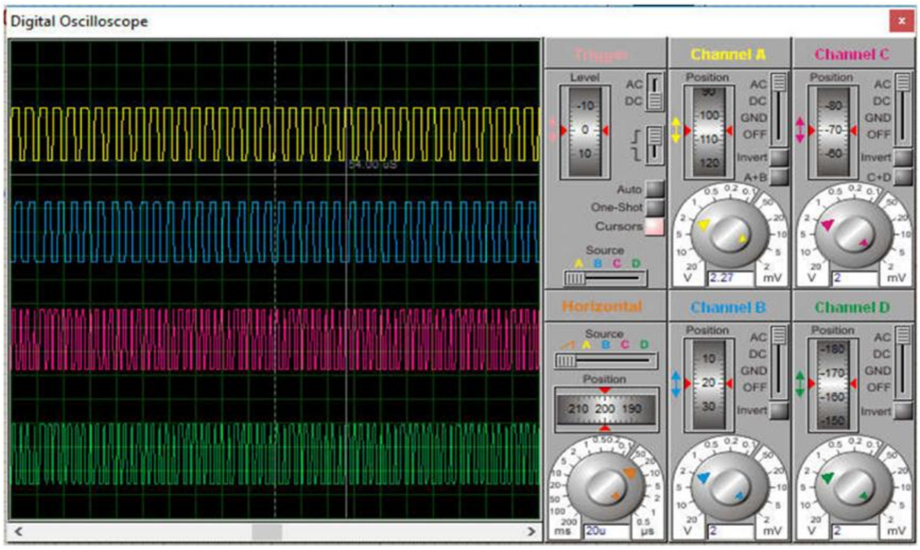

Figure 15: Output of PLL FM Demodulator circuit

Channel A: Input signal
Channel B: VCO Carrier signal
Channel C: PC1 output voltage signal
Channel D: LPF output signal

Analysis:
Frequency of input signal f_{in}

$$frequency = \frac{number\ of\ complete\ cycles}{time\ taken\ to\ complete\ one\ cycle}$$

From Figure 3.12, $f_{in} = \frac{19}{200\mu s} = 95kHz$ VCO Carrier frequency f_0

From Figure 3.12, $f_0 = \frac{19}{200\mu s} = 95kHz$

We notice that $f_0 = f_{in} = 95kHz$ as expected. That is, the FM demodulator synchronizes with any incoming signal whose frequency f_{in} is within the capture range

CR: $60kHz < f < 140kHz$

The FM Demodulator has been designed with a wide bandwidth or capture range CR: $60kHz < f < 140kHz$ so that the FM Demodulator can successfully detect or demodulate any FM modulated Signal whose frequency is within this capture range.

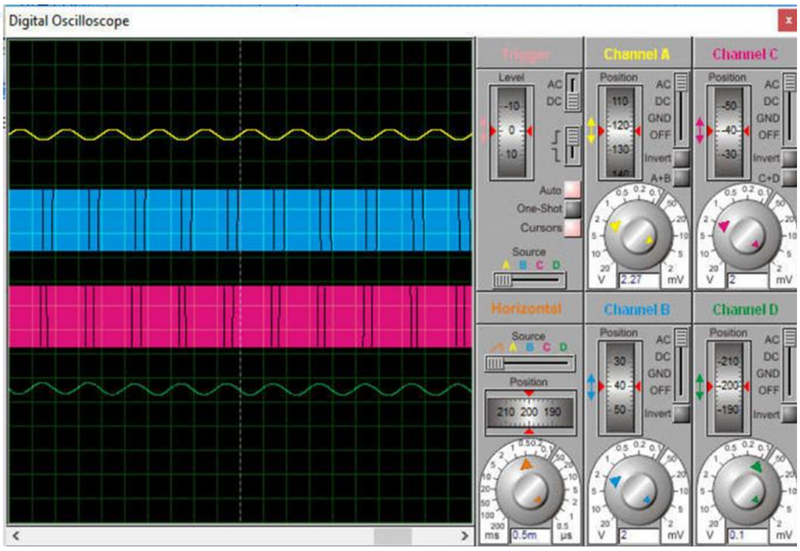

Figure 16: Output of Complete FM Modulator and Demodulator Circuit

Channel A: Information signal at 1KHz

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Channel B: FM modulated Signal
Channel C: Carrier signal of Demodulator
Channel D: Demodulated signal at 1KHz

Analysis

It is observed that both the information signal and demodulated signal have the same frequency (1KHz); meaning that the PLL modulator and demodulator circuits designed are reliable since they can successfully transmit and retrieve the same information.

