Elephant Detection and Localization Using Infrasound

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December 2016

Submitted in partial fulfillment of the requirements of the B. Sc in Computer Science 4th year individual project

Declaration

I certify that this dissertation does not incorporate, without acknowledgement, any material previously submitted for a degree or diploma in any university and to the best of my knowledge and belief, it does not contain any material previously published or written by another person or myself except where due reference is made in the text. I also hereby give consent for my dissertation, if accepted, be made available for photocopying and for interlibrary loans, and for the title and abstract to be made available to outside organizations.

Preface

This document is the thesis for the 4th Year Individual Project for partial fulfillment of the requirements of the Degree of Bachelor of Science in Computer Science, at University of Colombo School of Computing, University of Colombo, Sri Lanka.

The intended audience of this document is the academic staff of the University Of Colombo School Of Computing and it is intended to enable them to determine whether the project should be approved as proposed, approved with modifications, or not approved.

Contents

1 Introduction.

1.1 Goal and objectives.

The world elephant population has been on the decline [13] due to many reasons, among which the human elephant conflict is a major cause. Human settlements and cultivations adjoining the forest areas have resulted in the blocking of elephant migration routes and further the presence of crops attracts wild elephants, causing damages to livelihood of humans while threatening the lives of both elephants and humans. The wildlife conservation authorities worldwide do not possess an established method to manage this situation which is non-destructive to both elephants and humans, with most authorities having to resort to brute force, often consequently aggravating the situation in the long term [13]. At present, the primary solution introduced is the use of electric fences around elephant habitats to prevent elephants venturing beyond their habitat to encroach into human settlements; an expensive and potentially life threatening solution.

The objective of this research is to implement a cost effective input to a larger system that will help to solve the human elephant conflict building on and expanding upon the previous findings of related research. Research to date has found that elephants pass various messages using infra sound frequencies and this low frequency sound waves travel a greater distance than higher frequency waves due to high frequency waves being more easily absorbed by air molecules compared to the lower frequency waves [5]. In this research, an electronic system consisting of low cost sensors that have the capability of detecting infrasound calls emitted by the elephants as well as digital signal processing techniques are combined to identify elephant infrasonic vocalizations to localize the sound emitting sources. Further, attempts are made to use these information in various scenarios such as prior warning system before elephants enter a cultivation and elephant herd detection among other things.

1.2 Human elephant conflict

This includes brief intro on HEC.

1.3 Research question.

This research attempt to develop a method to distinguish an elephant call in a stream of sound data and to find an effective method of infrasound source localization using the phase difference of two infrasound waves captured by several sensors placed in different distances from the sound source. The research aims to answer two basic questions using a low cost sensor system consisting of off the shelf microphones capable of capturing infrasound:

1. How to identify an elephant call in an infrasound wave ?
2. How to localize a source emitting infrasound in a noisy environment ?

1.4 Background and Significance.

A typical human male voice in speech fluctuates around 110 Hertz, a female's voice at around 220 Hz and a child's at around 300 Hz. Among elephants, a typical male rumble fluctuates around a minimum of 12 Hz (more than 3 octaves below a man's voice), a female's rumble at around 13 Hz and a calf's around 22 Hz [1] [2]. In Asian elephants, this value fluctuates between 14 Hz to 24 Hz within 10 to 25 seconds [3] due to their smaller vocal cords compared to African elephants. Elephants produce a wide range of sounds from very low frequency rumbles to higher frequency snorts, barks, roars, cries as well as many other type of calls.

Audio waves below 20 Hz frequency is considered as Infrasonic waves [4]. As such, elephant rumbles can be considered as infrasonic waves and these rumbles follow all the properties of infrasonic waves. A significance of infrasonic waves is that it travels further than high frequency waves. Sound is a pressure wave vibration of molecules and as a result, whenever molecules move, there is an inevitable loss of energy as heat. As a result, sound is lost by heating the medium through which it propagates. Sound wave attenuation is frequency dependent in most media.

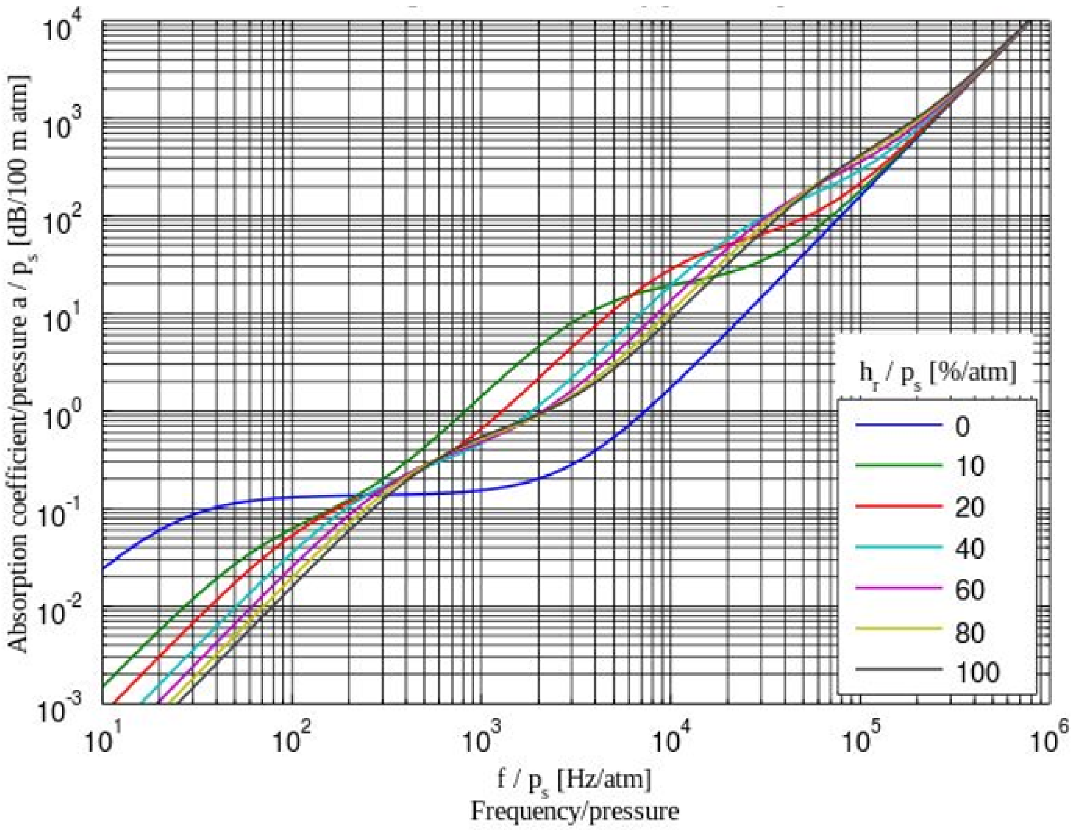


Figure 1.1: Sound absorption coefficient per atmosphere for air at 20 degree Celsius according to relative humidity per atmosphere.[14]

The above image 1.1 is a graph displaying the attenuation of sound at difference frequencies [5], which shows that low frequency waves have low absorption coefficient. Therefore, low frequencies are not absorbed well and travel further than high frequencies. This property of infrasound waves can be used for the acoustic detection of the wave from a greater distance from the source of the sound. The low-frequency sounds used by elephants for long-range communication travel a distance that exceeds 1 km [3]. As mentioned in the objectives, this research focusses on detecting elephant infrasonic calls from long distance and localizing them. Detecting and localizing elephants is an essential component of any viable solution to human elephant conflict. Attempts towards acoustic detection and localization of elephants exist in literature. However, due to high cost in infrasonic sensors and complexity in detection of these signals in noisy environment, no system exists to date, that operation ready in the field.

