DETERMINATION OF ELECTRONIC WARFARE RECEIVER'S INSTANTANEOUS DYNAMIC RANGE USING MUSIC METHOD

L. L. Liou, D. M Lin and J. B. Tsui,
Sensor Directorate, Air Force Research Laboratory
Wright-Patterson AFB, OH 45433

ABSTRACT

The purpose of this paper is to find the limit of instantaneous dynamic range (IDR) for electronic warfare (EW) receivers with two-signal detection capability. In order to make the limit independent of the order of music method, very high order is used. Two signal detection criterions are used. One criterion is through the eigenvalue distribution. The other is through the frequency identification using the music spectral analysis. General behavior of IDR vs. frequency shows a small IDR at small frequency separation and that IDR reaches a saturation value at some frequency separation. This is the expected result. In order to apply to practical EW receiver design, analog-to-digital converters (ADCs) with various numbers of bits are used to calculate the IDR. The calculated results for ADC with different number of bits are curve fitted to represent the IDR vs. frequency separation. These curves can be used as the limit for digital EW receiver designs.

INTRODUCTION

An electronic warfare (EW) receiver is a unique type of receiver that measures non-cooperative signals. Namely, EW receivers are required to measure time coincident signals without apriority information, a requirement that most communication receivers do not have. Let us refer to time coincident pulses as simultaneous signals. It is very desirable for EW receiver to detect multiple simultaneous signals. Some receivers are designed to receive up to four simultaneous signals, a considerably large number. The parameters used to evaluate this receiver will include the number of the detected signals and their relative strength, namely the dynamic range (DR). In the following paragraphs we give the general description of DR, and its implication to EW receivers design for single frequency detection and two simultaneous signal detections.

General description of DR: A DR is determined by two limits: the lower limit and the upper one. The lower limit is the weakest signal that can be detected. In some definitions of DR, the sensitivity level of the receiver is often the lower limit. The sensitivity is defined as an input signal that is strong enough to break a certain threshold, usually the background noise. The upper limit is the strongest signal that the receiver can measure without error or spurious outputs. If a receiver measures the amplitudes of the input signals, its capability of measuring the weakest to the strongest signals may be considered the DR. Under this situation, the maximum DR is usually about 6b dB, where b is the number of bits of the ADC. Sometimes frequency measurement capability is used as the criterion to determine the DR. The weakest input signal whose frequency can be encoded correctly (e.g., 90% of the time with 10^{-7} probability of false alarm) is used as the lower DR limit in this paper. A strong signal can drive some components in the receiver into saturation. When this happens, spurious signals may be generated. The level at which a strong signal causes the receiver to report a spurious signal is considered to be the upper limit. The DR in this case is the difference between these two limits. For example, if the lower limit is -60dBm, and the upper limit is +10dBm, the DR is 70 dB.

Single Signal Detection DR: There are two types of receivers for this discussion. If the receiver is designed for single signal detection only, the receiver can only encode one signal, and it will not produce false alarms on a second frequency. A very high DR can be anticipated. If the receiver is designed to process multiple simultaneous signals, then the single signal DR is limited by the type of ADC and the post detection circuitry following the FFT outputs. When the ADC is driven into saturation, the output will generate harmonics due to the spurious responses. If these harmonics are higher than a certain threshold, the receiver will consider them as signals, and false alarms occur. The strong signal that generates such spurs will be the upper limit of the DR of the receiver. In general, receiver designed to process multiple signals, has a lower single signal detection DR than that of the receiver designed to process only one signal.

