In the Name of God

University of Tehran

School of Electrical and
Computer Engineering

Digital Signal Processing

Computer Assignment #3

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Spring 2021

Question One

Part One:

Two signals are very similar to each other, in other words we can’t differentiate to channels by listening to them. They are different since they have been recorded with two different microphones, so they have phase difference.

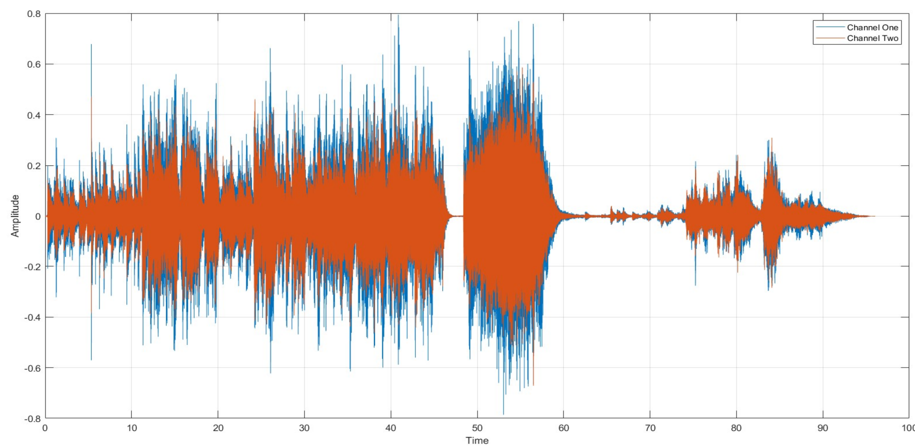

Figure 1

Part Two:

Fourier Transform of two channels has been shown below[1], like in time domain the absolute value of these two are very similar to each other but they are a little bit different in their value.

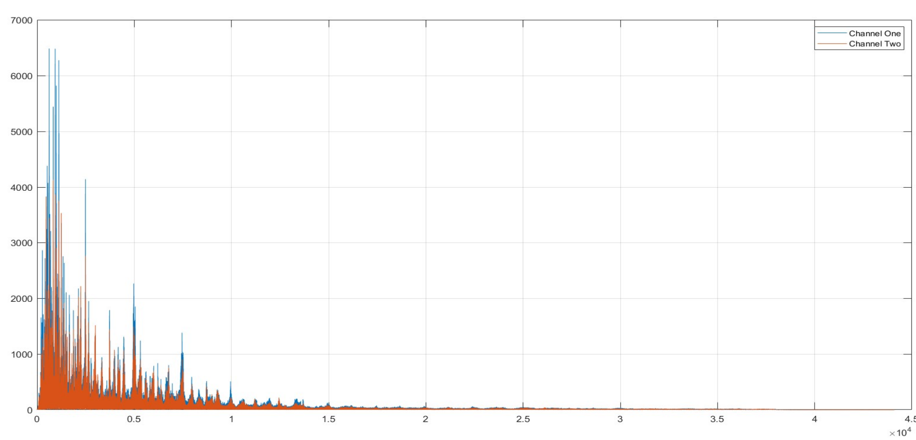

Figure 2

[1] Since FFT is an even function only half of signal has been shown.

As we can see bandwidth of signal is approximately 1.5 kHZ.

Part Three:

Spectral Density of signal is acquired by multiplying Fourier transform of signal to its complex conjugate. The plot below shows this. This plot is very similar to figure 2 which was showing absolute value of signal.