Various types of devices which are capable of capturing infrasonic waves exist and they were mostly used for detecting geographical phenomena such as earthquakes and volcanic eruptions. One such device used by most related research is Infitec Infra 20, which has a sampling rate of 50Hz. This device can be considered as a low cost infra sound recorder compared to other existing devices and this costs US$ 345 [7]. A significance of the research is the attempt to use a laboratory made sensor system which cost only US$ 15, which is capable of recording at a sampling rate of 44100 Hz. This sensor system consists of a Panasonic Omnidirectional Back Electret Condenser Microphone [15] (Model WM-64CWM-64K), LM358 IC for amplifying and a low pass filter[16].This sensor has the ability to capture infrasonic waves from a lower bound of 0 Hz to an approximate upper bound of 150 Hz.

The second research question is based on infrasound localization, which can be done by measuring physical quantities like sound pressure and particle velocity. The best localization technique which is used by nature (Animal sound localization) can be applied to this context in localizing a sound source digitally. Humans as well as most other land-living vertebrates use the time delay between the arrival of a sound wave at each ear to discern the direction of the source [8]. Similarly, localization can be done digitally by calculating the inter-aural time delay between two microphones lying on a specified distance, which will be further explained in the research methodology. Potential problems include the detection in noisy environment, sparsity and irregularity of elephant call and pattern recognition. These problems will be addressed using advanced digital signal processing techniques and the knowledge based on the existing literature.

1.5 Scope of the thesis.

As the final outcome, my intention is to introduce a low cost elephant detecting and localization system specialized for Asian countries using the sensor equipments made in the Sustainable Computing Research Group at University of Colombo School of Computing.

This research attempts to:

- Identify an elephant call in the infrasound range with low latency.

- Localizing an elephant call using the low cost sensors system consisting of a condenser microphone.

2 Review of Literature.

2.1 Elephant communication

Various research types exist in this specific domain. The experiments carried out by Katharine Payne, a researcher in the Bio acoustics Research Program at the Laboratory of Ornithology at Cornell University, show that elephants use infrasound in communication, which can be considered as the initial steps of all research work on this area. In 1984, she discovered that elephants communicate in low frequencies during her research carried out at the Portland Zoo. Further, her work with William Langbauer, Jr. and Elizabeth Thomas have shown that elephants were indeed making infrasonic calls. Subsequent studies, in association with Joyce Poole, William Langbauer, Cynthia Moss, Russell Charif, Rowan Martin and others, took place in Kenya, Namibia, and Zimbabwe, leading to the conclusion that elephants use their powerful deep calls in long distance communication [6] and elephants make these calls when coordinating family and larger group behaviors, when competing for resources and dominance, as well as when attracting mates and announcing reproduction. Large vocal cords of elephants were able to produce low frequency sound signals considered as rumbles, which were able to travel around 5 km in distance [6]. It is also revealed that the rumbles audible to human ears, are the harmonic waves created from the infrasonic fundamentals.

There has been comparatively less study of communication in Asian elephants. Acoustic communication in the Asian elephant by Dr. Shermin de Silva, a James Smithson Fellow at Smithsonian Conservation Biology Institute, can be considered as a comprehensive study on communication in Asian elephants, which was published in the Behaviour biological journal in 2010. She categorized acoustic features into 8 'single' calls, 5 'combination' calls and one possibly unique male call, for a total of at least 14 distinct call types [19]. Her observations and conclusions are based on the data collected at Udawalawa national park, Sri Lanka during 2007 and 2008. It is mentioned that 7 out of 14 distinct call types (rumble, rev, roar, cry, bark, grunt, husky-cry) are made out of elephants Larynx and most of the fundamentals of these calls were found to be infrasonic. While African elephants were the main subjects of considerable amount of existing literature with majority of these focusing on elephant detection in noisy environment; the human elephant conflict is more of a burning issue in South Asian countries like India and Sri Lanka.As a result, in 2002, there was an attempt at implementing a sensor system that detects infrasonic calls of elephants in Sri Lanka.

Elephant infrasound have not been recorded in wild Asian elephants anywhere in Asia prior to this research project in Sri Lanka in 2002. The prototype introduced by the above research was able to supports four infrasound sensors and is capable of standard DSP functions such as archiving and filtering. Sound detection has a long history although it was not specified to elephant infrasound calls. Recent researches have shown that this is possible using a template based or feature based technique. Matched filter method where two spectrograms of the pattern template and the signal are directly mapped, is a straightforward mechanism for detecting a pattern in a signal. Although it was optimal when finding the occurrence of a template in a recorded signal, it is sub optimal in the presence of complex noise [10]. Therefore, a novel spectro-temporal method for signal enhancement based on the structure tensor [11] was introduced by a group of researchers at University of Vienna, Austria.

2.2 Sound localization

There are many works related to sound localization using microphones. Many techniques and algorithms have been introduced during past decades to detect the direction of sound emitting source. These are mainly based on the following three type of principles.

1. Time difference of arrival, where the systems measure the difference in time between the signals received by the microphones to localize the sound source.

2. Direction of arrival, where the phase difference between the signals is used to locate the sound source [20].

3. Energy based sound localization, where the energy of sound wave decreases when the sound wave propagates in the air. By measuring the sound energy at different sensor locations, one may localize the sound source [21]

The most significant techniques is the time delay estimation, due to its simplicity and accuracy [22]. Several research works have compared the algorithms such as cross correlation method [23], phase transform [24] and maximum likelihood estimator [25] used to estimate the time delay. There are instances where they have used two sensors or array of sensors for the localization, where; when number of sensors increases, the mean error generated becomes relatively low [26].

Results of the experiments in the form of simulation results [26] show that all methods were able to estimate the time delay, where the peak position indicates the time delay estimation. However, the phase transformation method (PHAT) achieved a sharper peak than the other methods, which helps to estimate the real delay time more accurately in the real situation. The maximum likelihood method also achieved a sharp peak. Although cross correlation has the widest peak, it still can estimate the real time delay in the simulation conditions. Since these works can be directly incorporated to elephant call localization, we can guarantee that the accuracy of the localization results can be increased. As a result, we are able to select and use the most convenient and easily implemented method for the localization experiments. The environment of the localization scenario can result in the lagging of sound waves. As elephant localization is done in a forest environment, this is a factor that should be taken in to account. Many works have been done regarding the localization of sound in reverberation environment. The precedence effect describes the phenomenon, whereby echoes are spatially fused to the location of an initial sound, by selectively suppressing the directional information of lagging sounds (echo suppression). Echo suppression is a prerequisite for faithful sound localization in natural environments but its reliability depends on the behavioral context [27]. These works can be integrated with our findings to be produce an accurate infrasonic localization.

2.3 Acoustic detection of elephants

Although different literature exists with regard to acoustic detection of elephants and infrasound waves, there is no significant attempt towards an economically feasible solution for localizing Asian elephants through the use of infrasound calls, applicable to developing countries like Sri Lanka and India. The review will emphasize the importance towards a research on addressing the above problem.

2.4 Signal classification

Related works on :

- Biological researches on elephant communication.
- Behavior of infra sound waves.
- Sound localization.
- Signal classification.
- Acoustic detection of elephants.
- Infra sound recording devices

3 Design and Methodology.

3.1 Introduction

As mentioned in the background, this research will not produce an ultimate machinery to detect and localize elephants. Rather, it will be an input to a system of this calibre, which will increase the accuracy and also help validate the results of such a system. The first phase of the research will mainly focus on the infra sound localization from a relatively large distance.