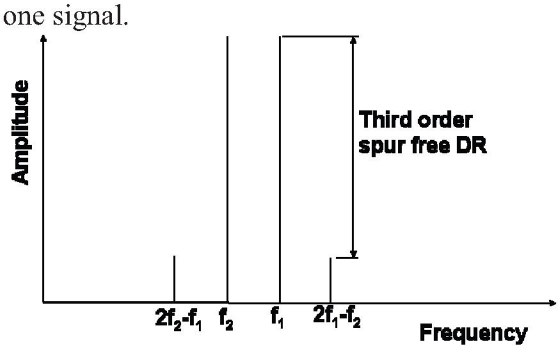


Figure 1: Two signal third order spur free DR

Two Signal Spur Free DR: There are two definitions of two-signal DRs. One is referred to as the spur free DR (or, third order spur free DR), and the other is referred to as the instantaneous DR (IDR). The spur free DR is defined as in Figure 1. Two frequencies f_1 and f_2 are drawn in the frequency domain and they have the same amplitude. If these two signals are strong enough to saturate some components in the receiver, third order spurious responses will be generated at $2f_1$-f_2 and $2f_2$-f_1. In a receiver measurement, one injects two signals of the same amplitude and separate them in frequency such that the two, third order spurs are also in the input band of the receiver. If the signals amplitude increases, then the third order spurs can be detected as false signals. When this false signals start to show, the input signal power level is considered as the upper limit of the DR. The concept of third order spurs is very important in receiver design. The amplification required before the ADC in an EW receiver is determined by the third order spur level. In an optimum design, when the spur level equals to the noise floor, the input power level should be about 6 dB below the ADC saturation level. The two signal spur free DR is a more of a receiver design criterion than a receiver performance criterion. This is because of its definition that requires two signals of the same amplitude and their summation must be close to the ADC saturation level to generate 3rd order spurs. This condition rarely occurs under real operational situation. Therefore, this parameter will not be discussed further in this paper.

Two Signal Instantaneous DR: The two signal instantaneous DR (IDR) is a very important parameter in measuring the performance of a receiver designed for multiple signal detection. It is the measure of the maximum difference between the strong and weak signals that can be simultaneously detected. Reference 2 studied the theoretical bound of the IDR using Cramer-Rao (CR) bound approach [2]. The variances of the frequency, amplitude and phase measurements are used to evaluate the CR bound. It shows that the CR bound of each signal depends on the frequency separation, the phase difference between the two signals and their S/N. In general, when two signals are close in frequency, the variance has a large value indicating a low IDR. When the frequency separation increases, the variance decreases indicating that IDR increases. As frequency separation keeps increasing, the CR bound reaches an asymptotic constant, so does the IDR. This constant depends only on the S/N of the measured signal [2]. This parameter is the main focus of this paper.

Prerequisite for DR Measurements: When EW receiver's performance is evaluated, it is important to isolate the receiver system from the measurement system, so that the measured DR does not depend on the measurement method. Controversial results often occurred when the same EW receiver were measured with a very different DR data. The difference can be of several tens of dB. Modern receivers are able to generate a pulse descriptor word (PDW) for every signal event so that it can be autonomous in determining signals without the interruption from outside the receiver system. Without a PDW, the DR measurement often involves the observation from an oscilloscope with human interpretation. The "process gain" due to the "human-in-the-loop" process may artificially increases the DR substantially. In reality, a “receiver” without a PDW encoder cannot be considered a complete receiver. For these receivers, the DR measurements are meaningless.

It is therefore, important to have a theoretical estimation of DR, against which a measurement can be compared. This paper presents numerical results of two-signal IDR. Section II describes the methods used in the calculation. The method is based on MUltiple SIgnal Classification (music) and its eigenvalue distribution. Section III presents the calculation results. The first part in this section presents the results of the digitized analog signal without an ADC. The second part of the calculation involves the digitization process of ADC. The practical digital receiver design parameters such as gain, ADC's saturation voltage, and bit number were used in the calculation. The calculated IDR as functions of frequency separation and the bit number of ADC was presented. The purpose of these results is to provide a practical guideline for the IDR measurements of digital EW receivers.

