HEAVY HYPERSONIC DUAL ACOUSTIC SYSTEM

A Project report submitted in partial fulfillment of the requirements for the award of the degree of

BACHELOR OF TECHNOLOGY

IN

ELECTRONICS AND COMMUNICATION ENGINEERING

Submitted by

M. John Kumar (317126512145)
P. Praveen (317126512162)
N. Dheeraj (318126512L30)
K.V.S. Harish Kumar (317126512140)

Under the guidance of

Mr. N. Ram Kumar, M.Tech, (Ph.D)

Assistant Professor, Department of ECE

PRAGNANAM BRAHMA

ANITS

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

ANIL NEERUKONDA INSTITUTE OF TECHNOLOGY AND SCIENCES

(UGC AUTONOMOUS)

(*Permanently Affiliated to AU, Approved by AICTE and Accredited by NBA & NAAC with 'A' Grade*)

Sangivalasa, Bheemili mandal, Visakhapatnam dist.(A.P) 2020-2021

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CERTIFICATE

This is to certify that the project report entitled **"HEAVY *HYPERSONIC DUAL ACOUSTIC SYSTEM*"** submitted by **M *John Kumar (317126512145), P Praveen (31712612162), N Dheeeraj (318126512L30), K V S Harish Kumar (317126512140)*** in partial fulfillment of the requirements for the award of the degree of ***Bachelor of Engineering** in **Electronics & Communication Engineering*** of Andhra University, Visakhapatnam is a record of bonafide work carried out under my guidance and supervision.

Project Guide

Mr. Ram Kumar . N

M.Tech,(Ph.D)

Assistant Professor

Department of ECE, ANITS

Assistant Professor
Department of E.C.E.
Anil Neerukonda
Institute of Technology & Sciences
Sangivalasa

Head of the Department

Dr. V Rajyalakshmi

M.E, Ph.D., MHRM, MIEEE, MIE, MIETE

Department of E.C.E, ANITS

Head of the Department
Department of E C E
Anil Neerukonda Institute of Technology & Sciences
Sangivalasa - 531 162

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PROJECT STUDENTS:

M.John Kumar (317126512145)
P.Praveen (317126512162)
N.Dheeraj (318126512L30)
K.V.S.Harish Kumar (317126512140)

ABSTRACT

A new Acoustic setup is being introduced to bound the sound waves produced by the heavy sound systems make directional. The sound waves are created using a directional sound system having sharp directivity and gain. We don't need any demodulator; due to the non-linearity of air, it acts as a demodulator. The main disadvantage of Ultrasound technology is even it means a small piece of paper blocks the sound waves. We are using this disadvantage to reduce the problem of heavy sound systems used in function halls, occasional parties, pubs etc..., Acoustic foam can be replaced with papers and make a fence around the area that confine the sound. The directive speaker array fires the sound directly either to the people or to the acoustic foam. After multiple reflections, sound-absorbing foam absorbs the maximum amount of sound and avoids reflections beyond the desired region. Heavy sounds will bound to a particular space and keep surroundings pollution-free; after installed the proposed device.

Keywords: ***Ultrasound, directive sound system, sound-absorbing foams, Heavy sounds, parametric grid.***

Team Members:

M. John Kumar (317126512145)
P. Praveen (317126512162)
N. Dheeraj (318126512L30)
K.V.S. Harish Kumar (317126512140)

Project Guide:

Mr. N. Ram Kumar, Asst. Prof., ECE

CONTENTS

List of Figures

List of Tables

CHAPTER 1

INTRODUCTION

1.1 Project Objective:

All over the World, people have been suffering from sound pollution, in order to reduce this problem, a device can be implement such that it will be helpful in many public areas like Function Halls, Colleges, Schools, heavy traffic places, Hospitals, Museums, Supermarkets, Crowded regions etc..., One of the solutions is to force the sound wave travel in particular area i.e., increasing the directivity to sound waves by decreasing its spreading region. In order to obtain this characteristic for sound waves, employing a new technology called “Holosonic”. By using this technology, a Directional Ultrasonic Speaker can be created which will integrated in modern Sound systems, Tv sets, Home theatres, Horns etc...,.

This Directional Sound System can be adjustable i.e., we can able to switch from normal mode to directional mode. Series of ultrasonic speakers are arranged in such a way that it converges all the sound signals given to it and finally gives the directivity for audio signals.

Audio signal are modulating with high frequency signals and transmitted directly into the air through parametric array of ultrasound speakers, the demodulation will be done air medium itself of its acoustic nature.

1.2 Motivation:

The Directional Sound system is an innovative sound platform started from the concept of making sound seamlessly tracking a designated user by harnessing the directivity of a mobilized ultrasonic speaker. The motivation of this system development is to enable a totally personalized ubiquitous sound delivery within a typical 3-dimensional space without using headphone or other external sound blocking device. Experienced example is the issue we faced during the college fest because of traditional sound system.

CHAPTER 2

NOISE POLLUTION

2.1 NOISE POLLUTION

1. Noise pollution, also known as environmental noise or sound pollution, is the propagation of noise with harmful impact on the activity of human or animal life.
2. The source of outdoor noise worldwide is mainly caused by machines, transport and transportation systems.
3. Poor urban planning may give rise to noise pollution, side-by-side industrial and residential buildings can result in noise pollution in the residential areas
4. High noise levels can contribute to cardiovascular effects in humans and an increased incidence of coronary artery disease.
5. In animals, noise can increase the risk of death by altering predator or prey detection and avoidance, interfere with reproduction and navigation, and contribute to permanent hearing loss.
6. Noise pollution, also known as environmental noise or sound pollution, is the propagation of noise with ranging impacts on the activity of human or animal life, most of them harmful to a degree.
7. The source of outdoor noise worldwide is mainly caused by machines, transport, and propagation systems

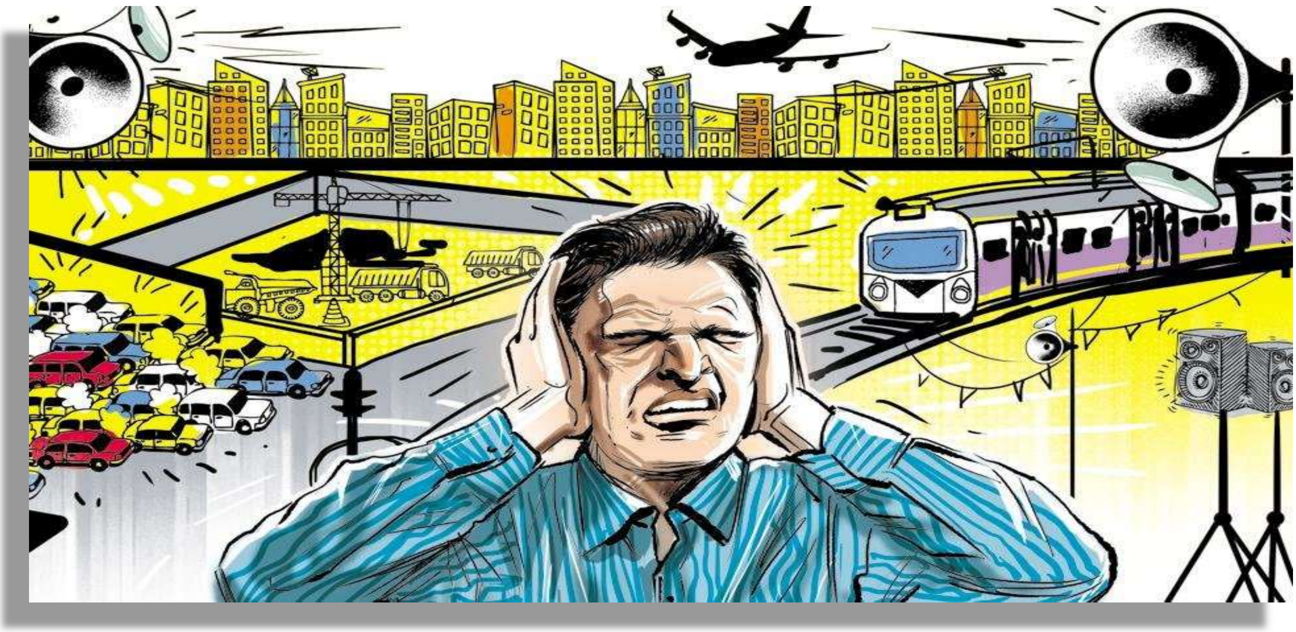


Fig 2.1

2.2 EFFECTS OF NOISE POLLUTION

1. Noise pollution can cause health problems for people and wildlife, both on land and in the sea. From traffic noise to rock concerts, loud or inescapable sounds can cause hearing loss, stress, and high blood pressure.

2. Noise from ships and human activities in the ocean is harmful to whales and dolphins that depend on echolocation to survive.

Fig 2.2

Fig 2.3

2.2.1 EFFECTS OF NOISE POLLUTION ON ANIMALS, CHILDREN AND SENIOR CITIZENS

Fig 2.4

Fig 2.5

Fig 2.6

1. While the elderly may have cardiac problems due to noise, according to the World Health Organization, children are especially vulnerable to noise, and the effects that noise has on children may be permanent.
2. Noise poses a serious threat to a child’s physical and psychological health, and may negatively interfere with a child's learning and behaviour.

2.2.2 EFFECTS OF NOISE POLLUTION IN FESTIVALS

1. The noise in Navratri makes our windows shake' Sight of activists recording noise pollution levels during events is quite normal. But this Navratri, many Citizens have taken to monitoring decibel levels via mobile phone applications

Fig 2.7

2. Citizens across the city recorded noise levels between 82 and 100 decibels during Navratri festival, much higher than the permissible limit of 55 db.
3. Festivals like Ganeshotsav and Navratri are getting noisier by the year. The sounds of dhol-tasha have been combined with recorded music blaring from loudspeakers in public places.
4. In Navratri, live music too reaches the decibel levels of recorded music via loudspeakers, with thousands dancing to the tunes at Garba celebrations.

Noise monitoring on Diwali

YEAR	Residential Zone	Commercial Zone	Silence Zone
2016	68.27 dB	74.96 dB	72.28 dB
2014	51.6 dB	61.6 dB	46.6 dB
2013	75.7 dB	79.1 dB	77.7 dB
2012	103.3 dB	103.0 dB	100.5 dB
2011	84.6 dB	84.3 dB	70.1 dB
2010	105.2 dB	104.5 dB	94.8 dB

Table 2.1

5. After the celebration of durgapuja, dusserah, it comes Diwali with it's new style and custom. The problem is that the new style is full of crackers, not the lights which causes air and sound pollution. Modern music system gives this problem an extra edge.
6. Any sound above 85 DB has the potential to harm our ears. Do you know, the sound level during diwali is in between 140-175 DB, a range that is high enough to cause permanent hearing damage.

CITY	HIGHEST NOISE LEVEL RECORDED dB(A)
DELHI	83
HYDERABAD	76
KOLKATA	75
CHENNAI	69
LUCKNOW	68
MUMBAI	66
BANGALORE	62

Table 2.2

2.2.3 HEALTH HAZARDS

1. Noise pollution affects both health and behaviour.

Area code	Category of area/zone	Limits in dB(A) leq*	
		Day time	Night time
(A)	Industrial area	75	70
(B)	Commercial area	65	55
(C)	Residential area	55	45
(D)	Silence zones	50	40

*dB(A) Leq denotes the time weighted average of the level of sound in decibels on scale A which is relatable to human hearing. Source: Central Pollution Control Board, India

Table 2.3

2. Unwanted sound (noise) can damage physiological health. Noise pollution can cause hypertension, high stress levels, tinnitus, hearing loss, sleep disturbances, and other harmful effects.
3. Sound becomes unwanted when it either interferes with normal activities such as sleep or conversation, or disrupts or diminishes one's quality of life.

