LOCALIZATION OF SMALL FAST MOVING OBJECTS USING DOPPLER RADAR

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ABSTRACT

This paper addresses issues linked with design of a Doppler radar localization system. At first a proposed design is presented, then hardware requirements are discussed. Detailed description is devoted to logarithmic amplifier theory as it plays crucial part in the final design. The last part describes a time-frequency analysis that is essential for further signal processing.

1 INTRODUCTION

The aim of the project is to design a system capable of detecting speed and position of very small objects moving fast at distances ranging from several centimetres up to few hundred meters by use of a Doppler radar. Prerequisite for such system is to take in account that its design should be made with respect to minimize noise which plays a crucial part for sake of subsequent signal processing, e.g. sampling of the signal will be cause of spectrum periodization and sum the noise that is present. The design should also ensure wide dynamic range of the amplifier, considering relatively high amount of energy in rebound signal when the object is close to the Doppler head aperture, and make sure to amplify signal enough when the object is at greater distance.

2 HARDWARE DESIGN

A block schematics of the localization system is depicted in Figure 1. It consists of a Doppler sensor head, a low noise preamplifier followed by a passband filter further reducing the noise bandwidth. A wide dynamic range is achieved by use of a logarithmic amplifier. It is possible to achieve dynamic range which exceeds 120 dB with existing circuit. The last stage of the localization is formed by a simple impedance matching circuit.
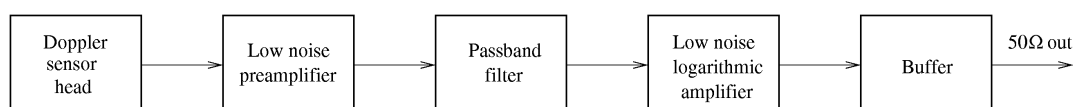
Figure 1: *A block scheme of the localization system*

2.1 LOGARITHMIC AMPLIFIER THEORY

In comparison with classical linear amplifiers logarithmic amplifiers perform more complex operation and their circuitry is significantly different. A logarithmic amplifier utilizes its amplification to achieve a function rather than to provide amplification of the signal. A wide dynamic range signal is compressed to its decibel equivalent. Due to its nature of converting the signal from one domain of representation to antother via a precise nonlinear transformation would be better to label a logarithmic amplifier as a logarithmic converter.

Logarithmic compression can lead to paradoxical or confusing situations. For instance, it can be confusing to add a voltage offset to the output of the logarithmic amplifier as it can be mistakenly regarded as a gain increase at its input. Usually when all the variables are voltages regardles of the particular structure, the relationshop between the variables can be expressed as:

$$V_{out} = K_y \cdot \log \left| \frac{V_{in}}{K_x} \right| \tag{1}$$
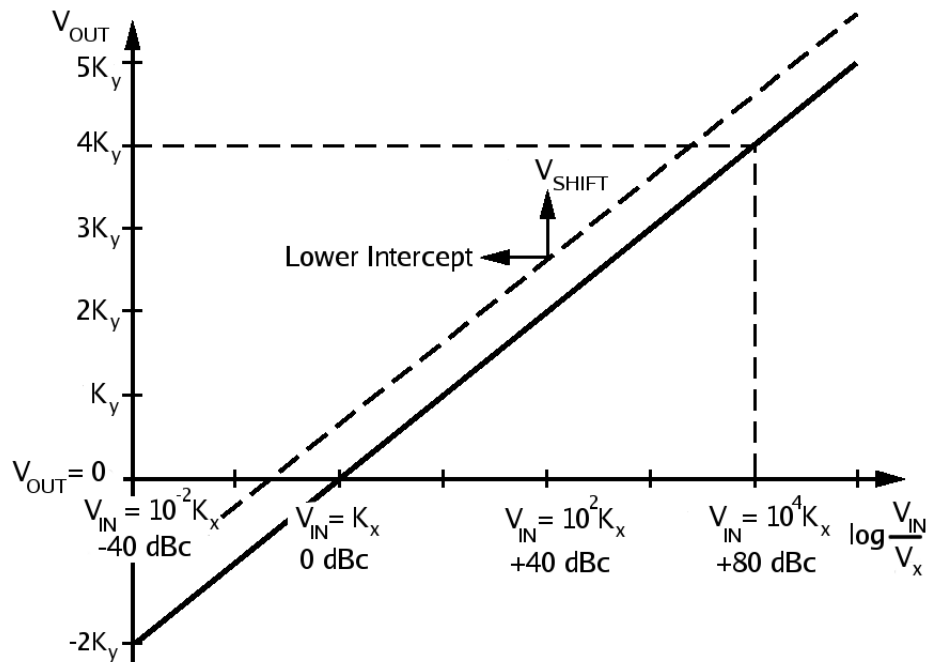
where:

V_{out} is the output voltage.

K_y is the slope voltage; the logarithm is usually taken to base 10.

V_{in} is the input voltage.

K_x is the intercept voltage.

The scaling of the circuit is implicitly determined by two references, here referred as K_x and K_y. They determine the absolute accuracy of a logarithmic amplifier.

Figure 2: *Ideal logarithmic amplifier function*

Ideal logarithmic amplifier input/output that conforms to (1) is depicted in Figure 2. The horizontal scale is logarithmic and spans over a wide dynamic range. The output traverses through zero at the logarithmic intercept point, where $V_{in} = K_x$, and ideally becomes negative for inputs below the intercept. In the ideal case, the function describing V_{out} for all values of V_{in} continues endlessly in both directions. The effect of adding an offset voltage V_{shift} to the output is shown

by the dotted line. The effective intercept voltage K_x is then lowered. The same result can be achieved by raising the gain or signal level ahead of the amplifier by the factor V_{shift}/K_y.

The logarithmic amplifer function is described by (1) and compared to the one of a linear amplifier the main difference is that the incremental gain $\delta V_{out}/\delta V_{in}$ is a very strong function of the momentary value of V_{in} as it is obvious by computing its derivative. For the case where the logarithmic base is δ.

$$\frac{\delta V_{out}}{\delta V_{in}} = \frac{K_y}{V_{in}} \tag{2}$$

That is, momentary value of the input voltage is inversely proportional to the incremental gain. This is true for any logarithmic base, which is chosen as 10 for all decibel related purposes. It follows that at a small signal conditions an infinite gain amplifier is required. This result indicates that whatever means are used to implement a logarithmic amplifier, accurate response under small signal conditions demands the provision of a very high gain bandwidth product. The consequence of the high gain is that, in the absence of an input signal, even very small amounts of thermal noise at the input of a logarithmic amplifier cause a finite output for zero input. This results in the response line curving away from ideal shown in Figure 2 toward a finite baseline, which can be either above or below the intercept.

A simpler formula is appropriate for specifying calibration attributes of a logarithmic amplifier though Equation 1 is fundamentally correct:

$$V_{out} = V_{slope}(P_{in} - P_o) \tag{3}$$

where:

V_{out} is the demodulated and filtered baseband output.

V_{slope} is the logarithmic slope, now expressed in V/dB.

P_{in} is the input power, expressed in decibels relative to some reference power level P_o is the logarithmic intercept, expressed in decibels relative to the same reference level.

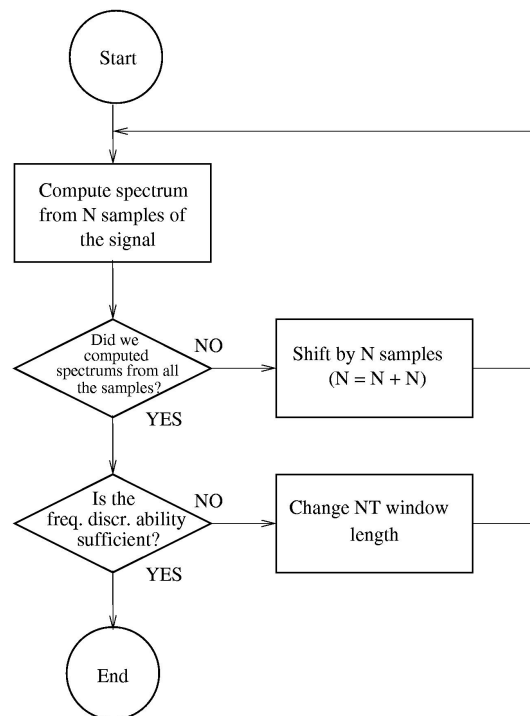
3   TIME-FREQUENCY ANALYSIS

During analysis of the signals that are changing its nature quickly is essential to take in account the frequency contents of the short signal segments. That means unwrapping the concept of short-term spectrum enabling us to define spectrum as a two dimensional function dependent on frequency and position in time. This concept is entitled as a time-frequency analysis.