Localization will be done using cross correlation. Two infrasonic sensors, 3m away from each other, are placed at different location away from an infra sound emitting source, which is a full range speaker with a larger membrane, capable of emitting low frequency sounds connected to a low frequency amplifier. A recorded sound clip of 15 seconds will be input to an automatic angle calculator program. Abstract overview of the above mentioned program is shown in figure 3.1. Before localization is tested for elephant sounds in forest environments, this will be tested at university premises taking recordings at different places. Main concern of this works will be identifying the factors responsible for the generated error. Results of the preliminary experiments regarding the localization are included in section 9.

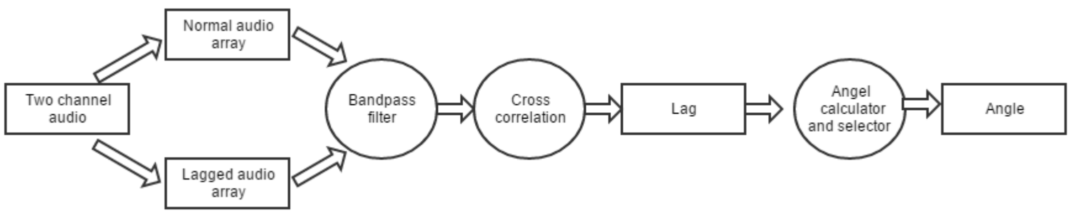


Figure 3.1: Overview of the cross correlation program

A high level overview of the integrated components is given in figure 3.2. An input from the sensor system (recorded signal) will be pre-processed in order to smooth-en the signal wave. The sample rate of the recorded wave will be 44100. Feature extraction is performed to feed this data to the machine learning module. The purpose of this is to be done using MFCC (Mel-Frequency Cepstral coefficients) at the initial stage and Green-wood feature extraction in the secondary stage.

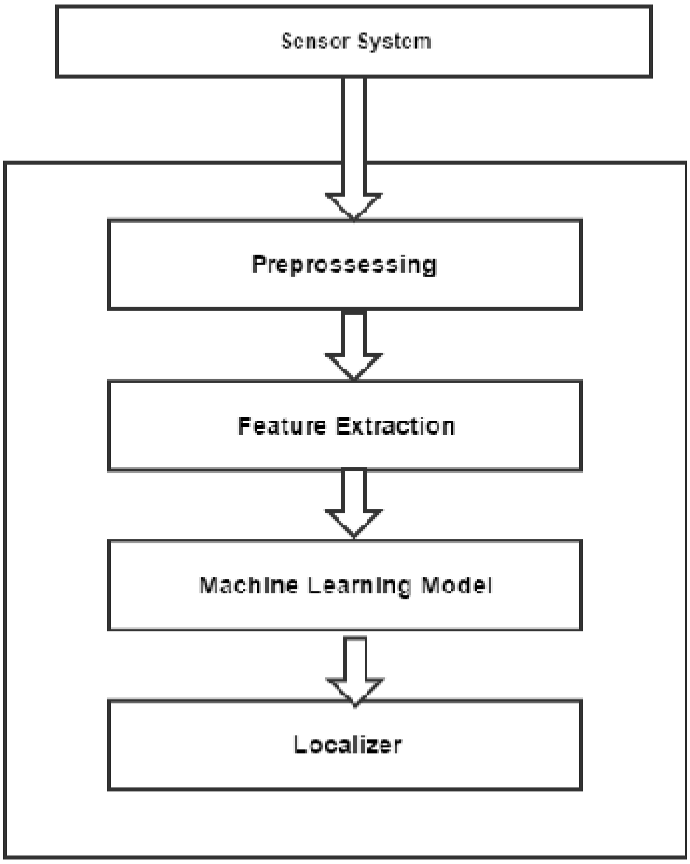


Figure 3.2: High level overview of the integrated components

An appropriate machine learning model will be selected based on the conclusion of the existing work and using the data sets recorded at different forest areas in Sri Lanka as well as the data sets received from the Smithsonian biology conservation institute (SBCI), which will be used to train this machine learning model. There will be a comparison between the signal properties of the recorded audios by SBCI and our recordings using sensors made by us at the laboratory. In the next phase of the research, there will be an integration of the localization work and the pattern recognition work.

Since a main objective of this research is to produce a low cost sustainable solution, for elephant localization and detection, the quality of the results obtained using laboratory made components will also be compared against the high cost devices currently available.

3.2 Infiltec INFRA-20

Among the infrasound recording devices available in the market Infiltec Infra 20 is widely used in academic researches.

Performance: Resolution is 0.001 Pascals (0.01 microbar or 0.0075 millitorr) over the range of + or - 25 Pascals (250 microbars or 187.5 millitorr). Mode of Operation: Microbarograph design with solid-state differential pressure sensor and high-pass pneumatic filter. Output: Serial output formatted for AmaSeis seismic software, RS232 9600 baud N81 ASCII, 50 samples/second. Frequency Pass-Band: 0.05 Hz to 20.0 Hz bandwidth digitized and stored on hard drive by AmaSeis at about 10 megabytes per 24 hours of operation. Temperature Range 40 to 90 F (4 to 32 C) with no adjustments required. Internal firware automatically adjusts for drifts. Requires constant temperature during operation that can be provided by insulation such as a small cooler. Housing: Diecast aluminum box encloses all electronics and sensor, 6.024x3.268x1.988 inches (153x83x50.5 mm). Shipping weight 2.2 lb (1 kg). Computer Connection: 15 ft (4M) serial cable (up to 100 ft (30M) extension optional) connects to PC DB9 connector or USB-Serial DB9 adapter. Adjustments: No leveling, alignment, damping, gain, zeroing, etc adjustments required. Power Supply: No external power supply required, all power (under 0.05 Watts) is supplied by PC serial port through the serial cable connected to the PC. Digital Processing: 16 bits voltage resolution from internal digitizer, 50 sample/second ASCII output, firmware adjusts for component drifts. Lowpass Filter: Very steep analog 8 Pole elliptic filter with 20 Hz corner frequency for anti-aliasing. Software: Windows PC based seismic data logging and analysis software AmaSeis available online for free download. Includes spectral analysis and bandpass filtering functions. Computer Requirements:PC running Windows (98/XP/V/7/8/8.1/10 32 or 64) can run AmaSeis in backgound while doing other tasks. Hard drive that can store 10 megabyte/day during 24/7 operation. DB9 serial port or USB port for supplied USB-Serial adapter.

The INFRA20 electronic noise level is about 20 counts (20 mPa or 60 dB SPL) over the full bandwidth, and this can be measured by cross connecting the ports on the internal differential pressure sensor. The lowest ambient infrasound level over the full bandwidth that you can expect to record with the INFRA20 is about 30 counts (30 mPa or 63.5 dB

SPL), typically during the middle of the night when the wind is calm and there are no nearby infrasound sources. The INFRA20 background noise level is low enough so that the microbarom peak at about 0.2 Hz can sometimes be detected above the ambient noise when the wind is calm.

3.3 Introducing Elocate sensors

In order to localize elephants using infrasonic emissions, it is necessary to have cost effective infrasonic detector systems. Most importantly, since we are deploying them in rural areas of Country in large numbers, they need to have a cheap unit cost. Moreover, they should be able to run with minimum maintenance due to the unavailability of technical experts in potential deployment areas. The ability for a rural villager with minimum technical knowledge to set up and run such an elephant localization system in the neighborhood to protect their premises would be a great advantage. The system has to be non-invasive to the elephants to avoid legal barriers of deploying such a system by individuals living in affected areas.