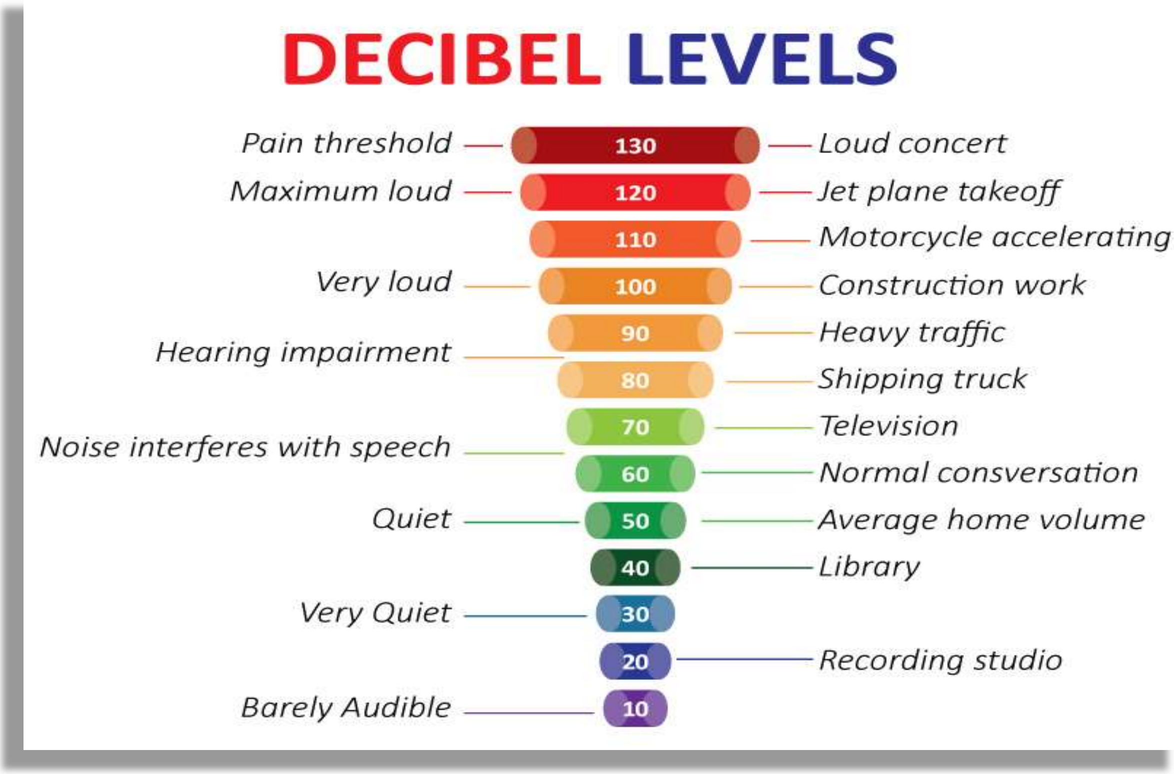


Table 2.4

4. Noise-induced hearing loss can be caused by prolonged exposure to noise levels above 85 A-weighted, decibels

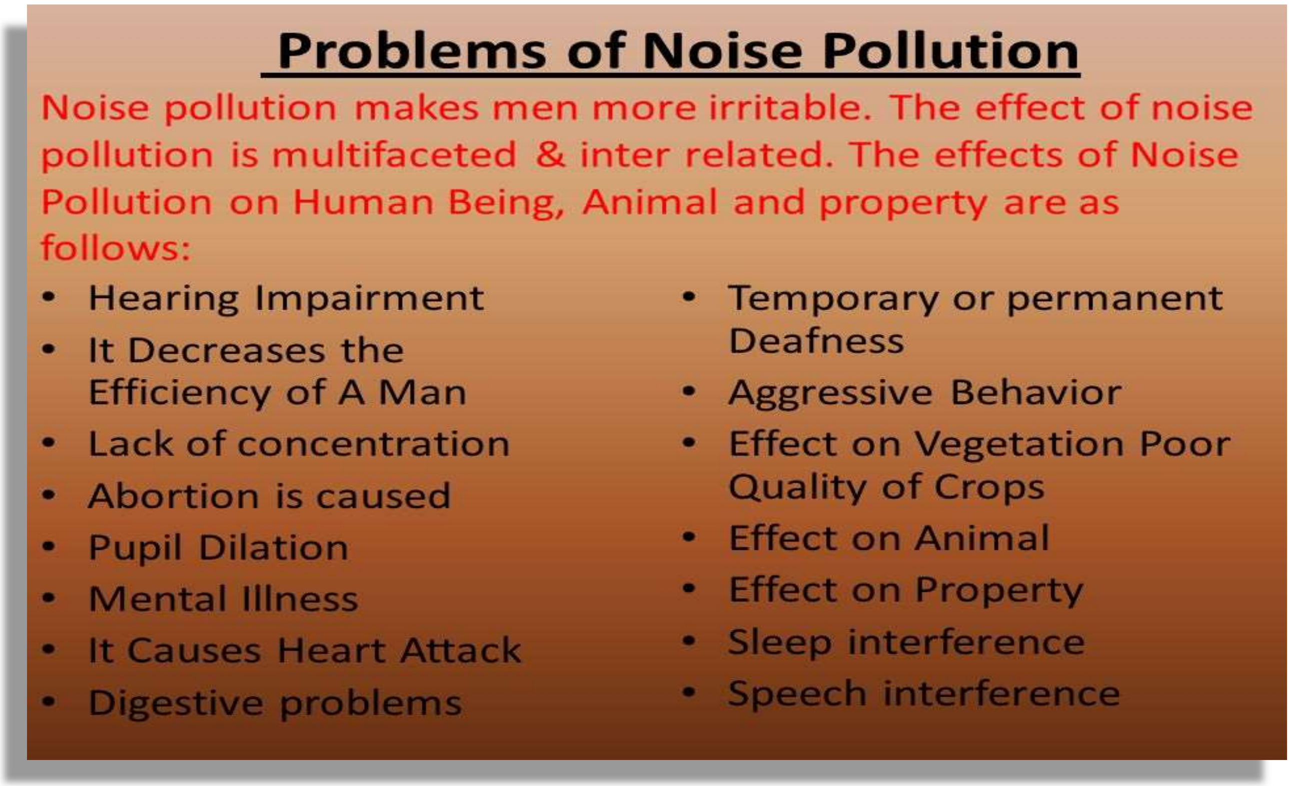


Fig 2.8

2.3 SOLUTION

1. The inherent directivity (narrowness) of all wave producing sources depends on little more than the size of the source, compared to the wavelengths it generates. Audible sound has wavelengths ranging from a few inches to several feet, and because these wavelengths are comparable to the size of most loudspeakers, sound generally propagates omnidirectionally.

Fig 2.9

2. Only by creating a sound source much larger than the wavelengths it produces can a narrow beam be created. In the past, loudspeaker manufacturers have created large speaker panels or used reflective domes to provide some directivity but, due to the sound's large wavelengths, the directivity of these devices is still extremely weak.

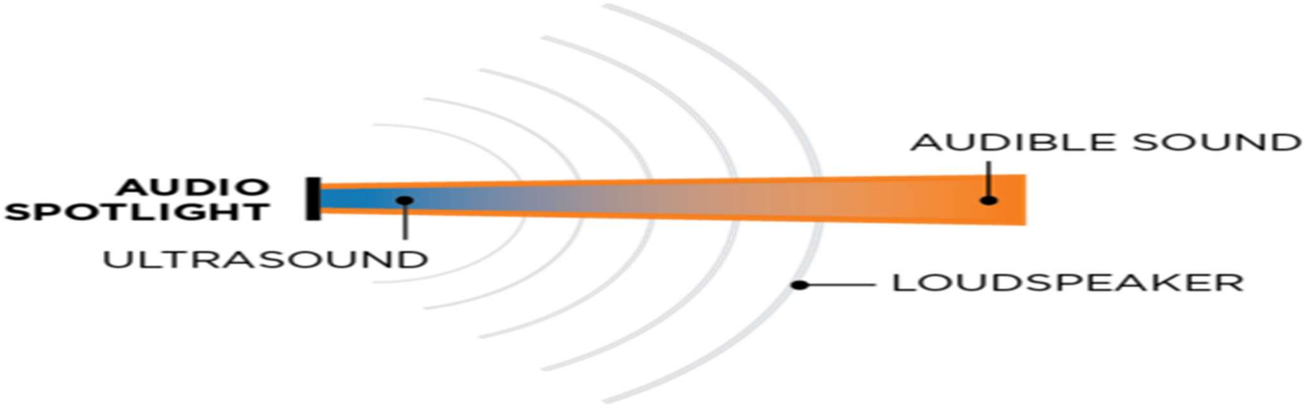


Fig 2.10

3. No loudspeaker can ever approach the directivity of Audio Spotlight technology.
4. Ultrasound as a sound source!
5. Since the goal is a small loudspeaker but strong directivity, the only possible solution is to generate very small wavelengths - such as those of high-frequency ultrasound. The ultrasound used in Holosonic technology has wavelengths only a few millimeters long, which are much smaller than the source, and therefore naturally travel in an extremely narrow beam.
6. Of course, the ultrasound, which contains frequencies far outside our range of hearing, is completely inaudible. But as the ultrasonic beam travels through the air, the inherent properties of the air cause the ultrasound to change shape in a predictable way. This gives rise to frequency components in the audible band, which can be accurately predicted, and therefore precisely controlled. By generating the correct ultrasonic signal, we can create, within the air itself, any sound desired.
7. Sound is literally made from thin air.

8. Note that the source of sound is not the physical device you see, but the invisible beam of ultrasound, which can be many meters long. This new sound source, while invisible, is very large compared to the audio wavelengths it's generating. So, the resulting audio is now extremely directional, just like a beam of light.
9. Often incorrectly attributed to so-called "Tartini tones", the technique of using high-frequency waves to generate low-frequency signals was pioneered over forty years ago. Over the past several decades, many others have attempted – and failed – to use this technique to make a practical audio source.
10. Through a combination of careful mathematical analysis and engineering insight based on pioneering work at MIT in the early 2000's, the patented Audio Spotlight sound system has become the very first, and still the only, truly directional audio system which generates high quality sound in a reliable, professional package.

CHAPTER 3

DIRECTIONAL SPEAKER

3.1 INTRODUCTION TO DIRECTIONAL SPEAKER

1. The piezoelectric transducers (Gray circles) in the directional speaker produce two ultrasonic waves (red and blue), both of which are at frequencies way too high to hear). The transducers pump out the waves in a focused column (like the light in a flashlight beam). The waves are actually modulated (like radio waves) and travel as one wave, but it's simplest to imagine them as two quite separate waves.
2. When the two waves hit something (or someone), they slow down and demodulate, producing a new wave (green) whose frequency is much lower—equal to the difference in frequencies between the two original waves. This is a wave you can hear.
3. When there's no-one standing in the beam, the waves keep on traveling without producing an audible sound wave—so if there's no-one standing in front of the speaker, there's nothing you can hear.
4. People standing outside the beam can't hear anything because (unlike with a conventional loudspeaker) the sound waves are not diverging from the source of sound to reach their ears.

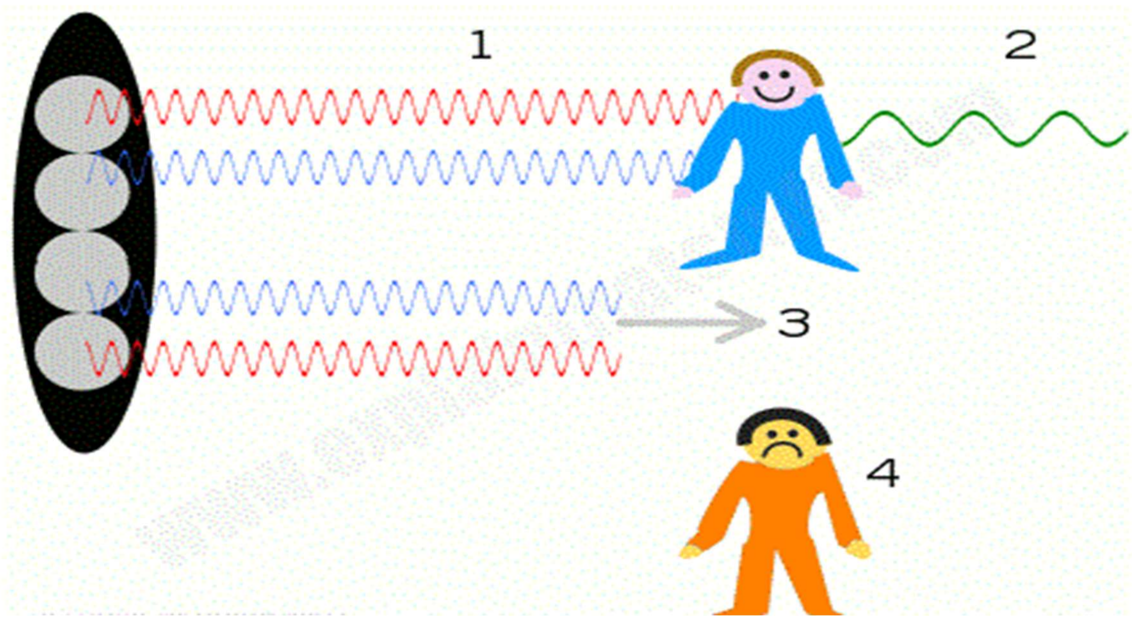


Fig 3.1

3.2 ACOUSTIC SOUND

1. Acoustics is a branch of physics that deals with the study of mechanical waves in gases, liquids, and solids including topics such as vibration, sound, ultrasound and infrasound.
2. A scientist who works in the field of acoustics is an acoustician while someone working in the field of acoustics technology may be called an acoustical engineer.
3. The application of acoustics is present in almost all aspects of modern society with the most obvious being the audio and noise control industries.
4. Hearing is one of the most crucial means of survival in the animal world and speech is one of the most distinctive characteristics of human development and culture.
5. Accordingly, the science of acoustics spreads across many facets of human society—music, medicine, architecture, industrial production, warfare and more. Likewise, animal species such as songbirds and frogs use sound and hearing as a key element of mating rituals or marking territories.
6. Art, craft, science and technology have provoked one another to advance the whole, as in many other fields of knowledge. Robert Bruce Lindsay's "Wheel of Acoustics" is a well accepted overview of the various fields in acoustics.
7. Physicists and acoustic engineers tend to discuss sound pressure levels in terms of frequencies, partly because this is how our ears interpret sound. What we experience as "higher pitched" or "lower pitched" sounds are pressure vibrations having a higher or lower number of cycles per second.
8. In a common technique of acoustic measurement, acoustic signals are sampled in time, and then presented in more meaningful forms such as octave bands or time frequency plots. Both of these popular methods are used to analyze sound and better understand the acoustic phenomenon.
9. The entire spectrum can be divided into three sections: audio, ultrasonic, and infrasonic. The audio range falls between 20 Hz and 20,000 Hz. This range is important because its frequencies can be detected by the human ear. This range has a number of applications, including speech communication and music. The ultrasonic range refers to the very high frequencies: 20,000 Hz and higher.

10. This range has shorter wavelengths which allow better resolution in imaging technologies. Medical applications such as ultrasonography and elastography rely on the ultrasonic frequency range. On the other end of the spectrum, the lowest frequencies are known as the infrasonic range. These frequencies can be used to study geological phenomena such as earthquakes.

11. Analytic instruments such as the spectrum analyzer facilitate visualization and measurement of acoustic signals and their properties.

12. The spectrogram produced by such an instrument is a graphical display of the time varying pressure level and frequency profiles which give a specific acoustic signal its defining character.

