Multirate digital signal processing Techniques in the receiver stage of RADARS

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Abstract : The aim of this paper is to study any analysis of latest Multirate Digital Signal Processing techniques for receiver of radar to include comparisons between latest techniques and earlier techniques with a view to improve target detection in radars. This includes understanding of practical applications of Multirate Digital Signal Processing and to compare some latest techniques such as FIR filters with IIR filters, poly phase filters, MRIMM, etc.

Key words: Moving target indicator (MTI), Analog-to-digital converter (ADC) and digital-to-analog converter (DAC)

I. INTRODUCTION

The exponential growth in digital technology along with the corresponding decrease in its cost has had a profound impact on the way radar systems are designed. More and more functions that historically were implemented in analog hardware are now being performed digitally, resulting in increased performance and flexibility and reduced size and cost.

Major advances in analog-to-digital converter and digital component technology have transformed the receiver front ends of radar systems, providing higher performance at lower cost. This dissertation will describe how these new technologies are being applied to radar systems and the benefits they bring to system performance. Receivers of military or even commercial radars will mostly have similar electronic subunits and assemblies. These subunits execute different signal processes on the signal where in the signal goes through different fundamental changes in frequency, phase and bandwidth or different sampling rates in case of digital signals.

Moving target indicator (MTI) radar has demonstrated its powerful surveillance/ reconnaissance capability in many military operations. Its ability to provide timely detailed information throughout the theatre is peerless and critical to real-time command and control in a battlefield. In the near future, a surveillance operation may include a network of MTI radars. With each MTI radar operating at several modes, including wide area search and sector search, this MTI radar network will provide an unprecedented amount of data regarding numerous targets in a surveillance region, from small units to major enemy forces.

This network of MTI radars can bring to its user a ‘totally new and unique kind of information whose military value and significance was equal to that of overhead imagery and signals intelligence’. As there are thousands of objects in a surveillance region, an enormous tracking effort is needed to correlate radar measurements into target trajectories. Meanwhile, under combat conditions, the measurements from multiple platforms will not be synchronized, because each individual platform has its own scan rate and the communication network cannot guarantee to deliver measurements on time. Asynchronized measurements inevitably make the order of measurements uncertain, which creates an out-of-sequence (OOS) reporting phenomenon. Whenever a signal at one rate has to be used by a system that expects a different rate, the rate has to be increased or decreased, and some processing is required to do so. “Mulirate DSP” refers to the art or science of changing sampling rates i.e. decimation and interpolation. The process is not only limited to up or down sampling but all the latest aspects which can be exploited to improve target detection.

There are numerous applications of Multirate Digital Signal Processing and some of them as mentioned in this paper are like Interface of digital filters with different sampling rates, implementation of Narrow band filters being possible by use of Multirate Digital Signal Processing, implementation of digital filter banks, integration of multiple target/subject data inputs sources having different sampling rate for latest target position update, reduction of white noise and other identified interferences, reduction of hardware load and cost, reduction of data crunching efforts.

II. RADAR RECEIVER

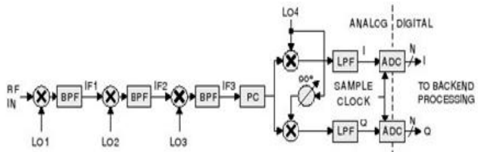


Figure 1 : Typical Radar Receiver Front End from 1990

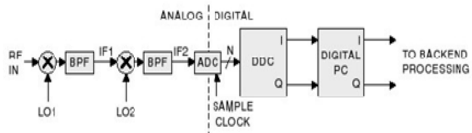


Figure 2 : Typical Digital Front End design

Radar is an object-detection system that uses radio waves to determine the range, angle, or velocity of objects.

It can be used to detect aircraft, ships, spacecraft, guided missiles, motor vehicles, weather formations, and terrain. A radar transmits radio waves or microwaves that reflect from any object in their path. A receive radar, which is typically the same system as the transmit radar, receives and processes these reflected waves to determine properties of the object(s) [3]. Advances in analog-to-digital converter (ADC) and digital-to-analog converter (DAC) technologies are pushing the border between analog and digital processing closer and closer to the antenna. For example, Figure 1 shows a simplified block diagram of the receiver front end of a typical radar system that would have been designed around 1990. Note that this system incorporated analog pulse compression (PC).It also included several stages of analog down conversion, in order to generate baseband in-phase (I) and quadrature, signals with a small enough bandwidth that the ADCs of the day could sample them. The digitized signals were then fed into digital Doppler/ MTI and detection processors.