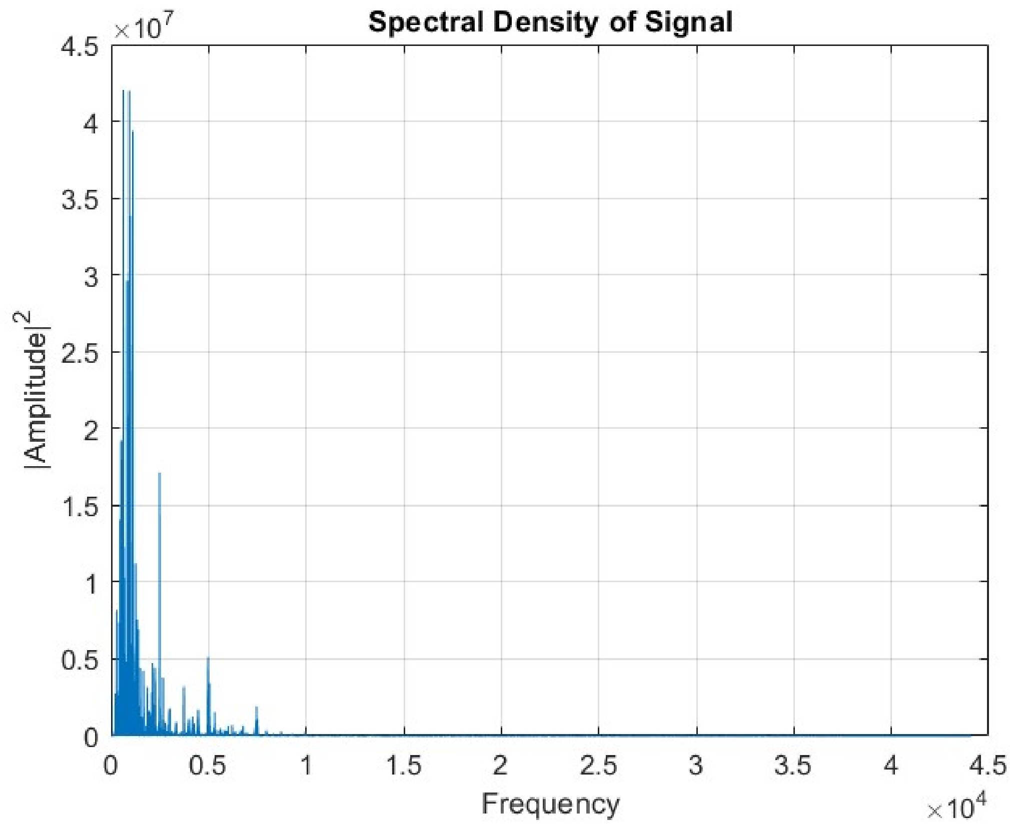

Figure 3

Part Four:

First briefly spectrogram arguments will be explained:

Spectrogram(x, window-size, noverlap, nfft, fs)

X: signal which we want to examine it.

Window-size: spectrogram performs windowing, this parameter determine window size.

Noverlap: windowing causes error in our spectrogram, this parameter shows how samples overlap between windows.

Nfft: uses NFFT number to calculate FFT of each window.

Fs: Sampling frequency.

Spectrogram help us to observe signal in both frequency domain and time domain, output of this function contains an estimate of the short-term, time-localized frequency content of input.

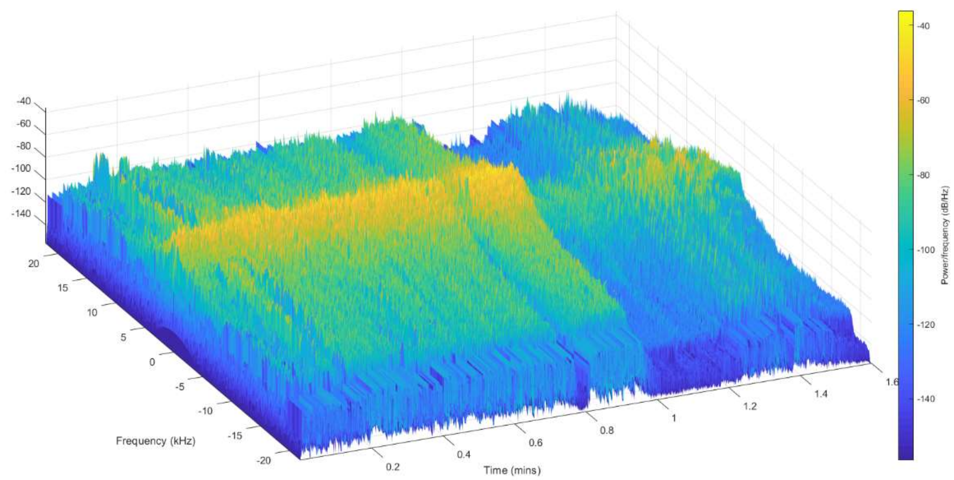

Figure 4

Figure 4 can explain us how spectrogram help us to observe in both domains, if we look at figure 4 from left side we can see it is very simillar to figure 2, the Fourier Transform of signal, and if we look at it from right side it shows us in time domain which is figure 1.

