DESIGN AND SIMULATION OF AUDIO OVER IP (AoIP) FREQUENCY MODULATED (FM) BROADCAST STUDIO SYSTEM (88-108MHZ)

BY

NWACHUKWU CHINWEOKE (B.ENG)

(20065637398)

A THESIS SUBMITTED TO THE POST GRADUATE SCHOOL,

FEDERAL UNIVERSITY OF SCIENCE AND TECHNOLOGY,

OWERRI

IN PARTIAL FULFILLMENT OF THE REQUIREMENT FOR THE AWARD OF THE DEGREE OF MASTERS OF ENGINEERING (M.ENG) IN ELECTRONIC AND COMPUTER ENGINEERING

DECEMBER, 2010

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UNDER THE SUPERVISION OF ENGR DR. F.K. OPARA

DECEMBER, 2010

CERTIFICATION

This is to certify that Engr. Nwachukwu Chinweoke, a post graduate student with Registration Number 20065637398 has satisfactorily completed the requirement for the research work towards the award of the degree of Master of Engineering (M.Eng) in Electrical Electronic Engineering.

ENGR. DR F.K OPARA ----------------------------

SUPERVISOR SIGNATURE & DATE

ENGR. DR E.N.C OKAFOR ----------------------------

H.O.D E.E.E. SIGNATURE & DATE

ENGR. PROF. C.D OKEREKE ---------------------------

DEAN PGS. SIGNATURE & DATE

EXTERNAL EXAMINER SIGNATURE & DATE

DEDICATION

This work is dedicated to almighty God who has given me the grace to accomplish this task and to my mentor Late Ven M.S Bello.

ACKNOWLEGMENT

My utmost gratitude goes to my advisor, Engr. Dr F.K. Opara, who has been a wonderful source of encouragement for his guidance, patience, and support. I am grateful to the head of department Engr. Dr E.N.C. Okafor for the wonderful role he played and to the entire staff of my Department and my colleagues especially Etus Chukwuemeka for their support and encouragement.

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To all those who contributed to my success, thanks and God bless.

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LIST OF ABBREVIATIONS

IP	Internet Protocol
TMT	Technology, Media and Telecommunication
TCP	Transmission Control Protocol
UDP	Universal Datagram Protocol
LAN	Local Area Network
WAN	Wide Area Network
TDM	Time Division Multiplexing
FM	Frequency Modulation
CB	Citizens Band
AES/EBU	Audio Engineering Society/European Broadcasting Union
RU	Rack Unit
DAB	Digital Audio Broadcasting
IBOC	Inband On Channel
ISDB-T	Integrated services Digital Broadcast-Terrestrial
AM	Amplitude Modulation
SW/MW	Short Wave/Medium Wave
HD	High Definition
DRM	Digital Radio Mondiale

BBC	British Broadcasting Coporation
SR	Satellite Radio
ESN	Electronic Serial Number
ITC	In The Air Channel
PAD	Program Associated Data
RTP	Real –Time Transport Protocol
NIC	Network Interface Card
UTP	Unshielded Twisted Pairs
TIA/EIA	Telecommunications Industry Association/Electronic Industries Alliance
MAC	Media Access Control
ARP	Address Resolution Protocol
AoE	Audio over Ethernet
FTP	File Transfer Protocol
SMTP	Simple Mail Transfer Protocol
ToS	Type of Service
TTL	Time to Live
NAT	Network Address Translation
DHCP	Dynamic Host Configuration Protocol
IGMP	Internet Group Management Protocol
VoIP	Voice over Internet Protocol
AoIP	Audio over Internet Protocol
Bps	Bit per second
EMI	Electromagnetic interference
SNR	Signal to Noise ratio

CODEC	Coding Decoding
IEEE	Institute of Electrical Electronics Engineering
STL	Studio Transmitter Link

ABSTRACT

Communication Protocols are the key tools for data transmission; most of them were defined for asynchronous data transport. Underpinning the Internet is the Internet Protocol (IP), which, has offered an ideal pathway for general Information Technology convergence with its capability of carrying large variety of signals (audio, control, voice, video, data, etc.). This thesis is on "the Design and Simulation of Audio Over IP (AoIP) frequency modulated broadcast studio system". Using a digital microphone, radio signal is introduced into the system, the introduced signal transported within an IP based Local Area Network is mixed with a combination of digitally networked console/mixer, CD/Tape recorder and computer system to produce an output signal routed across the network and modified/edited to produce radio programs, jingles, commercials, intro, call sign etc. A dedicated server within the network is used to store these programs from where the programs are cued for airing. The studio output signal is excited and amplified severally with a transmitter to produce radio frequency signal within the FM frequency range. This can be received by multiple listeners within the given range. With the aid of a Studio Transmitter Link (STL) and a Satellite link the studio output signal is sent to other network stations. This research work is a new studio automation technology for audio signal transmission in radio stations within FM range (88-108MHz). The work proves a perfect replacement for traditional form of audio signal transport. Educational Studios, Radio Broadcast Studios, Music Studios would find this very resourceful.

CHAPTER ONE

1.0 INTRODUCTION

This section deals with AoIP broadcast system introduction which entails thesis motivation, objectives, scope, justification and organization. For many decades, analog techniques were used in broadcast industry for distribution of television and radio programmes worldwide. These techniques served well in time past. However, the advent of digital technologies which have enabled diverse devices such as GSM telephone system, i-pod, compact disk, DVD's etc has also revealed the limitations of analog technologies.

Nigeria is one of the countries that is still broadcasting analog from program production, content distribution to transmission using traditional purpose-built equipment. This poses challenges and operational hitches in the area of quality of transmitted signals, air time, operational cost etc. Use of conventional analog and digital mixers, studio links etc have not completely eliminated these problems.

Presently, many countries have embarked on the programme of 'analogue switch-off' which should see existing analogue broadcast services replaced by superior digital solutions, economizing the valuable radio spectrum and freeing up frequencies to serve the growing demand for more diverse and advanced services. There are many radio stations all over the world presently broadcasting using a variety of/different types of digital transmission technique. Radio broadcast equipment from transmitters, mixers, receivers, antennas, studios and relay links are also widely available in developed countries, with new standards for digital transmission which includes DAB Digital Radio, HD radio/IBOC, ISDB-SB FMeXtra, DRM etc. The proposed AoIP broadcast studio system is one of the new digital transmission technologies for implementing digital changeover that would eliminate problems associated with older transmission mode at primary and intermediary levels.

1.1 THESIS MOTIVATION

With the year 2015 as the global deadline for digitization (the conversion of broadcast and communication systems from analogue to digital) an important global movement driven by the international Telecommunications Union (ITU), audio over IP is rapidly becoming the technology by which broadcasters are planning future broadcast network infrastructures. The development of an Audio over IP system for broadcasting is one of the technologies for implementing digitization. It is also one of the most significant and ideal changes in broadcast environment to meet the current conventional trend in digital broadcasting, digital electronics, computers and tremendous flow of global information.

The idea of audio over IP for broadcasters is relatively new; with it, broadcasting is completely digitized from the point of content delivery to the end point of its transmission from broadcasting consoles, to routing of signals to different studios. The proposed Audio over IP is another digitization technology that would bring transformation to the broadcast industry [1]. With it, audio routing in stations can be managed remotely, system can run completely on an automated way and the broadcasting industry would be at the verge of an IP-fueled revolution in distribution and infrastructure design [2].

1.2 THESIS OBJECTIVES

This thesis has three objectives.

- To design a robust, reliable and flexible IP - based audio networking (AoIP) system for delivering high quality radio services to listeners in multiple destinations using ethernet switches.
- To automate the studio system to address problems and hitches related to time loss and reduce organizational and operation cost.
- To simulate the AoIP FM studio system using Visual Studio 6.0 and to run the simulated design.   .

1.3 THESIS JUSTIFICATION

AoIP is one of the technologies that characterize the ongoing transition to digital transmission. With the digital transition mandate, many broadcasting firms, both large and small, Government and Private, are either thinking of or have already begun the process of replacing their existing broadcast infrastructure with audio over IP (AoIP) solutions. Today's economic situation has made it more crucial than ever for decision makers to consider the type of return their organizations can expect from prospective investments. Enterprises that are investing in IP Communications have two primary expectations — reducing operational costs and improving their organization's communications capabilities. IP communication is a viable technology that when implemented, by converging existing audio, voice, video and data networks onto a single IP-based network meets this expectation and provides solid foundation for the deployment of advanced, feature-rich services and solutions. IP telephony, unified messaging, and multi-channel contact center applications are just a few examples of such solutions. While some organizations have expressed concern about migrating to the converged model, industry studies show that those that have already implemented IP Communications indicated that the quality, resiliency, and scalability this technology delivers met or exceeded their expectations [3]. Other advantages of IP system include flexibility and ease of management. Detailed discussion on the advantages offered by this system is contained in Chapter four.

More so, this system solves the problems and bottlenecks created by a studio failure. Usually when a studio fails, the studio manager, news reader and presenters would have to physically move from one studio to another studio to continue operation, this usually results in air-time loss. But with AoIP studio system, this problem is eliminated as the AoIP signal is available throughout the network steam and would be routed directly to any desired destination. Cisco Systems in its white paper titled The Strategic and Financial Justification for IP Communications gave a generalized detailed justification for investing in IP communications that I believe would further justify this system [4].

1.4 SCOPE OF THE STUDY

The ability to deliver high level and quality service to listeners is an essential aspect of radio broadcasting service. This research focuses on the design and simulation of IP – based networked audio broadcast system - an automation of existing technology, with the aim of replacing the existing mode of signal transmission flow with this alternative technology. The analysis and testing of the simulated design was also captured in this work.

1.5 THESIS ORGANISATION

To achieve the objectives, the project is divided into different stages organized as follows:

• Chapter one gives a detailed introduction to Audio over IP technology concepts and background.

• Chapter two presents the literature review of AoIP broadcast studio system technology and protocol.

• Chapter three focuses on methodology, design analysis and simulation of AoIP broadcast system.

- Chapter four is on the AoIP broadcast system configuration, running and evaluation.
- Chapter five is on conclusion and recommendation.

CHAPTER TWO

2.0 LITERATURE REVIEW OF AoIP BROADCAST SYSTEM TECHNOLOGY AND PROTOCOL

This section deals with the literature review of AoIP broadcast system which covers historical background, trends in audio broadcast technology, broadcasting techniques and the concept of internet protocol.

2.1 HISTORICAL BACKGROUND

Before the development of digital broadcast system, broadcast audio/video production, distribution and transmission was purely analog and relied completely on analog process using continuous variable signal and manual processes (traditional techniques). In analog form, signals are represented as a continuous electromagnetic wave forms whereas digital signals are represented in binary forms (1's and 0's) through a series of discrete bits. It has only two states 'ON' and 'OFF' representing logic states '1' and '0'. An example of this is the Push Switch. Digital systems cannot process analog signals and their varying degrees of amplitude, but utilizes digital signals to understand, interpret and process commands and data [5]. AoIP system utilizes digital transmission technology.

2.2 ANALOG VERSUS DIGITAL SIGNAL

Analog audio signal has to do with the representation of series of sounds through the use of sinusoidal waveform (signals) that changes continuously. Analog signal can be explained by examining the transmission of a natural form of information, such as sound or human speech, over an electrified copper wire. In its native form, human speech is an oscillatory sound in the air which varies in its volume or power (amplitude) and its pitch or tone (frequency). As audio signals from the studio fall onto a transmitter, variations in electrical waveforms are created over the electrical circuit; these waveforms maintain their various shapes across the wire until they fall on the receiver or speaker, which converts them back into their original form of variations in air pressure. A similar but more complicated conversion process is employed to transmit video over networks. Conversely, digital signal is the representation of information in discrete formats. Figure 2.1 depicts digital and analog signal representation.

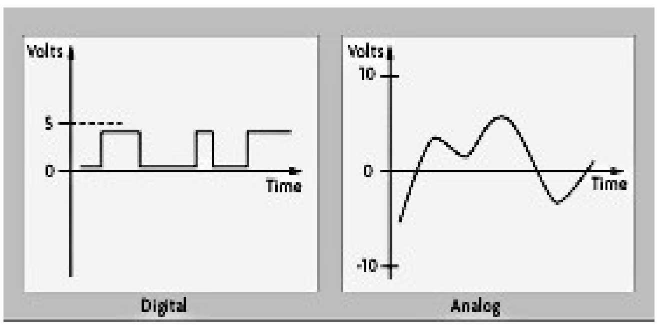


Figure 2.1: Digital and Analog Signal Representations

Bandwidth

A voice channel is approximately 4,000 Hz (4 kHz). Between 200 to 3,500 Hz of this is actually used for voice signal, the remaining bandwidth is used for purposes of network signaling, control and to maintain separation between information channels. Analog signal is more bandwidth-conservative; a raw information stream consumes less bandwidth in analog form than in digital form. Band-limiting filters are used in carrier networks to constrain the amount of bandwidth provided for voice applications. Analog signal is commonly converted to a digital bit stream, requiring a maximum of 64 Kbps for full fidelity/quality and 4 kHz voice bandwidth. By Nyquist's theorem, the frequency bandwidth must be twice the maximum required to produce good quality voice. Thus results in producing 8 kHz frequency bandwidth [6].

Digitally, bandwidth is measured in bits per second (bps). The amount of bandwidth required depends on the amount of raw data to be transmitted, the desired speed of transmission, issues of finance etc. Data can be routinely compressed in order to reduce bandwidth requirement, enhance efficiency of transmission and reduce transmission costs.

Compression and Security

Digital data can be relatively easily compressed, thereby increasing the efficiency of transmission after which substantial volumes of voice data, video and image can be transmitted using very small bandwidth. Digital systems offer better security; analog systems offer small level of security through the scrambling, or intertwining of several frequencies. Digital information, on the other hand, can be encrypted to create the appearance of a single, pseudo-random bit stream. The true meaning of individual bits, sets of bits, or the total bit stream cannot be determined without having the key to unlock the encryption algorithm employed [7].

Quality and Attenuation

Audio signals tend to pick up noise/interference as they transverse the network. This is particularly true of twisted pair copper wire; such wires tend to act as antennae thereby absorbing noise from outside. Thus, the quality of the signal degenerates and is distorted by the noise. Analog signals unlike digital signals tend to weaken, or attenuate, over a distance; this is particularly true of electrical signals carried over twisted pair copper wire, due to the level of resistance in the wire. Attenuation is sensitive to carrier frequency and usually higher at higher frequency.

The signal to noise ratio (SNR) in dB is computed as 10log(Power out/ Power in)

$$dBm = 10\log(\text{Power measured in mW})/1\ \text{mW}) \ldots. (2.1)$$

Where dB = decibel, dBm = decibel relative (with reference to) 1 milliwatt and dBrn = decibel above reference noise

The reference power used in describing noise is given as –90dBm

To describe the noise in circuits, we use dBrn = dBm + dB

0 dBrn = –90dBm + xdB, we need x = 90

Thus, the SNR for 0 dBr is 90dB, this is the SNR used for most sound cards.

There are also several advantages of digital systems that are of practical value. Digital copies of media and its resources can be reproduced with mathematical precision and without any degradation. This is certainly not the case with the old analog dubbings. Digital media allows dynamic range non-sequential (random) access to files, making audio editing with software tool on a computer workstation, a very user-friendly and non-laborious process. It also offers greater flexibility in playback and recordings. Digital audio can be stored in computer-based data storage server and scheduled ahead for playback automatically. Digital systems also allow non-audio data to be inserted into the database together with the audio such as the artist's names, track titles, plus other useful information. These can be easily retrieved by the disc-jockeys (DJs), audio engineers and studio managers from theoretically almost anywhere like the reporters corps, production studio and on-air studio [8]. Other excellent features of digital systems are less tedious and easy maintenance and troubleshooting which are usually few, improved signal quality, superior technical specifications in the area of frequency response, signal to noise ratio.

Few disadvantages occur in issues of training and retraining and unavoidable noise generated by cooling systems.