By contrast, Figure 2 depicts a typical digital receiver for a radar front end. The RF input usually passes through one or two stages of analog down conversion to generate an Intermediate Frequency (IF) signal that is sampled directly by the ADC. A digital down converter (DDC) converts the digitized signal samples to complex form at a lower rate for passing through a digital pulse compressor to backend processing. Note that the output of the ADC has a slash through the digital signal line with a letter above. The letter depicts the number of bits in the digitized input signal and represents the maximum possible dynamic range of the ADC. The use of digital signal processing (DSP) can often improve the dynamic range, stability, and overall performance of the system, while reducing size and cost, compared to the analog approach.

III. DIGITAL SIGNAL PROCESSORS

Digital signal processors are sampled signal systems. Sampling is the process by which a continuous (analog) signal is measured at regular intervals of time (the sampling interval), producing a sequence of discrete numbers (samples) that represents the values of the signal at the sampling instants. The sampling frequency is the inverse of the sampling interval and is typically designated fs. Sampled systems are subject to the Nyquist limit, which lower bounds the sampling rate at which reconstruction of the unsampled signal from its samples is possible without corruption by aliasing, the overlapping of spectral components.

The bound, termed the Nyquist frequency or Nyquist rate, is equal to the two-sided signal bandwidth B, the bandwidth considering components at both positive and negative frequencies. The Nyquist rate is often said to be twice the signal bandwidth, but that refers to a one-sided bandwidth, positive frequencies only, of a real signal. Our definition refers to the two-sided bandwidth, both positive and negative frequencies, of a signal that, in general, is complex with a real signal as a special case.

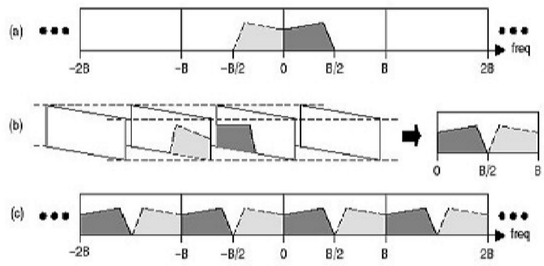


Figure 3 (a)Bandlimited real signal spectrum before sampling, (b) Portion of sampled spectrum from o to B, (c) Full sampled signal spectrum.

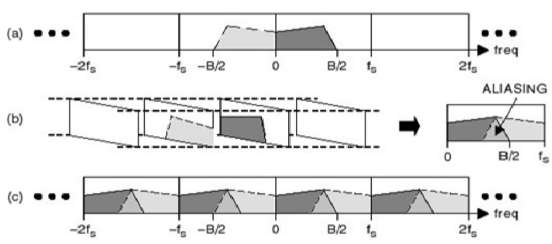


Figure 4 (a)Bandlimited low pass signal spectrum before sampling, (b) alaised low pass signal spectrum after sampling rate fr>B, (c) alaised sampled signal spectrum.

The following figures illustrate the origin of the Nyquist rate. Imagine that a real signal with a low pass signal spectrum of two-sided bandwidth B is plotted on a long piece of paper, as shown in Figure 3a. In the figure, the positive-frequency spectral components of the signal are darkly shaded, and the negative-frequency components are lightly shaded. To see the effect of sampling this signal at Nyquist rate B, the long sheet is cut into smaller sheets, with the first cut at zero frequency and subsequent cuts at sample-rate (B, in this case) intervals in positive and negative frequency. The sheets are stacked one on top of the other as shown on the left side of Figure 3b, and the resulting portion of the sampled signal spectrum from 0 to the sampling rate of B is generated by adding the spectra of the stacked pages together, as shown on the right.

Note that the lightly shaded negative-frequency portion of the spectrum now appears on the right of the sampled spectrum and doesn't overlap the darker positive-frequency portion. As long as the two portions of the sampled signal don't overlap, the signal is not aliased. The full sampled-signal spectrum is obtained by laying copies of this page end-to-end, as shown in Figure 3c, producing copies of the 0 to B portion of the sampled signal spectrum at B intervals. Figure 4 shows the result of sampling below the Nyquist rate. Figure 4a shows the same band limited signal as the previous example, but this time it is sampled at some rate that is less than Nyquist rate B. The resulting sampled spectrum, shown in Figure 4b and c, contains overlapped, or aliased, spectral components that add and represent corruption of the signal.