If we compare figure 4 with figure 2 we can understand that at t = 1min signal has more silence sound (blue region) and after that from changing frequency component values, violon sound can be understood.

Part Five: Why we use two channels?

For better listening experience stereo sound (multi channel sound) is used. In these audio signals distinguishing familiar sounds are much easier because in most of cases two or multiple microphone is used to record sound, so recorded sounds is so much familiar to each other but different in some specs, this help us to get better sound quality specially in music.

Part Six: Impact of Arguments

Window size: decreasing window size will lead to smaller frequency tabs, so it increases accuracy in frequency domain, in other hand bigger window size reduces accuracy, but it will decrease computations.

Noverlap: increasing Noverlap decreases error causes by windowing

Nffts: if we pad our signal to 2 powers we will have better accuracy at same speed but increasing more than next power two is useless at just increases computations.

'centered' : acts like fftshift.

Fs: changes sample rate.

Question Two

Part One and Two

After reading V_1 , V_2 , V_3 from given files, and plotting them in time and frequency domain these figures will yield.

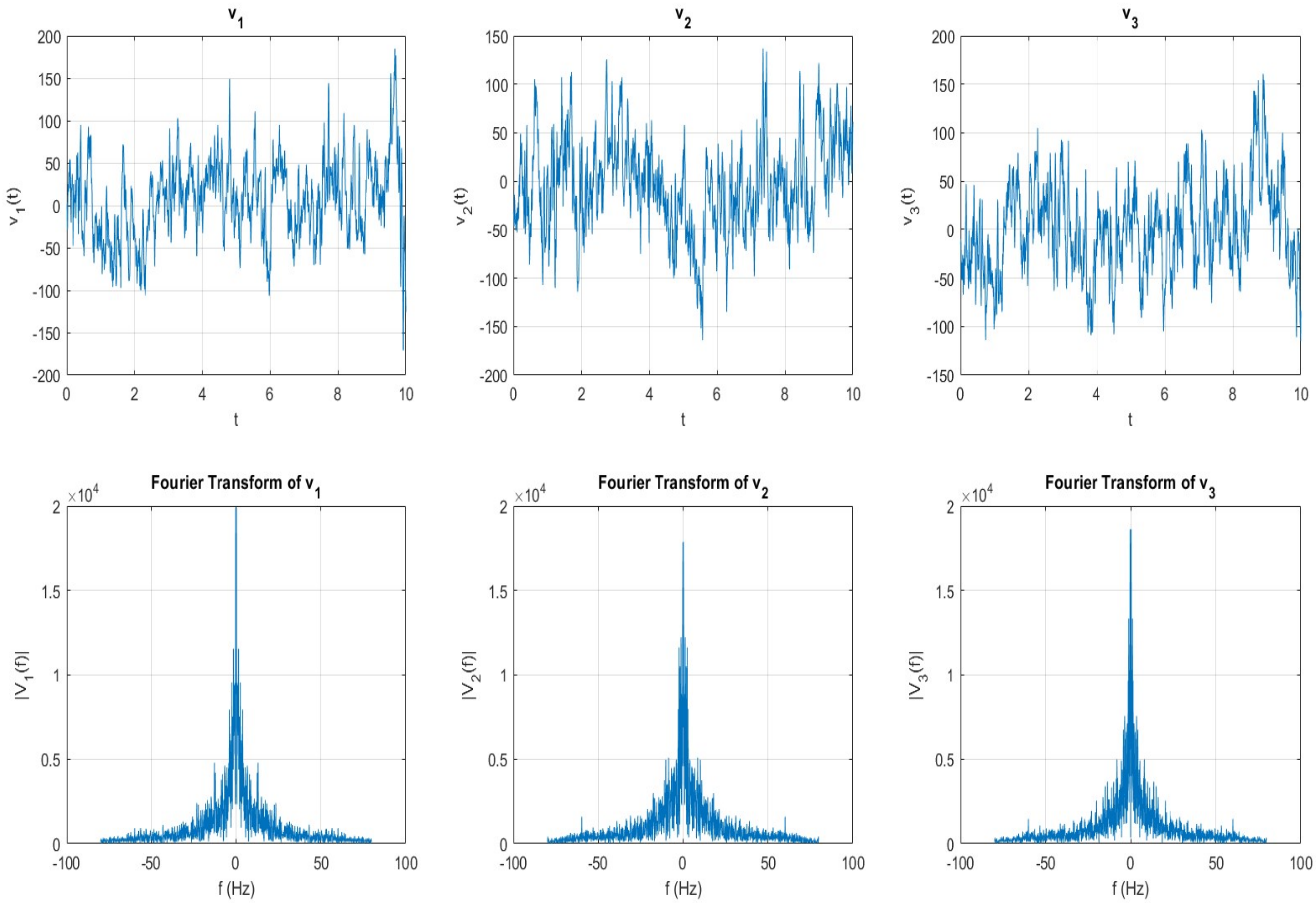

Figure5

Part Three:

Generally, an EEG signal with higher amplitude and lower frequency has more harmonic brain cells (most of them work together in order to reach same goal), and EEG with higher amplitude and higher frequency component indicates less coordinated brain cells.

Part Four

After picking up high amplitude frequency component with a bandpass filter(MATLAB code was used to create something act like a bandpass filter) and perform iFFT on it this five figures will be acquired.