To match our needs, we design and implement an infrasonic detector which we call Eloc node. The heart of an Eloc node is a Panasonic WM-61A omnidirectional back electret condenser microphone [15]. Compared to ordinary condenser microphones, this microphone consumes less electric current and is therefore suitable for using in low power devices such as battery-operated embedded systems. An Eloc node consists of such a microphone and a small preamplifier circuitry connected to it inside a sealed plastic container.

As shown in Figure , the pre amplifier consists of an operational amplifier in inverting mode and a low-pass filter. The low-pass filter attenuates frequencies above 150 Hz since we are interested only in infrasonic frequency components. Eloc nodes get power from a 9V battery attached to the plastic container. This complete unit draws a current less than 50mA from the battery when the microphone is active. Hence, Eloc nodes could be powered by solar cells to avoid the need of replacing batteries once deployed in the field. The output of the Eloc nodes is an analog signal that we can sample directly using a suitable analog-to-digital converter before sending it over a wireless network for processing in the back-end. For the experimental purposes, we connect it directly to the audio input port of a computer for processing on-site.

The microphone manufacturer has specified a sensitivity range from 20 Hz to 20 kHz. As mentioned in Section 2, acoustic calls of Asian elephants have, however, fundamental frequency components from 14 Hz to 24 Hz [14]. To evaluate the usability of the microphone for low audio frequencies, we conduct an experiment in which we play a chirp from 10 Hz to 100 Hz using a subwoofer. Our earlier work has shown [3] that subwoofers can replay elephant sounds that include fundamental frequency components in the infrasonic range with sufficient output power to emulate a real elephant. Figure 2 shows that the microphone is sensitive to much lower frequencies than 14 Hz and hence can be used for the task at hand. Meanwhile, Figure 3 illustrates the variation of sensitivity of an Eloc node with the distance to an infrasonic source. The figure shows that Eloc nodes outperform the expensive Infiltec Model INFRA-20 device at all distances.

Since we are interested in low frequencies, the noise imposed by the wind plays an important role. Initial experiments indicated that the Eloc nodes are unable to detect low frequency sources since they are submerged under the wind noise floor. Therefore we design and built a wind barrier around the microphones to protect them from wind noise as shown in Figure 4 a and b. This wind barrier consists of a wireframe with a soft material attached to it. An Eloc node is placed inside the wireframe using a vibration-proof mounting. We used different soft materials for the fur layer on the wireframe as well as several shapes for the wire frame to identify the most appropriate combination. The selected fur layer consists of a cheap artificial fur material we found off the shelf.

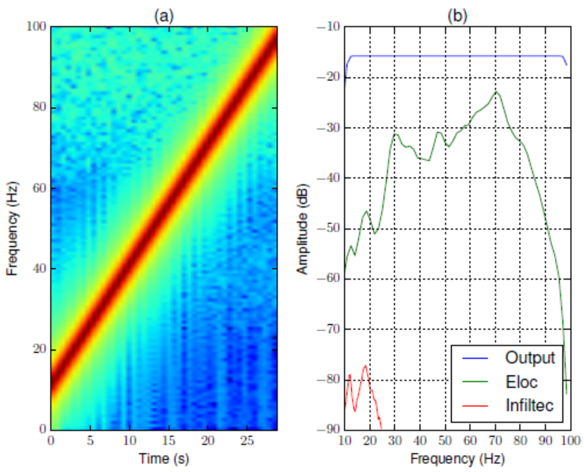


Figure 3.3: Sensitivity comparison of the microphones for different frequencies. The spectrogram (a) shows the played chirp from 10 Hz to 100 Hz and the graph (b) shows the received power by an Eloc node and an Infiltec device. Eloc node is sensitive even to much higher frequencies above the infrasonic range.

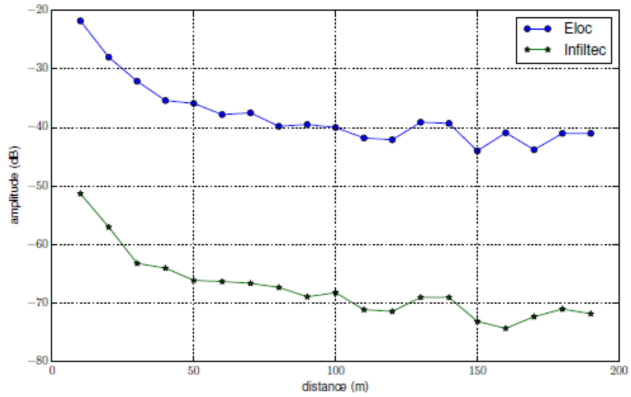


Figure 3.4: Sensitivity variation of the microphones with distance. The received power is measured for a 20 Hz tone for different distances from a subwoofer. Eloc node outperforms Infiltec device even in longer distances.

3.4 Other equipment

3.5 Localization

3.6 Feature extraction

3.7 Rumble detection

3.8 Signal enhancement

3.9 Classification using SVM

4 Implementation.

4.1 Electronic circuit of the sensors

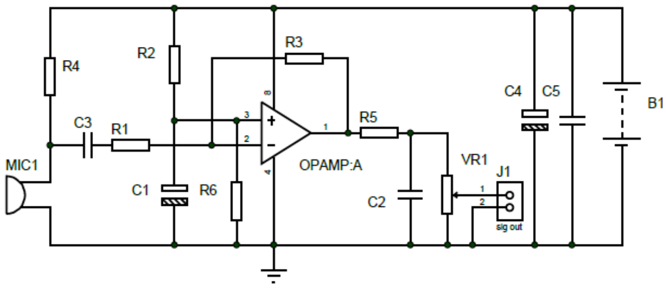
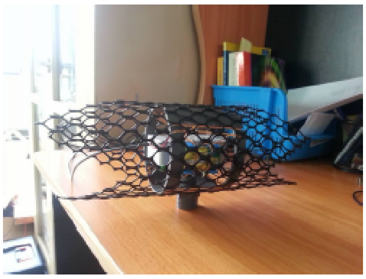
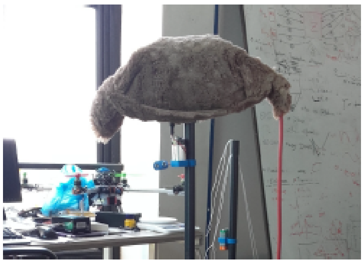


Figure 4.1: The pre amplifier schematic used in an Eloc node. It consists of an operational amplifier in inverting mode and a low-pass filter. The low-pass filter attenuates frequencies above 150 Hz.



(a)



(b)

(c)

(d)

Figure 4.2: (a) The microphone together with pre-amp circuitry is placed inside a sealed plastic box and then placed inside the wire-frame with shock mounting. (b) This wire-frame was covered with a soft material to cancel wind noise. In (c) and (d), two of these microphones are mounted in the front and back of a vehicle for field experiments.

4.2 Localization using cross correlation

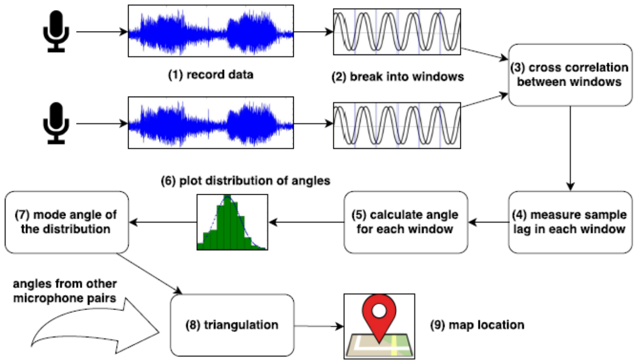


Figure 4.3: Process of calculating the location to an elephant. A pair of Eloc nodes capture elephant sounds which we break into windows, cross-correlate, calculated the sample lag and finally use it as an input for the angle calculation.

