In [2]:
```
# STEP 1: ACCEPT AUDIO INPUT
import os
import IPython.display as ipd

muffled_file = 'muffled-talking-6161.mp3' # source url: https://pixabay.com/so
sample_rate = 44100 # default sample rate for audio processing is 44.1 Hz

ipd.Audio(muffled_file, rate=sample_rate)
```

Out[2]:

0:03 / 0:18

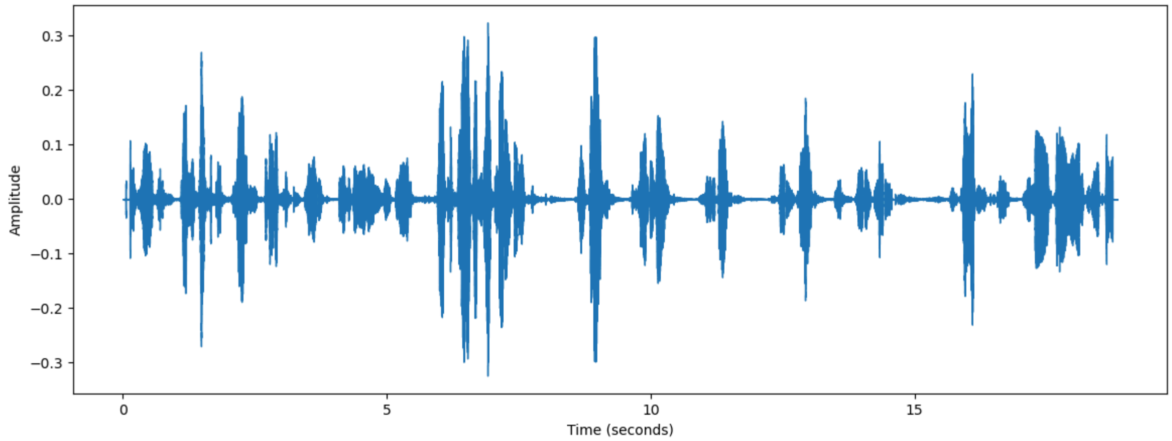
In [3]:
```
%pip install librosa
```

```
Defaulting to user installation because normal site-packages is not writeabl
e
Requirement already satisfied: librosa in c:\users\vaishnavi pachava\appdata
\roaming\python\python311\site-packages (0.10.1)
Requirement already satisfied: audioread>=2.1.9 in c:\users\vaishnavi pachav
a\appdata\roaming\python\python311\site-packages (from librosa) (3.0.1)
Requirement already satisfied: numpy!=1.22.0,!=1.22.1,!=1.22.2,>=1.20.3 in
c:\programdata\anaconda3\lib\site-packages (from librosa) (1.24.3)
Requirement already satisfied: scipy>=1.2.0 in c:\programdata\anaconda3\lib
\site-packages (from librosa) (1.11.1)
Requirement already satisfied: scikit-learn>=0.20.0 in c:\programdata\anacon
da3\lib\site-packages (from librosa) (1.3.0)
Requirement already satisfied: joblib>=0.14 in c:\programdata\anaconda3\lib
\site-packages (from librosa) (1.2.0)
Requirement already satisfied: decorator>=4.3.0 in c:\programdata\anaconda3
\lib\site-packages (from librosa) (5.1.1)
Requirement already satisfied: numba>=0.51.0 in c:\programdata\anaconda3\lib
\site-packages (from librosa) (0.57.1)
Requirement already satisfied: soundfile>=0.12.1 in c:\users\vaishnavi pacha
va\appdata\roaming\python\python311\site-packages (from librosa) (0.12.1)
Requirement already satisfied: pooch>=1.0 in c:\users\vaishnavi pachava\appd
ata\roaming\python\python311\site-packages (from librosa) (1.8.1)
Requirement already satisfied: soxr>=0.3.2 in c:\users\vaishnavi pachava\app
data\roaming\python\python311\site-packages (from librosa) (0.3.7)
Requirement already satisfied: typing-extensions>=4.1.1 in c:\programdata\an
aconda3\lib\site-packages (from librosa) (4.7.1)
Requirement already satisfied: lazy-loader>=0.1 in c:\programdata\anaconda3
\lib\site-packages (from librosa) (0.2)
Requirement already satisfied: msgpack>=1.0 in c:\programdata\anaconda3\lib
\site-packages (from librosa) (1.0.3)
Requirement already satisfied: llvmlite<0.41,>=0.40.0dev0 in c:\programdata
\anaconda3\lib\site-packages (from numba>=0.51.0->librosa) (0.40.0)
Requirement already satisfied: platformdirs>=2.5.0 in c:\programdata\anacond
a3\lib\site-packages (from pooch>=1.0->librosa) (3.10.0)
Requirement already satisfied: packaging>=20.0 in c:\programdata\anaconda3\l
ib\site-packages (from pooch>=1.0->librosa) (23.1)
Requirement already satisfied: requests>=2.19.0 in c:\programdata\anaconda3
\lib\site-packages (from pooch>=1.0->librosa) (2.31.0)
Requirement already satisfied: threadpoolctl>=2.0.0 in c:\programdata\anacon
da3\lib\site-packages (from scikit-learn>=0.20.0->librosa) (2.2.0)
Requirement already satisfied: cffi>=1.0 in c:\programdata\anaconda3\lib\sit
e-packages (from soundfile>=0.12.1->librosa) (1.15.1)
Requirement already satisfied: pycparser in c:\programdata\anaconda3\lib\sit
e-packages (from cffi>=1.0->soundfile>=0.12.1->librosa) (2.21)
Requirement already satisfied: charset-normalizer<4,>=2 in c:\programdata\an
aconda3\lib\site-packages (from requests>=2.19.0->pooch>=1.0->librosa) (2.0.
4)
Requirement already satisfied: idna<4,>=2.5 in c:\programdata\anaconda3\lib
\site-packages (from requests>=2.19.0->pooch>=1.0->librosa) (3.4)
Requirement already satisfied: urllib3<3,>=1.21.1 in c:\programdata\anaconda
3\lib\site-packages (from requests>=2.19.0->pooch>=1.0->librosa) (1.26.16)
Requirement already satisfied: certifi>=2017.4.17 in c:\programdata\anaconda
3\lib\site-packages (from requests>=2.19.0->pooch>=1.0->librosa) (2023.7.22)
Note: you may need to restart the kernel to use updated packages.
```