2.3 TRENDS IN AUDIO BROADCAST TECHNOLOGY

The radio broadcast industry has evolved strongly over the years. This evolution has taken place step by step. The move from analog to digital world has not only meant moving from turntable to Compact Disc, it has also involved the 'dematerialization' of the media [9]. When digitization came along, the workflow changed for most radio stations. Digital audio recorders replaced analog ones, signal processing and switching became all digital, and signal transport became AES/EBU (Audio Engineering Society/European Broadcasting Union). Carts, reels of tape and CD players have been supplanted by sound file editors and PC delivery machines which had only replaced turntables few years earlier. Editing workstations have replaced Scissors, razor and tapes. With the introduction of M-audio, Digigram sound cards, with specific functions tailored for the radio broadcast industry, manual functions have been replaced on a one-on-one basis by editing functions. Most programs are now recorded, edited, cued and played out of computer system. Now, we are smacked inside the center of the conversion to digital for mixing, routing, and processing – AoIP technology [10]. This would enable radio automation and transform the way radio operations are organized. Gone are the days of playing from carts, vinyl, cassette and reel of tape.

2.4 BROADCAST STUDIO OPERATION TECHNIQUES

Over the years, digital technology has revolutionalized the radio industry, the production person can record, edit, produce and otherwise manipulate an audio signal [11]. All the manual process of "button pushing" have been replaced with manipulating a mouse or keyboard. Audio tape cartridge machines and rotary pot consoles are obsolete, while turntables and reel to reel tape recorders sit idle in many studios. From the advent of the compact disk player in early 1980's to the total digital production studio in the early 2000's, radio has eagerly embraced digital technology. This technological change cuts across the equipment used as well as the physical setup [12].

2.5 AUDIO RECORDING

Older studios are designed to use traditional analog recording equipment namely reel to reel, cart, cassette/tape recorders etc. These have been replaced by newer CD and DVD digital equipment using digital audio workstations (DAW) and digital studio editors [13]. In the analog recording process a duplicate or electromagnetic representation of sound wave of the original/source sound is stored on magnetic tape. e.g the microphone converts the sound/acoustic energy (changes in sound pressure) to electrical form (changes in voltage) which is sent down via the microphone cable through the mixing console to the audio tape recorder where it would be recorded as change in microphone strength. Each time the analog signal is recorded or processed in some fashion, it is subject to degradation due to slight change/variation in signal shape due to noise, interference and reduced signal strength. Analog encoding is similar to creating a line graph to show statistical analysis. It has all its measurements on a continuous line that curves up and down with no discrete point(s). Recording process is like trying to retrace a curve on a graph; the production (output) is usually slightly different from the original.

Analog recording relies on magnetic pulses stored on tapes as such; any defect or decrease in magnetic properties of the tape implies reduced signal quality.

Other notable problems of analog recording include noise, distortion, cross talk, flutter, hiss, hum etc.

In modern production studios, computers and other equipment are used in digital recording process whereby audio encoding is accomplished in a discrete fashion similar to viewing individual numbers in a statistical analysis and writing them down in a set order. The audio signal starts out as analog and then converted to digital through four stages namely: Filtering, Sampling, Quantization and Coding. These stages are illustrated using sample signal amplitude values over time interval of 0 to 10s as given in table 2.1.

Table 2.1: Sample Amplitude between 0 – 10s

Time (s)	Amplitude (V)
0	0
1	3
2	5
3	5.9
4	4
5	2
6	1.2
7	2
8	0

By plotting the graph of Amplitude versus Time, the original analog signal would be drawn as shown in fig 2.2.

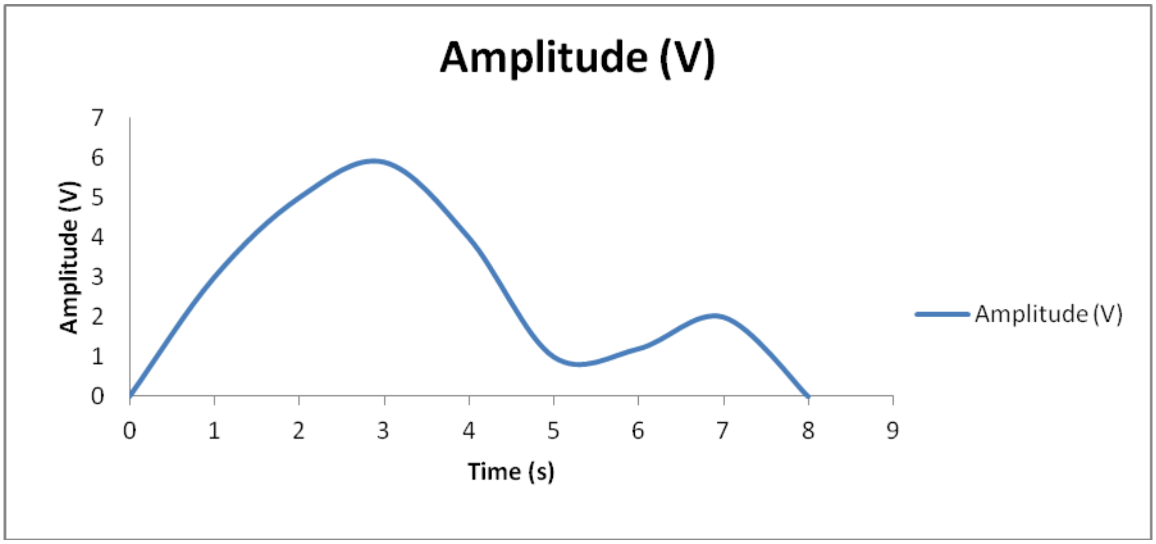


Fig 2.2: Graph of Source/Input Analog Signal

Filtering and Sampling: These are the first two stages, during these stages, the original signal above is sent through a low pass filter called anti-alias filter that strips off/removes frequencies above the range of human hearing and frequencies above Nyquist frequency (one-half the sampling rate) that would alias during sampling, if not removed. These frequencies although originally inaudible, can be shifted/aliased into audible range during recording or anti-aliased during playback. The filtered analog signal is then divided many times through a process of analog-to-digital conversion known as sampling as shown in fig 2.3.

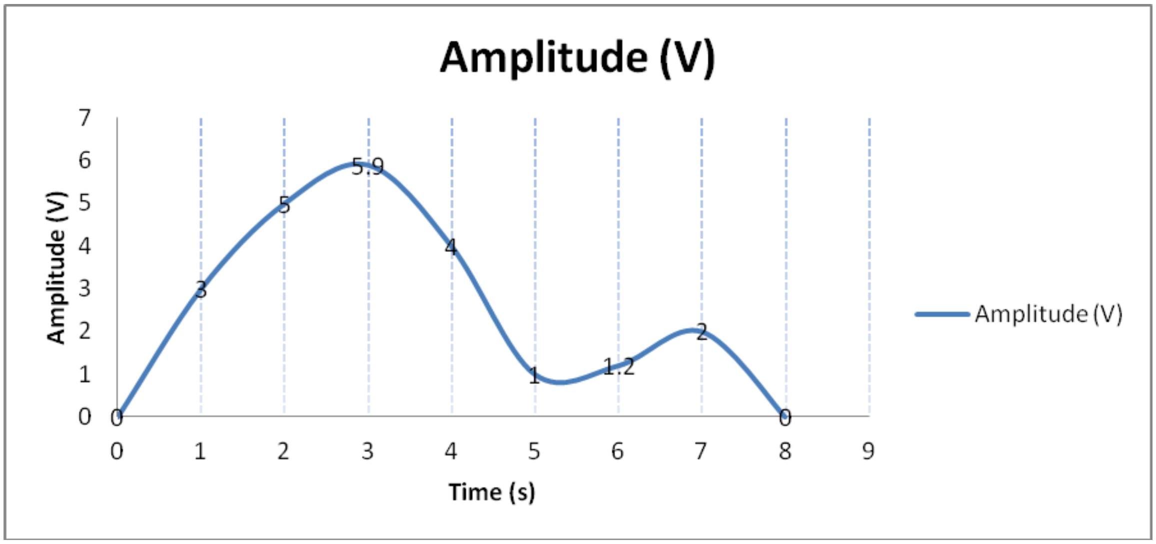


Fig 2.3: Filtering/Sampling

Each sample represents the amplitude of the audio signal at the moment the sample is taken. The more the samples taken and converted to binary data, the more the accuracy/exact production of the original sound can be recorded unto the storage medium. This process is much like that of film, where 24 still photographs are taken per second, when these photographs are played back at the same rate; they create an illusion of movement. When audio samples are played back, they create an illusion of a contiguous sound. However, to actually achieve this, more than 24 samples must be taken per second. Many digital audio equipment utilizes sampling rates of 32, 44.1 and 48 thousand samples per second. The sampling rate which defines the number of samples taken per second must be at least twice the highest frequency in the audio signal to enable high quality and accurate encoding and decoding. As stated earlier, since we can't hear sounds above 20kHz, the sampling rate should be slightly greater than twice 20kHz, this is one reason why the most common digital sampling rate is 44.1kHz. Lower sampling rates can be utilized for lower quality recordings. Very high sampling rates produce audiophile-quality sound, which must be played on highest fidelity equipment to be appreciated. This would be too bandwidth-intensive to broadcast.

Quantization and Coding: These are the last two stages in audio digitization. In these stages numerical values are assigned to each individual sample. The samples of the signal's amplitude taken can fall at any point within the given range of amplitudes, from absolutely silent to very loud, between the two extremes 0 and six in this case or between any other two points in the geometry. Quantization breaks up these infinite points into a more manageable digital number, rounding samples up or down to the nearest value. Bit depth has to do with the quantizing levels, the more the level, the more accurate the information you would have about the signal as samples would be rounded up or down. For example, 1-bit system would signify just 2 quantizing levels - either minimum or maximum. This does not give much information about the signal. Each additional bit doubles the number of levels – 2 bits gives four levels, 3 bits corresponds to eight levels etc. The standard bit rate for most digital recording is 16 bits, with some compact disk and DVD recording now being done at 20 or 24 bits technology. The 16 bits represents up to 65,536 values. Higher bit depths equal lower noise and better fidelity of the digital recording. Coding involves putting 0's and 1's in a precise order corresponding to the values measured during quantizing process. This binary, or digital, "word" presented in the table and figure below represents each individual sample's quantized (rounded up or down) voltage level at that particular moment. This is accomplished with the analog-to-digital (A/D) converter and having in mind that the actual data recorded is digital not its analog representation.

Table 2.2: Amplitude Quantized/Coded to the nearest bit.

Amplitude (V)	Binary Equivalent	Nearest bit
0	000	000
3	011	011
5	101	101
5.9	101.11100110011	110
4	100	100
1	001	001
1.2	001.00110011	001
2	010	010
0	000	000

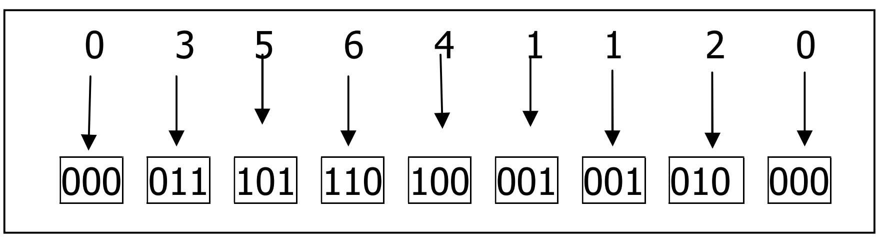


Fig2.4: Amplitude Quantized/Coding to the nearest bit.

With digital technology, we can copy from tape to tape with no measurable loss of quality. Along with the improved frequency response, wide dynamic range, and drastically reduced noise and distortion, the ability for excellent recording with no quality reduction has contributed greatly to the acceptance of digital radio production.

Finally, to transform the digitized signal back to analog, the recorded bits are sent to a digital-to-analog (D/A) converter for reading and decoding(reverse production of the original signal) and also to another low-pass filter set to the Nyquist frequency to eliminate frequencies above the Nyquist rate, and may include a correction for the zeroth-order hold. This output filter is called a reconstruction filter**.** A signal of corresponding analog amplitude is thereby produced and "held" for the individual samples, producing a stair-step signal that represents the analog signal as shown in fig 2.5.

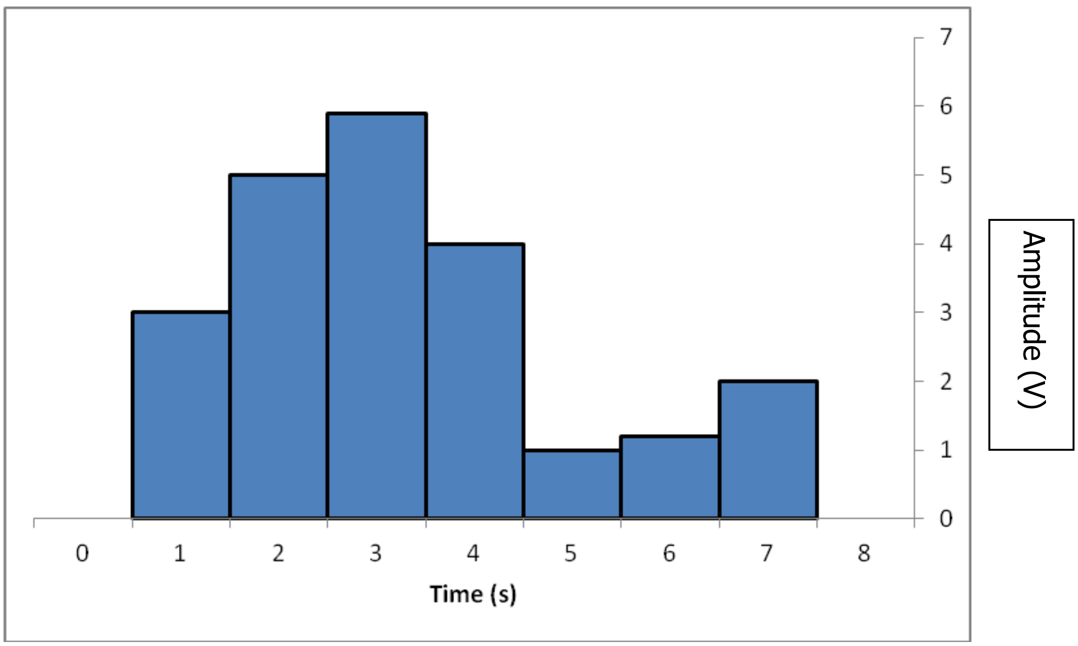


Fig 2.5: Stair Step Signal Representation

The filter employs exact mathematical attribute for each sample thereby creating a correct reproduction of the original analog signal as shown in figure 2.6 below.

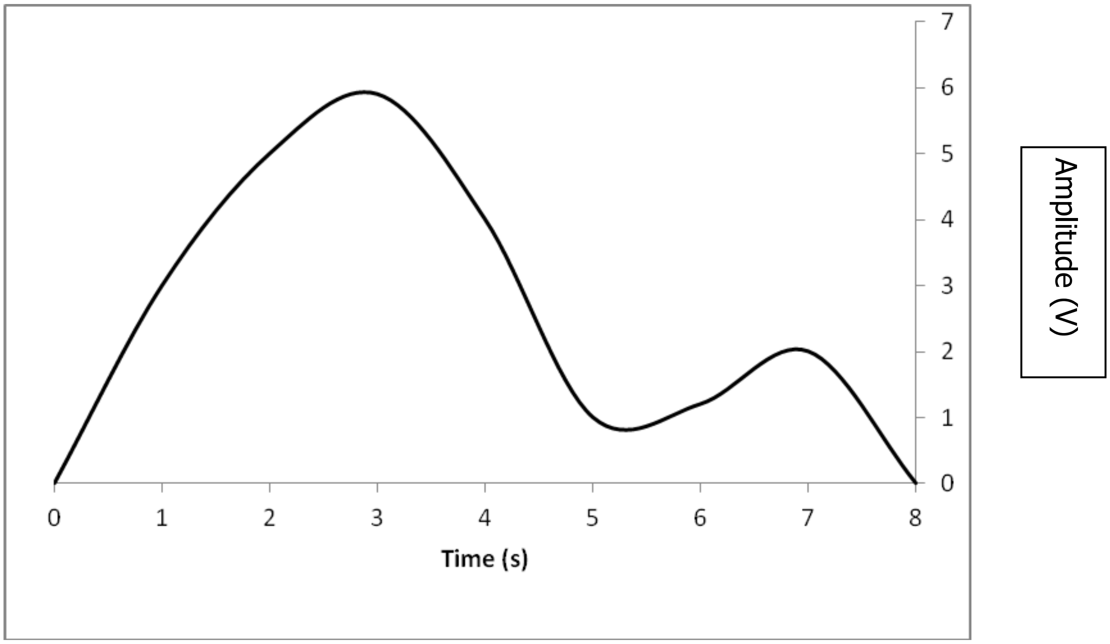


Fig 2.6: Recreated Analog Signal

AoIP is digital audio workstation (DAW) based and uses digital technology; producing radio programs, commercials, jingles, music etc and storing them in several variations while maintaining good digital quality. It also allows for On–Air recording play out and transmission using a digital transmitter and logging of transmitted data and information about aired programs. This system is now being deployed by more and more radio stations.