7. Simulation of Audio Amplifier Circuit

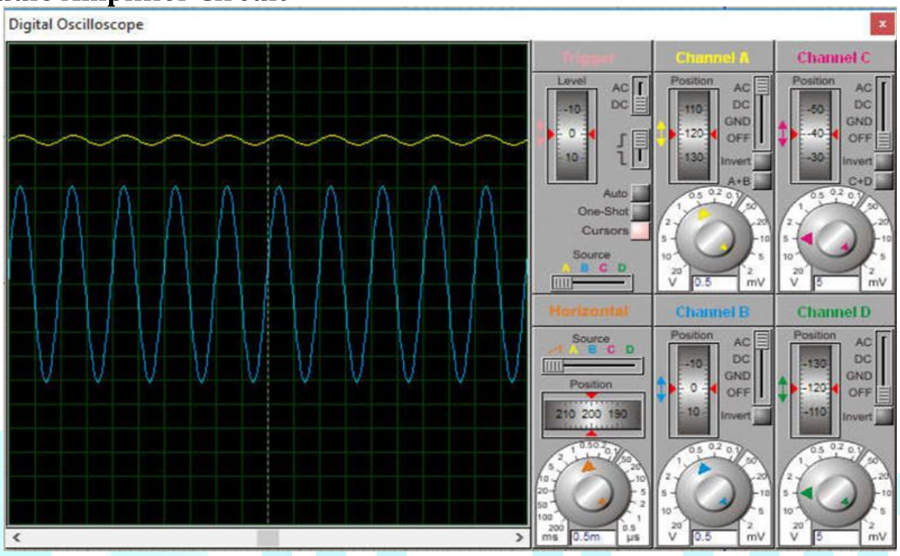

Figure 17: Output of low power Audio Amplifier

Channel A: Input signal (100mV)
Channel B: Amplified output Signal (2.02V)

Analysis

The input signal is amplified by a factor of 20. The display from the oscilloscope could not provide the amplitude of the input and output signals but these measurements (100mV and 2.02V) were obtained during simulation using the “Cursors” option available on the oscilloscope.

7.1. Findings

7.1.1. VCO transfer characteristics

VCO characterization is carried out in order to determine the linearity and frequency sensitivity gain (K_{VCO}) of the VCO. The input voltage of VCO is varied while the corresponding output frequency is being recorded.

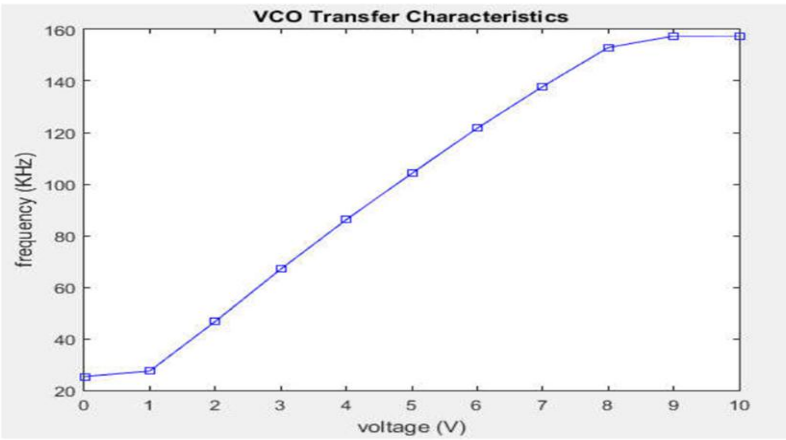

Figure 18: VCO Transfer Characteristics

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The slope of VCO transfer curve represents the linearity of VCO input voltage and output frequency. From the VCO transfer curve, the VCO of both modulator and demodulator circuits is designed with a frequency sensitivity gain

$$K_{VCO} = \frac{\Delta f}{\Delta V} = \frac{152.96 - 27.43}{8-1} = \frac{125.43}{7} = 17.93 \cong 18kHz/V$$

We observe that this value differs from the theoretical value computed in Section 3.2.2 -v ($K_{VCO} = 15kHz/V$).

7.1.2. Signal Analysis

A spectrum analyzer is used to analyze the signals we intend transmitting over our system. A spectrum analyzer is a test instrument that displays the amplitude versus frequency of an input signal. It can be used to monitor FM broadcasts and transmissions and also to test RF components/ subsystems.