13. A transducer is a device for converting one form of energy into another. In an electroacoustic context, this means converting sound energy into electrical energy (or vice versa). Electroacoustic transducers include loudspeakers, microphones, particle velocity sensors, hydrophones and sonar projectors. These devices convert a sound wave to or from an electric signal. The most widely used transduction principles are electromagnetism, electrostatics and piezoelectricity.

14. The transducers in most common loudspeakers (e.g. woofers and tweeters), are electromagnetic devices that generate waves using a suspended diaphragm driven by an electromagnetic voice coil, sending off pressure waves. Electret microphones and condenser microphones employ electrostatics—as the sound wave strikes the microphone's diaphragm, it moves and induces a voltage change.

15. The ultrasonic systems used in medical ultrasonography employ piezoelectric transducers. These are made from special ceramics in which mechanical vibrations and electrical fields are interlinked through a property of the material itself.

3.2.1 TECHNICAL TERMS

Archaeoacoustics:

- Archaeoacoustics, also known as the archaeology of sound, is one of the only ways to experience the past with senses other than our eyes. Archaeoacoustics is studied by testing the acoustic properties of prehistoric sites, including caves. Iegor Rezkinoff, a sound archaeologist, studies the acoustic properties of caves through natural sounds like humming and whistling.

- Archaeological theories of acoustics are focused around ritualistic purposes as well as a way of echolocation in the caves. In archaeology, acoustic sounds and rituals directly correlate as specific sounds were meant to bring ritual participants closer to a spiritual awakening.

- Parallels can also be drawn between cave wall paintings and the acoustic properties of the cave; they are both dynamic. Because archaeoacoustics is a fairly new archaeological subject, acoustic sound is still being tested in these prehistoric sites today.

Aeroacoustics:

- Aeroacoustics is the study of noise generated by air movement, for instance via turbulence, and the movement of sound through the fluid air.

- This knowledge is applied in acoustical engineering to study how to quieten aircraft. Aeroacoustics is important for understanding how wind musical instruments work.

Acoustic signal processing:

- Acoustic signal processing is the electronic manipulation of acoustic signals.

- Applications include: active noise control; design for hearing aids or cochlear implants; echo cancellation; music information retrieval, and perceptual coding (e.g. MP3 or Opus)

Echo suppression and cancellation:

- Echo suppression and echo cancellation are methods used in telephony to improve voice quality by preventing echo from being created or removing it after it is already present.

- In addition to improving subjective audio quality, echo suppression increases the capacity achieved through silence suppression by preventing echo from traveling across a telecommunications network.
- Echo suppressors were developed in the 1950s in response to the first use of satellites for telecommunications.

3.3 NORMAL SPEAKER VS DIRECTIONAL SPEAKER

A. Loud Speaker:

1. A loudspeaker is an electroacoustic transducer; A device which converts an electrical audio signal into a corresponding sound.
2. The most widely used type of speaker is the dynamic speaker. The sound source (e.g., a sound recording or a microphone) must be amplified or strengthened with an audio power amplifier before the signal is sent to the speaker.
3. The dynamic speaker was invented in 1925 by Edward W. Kellogg and Chester W. Rice issued as US Patent 1,707,570. Apr 2, 1929. The dynamic speaker operates on the same basic principle as a dynamic microphone, but in reverse, to produce sound from an electrical signal.
4. When an alternating current electrical audio signal is applied to its voice coil, a coil of wire suspended in a circular gap between the poles of a permanent magnet, the coil is forced to move rapidly back and forth due to Faraday's law of induction, which causes a diaphragm (usually conically shaped) attached to the coil to move back and forth, pushing on the air to create sound waves.
5. Besides this most common method, there are several alternative technologies that can be used to convert an electrical signal into sound.
6. Speakers are typically housed in a speaker enclosure or speaker cabinet which is often a rectangular box made of wood or sometimes plastic. The enclosure's materials and design play an important role in the quality of the sound.

7. The enclosure generally must be as stiff and non-resonant as practically possible. Where high fidelity reproduction of sound is required, multiple loudspeaker transducers are often mounted in the same enclosure, each reproducing a part of the audible frequency range (picture at right).

8. In this case, the individual speakers may be referred to as *drivers* and the entire unit is called a loudspeaker. Drivers made for reproducing high audio frequencies are called tweeters, those for middle frequencies are called mid-range drivers and those for low frequencies are called woofers. Extremely low frequencies (16Hz-~100Hz) may be reproduced by separate subwoofers.

9. Smaller loudspeakers are found in devices such as radios, televisions, portable audio players, computers, and electronic musical instruments.

10. Larger loudspeaker systems are used for music, sound reinforcement in theatres and concert halls, and in public address systems.

B. Diaphragm:

1. The diaphragm is usually manufactured with a cone- or dome-shaped profile. A variety of different materials may be used, but the most common are paper, plastic, and metal.

2. The ideal material would 1) be rigid, to prevent uncontrolled cone motions; 2) have low mass, to minimize starting force requirements and energy storage issues; 3) be well damped, to reduce vibrations continuing after the signal has stopped with little or no audible ringing due to its resonance frequency as determined by its usage. In practice, all three of these criteria cannot be met simultaneously using existing materials; thus, driver design involves trade-offs.

3. For example, paper is light and typically well damped, but is not stiff; metal may be stiff and light, but it usually has poor damping; plastic can be light, but typically, the stiffer it is made, the poorer the damping. As a result, many cones are made of some sort of composite material

4. . For example, a cone might be made of cellulose paper, into which some carbon fiber, Kevlar, glass, hemp or bamboo fibers have been added; or it might use a honeycomb sandwich construction; or a coating might be applied to it so as to provide additional stiffening or damping.

C. **Basket**

1. The chassis, frame, or basket, is designed to be rigid, preventing deformation that could change critical alignments with the magnet gap, perhaps allowing the voice coil to rub against the magnet around the gap. Chassis are typically cast from aluminium alloy, in heavier magnet-structure speakers; or stamped from thin sheet steel in lighter-structure drivers.

2. Other materials such as molted plastic and damped plastic compound baskets are becoming common, especially for inexpensive, low-mass drivers. Metallic chassis can play an important role in conducting heat away from the voice coil; heating during operation changes resistance, causes physical dimensional changes, and if extreme, broils the varnish on the voice coil; it may even demagnetize permanent magnets.

5. The suspension system keeps the coil centered in the gap and provides a restoring (centering) force that returns the cone to a neutral position after moving.

6. A typical suspension system consists of two parts: the *spider*, which connects the diaphragm or voice coil to the lower frame and provides the majority of the restoring force, and the *surround*, which helps center the coil/cone assembly and allows free pistonic motion aligned with the magnetic gap.

7. The spider is usually made of a corrugated fabric disk, impregnated with a stiffening resin. The name comes from the shape of early suspensions, which were two concentric rings of Bakelite material, joined by six or eight curved "legs."

8. Variations of this topology included the addition of a felt disc to provide a barrier to particles that might otherwise cause the voice coil to rub. The German firm Rulik still offers drivers with uncommon spiders made of wood.

D. Cone materials

1. The cone surround can be rubber or polyester foam, treated paper or a ring of corrugated, resin coated fabric; it is attached to both the outer cone circumference and to the upper frame. These diverse

surround materials, their shape and treatment can dramatically affect the acoustic output of a driver; each implementation has advantages and disadvantages.

2. Polyester foam, for example, is lightweight and economical, though usually leaks air to some degree, but is degraded by time, exposure to ozone, UV light, humidity and elevated temperatures, limiting useful life before failure. Treated paper surrounds will eventually fail.

9. The wire in a voice coil is usually made of copper, though aluminium—and, rarely, silver—may be used. The advantage of aluminium is its light weight, which reduces the moving mass compared to copper. This raises the resonant frequency of the speaker and increases its efficiency.

10. A disadvantage of aluminium is that it is not easily soldered, and so connections are instead often crimped together and sealed. These connections must be made well or they may fail in an intense environment of mechanical vibration. Voice-coil wire cross sections can be circular, rectangular, or hexagonal, giving varying amounts of wire volume coverage in the magnetic gap space.

11. The coil is oriented co-axially inside the gap; it moves back and forth within a small circular volume (a hole, slot, or groove) in the magnetic structure.

12. The gap establishes a concentrated magnetic field between the two poles of a permanent magnet; the outside ring of the gap is one pole, and the center post (called the pole piece) is the other. The pole piece and backplate are often made as a single piece, called the poleplate or yoke.

13. Modern driver magnets are almost always permanent and made of ferrite, alnico, or, more recently, rare earth such as neodymium and samarium cobalt. Electrodynamic drivers were often used in musical instrument amplifier/speaker cabinets well into the 1950s; there were economic savings in those using tube amplifiers as the field coil could, and usually did, do double duty as a power supply choke.

14. A trend in design — due to increases in transportation costs and a desire for smaller, lighter devices (as in many home theatre multi-speaker installations) — is the use of the last instead of heavier ferrite

types. Very few manufacturers still produce electrodynamic loudspeakers with electrically powered field coils, as was common in the earliest designs; one of the last is a French firm.

15. When high field-strength permanent magnets became available after WWII, alnico, an alloy of aluminium, nickel, and cobalt became popular, since it dispensed with the problems of field-coil drivers. Alnico was used almost exclusively until about 1980,[*citation needed*] despite the embarrassing problem of alnico magnets being partially degaussed (i.e., demagnetized) by accidental 'pops' or 'clicks' caused by loose connections, especially if used with a high-power amplifier.
16. The damage can be reversed by "recharging" the magnet, but this requires uncommon specialist equipment and knowledge.
17. After 1980, most (but not quite all) driver manufacturers switched from alnico to ferrite magnets, which are made from a mix of ceramic clay and fine particles of barium or strontium ferrite. Although the energy per kilogram of these ceramic magnets is lower than alnico, it is substantially less expensive, allowing designers to use larger yet more economical magnets to achieve a given performance.
18. The size and type of magnet and details of the magnetic circuit differ, depending on design goals. For instance, the shape of the pole piece affects the magnetic interaction between the voice coil and the magnetic field, and is sometimes used to modify a driver's behaviour.
19. A "shorting ring", or Faraday loop, may be included as a thin copper cap fitted over the pole tip or as a heavy ring situated within the magnet-pole cavity. The benefits of this complication is reduced impedance at high frequencies, providing extended treble output, reduced harmonic distortion, and a reduction in the inductance modulation that typically accompanies large voice coil excursions.
20. On the other hand, the copper cap requires a wider voice-coil gap, with increased magnetic reluctance; this reduces available flux, requiring a larger magnet for equivalent performance.

21. Driver design—including the particular way two or more drivers are combined in an enclosure to make a speaker system—is both an art, involving subjective perceptions of timbre and sound quality and a science, involving measurements and experiments.

22. Adjusting a design to improve performance is done using a combination of magnetic, acoustic, mechanical, electrical, and material science theory, and tracked with high precision measurements and the observations of experienced listeners.

23. A few of the issues speaker and driver designers must confront are distortion, radiation lobing, phase effects, off-axis response, and crossover artifacts. Designers can use an anechoic chamber to ensure the speaker can be measured independently of room effects, or any of several electronic techniques that, to some extent, substitute for such chambers.

24. Some developers eschew anechoic chambers in favor of specific standardized room setups intended to simulate real-life listening conditions.

25. Fabrication of finished loudspeaker systems has become segmented, depending largely on price, shipping costs, and weight limitations. High-end speaker systems, which are typically heavier (and often larger) than economic shipping allows outside local regions, are usually made in their target market region and can cost $140,000 or more per pair.

Fig 3.2

E. Full-range drivers

1. A full-range driver is a speaker designed to be used alone to reproduce an audio channel without the help of other drivers, and therefore must cover the entire audio frequency range.
2. These drivers are small, typically 3 to 8 inches (7.6 to 20.3 cm) in diameter to permit reasonable high frequency response, and carefully designed to give low-distortion output at low frequencies, though with reduced maximum output level.
3. Full-range (or more accurately, wide-range) drivers are most commonly heard in public address systems, in televisions (although some models are suitable for hi-fi listening), small radios, intercoms, some computer speakers, etc.
4. In hi-fi speaker systems, the use of wide-range drive units can avoid undesirable interactions between multiple drivers caused by non-coincident driver location or crossover network issues.
5. Fans of wide-range driver hi-fi speaker systems claim a coherence of sound due to the single source and a resulting lack of interference, and likely also to the lack of crossover components.
6. Detractors typically cite wide-range drivers' limited frequency response and modest output abilities (most especially at low frequencies), together with their requirement for large, elaborate, expensive enclosures—such as transmission lines, quarter wave resonators or horns—to approach optimum performance.
7. With the advent of neodymium drivers, low-cost quarter-wave transmission lines are made possible and are increasingly made available commercially.
8. Full-range drivers often employ an additional cone called a *whizzer*: a small, light cone attached to the joint between the voice coil and the primary cone. The whizzer cone extends the high-frequency response of the driver and broadens its high frequency directivity, which would otherwise be greatly narrowed due to the outer diameter cone material failing to keep up with the central voice coil at higher frequencies.

9. The main cone in a whizzer design is manufactured so as to flex more in the outer diameter than in the center. The result is that the main cone delivers low frequencies and the whizzer cone contributes most of the higher frequencies.

10. Since the whizzer cone is smaller than the main diaphragm, output dispersion at high frequencies is improved relative to an equivalent single larger diaphragm.

11. Limited-range drivers, also used alone, are typically found in computers, toys, and clock radios. These drivers are less elaborate and less expensive than wide-range drivers, and they may be severely compromised to fit into very small mounting locations.

12. In these applications, sound quality is a low priority. The human ear is remarkably tolerant of poor sound quality, and the distortion inherent in limited-range drivers may enhance their output at high frequencies, increasing clarity when listening to spoken word material.