A major recording equipment is the microphone. The AES 42 standard, published by the Audio Engineering Society (AES), defined a digital interface for microphones. Microphones conforming to this standard directly output a digital audio stream through the male connector (XLR) connector, as opposed to producing an analog output. Such digital microphones are used either with new audio equipment which has the appropriate input connections conforming to the AES 42 standard, wireless or with a suitable interface box. Studio-quality microphones which operate in accordance with the AES 42 standard are now appearing from a number of microphone manufacturers. AoIP system microphones use the AES 42 standard.

2.6 AUDIO PLAYBACK

Older studios use turntables, cassette players etc for playback. These equipment are obsolete and no longer used in modern studios. At this transition stage, many stations now playback their programs, music, commercials, reports and other programming elements from a digital audio storage system with mouse click, drag and drop operations and digital audio tape (DAT) recorders. Similar technique as explained in audio recording section applies here using digital audio workstations (DAW) and digital studio editors.

2.7 AUDIO MIXING, MONITORING AND ROUTING

The heart of any radio station is the audio console. Signals from other equipment go through it. Audio consoles perform three major functions, selecting, monitoring and routing audio signals. Individual buttons and slide controls (faders) assigned to different channels and equipment are used to actualize these functions.

Older studios are equipped with gigantic consoles which uses daunting looking knobs known as potentiometer and purely analogue operations. Presently, consoles are digital offering many more features, equalizers, increased channels and its associated switches allow operators to control other equipment in the studio [14]. In place of the knobs, it is now usually designed with faders which offer higher efficiency. AoIP compatible consoles accommodates a collection of dedicated processing features such as encoding/decoding, routing, mixing, time stretching, equalization etc to handle radio automation tasks. These consoles are also equipped with or interfaces with a companion “audio engine”, switch or router with other audio equipment wired to and controlled by it.

Loudspeakers and Headphones are the major equipment for audio monitoring. The fundamental technology behind loudspeaker design has remained very much the same since the first electro-acoustic transducer was invented by Alexandra Graham Bell in 1876. Today, passive speakers have made way for amplified speakers, or more commonly known as the active studio monitors, and have dominated most studios [15]. The advantages of a dedicated amplifier-speaker match are further enhanced with active digital crossover filter design instead of a passive filter topology. The Adaptive digital signal processing (DSP) based digital equalizer techniques, first adopted by hi-fi industry provides a viable and inexpensive solution to poor speaker output.

2.8 AoIP BROADCAST SYSTEM

Broadcasting centers on distribution and transmission of audio and/or video using electrical signals, so that it can be heard and seen over a large area. An audio network system is a networking system designed to accept audio input signals, manipulate them with processes such as equalization and mixing, and make them available throughout the system. AoIP broadcast system differs from traditional broadcast system in that a mechanism is provided to transport audio in digital format throughout the system via some sort of high speed interconnects. In a typical audio over IP network system, input-output satellite rack or node is localized wherever there are audio sources or destination devices. The inputs and/or outputs of the audio devices are wired to the local rack with generally short, simple and standardized cables. The local racks in turn connect together via the high speed interconnects which are usually Cat - 5 or 6 cables [16]. This technology offers a centralized content storage, redefines workflow through a decentralized signal processing and management thereby improving efficiency and productivity.

The concept of Audio over IP is described by Banes and Noble's as one of the broadcast's hottest topics. Two of the best-known names in broadcast engineering have teamed up for a book on one of broadcasting's hottest topics: "Building audio networks using Audio over IP". Steve Church, founder and CEO of Telos Systems, and Skip Pizzi, well-known consultant and Contributing Editor for Radio World newspaper, have collaborated on Audio Over IP: Building Pro AoIP Systems with Livewire [17].

Steve Church and Skip Pizzi's book, "Audio Over IP" a two hundred and eighty (280) page book is described as a comprehensive look at AoIP that not only introduces readers to the technology and advantages of IP-based distribution systems, but also acts as a reference for IP audio distribution system development. The book also provides thorough coverage of how to design, build, configure and troubleshoot an IP Audio system, and how to interface it to PCs and the Internet.

The contributing editor of Radio World, also having recognized the significance of this developing trend, devoted an entire publication to the subject of Internet Protocol Audio titled "Using Internet Protocol for Audio and Broadcast – An Internet Protocol guidebook for radio. It was written by Skip Pizzi, in that special edition, Pizzi recognized the significance of Internet Protocol – based audio trend and saw it as an entirely developing new way of studio planning, design and system integration that would provide universal connectivity between many different types of products in the broadcast signal chain and throughout the installed network [18].

Carl Lindemann, in his article on www.proaudioreview.com March 2008 titled IP-based audio is foundation for major makeovers depicts Internet Protocol based audio as the major foundation for makeovers he said that “a massive transformation in radio’s technical infrastructure is underway replacing the analog infrastructure with next-generation digital systems”. He pointed out the usefulness of Internet Protocol-based systems as he quoted John Voci, director of radio stations for WGBH Radio Boston; “Internet Protocol-based systems have the obvious advantage of tremendous flexibility as it enables any studio, control room or device to be accessible throughout the facility. From an installation perspective, it also has tremendous advantages [19].”

Embracing Internet Protocol based system not only solves the immediate problem associated with conventional systems but also helps to prepare for the future in realization of vision 2015. Gary Kline Cumulus vice president of engineering and IT stated thus “we’re preparing for the future by adding Internet Protocol Technology into our infrastructure in a major way, we’re also eliminating sound cards by formalizing our IP driver policy [20].” They opted for this technology after careful consideration and with the convictions that implementing Internet Protocol based system would lift their company to a different level other than the traditional way of doing things. He commented thus “Much of the discrete and traditional circuit-switched approaches used in radio can now be replaced by a more modern, packet-switched interface style; it is expected that the technology eventually will become the dominant mode of audio transport in broadcast facilities”. All these articles provide the basis for the use of Internet Protocol audio technology in the studio today. They also provide insight to the benefits of this technology and descriptions of the advancement of the studio technology.

2.9 INTERNET PROTOCOL

The Internet Protocol is the lingua franca of the internet that defines the set of rules and standard for communication via the internet. It is designed for use in internetworks. Internet Protocol provides extremely flexible routing as well as reduced communication cost, it provides a manufacturer independent standard for transporting data over short and long distances. Internet protocol provides for transmitting datagram from sources to destinations, fragmentation and reassembly of long datagram if necessary. The function or purpose of Internet Protocol is to move datagram through an interconnected set of networks. This is done by passing the datagram from one module to another until the destination is reached. The datagram are routed from one module to another through individual networks based on the interpretation of its address. IP Addresses are fixed length of four octets (32 bits) or (128 bits for IPV6) for unique identification of devices on a network. An address begins with a network number, followed by local address (called the "rest" field). There are five classes of internet Addresses. Class A, Class B, Class C, Class D and Class E.

2.9.1 STRUCTURE OF IP

The Internet Protocol is layer structured to provide efficient communication between different kinds of computers. Several versions of this layering model are documented. They have five layers as illustrated in fig 2.7:

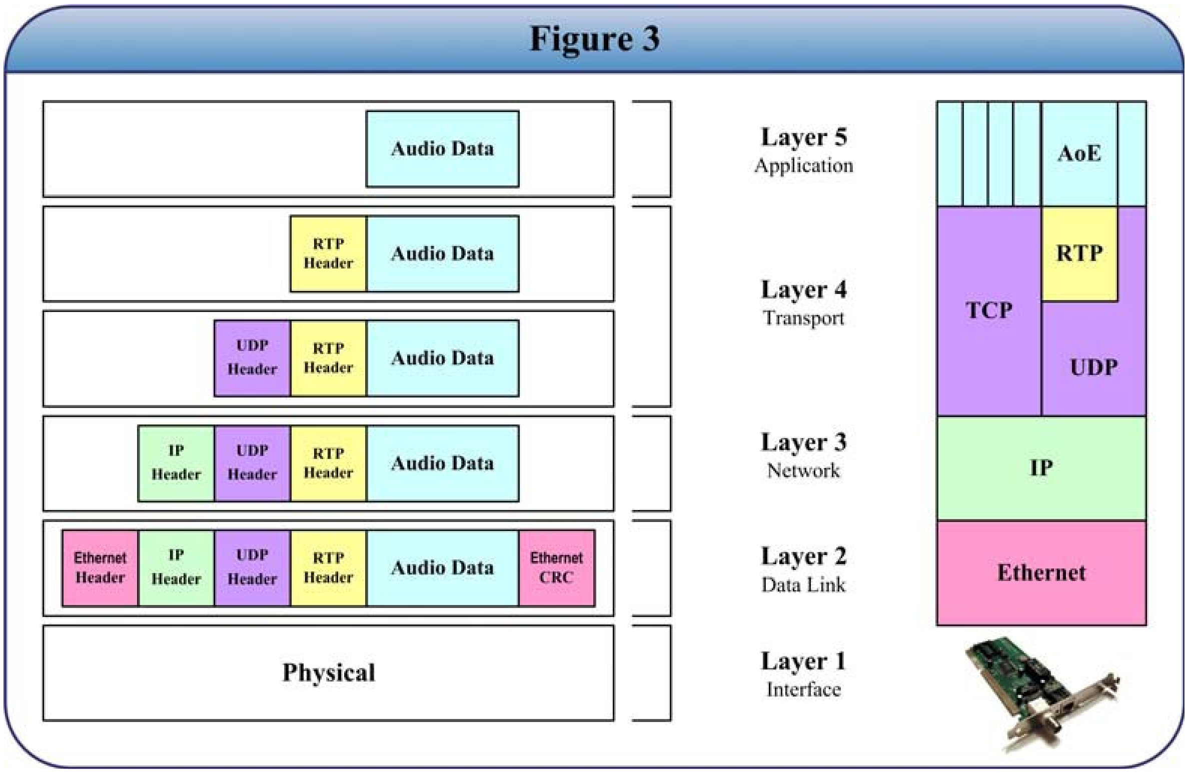


Fig 2.7: A five layered concept for IP audio over ethernet

2.9.2 IP MULTICASTING

IP multicasting is defined as the transmission of an IP datagram to a "host group", a set of zero or more hosts identified by a single IP destination address. A multicast datagram is delivered to all members of its destination host group with the same "best-efforts" reliability as regular unicast IP datagrams, i.e. the datagram is not guaranteed to arrive at all members of the destination group or in the same order relative to other datagram. This system utilizes IP multicasting.

An IP based system is built using conversion boxes that take analog or digital signals in (or out) and convert them to (or from) Ethernet packets which are then distributed LAN fashion using one or more intelligent Ethernet switches that have been set up to manage the audio distribution. This is a complex process that requires attention to detail and careful planning. AoIP broadcast system is a typical IP based system.

The broadcast chain still consists of chain of specific tasks with little overlap. From production studio to On Air studio, to Master Control Room, to Studio Transmitter Link (STL) and finally to the transmitter, a chain of highly specific tasks is established with defined interaction but virtually little or no interoperability. The technician sitting at the On Air console has almost no link to what happens at the transmitter site. The common data platform of IP, already well-established in the computer networking environment and AoIP, have opened doors for the cost-effective transfer of digital audio and program-associated data in the professional production space, providing universal connectivity between many different types of products in the broadcast signal chain and throughout the installed base of IP networks.

Furthermore, broadcasters that operate nationwide networks need to manage highly complex interactions to deliver their programs, including: multiple contributions, localized programming, localized advertising, complex audio transport, remote transmitters, etc. Using traditional technology, these tasks often turn into nightmares and each failure rapidly turns into a catastrophe. With AoIP, the future need of radio broadcasting industry is addressed by providing reliable and smart equipment that takes care of both Radio automation tasks in the studios; as well as program sharing and distribution throughout the network through IP audio networking.

CHAPTER THREE

3.0 METHODOLOGY, DESIGN ANALYSIS AND SIMULATION OF AoIP BROADCAST SYSTEM

This chapter deals with the complete design, analysis and simulation of AoIP broadcast system. It gives a detailed explanation of AoIP system hardware and software design with their respective functions.

3.1 AoIP BROADCAST SYSTEM PLAN

The planning of AoIP broadcast system has this primary objective - to develop a platform that will work as model for audio data communication on a single IP. The block diagram below represents its developmental stages.

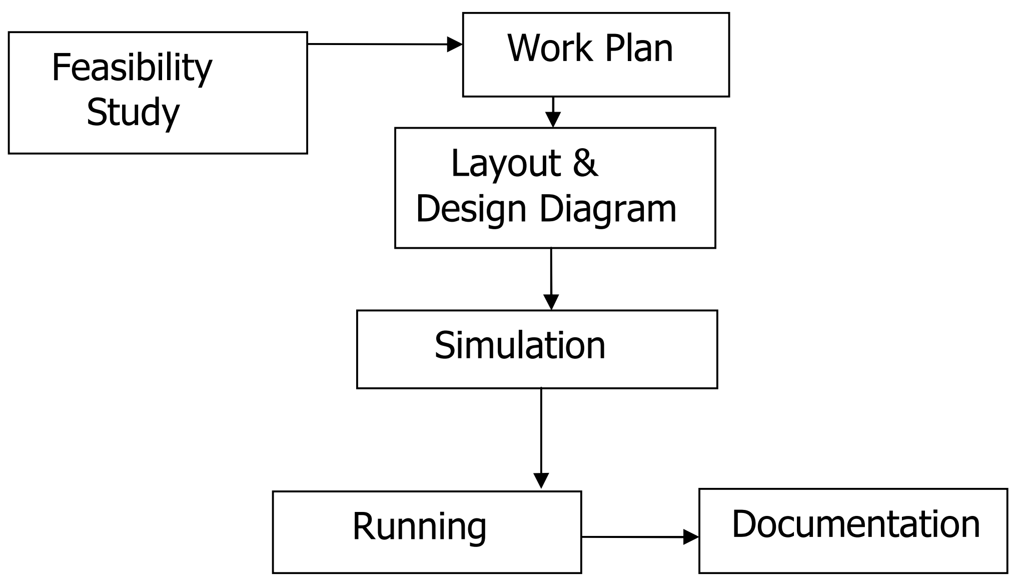


Fig 3.1: Block Diagram showing AoIP System Platform Development

In a nutshell, this platform is a six–stage platform comprising of feasibility study and scope definition, Work Plan, System Layout and Design diagram, Test running and Documentation. These are further discussed in the sub-sections below.

3.2 FEASIBILITY STUDY AND WORK PLAN

Feasibility study and work plan is the AoIP broadcast system initial stage which involved thorough inspection and survey of station studios, editing office and master control room. AoIP broadcast system was carefully planned through thorough feasibility study and assessment of the studios and the editing offices of Federal Radio Corporation of Nigeria, Headquarter, Abuja with an interview session with the studio manager after which a sketch/work plan of the proposed system was made to cover the infrastructural requirements. Table 3.1 summarizes the observations and findings; as well as the proposed system.

Table 3.1: Summary of observations and findings`

Mode of Transmission	Analogue
Studio Connection Method	AES-3 , Patch Panel
Type of Operation	Traditional and Stereotype
Transmission Method	Vsat, Audio Streaming, TVRO
Type of Mixer	Standard 24 channel non AoIP compliant analog/digital mixer
Cabling and Networking	Cat 5 Cabling
Signal Output	Good

Table 3.2: Average Percentage rating of the existing system`

S/NO	**FACTOR**	**RATING**
1.	Office Space	35%
2.	Equipment Stability	40%
3.	Redundancy and Server implementation	5%
4.	Equipment Upgrade	50%
5.	Studio Staff Strength	50%
6.	Staff Training	30%
7.	Quality of output	60-65%

Table 3.3: Proposed AoIP System`

Mode of Transmission	Digital
Studio Connection Method	Ethernet based
Type of Operation	Digital and Flexible
Transmission Method	Vsat, Audio Streaming, TVRO
Type of Mixer	Standard 24 Channel digital AoIP mixer with mixing engine
Cabling and Networking	Cat 6 Cabling/ Fibre Optic
Signal Output	Excellent

Table 3.4: Percentage rating for the Proposed AoIP System

S/NO	FACTOR	RATING
1.	Office Space	80%
2.	Equipment Stability	85%
3.	Redundancy and Server implementation	95%
4.	Equipment Upgrade	85%
5.	Studio Staff Strength	50%
6.	Staff Training	70%
7.	Quality of output	90%

The bar chart below analyses the two systems.