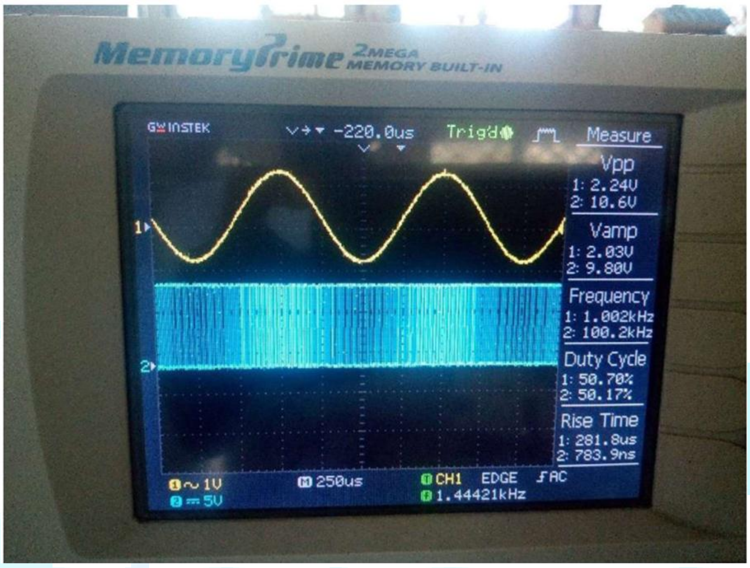

Figure 19: FM Modulated square wave signal with sine wave

A 1KHz sine wave modulated by 100KHz frequency carrier signal from modulator. The amplitude of FM signal is clipped at 5V which is the same as the FM signal generated from simulation.

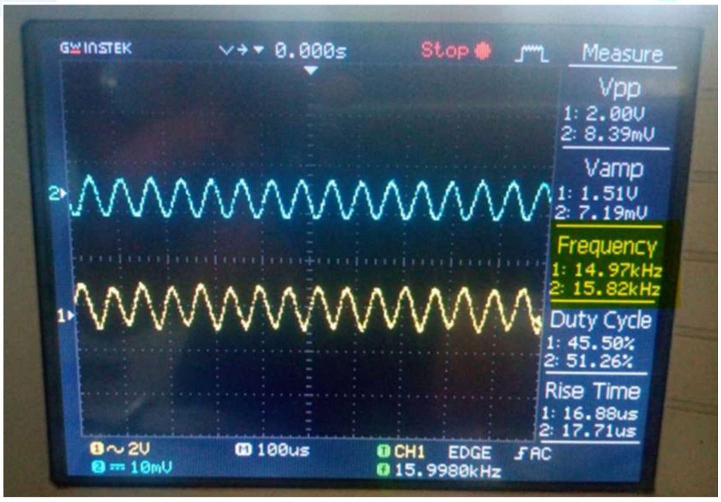

Figure 20: Modulation and demodulation of 15kHz sine wave

The frequency of the modulating signal is varied over the frequency range 1KHz to 40KHz to find out whether or not the system will modulate the audio range. The designed circuit could only demodulate signals within the range 1KHz to 30KHz making the proposed system suitable for transmission and reception of audio signals since the audio frequency range is 20Hz – 20kHz. Results from the modulation and demodulation of sine wave by PLL are the same as those obtained by, [31].

7.2. Prototype Layout

PCB editor (ARES) of proteus 8 professional IDE was used to draw the PCB layout of all the circuits to be mounted on breadboard.