IDR CALCULATION USING MUSIC AND EIGENVALUE DISTRIBUTION METHOD

Defining problem: In an EW receiver, the number of signals is usually unknown. In this study, we limit the number of signal to be two. A data set contains 256 time series data points. This is a typical number of data point used in the EW receiver's data block process. If other number of data points is desired, the same approach can be used to determine the IDR. The input contains two continuous wave (cw) signals embedded in Gaussian noise. The noise in the input signal is a function of receiver input bandwidth. We use real data as the input data, and the sampling frequency is 2.56 GHz. The bandwidth is therefore, 1.28 GHz, and the noise floor is about -83 dBm. Assuming that the receiver has a noise figure of 3 dB, the overall noise floor is at -80 dBm. The input condition of the two signals under this study includes their relative amplitudes and frequency separation.

In the first part of the calculation, the input data are digitized without the accuracy restriction imposed by an ADC. In the second part of the calculation, the input data are digitized by an ADC with different number of bits.

Music Method Approach: The accuracy using Music analysis depends on proper choice of two important parameters. One is the number of signal. If this number is larger than that of the real one, the music process will produce spurious outputs [2]. If the number is smaller than that of the real one, it simply misses signals. It is, therefore, important to determine the correct number of the signals from the measured data using Music method. In the next subsection, we discuss the possibility using the eigenvalue distribution of the autocorrelation matrix to determine the number of signals.

The other important parameter is the order of the process. This number determines the final dimension of the autocorrelation matrix. Let the current problem described above as an example. If a low order such as 5 or 6 is used, the computation time is short. However, it does not provide enough resolution to resolve two signals separated by a small frequency, e.g., 1 MHz. If an order between 80 to 100 is used, Music method provides enough resolution to resolve two signals with 1 MHz frequency separation. However, the computation time is long.

In this study, we use an order of 80 to determine the limits of amplitude and frequency separation. This is because that the general rule for the order is about one third to one half of the input data points of 256 [2]. We verify this rule with some numerical experiments. If order of 10, 20 and 40 were used, it did not provide satisfactory resolution. If order of 80 and 100 were used, the results are similar, and both provide enough resolution.

Eigenvalue Distribution generated with noise and noise plus signals: The autocorrelation with an order of 80 resulted in a matrix with a dimension of 80 by 80. If the complex time series data were used (i.e., the I Q channel data), the matrix is a Hermitian and there are 80 eigenvalues of real numbers. These eigenvalues are sorted according to their values and are separated into two groups. One is the signal group, and the other is the noise group. The signal group contains the largest p eigenvalues, where p is the number of the input signals. The noise group contains the rest of the eigenvalues. Since the input of this study is limited to two signals, only the three largest eigenvalues (two correspond to the signal and one corresponds to noise) will be evaluated. From these three, we may be able to determine the number of the signals. If the input time series is real, the corresponding eigenvalues are in pairs. For example, if there is only one input signal, there will be two large eigenvalues. The three eigenvalues to be considered are the first, third and fifth largest ones.

We need to calculate the eigenvalue distribution in order to set up threshold. The first test is to use noise only in the input signal. The eigenvalue distribution is obtained from 1000 runs. For each run, one random Gaussian noise of 256 points is generated. The real input data were used in this study, therefore, the first, third and fifth eigenvalues were evaluated, and they are referred to as the largest three eigenvalues. The autocorrelation matrix, thus generated, belongs to a classification known as the random matrix [3,4]. The eigenvalues of the resulting matrix are thus random numbers with certain probability density functions (pdf). The numerical simulation results show that these pdfs are skew-Gaussian-like. Table I lists the first two moments of the distributions. It appears that the higher the means the higher the standard deviations are. The pdf of the eigenvalues will be discussed further in the future paper.

For the second test, an idealistically strong signal-to-noise ratio (S/N) = 100 dB is used as input. Under normal condition, the input signal may not be this high. There are two reasons for this strong signal. One is to minimize the noise effect. If the signal is not strong enough, the detection of the weaker signal might be limited by noise. The other reason is to check the signal effect on all three largest eigenvalues. The results are obtained from 1000 runs with random Gaussian noise. Their first two moments of the eigenvalue distribution are listed in Table 2. It is seen that the strong signal only affects the largest eigenvalue which is a desired result. If one signal affects more than one eigenvalue, it is difficult to use this method to determine the number of the signals.