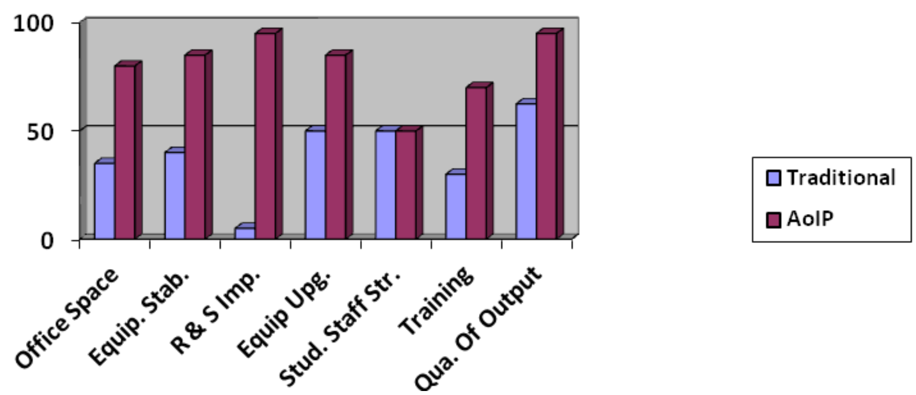


Fig 3.2: Bar Chart analyzing the two systems.

From the above chart, it is obvious that the existing system unlike the proposed AoIP system rates low in most of the factors. Space for future expansion is an issue for the existing system. The existing system has record of unallowable loss of airtime. In broadcasting, this is a critical performance and service delivery factor, as loss of air time, downtime is not allowed. Secondly, it is characterized by occasional program/failure due to problem of equipment instability and lack of redundancy. This system is at the transition stage as such is still broadcasting analog and old fashioned with some features of digital technology; the AES-3/EBU connection method in use offers one-way single source per cable. The reliability of workforce is not supported by redundancy and server operation. On the other hand, the proposed AoIP system would be designed to be spacious, equipped with highly stable modern digital equipment, AoIP compatible mixer, intelligent switch etc. Redundancy and Server application is one of its attractive features. It entails staff training and retraining program, with all these, service delivery and improved output would be guaranteed even with the same number of staff; thereby eliminating all the hitches of the existing system.

3.3 PROPOSED AoIP BROADCAST SYSTEM

The block diagram below depicts the complete AoIP broadcast system.

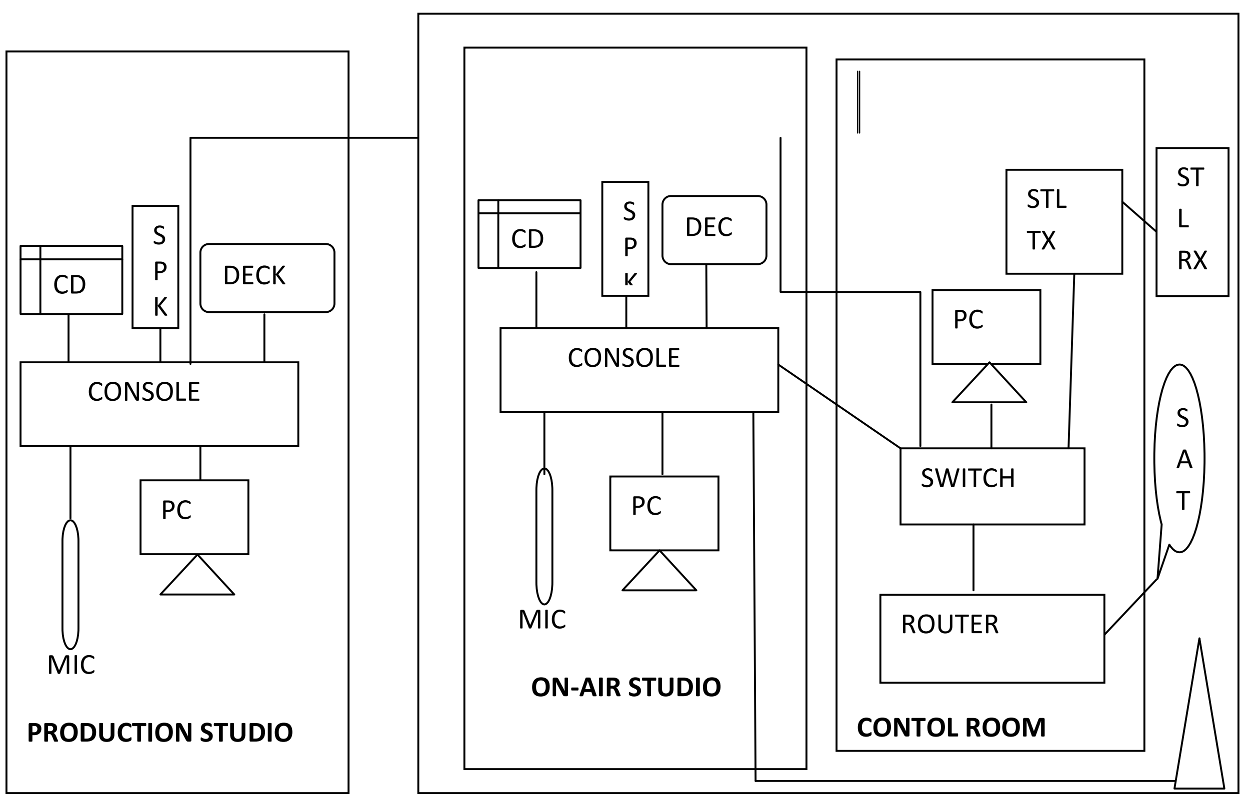


Fig 3.3: Block diagram of complete AoIP broadcast system

3.4 AoIP BROADCAST SYSTEM HARDWARE ANALYSIS

AoIP broadcast system comprises of two identical studio units, the Production or recording studio and the ON-Air studio. The ON-Air studio is sub divided into the studio unit, the control room unit and the transmission unit. AoIP studio hardware is comprised of CD player, radio deck, Microphone, speaker, mixing console, and personal computer, Studio transmitter link, internetworking switch, router, transmitter, and the satellite dish. They are grouped into audio input devices, audio mixing devices, audio output devices, audio mixing and routing devices, audio storage devices and audio transmitting devices.

3.4.1 AUDIO INPUT DEVICES

The audio input devices for this design include microphone, CD player and radio deck. These devices are used to send (Play-in) and transfer audio signals into the mixer. The microphone is the first input device of the system. Microphones are highly sensitive devices used to input voice signals and other sounds. For this design work, Shure KSM 27 microphones is ideal. It has the following excellent capabilities - extended frequency response, low self noise, exceptional low-frequency reproduction, high output level, high input SPL capability, no crossover distortion and extremely uniform cardioids polar response [21]. It is connected to the console via any of output lines 1 to 24. Shure microphone accepts acoustic sound signal from the producer, announcer, newsreader and presenter, converts the accepted signal into audio signal and transfers same to the mixing console.

The CD player and Radio deck like the microphone send in audio signal into the system. They are also used to cue in music, cut inserts; the CD/DVD player and radio deck are similar to domestic models in shape and operation. These devices are similar in operation, they accepts digital input signal from the compact disks and transfers the audio signal to the mixing console. For this design, Denon advanced dual CD/DVD and cassette player is used with the following unique features - easy maintenance removable drives, alpha Track Play permits independent, simultaneous playback of 2 tracks from the same disc, create up to 4 seamless loops per track or disc, 4 hot starts, 2 - 15-second onboard samplers, Output volume, reverse mode, looping, scratch & pitch, simultaneous seamless looping and sampler playback 6 stutter-cue play points. These devices are connected to the mixing console through the audio-input interface.

3.4.2 AUDIO RECORDING AND PLAYBACK/PLAYOUT DEVICES

The audio recording and playback devices for this design include the desktop computer, Marantz and MP3 player. The last two are used for outside assignment hence they are not reflected on the design diagram. For this design, HP dx 7800 is used with the following specifications: Intel Pentium dual core processor, 320Gb hard disk, 2GB DDR2, Windows XP operating system, 17" Monitor. HP dx 7800 is installed with the Vx222 digigram sound card, audio vault 10.0 Adobe audition 3.0 and a good antivirus. HP dx 7800 is connected to the mixing console via the AES digital output interface cable of the digigram. The mixed audio signal from the mixing engine is transferred to the computer via the AES digital output port. This signal is edited and stored in the appropriate folder in the server in the control room. For playout, the audio vault/ cool edit installed in the HP dx 7800 is used to access the required program, jingle, commercial, music from the server, the accessed file is scheduled appropriately and played out. This is actualized with the audio vault and cool edit. The audio vault AV-air has the capability to schedule between ten to fifteen files at a time.

3.4.3 AUDIO MIXING DEVICE

The audio mixing device for this design is the console. The console is the centrepiece/central processing and control device of the AoIP broadcast system; all other equipment microphones, CD-players, cassette recorders, mini disc etc are linked to it. The console controls what goes on air, what is produced and recorded. It allows one to switch from one source of sound to another and lets one mix together the sound from other hardware devices. The console slide control or rotary knob faders determine the level of audio signal from the device. The wheatnet E-6/24 mixing console is used for this design. Its frame is a two rack unit fanless network I/O cage with eight digital inputs and outputs and 24 logic ports. This mixer have the following attractive features- 24 input faders with E-Sat I/O equipped with E-Net card, four analog and digital input cards, inbuilt sound level indicator meter, ESAT 24 logic ports, event recall, bus minus, mix minus, four aux mixes all with dedicated talk back system and four monitor outputs. It integrates flawlessly with E-series satellite digital audio network router and bridge. This mixer allows for easy creation of small and large platforms that are exceptionally user friendly and flexible. It is equipped with a user supplied VGA display monitor via the embedded E-6 graphical user interface (GUI) providing real time graphic displays, production tools and setup screens with an RJ-45 Audio transport mixer connector to link it to the network. This device is connected to the switch via the RJ-45 connector and routes the audio input and output signal and control logic using its ESAT 3232-24. It accepts E Series I/O AI-NC4 – 4 Stereo (8 Mono) Analog Line Input Card, AO-NC4 – 4 Stereo (8 Mono) Analog Output Card, DI-NC4 – 4 AES Input Card, DO-NC4 – 4 AES Output Card, LIO-NC –12 Port Universal Logic Card.

AoIP signals from various input and storage devices are mixed using this device and ready for routing.

3.4.4 AUDIO OUTPUT DEVICE

The audio output device for this design is comprised of the speakers and headphone. These devices are used to play out and monitor the quality of the studio output from the mixer and control room. If the output is poor, necessary adjustment would be made. Loudspeakers and the fundamental technology behind speaker design has remained very much the same since the first electro-acoustic transducer was invented by Alexandra Graham Bell in 1876. Today, passive speakers have made way for amplified speakers, or more commonly known as the active studio monitors, and have dominated most studios. The advantages of a dedicated amplifier-speaker match are further enhanced with active digital crossover filter design instead of a passive filter topology. The Adaptive digital signal processing (DSP) based digital equalizer techniques, first adopted by hi-fi industry provides a viable and inexpensive solution to poor speaker output [22]. For this design, Behringer Speakers are used. Speakers are connected to the console via any of output lines 1 to 24.

3.4.5 AUDIO SWITCHING AND ROUTING DEVICE

The audio switching and routing device for this design is comprised of the switch and the router. These devices are used to interconnect every source and destination in the system and allows routing via TCP/IP.

The switch is used to interconnect and network the AoIP broadcast system devices. It links the production studio, ON-Air studio and other units. The HP Procurve 2600 -8 PWR switch is used for this design. This switch is an ideal switch for Power over Ethernet and provide up to 15.4W per port. It has many powerful inbuilt diagnostic features including alert for various error conditions and a full set of statistics for each port. A redundant external power supply is also available as an accessory. It support trunking across modules up to six (6) trunks, each with up to four (4) links /ports; and provides redundant links while preventing network loop with IEEE 802.1D and IEE 802.1W (rapid convergence spanning tree protocol) allows for increase in network uptime through faster recovery from failed links. Its basic IP routing feature enables automatic routing to the connected VLANs and up to 16 static routes including one default route in IP networks.

The audio routing device is used to route packets across the AoIP network. The routing device used for this design is the wheatstone bridge. This device is a four rack unit Networked I/O Cage that accepts bridge series Input/Output plus. Its design consists of seven inch rack mount digital routing cages, each capable of handling 512 simultaneous audio channels on its backplane compact enough for small applications, yet stackable for tremendous growth potential. It houses the ET -2001 Card a 16 x 16 Streaming AoIP Card. It contains a frame capacity of 19 Universal Slots plus Network card. The cage design offers a rear backplane that accepts the input and output connector cards, while the processor cards slide easily into the front. This design seems to be ideal for switching cards between each other for troubleshooting, or switching components to replace a dead audio source. The backplane is a dual-sided connection point for the front and rear components. The rear I/O card is inserted to match with its corresponding processing card in the front. Cards can easily be swapped. The backplane connectors are fixed and are the only potential point of failure. In addition, the ET-2001 card is smart enough to block changes when the fader is turned on, thereby eliminating the accidental switch of a source while it is still live. The IP address and subnet mask are stored in EEPROM memory of ET-2001 card.

All AoIP hardware devices must be attached to the switch at 100Mbps, full duplex network links to provide sufficient bandwidth and eliminate collision. AoIP system requires good bandwidth management. The HP Procurve power switch and wheatnet bridge are connected to each other via the RJ-45 port using a straight cable.