Sampling Rate conversion can be done by:

(a) D/A conversion followed by A/D conversion at a different rate –(i) Advantage: arbitrary sampling rate

(ii) Disadvantage: distortion during A/D and quantization noise during D/A;

(b) Sampling rate conversion in digital domain

IV. MULTIRATE TERMINOLOGIES AND SAMPLING RATE CONVERSION

. Multirate Terminologies.

(a) Decimation i.e Downsampling

(b) Interpolation i.e Upsampling

Sampling rate conversion can be viewed as a linear operation:

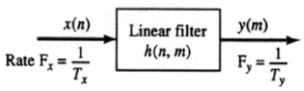


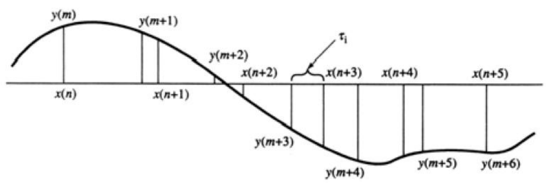
Figure 5 Sampling Rate Conversion

Tx and Ty are the corresponding sampling intervals for x(n) and y(m)

Constrain;
Fy = I
FX D

Where I and D are the integer up sampling and down sampling factors.

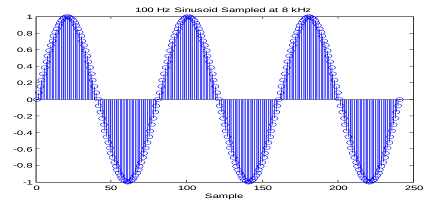
Sampling rate conversion can also be viewed as resembling of the same analog signal.



Down-Sampling a Sinusoid Wave (MATLAB).

Program Instructions.

```
>> [ysin,tsin]=analog(100,1,30,8000);

>> stem(ysin),title('100 Hz Sinusoid Sampled at 8 kHz')

>> xlabel('Sample')

>> dysin=downsample(ysin,4);

>> figure,stem(dysin);

>> title('100 Hz Sinusoid Sampled at 8 kHz, Down-Sampled by 4x')

>> xlabel('Sample')
```



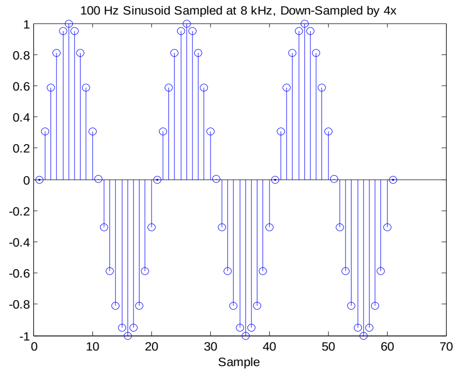
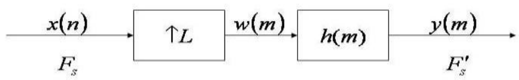


Figure 6.Down-Sampling Example to Include Solution of Aliasing (Using MATLAB).

V. MULTISTAGE IMPLEMENTATION

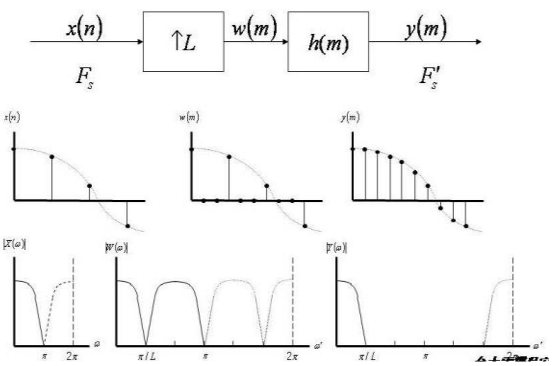
Block Diagram representation of multistage implementation.



$$F_s' = LF_s$$

$$\frac{T'}{T} = \frac{1}{L} = \frac{F_s}{F_s'}$$

$$H_I(\omega') = \begin{cases} G, & |\omega'| < \frac{\pi}{L} \\ 0, & \text{otherwise} \end{cases}$$



Conversion by a Rational Factor M/L

$$\frac{T'}{T}=\frac{M}{L}=\frac{F_s}{F_s'}$$

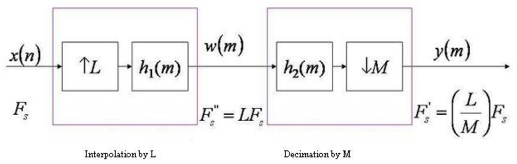
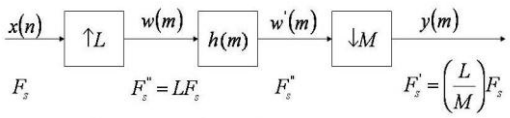