The final test uses two strong signals. Both signals are 100 dB above noise. The results are shown in Table 3. The first signal is randomly selected between 10 and 1270 MHz and the second signal is 2 MHz below the first one. The first two largest eigenvalues are quite different and they differ by more than two orders in magnitude. The important result here is that the third eigenvalue stays about the same value as those small eigenvalues in Table 1 and 2. These results indicate the feasibility to determine the number of signals from the eigenvalue distribution.

RESULTS AND DISCUSSION

Instantaneous dynamic range determination through eigenvalues: In order to determine the IDR through the eigenvalues, thresholds are needed. We choose the threshold to be the mean value plus 10 times the standard deviation. This approach with a high threshold, should generate low false alarm. The thresholds for the three largest eigenvalues are [6.45 4.23 3.54]. Comparing with the mean values in Table 3 they are quite large. To calculate IDR, the weak signal is gradually decreasing from the same magnitude as the strong one. At each decreased value, eigenvalues are calculated. IDR is recorded when the calculated 2nd largest eigenvalue is smaller than the threshold. The final IDR is obtained from an average of 100 run with different noise distributions. The strong signal is 100 dB above noise and the frequency is randomly selected between 60 to 1270 MHz. The higher frequency limit is set to exclude the upper edge band of 10 MHz. The lower frequency limit is set so that when the weaker signal is 50 MHz below the strong one, it excludes the lower edge band of 10 MHz. Figure 2 shows the IDR vs. frequency separation between the two signals. It is seen that when two signals are separated by only 1 MHz, the IDR is over 60 dB. This result is far beyond the fast Fourier transform (FFT) capability, which is limited by the frequency bin resolution (10 MHz in this case).

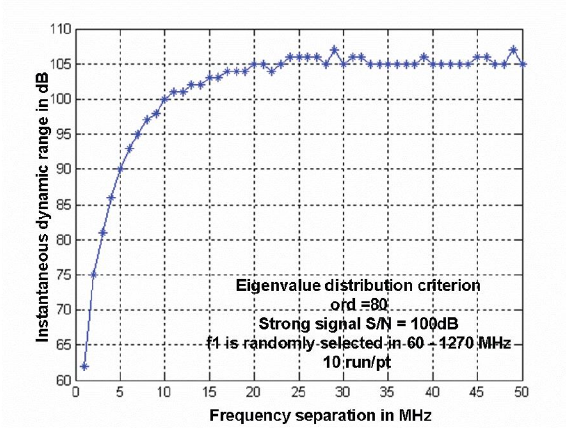


Figure 2: Two signal IDR obtained with 80 order music method using eigenvalues as threshold.

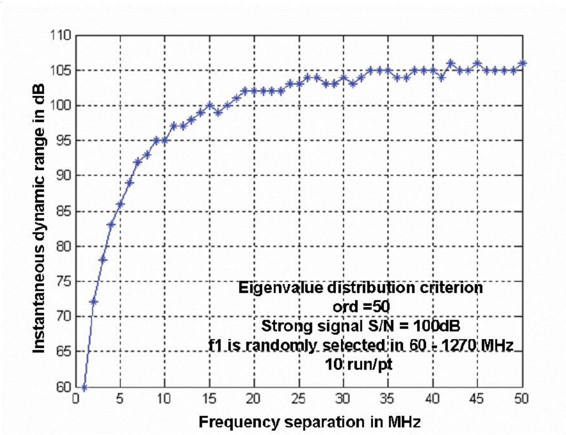


Figure 3: Two signal IDR obtained with 50 order music method using eigenvalues as threshold.