3.4.6 AUDIO STORAGE DEVICE

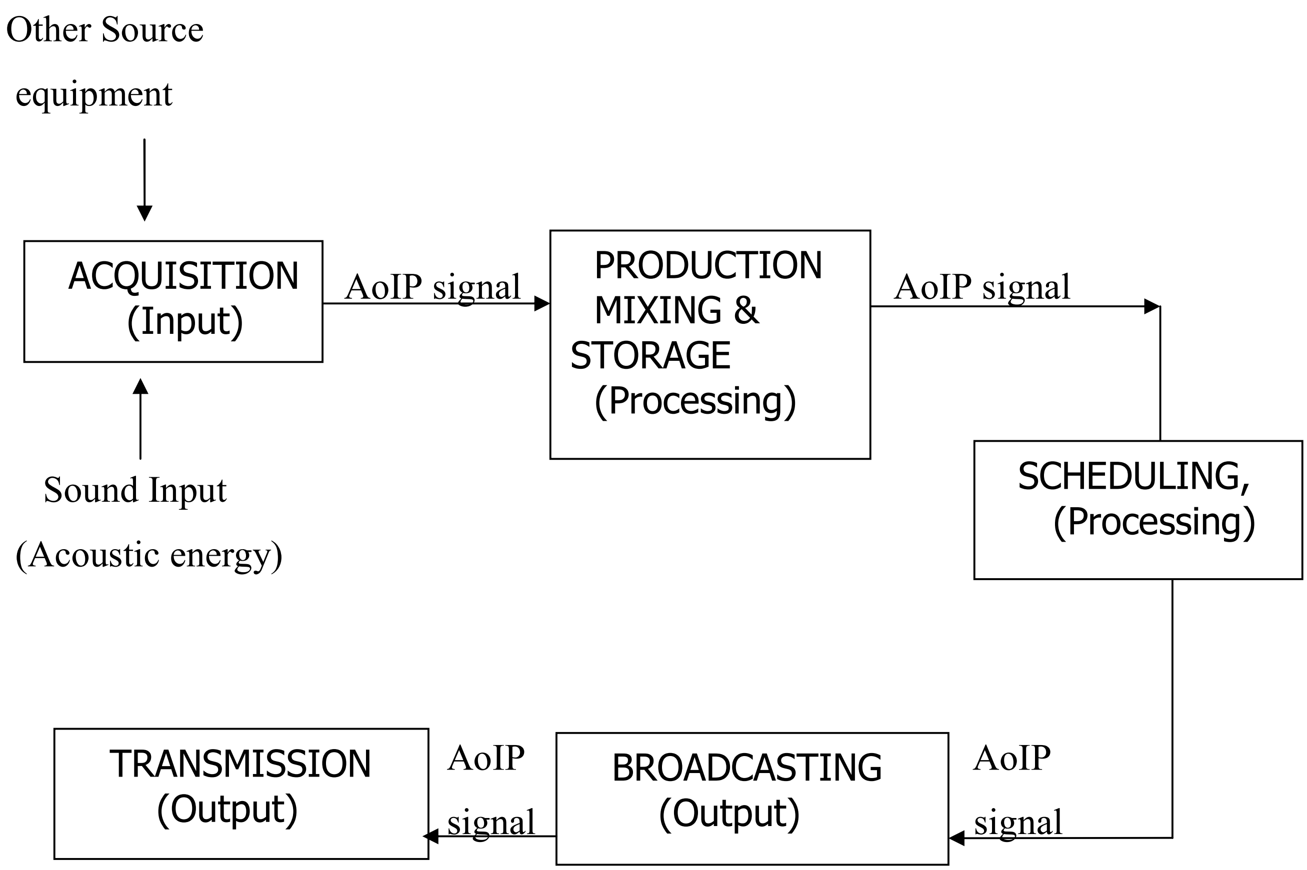
The audio storage device for this design is the server. The server houses the database of the AoIP broadcast system and stores all the audio files of the system, the programme, commercials, jingles, news, music etc. For this design, DELL 560TM is deployed with the following specification – Intel Dual Core processor, 1.5TB Harddisk, 8GB RAM, Microsoft server 2008 operating system, SVGA display, 17" flatscreen monitor, Audio Vault 10.0, Adobe audition, Vx222 Digigram, a good server antivivirus and WsNetServer utility for discovering and changing of the assigned IP address of ET-2001 card. This system allows for maintenance of a completely digital audio chain. The station's music is ripped from CD/DVD stored in music folder of the server, programs, news, commercials, adverts and jingles are also stored accordingly, played out through the AES-3 output of the automation system and routed digitally until it becomes RF. The WSNetserver software shows a visual overview of the system routing and allows the user to make changes and restrict or permit specific routing. The server is installed in the control room and connected to the HP procurve switch with a straight cable via the RJ-45 port

3.4.7 AUDIO TRANSMITTING AND OUTPUT DEVICE

The audio transmitting device is the studio transmitter link (STL), transmitter, the satellite dish and the streaming device. The STL is wired to the mixing console. The AoIP signal leaves via the output line of the mixer, STL receives the AoIP signals from the output port of the mixing console, and is amplified severally. AoIP broadcast system uses IP studio transmitter link. This STL comes in pair - the transmitter and the receiver (Tx and Rx). The STL Tx is installed in the control room; AoIP signal routed from the mixer to the STL Tx is thereby transmitted with this device and then encoded and broadcast using the satellite link. The transmitted signal is received by various network stations using STL Rx. The received signals are finally converted to radio signal and sent to air via the antenna of the transmitter as final output. This can be picked or received by various listeners/ audience with radio receivers within the frequency range area. The frequency range for transmitted signals is purely within the FM frequency range (88-108MHZ). The streaming device provides a redundancy/ backup link used to broadcast AoIP signal via the internet.

3.5 AoIP BROADCAST SYSTEM SIGNAL FLOW

The signal flow for the AoIP system is represented in the block diagram shown in fig 3.4:



ACQUISITION - Getting content into the Library

PRODUCTION - Putting content into a suitable format

SCHEDULING - Arranging content into a suitable 'running order'

MIXING- Combining content to a quality and desired form using console

STORAGE – Saving content in the computer

BROADCASTING – Sending out programme either 'Live' or Recorded

TRANSMISSION - Delivering the content to the listeners

Fig 3.4: Signal flow for AoIP Broadcast System

3.6 AoIP BROADCAST SYSTEM RECORDING PROCESS

Continuous Analog signals from the Presenter, Announcer or Newsreader are sent to the microphone, the microphone converts the sound energy to electrical energy still represented in continuous varying amplitude. This is transferred to the mixing console for mixing. During digital recording of the signal, analog to digital (A/D) conversion, filtering, sampling, coding, quantization takes place. Signals are converted from continuous time-amplitude coordinates to discrete time-amplitude coordinates. The difference between the instantaneous analog signal and the digital representation is digital error introduced in the signal – digital noise by conversion. Fig 3.5 below depicts this.

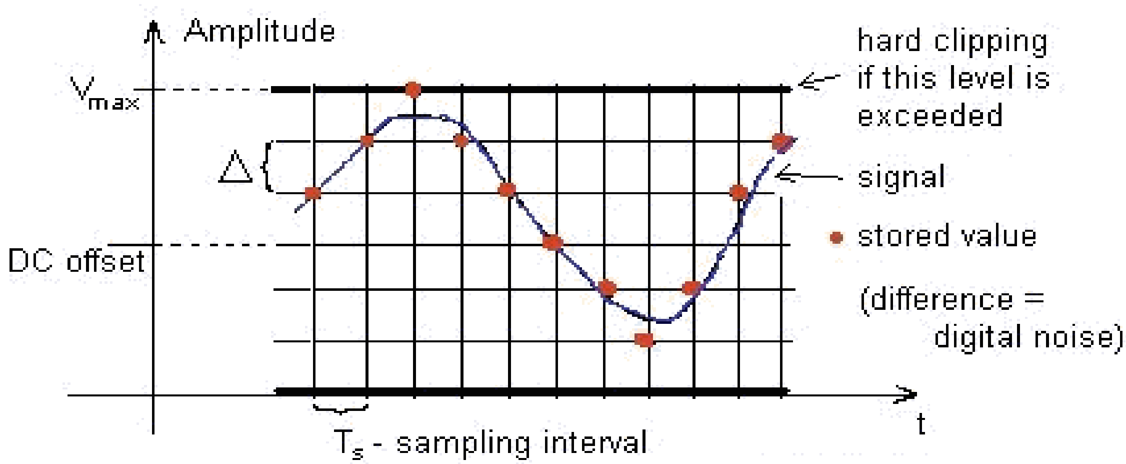


Fig 3.5: Conversion of analog signal to digital signal using an A/D (or D/A) converter.

The four staged process have been explained in section 2.5. The audible human hearing frequency is 20KHz, as such AoIP signal is recorded to the compact disks and sampled at a frequency higher than twice this frequency (Nyquist theorem) ie higher than 40KHz. therefore sampling rate of 44.1KHz is chosen.

For the t coordinate (6•N [dB]) Nyquist theorem states that if a signal V(t) does not contain frequencies higher than $f_s/2$ (where $f_s = 1/T_s$), then it can be fully recovered from its sampled values $V(nT_s)$ at discrete times $t_n = nT_s$ where n = .. -1, 0 , 1 , 2 , 3...

$$V(t) = \sum_{n=\infty}^{\infty} V(n \bullet T_s) \frac{\sin[\pi \bullet f_s(t - n \bullet T_s)]}{\pi \bullet f_s \bullet (t - n \bullet T_s)} \quad \ldots\ 3.1$$

where:

$f_s = 1/T_s$, the sampling frequency

V(t) = value of signal at arbitrary time t.

The recovered signal will have all frequencies in the range of 0 to $f_s/2$ Hz.

For the amplitude (y) coordinate, generally, digital systems has signal levels and bit resolutions with logarithm to base 2 as the conversion index for converting number of available quantization levels to number of bits.

From the sample signal in section 2.2, the maximum amplitude is 6V.

Using $2^N >= V$, for 6V, we have $2^3 >=8$ …. 3.2

This is equivalent to 3 no of states. It implies that this system can store 3 bits of information with maximum of 8 possible states derived from the expression given below.

Amount of digital information $=\log_2 2^3$. Thus it requires a 3-bit resolution A/D and D/A converter.

When voltage amplitude of 0 to V_{max} is used, then one quantization step will be:

$\Delta = V_{max}$ / No of levels $= V_{max}/2^N$ …… 3.3

At adequately high level and complex input signal V(t) value, the digital error (difference between analog signal and stored digital value) would be statistically independent and uniformly distributed in the range of [$-\Delta/2$, $\Delta/2$] where Δ is the step size in the A/D converter.

Thus, the maximum Signal-to-Noise Ratio (S/N) in decibels is calculated as:

$S/N \equiv 20x\log(V_{SIGNAL\ RMS}/V_{NOISE\ RMS}) = 6.02 \cdot N + 1.76$ [dB] $\approx 6 \cdot N$ [dB]… 3.4

(for all, practical purposes).

Thus for our sample signal we have $6 \cdot 3$[dB] = 18dB. A/D converter resolutions of 8, 12,16, and 20 bits would allow a 48, 72, 96, and 120 dB S/N ratio respectively.

The block diagram below depicts the technical analysis of typical digital recording process which occurs as AoIP broadcast system signal is being recorded with the studio equipment.

The processes at each of the numbered blocks 1 to 7 are described below:

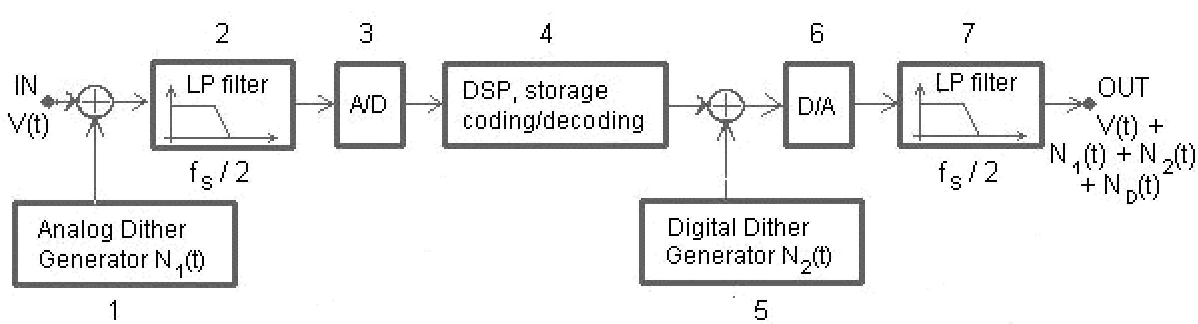


Fig 3.6: Block diagram of a typical digital recording processing system.

1. Optional analog dither generator is added to the input signal in order to

a) linearize the A/D converter

b) make possible improvement of S/N by averaging process according to

formula: (S/N) after averaging = (S/N) before averaging $\cdot\, n^{1/2}$ …… 3.5

where: n = No. of averaged signals.

c) eliminate harmonic distortions and intermodulation (created when digital noise $N_D(t)$ is coherent with signal V(t)).

d) eliminate "digital deafness" (when the signal V(t) falls below $\pm\Delta$, where Δ is the step size in the A/D converter, the signal will not be recorded at all unless there is a noise $N_1(t)$ on the input) and to eliminate noise modulation by the signal

2. Input low pass filter (antialiasing filter) eliminates all frequencies above $f_s / 2$, where f_s = sampling frequency, in order to avoid aliasing distortion (Folding of frequencies into passband: $f_{new} = f_s - f_{original}$ where $f_{original} \geq f_s / 2$).
3. A/D converter converts analog signal into a digital number (for example, 101 represents the binary coded 3-bit amplitude). Sampling speeds range from 2 kHz to 10 GHz and amplitude resolution ranges from 4 bits to 20 bits.
4. Add optional digital dither $N_2(t)$ to avoid digital distortions and coherent noise N_D (t) on the output of D/A converter for digital processing.

5. Storage of digital AoIP data is performed on optical disk and harddisk. Prior to storage, extra code is generated to allow for error correction. This error correction code allows detection and *correction* of errors during playback of the audio signal. Redundant information must be added to the original signal in order to combat noise inherent in any storage/communication system.
6. D/A converter converts digital numbers into analog signal. At conversion speeds between 2 kHz and 200 MHz and amplitude resolution between 4 bits to 20 bits.
7. Output low pass filter would eliminate all frequencies above f_s /2 generated during D/A conversion.

3.7 AoIP BROADCAST SYSTEM SIMULATION

AoIP broadcast system application was designed and developed with Visual Studio 6.0 Enterprise Edition. Visual Studio 6.0 has thirteen (13) types of project used to create applications namely:

- Standard EXE
- ActiveX EXE
- ActiveX DLL
- ActiveX control
- ActiveX Document EXE
- ActiveX Document DLL
- VB Application Wizard
- VB Wizard Manager
- Data Project
- DHTML Application
- IIS Application

- Addin
- VB Enterprise Edition controls

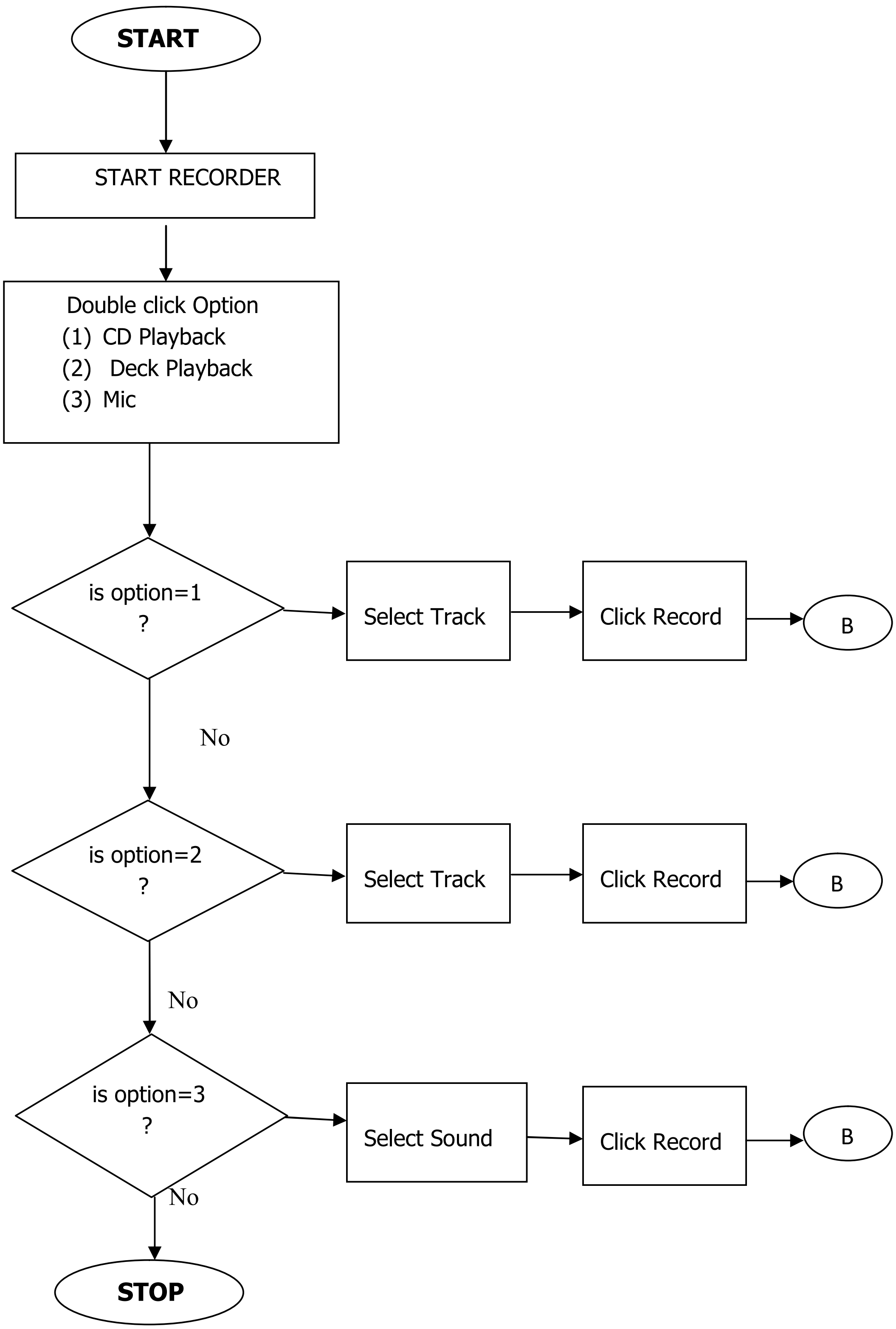
The Standard EXE type of project was used to design AoIP broadcast system simulator application. The application is basically designed in three modules namely:

- the production studio module
- the on-air studio module, the ON-Air studio comprises of the studio, control room and transmitter hall.
- the visual basic module

The system is divided into two for easy understanding of how the studio functions starting from when audio is introduced to its processing and finally its output.