In [4]:
```
# STEP 2: DISPLAY AUDIO AS A SIGNAL
import librosa # package for music & audio analysis
import numpy
from pylab import plot, figure, show, xlabel, ylabel

# PART A: WORKING WITH INPUT AUDIO
muffled_c4, m_sr = librosa.load(muffled_file)

# plot the signal's waveform
figure(figsize=(14, 5))
librosa.display.waveshow(muffled_c4, sr=m_sr)
xlabel('Time (seconds)')
ylabel('Amplitude')
show()
```




In [5]:
```
# PART B: TRYING SIMPLE MODIFICATIONS -> THEY AREN'T GOOD ENOUGH

# amplitude multiplier
amp_factor = 100
amp_signal = muffled_c4 * amp_factor
ipd.Audio(amp_signal, rate=m_sr)
```

Out[5]:

0:16 / 0:18

In [6]:
```
# slowing down audio
stretch_signal = librosa.effects.time_stretch(muffled_c4, rate=0.85)
resampled_stretch_signal = librosa.resample(stretch_signal, orig_sr=len(stretc
ipd.Audio(stretch_signal, rate=m_sr)
```

Out[6]:

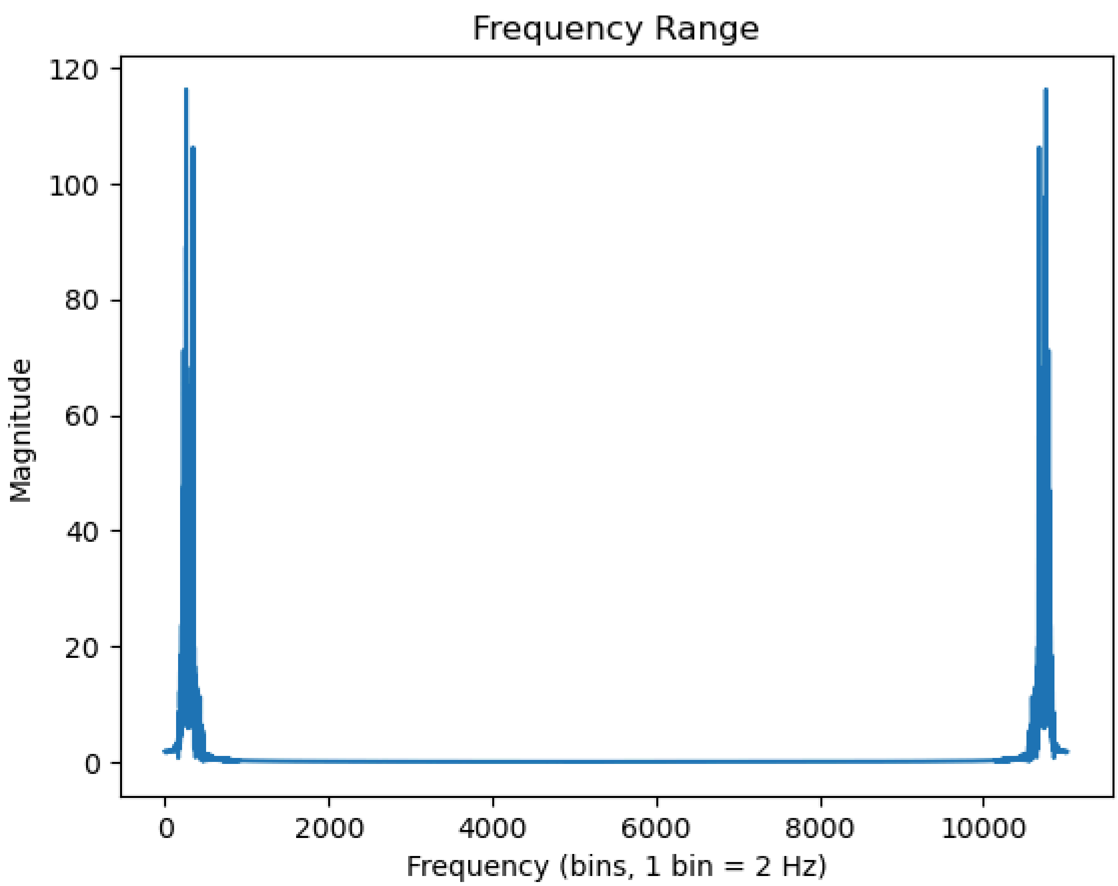
0:15 / 0:22

In [7]:
```
# STEP 3: EXTRACT FREQUENCIES VIA FOURIER TRANSFORM
from numpy import zeros
from cmath import exp, pi
from pylab import title

# define discrete fourier transform
def dft(y):
    N = len(y)
    c = zeros(N, complex)
    for k in range(N):
        for n in range(N):
            c[k] += y[n]*exp(-2j*pi*k*n/N)
    return c

# decide a subset for the transform -> where there is the most audio modulatio
startIdx = 6 * 22050 # find the sample # at 6s mark
endIdx = 11025 + startIdx # find the sample # at 6.5s mark

# plot transform of signal
transform = dft(muffled_c4[startIdx:endIdx])
plot(abs(transform))
xlabel('Frequency (bins, 1 bin = 2 Hz)')
ylabel('Magnitude')
title('Frequency Range')
show()
```