This was implemented in Python as well as in Matlab for different experiments. Following is the python implementation.

```
import numpy as np
from scipy.io import wavfile
import math
#this work only for 16bit wave format

def getAngle(fi):
    fs, data = wavfile.read(fi)
    data=data.astype('float64')
    data=data.T/32767
    s1=data[0]
    s2=data[1]
    limit=len(data[0])
    fs = 44100
    d = 3 #distance between two microphones
    vs=335.0
    start_window = 1
    end_window = 10000
```

```
    window = 10000
    total = 0
    arr=[]
    while(end_window<limit):
        s1=data[0][start_window:end_window]
        s2=data[1][start_window:end_window]
        xcor=np.correlate(s1,s2,'full')
        m=max(xcor)
        #print(xcor)
        im=np.argmax(xcor)
        start_window = start_window + window
        end_window = end_window + window
        deference = abs(im - window)
        #print deference
        #break
        #print(deference*vs/fs/d)
        ang = deference*vs/fs/d
        if(ang<=1):
            angle=np.arcsin(ang)
            arr.append(math.degrees(angle))
    plt.hist(arr)
    plt.show()

    return(arr)
```

Following is the Matlab implementation of the above implementation to find the angles.

```
function angle = get_angle_mode(name)
data = audioread(name);
array = [];
[limit,b] = size(data);
data = data';
```

```
fs = 44100;
d = 3;
vs = 335.0; %340.29; % 343.2 /
start_window = 1;
end_window = 10000;
window = 10000;
total = 0;
i = 0;
a=0;
while end_window < limit
    s1 = data(1,start_window:end_window);
    s2 = data(2,start_window:end_window);
    [m,im]=max(abs(xcorr(s1,s2)));
    start_window = start_window + window;
    end_window = end_window + window;
    deference = abs(im - window);
        break;
        ang=abs(deference*vs/fs/d);
        if ang<=1
        angle = asind(ang);
        array = [array,angle];
        end


end
%disp(array);
angle=array;
```

4.3 Feature extraction

Implementation of the feature extraction was done in Python. MFCC was used in the initial phase which is in Python speeches. Default frequency range of MFCC is shifted down where maximum frequency will be 300Hz which is suitable for elephant range.

```
from python_speech_features import mfcc
from python_speech_features import logfbank
import scipy.io.wavfile as wav




def extract_features(wavefile,winlen):
    (rate,sig) = wav.read(wavefile)
    data=sig.T
    #checking the audio is dual channel or single
    if(len(data)==2):
        mfcc_feat = mfcc(data[0],rate,winlen,winstep=1,highfreq=300)
    else:
        mfcc_feat = mfcc(data.T,rate,winlen,winstep=1,highfreq=300)

    return mfcc_feat
```

This will return a feature vector consist of 13 features which described in the methodology. Length of a window can be parsed according to different scenarios.

4.4 Spectrogram visualization

```
#!/usr/bin/env python
#coding: utf-8
import numpy as np
from matplotlib import pyplot as plt
import scipy.io.wavfile as wav
from numpy.lib import stride_tricks

""" short time fourier transform of audio signal """
def stft(sig, frameSize, overlapFac=0.5, window=np.hanning):
    win = window(frameSize)
    hopSize = int(frameSize - np.floor(overlapFac * frameSize))

    # zeros at beginning (thus center of 1st window should be for sa
    samples = np.append(np.zeros(np.floor(frameSize/2.0)), sig)
    # cols for windowing
    cols = np.ceil( (len(samples) - frameSize) / float(hopSize)) + 1
    # zeros at end (thus samples can be fully covered by frames)
    samples = np.append(samples, np.zeros(frameSize))

    frames = stride_tricks.as_strided(samples, shape=(cols, frameSiz
    frames *= win

    return np.fft.rfft(frames)

""" scale frequency axis logarithmically """
def logscale_spec(spec, sr=44100, factor=20.):
    timebins, freqbins = np.shape(spec)

    scale = np.linspace(0, 1, freqbins) ** factor
    scale *= (freqbins-1)/max(scale)
    scale = np.unique(np.round(scale))
```

```
    # create spectrogram with new freq bins
    newspec = np.complex128(np.zeros([timebins, len(scale)]))
    for i in range(0, len(scale)):
        if i == len(scale)-1:
            newspec[:,i] = np.sum(spec[:,scale[i]:], axis=1)
        else:
            newspec[:,i] = np.sum(spec[:,scale[i]:scale[i+1]], axis=

    # list center freq of bins
    allfreqs = np.abs(np.fft.fftfreq(freqbins*2, 1./sr)[:freqbins+1]
    freqs = []
    for i in range(0, len(scale)):
        if i == len(scale)-1:
            freqs += [np.mean(allfreqs[scale[i]:])]
        else:
            freqs += [np.mean(allfreqs[scale[i]:scale[i+1]])]

    return newspec, freqs

""" plot spectrogram"""
def plotstft(audiopath, binsize=2**10, plotpath=None, colormap="jet"
    samplerate, samples = wav.read(audiopath)
    s = stft(samples, binsize)

    sshow, freq = logscale_spec(s, factor=1.0, sr=samplerate)
    ims = 20.*np.log10(np.abs(sshow)/10e-6) # amplitude to decibel

    timebins, freqbins = np.shape(ims)

    plt.figure(figsize=(15, 7.5))
    plt.imshow(np.transpose(ims), origin="lower", aspect="auto", cma
```

```
    plt.colorbar()

    plt.xlabel("time (s)")
    plt.ylabel("frequency (hz)")
    plt.xlim([0, timebins-1])
    plt.ylim([0, freqbins])

    xlocs = np.float32(np.linspace(0, timebins-1, 5))
    plt.xticks(xlocs, ["%.02f" % l for l in ((xlocs*len(samples)/tim
    ylocs = np.int16(np.round(np.linspace(0, freqbins-1, 10)))
    plt.yticks(ylocs, ["%.02f" % freq[i] for i in ylocs])

    if plotpath:
        plt.savefig(plotpath, bbox_inches="tight")
    else:
        plt.show()

    plt.clf()

plotstft("16040201Mcropped.wav")
```

4.5 SVM model

```
from feature_extraction import *
from sklearn import svm
from os import listdir,getcwd

negative_files=listdir('./Negative')
positive_files=listdir('./Positive')

postive_features=[]
negative_features=[]
y=[]
for fil in positive_files:
    feature_vector=extract_features('Positive/'+fil,1)
    for f in feature_vector:
        postive_features.append(f)
        y.append(1)
for fil in negative_files:
    feature_vector=extract_features('Negative/'+fil,1)
    for f in feature_vector:
        negative_features.append(f)
        y.append(0)

x=postive_features+negative_features
clf = svm.SVC()
clf.fit(x, y)


#Saving the model
from sklearn.externals import joblib
joblib.dump(clf, 'Model/model.pkl')
```

4.6 Testing model

```
from sklearn.externals import joblib
from feature_extraction import *
from os import listdir

def test(ffile):
    clf = joblib.load('Model/model.pkl')
    feature_vector=extract_features(ffile,1)

    return clf.predict(feature_vector)



positive=listdir('Testing/Positive')
negative=listdir('Testing/Negative')



print "\n\nTesting positive set\n\n"
for i in positive:
    print test('Testing/Positive/'+i)
print "\n\nTesting negative set\n\n"
for i in negative:
    print test('Testing/Negative/'+i)
```

- Electronic circuit of the sensors.
- Noise reduction techniques.
- Implementation of localization.
- Data collection.
- Implementation of pre processing.
- Training SVM.
- Testing the model.