F. Subwoofer

1. A subwoofer is a woofer driver used only for the lowest-pitched part of the audio spectrum: typically below 200 Hz for consumer systems, below 100 Hz for professional live sound,] and below 80 Hz in THX-approved systems.

2. Because the intended range of frequencies is limited, subwoofer system design is usually simpler in many respects than for conventional loudspeakers, often consisting of a single driver enclosed in a suitable box or enclosure.

3. Since sound in this frequency range can easily bend around corners by diffraction, the speaker aperture does not have to face the audience, and subwoofers can be mounted in the bottom of the enclosure, facing the floor.

4. This is eased by the limitations of human hearing at low frequencies; such sounds cannot be located in space, due to their large wavelengths compared to higher frequencies which produce differential effects in the ears due to shadowing by the head, and diffraction around it, both of which we rely upon for localization clues.

5. To accurately reproduce very low bass notes without unwanted resonances (typically from cabinet panels), subwoofer systems must be solidly constructed and properly braced to avoid unwanted sounds of cabinet vibrations.
6. As a result, good subwoofers are typically quite heavy. Many subwoofer systems include integrated power amplifiers and electronic subsonic (sub)-filters, with additional controls relevant to low-frequency reproduction (e.g., a crossover knob and a phase switch).
7. These variants are known as "active" or "powered" subwoofers, with the former including a power amplifier. In contrast, "passive" subwoofers require external amplification.
8. In typical installations, subwoofers are physically separated from the rest of the speaker cabinets. Because of propagation delay, their output may be somewhat out of phase from another subwoofer (on another channel) or slightly out of phase with the rest of the sound.
9. Consequently, a subwoofer's power amp often has a phase-delay adjustment (approximately 1 ms of delay is required for each additional foot of separation from the listener) which may improve performance of the system as a whole at subwoofer frequencies (and perhaps an octave or so above the crossover point).
10. However, the influence of room resonances (sometimes called standing waves) is typically so large that such issues are secondary in practice. Subwoofers are widely used in large concert and mid-sized venue sound reinforcement systems. Subwoofer cabinets are often built with a bass reflex port (i.e., a hole cut into the cabinet with a tube attached to it), a design feature which if properly engineered improves bass performance and increases efficiency.

G. Woofer

1. A woofer is a driver that reproduces low frequencies. The driver works with the characteristics of the enclosure to produce suitable low frequencies (see speaker enclosure for some of the design choices available).

2. Indeed, both are so closely connected that they must be considered together in use. Only at design time do the separate properties of enclosure and woofer matter individually. Some loudspeaker systems use a woofer for the lowest frequencies, sometimes well enough that a subwoofer is not needed.

3. Additionally, some loudspeakers use the woofer to handle middle frequencies, eliminating the mid-range driver. This can be accomplished with the selection of a tweeter that can work low enough that, combined with a woofer that responds high enough, the two drivers add coherently in the middle frequencies.

H. Mid-range driver

1. A mid-range speaker is a loudspeaker driver that reproduces a band of frequencies generally between 1–6 kHz, otherwise known as the 'mid' frequencies (between the woofer and tweeter). Mid-range driver diaphragms can be made of paper or composite materials, and can be direct radiation drivers (rather like smaller woofers) or they can be compression drivers (rather like some tweeter designs).

2. If the mid-range driver is a direct radiator, it can be mounted on the front baffle of a loudspeaker enclosure, or, if a compression driver, mounted at the throat of a horn for added output level and control of radiation pattern.

I. Tweeter

Fig 3.3

1. A tweeter is a high-frequency driver that reproduces the highest frequencies in a speaker system. A major problem in tweeter design is achieving wide angular sound coverage (off-axis response), since high frequency sound tends to leave the speaker in narrow beams.
2. Soft-dome tweeters are widely found in home stereo systems, and horn-loaded compression drivers are common in professional sound reinforcement.
3. Ribbon tweeters have gained popularity in recent years, as the output power of some designs has been increased to levels useful for professional sound reinforcement, and their output pattern is wide in the horizontal plane, a pattern that has convenient applications in concert sound.

3.4 DIRECTIONAL SPEAKER

A. THEORY

Ultrasonic Nonlinear Acoustics :

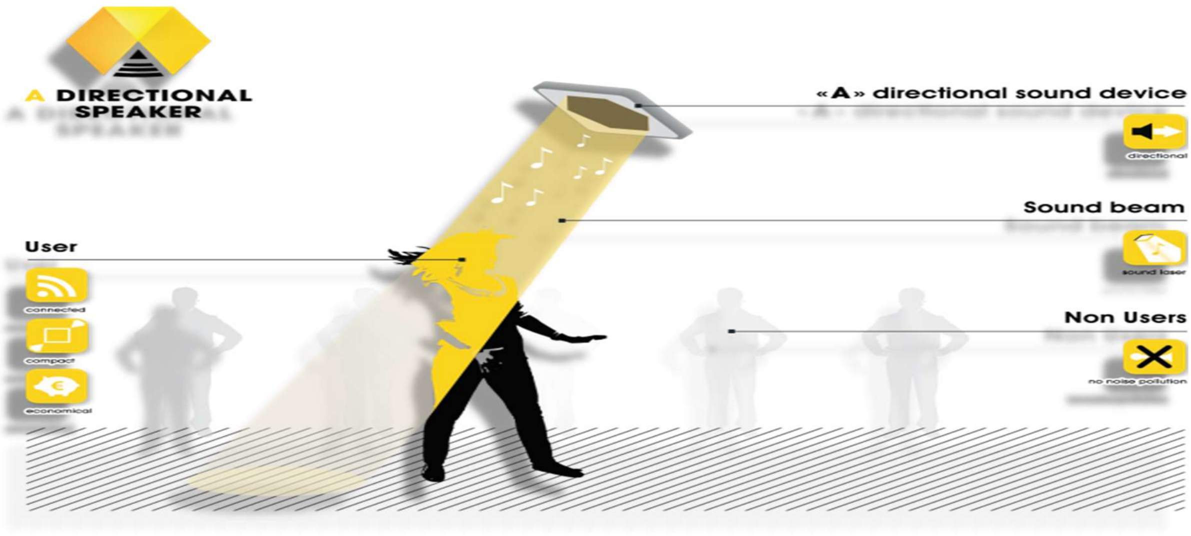


Fig 3.4

1. A chronological summary of the experimental approaches taken to examine Audio Spotlight systems in the past will be presented here. At the turn of the millennium working versions of an Audio Spotlight capable of reproducing speech and music could be bought from Holosonics, a company founded on Dr. Pompei's work in the MIT Media Lab.

2. Related topics were researched almost 40 years earlier in the context of underwater acoustics.
3. The first article consisted of a theoretical formulation of the half pressure angle of the demodulated signal.
4. The second article provided an experimental comparison to the theoretical predictions.
5. Both articles were supported by the U.S. Office of Naval Research, specifically for the use of the phenomenon for underwater sonar pulses.
6. The goal of these systems was not high directivity *per se*, but rather higher usable bandwidth of a typically band-limited transducer.
7. The 1970s saw some activity in experimental airborne systems, both in air and underwater. Again supported by the U.S. Office of Naval Research, the primary aim of the underwater experiments was to determine the range limitations of sonar pulse propagation due to nonlinear distortion.
8. The airborne experiments were aimed at recording quantitative data about the directivity and propagation loss of both the ultrasonic carrier and demodulated waves, rather than developing the capability to reproduce an audio signal.
9. In 1983 the idea was again revisited experimentally but this time with the firm intent to analyze the use of the system in air to form a more complex base band signal in a highly directional manner.
10. The signal processing used to achieve this was simple DSB-AM with no precompensation, and because of the lack of precompensation applied to the input signal, the THD Total harmonic distortion levels of this system would have probably been satisfactory for speech reproduction, but prohibitive for the reproduction of music.
11. An interesting feature of the experimental set up used in was the use of 547 ultrasonic transducers to produce a 40 kHz ultrasonic sound source of over 130db at 4 m, which would demand significant safety considerations.

12. Even though this experiment clearly demonstrated the potential to reproduce audio signals using an ultrasonic system, it also showed that the system suffered from heavy distortion, especially when no precompensation was used.

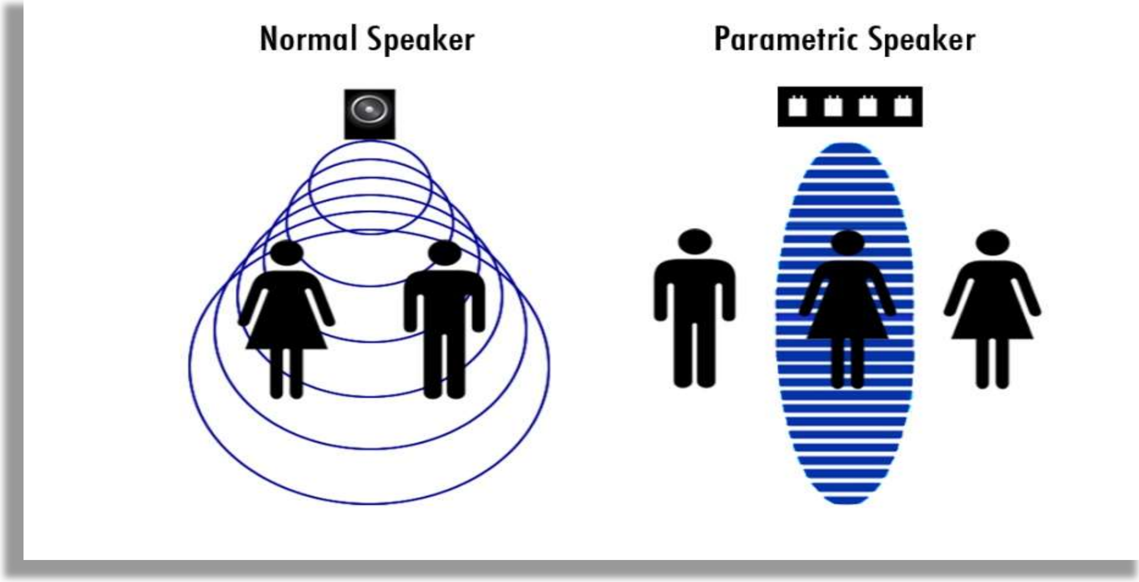


Fig 3.5 Theoretical ultrasonic nonlinear acoustics

13. The equations that govern nonlinear acoustics are quite complicated and unfortunately, they do not have general analytical solutions.

14. They usually require the use of a computer simulation. However, as early as 1965, Berktay performed an analysis under some simplifying assumptions that allowed the demodulated SPL to be written in terms of the amplitude modulated ultrasonic carrier wave pressure P_c and various physical parameters.

15. Note that the demodulation process is extremely lossy, with a minimum loss in the order of 60 dB from the ultrasonic SPL to the audible wave SPL.

16. A precompensation scheme can be based from Berktay's expression, shown in Equation 1, by taking the square root of the base band signal envelope E and then integrating twice to invert the effect of the double partial time derivative.

17. The analogue electronic circuit equivalents of a square root function is simply an op-amp with feedback, and an equalizer is analogous to an integration function. However, these topic areas lie outside the scope of this project.

$$P(x,t) \ = \ K.P^{2} \cdot \frac{\partial^2}{\partial t^2} \ E(x,t)^2$$

Audible secondary pressure wave misc. physical parameters SPL of the ultrasonic carrier wave Envelope function (such as DSB-AM)

18. This equation says that the audible demodulated ultrasonic pressure wave (output signal) is proportional to the twice differentiated, squared version of the envelope function (input signal). Precompensation refers to the trick of anticipating these transforms and applying the inverse transforms on the input, hoping that the output is then closer to the untransformed input.

19. By the 1990s, it was well known that the Audio Spotlight could work but suffered from heavy distortion. It was also known that the precompensation schemes placed an added demand on the frequency response of the ultrasonic transducers.

20. In effect the transducers needed to keep up with what the digital precompensation demanded of them, namely a broader frequency response. In 1998 the negative effects on THD of an insufficiently broad frequency response of the ultrasonic transducers was quantified with computer simulations by using a precompensation scheme based on Berktay's expression.

21. In 1999 Pompei's article discussed how a new prototype transducer met the increased frequency response demands placed on the ultrasonic transducers by the precompensation scheme, which was once again based on Berktay's expression. In addition impressive reductions in the THD of the output when the precompensation scheme was employed were graphed against the case of using no precompensation.

22. In summary, the technology that originated with underwater sonar 40 years ago has been made practical for reproduction of audible sound in air by Pompei's paper and device, which, according to

his AES paper (1998), demonstrated that distortion had been reduced to levels comparable to traditional loudspeaker systems.