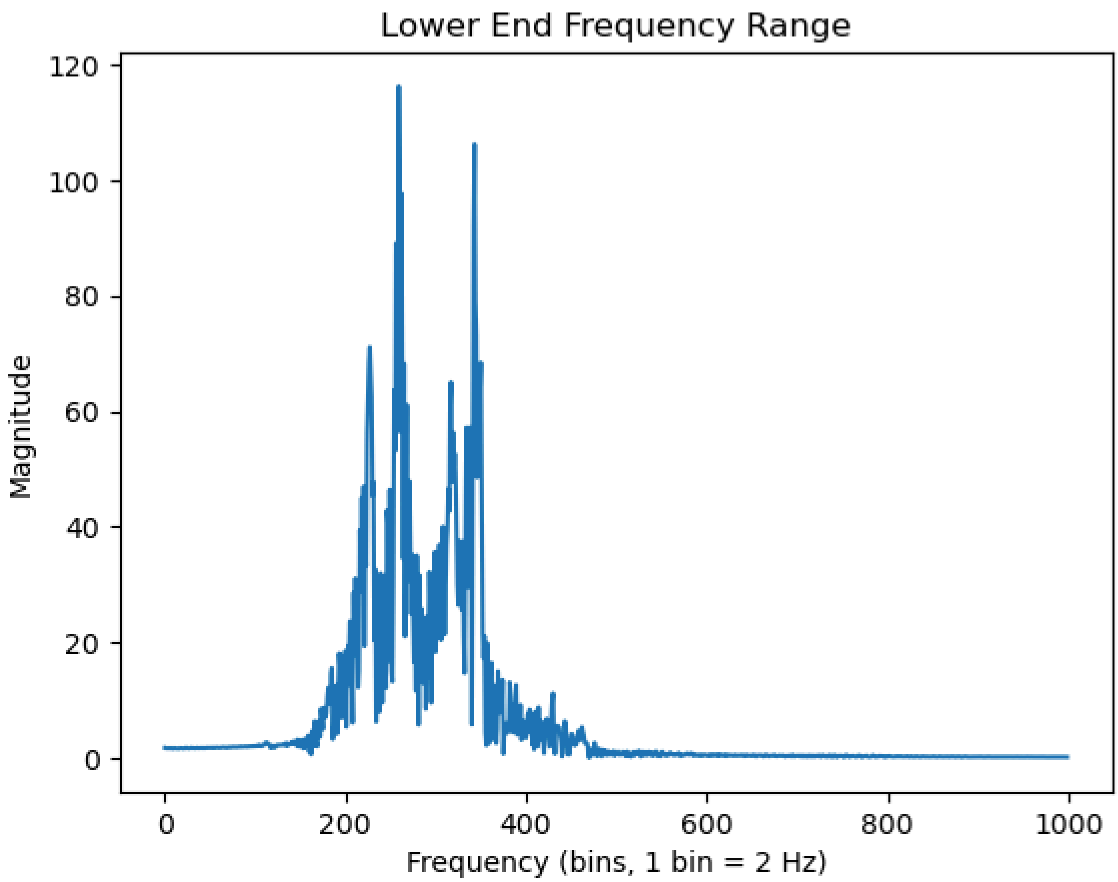


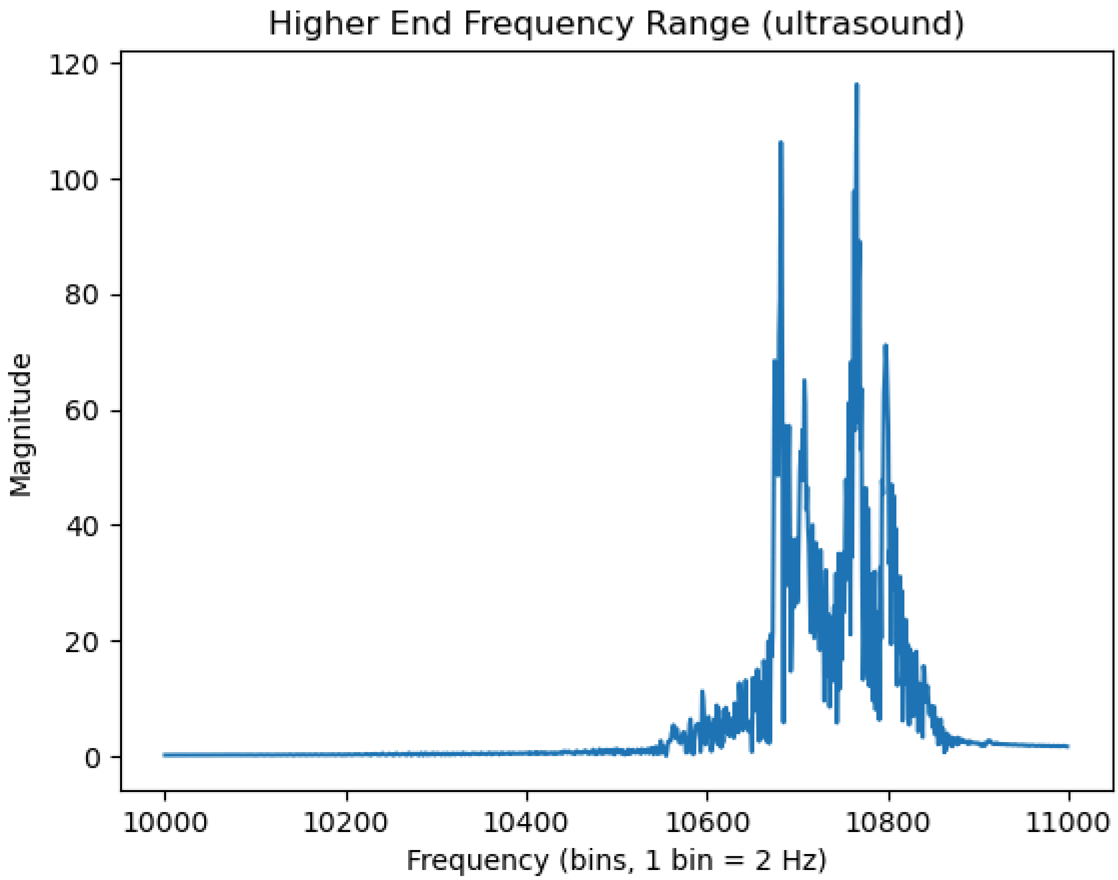
In [8]:
```
# zoom into interesting sections of the plot i.e. sections with the most chang

# lower end frequency range
lowerend_transform = abs(transform[0:1000])
plot(lowerend_transform)
xlabel('Frequency (bins, 1 bin = 2 Hz)')
ylabel('Magnitude')
title('Lower End Frequency Range')
show()

# higher end frequency range
from numpy import arange

higherend_transform = abs(transform[10000:11000])
plot(10000 + arange(1000), higherend_transform)
xlabel('Frequency (bins, 1 bin = 2 Hz)')
ylabel('Magnitude')
title('Higher End Frequency Range (ultrasound)')
show()
```




Higher End Frequency Range (ultrasound)
Magnitude
0
20
40
60
80
100
120
10000
10200
10400
10600
10800
11000
Frequency (bins, 1 bin = 2 Hz)

In [9]:
```
# STEP 4: DEVISE METHODS TO SELECTIVELY AMPLIFY COMPLEX SOUNDS
from numpy import sqrt

# PART A: FREQUENCY ANALYSIS VIA GOLDEN RATIO SEARCH FOR LOCAL & GLOBAL MAXIMA

# local maxima in an audio signal are all harmonic frequencies
def find_harmonics(signal, tolerance):
    golden_ratio = (sqrt(5) - 1) / 2

    a = 0
    b = len(signal) - 1

    freq_maxima = []

    while abs(b - a) > tolerance:
        c = b - (b - a) * golden_ratio
