IMPLEMENTATION OF ADAPTIVE STFT ALGORITHM FOR LFM SIGNALS

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Abstract

Normally Time-Frequency analysis is done by sliding a window through the time domain data and computing the Fourier Transform of the data within the window. The choice of the window length determines whether specular or resonant information will be emphasized. A narrow window will isolate specular reflections but will not be wide enough to accommodate the slowly varying global resonances; a wide window cannot temporally separate resonance and specular information. So we will adapt window length according to changes in frequencies. In this case we are realizing the specifications of Linear Frequency Modulation (LFM) signal.

***Index Terms*—LFM, FFT, DFT, STFT and ASTFT**.

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1. INTRODUCTION

Present scenario Electronic Warfare receiver need to handle wide variety of signals comprising high signal density. There are two types of signals present: one, Stationary signals, whose frequency doesn‟t vary with time and the other signals, whose frequency vary with time. The stationary signals can be analyzed by using the Fast Fourier transforms (FFT) whereas for other type of signals cannot be analyzed with FFT since their frequency varies with time. We will realize these signals by using Short Time Fourier Transform (STFT) whereas in STFT the window length will be constant throughout the analysis. So there will be some time and frequency resolution problems that may occur during analysis because of fixed window length. So we will go for Adaptive Short Time Fourier Transform (ASTFT) [1], [2].

Normally Time-Frequency analysis is done by sliding a window through the time domain data computing the Fourier Transform of the data within the window. The choice of the window length determines whether specular or resonant information will emphasized. A narrow window will isolate specular reflections but will not be wide enough to accommodate the slowly varying global resonances; a wide window cannot temporally separate resonances and specular information. So we will adapt the window length according to changes in frequencies. Electronic warfare is a silent war. The basic concept of EW is to exploit the enemy's electromagnetic emissions in all parts of the EM spectrum in order to provide intelligence on the enemy's order of the battle, intensions and capabilities and to the foe's electromagnetic discharges in all parts of the EM range to give knowledge on the foe's request of the fight, intensions and capacities and to use counter measures to deny effective use of communications and weapons while protecting one's own use of the same spectrum. In this chapter, the role of electronic warfare in the national security, its different applications is discussed. An introduction to the radar systems and their evolution in the electronic warfare is discussed. The various radar signal parameters analyzed by a radar receiver along with the different kinds of radar receivers [3].

2. SHORT TIME FOURIER TRANSFORM (STFT)

The Short Time Fourier Transform (STFT), is a Fourier-related change used to focus the sinusoidal recurrence and stage substance of nearby areas of a sign as it changes after some time [4].

2.1. Continuous-Time STFT

Just, in the persistent time case, the capacity to be changed is reproduced by a window function which is nonzero for just a brief time of time. The Fourier change (an one-dimensional capacity) of the subsequent sign is taken as the window is slid along the time pivot, bringing about a two-dimensional representation of the sign. Scientifically, this is composed as:

$$\mathrm{STFT}\{\mathrm{x}(\mathrm{t})\} = X(\tau, \omega) = \int_{-\infty}^{+\infty} x(t) w(t-\tau) e^{-j\omega t} dt \qquad (1)$$

where w(t) is the window capacity, ordinarily a Hann window or Gaussian ringer based on zero, and x(t) is the sign to be changed. X(τ,ω) is basically the Fourier

Transform of x(t)w(t-τ), a mind boggling capacity speaking to the stage and extent of the sign after some time and recurrence. Frequently stage unwrapping is utilized along either or both the time hub, τ, and recurrence hub, ω, to stifle any hop brokenness of the time record τ is typically thought to be "moderate" time and more often than not communicated in as high determination as time *t* [5].