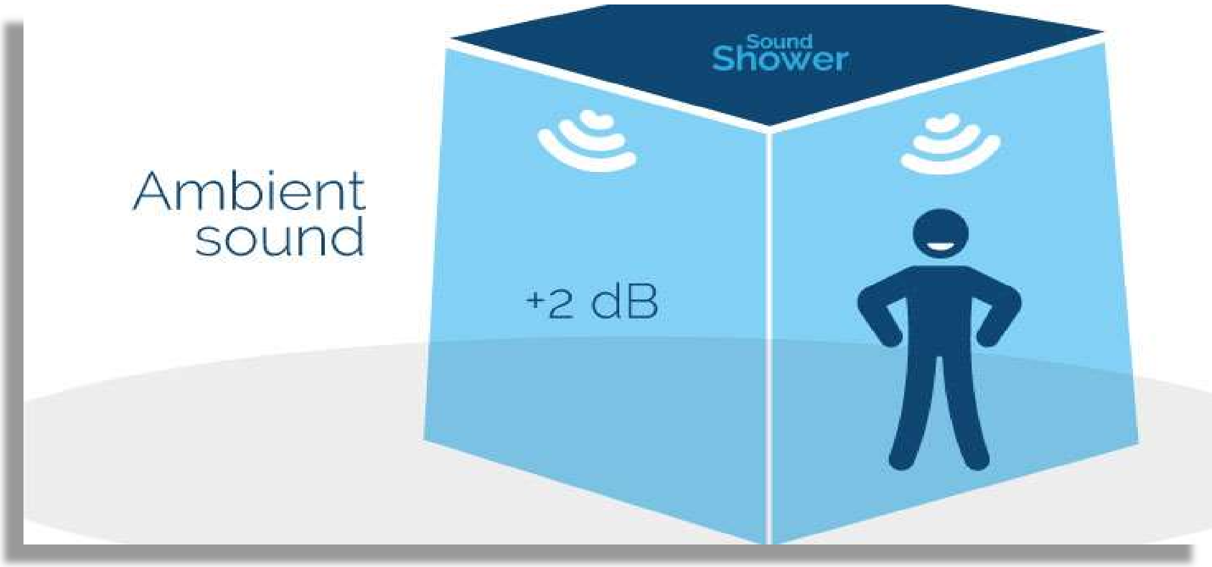


Fig 3.6

B. Hypersonic Sound:

1. Elwood "Woody" Norris, founder and Chairman of American Technology Corporation (ATC), announced he had successfully created a device which achieved ultrasound transmission of sound in 1996.
2. This device used piezoelectric transducers to send two ultrasonic waves of differing frequencies toward a point, giving the illusion that the audible sound from their interference pattern was originating at that point.
3. ATC named and trademarked their device as "Hypersonic Sound" (HSS). In December 1997, HSS was one of the items in the Best of What's New issue of Popular Science.
4. In December 2002, Popular Science named Hypersonic Sound the best invention of 2002.[*citation needed*] Norris received the 2005 Lemelson-MIT Prize for his invention of a "hypersonic sound".

5. ATC (now named LRAD Corporation) spun off the technology to Parametric Sound Corporation in September 2010 to focus on their Long-Range Acoustic Device products (LRAD), according to their quarterly reports, press releases and executive statements.

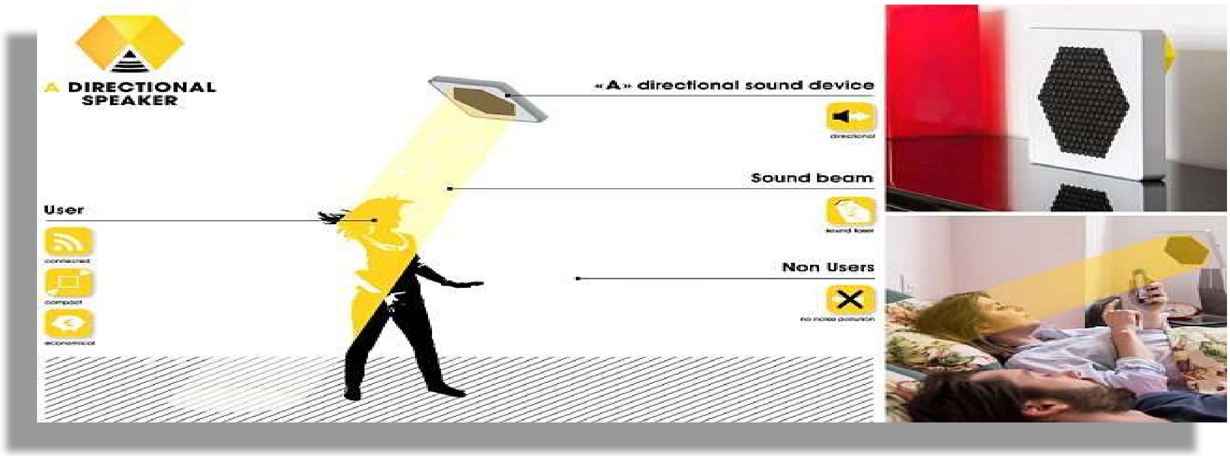


Fig 3.7

C. Safe use of high-intensity ultrasound

1. For the nonlinear effect to occur, relatively high intensity ultrasonics are required. The SPL involved was typically greater than 100 dB of ultrasound at a nominal distance of 1 m from the face of the ultrasonic transducer.[*citation needed*]

2. Exposure to more intense ultrasound over 140 dB[*citation needed*] near the audible range (20–40 kHz) can lead to a syndrome involving manifestations of nausea, headache, tinnitus, pain, dizziness and fatigue, but this is around 100 times the 100 dB level cited above, and is generally not a concern.

3. Dr Joseph Pompei of Audio Spotlight has published data showing that their product generates ultrasonic sound pressure levels around 130 dB (at 60 kHz) measured at 3 meters.

4. The UK's independent Advisory Group on Non-ionising Radiation (AGNIR) produced a 180-page report on the health effects of human exposure to ultrasound and infrasound in 2010.

5. The UK Health Protection Agency (HPA) published their report, which recommended an exposure limit for the general public to airborne ultrasound sound pressure levels (SPL) of 100 dB (at 25 kHz and above).
6. OSHA specifies a safe ceiling value of ultrasound as 145 dB SPL exposure at the frequency range used by commercial systems in air, as long as there is no possibility of contact with the transducer surface or coupling medium (i.e. submerged). T
7. his is several times the highest levels used by commercial Audio Spotlight systems, so there is a significant margin for safety[*citation needed*]. In a review of international acceptable exposure limits
8. Howard et al. (2005)noted the general agreement amongst standards organizations, but expressed concern with the decision by United States of America's Occupational Safety and Health Administration (OSHA) to increase the exposure limit by an additional 30 dB under some conditions (equivalent to a factor of 1000 in intensity).
9. For frequencies of ultrasound from 25 to 50 kHz, a guideline of 110 dB had been recommended by Canada, Japan, the USSR, and the International Radiation Protection Agency, and 115 dB by Sweden in the late 1970s to early 1980s, but these were primarily based on subjective effects.
10. The more recent OSHA guidelines above are based on ACGIH (American Conference of Governmental Industrial Hygienists) research from 1987.

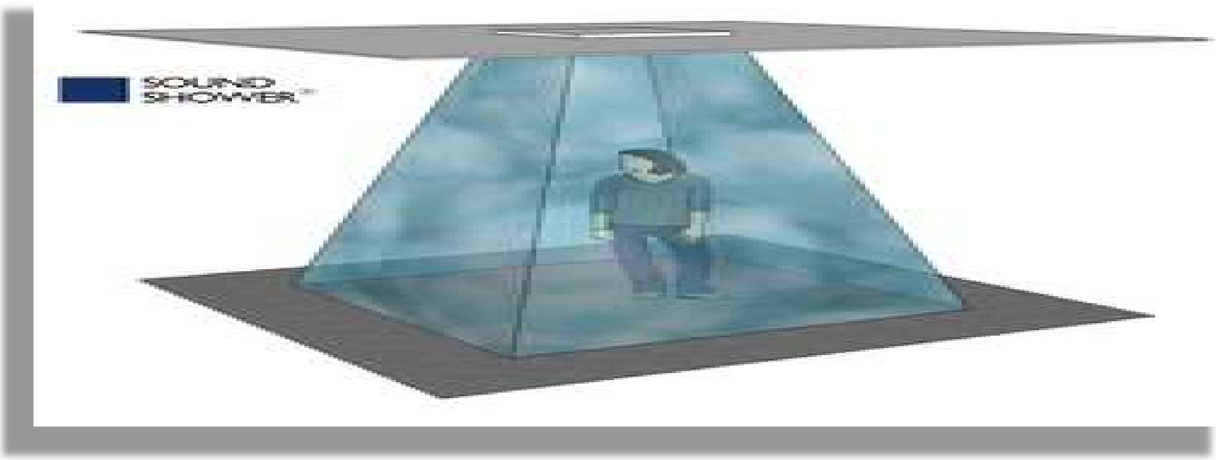


Fig 3.8

CHAPTER 4

DESIGN PARAMERS AND

EXPERIMENTAL SETUP

4.1 . PARRAMETRIC ARRAY CONSIDERATION

1. Since the early 1960s, researchers have been experimenting with creating directive low-frequency sound from nonlinear interaction of an aimed beam of ultrasound waves produced by a parametric array using heterodyning.
2. Ultrasound has much shorter wavelengths than audible sound, so that it propagates in a much narrower beam than any normal loudspeaker system using audio frequencies. Most of the work was performed in liquids (for underwater sound use).
3. The first modern device for air acoustic use was created in 1998, and is now known by the trademark name "Audio Spotlight", a term first coined in 1983 by the Japanese researchers who abandoned the technology as infeasible in the mid-1980s.
4. A transducer can be made to project a narrow beam of modulated ultrasound that is powerful enough, at 100 to 110 dBSPL, to substantially change the speed of sound in the air that it passes through.
5. The air within the beam behaves nonlinearly and extracts the modulation signal from the ultrasound, resulting in sound that can be heard only along the path of the beam, or that appears to radiate from any surface that the beam strikes.
6. This technology allows a beam of sound to be projected over a long distance to be heard only in a small well-defined area; for a listener outside the beam the Sound pressure decreases substantially. This effect cannot be achieved with conventional loudspeakers, because sound at audible frequencies cannot be focused into such a narrow beam.
7. There are some limitations with this approach. Anything that interrupts the beam will prevent the ultrasound from propagating, like interrupting a spotlight's beam. For this reason, most systems are mounted overhead, like lighting.

8. The main component in the bounded hypersonic heavy acoustic system is parametric array(PA) consideration. A uniform planar having a triangular lattice grid contains 25 elements for testing our assumptions as shown in Fig .2. .

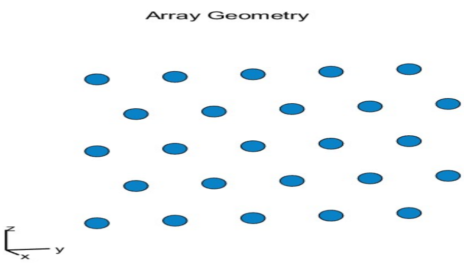


Fig 4.1

Fig 4.2

4.2 ULTRASONIC TRANSDUCER CONSIDERATION

- Ultrasonic transducers convert AC into ultrasound, as well as the reverse. Ultrasonics, typically refers to piezoelectric transducers or capacitive transducers.
- Piezoelectric crystals change size and shape when a voltage is applied; AC voltage makes them oscillate at the same frequency and produce ultrasonic sound. Capacitive transducers use electrostatic fields between a conductive diaphragm and a backing plate.
- The beam pattern of a transducer can be determined by the active transducer area and shape, the ultrasound wavelength, and the sound velocity of the propagation medium.
- The diagrams show the sound fields of an unfocused and a focusing ultrasonic transducer in water, plainly at differing energy levels.
- Since piezoelectric materials generate a voltage when force is applied to them, they can also work as ultrasonic detectors.
- Some systems use separate transmitters and receivers, while others combine both functions into a single piezoelectric transceiver.

- Ultrasound transmitters can also use non-piezoelectric principles. such as magnetostriction. Materials with this property change size slightly when exposed to a magnetic field, and make practical transducers.
- A capacitor ("condenser") microphone has a thin diaphragm that responds to ultrasound waves. Changes in the electric field between the diaphragm and a closely spaced backing plate convert sound signals to electric currents, which can be amplified.

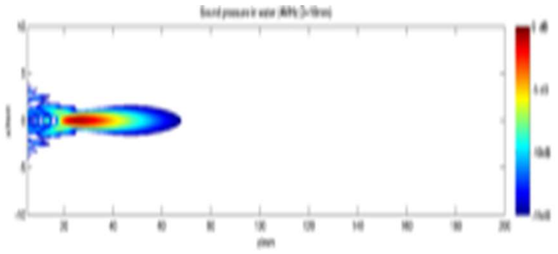


Fig 4.3

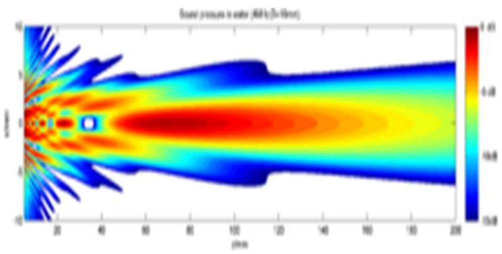


Fig 4.4

- The diaphragm (or membrane) principle is also used in the relatively new micro-machined ultrasonic transducers (MUTs). These devices are fabricated using silicon micro-machining technology (MEMS technology), which is particularly useful for the fabrication of transducer arrays.
- The vibration of the diaphragm may be measured or induced electronically using the capacitance between the diaphragm and a closely spaced backing plate (CMUT), or by adding a thin layer of piezo-electric material on diaphragm (PMUT).
- Alternatively, recent research showed that the vibration of the diaphragm may be measured by a tiny optical ring resonator integrated inside the diaphragm (OMUS).
- Ultrasonic Transducers are also used in acoustic levitation.
- Choosing an ultrasonic transducer for converting electrical signals into sound waves having a very high frequency which is optimum in both cost and performance-wise. Modern speakers produce a frequency range up to 40hz to 20khz.
- But the speakers' requirements have to handle high frequencies for sound transmission in place of the antenna.The need is fulfilled by choosing an appropriate transducer, as shown in Fig 3. .

Fig 4.5

4.3 ACOUSTIC ABSORPTION FOAM CONSIDERATION

- The next one is Acoustic foam consideration for sound absorption and proofing. This paper used YGM Acoustic Foams Pyramid Soundproofing Studio Acoustic Foam for testing as shown in Fig 5..After fired ultrasound from the parametric array;
- Whenever it strikes the people, it will demodulate. Then the sound spreads surroundings in the desired direction. The main drawback is the gain of the directive sound will exponentially decrease for a single piece of paper obstruction.
- By using this disadvantage to make a heavy hypersonic bounded acoustic system by arranging acoustic foams around the surroundings for absorbing the reflections created by the directive speaker array.
- Acoustic foam is an open celled foam used for acoustic treatment. It attenuates airbone sound waves, reducing their amplitude, for the purposes of noise reduction or noise control.The energy is dissipated as heat.
- Acoustic foam can be made in several different colors, sizes and thickness.