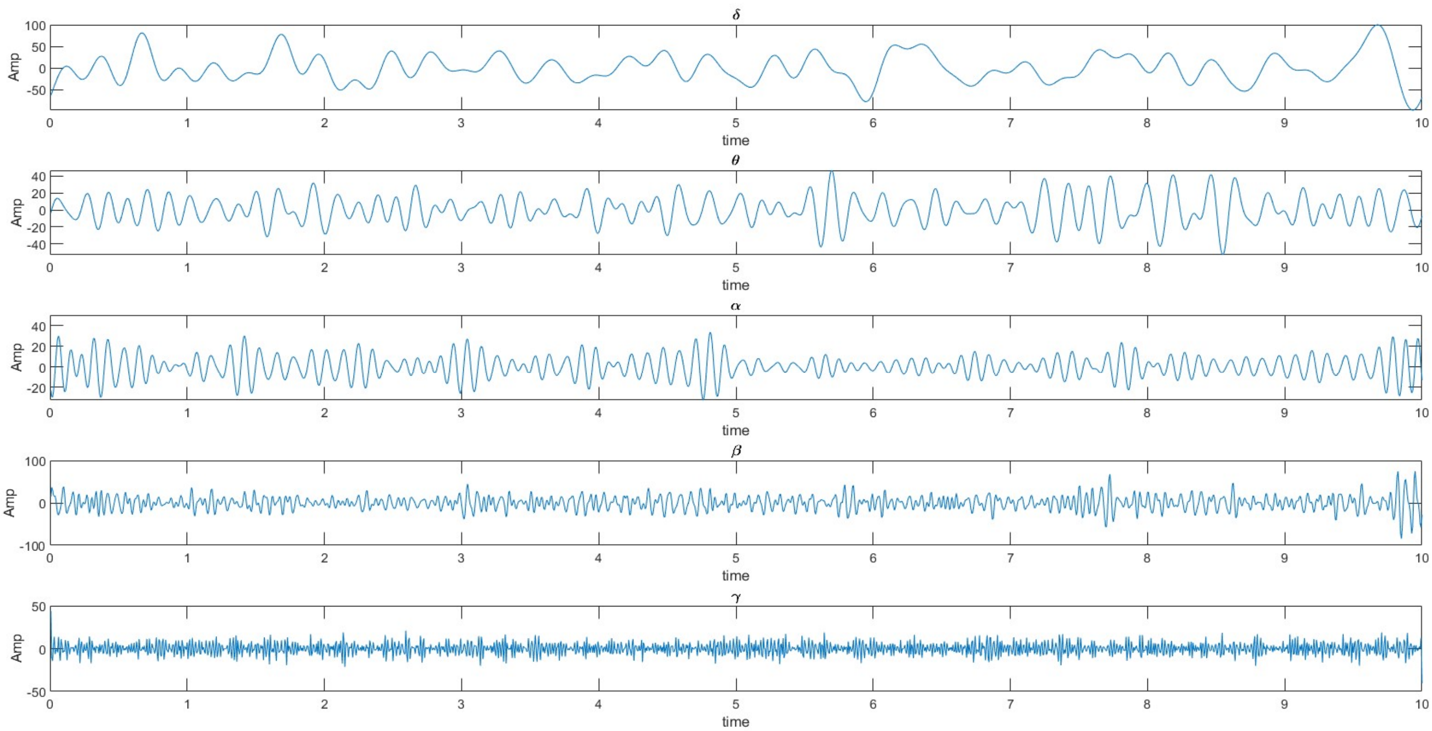

Figure6

Part Five

No we cannot find a reasonable relation between amplitude of EEG and frequency from this figures since δ , θ, α, β, γ is increasing in frequencies but they have these peak of time domain amplitude orderly: 73, 52, 41.2, 55, 51. We can see there is no such relation..

Part Six

From amplitude and density of parameters we can understand what person is doing during this 10 second, from amplitude and density of δ I suggest he was sleeping.

Question Three

Part One to Three

After creating cepstrum function, giving first row of $V_1, V_2 . V_3$ to it and plotting by Quefrency figure below will be got:

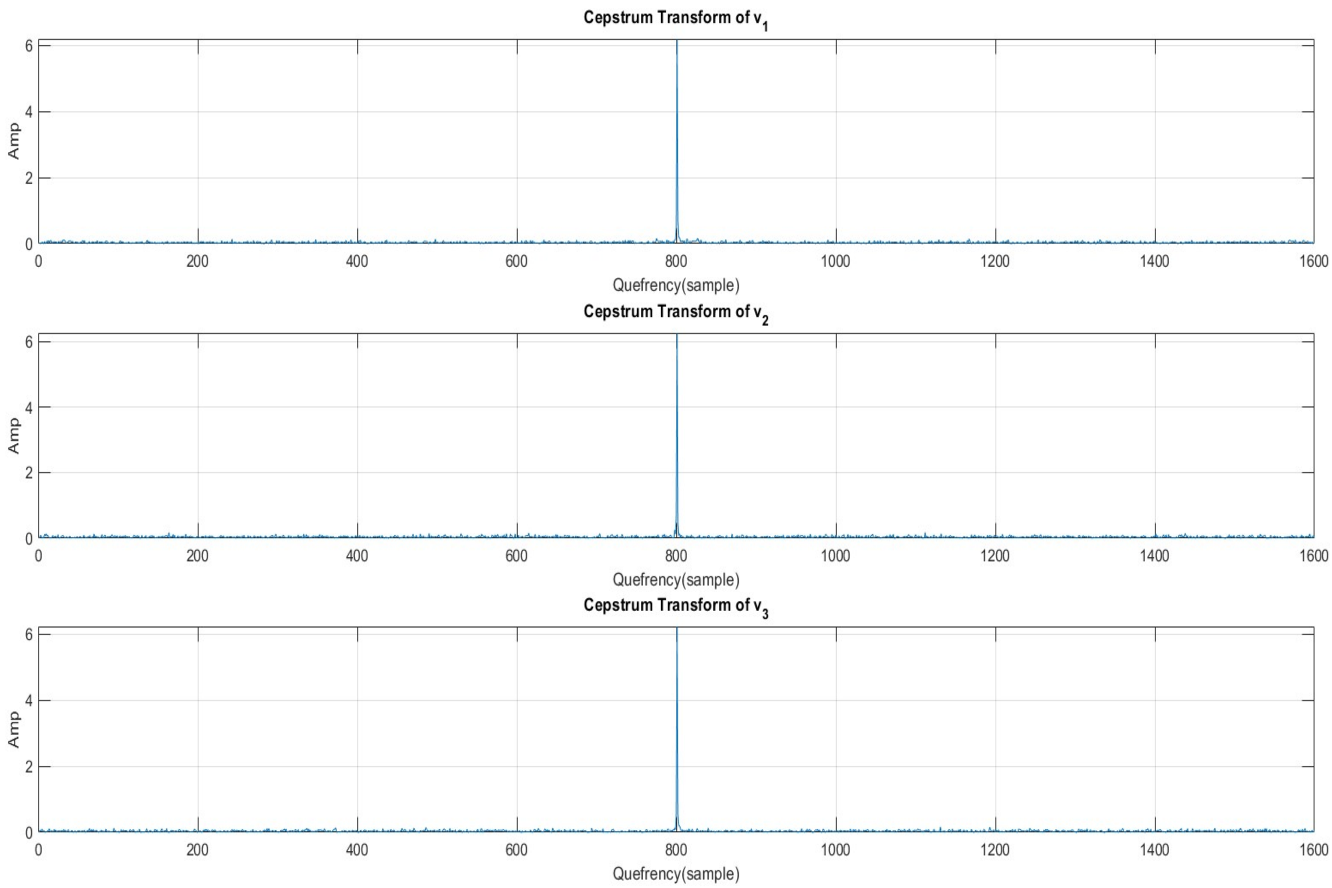

Figure7

As we can see the peak of cepstrum is at Quefrency = 800.

Part 4

As we know quefrecy has reverse relation with frequency so:

$$F_S = \frac{sample}{second} \rightarrow freq = \frac{F_s}{quefrency}$$

$$for\ instance\ V_2\ (other\ are\ simillar)$$

$$f: 0.1hz; freq \rightarrow \frac{80}{801} = 0.997$$

As we can see resulting number is very close to what we expected.

Part five

As mentioned earlier quefrecy has reverse relation with frequency and it kind of time domain unit per sample.