2.2 Discrete-Time STFT

In the discrete time case, the information to be changed could be separated into lumps or casings (which for the most part cover one another, to lessen ancient rarities at the limit). Every lump is Fourier changed, and the intricate result is added to a network, which records size and stage for every point in time and recurrence. This can be communicated as:

$$\mathrm{STFT}\{x[n]\} = X(m,\omega) = \sum_{n=-\infty}^{\infty} x[n]w(n-m)e^{-j\omega n} \quad (2)$$

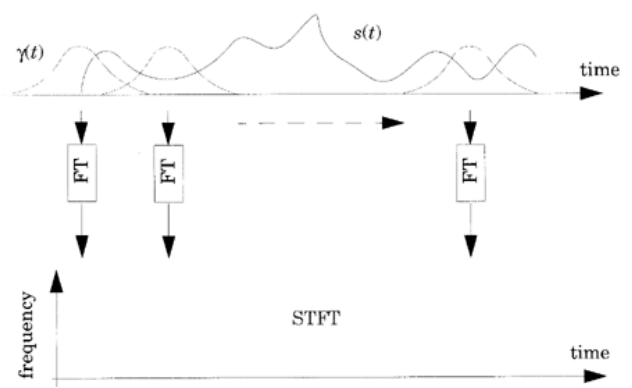

Fig. 1 Short Time Fourier Transform implementation Result of the STFT.

For this situation, m is discrete and ω is constant, however in most common applications the STFT is performed on a PC utilizing the Fast Fourier Transform, so both variables are discrete and quantized. The greatness squared of the STFT yields the spectrogram of the capacity

$$\mathrm{spectrogram}\{x(t)\} = |X(\tau,\omega)|^2 \quad (3)$$

3. ADAPTIVE SHORT TIME FOURIER TRANSFORM (ASTFT)

The important characteristics of a signal are its amplitude, frequency and phase and its computation is very important in many applications. For instance, the measurement of voltage and current in the power system is performed through the traditional DFT-frequency spectrum method. When the signal is non-inter-period sampled, the leakage error occurs, which makes the frequency, amplitude and phase deflect from the real value especially the phase. All together for evaluation of the these parameters numerous strategies were proposed of which a novel calculation of high accuracy parameter estimation of sinusoidal signal based on all phase FFT spectrum analysis is put forward in this project, which is better than all algorithms nowadays not only in the precision and the real time property, but also in the calculation complexity. The system of programming is basic, so it is a perfect estimation strategy. Compared with classical algorithm Discrete Fourier Transform (DFT), Adaptive STFT reduces spectral leakage caused by truncation error efficiently. Another merit of ASTFT is that the changes Frequency is calculated by ASTFT need not being adjusted, and are not affected by Frequency offset. Thus whether the signal is truncated at entire cycles, real and less error result could be got through ASTFT. Now ASTFT applies broadly in digital signal processing such as digital filter design, image processing, etc. Adaptive Short Time Fourier Transform calculation enhances the exactness of Fast Fourier Transform Algorithm by utilizing an uncommon information truncation window [2], [3].

Generally speaking, there are two methods of phase measurement: analog and digitalized method. Due to the advantages of low hardware cost and strong adaptability, in recent years, the latter becomes the mainstream technique. And its strong adaptability embodies in that just simply replacing the internal procedure algorithms of programmable devices (e.g.FPGA, DSP etc.) will adapt itself to different measurement objectives. Obviously, the key problem is to develop a precise phase-measuring algorithm. Discrete Fourier Transform, which usually generates the outcome of complex number containing rich phase information, is a commonly used means of digitalized measuring phase method. A sinusoidal signal with basic parameters (amplitude, phase, frequency) can be estimated using many algorithms of which traditional FFT is significant. A novice method called Adaptive STFT has been proposed. ASTFT has faultless characteristics of phase analysis [1], [3].

ASTFT Algorithm

Algorithm for the Adaptive STFT is:
Set $W_l[n]=W_r[n]=W[n]$.
Set p=q=1.
Compute the spectral kurtosis of the current left short segment
$X_l=X[n]W_r[pL-n]$
Compute the spectral Kurtosis of the right short segment
$X_r=X[n]W_r[(p+q)L-n]$
Set $W_p[n] = \sum_{k=p}^{p+q} W[kl-n]$ and compute the Spectral Kurtosis for the combined frame $Xp=X[n]W_p[pL-n]$
If $C(x_p) > C(x_l) + C(x_r)$ then set $W_l[n]=W_p[n]$ and q=q+1 otherwise set p=p+q, q=1 and set $W_l[n]=W[n]$.