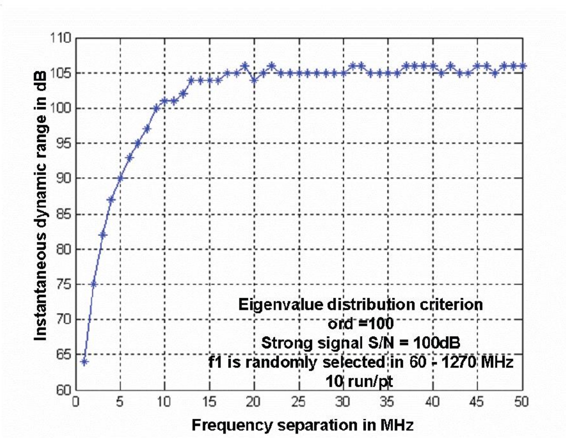
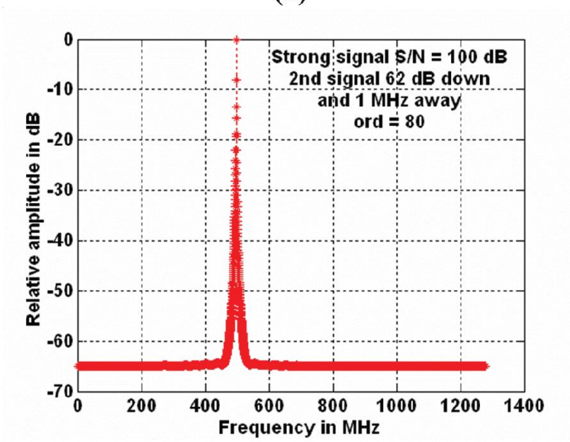


Figure 4: Two signal IDR obtained with 100 order music method using eigenvalues as threshold.

In order to test the effect of the order on the IDR, we repeat this calculation using orders of 50 and 100. The thresholds for determining the number of signals must be generated for these two cases. The eigenvalue distribution parameters for the noise-only input are shown in Tables 4 and 5, for orders of 50 and 100, respectively. The IDR vs. frequency separation results are shown in Figures 3 and 4 for order of 50 and 100, respectively. These results are similar to those in Figures 2. It shows that the IDR reaches a constant value of about 105 dB when the frequency separation is about 20 MHz. In Figure 3, the IDR reaches the same constant value at somewhere between 30 to 35 MHz. In Figure 4, the IDR reaches the same constant value at somewhere between 15 to 20 MHz. The constant IDR of 105 dB indicates that the detectable weak signal is about -5 dB below the noise. The difference between order 80 and 100 is small. This study uses the order of 80 as the default order for the following studies.

Instantaneous dynamic range determined by Frequency identification: The IDR obtained through the eigenvalues in the previous subsection are very high. For example, in Figure 2, the IDR is 62 dB when the two signals are separated by 1 MHz. It is desired to characterize these two signal using music spectral analysis. In order to resolve two signals separated by 1 MHz, a frequency step of 0.25 MHz is used in music spectral calculation. Since the total frequency coverage is 1280 MHz, the total output data are 5120 (4×1280). The result of the music-defined power spectrum is shown in Fig. 5. Figure 5a shows the overall frequency plot, which appears containing only one peak. Figure 5b is a close up plot of the peak and it also shows only one frequency. This means that although the eigenvalues indicate the presence of two signals, the music spectral analysis can not identify both of them. In order to find two signals separated by 1 MHz, the amplitude separation must be decreased. The new definition of IDR used in this section is that when two signals can be identified by their frequencies. We define that the frequencies are identifiable if the spectral power density has a minimum between two maxima, and the difference between the minimum and any of the maxima is larger than 0.1 dB. With this new definition, the IDR is found to be reduced to 39 dB when the frequency separation is 1 MHz. Under this condition, a close-up music-defined power spectral is shown in Figure 6 in which two peaks separated by 1 MHz is clearly shown. It should be noted that the peaks generated by the music method do not represent amplitude of the signals and they only represent the position of the signal.