Figure 7 : Cascade Processes (Inerploation and Decimation)

Cascade of Two Processes

A more efficiency implementation;



$$H_I(\omega'')=\begin{cases} L, & |\omega''|\leq \min\left|\frac{\pi}{L},\frac{\pi}{M}\right| \\ 0, & \text{otherwise} \end{cases}$$

Multirate Digital signal processing

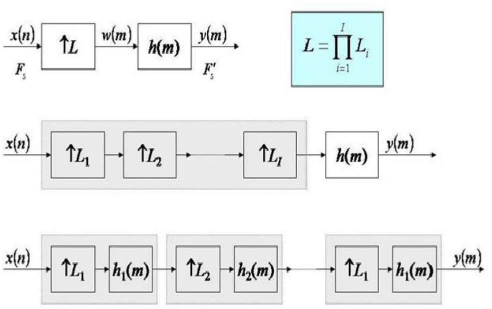


Figure 8: Universal block diagram for multistage receiver

(a) Advantages

(i) Reduce the complexity

(ii) Reduce storage devices (registers)

(iii) Simplify (relax) filter design problem

(iv) Reduce the finite word-length effect

(b) Disadvantages

(i) Increase the control circuit

(ii) Difficulty in choosing I and best L_j for $1\leq i\leq I$

The spectral content of a digital signal is replicated at integer multiples of the sampling frequency. Oversampling (sampling well beyond the Nyquist rate) spreads the frequency content over a wider frequency range The requirements on the analog anti-aliasing filter can be relaxed by oversampling followed by digital low-pass filtering.

The extra samples in an oversampled and bandwidth-limited signal can be removed by downsampling or decimation, thereby reducing the effective sampling frequency. A signal can be restored to a higher sampling frequency by the processes of upsampling or interpolation . Since Gaussian noise is uniformly distributed in the frequency domain, the combination of oversampling and filtering can effectively de-noise a signal in a bandwidth of interest. Oversampling can be maximized by delta-sigma quantization, which is quantization with one bit. Delta-sigma quantizers have the property of noise-shaping, which allows the elimination of quantization noise by low-pass filtering.

VI. CONCLUSION

Major advances in analog-to-digital converter and digital component technology have transformed the receiver front ends of radar systems, providing higher performance at lower cost. New technologies are being applied to radar systems and the benefits they bring to system performance.

Receivers of military or even commercial radars mostly have similar electronic subunits and assemblies. These subunits execute different signal processes on the signal where in the signal goes through different fundamental changes in frequency, phase and bandwidth or different sampling rates in case of digital signals.

Multirate DSP not only limited to up or down sampling but all the latest aspects which can be exploited to improve target detection in radars. Hence in most applications Multirate Digital Signal Processing systems are used to improve the performance, or for increased computational efficiency which in a way improves the most important part of a radar which is receiver which is ultimately required to have faster detection and efficient tracking.

REFERENCES

[1] Website: www.dspguru.com as on 15 Nov 2013

[2] Book: Digital Signal Processing using MATLAB by Vinay K Ingle and John G Proakis.

[3] Book: Digital Signal Processing using MATLAB and Wavelets by Machael Weeks.

[4] Book: Radar Digital Signal Processing by James J. Alter and Jeffrey O. Coleman,Naval Research Laboratory

[5] Book: Signal Analysis :Wavelets, Filter Banks, Time-Frequency Transforms and Applications by Alfred Mertins, University of Wollongong, Australia

[6] IEEE paper on ARMA Processes in Multirate Filter Banks with Applications to Radar Signal Classification by Kie B. Eom and Rama Chellappa.

[7] IEEE paper on MRIMM and OOS by L. Hong, S.Cong and D. Wicker.

[8] IEEE paper on Engineering Multirate Convolutions for Radar Imagaing by Laurens Bierens.

[9] IEEE paper on Modeling of Multirate Signal in Radar Target Recognition by LIU Yong-Xiang, LI xiang and ZHUANG Zhao-wen.

[10] IEEE paper on Extracting In-Phase and Quadrature Signal Components from a Bandlimited Real Signal Using a Half Band Multirate Filter Design and its Implementation by Phillip D West and Mark D Austin.