3.7.1 AoIP PRODUCTION STUDIO PROGRAM DESIGN AND ANALYSIS

This module is the first module (Module 1), it illustrates the production studio, where devices like CD Player, Radio Deck, Microphone, Speakers and Recording Computer (known as editing PC) are connected to a digital Console. The flowchart given below represents and analyses AoIP production studio.


START
START RECORDER
Double click Option
(1) CD Playback
(2) Deck Playback
(3) Mic
is option=1
?
Select Track
Click Record
B
No
is option=2
?
Select Track
Click Record
B
No
is option=3
?
Select Sound
Click Record
B
No
STOP

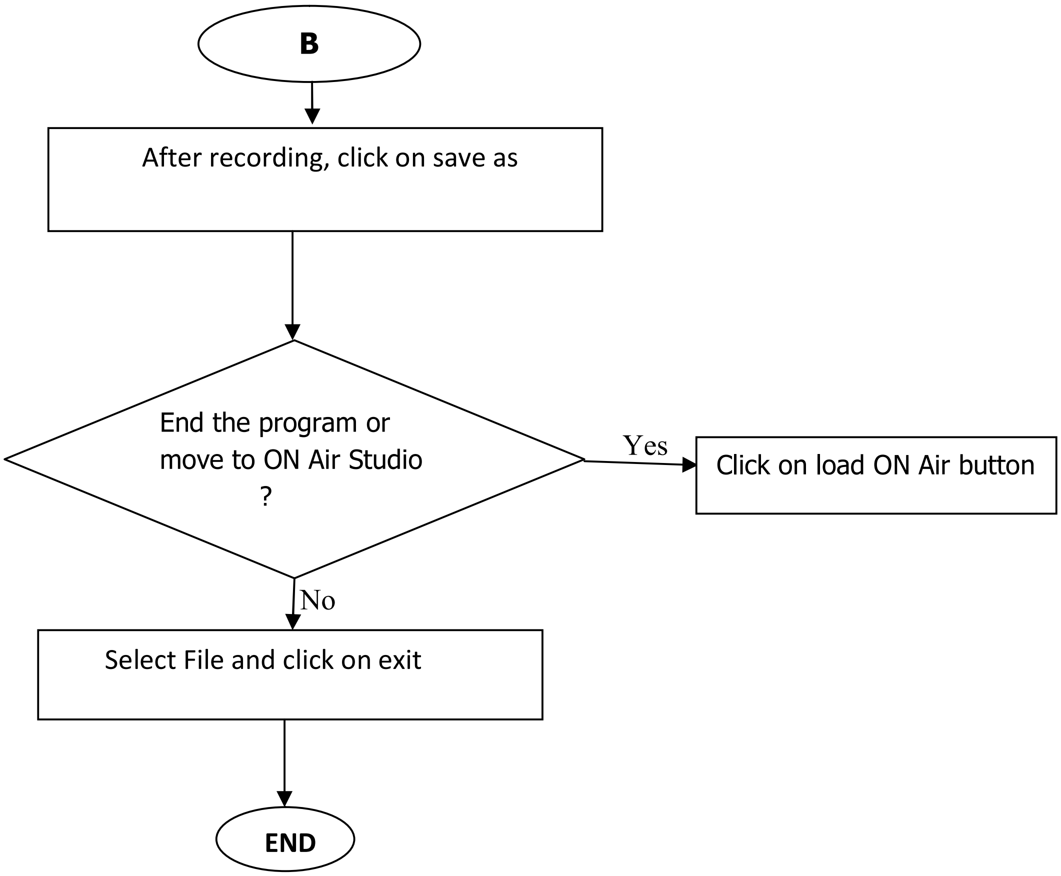


Fig 3.7: AoIP Production Studio Flowchart

The production studio module uses a standard form in which was placed image boxes that shows/represents the devices in a studio namely:

- CD Player
- Speakers
- Microphone
- PC
- Console

Double clicking on any of the CD player, Radio Deck or Microphone triggers the double click event procedure of the image box. This sub procedure contain lines of codes which activates the windows “OPEN” dialogue box that allows the user to browse the windows folders and locate a sound file already in the system and clicks open.

After the click of “OPEN” in the dialogue box, the sound file selected is attached to a public variable (declared in the module) and another form that has an instance of window media player is activated. In the “Load” event of the windows media form, the public variable with the sound file is attached to the media player’s “URL” property. This enables the media player to playback the selected sound file.

As the file is playing back the virtual speakers seems to be activated too. Also when the media player stops playing, the speakers goes back to normal. This is achieved by placing invisible circle shapes on the speakers. These circle shapes are controlled on and off by the window media player “STATUS_CHANGE” event procedure which is triggered by the state of the media player namely:

- play state
- pause state
- stop state

The double click event procedure of the recording PC system image contains codes that activate an application which records any sound the PC is playing. So any sound that either the radio deck, the CD player or the microphone is playing are being recorded by this recorder application.

The “Start Recording” button click event sub procedure of the recorder application uses a Windows API function called “Public Declare Function mciSendString Lib "winmm.dll" Alias "mciSendStringA" (ByVal lpstrCommand As String, ByVal lpstrReturnString As Any, ByVal uReturnLength As Long, ByVal hwndCallback As Long) As Long", which was declared in the module part of the application, to capture any sound playing in the PC system.

API stands for Application Programming Interface. There are many of them. These APIs are functions which Windows uses in its activities. But they are also available to programmers to be used within their applications. It provides specific functions just like the example cited above which captures sounds.

Included is a sound recorder to demonstrate the recording process the recorder was developed by a VB programmer called VB BEGINNER. The application is free on the internet (www.freevbcode.com). With little modifications, the application was incorporated in the studio simulation application. The "Stop Recording", "Playback Recording", "Stop Playback" buttons' click event sub procedures equally uses the API function above to achieve the stated functions of the recorder application.

The "Load On-Air" button click event sub procedure activates the On-Air module of the application.

The simulated output is as shown below:

Fig 3.8: Simulated AoIP Production Studio

3.7.2 AoIP PRODUCTION STUDIO OPERATION

1. Double click the cd player icon
2. A folder where you can select a music file appears, Select the music of your choice
3. Click on Open button
4. A media player appears at the right side of the form) as shown in figure 3.9.

Since we are simulating the studio, it is assumed that each device is connected to the console and the console to the switch and router. The same process is applicable to other devices except during recording.

3.7.3 AoIP PRODUCTION STUDIO RECORDING

The production studio is primarily used for recording and programme production. When you double click on recording system, the figure below appears.

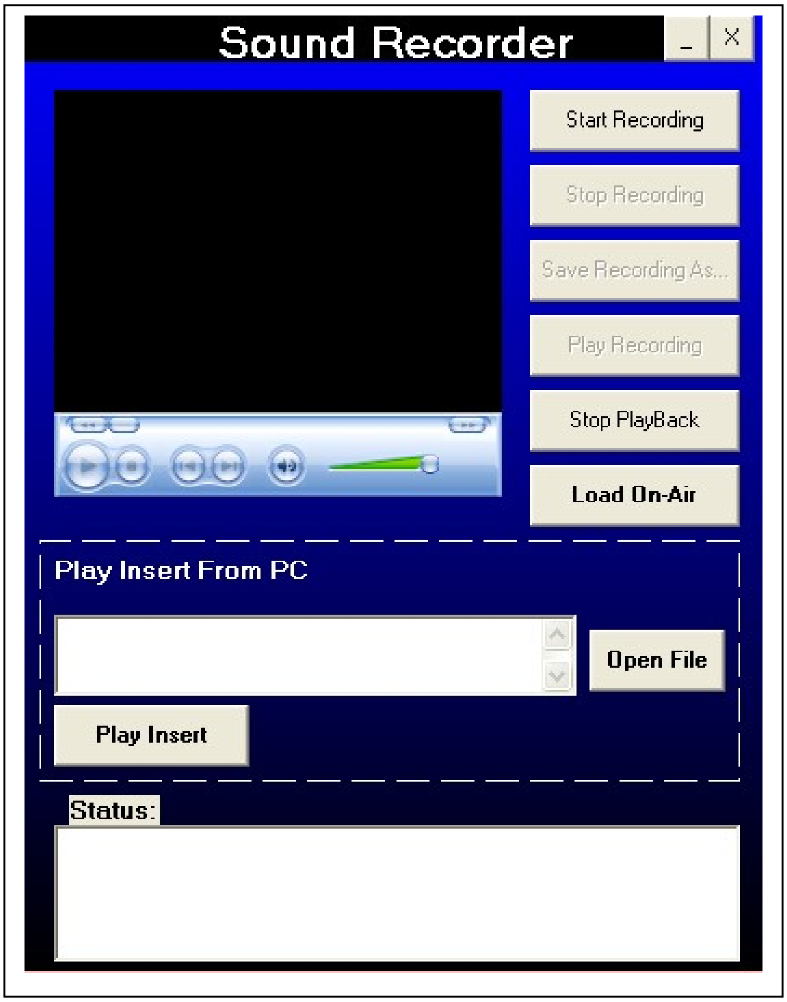


Fig 3.9: AoIP Production Studio Sound Recorder

3.7.4 AoIP ON-AIR STUDIO PROGRAM DESIGN AND ANALYSIS

This module is the second module (Module 2). It illustrates the ON-Air studio, where devices like CD Player, Radio Deck, Microphone, Speakers and Vault computer (known as airing PC) are connected to a digital Console to form a virtual studio. The virtual studio which comprise of the console, the CD player, Computer system, microphone and speakers, the virtual control Room comprise of transmitting STL, Network switch, router, and a computer system while the virtual transmitter hall comprise of the transmitter, receiving STL and a satellite dish are all embedded in the On-Air Module.

Like Module 1, this module also uses a standard form. The studio section was designed with image boxes that represent the components in a studio namely:

- CD Player
- Speakers
- Microphone
- A PC
- Console

The double clicks on either the CD player, the PC system or the Microphone triggers the double click event procedure of the image box. This sub procedure contain lines of codes which activates the windows “OPEN” dialogue box that allows the user to browse the windows folders and locate a sound file already in the system and clicks open.

After the click of “OPEN” in the dialogue box, the sound file selected is attached to a public variable (which was declared in the module) and another form which has an instance of window media player is activated. In the “Load” event of the windows media form, the public variable with the sound file is attached to the media player’s “URL” property. This enables the media player to playback the selected sound file.

As the file is playing the virtual speakers seems to be activated too. Also when the file stops playing, the speakers goes back to normal. This is achieved by placing invisible circle shapes on the speakers. These circle shapes are controlled on and off by the window media player "STATUS_CHANGE" event procedure which is triggered by the state of the media player namely:

- play state
- pause state
- stop state

Control Room section

The STLs indicates that sound signals are coming into it by the movement of the virtual equalizer on it. This was achieved by placing invisible circle shapes on the STL. With timer events, these circles were triggered on and off. Using random numbers generator in the timer event sub procedure of the timer object, the circles are made to appear in random order thereby making it look as if the played sound is controlling it. Also the timer object is controlled by the media player's "STATUS_CHANGE" event procedure which turns the timer object on and off depending on the state of the player.

The switch image also shows some activity in the form of little green lights to indicate that the network is working. This is achieved by placing small lines on the switch image.

These lines were controlled by timer objects. In the timer event sub procedure, these lines are flipped from black to green. Also random numbers are used to ensure that the process does not maintain definite pattern just like a real switch. In the PC system and Streaming router images' "MouseUp" event sub procedure, the mouse buttons actions are captured (in this case the right click button). When the right click button is clicked, a popup menu "Enter IP address” appears. If this menu is clicked, an IP Address form is activated where the user enters a Class C IP address format. If the two IP addresses (ie PC system and Streaming router) are correct, it activates a timer object. The timer object controls a group of lines which is flipped between visible and invisible thereby creating an illusion of movement.

Transmission Hall Section

In front of the satellite dish, three circles appear in sequential order, one bigger than the next and thereby creating an illusion of the dish emitting signals to the space. These three circles are also controlled by the same timer object that controls the lines between streaming router and satellite dish.

On the “Form Load” event sub procedure, a timer object is activated. This timer object controls the group of lines on top of the transmitting tower which comes off and on indicating that whatever that is produced in the On-Air studio is being transmitted into the atmosphere.

The flowchart given below represents and analyzes AoIP ON-Air studio.