(a)



(b)

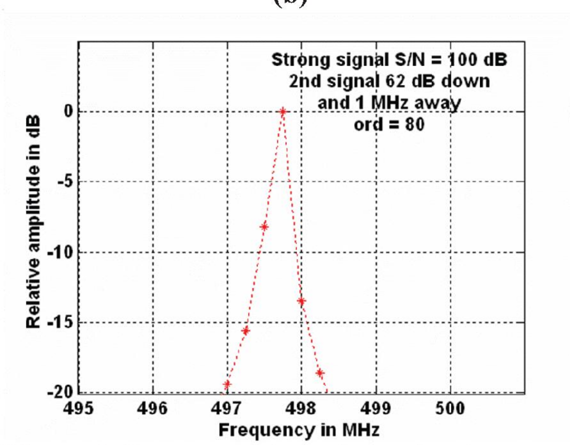


Figure 5: Frequency plots of two signals separated by 1 MHz and 62 dB a) Over entire frequency range b) frequency close to signals

Using the frequency reading capability of the music method, the newly defined IDR vs. frequency separation is shown in Figure 7. The data shown in this figure is the result obtained through an average of 100 runs with different noise distributions. The newly defined IDR is about 24 dB (62-38) smaller at 1MHz frequency separation and 18 dB (75-57) smaller at 2 MHz frequency separation, compared to those shown in Figure 2. A saturation occurs at 20 MHz which is similar to those shown in Figure 2. The saturation IDR is 100 dB which is about 5 dB lower than that in Figure 2.

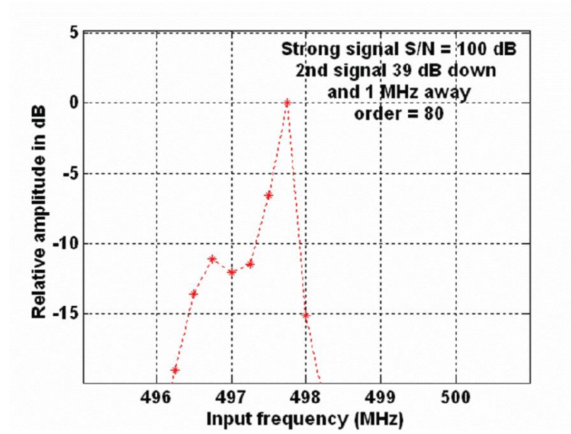


Figure 6: Frequency plots of two signals separated by 1 MHz and 39 dB a) Over entire frequency range b) frequency close to signals

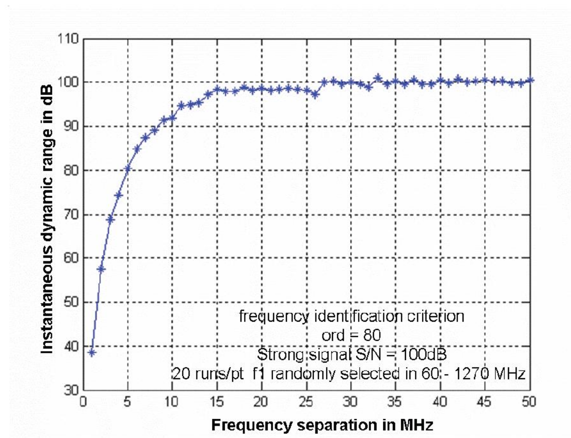


Figure 7: Two signal IDR obtained with 80 order music method using frequency identification

The IDR using eigenvalue to determine the number of the input signals depends on the threshold which is arbitrarily chosen. One would expect the IDR to alter if different threshold is used. The IDR using music spectral analysis to identify two input frequencies correctly is closed to a practical situation. This latter definition will be used in future discussion.

Introducing digitization in the instantaneous dynamic range calculation: The previous subsection shows the IDR results based on the idealistic input signal which is not affected by ADC. In building digital RF receivers, the input signal is digitized by ADC. It is desirable to add this digitization effect.

The maximum voltage of an ADC is half of the peak to peak voltage. For example, if the ADC has +1 to -1 voltage, the maximum voltage is 1 v. There are two different maximum voltages depending on the bit number. For the 4 to 8 bit ADC, the maximum voltage is assumed to be 0.27 v and for the 8 to 16 bits the maximum voltage is assumed to be 1 v. In this study, the 8-bit ADC can assume two maximum values.