Although integral Fourier transform in theoretical representation is dedicated to operate on signals of infinite length, practical analysis comes out with finite segments of the signal limited by type of window used. This type of approach could be used for time-frequency analysis if the window is of suitable length and is set as sliding on time axis. While use of classical frequency analysis is limited to the amplitude and phase search of harmonic components of various frequencies and infinite duration, the time-frequency approach requires the amplitude and phase to be found as precisely as possible.

Different approach between previously mentioned methods asks for different methodology for designing the observational periods. When performing classic analysis, it is reasonable to extend the period in order to minimize distortion, while during time-frequency analysis the observation period is given as a compromise between differentiation ability in the frequency domain (i.e. distinguishable frequency difference is inversely proportional to the window length) and

effort for the highest resolution in time domain as possible (i.e. minimum distinguishable time difference is equal to the window length)

One of the above mentioned requirements is dominant in a particular application and the window length needs to be adjusted to it. Resulting from the use of discrete Fourier transform upon such fixed sections of the signal, short-term spectrums will develop. They characterize frequency contents and phase relationships of the signal with the corresponding time segment. Notice that values of the short-term spectrum conform to the whole time segment of the window and it is not possible to distinguish if some signal development took place inside the period. This pinpoints the discrimination ability, which is approximately N-ary part of the range $< -\omega_s/2, \omega_s/2 >$ and is constant over the range of processed frequencies. Flow chart shown in Figure 3 illustrates whole process of the time-frequency analysis.

Figure 3: *Flow chart showing progress of the time-frequency analysis*

Short-term spectrums are usually computed from longer segments of the signal where the high frequency signal evolution is essential to observe. In the simplest case is possible to use sequences of the signal samples of length N, take whole sampled signal of length M and obtain int(M/N) of the spectrums. The frequency discrimination ability is given by the window length NT and it also specifies the smallest distinguishable difference in time. The discrimination ability can be increased further by overlapping the particular windows, e.g. by half of their size. As a result a reasonable number of the spectrums will appear along the time axis enabling us to trace possibly quick signal evolution, especially of high frequency components. Such a set of spectrums, labelled as spectrogram, can be clearly depicted as a two dimensional chart, where the first coordinate corresponds to the frequency, whereas the second one represents the time. The amplitude of the corresponding spectrum coefficients is distinguished by a color.

Disadvantage of discrete Fourier transform is unveiled when a plain spectrogram is used because its absolute frequency discrimination ability is constant. The window length is determined by required frequency discrimination ability for the low frequencies and it limits discrimination in

time. But for practical applications would be more useful to have relative discrimination ability constant, i.e. having distinguishable frequency interval as a certain fraction of the frequency. Then, on the other hand, absolute frequency discrimination is sufficient smaller for high frequencies utilizing shorter window enabling detailed time resolution for high frequencies. For our purpose, one detected frequency, not too much changing over the time, is enough. As a result we do not have to pay attention to above mentioned problem.

4 CONCLUSION

Detecting of small objects results in small amplitude of rebound signal, especially at greater distances from the Doppler head, where the signal to noise ratio drops to several decibels. Because of the signal sampling is followed by a spectrum periodization and summation of the noise it is convenient to design a system that has minimal intrinsic noise. This is achieved by selecting ultra-low noise active components. For example preamplifier's intrinsic noise is $1.3 \mathrm{n} V / \sqrt{\mathrm{Hz}}$ which is comparable to 100 ohm resistor's intrinsic noise of $1.27 \mathrm{n} V / \sqrt{\mathrm{Hz}}$.

The aim of the project is to find trajectory of moving object by use of multiple Doppler sensor heads. Samples of the signal from sensor heads will be collected by embedded computer, which will compute the trajectory.

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