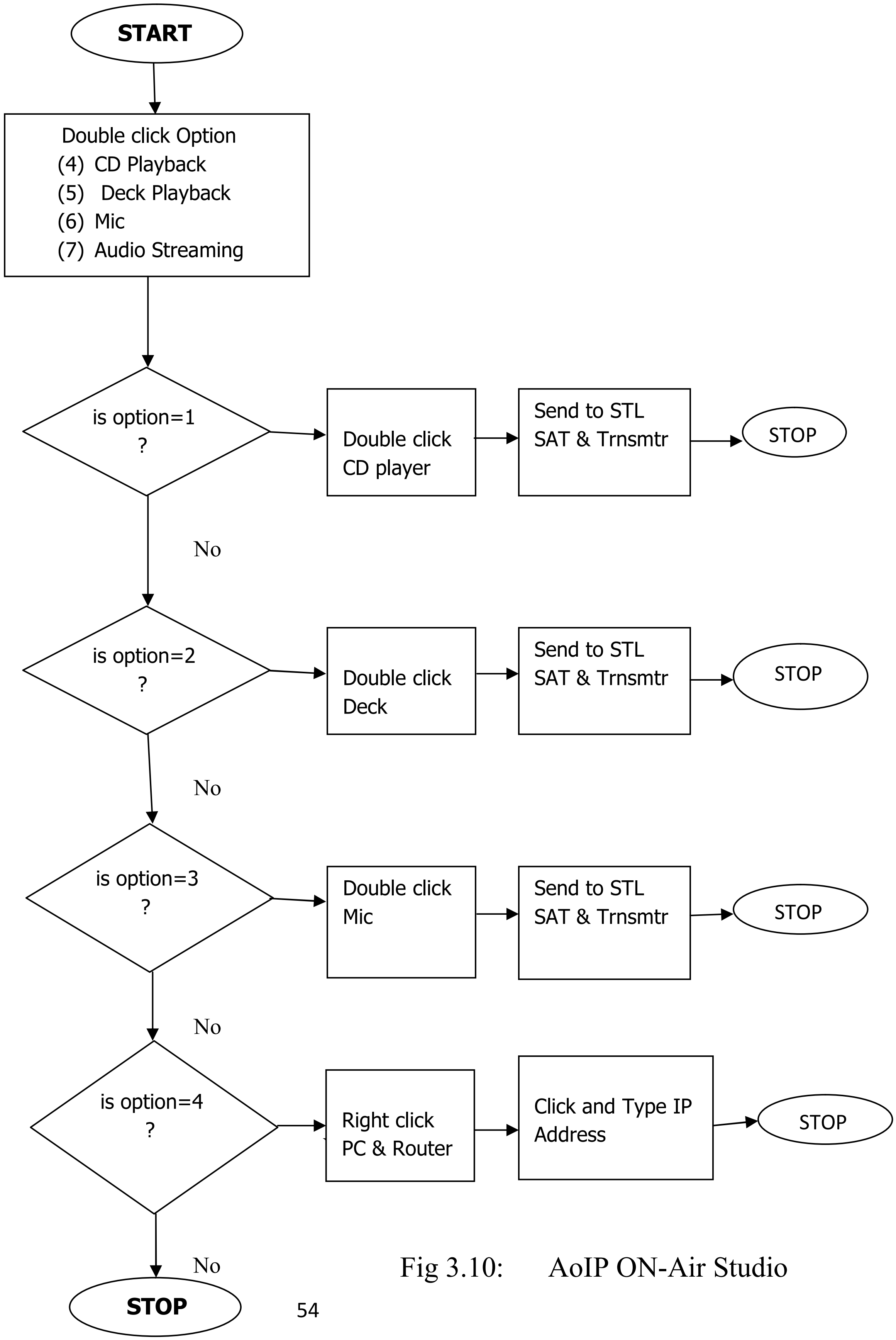
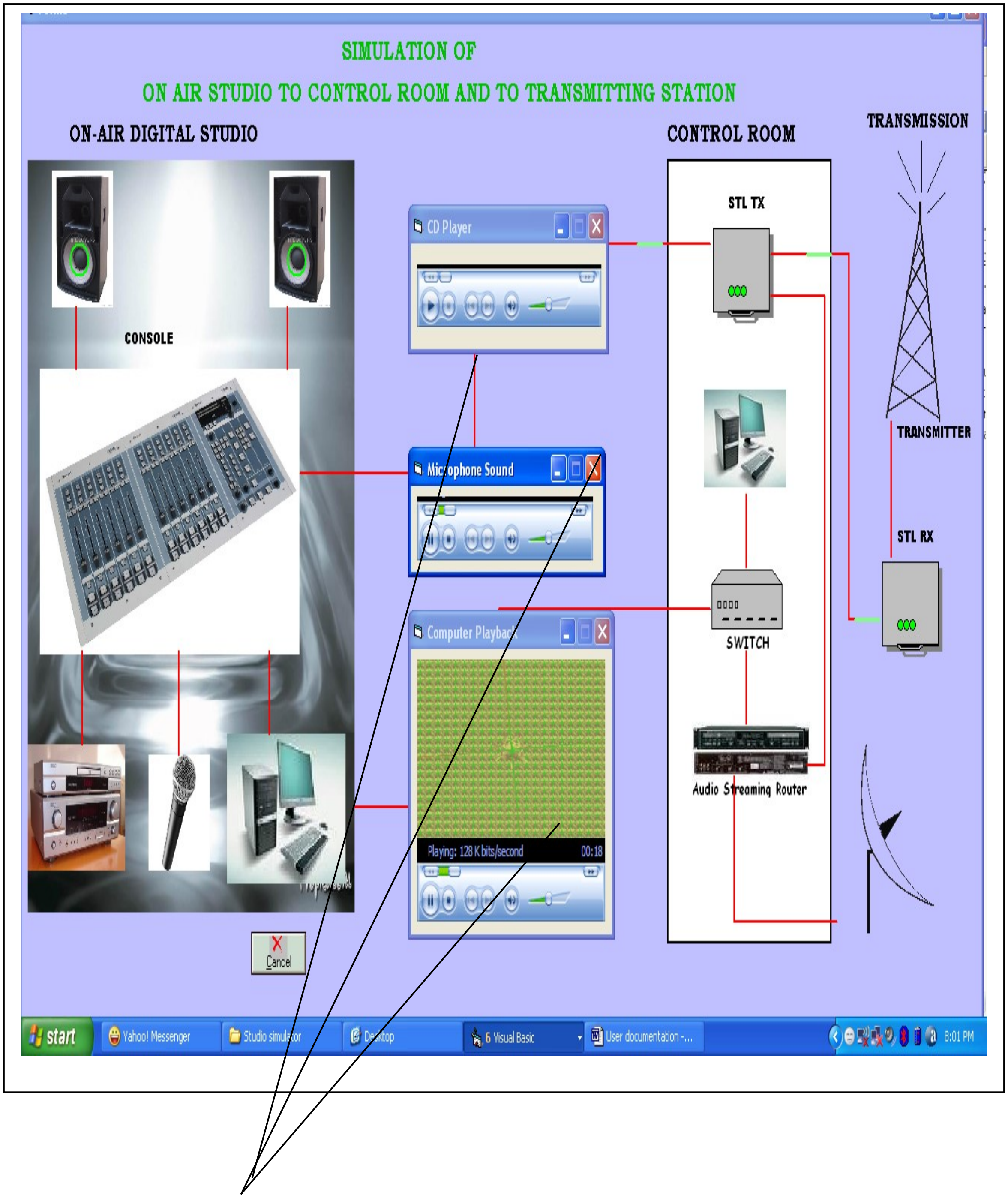


Fig 3.10: AoIP ON-Air Studio

The simulated output is as shown below:



CD playback, Microphone sound and PC playback respectively

Fig 3.11: Simulated AoIP ON-Air studio

3.7.5 AoIP ON-AIR STUDIO OPERATION

1. Double click load on air, or go to file select load on air.
2. Right click the router and then the computer to configure their IP address to stream.
3. Double click the computer in the on air to cue, schedule and air the program, music, commercial etc .

3.7.6 VISUAL BASIC MODULE

This module contains all public variable declarations, all the Windows APIs and public function procedure used in the application.

3.7.7 AoIP BROADCAST SYSTEM REQUIREMENT

The minimum system requirement for AoIP broadcast system is given below:

- Pentium III (PIII) system.
- Windows Operating system (Window XP)
- 128MB of RAM.
- 5GB of Hard Drive space free.
- Visual Studio 6.0 Enterprise Edition
- Microsoft Office (especially MS Access)

CHAPTER FOUR

4.0 CONFIGURATION, TESTING AND EVALUATION OF AoIP BROADCAST SYSTEM

This section brings us to the AoIP system configuration and testing. It outlines the vital configurations for AoIP broadcast hardware devices especially the HP Procurve Switch, the IP Aud WDM driver and the USB Security Dongle, the running and the testing of the virtual studio and the cost analysis. The HP Procurve 2600 switch configuration is given as follows. This can be achieved using a computer with an RS-232 serial connection in real life situation.

- Reset the switch to factory default by pressing the Reset and Clear button simultaneously.
- Continue to press the Clear button while releasing the Reset button until the Self Test LED indicator begins to flash.

  Using a Hyper Terminal emulator, the following standard configuration and settings are made:

Table 4.1: HP Procurve 2600 Switch configuration

Baud rate	115200
Parity & Flow Control	NIL (No Parity, No flow control)
No of Bits	8 bits
Stop bits	1 stop bit
Terminal Emulator	ANSI

- Press the Reset Button for the switch to perform a self test.
- Continue to press <Return> until the firmware revision prompt is displayed on the screen. The firmware revision displayed must be

H.08.60 or greater else follow the instruction on the manual to update the software.

- Press <Return> until the basic configuration menu screen appears.

Table 4.2: HP Procurve 2600 Switch Settings

System Name/System Contact	User defined eg “FRCN Procurve Switch”
Manager Password	User defined
Logon Default	Set to Command Line Interface(CLI)
Time Zone	Can be left to zero
Community Name	User defined eg “FRCN AoIP”
Spanning Tree	Enabled (This allows the switch to communicate with other network devices to support redundant link and prevent network loops.)
Default Gateway	This would be sets to the IP address you assign to the gateway on the local network to which the switch is attached eg 192.168.1.1
Time Synch Method/TIMEP Mode	TIMEP/disabled
IP address/Subnet Mask	192.168.1.2/255.255.255.0
Proxy ARP	No

- Then Select Save to save this configuration.
- Configure the Switch to map IP precedence to QoS.

    - Then set aside/reserve some ports for Non AoIP devices by default, the last 1/3 of switch ports are reserved for this. Block these ports from receiving multicast packets from other ports on the Switch with the “vlan 1 ip igmp block” command.
    - With the PC configured to the same subnet as the Switch for connectivity, log on to a web browser and type in the switch IP address to verify the set configurations.
- Finally, set the “Diagnostics” tab and press the “Configuration Report” button. This can be printed out to form part of the documentation with ports to all the equipment properly labeled.

4.1 IP Aud WDM DRIVER CONFIGURATION

The steps to install and configure the IP Aud WDM driver on the computers that houses the USB Security dongle and audio vault license is summarized below: the installation requires about 3.8MB harddisk space.

- Insert the installation CD and run setup.
- Read the licence agreement, then click “I agree”
- Select the following installation options – driver, PC X controller, configure firewall and install shortcuts.s Click next and click continue anyway (if windows indicates that the driver is not signed).
- Click close after successful installation.

To configure the required parameters for Audio Over IP driver control application on the audio automation or editing computer to play out AoIP streams onto the network and to play back AoIP streams from the network use the following parameters.

Table 4.3: IP Aud WDM driver configuration parameters

Network Interface	Default Adaptor for 1 adaptor or select the preferred adaptor for more than 1.
IP address	This would display automatically.
System ID	1,2,3 etc
Status Report Interval(s)	20 seconds
Number of channels (full duplex stereo)	6
Source Stream ID	1,2,3 etc
Multicast Group	This would display automatically.
Stop Audio on disconnect	Place a check on this option.

The first system would be assigned ID 1, subsequently increment the number by 1.

4.2 USB SECURITY DONGLE INSTALLATION & IP ADDRRESSING FOR AoIP BROADCAST SYSTEM

The USB security dongle is simply installed on the AoIP sources and destination computers by plugging it into any of the USB ports. The system uses a static IP addressing scheme with all the devices residing on the same subnet. Each device has a unique IP address but the same subnet mask. For this system Class C addressing scheme would be deployed with the following IP addresses assigned.

Table 4.4: AoIP System IP addresses

Device	IP address
HP Procurve 2600 switch	192.168.1.2
ET- 2001 card	192.168.1.141
Computers with IP Aud drivers	192.168.1.212,213,etc
Subnet mask	255.255.255.0
Primary Computer/Server	192.168.1.160
Failover/redundancy computers	192.168.1.161,162
Mixing Console	192.168.1.150,151

These configurations are automatically stored in the EEPROM memory on the ET-2001 card.

AoIP device configuration and settings are not limited to the ones highlighted above, the complete and detailed configurations and settings would be more applicable in real life implementation.

4.3 AoIP SYSTEM QUALITY OF SERVICE AND BANDWIDTH ISSUES

Good quality of service is a must. To ensure this, the HP Procurve switch device should be configured with the new 802.1p QoS setting used to prioritize the network traffic ie to give priority. With this, AoIP data could be sent at a higher priority so it gets first call for available bandwidth other real time stream for applications such as Voice-Over IP data are also accommodated. This makes it a perfect fit AoIP technology as well. Other normal office traffic and are moved across the network at a default low priority and ports without devices requesting multicast data would never be sent any. No device will source more data than what can be carried on a 100Mbps link. Ports can handle either a single ET card request or two fully populated AoIP PC's without overflowing the available bandwidth.

AoIP devices make use of TOS precedence field in the IP header to indicate the relative priority of each IP packet; normal network traffic has a precedence of 0. The HP Procurve switch from its configuration, would use the IP TOS precedence field to map the 802.1p QoS value. With this QoS prioritization scheme, the AoIP network would be protected from other network data.

4.4 AoIP BROADCAST SYSTEM PACKAGING

Microsoft Visual Studio 6.0 package and deployment wizard shown below is a good wizard that can be used to package AoIP broadcast system. This wizard enables the inclusion of crucial files into a distributable package and for proper running of software; as well as sending the package to a distributable environment such as a server. It is also equipped with object linking and embedding OLE an application creation and editing feature. This wizard has three options: one can package, deploy or manage the scripts in case of any administration or management modification.

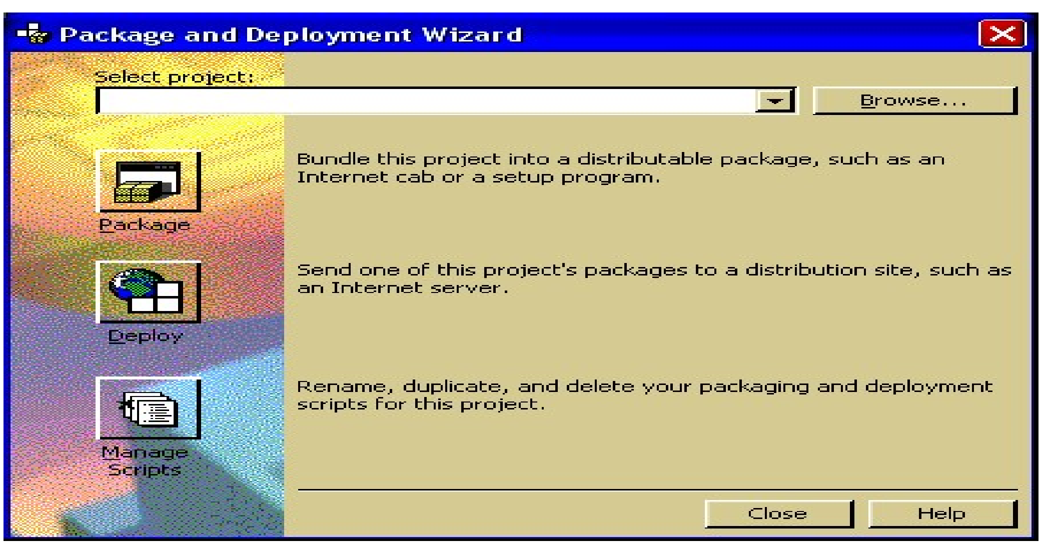


Fig 4.1: VB Package and Deployment Wizard

4.5 AoIP BROADCAST SYSTEM RUNNING AND TESTING

Running the simulated AoIP broadcast system is very simple and easy, the program code has been compiled to an executable file named FM simulator.exe. It requires approximately 8.22MB Harddisk space and runs on windows XP, Vista, and Windows 7. The system requirements have been given in a previous section. After installing, the location folder is C:/Users/friend/Documents/FMsimulator.exe. To run, double click the folder (if it is saved in a folder), Double click the .exe file, Click run (for windows vista). On starting, the simulator name displays my name, my matric number and institution and department and later loads the AoIP product ion studio. The virtual studio can be tested for recording with virtual microphone by Double click the computer icon, Click start recording, Double click the microphone, Speak/ talk/read to send in input signal. The system picks and records it automatically. To stop click on stop recording, To save, click on SAVE, To playback, click on playback, To cut insert into the program, select the insert.

The same process follows for recording with cd player or cassette player, it can be demonstrated with a live cd or cassette. Or with the icons on the written program. Demonstration with live players will give a better quality output during play back.

To load on air, Double click load on air, or go to file select load on air, Right click the router and then the computer to configure their IP address, Double click the computer in the on air to cue, schedule and air the program. The source code for simulated AoIP broadcast system is attached as appendix page 84.

4.6 AoIP USER GUIDE

Every information on the user guide has been explained in the previous sections; the following is a simple addition. The software can come in a CD or flash, to use, simply insert the device, then open the device and copy the entire folder into the harddrive, then send the icon to desktop for easy access (this is optional) then double click on the icon to run and operate the simulated AoIP simulator.

4.7 AoIP BROADCAST SYSTEM COST EVALUATION AND ANALYSIS

The cost analysis given below was gotten during the feasibility studies and marker survey phases of the AoIP broadcast system design. The cost implication is given in Table 4.5.

Table 4.5: Cost Implication for Traditional and AoIP broadcast system device.

S/NO	**HARDWARE/SOFTWARE**	**QTY**	**UNIT PRICE**	**TOTAL**
1	Studio Desktop Computer(HP dx 2300 with monitor)	2	110,000.00	220,000.00
2	Editing Desktop Computer(HP dx 2300 with monitor)	4	110,000.00	440,000.00
3	12 Port Netgear Switch	1	8,000.00	8,000.00
4	Denon Advanced DVD player(DN-9000)	2	753,750.00	1,507,500.00
5	BBC Receiver	1	75,750.00	75,750.00
6	Technica ATHM 50 Headphone	4	13,498.00	53,994.00
7	Shure Microphone	8	55,000.00	440,000.00
8	Behringer Speaker	4	40,041.00	160,164.00
9	Digigram	2	50,000.00	100,000.00
10	M-Audio	5	22,500.00	112,500.00
11	GPC-3 Desktop Turret List	2	35,700.00	71,400.00
12	Miscellenous		200,000.00	200,000.00
12	Total			3,389,308.00

NB: The above cost for device totaling three million, three hundred and eighty nine thousand, three hundred and eight naira (₦3,389,308.00) or ($22,595,387.00) is the same for both traditional and AoIP networks

Table 4.6: Wiring Materials cost comparison for a Typical Studio Facility (US$)

Materials	**Traditional Wiring**	**AoIP Network**
CAT-6 cable or fiber	0	$ 600.00
Multi-pair audio cable	$ 2,800.00	0
Punch Blocks and wiring guides	$ 1,600.00	0
Distribution Amps	$ 2,400.00	0
Central Audio Router or Ethernet switches	$ 60,000.00	$ 18,000.00
Satellite Router Cage or Audio Nodes	0	$ 32,300.00
Audio Console/Control Surface (Wheatnet & its accessories for AoIP)	$ 76,000.00	$ 68,000.00
COMTEC Studio Transmitter Link (STL)	$10,000	$12,000.00
Automation Server	-	$3,000.00
Audio cables & connectors for studio and Control Center equipment	$ 900.00	$ 1,200.00
Total cost for consoles, routing & wiring	**$153,700.00**	**$133,900.00**

Table 4.7: Audio Wiring Labour for a typical Studio Facility and Control Centre

Task	Traditional Wiring	AoIP Network
Studio: Source/Destination equipment to punch blocks or nodes	96	32
Studio: Console to punch blocks	32	0
Studio & Control Center: Multi-pair cable runs and terminations	48	0
Studio & Control Center: CAT-6 cable terminations	0	16
Control Center: Central Audio Router to punch blocks	32	0
Control Center: Source/Destination equipment to punch blocks or nodes	24	4
Control Center: Distribution amps to/from punch blocks	8	0
Programming of audio router or nodes and consoles	4	16
Total labor hours	244	68
Installation labor expense	$12,200.00	$3,400.00
Total equipment and labor	**$155,900.00**	**$123,500.00**

From the above, the total cost implication of $22,595,387.00 for AoIP system as against $35,552,208.00 for traditional studio offers tremendous cost advantage. Also AoIP system reduces labor hours input as well as cost of labor.