The RF signal passes through an amplifier before reaching ADC. The amplification required depends on the bit number of the ADC. Table 6 shows the amplification required vs. bit number of the ADC [2]. These data were derived based on a condition maximizing the product of differentiated sensitivity and dynamic range vs. amplification. This topics will be presented in a future report. One notes that ADC with small bit number usually requires high gain. One also notes that 16-bit ADC requires no gain. This design may be interesting due to its simplification.

The IDR is calculated using frequency identification as a criterion. Its dependence on frequency separation and ADC's bit number are shown in Figure 8(a). Each data point is obtained through 10 runs of calculation with different Gaussian noise. When the number of bits is low, such as from 4 to 10 bits, the IDR is less than zero. It means that music method can not detect two simultaneous signals separated by 1 MHz even if they have the same amplitude. When the number of bits is 12 or higher, signals separated by 1 MHz with amplitude within IDR can be detected. Figure 8 also shows that the IDR obtained from the two 8 bit ADCs (with 0.27 v and 1v maximum voltage) are very close. Thus, the dynamic range is determined by the number of bits rather than the maximum voltage.

The calculated IDR vs. frequency separation in Figure 8 are least-square fitted using the following equation:

$$IDR(\Delta f) = A\{1 - \exp[-(\Delta f - \Delta f_0)/B]\}$$

The least-square-fitted parameters of A, B and Δf_0 are listed in Table 7 for ADC with different bit number. The fitting curve results are also shown in Figure 8.

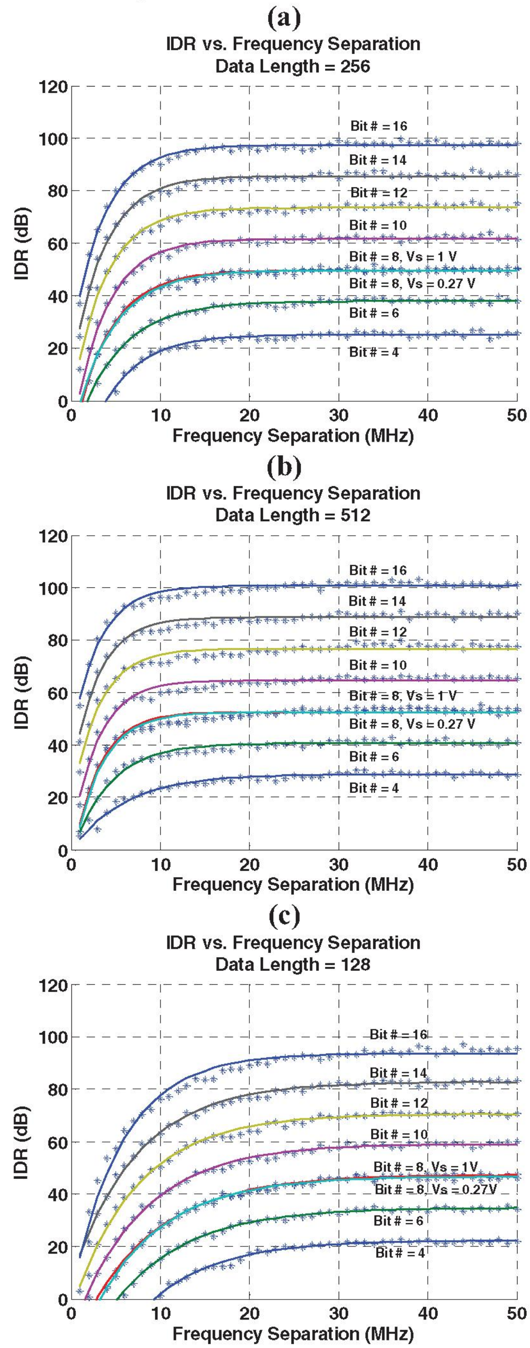


Figure 8: IDR with digitization effect and using frequency identification method. The best fitting results are also shown.

IDR vs. data-length in digitization: In this last section, the effect of data length will be investigated. The procedure is identical to those used in the previous subsection, except the order used for the 128-data-length case is 40. The results are shown in Figure 8(b) and (c) for 512- and 128-data-length, respectively. Together with the 256-data-length case, the IDR is increased as the data length is increased for the same ADC's bit number.

For ADC with 16-bit and frequency separation of 1 MHz, the IDR is 3, 37.2 and 54.9 dB for the 128, 256, and 512-data-length cases, respectively. It also shows that the smaller data length case reaches a slightly smaller saturation value at a larger frequency separation. Within each data-length case, the saturation IDR vs. bit number of the ADC indicates a 6 dB per bit improvement which is as expected. Figure 8 shows that IDR for 16-bit ADC reaches saturation of about 94, 97.4 and 100.6 dB for 128-, 256- and 512-data-length cases, respectively. The difference among these cases is accountable to the difference in data length.

CONCLUSION

Instantaneous dynamic range is an important parameter evaluating high performance digital EW receiver. It indicates the ability to detect simultaneous multiple signals with a large amplitude difference. This study focuses on the stand-alone EW receiver capable of generating PDW for autonomous signal detection. Typical FFT process, requiring less computing time, but the resolution is limited by the frequency bin determined by the integration time. The music method, though requiring long computing time, can provide high resolution results. The results of IDR vs. frequency separation depend on the order used in music, and the definition of the signal detection. For 256 real data points sampled at a frequency of 2.56 GHz, an order of 80 seems to be optimized. There are two signal detection definitions in this study. One is through the eigenvalue distribution, and the other is through the signal characterization in the spectral domain. The IDR using the latter definition is smaller than those using the former definition. Finally, the practical design parameters of ADC are used to calculate IDR vs. frequency separation and ADC's bit number using the latter definition of signal detection. The calculated results are least square fitted with fitting parameters. The resulting IDR vs. frequency separation for ADC with various numbers of bits are presented. These results can be used as the limit of IDR in EW receiver design for two-signal detection.

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Table 1 Eigenvalues generated from noise alone; order = 80

	Mean	Standard deviation
Eigenvalue 1	2.08	0.43
Eigenvalue 2	1.73	0.26
Eigenvalue 3	1.55	0.21

Table 2 Eigenvalues generated from one strong signal (S/N=100 dB) and noise; order = 80

	Mean	Standard deviation
Eigenvalue 1	1.42×10^{11}	7.14×10^{9}
Eigenvalue 2	2.04	0.42
Eigenvalue 3	1.70	0.26

Table 3 Eigenvalues generated from two strong signals (S/N=100 dB) and noise; order =80

	Mean	Standard deviation
Eigenvalue 1	3.7×10^{10}	2.14×10^{9}
Eigenvalue 2	4.37×10^{8}	2.23×10^{7}
Eigenvalue 3	2.06	0.43

Table 4 Eigenvalues generated from noise alone; order = 50

	Mean	Standard deviation
Eigenvalue 1	1.84	0.35
Eigenvalue 2	1.54	0.21
Eigenvalue 3	1.39	0.16

Table 5 Eigenvalues generated from noise alone; order = 100

	Mean	Standard deviation
Eigenvalue 1	2.19	0.47
Eigenvalue 2	1.81	0.30
Eigenvalue 3	1.61	0.23

Table 6 Gain required before ADC

No. bits	4	6	8 (.27V)	8 (1V)
Gain (dB)	52	40	28	40

No. bits	10	12	14	16
Gain in dB	28	16	4	-8

Table 7 the fitting parameters of A, B and df_0 in $DR(df)=A\{1-\exp[-(df- df_0)/B]\}$

	4 bits	**6 bits**	**8 bits (V_0=0.27V)**	**8 bits (V_0=1.0V)**
df_0	3.97	1.86	1.24	1.12
B	4.48	4.98	4.07	4.37
A	25.24	37.95	49.69	49.69

	10 bits	**12 bits**	**14 bits**	**16 bits**
df_0	0.86	0.13	-0.41	-0.89
B	3.68	3.68	3.64	3.61
A	61.65	73.54	85.63	97.39