- Acoustic foam can be attached to walls, ceilings, doors, and other features of a room to control noise levels, vibration, and echoes.
- Many acoustic foam products are treated with dyes and/or fire retardants.

Fig 4.6

4.4 CIRCUIT FLOW OF HEAVY HYPER SOUND ACOUSTIC SYSTEM CONSIDERATION

I. Input signal and preamp stage :

a. The 'non linear characteristic' is due to the fact that it takes more time for air molecules to be restored to their original density than to be compressed .

b. When the sound pressure is high, and frequency too, a shock wave may be produced by returning air molecules colliding with the ones being compressed. In fact, an audible sound is produced by any molecule not completely 'returning'.

c. When the frequency of the vibration rises, the 'non-linear characteristic" tends to become noticeable by an effect best described as 'air viscosity'. There is another reason for the high directivity (i.e. small beamwidth) exhibited by a parametric speaker array.

d. The supersonic wave is actually generated by a large number of small loudspeakers called transducers. The piezo-electric transducer is widely used both as a sensor and a transmitting device in car and home automatic systems.

e. The directivity of the piezo transducer by itself is not too high. However, strength is in numbers, meaning the high directivity is due to many small transducers arranged in a plane-like shape. This is essential for making a truly directional speaker unit.

Fig 4.7

II. Modulation schemes:

- The nonlinear interaction mixes ultrasonic tones in air to produce sum and difference frequencies. A DSB-AM modulation scheme with an appropriately large baseband DC offset, to produce the demodulating tone superimposed on the modulated audio spectrum, is one way to generate the signal that encodes the desired baseband audio spectrum.
- This technique suffers from extremely heavy distortion as not only the demodulating tone interferes, but also all other frequencies present interfere with one another. The modulated spectrum is convolved with itself, doubling its bandwidth by the length property of the convolution.
- The baseband distortion in the bandwidth of the original audio spectrum is inversely proportional to the magnitude of the DC offset (demodulation tone) superimposed on the signal. A larger tone results in less distortion.
- Further distortion is introduced by the second order differentiation property of the demodulation process. The result is a multiplication of the desired signal by the function $-\omega^2$ in frequency. This distortion may be equalized out with the use of preemphasis filtering (increase amplitude of high frequency signal).
- By the time convolution property of the Fourier transform, multiplication in the time domain is a convolution in the frequency domain. Convolution between a baseband signal and a unity gain pure carrier frequency shifts the baseband spectrum in frequency and halves its magnitude, though no energy is lost. One half-scale copy of the replica resides on each half of the frequency axis. This is consistent with Parseval's theorem.
- The modulation depth m is a convenient experimental parameter when assessing the total harmonic distortion in the demodulated signal. It is inversely proportional to the magnitude of the DC offset. THD increases proportionally with m_1^2.
- These distorting effects may be better mitigated by using another modulation scheme that takes advantage of the differential squaring device nature of the nonlinear acoustic effect. Modulation of

the second integral of the square root of the desired baseband audio signal, without adding a DC offset, results in convolution in frequency of the modulated square-root spectrum, half the bandwidth of the original signal, with itself due to the nonlinear channel effects. This convolution in frequency is a multiplication in time of the signal by itself, or a squaring.

- This again doubles the bandwidth of the spectrum, reproducing the second time integral of the input audio spectrum. The double integration corrects for the $-\omega^2$ filtering characteristic associated with the nonlinear acoustic effect. This recovers the scaled original spectrum at baseband.
- The harmonic distortion process has to do with the high frequency replicas associated with each squaring demodulation, for either modulation scheme. These iteratively demodulate and self-modulate, adding a spectrally smeared out and time exponentiated copy of the original signal to baseband and twice the original center frequency each time, with one iteration corresponding to one traversal of the space between the emitter and target.
- Only sound with parallel collinear phase velocity vectors interfere to produce this nonlinear effect. Even-numbered iterations will produce their modulation products, baseband and high frequency, as reflected emissions from the target. Odd-numbered iterations will produce their modulation products as reflected emissions off the emitter.
- This effect still holds when the emitter and the reflector are not parallel, though due to diffraction effects the baseband products of each iteration will originate from a different location each time, with the originating location corresponding to the path of the reflected high frequency self-modulation products.
- These harmonic copies are largely attenuated by the natural losses at those higher frequencies when propagating through air.

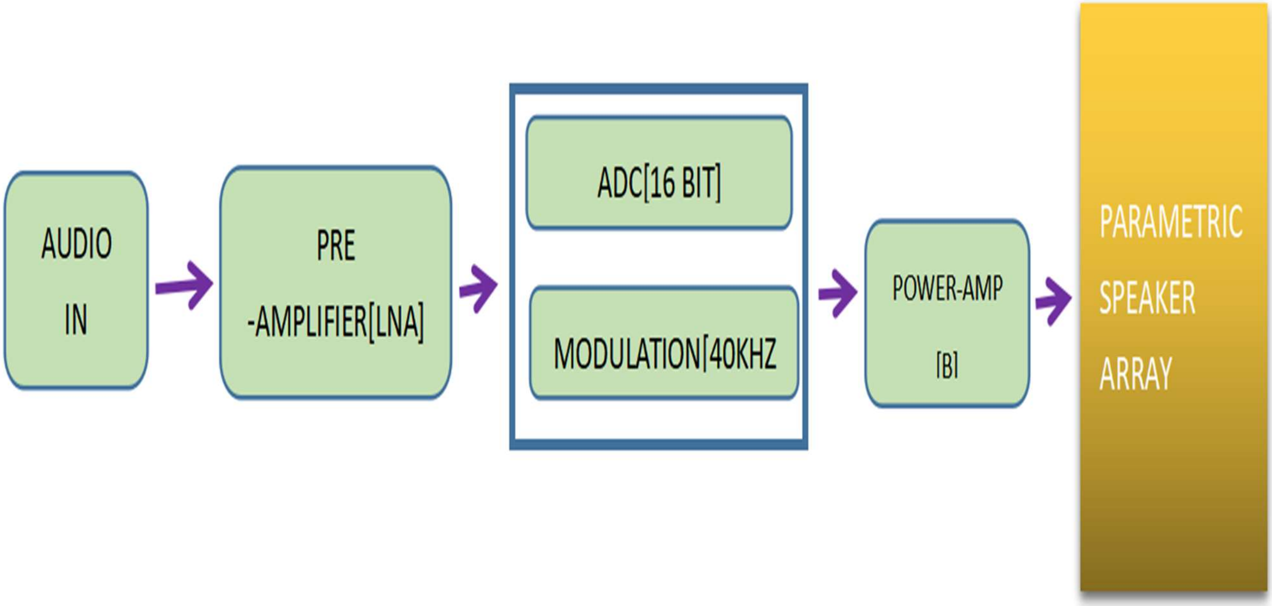


Fig 4.8

III. Power amplification and Tuning:

- Double sideband modulation (DSB) is easily implemented using analogue switches. Frequency Modulation (FM) has the same effects basically if you look at the way supersonic sound waves compress air and interact.
- The first attempted a DSB modulator. The result: big sound, lots of distortion and the method might be suitable for a sound beam weapon. Next, a PWM system was built. Looking at the net result, PWM is very similar to FM. The audible sound obtained from PWM is weaker than from DSB, but of a better quality. A PWM modulator may be compared to a class-D amplifier without its low pass filter! The schematic of a 2-channel PWM modulator is given in Figure 3. There are no special components. The TL494 PWM control circuit and the IR2111 half bridge MOSFET driver are used in their standard application circuits.
- The TL494 has an internal oscillator whose frequency is determined by trimpot R2 and capacitor C1. The basic pulse width is adjusted with R1. You need to set up optimum modulation with trimpots R1 and R2. The audio input signal is connected to K2 (loudspeaker level required, not microphone or line). The board has two outputs, A and B, each driving an array of piezo transducers, optionally

through an inductor (see below). Each channel is suitable for up to 200 transducers. The normal supply voltage is 20-24 DVC to K1.

- The FET stages may be powered by an external supply via the EXT terminal after removing wire link J1. Heatsinks may be required on the IRF540 FETs depending on the supply voltage and the transducers' ratings (up to 60 VDC may be possible).
- The U/S speaker schematic is large but unsurprising, see Figure 4. It represents one channel and a 'mini' version with just 50 transducers. Speaker unit and optional coil There are several type of ultrasonic transducer around. The author used 16 mm diameter devices specified for 40 kHz and 28 kHz. A minimum of 50 transducers is required to make an effective speaker unit. You need more than 100 transducers if you want to the unit to have any sort of range outdoors.
- All transducers should be carefully distributed to maintain phase. Remember, the wavelength is about 8 mm so a positioning error of 1 mm causes phase errors and loss of SPL. Ultrasonic transducers are made from piezoelectric ceramic materials.
- When a voltage is applied to the device, a special type of foil is deformed inside, generating a supersonic sound wave of a specific frequency. Typically, the transducer's sound output reaches 105–120 dB (at 30 cm distance) when a voltage of 10–20 Vrms is applied

CHAPTER 5

SOFTWARE SIMULATION

5.1 MATLAB AND TOOL BOX

Phased Array System Toolbox:

- Phased Array System Toolbox™ provides algorithms and apps for designing and simulating sensor array and beamforming systems in wireless communication, radar, sonar, acoustic, and medical imaging applications.

- You can model and analyze the behaviour of active and passive arrays, including subarrays and arbitrary geometries. Simulated signals can be transmitted and received by these arrays for beamforming and signal processing algorithm design.

- For 5G and LTE cellular, SATCOM, and WLAN communications systems, you can design multibeam and electronically steerable antennas.

- The toolbox includes algorithms for simulating hybrid and full digital beamforming architectures for massive MIMO and millimeter wave systems. You can simulate multipath fading environments to test the performance of beamforming antenna arrays.

- For radar, sonar, and acoustic system design, the toolbox includes signal processing algorithms for beamforming, space-time adaptive processing (STAP), direction of arrival (DOA) estimation, matched filtering, and signal detection.

- The toolbox also provides continuous and pulsed waveforms that you can use to generate test signals and simulate target echoes, interferences, and propagation effects.

- For simulation acceleration or desktop prototyping, the toolbox supports C code generation. Reference examples provide workflows for generating HDL code from Simulink® models.

Simulation Result:

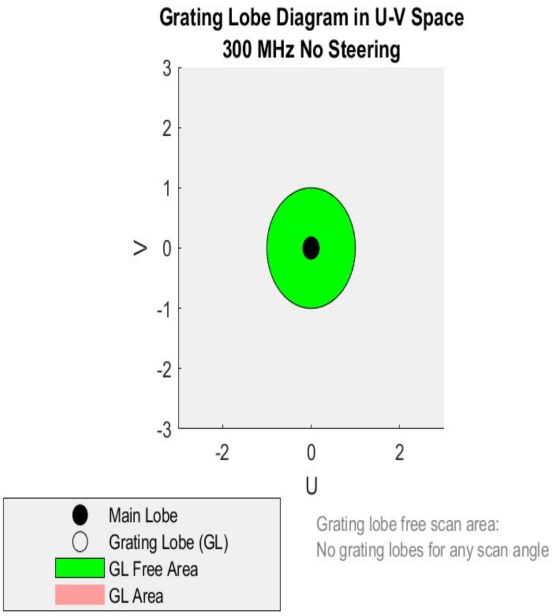


Fig 5.1

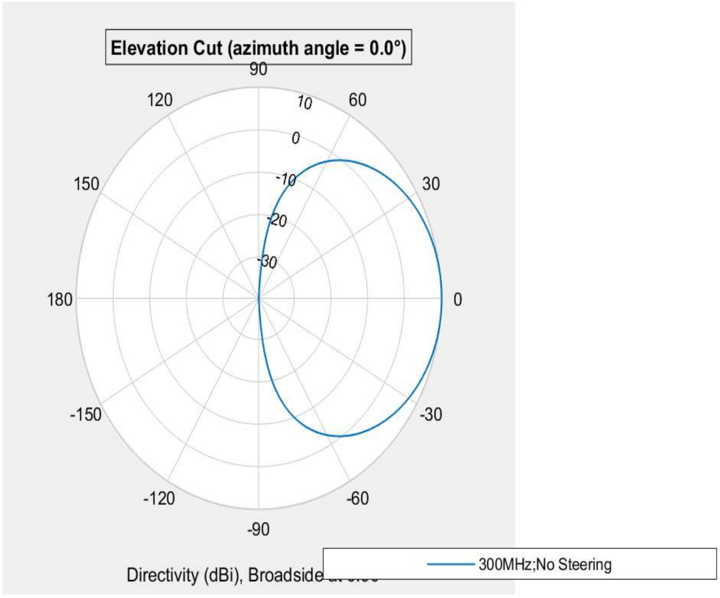


Fig 5.2

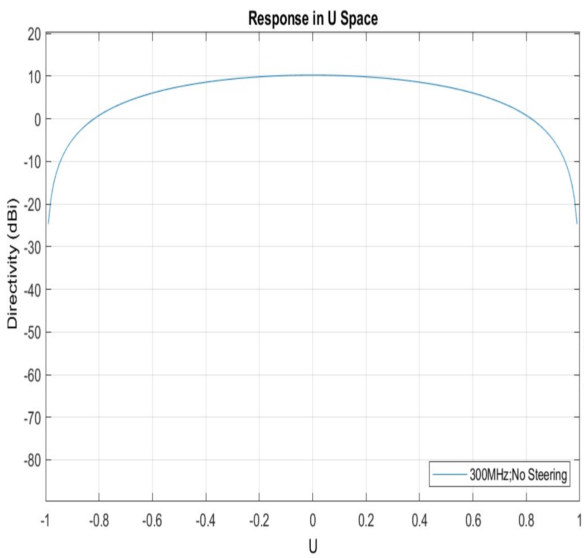


Fig 5.3

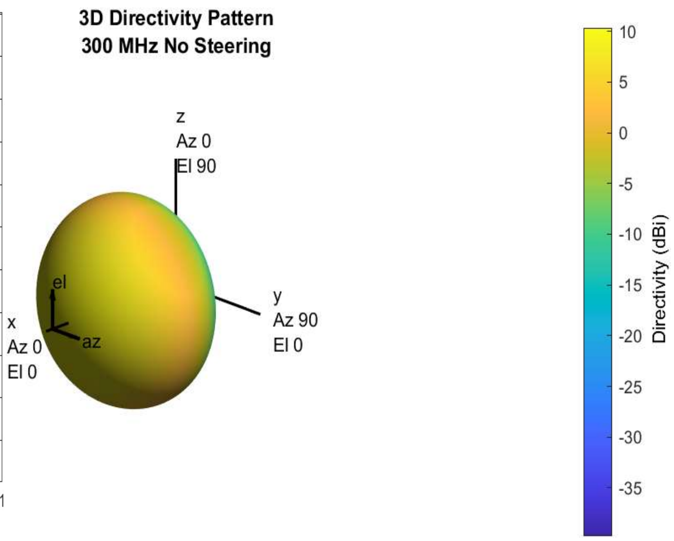


Fig 5.4

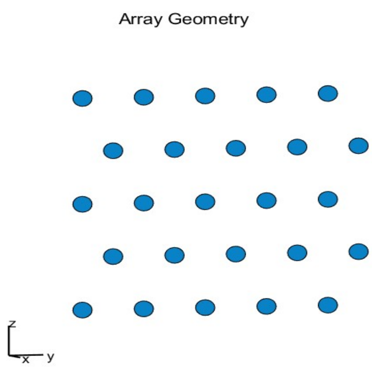


Fig 5.5

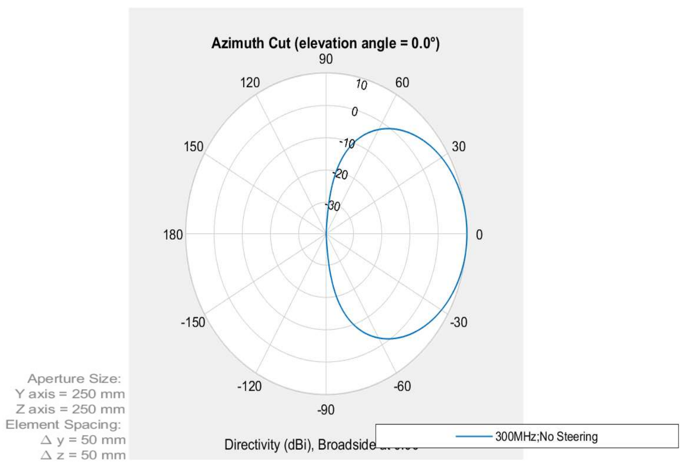


Fig 5.6

➢ The parametric array is simulated using 300Mhz frequency through software tools. The directional pattern and elevation angle are indicated in Fig 7. and Fig 8. respectively.

➢ The direction pattern depicts the ultrasound will be propagates in a single direction. The elevation angle tells; how much space will be covered.

➢ When considering these conditions for 40 kHz, the parameters are not much good because of the decrease in the frequency.

➢ In general, antennas use very high frequencies, unlike less than 100 kHz. From the Fig 9. and Fig 10. the directivity and the Elevation angle of the parametric array is substantial when compared to the higher frequencies.

Audio from directed speaker:

➢ Now examine the acoustic setup furnished in this paper. Measure the audio signal directed towards the wall and the acoustic foam, then find the difference between them.

➢ The audio file is recorded from the directive speaker, and it is processed using software simulation. The Fig 11. and Fig 12. demonstrates the histogram and spectrogram of the recorded audio signal taken from the directive speaker.

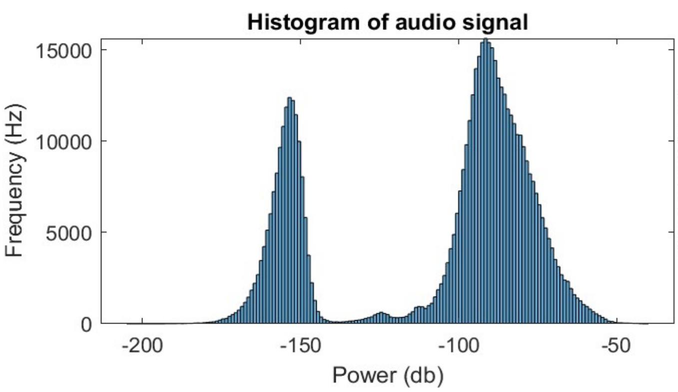


Fig 5.7

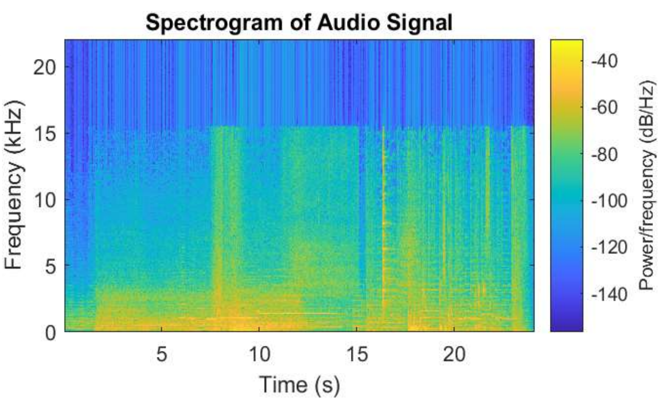


Fig 5.8

- From the histogram Fig 11. and Spectrogram Fig 12. ; the audio signal is in the range of 0-25 kHz, and max power occurs at higher frequencies, nearly 100 dB. High peaks of audio signal occurred in 170 and 100 dB having 10kHz and 15kHz frequency, respectively.
- After the directive sound reflected by the acoustic foam, the power level will reduce as per Fig 13. and Fig 14. which are simulated using the software. For every strike with acoustic foam, the power drops; after some time, sound level seizes to zero.
- This also reduces the reflections which are created when it strikes with the people or things, as shown in Fig 1. .

Code:

```
[x,fs]=audioread('j.wav');
x=x(:,1);
N=length(x);
t=(0:N-1)/fs;
%%%Time domain audio signal
subplot(2,2,1);
plot(t,x);
grid on
xlabel("Time  (sec)");
ylabel("Amplitude  (volts)");
title("Audio Signal in Time Domain");
%%%time domain values
maxvalue=max(x);
minvalue=min(x);
meanvalue=mean(x);
stdvalue=std(x);
%spectrogram of audio signal
subplot(2,2,2);
spectrogram(x,1024,512,1024,fs,'yaxis');
title("Spectrogram of Audio Signal");
%periodogram Plot
subplot(2,2,3);
w=hanning(N,'periodic');
```

```
periodogram(x,w,N,fs,'power');
%histogram of input audio sinal
subplot(2,2,4);
histogram(y);
xlabel("Power (db)");
ylabel("Frequency (Hz)");
title("Histogram of audio signal");
```

5.2 MULTI SIM CIRCUIT DESIGN

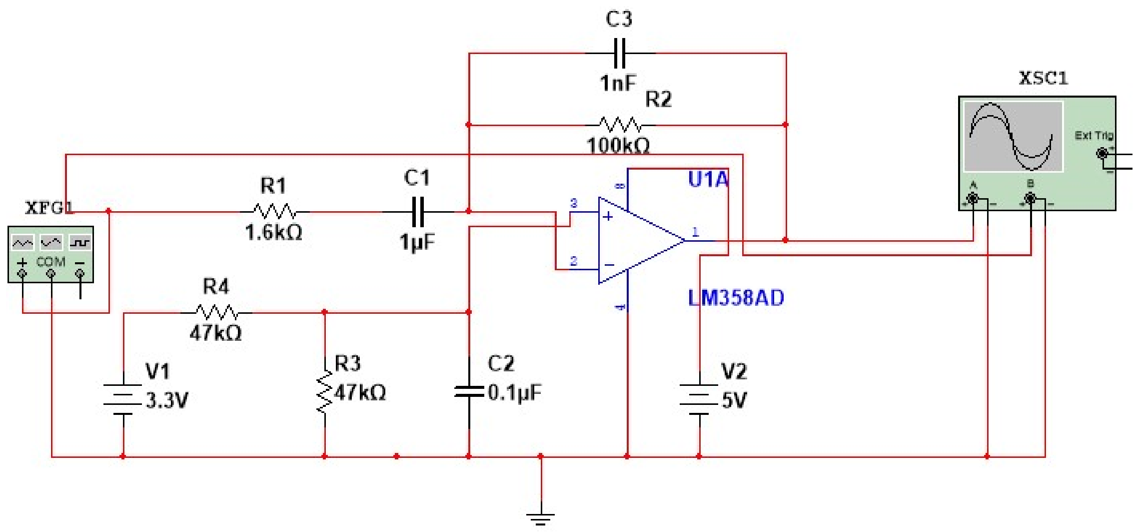


Fig 5.9

- Simple amplifier is designed using multi sim software and will be verified after considering final output
- LM358 is used to amplify the input signal and then passed into modulator which modulates the signal with 40khz frequency
- The high frequency signal will be passed through the power amplifier for power amplification such that the square wave signal will be amplified significantly
- The amplified signal will be tuned such that the 40khz signal will retrieved back using parametric array
- The tuning process is done by using inductor and internal capacitance the parametric array.

5.3 APPLICATIONS

1. Digital signage

Today, an effective digital signage presentation includes both video and audio. But once sound is introduced in a public environment problem are sure to follow: blaring audio will quickly get your sound turned off and probably result in your entire display getting unplugged by nearby employees. Not surprisingly, many forego the audio component completely. Our Sonic Beam directional speakers with Smart Volume incorporate the technology necessary to ensure a successful isolated audio presentation for your digital signage applications.

Fig 5.10

2. Museum

In a museum setting, audio isolation is of paramount importance. When using standard speakers, reflective surfaces bounce sound everywhere. The solution? Our complete range of directional speakers from flush mountable models for discreet audio delivery to custom solutions – keep audio right where you want it even in the most sound-reflective environments.

Fig 5.11

3. Restaurant

Want a unique listening experience for each party seated in your restaurant? Our Quadeo Sound Dome speaker mounts above a table and directs sound to just those seated around the table, keeping audio so focused it cannot be heard by those at the next table. Use our SB-47 Sonic beam directional speaker to focus sound just to those sitting in the bar area. Patrons can listen to the game leaving others undisturbed to enjoy their meal.

Fig 5.12

4. Video conferencing

Our videoconferencing technology allows for private conversation in public environments. A special microphone allows users to keep their voice at a whisper, while audio precisely focused to their ears cannot be heard by others steps away. From our Vocalizer product to custom group-conferencing systems, like the MIT - Stanford WormHole, we offer videoconferencing solutions for any environment, no matter how public or intimate.

Fig 5.13

5.4 Limitations

- Yoneyama presented the implementation of PAA which used the amplitude modulation (also known as double sideband amplitude modulation) technique where the 40 kHz sinusoidal carrier was used, which lies far beyond the human hearing range but introduces a high level of total harmonic distortion (THD).
- In DSBAM the high frequency carrier interacts with upper and lower side-band components in a nonlinear manner. In order to reduce the level of THD Kamakura et al. presented the square-root AM (SRAM) method. Hitherto, from all known modulations, the single sideband amplitude modulation (SSB-AM is characterized by the lowest level of THD and the highest power efficiency). However, the main drawback of SSB-AM is related to the envelope errors in case where the emitted signal consists of a broadband signal (such as speech or music) which modulates the carrier. To overcome this issue, Croft et al. proposed application of the recursive SSB-AM (RSSB-AM).
- The directivity of all traditional loudspeaker devices is fundamentally limited by nothing more than the size of the source compared to the wavelengths it is generating.
- A large loudspeaker will be more directive than a small loudspeaker, or a loudspeaker specified at higher frequency (smaller wavelength) will also have more directivity. No amount of phasing, shading, focusing, or other method can overcome this fundamental limit; in fact, any of these methods will always reduce directivity.
- Very less sound, upto 10cm for 50 ultrasonic transducers. Ultrasonic transducers are made from piezoelectric ceramic materials. When a voltage is applied to the device, a special type of foil is deformed inside, generating a supersonic sound wave of a specific frequency. Typically, the transducer’s sound output reaches 105–120 dB (at 30 cm distance) when a voltage of 10–20 Vrms is applied.

CHAPTER 6

RESULTS AND DISCUSSIONS

INTRODUCTION TO IMPLEMENTATION OF PROBLEM

Fig 6.1

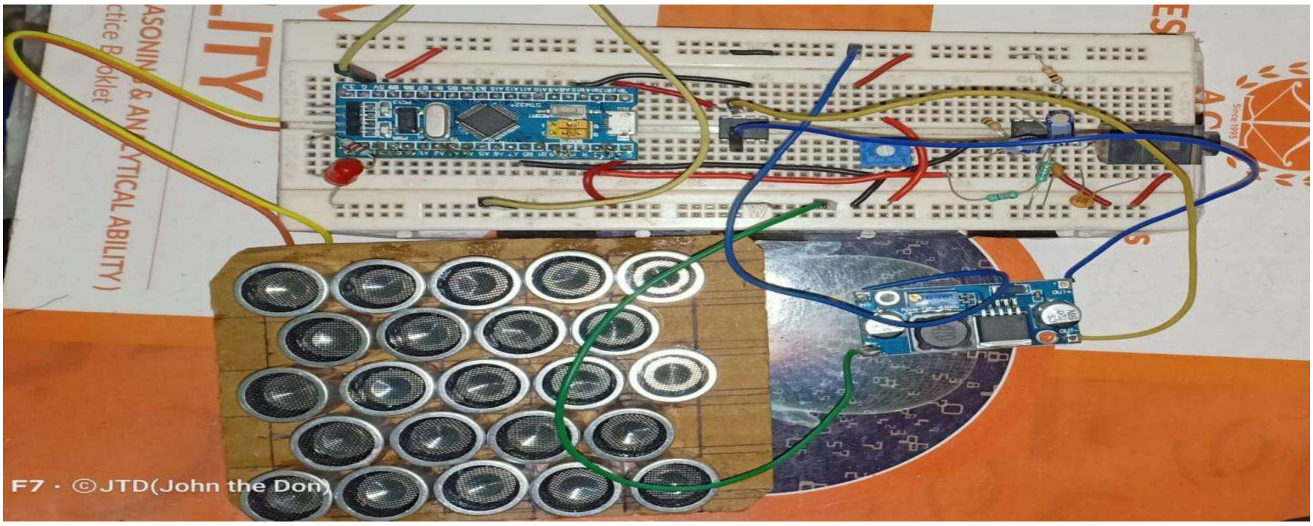


Fig 6.2

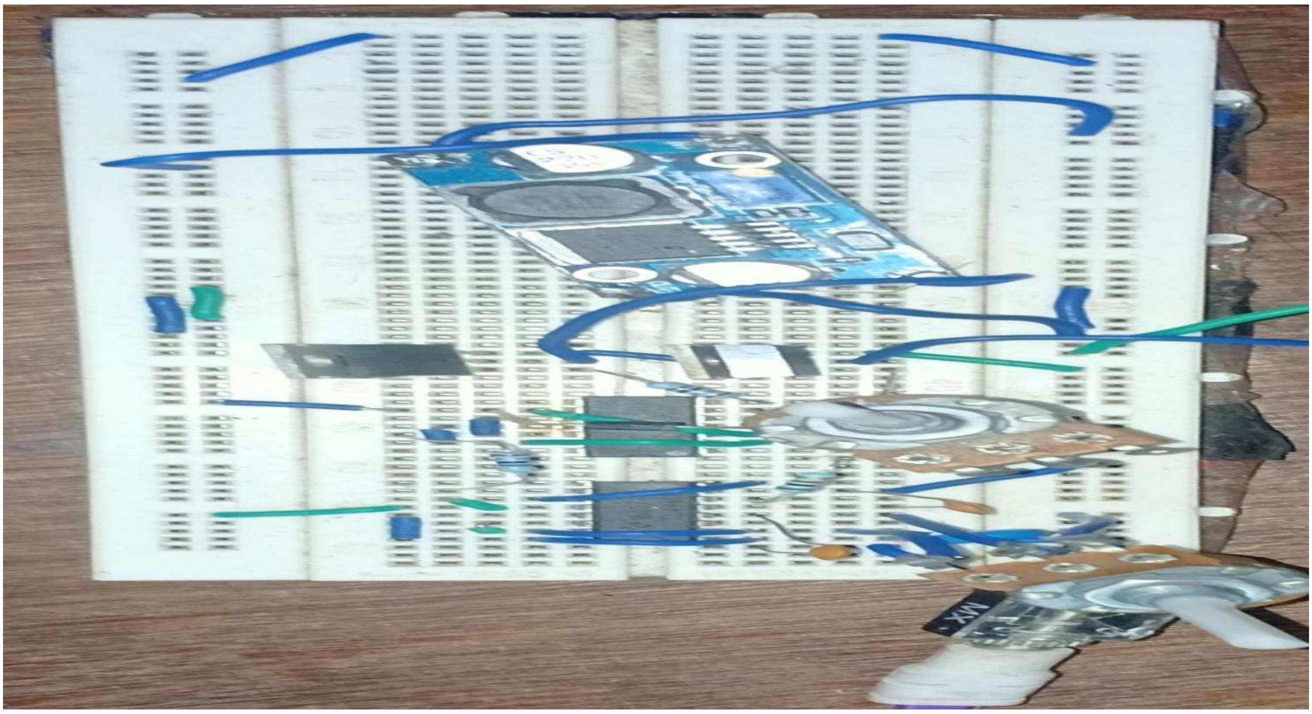


Fig 6.3

Circuit Implementation:

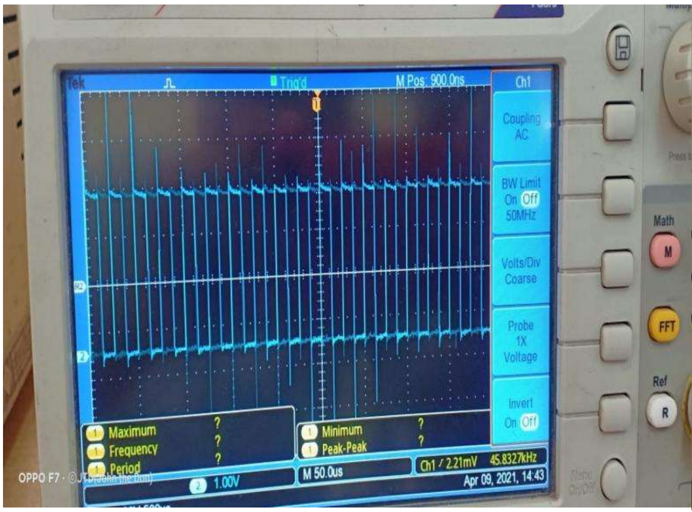
- ❖ Circuit implemented on breadboard after tested in simulation using multisim software.
- ❖ The directivity is tested on matlab using phased array toolbox which gives Elevation and Power graphs for a given input signal.

Problems Faced while Implementing Hardware Segment:

- ➢ The main problem faced while implementing circuit is output power. Due to high impedance of the Parametric array loss the total power signal at the input
- ➢ This can be overcome by using tuning process with a variable inductor and considering internal capacitance of parametric array.

- Even for 25 ultrasonic transducer the directivity and output power is very poor. Excellent circuit design and power implementation gives the desired output.
- Ultrasonic transducers are made from piezoelectric ceramic materials. When a voltage is applied to the device, a special type of foil is deformed inside, generating a supersonic sound wave of a specific frequency.
- Typically, the transducer’s sound output reaches 105–120 dB (at 30 cm distance) when a voltage of 10–20 Vrms is applied,

6.1 RESULTS



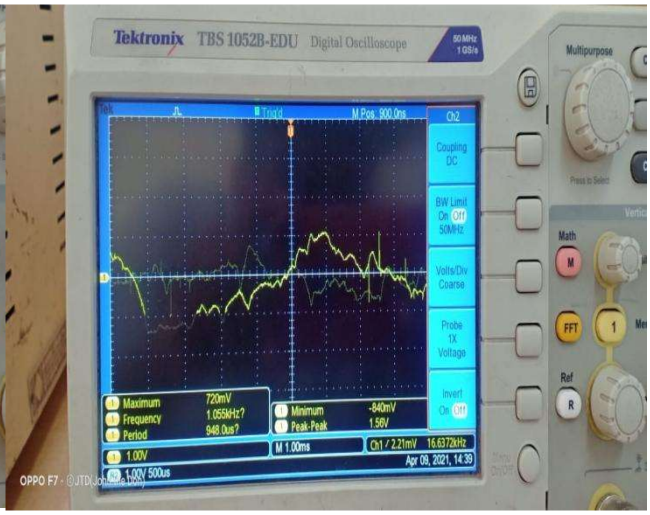


Fig 6.4

Fig 6.5

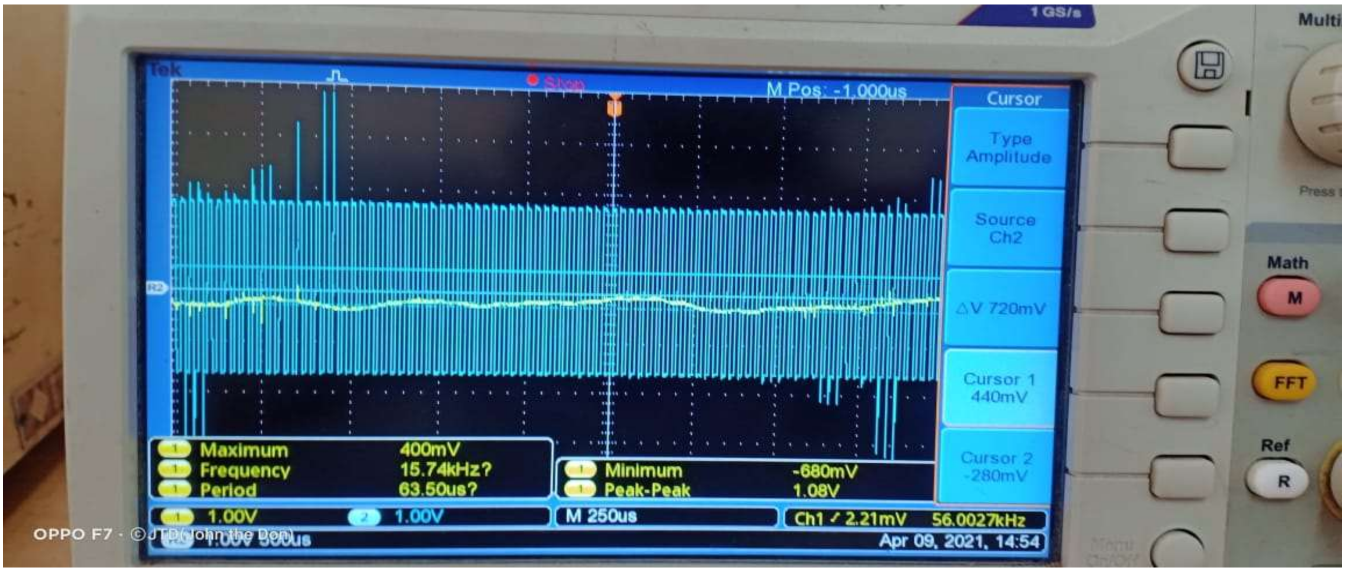


Fig 6.6

6.2 CONCLUSION:

- The directive sound system has made the sound directional; using this can produce hypersonic sound waves. This method can embed in Heavy Hypersonic Acoustic Systems (HHAA).

- The directivity of HHAA increases when we increase the number of ultrasonic transducers. If the number of transducers is multiplied to 50, it increases the directivity up to 110 dB. HHAA can be used in public areas and parties. But the problem is reflections and their power.

- To reduce reflected waves from the speaker, employing a specific bounded setup makes the reflections confined, and strength will be reduced when using acoustic foam. The simulation measured the required parameters and produced graphs by recording audio from the directive beam speaker.

- Simulation results show the directed waves that strike the acoustic foam reduce the power significantly, and it is confined into a setup. It differentiated between audio files taken directly from the directive array, reflected from the wall, and reflected from the acoustic foam.

- The results show that we can constrain the sound using acoustic foam as a barrier in a particular space without disturbing the surroundings. In the future, our goal is to make a real-life Heavy Hyper Sonic Acoustic Setup that will freely work in public areas.

REFERENCES:

[1]Copeland, Darren. “The Audio Spotlight in Electroacoustic Performance Spatialization.” eContact! 14.4 — TES 2011: Toronto Electroacoustic Symposium / Symposium électroacoustique de Toronto (March 2013). Montréal: CEC.

[2] M.F. Hamilton, D.T. Blackstock, Nonlinear Acoustics, Academic Press, San Diego 1998,Chap. 3.

[3] W. Gan, J. Yang, T. Kamakura, A review of parametric acoustic array in air, J. Applied Acoust 73(12), 1211-1219 (2012).

[4] https://en.wikipedia.org/wiki/Sound

[5] https://en.wikipedia.org/wiki/Nonlinear_acoustics

[6] DEVELOPMENT OF A DIRECTIONAL LOUDSPEAKER SYSTEM FOR SOUND REPRODUCTION Peifeng Ji, Chao Ye, and Jing Tian The Institute of Acoustics, Chinese Academy of Sciences, Beijing, China 100080

[7] Jacqueline Naze, Tjotta; Tjotta, Sigve (1981). "Nonlinear equations of acoustics, with application to parametric acoustic arrays". *Journal of the Acoustical Society of America*. 69 (6): 1644–1652. Bibcode:1981ASAJ...69.1644T. doi:10.1121/1.385942.]

[8] P.J. Westervelt, Parametric Acoustic Array, J. Acoust. Soc. 35(4), 535-537 (1963).

[9] https://en.wikipedia.org/wiki/Sound_from_ultrasound

[10] https://www.nti-audio.com/en/applications/room-building-acoustics/reverberation-time-rt60-measurement

[11] https://en.wikipedia.org/wiki/Soundproofing#:~:text=Sound%20absorbing%20material%20controls%20reverberant,a%20cavity%2C%20enclosure%20or%20room.&text=Both%20fibrous%20and%20porous%20absorption,a%20room%2C%20improving%20speech%20intelligibility.

[12] Datasheet of MCUST16A40S12RO

[13] Assessment of noise pollution in and around a sensitive zone in North India and its non-auditory impacts Author links open overlay panelKhaiwalRavindraaTanbirSinghbJayaPrasadTripathyacSumanMorbdSanjayMunjaleBinodPatrofNareshPandae

[14] https://en.wikipedia.org/wiki/Loudspeaker_measurement#:~:text=While%20the%20very%20best%20modern,are%20within%20%C2%B112%20dB.