        d = a + (b - a) * golden_ratio

        fc = signal[int(c)]
        fd = signal[int(d)]

        if fc < fd:
            a = c
            freq_maxima.append(int(c))
        else:
            b = d
            freq_maxima.append(int(d))

    return freq_maxima

# the global maximum is the most common frequency, which is the fundamental fr
def find_fundamental(signal, local_maxima):
    max_val = 0
    max_freq = 0

    for m in local_maxima:
        if signal[m] > max_val:
            max_val = signal[m]
            max_freq = m

    return max_freq

# test on lower end frequency range
lowerend_harmonics = find_harmonics(lowerend_transform, 1e-05)
lowerend_fundamental = find_fundamental(lowerend_transform, lowerend_harmonics

print('Fundamental Frequency: ', lowerend_fundamental, ' Hz')
print(lowerend_transform[259])
print(lowerend_transform[260])
print(lowerend_transform[258])

print('Harmonics: ')
for h in lowerend_harmonics:
```

```
    print(h * 2, ' Hz')
```

```
Fundamental Frequency:  518  Hz
116.22455061032198
97.98088651687793
57.85533431339062
Harmonics:
1234  Hz
762  Hz
290  Hz
582  Hz
402  Hz
470  Hz
540  Hz
496  Hz
514  Hz
530  Hz
524  Hz
520  Hz
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518  Hz
518  Hz
```