Question Four

Part One , Two and Three

In this section we want to remove an specific word which is California, firstly we read Arnold.wav and save it in a vector. We are going to remove remove California from this file.

After calculating normalized Cross Correlation of these two signals and finding its maximum values, we will find California position in audio file .

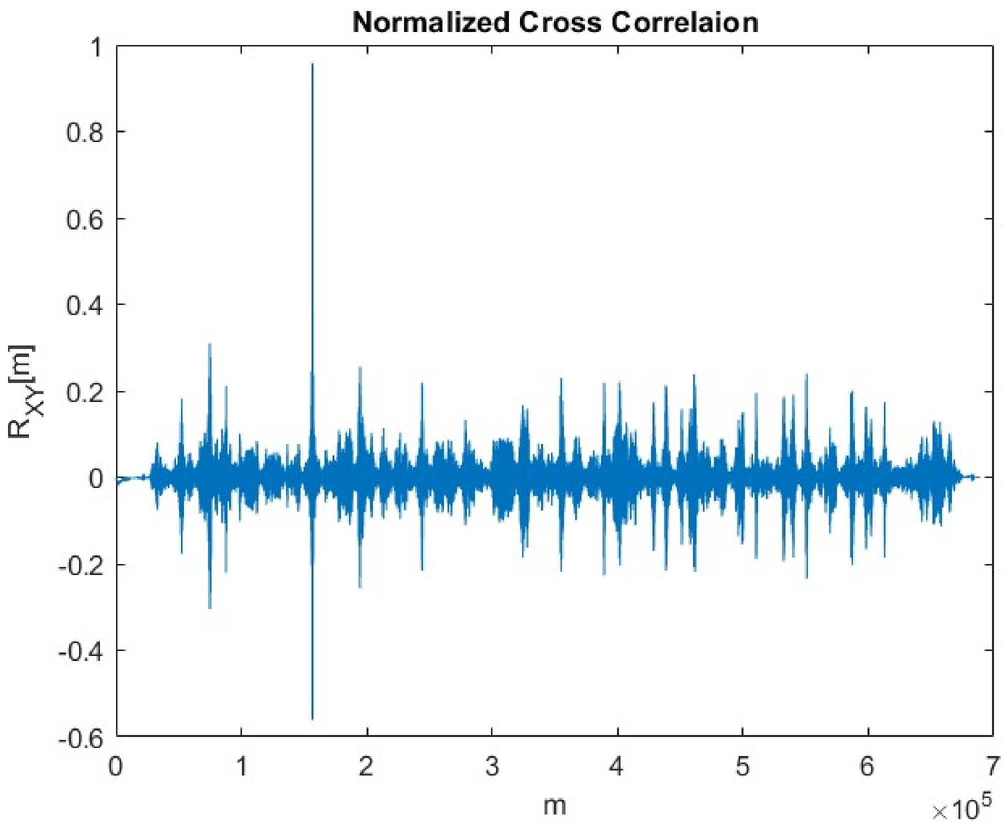

Figure 8

California sound has been removed and replaced by a bleep and saved.