5 Results and Evaluation.

5.1 Introduction

Majority of the preliminary works are based on the correct localization of infrasound. Various experiments have been conducted at the University to achieve a better accuracy level of the localized angles using signal cross correlation technique. The method used for this experiment is explained in the methodology and design sections and only the result of these preliminary experiments are discussed in this section. The objectives of the following experiments are to calculate an angle of the direction of an infrasonic emitting sound source at different distances away from the source, estimate a formula for the error term, heuristically minimize the error by choosing a better sample rate and a statistical measure in calculating the angle. The distinction of these experiments is the use of a low-cost sensor made by us in the using off-the-shelf condenser microphones. This sensor system consist of two microphones with a band pass filter.

5.2 Localization experiment 01.

The objective of this experiment was to calibrate our sensors and angle calculating program. This was done in indoor premises placing the speaker 10m away from the sensors and placing the two sensors 3m away from each other. Several recordings were taken by changing the angle between the imaginary line joining two sensors of a infrasonic tone played at 20hz frequency.

Actuale Angle	**Calculated Angle**	**Difference**
0	**1.3058**	**1.3058**
30	**22.7937**	**7.2063**
45	**43.7307**	**1.2693**
60	**51.9516**	**8.0484**
90	**72.5378**	**17.4622**

5.3 Localization experiment 02.

After correcting some errors in the angle calculating program and modification of the gain in the sensors, the next experiment was conducted outdoors at the university grounds. Similar to the indoor experiment, a low frequency was played from a specific position of the ground and recordings were taken from different places of the university. Before calculating the angles, FFT was applied to the signals and frequency against sample count was plotted, to check whether the played tone is received by the microphones. FFT graph of some of the audio clips are shown below. The frequency of the played wave during the experiment was 20Hz.

These angles were calculated by considering several windows from the recorded audio and the mode of the angles calculated in different windows were considered in this initial experiment.

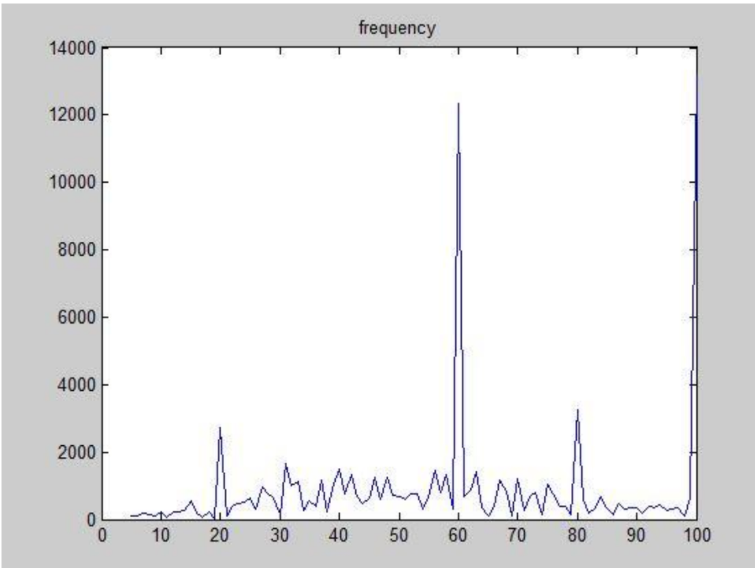


Figure 5.1: FFT at position 3

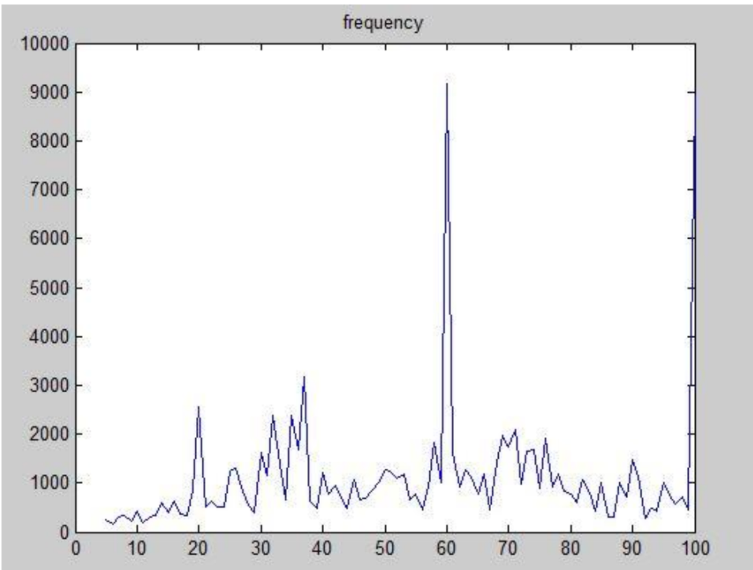


Figure 5.2: FFT at position 7

By looking at the above graphs, it is clear that the infrasonic waves reached two of the furthest recording places away from the speaker. Figure 5.3 show the locations of the places where recordings were done.

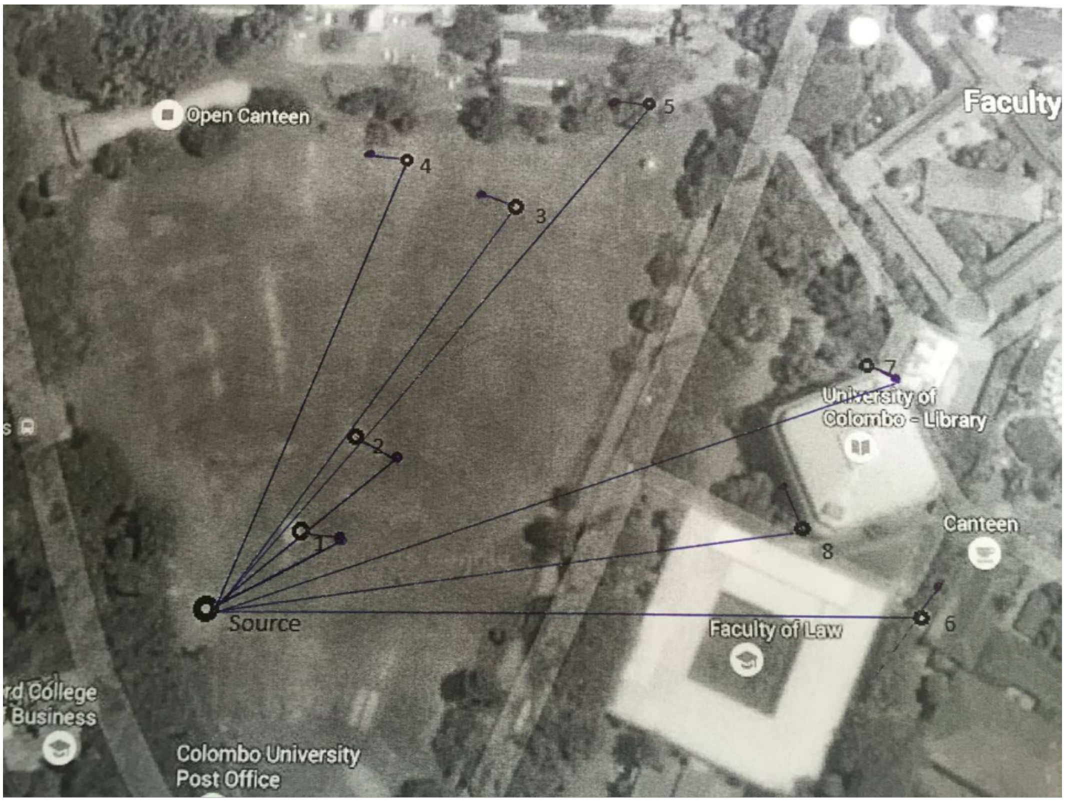


Figure 5.3: Recording locations

The estimated angle and actual angles of each positions can be found in the following table. Note that the actual angles are estimated geometrically using Google map.