In [18]:
```
# PART B: FREQUENCY TRIMMING

# remove unnecessary frequencies
def freq_trimmer(signal, harmonics, limit):
    N = len(signal)
    c = zeros(N, complex)

    for n in range(N):
        if n in harmonics:
            continue
        if signal[n] < limit:
            c[n] = 0
        else:
            c[n] = signal[n]

    return c.real
```

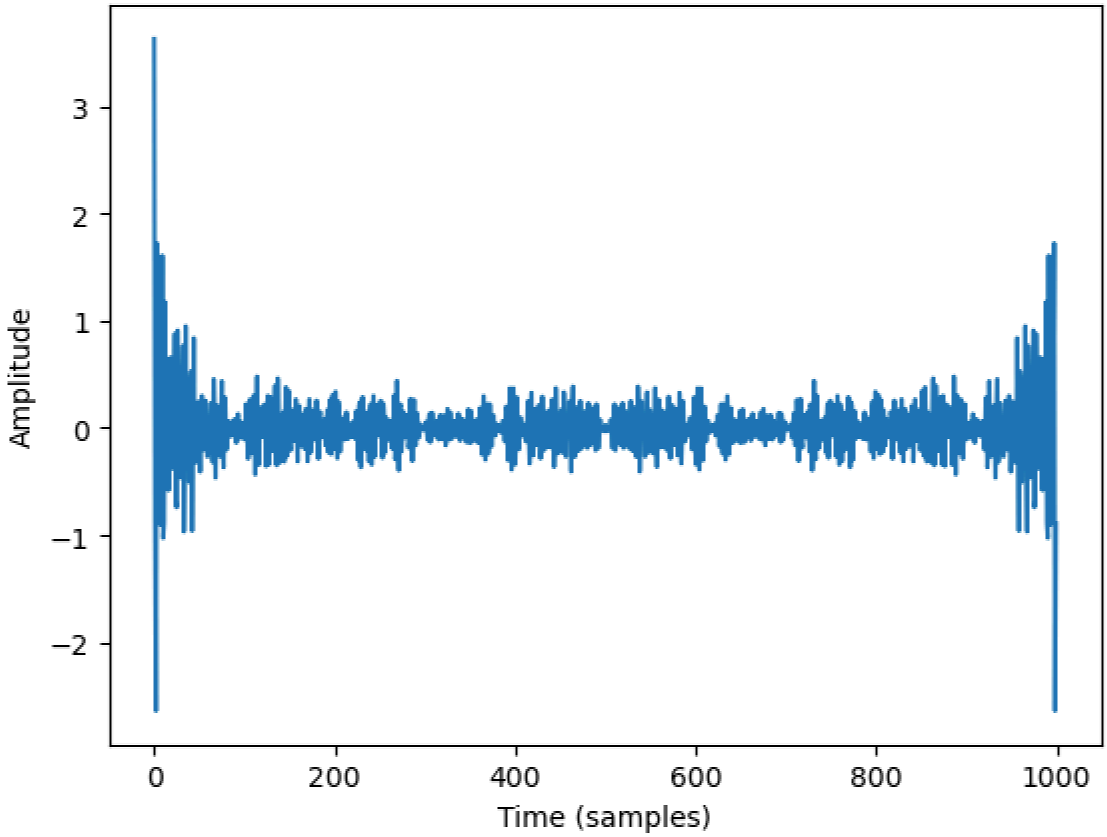
In [60]:
```
# PART C: PITCH SHIFTING VIA PHASE MANIPULATION
def pitch_push(signal, freq_shift, sr):
    # compute the phase shift based on the frequency shift
    phase_shift = 2 * pi * freq_shift / sr

    # normalize time axis that helps incorporate shift
    N = len(signal)
    t = arange(N)

    # apply phase shift to direct audio signal
    shifted_audio = zeros(N, complex)
    for n in range(N):
        shifted_audio[n] = signal[n] * exp(phase_shift*t[n]/sr)

    return shifted_audio.real
```

In [61]:
```
# PART D: NOISE REDUCTION BY SPECTRAL SUBTRACTION
from numpy import mean, where, max

def noise_reduce(signal, threshold):
    # calculate power spectrum from fourier transform
    power_spectrum = abs(signal) ** 2

    # apply noise condition to find general noise spectrum from power spectrum
    noise_spectrum = mean(power_spectrum[where(power_spectrum < threshold * ma

    # perform spectral subtraction
    clean_spectrum = power_spectrum - noise_spectrum
    plot(clean_spectrum)
    return clean_spectrum
```