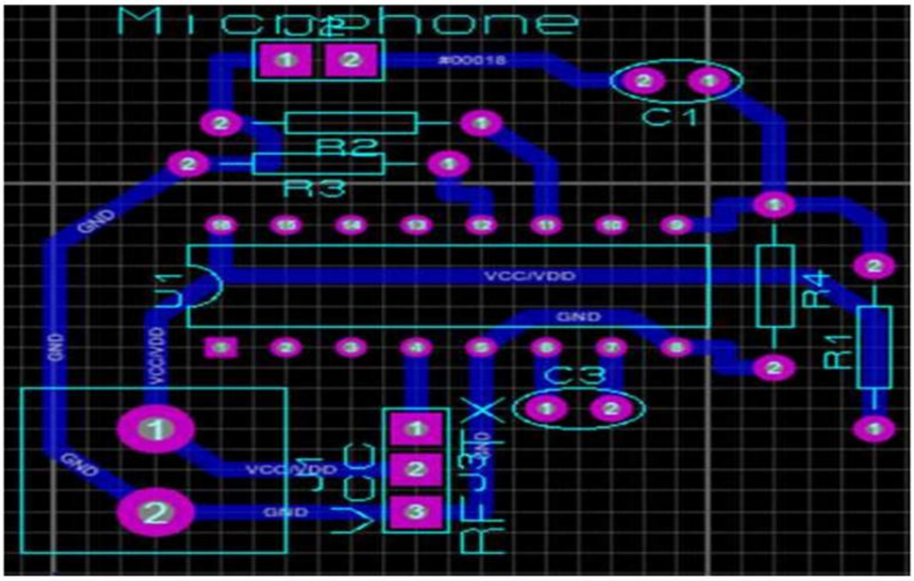

Figure 21: PCB layout of PLL based Audio FM transmitter

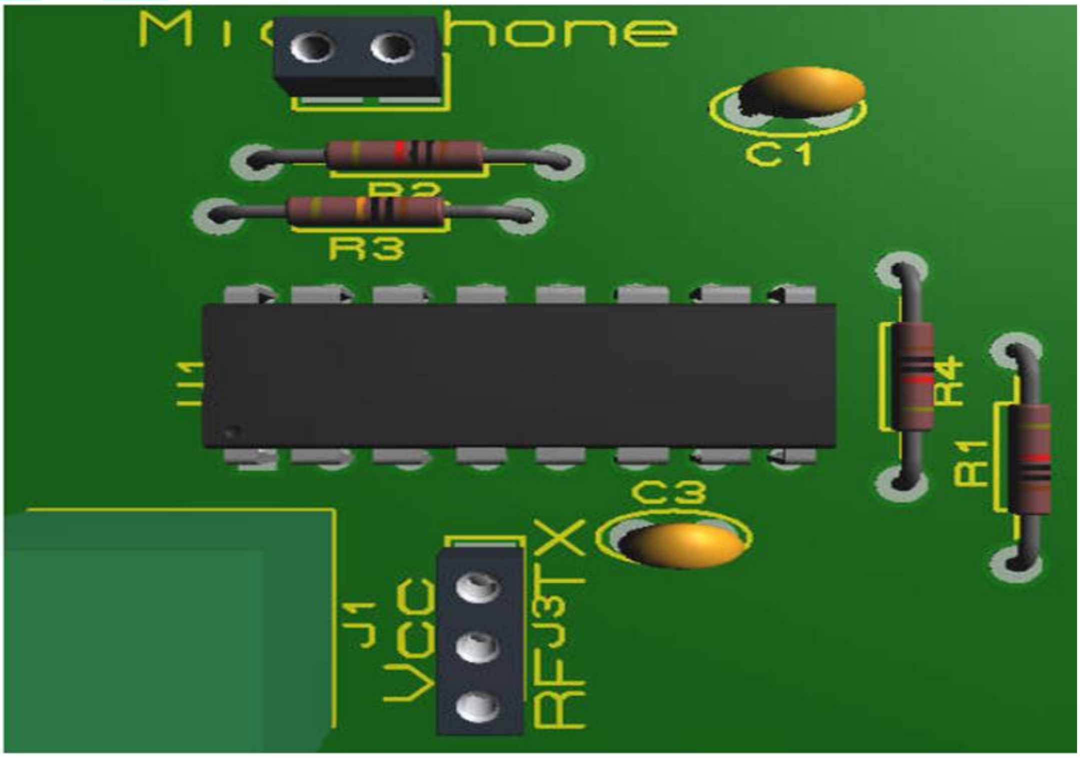

Figure 22: 3D visualization of PLL based Audio FM transmitter

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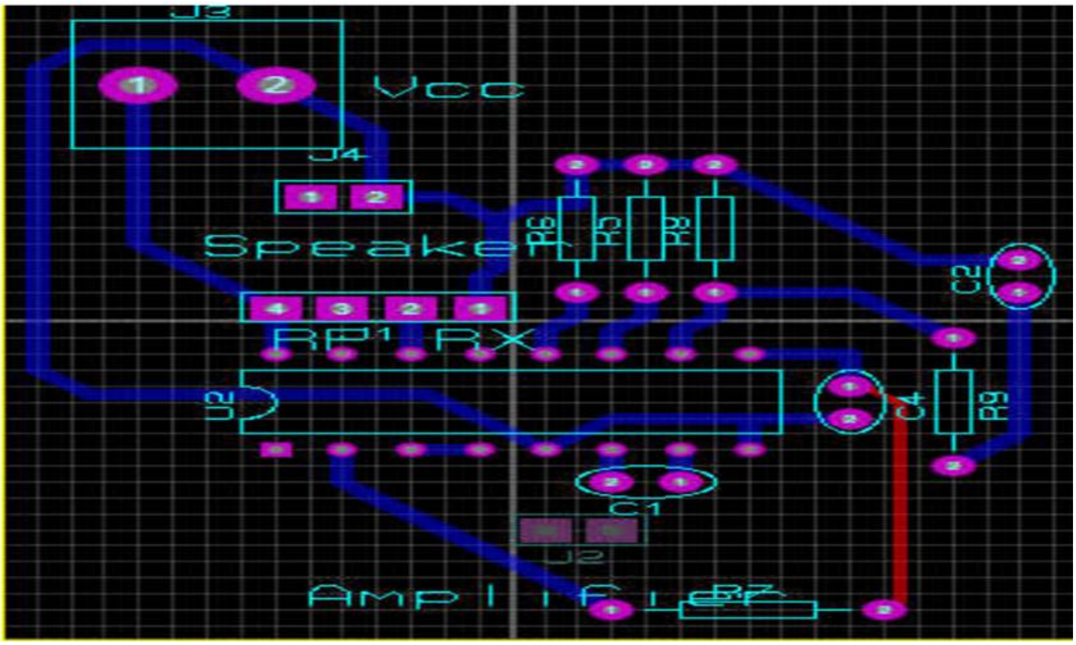

Figure 23: PCB layout of PLL based audio FM receiver

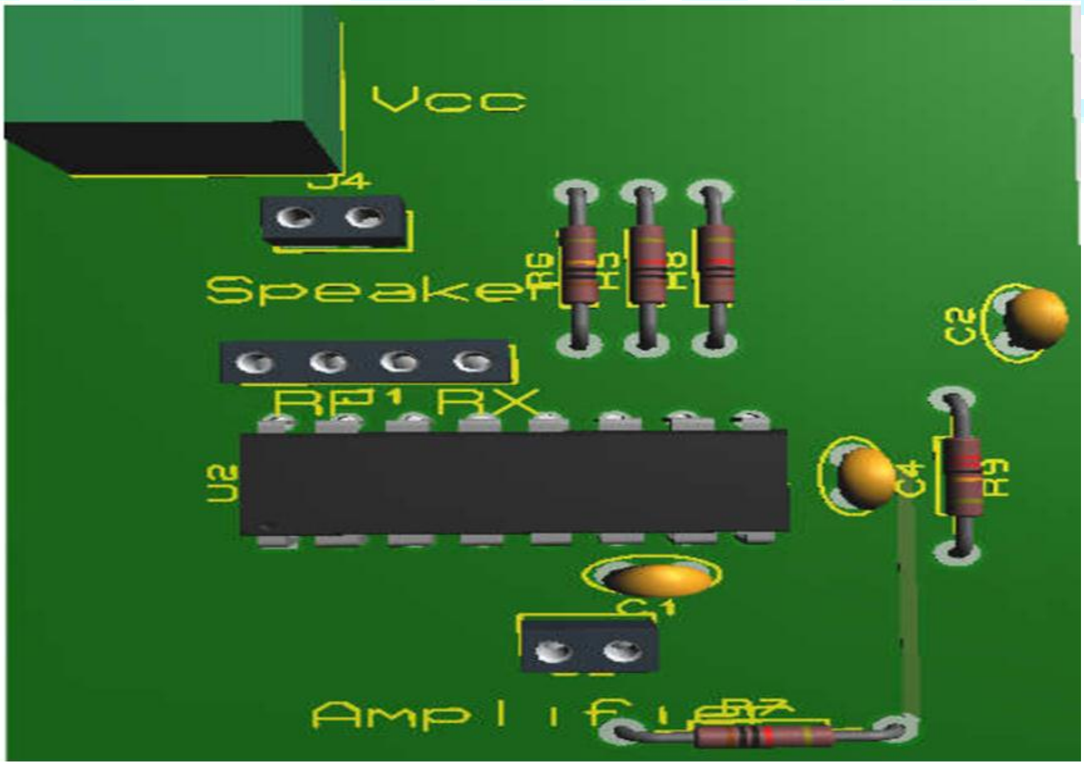

Figure 24: 3D visualization of PLL based audio FM receiver

7.3. Other parameters

7.3.1. Signal to noise Ratio (SNR)

This is a measure used to compare the level of a desired signal to the level of background noise. It is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 indicates more signal while a ratio lower than 1:1 indicates more noise.

From (Junho & Lyon, 1994), the standard SNR at the output of an FM receiver is given by

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$(SNR)_0 = \frac{3A_C^2 h_f^2 P_m}{2N_0 W^3}$ 4.1

Where:

A_C = carrier amplitude

h_f = modulation index

P_m = Power of message signal at receiver output

N_0 = AWGN

W = message signal bandwidth

For the FM system designed in this work,

A_c = 5V

$h_f = \frac{\Delta f}{f_m} = \frac{40kHz}{16kHz} = 2.5$

P_m = 0.7W

$N_0 = KT_0 = 3.77 \times 10^{-21}$ W/Hz

$W = f_m$ = 16kHz

Therefore

$(SNR)_0 = \frac{3A_C^2 h_f^2 P_m}{2N_0 W^3}$

$= \frac{3 x\ 5^2 \times 2.5^2 \times 0.7}{2 \times 3.77 \times 10^{-21} \times 16 \times 100}$

= 21dB

7.3.2. Coverage Range

The expected coverage range for the prototype is a transmission distance of 100m when the system is supplied with a 9V battery supply. The characteristics (ACOPTEX, 2017) of RF TX module indicate that, a maximum transmission distance of 300m is covered when the RF TX module is supplied with a maximum voltage, 12V. This distance makes the system suitable for long range monitoring and tracking.

7.3.3. System Bandwidth Analysis

The transmission bandwidth (B_T) an FM signal is given by Carson's rule

$B_T = 2(\Delta f + f_m)$ 4.2

Where:
Δf = maximum deviation from center frequency

f_m = highest frequency of audio signal

$B_T = 2(40 + 16)kHz = 112kHz$

The transmission bandwidth of the designed system is less than the standard 200KHz channel bandwidth of GSM carriers. The transmission bandwidth is designed to be large enough to increase the bandwidth efficiency of the system.

8. CONCLUSION AND RECOMMENDATIONS

8.1. Summary of findings

Ensuring awareness and security is a necessity for everyone. Tracking acoustic information of a milieu provides knowledge and consciousness about the type of ongoing activities and happenings within that environment. Several approaches have been developed for monitoring and tracking people. This research work employed audio surveillance based on an RF communication system designed for monitoring a local area.

This dissertation presented the design of a PLL based audio FM transmitter and receiver used for security, addressing and tracking of population. Understanding the process of FM modulation and demodulation is of outmost importance in communication systems. The design of the modulator and demodulator circuits was achieved by applying the knowledge gained from the fundamentals and concept of Phase Locked Loop. The design was based on mathematical analysis followed by computer simulations. The simulations verified that the design expectations were preserved from theory to realization of various circuits. The transmission distance covered depends on the supply voltage of the RF transmitter module.

A coverage range of 100meters was attained with a supply voltage of 9V without any degradation in signal quality. Filtering the audio signal at the output of the receiver ensures better signal quality and reduces the effect of noise from circuit components and other sources of interference.

8.2. Contribution to Education and Technology

The design and implementation of the PLL based Audio FM transmitter and Receiver system proposed in this research work has a vast domain of applications, where audio information is important.

In education, it will serve to improve on the awareness and security of administrators, teachers, students and other personnel on campus. This will help make the campus more conducive for the learning process as the people on campus are sure of a reliable communication system to alert them in case of impending danger.

In technology, the system proposed can be used as a substitute to cell phones. It is more cost effective as it

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does not require airtime to function. It is also power effective; it van be powered with a 9V supply.

8.3. Difficulties Encountered

Developing this system was not effortless and as such, some difficulties encountered are cited below.

- Equipment direly needed to achieve this work was not readily available like the spectrum analyzer.

8.4. Recommendations

To improve on the results achieved in this research paper, we have the following recommendations

- Integrating a high sensitive sound sensor will increase the monitoring range of the system.
- Increasing the supply voltage from 9V to 12V could increase the coverage range from about 100m to about 300m.

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