Actuale Angle	**Calculated Angle**	**Difference**
30	**25.209**	**+4.791**
60	**59.574**	**+0.426**
50	**48.457**	**+1.543**
55	**59.909**	**-4.909**
45	**45.153**	**-0.153**
70	**65.692**	**+4.308**
30	**26.912**	**+3.088**

5.4 Localization experiment 03.

In this experiment, instead of taking out the mod value of the angles calculated in different windows, we plotted the histogram of the angle distribution. Similar to the above experiment, the sound source was placed at the same location and recordings were taken at different angles between the source and the sensors. This time, actual angles were calculated geometrically at the location using basic simple geometric construction theories. The following table shows the results related to the experiment.

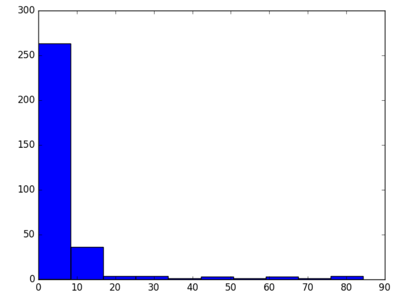


Figure 5.4: 0 degree histogram.

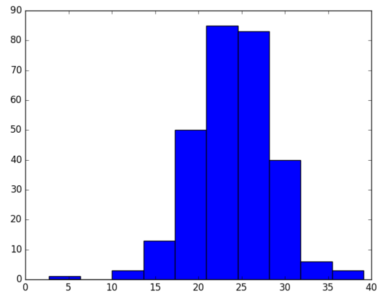


Figure 5.5: 20 degree histogram

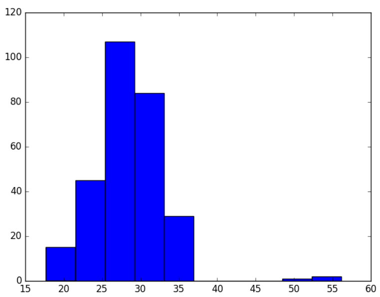


Figure 5.6: 30 degree histogram.

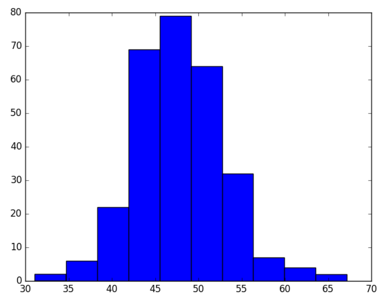


Figure 5.7: 50 degree histogram

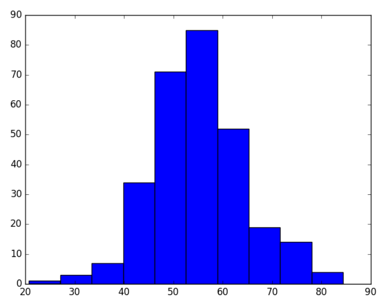


Figure 5.8: 60 degree histogram.

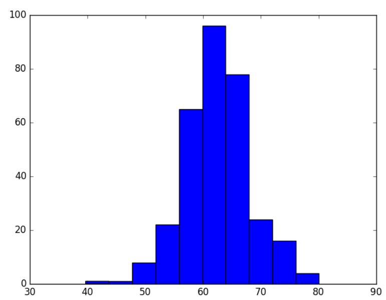


Figure 5.9: 70 degree histogram

Following table shows the number of windows have been produced in the correct range as well as the total number of windows.

Actuale Angle	**Calculated Range**	**Window count**	**Total window count**
0	**0-7.2**	**250**	**402**
10	**7.9-11.6**	**85**	**345**
20	**24.7-27.4**	**65**	**347**
30	**26.4-30.8**	**120**	**331**
40	**30.8-34.9**	**120**	**310**
50	**34-41.5**	**100**	**332**
60	**46-50**	**110**	**460**
70	**53-62**	**55**	**366**
80	**65-69**	**110**	**431**
90	**65-79**	**110**	**384**
90	**70-73**	**90**	**507**

We can observe that when angles get close to 90 degrees, the total error gradually increases. The next phase of the localization experiment is to heuristically find an angle with a minimum error or defining the factors affecting the error and using those factors building an equation for the angle error.

Apart from the localization experiments, there were few attempts to record vocals of the wild elephants in Sri Lanka at Kalawewa and Yala. During the visit to Kalawewa, the recordings were done in the presence of an elephant herd and a total hour of 8 was recorded. All visible and audible behavior of the elephants were recorded. Thereafter, these data were successfully backed up and logged in a digital spreadsheet.

During the visit to Yala, we were able to place the sensors, so that recorded data could also be used for the purpose of localization. There were few significant incidents during the visit and the analyzed results of a recorded audio at one such instance is shown in the following figure.

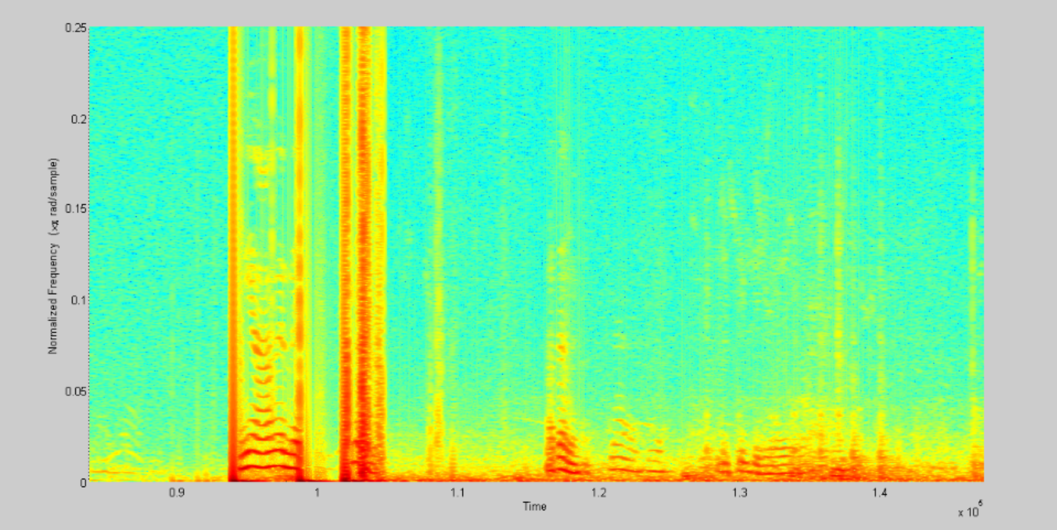


Figure 5.10: Spectogram of the signal under 44100 sample rate

The above image shows the spectrogram of a recorded clip at a sample rate of 44100 . The patterns of elephant rumbles are shown by the small red curves [19]. If the above clip re-sampled at sample rate of 250 , the infrasonic rumble patterns become visible.

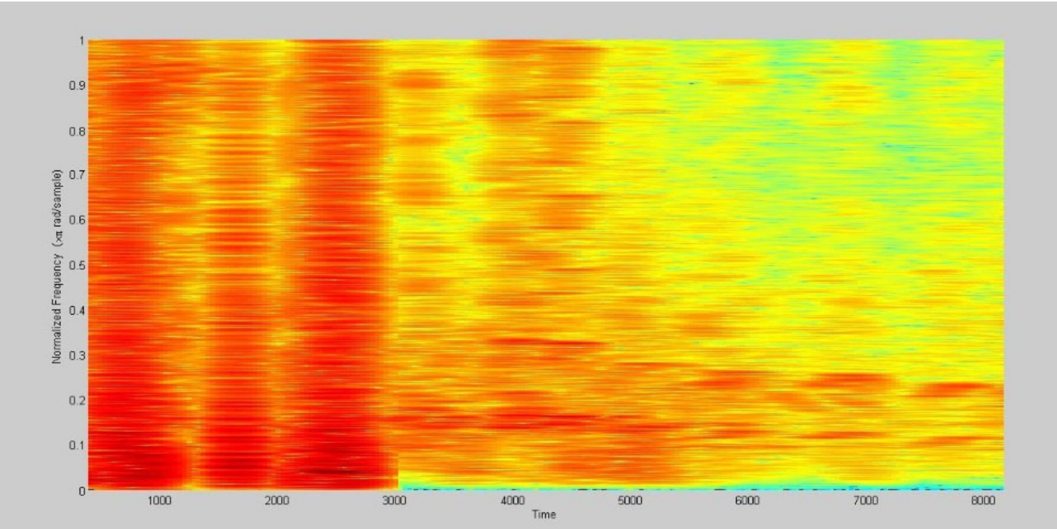


Figure 5.11: Spectogram of the signal under 200 sample rate

As this was recorded by the localizable configurations of the devices, this signal can also be analyzed to calculate angles of the elephant location. Some clips where rumbles are visible, were cropped using the spectrogram. These clips are cropped one after the other, so that it indicates the rumbles of the moving elephant herd. Estimated angle ranges from the plotted histograms are shown below.

Clip	**Angle range**
160402001Mcropped1.wav	**40-45**
160402001Mcropped2.wav	**47-50**
160402001Mcropped3.wav	**54-56**
160402001Mcropped4.wav	**55-57**
160402001Mcropped5.wav	**50-60 (a rather longer clip)**
160402001Mcropped6.wav	**55-60**

5.5 Error and correction of the angle

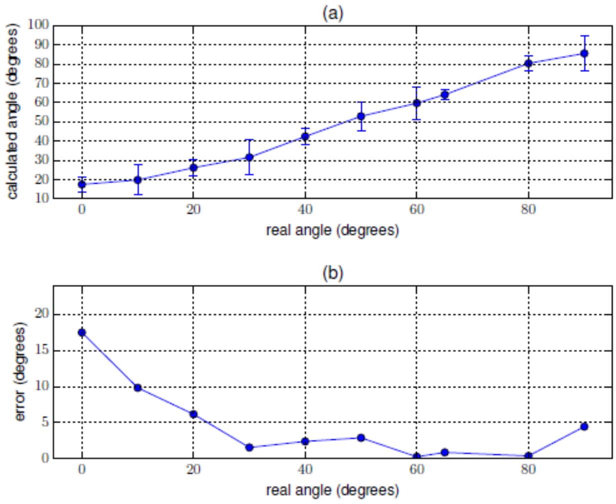


Figure 5.12: Accuracy of calculating the direction to an infrasonic source using a pair of Eloc nodes. The figures top part shows the relationship between real angles and respective calculated angles. The lower part shows the variation of the error in calculation against the real angle in consideration. For angles above 30 degrees the error is low.

5.6 Elephant Sound Recording in the Wild

The previous experiment confirms that Eloc nodes together with the wind shield can identify infrasonic frequencies inside the controlled environment of a laboratory even when there is wind. However, real deployments pose further challenges such as vehicle noise from nearby areas, vibrations that occur on the Eloc nodes due to the impact of dust and vegetation in the deployed location. Therefore, we evaluate the effectiveness of Eloc nodes to identify low frequency elephant sounds in the wild.

In order to capture real elephant sounds, we take a pair of Eloc nodes to a national park in Country where free ranging elephants live. We mount the nodes inside an offroad vehicle and park it in a location inside the national park where we can observe elephants visually while recording sounds. The vehicle engine is turned off during the recording time. Other external noise sources such as wind and vehicles in the nearby areas are present. We note the presence of elephants and their behavior in the visual range and compare them against the data we record. We take support of a zoologist to verify whether the recorded patterns are from an elephant

Figure 5.13 illustrates the waveform, frequency domain and the spectrogram of an elephant sound identified by a zoologist from our field recordings. As the figure shows, Eloc nodes can clearly capture the fundamental frequency component of the sound that is below the 25 Hz in addition to the higher frequency harmonics. However, due to the effect of the low-pass filter in Eloc nodes, the frequencies above 150 Hz are significantly attenuated.

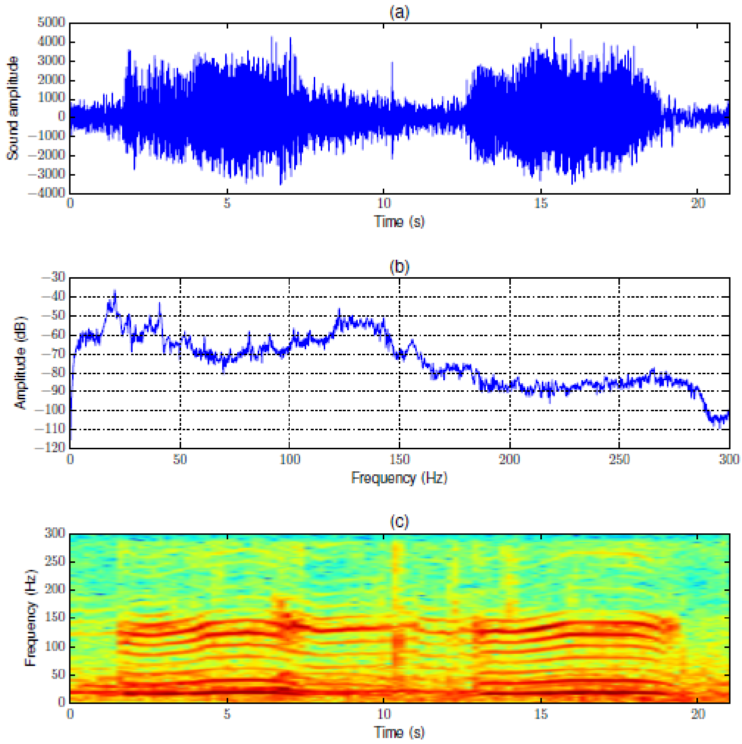


Figure 5.13: An excerpt from the elephant sound recordings using a pair of Eloc nodes. The graph (a) shows the waveform of the signal while the graph (b) shows its frequency domain. Graph (c) illustrates the spectrogram of the signal where a number of higher harmonics of the fundamental frequency of the elephant sound is visible. (The waveform shown in graph (a) has the amplitude as a scalar value between +215 and +215 since we store the recorded data in wave files which uses 16-bit Pulse Code Modulation (PCM))

5.7 Elephant detection model

5.7.1 Data set

5.7.2 Accuracy

5.7.3 Improvements

6 Conclusion and Future Works.

- New possibilities discovered.
- Problems encountered.
- Increasing the accuracy of detection and localization.
- Summary

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