In [12]:

```python
# STEP 5: PLAYBACK AMPLIFIED AUDIO & COMPARE WITH ORIGINAL

# define inverse discrete fourier transform to convert fourier coefficients ba
def inverse_dft(c):
    N = len(c)
    y = zeros(N, complex)
    for k in range(N):
        for n in range(N):
            y[k] += c[n]*exp(2j*pi*k*n/N)/N
    return y.real

# plot the modified audio signal
def playback_plot(signal):
    # plot the signal's waveform
    plot(signal)
    xlabel('Time (samples)')
    ylabel('Amplitude')
    show()
```

In [22]:
```
# PART A: TEST FREQUENCY TRIMMING
trimmed_signal = freq_trimmer(lowerend_transform, lowerend_harmonics, 30)

inv = inverse_dft(trimmed_signal)
playback_plot(inv)

ipd.Audio(inv, rate=m_sr)
```




Out[22]:

0:00 / 0:00

In [13]:
```
# PART B: TEST PITCH SHIFTING
pitchpushed_signal = pitch_push(muffled_c4, 400, m_sr)

ipd.Audio(pitchpushed_signal, rate=m_sr)
```

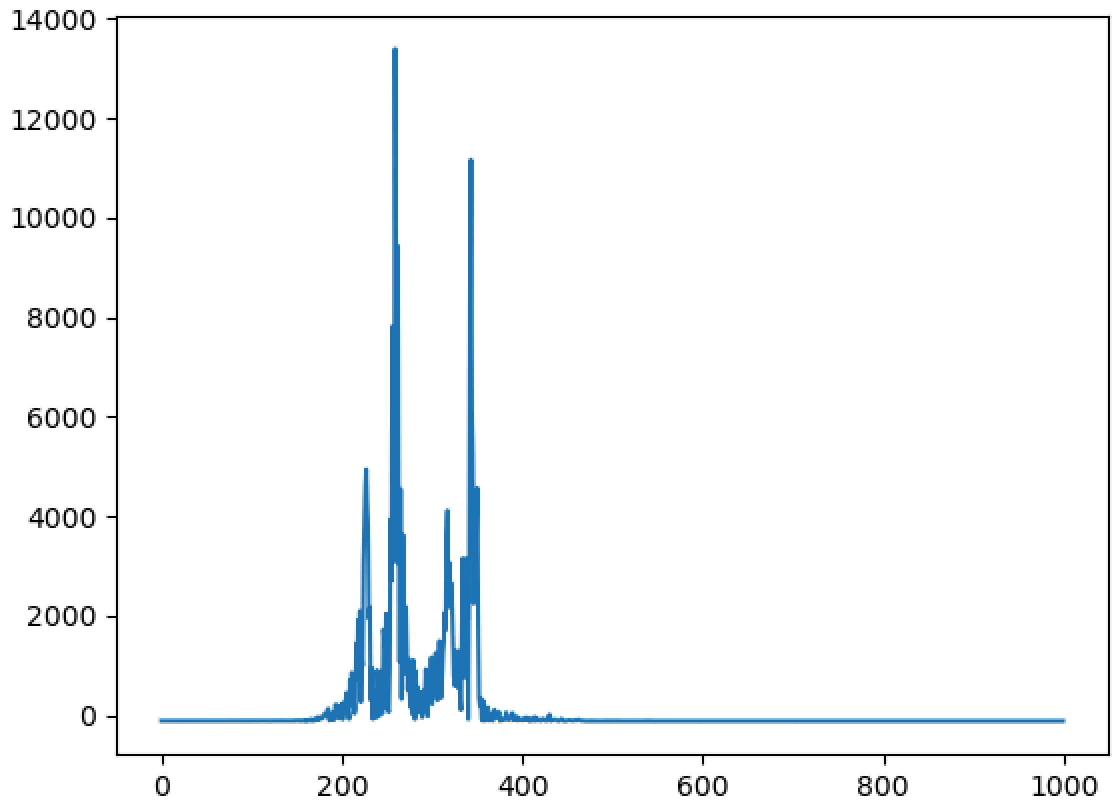
Out[13]:

0:15 / 0:18

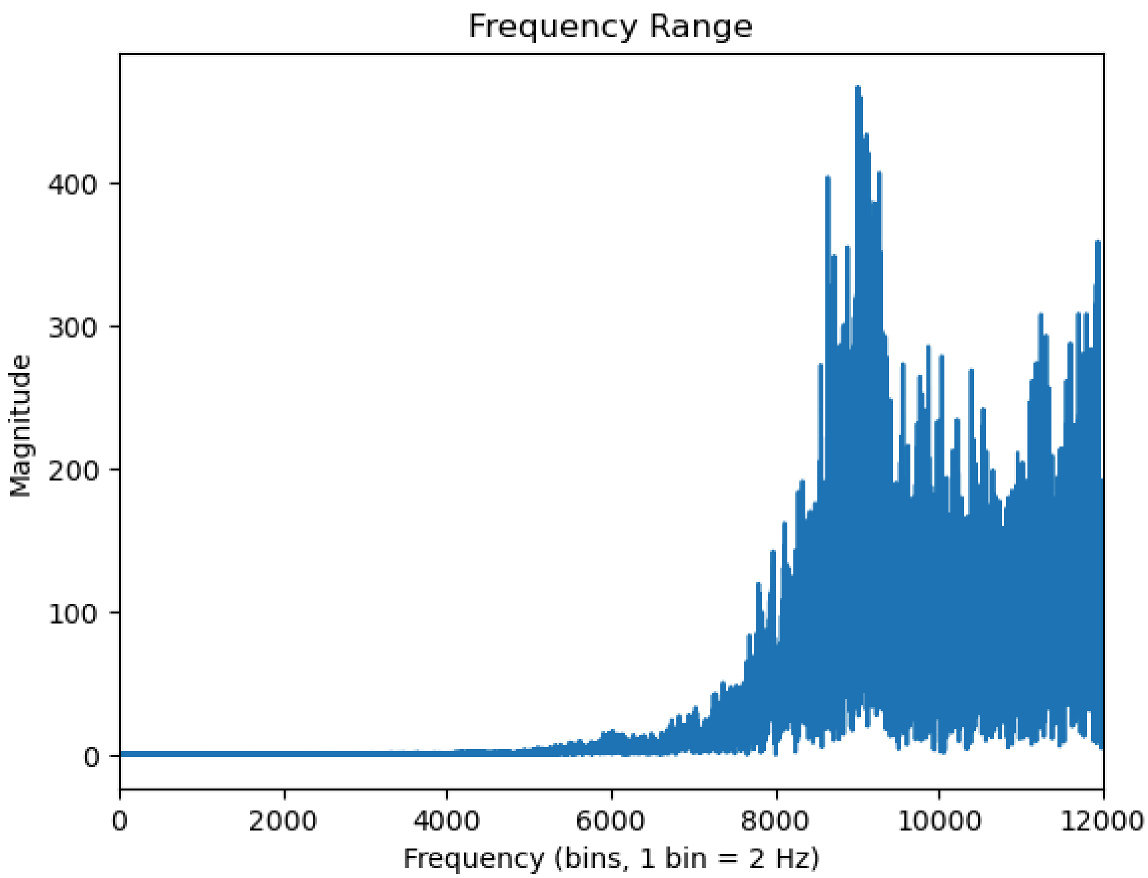
In [49]:
```
# PART C: TEST NOISE REDUCTION
reduced_signal = noise_reduce(lowerend_transform, 0.18)

inv_r = inverse_dft(reduced_signal)

ipd.Audio(inv_r, rate=m_sr)
```

Out[49]:

0:00 / 0:00



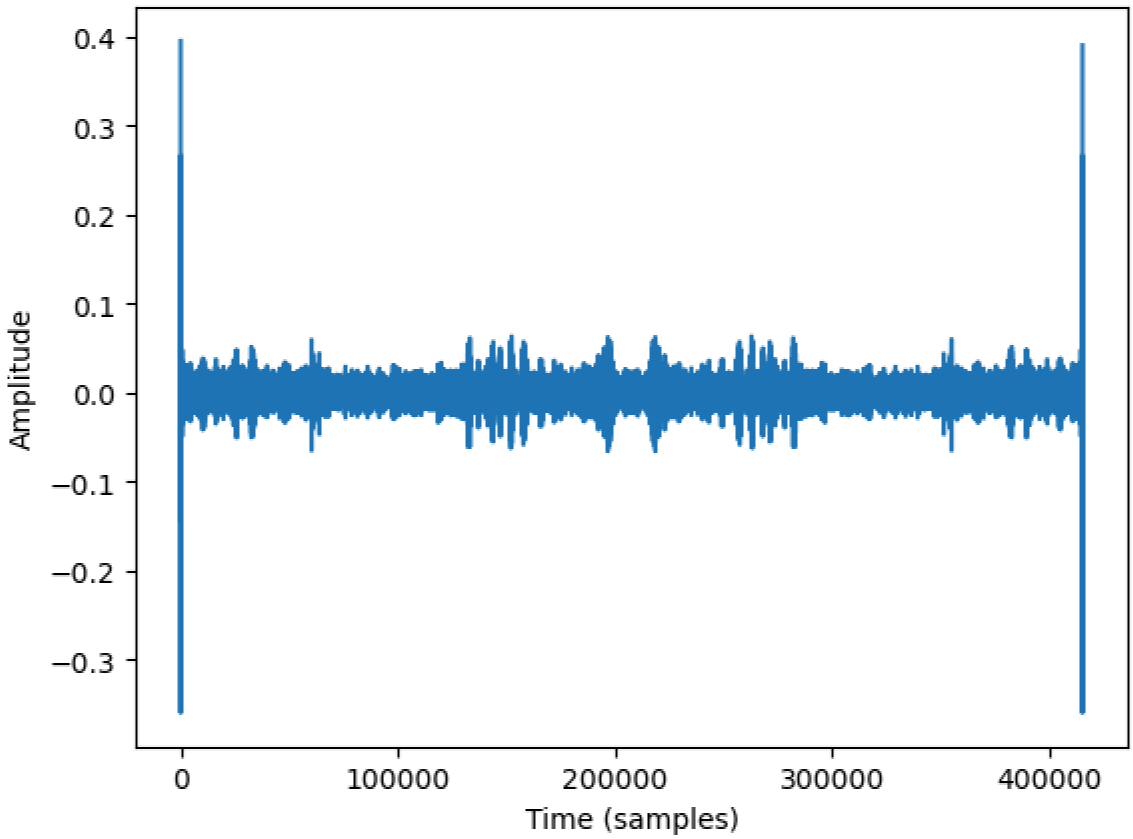
In [93]:
```
# PART D: COMBINE IT ALL -> LIMITED USE ON LARGE NUMBER OF SAMPLES
from numpy.fft import fft as fast_ft # use in-built fast fourier transform to
from numpy.fft import ifft as fast_ift # use in-built fast inverse fourier tr
from pylab import xlim

fulltr = fast_ft(muffled_c4)
plot(abs(fulltr))
xlim(0, 12000)
xlabel('Frequency (bins, 1 bin = 2 Hz)')
ylabel('Magnitude')
title('Frequency Range')
show()
```