4.8 ADVANTAGES OF AoIP BROADCAST SYSTEM

The following are the advantages of this system:

Transport: Computer cable can transport the audio signal faster. Fibre optics and copper wire can also be used for more speed ad efficiency.

Distribution: The receiving devices just “listen” to the feed from the source devices in multicast mode.

Routing: Multicast data packets are seamlessly routed across the network.

Flexibility: Compared to a traditional fixed studio installation, a modular IP audio network can be configured, modified and upgraded much more easily.

Scalability: Perhaps the most fundamental change between IP based audio systems and traditional approaches – analog or digital – is the ability of IP architectures to adapt to change and growth. An IP Audio can easily adapt to any format- mono, stereo, surround. This system implements packet switching and serial design. This allows great flexibility and responsiveness in accommodating changes in I/O configuration.

Cost effectiveness: At almost any reasonable size, an IP based audio system will compare favorably with the cost of a traditional system – both in terms of hardware/materials pricing and installation cost. The reduction in wire alone provides substantial economy. Maintenance and operational expenses are generally also lower. These cost differentials increase with the size of the facility, which is why so many larger installations have already moved to IP based solutions as their needs have called for new technical plants.

Convenience: The small physical footprint, low operating cost, ease of reconfiguration or upgrade, and fast installation of IP Audio systems make them extremely convenient for engineering and operations alike at the audio studio facility. From initial design to implementation to daily operation, IP based systems make life easier.

Future proofing: Nothing strikes fear in the heart of the engineer or manager more than making a poor major purchasing decision. Moving to an IP based audio architecture takes a lot of the pressure off, since it offers such flexibility and allows broad ability for reconfiguration down the road. Provisioning for unforeseen changes is much less problematic and cheaper.

This is not to say that there aren't some challenges to the optimal use of IP for studio audio transport. Primary among these is the latency that the encapsulation process of audio data into IP packets can cause, and their serial routing through a packet switched network can be prone to data collisions. As mentioned earlier, various methods have been put forth to ameliorate this, but proper configuration of available devices is usually adequate to resolve any such difficulties. Such configuration of standard IP equipment (e.g., Ethernet routers and switches, buffer sizes and network speeds) can be set to optimally serve the specific needs of a studio audio system. For this reason – as well as for obvious security purposes – it is important to use a separate, dedicated network for the design. This network can carry all audio content, control signals and metadata related to production, but should be isolated from the general data network of the facility. In addition, there generally should not be a direct connection of the studio audio system to the public Internet. When Internet connections are required for access to offsite audio sources or destinations, they should be routed through a proxy server or other isolating path. Setting IP packet prioritization at a higher level for audio content packets than for general network data yields more efficiency. Of course, the need to retain compatibility with analog and AES3 digital audio will likely remain for some time to come at any IP based audio facility. At the very least, live microphone signals will need to be converted from their native analog audio this can be accomplished through careful selection of selection of a single vendor for the supply of the system equipment or at least the verification that compatibility among different vendors' IP Audio equipment is assured. Working with a single vendor also ensures that updates and upgrades will be delivered in a timely fashion.

AoIP broadcast system provides a standardized platform to overcome most challenges encountered in traditional method. Another development that is closely related to IP Audio is Voice over IP (VoIP), which has rapidly gained ground in the telephony space as a replacement for traditional voice service, in both consumer and enterprise applications. Also gaining ground is the IPTV a technology similar to IP Audio and widely deployed in the TV industry.

CHAPTER FIVE

5.0 CONCLUSION AND RECOMMENDATION

This section is on the conclusion and recommendations for design of AoIP broadcast system.

5.1 CONCLUSION

The proposed AoIP broadcast system was designed and simulated. The simulated Production and ON-air studio was demonstrated, tested and runned as a stand alone. Hence the Audio Over Internet Protocol (AoIP) Broadcast Studio addresses and eliminates the hitches related to time loss, signal quality and cost by ensuring reduced cost, good and quality signal transmission. As such it has met the objective of this thesis. It is more computer friendly, flexible and scalable. The entire system finds useful application in radio stations, music studios, mass communication department studios in higher institutions etc. With this solution, it is quite obvious that the broadcast industry is ready for digitization. It will also give rise to quantitative growth in services, efficiency and improved productivity. More streams, more audio channels, more data, more responsiveness to audience demands, and more still, are all on the path that lies ahead for broadcasters. Other valuable advantages of this solution are:

5.2 CONTRIBUTION TO KNOWLEDGE

AoIP Broadcast System comes with notable improvement on the sustained economic wealth and national development of her citizenry in the following areas:

- Digital Broadcasting

- Building and maintaining National or State Broadcasting
  database and e-archive.
- National Voice
- Technological Development and advancement
- Information Communication technology
- Educational learning system

5.3 PROBLEMS ENCOUNTERED

No good work comes without its challenges and pitfalls. Challenges of this system design manifested in diverse forms. The simulator had compatibility issues with older edition of Windows vista and 7 Microsoft operating system especially with its library link files. For optimal performance, it is highly recommended to run on Windows XP professional edition. Books and literature were not locally available for reference as the concept of AoIP is a fairly new technology. Due to the huge capital involvement, the work cannot be demonstrated beyond the simulation level.

5.4 FUTURE WORK

The following are recommendation for future work. The work can be expanded to eliminate and reduce the above challenges and to integrate with multiple subnet router for remote site access, IP telephony system, HD Radio system (multichannel), Podcasting etc would be a plus. Development of an effective system to combat latency issue and accommodate the above expansion is a subject for further study. Development and installation of aggressive digital broadcast infrastructure in the nation broadcast industry is a must for this solution to be utilized.

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APPENDIX

```
frmmic
Private Sub Form_Load()
frmmic.Left = FormPositionLeftMic
frmmic.Top = FormPositionTopMic
Call SetWindowPos(hwnd, HWND_TOPMOST, 0, 0, 0, 0, SWP_NOMOVE Or SWP_NOSIZE)
Dim i As Integer
i = Rnd * 4
CommonDialog1.InitDir = App.Path & "\micsound"
CommonDialog1.ShowOpen
WindowsMediaPlayer1.URL = CommonDialog1.Filename
End Sub
Private Sub Form_Unload(Cancel As Integer)
timerStateForMic = "stop"
SpeakerState2 = "stop"
End Sub
Private Sub WindowsMediaPlayer1_StatusChange()
SpeakerState2 = ""
timerStateForMic = ""
 If WindowsMediaPlayer1.playState = wmppsStopped Then
   SpeakerState2 = "stop"
   timerStateForMic = "stop"
 ElseIf WindowsMediaPlayer1.playState = wmppsPaused Then
   timerStateForMic = "stop"
   SpeakerState2 = "stop"
End If
```

```
End Sub

Cdplayer.frm

Private Sub Form_Load()

frmcd.Left = FormPositionLeftCD

frmcd.Top = FormPositionTopCD

Call SetWindowPos(hWnd, HWND_TOPMOST, 0, 0, 0, 0, SWP_NOMOVE Or SWP_NOSIZE)

On Error GoTo Error_Handler

 WindowsMediaPlayer1.URL = deckfile

    WindowsMediaPlayer1.Controls.play

     Error_Handler:

End Sub

Private Sub Form_Unload(Cancel As Integer)

timerstateForCD = "stop"

SpeakerState1 = "stop"

End Sub

Private Sub WindowsMediaPlayer1_StatusChange()

timerstateForCD = ""

SpeakerState1 = ""

 If WindowsMediaPlayer1.playState = wmppsStopped Then

    timerstateForCD = "stop"

    SpeakerState1 = "stop"

 ElseIf WindowsMediaPlayer1.playState = wmppsPaused Then

    timerstateForCD = "stop"

    SpeakerState1 = "stop"

End If

End Sub
```

Loading.frm

```
Private Sub Timer1_Timer()
Timer1.Enabled = True
shl1.BackColor = &HFF&
Timer2.Enabled = True
End Sub
Private Sub Timer2_Timer()
Timer2.Enabled = True
shl2.BackColor = &HFF&
Timer3.Enabled = True
End Sub
Private Sub Timer3_Timer()
Timer3.Enabled = True
shl3.BackColor = &HFF&
Timer4.Enabled = True
End Sub
Private Sub Timer4_Timer()
Timer4.Enabled = True
shl4.BackColor = &HFF&
Timer5.Enabled = True
End Sub
Private Sub Timer5_Timer()
Timer5.Enabled = True
shl5.BackColor = &HFF&
Timer6.Enabled = True
```

```
End Sub

Private Sub Timer6_Timer()

Timer6.Enabled = True

shl6.BackColor = &HFF&

Timer7.Enabled = True

End Sub

Private Sub Timer7_Timer()

Timer7.Enabled = True

shl7.BackColor = &HFF&

Timer8.Enabled = True

End Sub

Private Sub Timer8_Timer()

Timer8.Enabled = True

shl8.BackColor = &HFF&

Timer9.Enabled = True

End Sub

Private Sub Timer9_Timer()

Timer9.Enabled = True

shl9.BackColor = &HFF&

Timer10.Enabled = True

End Sub

Private Sub Timer10_Timer()

Timer10.Enabled = True

shl10.BackColor = &HFF&

Timer11.Enabled = True

End Sub
```

```
Private Sub Timer11_Timer()

Timer11.Enabled = True

shl11.BackColor = &HFF&

Timer12.Enabled = True

End Sub

Private Sub Timer12_Timer()

Timer12.Enabled = True

shl12.BackColor = &HFF&

Timer13.Enabled = True

End Sub

Private Sub Timer13_Timer()

Timer13.Enabled = True

shl13.BackColor = &HFF&

Unload Me

frmonair.Show

End Sub
```

Production.sub

```
Private Sub cd_Click()

frmcd.Show

End Sub

Private Sub deck_Click()

frmdeck.Show

End Sub

Private Sub Image1_DblClick()

frmsample.Show
```

```
End Sub

Private Sub Image1_MouseUp(Button As Integer, Shift As Integer, X As Single, Y As Single)

If Button = vbRightButton Then PopupMenu computer

End Sub

Private Sub Image2_MouseUp(Button As Integer, Shift As Integer, X As Single, Y As Single)

If Button = vbRightButton Then PopupMenu console

End Sub

Private Sub Image4_MouseUp(Button As Integer, Shift As Integer, X As Single, Y As Single)

If Button = vbRightButton Then PopupMenu server

End Sub

Private Sub MMControl1_PlayClick(Cancel As Integer)

MMControl1.Filename = App.Path & "\Windows XP Startup.wav"

MsgBox "please continue", vbInformation, "that is what it does"

MMControl1.Enabled = True

End Sub

Private Sub Image3_DblClick()

frmsample.Show

End Sub

Private Sub Image5_DblClick()

On Error GoTo Error_Handler

   CommonDialog1.CancelError = True

   Unload frmdeck

   CommonDialog1.Filter = "All Supported Media Files|*.*"

   CommonDialog1.Flags = &H2 Or &H400

   CommonDialog1.ShowOpen

   deckfile = CommonDialog1.Filename
```

```
    FormPositionLeftDeck = 13000
    FormPositionTopDeck = 4200
    frmdeck.Show
    Timer5.Enabled = True
Error_Handler:
End Sub
Private Sub Image6_DblClick()
On Error GoTo Error_Handler
    Unload frmcd
    CommonDialog1.CancelError = True
    CommonDialog1.Filter = "All Supported Media Files|*.*"
    CommonDialog1.Flags = &H2 Or &H400
    CommonDialog1.ShowOpen
    deckfile = CommonDialog1.Filename
    FormPositionLeftCD = 13000
    FormPositionTopCD = 2200
    frmcd.Show
    Timer1.Enabled = True
Error_Handler:
End Sub
Private Sub Image7_DblClick()
Unload frmmic
FormPositionLeftMic = 13000
FormPositionTopMic = 6000
```

```
frmmic.Show
Timer2.Enabled = True
End Sub
Private Sub mic_Click()
Unload frmmic
frmmic.Show
End Sub
Private Sub onair_Click()
Form2.Show
Unload Me
Unload frmcd
Unload frmdeck
frmsample.Hide
Unload frmmic
End Sub
Private Sub mnuexit_Click()
End
End Sub
Private Sub mnuonair_Click()
Unload Me
frmonair.Show
End Sub
Private Sub Timer1_Timer()
If SpeakerState1 = "" Then
   Timer4.Enabled = True
End If
```

```
End Sub
Private Sub Timer10_Timer()
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape23.Visible = True
Shape3.Visible = True
Shape6.Visible = True
Shape1.Visible = False
Shape4.Visible = False
Shape7.Visible = False
Timer11.Enabled = True
Timer10.Enabled = False
End Sub
Private Sub Timer11_Timer()
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape1.Visible = False
Shape23.Visible = False
Shape6.Visible = False
Shape7.Visible = False
Shape4.Visible = False
Shape3.Visible = False
Timer9.Enabled = True
Timer11.Enabled = False
```

```
End Sub
Private Sub Timer2_Timer()
If SpeakerState2 = "" Then
   Timer4.Enabled = True
End If
End Sub
Private Sub Timer3_Timer()
If SpeakerState3 = "" Then
   Timer4.Enabled = True
End If
End Sub
Private Sub Timer4_Timer()
If SpeakerState1 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState2 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState3 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape1.Visible = True
Shape4.Visible = True
```

```
Shape7.Visible = True
Shape23.Visible = False
Shape6.Visible = False
Shape3.Visible = False
Timer6.Enabled = True
Timer4.Enabled = False
End Sub
Private Sub Timer5_Timer()
If SpeakerState4 = "" Then
   Timer9.Enabled = True
End If
End Sub
Private Sub Timer6_Timer()
If SpeakerState1 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState2 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState3 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape23.Visible = True
```

```
Shape3.Visible = True
Shape6.Visible = True
Shape1.Visible = False
Shape4.Visible = False
Shape7.Visible = False
Timer8.Enabled = True
Timer6.Enabled = False
End Sub
Private Sub Timer7_Timer()
Shape1.Visible = False
Shape23.Visible = False
Shape6.Visible = False
Shape7.Visible = False
Shape4.Visible = False
Shape3.Visible = False
Timer4.Enabled = False
Timer9.Enabled = False
Timer6.Enabled = False
Timer8.Enabled = False
Timer7.Enabled = False
End Sub
Private Sub Timer8_Timer()
If SpeakerState1 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState2 = "stop" Then
```

```
   Timer7.Enabled = True
End If
If SpeakerState3 = "stop" Then
   Timer7.Enabled = True
End If
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape1.Visible = False
Shape23.Visible = False
Shape6.Visible = False
Shape7.Visible = False
Shape4.Visible = False
Shape3.Visible = False
Timer4.Enabled = True
Timer8.Enabled = False
End Sub
Private Sub Timer9_Timer()
If SpeakerState4 = "stop" Then
   Timer7.Enabled = True
End If
Shape1.Visible = True
Shape4.Visible = True
Shape7.Visible = True
Shape23.Visible = False
Shape6.Visible = False
```

```
Shape3.Visible = False

Timer10.Enabled = True

Timer9.Enabled = False

End Sub
```