4. RESULTS AND DISCUSSIONS

Input signals which we are using for testing of algorithm was generated in arbitrary waveform generator and applied to algorithm. LFM signal, Extracted signal, Left segment, Right segment and window for total segment are shown in Fig. 2, Fig. 3, Fig. 4, Fig. 5, Fig. 6 and Fig. 7 respectively.

STEP 1: Generation of LFM signal

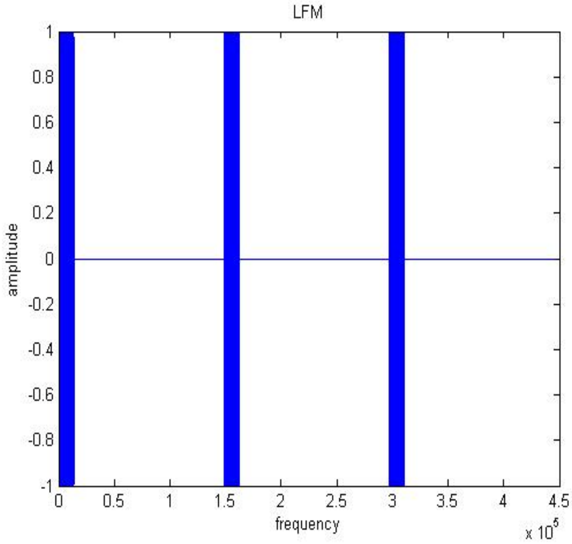

Fig. 2 LFM signal

STEP 2: Extracting the 1024 samples data from it

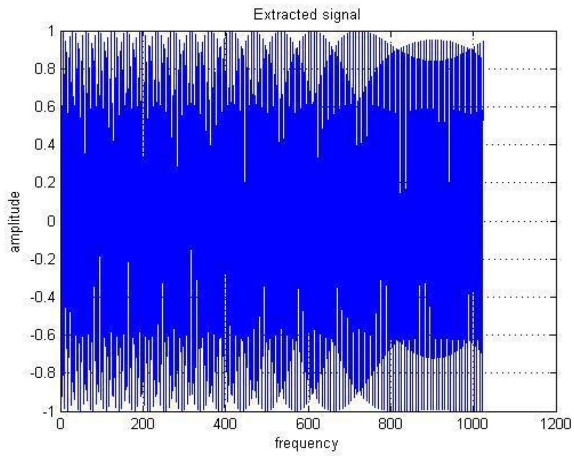

Fig. 3 Extracted signal

Step 3: Left segment data

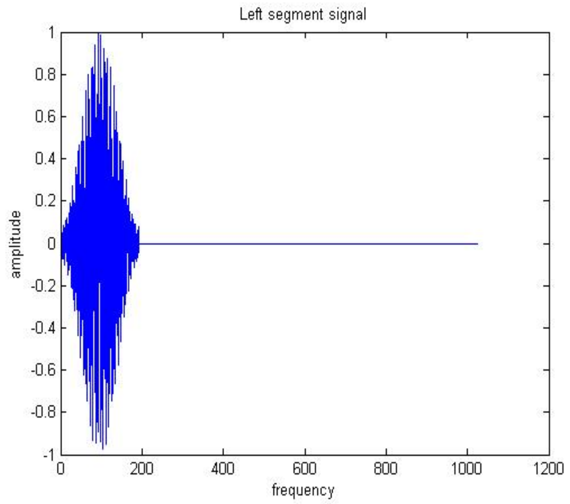

Fig. 4 Left segment signal

Step 4: Right Segment data

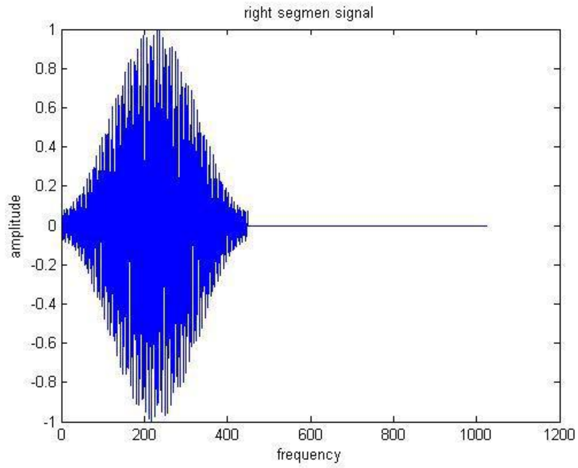

Fig. 5 Right segment data

Step 5: Windows for calculation of total segment data
The calculations made for normal LFM signal

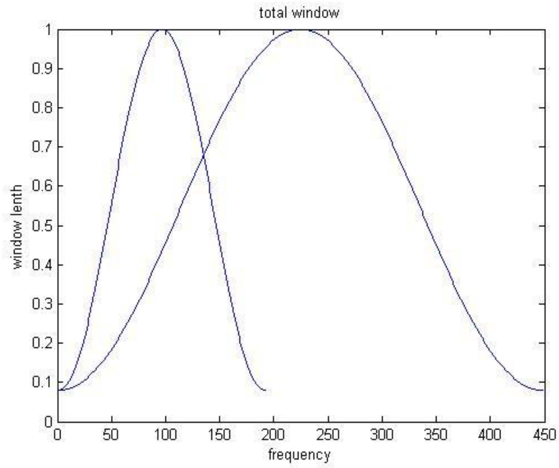

Fig. 6 Window for total segment data

Step 6: Total Segment data:

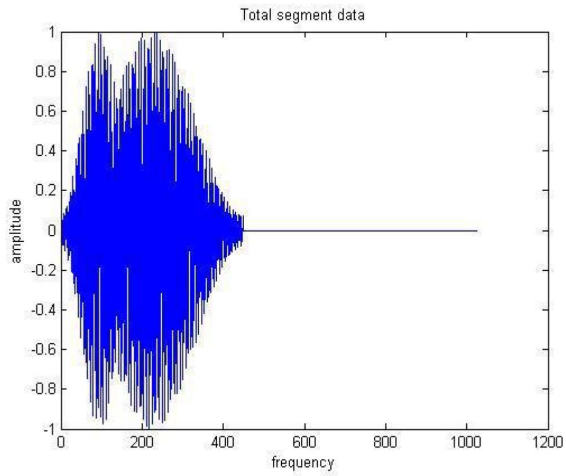

Fig. 7 Total segment data

The input parameters of the signal are start frequency is 1.5 GHz, end frequency is 1.6 GHz, slope is 100 MHz/us, on time is 1us and off time is 10us.

CALCULATION RESULTS USING MATLAB:

i. This calculations are done for first 1024 samples of LFM signal Spectral Kurtosis which we obtain from the matlab are:
Left Segment data Spectral Kurtosis is, C=1.
Right Segment data Spectral Kurtosis is, V=3.9707
Total Segment data Spectral Kurtosis is, K=2.6382.
As, K<C+V;
According to the algorithm implemented the left segment window length, p and q values are as follows:
wl=256 and p=2 q=1 are obtained.

ii. This calculations are done for next 1024 i.e. for 1025 to 2048 samples of LFM signal Spectral Kurtosis which we obtain from the Matlab are:
Left Segment data Spectral Kurtosis is, C=0.6.
Right Segment data Spectral Kurtosis is, V=1.8037
Total Segment data Spectral Kurtosis is, K=2.0437.
As, K<C+V;
According to the algorithm implemented the left segment window length, p and q values are as follows:
wl=256 and p=2 q=1 are obtained.

iii. This calculations are done for next 1024 i.e. 2049 to 3072 samples of LFM signal Spectral
Kurtosis which we obtain from the Matlab are:
Left Segment data Spectral Kurtosis is, C=1.
Right Segment data Spectral Kurtosis is, V=1.4287
Total Segment data Spectral Kurtosis is, K=3.3082.
As, K>C+V;
According to the algorithm implemented the left segment window length and q values are as follows:
wl=512 and q=2 are obtained.

5. CONCLUSION

The system can be used for accurate parameter measurement of LFM signals even in hostile environments and the system designed is very compact. Taking all these aspects into consideration, it can be concluded that the Adaptive STFT that has been designed in MATLAB can be used widely and hence improvise the existing Electronic Warfare and simplifies the existing complex RADAR techniques to provide a simple and the best Electronic Protection of targets from enemy RADARs.

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AUTHOR'S PROFILE

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