In [63]:
```
# PART I: DO FREQUENCY ANALYSIS
full_harmonics = find_harmonics(fulltr, 1e-05)
```

In [96]:

```
# PART II: DO FREQUENCY TRIMMING
full_trim = freq_trimmer(fulltr, full_harmonics, 100)

inv_ft = fast_ift(full_trim)
playback_plot(inv_ft)

ipd.Audio(inv_ft, rate=m_sr)
```



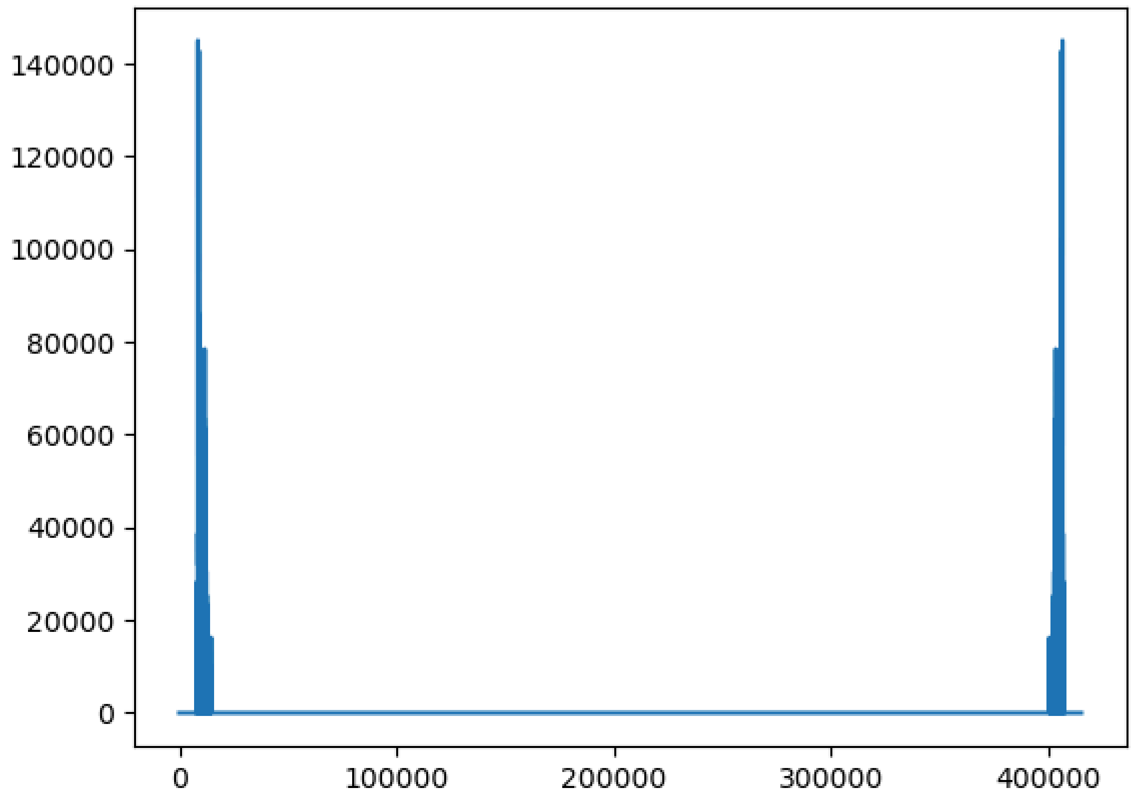

Out[96]:

0:13 / 0:18

In [115]:
```
# PART III: DO NOISE REDUCTION ON TRIMMED WAVE
full_reduce = noise_reduce(fast_ft(inv_ft), 1000)

inv_fr = fast_ift(full_reduce)

ipd.Audio(inv_fr, rate=m_sr)
```

Out[115]:

0:09 / 0:18




In [118]:
```
# PART IV: DO PITCH SHIFTING ON NOISE REDUCED WAVE
full_pitchpush_nr = pitch_push(inv_fr, 50, m_sr)

ipd.Audio(full_pitchpush_nr, rate=m_sr)
```

Out[118]:

0:16 / 0:18

In [ ]: