Project code explanation.

- We are importing the necessary libraries for our program
- First, we create two plots one is for MAS and other is for FFT
- Next, we prepare the plotting environment with random starting values
- Separately we set the limits for raw audio as well as FFT audio.
    - In plot 0 ax[0] is MAS
    - In plot 1 ax[1] is FFT
- In the next step we show the plot without blocking updates in a tight layout.
- Then we initialized the required variables such as format which takes pyaudio library and gets the input in 16 bit format per sample from the user's microphone.
- Channel is the number of channels our microphone has, next is the sampling rate which is set to 44100, chunk is the number of samples which will be displayed at a time instance, then we set the recording seconds to 0.1 seconds per sample.
- We create a variable audio and get the raw audio from user's microphone using PyAudio()
- Then, we use the open method on the object p and pass on the variables
- we initialized as parameters.
- Then we convert the data which we got from microphone into 16 bit floating
- integers, this step is done because to convert time domain to frequency domain.
- To set the fast fourier transform to decibels (dB), we use 10*log10(abs)
- Next step we forced the new data into the plot but without redrawing it.
- Then print the audio data and fft data for 10 intervals
- Then setting the X limit as audio data for the first plot then for second plot
- setting the X limit as fft data.
- We keep on updating the plot without blocking using plt.pause
- Then in the main method we open the connection and start streaming the data.
- In our program we included a short cut which breaks the recording and interrupts the code while Ctrl + C is pressed.