Improving Spectral Efficiency and Reducing Adjacent Channel Interference of a Wireless Emergency Communication System

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***Abstract*— Cellular services have been ineffective in providing defense and disaster recovery communications. Overloads that occur after disasters cause degraded resource access to all users, no matter how important. This has led to the development of the Wireless Priority Service, which will queue emergency calls if they are first blocked. The next generation of Wireless Priority Services, however, must also be able to support more numbers of calls after disaster strikes, which can only be achieved by increasing the spectral efficiency of a system. In this work, we propose to use linear, higher order modulation techniques for priority services instead of the currently deployed non linear modulation schemes used in GSM systems. Linear, higher order modulation techniques increase spectrum efficiency as needed but suffer from spectral regrowth, resulting in increased Adjacent Channel Interference and Inter Symbol Interference. Our work deals with Adjacent Channel Interference, and we presented a new technique to model Adjacent Channel Interference in our earlier work. Here we follow by designing filters to reduce Adjacent Channel Interference. Finally, we show how these linear modulation schemes and interference minimizing filters can be activated as needed using software defined radio.**

***Index Terms*— Autocorrelation, Probability Distribution function, Linear Adaptive Filter, FIR filter.**

I. Introduction

Disasters, terrorist attacks, and major accidents have always triggered tremendous telephone traffic in the land line and wireless network. During emergency situations, the availability of a cellular system can be greatly impaired due to the heavy traffic demand placed upon the surviving cellular systems in the aftermath of disasters. This results in a high call blocking rate to critical disaster relief officials when communication is needed the most. This justifies the need for providing some high quality wireless priority service, where some set of user groups are given different, higher, and appropriate level of service.

In the United States, this resulted in the Wireless Priority Service being developed and managed by National Communication System. The Wireless Priority Services can be defined as an enhancement to the basic wireless service that allows the National Security/Emergency Preparedness (NS/EP) calls to queue for priority service instead of being blocked. In times of emergency or crisis, the system enables designated WPS users to have a greater chance of being able to complete calls. Not every individual can subscribe for WPS; only NS/EP personnel can benefit from the advantages of WPS. The WPS provides only the qualified and authorized NS/EP users with means to obtain priority access to the next available radio channel in a wireless call path when emergency calls are placed. When an NS/EP user places a WPS call and the service is activated, the mobile handset requests a radio channel via control channel messaging. In non-congested environments, a radio channel is allocated to the mobile set and the call is connected. However, in congested environments a radio channel may not be available and so the network will not be able to grant a channel. This situation invokes WPS. When congestion is encountered at call origination, the NS/EP caller is placed in a queue and will be given the next available radio channel ahead of any other non-emergency request. WPS does not preempt calls in progress and is intended for use in emergency situations where network congestion is blocking call attempts. During normal everyday situations, noone will observe a change in the ability to make calls. Channels are not held in reserve in anticipation of NS/EP events.

Of the currently used cellular technologies, only GSM has the ability to prioritize calls. And of the GSM service providers in the U.S., only T-Mobile has deployed the Wireless Priority Service. Since only one service provider has implemented WPS, it is not able to meet the high call demands; this was experienced during the Northeast blackout of August 2003. Therefore, in order to improve this situation and make priority services fully functional, one of two things must happen.

- All of the carriers could implement priority service, either all of the carriers or at least all of the GSM carriers. This would enable most police, firefighters, and first responders to get priority for their cell phone calls after disasters.
- The Wireless Priority Service provider could increase its capacity to support more users during emergencies. This would allow all priority service users to subscribe to the same service provider to facilitate communication amongst themselves during emergencies.

This paper facilitates the second option by focusing on improving the existing Wireless Priority Service by proposing the use of a spectrally enhanced WPS system for voice communication. This system should be used only in emergency situations and not as an underlying GSM system. So the GSM architecture will consist of two sub-architectures. One of them will be the existing architecture complying with the current GSM standards to be used during non-emergency conditions and the other will be the high capacity architecture which can be used during emergencies. So typically the existing architecture should be used and then the new architecture can be used during emergency situations. The emergency system will use linear, higher order modulation techniques like M-QAM and M-PSK which have higher spectral efficiency; the system will also comprise of interference minimizing filters. The incorporation of two different architectures within the same piece of equipment and existing cellular architecture can be accomplished by using Software Defined Radio (SDR).

In this paper we compare the performances of linear modulation schemes, i.e., Quadrature Amplitude Modulation and Phase Shift Keying of the order of 4, 8, 16, 32 and 64, with that of Gaussian Minimum Shift Keying in the presence of Adjacent Channel Interference. Adjacent channel interference is modeled and added to the transmitted signal using the approach mentioned in [1]. The use of linear, higher order modulation schemes results in a higher BER than that obtained with non-linear modulation schemes. We reduce the BER for the proposed system with the incorporation of FIR filters to minimize the effect of Adjacent Channel Interference for M-QAM systems and with Linear Adaptive filters to minimize the effect of Adjacent Channel Interference in M-PSK systems.

II. Related Work

The introduction mentioned the emergence of Wireless Priority Services and the initial work done with it. Since Wireless Priority Services is relatively new, not much work had been done to study or improve its capabilities. A service with the same objective has been proposed in [2]; here researchers investigate a rapidly deployable wireless communication system for disaster and emergency response. But this is not related to cellular communications.

Although little work has been done in the field of Wireless Priority Services, a significant amount of work has been done which focuses on improving the spectral efficiency of a wireless communication system. The EDGE (Enhanced Data Rates for GSM and TDMA/136 Evolution) technology [3] provides significantly higher user bit rates and spectral efficiency. In [4], work similar to what we have proposed above is done except that coding schemes are used to reduce the error rates of the system as compared to filtering schemes. In order to satisfy high bandwidth requirements of broadcast networks, higher order modulation schemes like 8PSK or 16QAM are chosen to increase the data rates through the existing transmission link. Also, a new power and spectrally efficient family of modulations, FQPSK (Feher-patented quadrature-phase-shift keying), are proposed for wireless mobile and personal communication services in [5]. Several evolutionary steps like using multi-modulation, where higher-order modulation techniques such as QPSK and 16QAM, use of multi-slot, where more than one time slot per frame is allocated to a user and multi-carrier transmission, in which more than one GSM carrier can be simultaneously allocated to a user are being considered in [6] to enable higher bit rate services to be mixed with low bit rates in a combined scheme.

In our system we use higher order modulation techniques as in many of the proposals above, but we compensate for the degradation due to Adjacent Channel Interference with the help of filters, designed using the parameters obtained in the model proposed in our earlier work in [1].

III. Filter Design

Adjacent Channel Interference is the interference due to signals with different carrier frequencies which are close enough to cause spectral overlap. This degrades the receiver performance depending upon relative power levels of interfering signals, modulation and bandwidth. Adjacent Channel Interference limits the capacity and performance of a digital wireless communication system and hence we need to design systems to mitigate ACI so as to improve the performance and capacity for cellular, mobile and land mobile systems.

Using the procedure mentioned in [1], we can find the statistical characteristics, namely the probability density function of the interference. In this paper we explain system designs which minimize the degradation caused by Adjacent Channel Interference.

Earlier ACI mitigation was done using equalization methods and subtractive demodulation. In [7], coherent maximum likelihood sequence estimation receivers are developed for demodulation of adjacent channel signals. The use of a linear equalizer or a decision feedback equalizer to suppress all received adjacent channel, intersymbol, and cochannel interference is explained in [8]. But the analysis in the above papers makes an assumption about the channel response. A completely different approach based on SAW filter design is presented in [9]. Here the author presents a sophisticated design procedure for a triple mode SAW filter, applying extended eigen-mode analysis.

A. *Design of the FIR Filter*

Here we design a digital FIR receive filter as mentioned in [10], using a cost function which minimizes the total mean square error by considering the effect of Adjacent Channel Interference. The filter in [10] is designed for minimizing jointly the mean square error value of the channel noise, Inter Symbol Interference and Adjacent Channel Interference. However, since our work deals mainly with Adjacent Channel Interference, the derivations below are simplified from those in [10].

Consider the model of a communication system in which the output signal $y(x)$ is a combination of the transmitted signal and adjacent channel interference. Eq. (1), defines an error component that indicates the deviation from the desired signal $s(x)$.

$$e_0 = y(x) - s(x) = i(x) \tag{1}$$

In order to minimize the above error component, we first evaluate the mean square error (MSE). Assuming uncorrelated ACI, the MSE can be expressed as

$$E\{e_0^2\} = E\{i(x)^2\} \tag{2a}$$

$$E\{e_0^2\} = \sigma_{aci}^2 \tag{2b}$$

where $E\{.\}$ denotes the expected value of the argument and σ^2_{aci} is the variance of the ACI, which is same as the average power.

The ACI term can be elaborated as follows. After recive filtering, the time domain signal is $i_R(t)$=$i_T(t)$*$h_r(t)$, where $i_R(t)$ and $i_T(t)$ are the time domain representations of the received and the transmitted signals respectively and $h_r(t)$ is the impulse response of the receive filter, which might be either rectangular or root raised cosine. The variance of ACI can be expressed as

$$\sigma^2_{aci} = E\{i_R(t)^2\} = \int_{-1/2}^{1/2} S_i(v)|H_R(v)|^2 dv \tag{3}$$

where $S_i(v)$ is the power spectrum of the ACI signal and $H_R(v)$ is the Fourier transform of the receive filter response. The power spectrum of the ACI can be defined as

$$S_i(v) = \sum_{-\infty}^{\infty} r_i(k) e^{-j2\pi kv} \tag{4}$$

where $r_i(k)$is the autocorrelation sequence of ACI:

$$r_i(k) = E\{i_T(nT) i_t^*((n+k)T)\} \tag{5}$$

The interference signal $i_T(t)$ can be represented as

$$i_T(t) = s_2(t) + s_3(t) \tag{6a}$$

$$i_T(t) = \sum_{p=-P}^{p=P} \sqrt{\frac{2E_s}{T_s}} cos(2\pi pB_c t + \theta_p) \sum_{i=-\infty}^{\infty} a_i h_T(t - iT - \tau_p) \tag{6b}$$

where $s_1(t)$ and $s_2(t)$ are interfering signals, θ_p and τ_p are the phase shift and time delay of the ith symbol, B_c is the frequency spacing between the adjacent channels, a_i is the amplitude of the symbol and 2P is the total number adjacent channels. Using Eq. (6) and assuming that E$\{\theta_p\}$ and E$[\tau_p\}$ to be zero and E$\{a_i\ a_j\}$=A^2, Eq. (4) can be elaborated as

$$S_i(v) = A^2 \sum_{p=-P}^{p=P} |H_T(v + pB_c T)|^2 \tag{7}$$

Substituting the above into Eq. (3) we get

$$\sigma^2_{aci} = A^2 \int_{-1/2}^{1/2} \sum_{p=-P}^{p=P} |H_T(v + pB_c T)|^2 |H_R(v)|^2 dv \tag{8a}$$

$$\sigma^2_{aci} = h_R^T R_{aci} h_R \tag{8b}$$

where R_{aci} is the autocorrelation matrix of the ACI. The elements of R_{aci} are calculated by first calculating the autocorrelation vector of the interference signal and then forming a symmetrical matrix from its elements.

Now we define a Lagrange function to be minimized as

$$\ell(h_R) = \beta_{aci} A^2 h_R^T R_{aci} h_R \tag{9}$$

along with the constraint

$$h_r^T w_0 = 1 \tag{10}$$

In Eq. (9) weight parameter β_{aci} has been introduced in order to be able to experiment with parameter values to determine if non-unity values will lead to better BER performance. The minimizing solution is found by setting the derivative with respect to h_R to be zero. On taking the derivative, we get

$$h_R = \frac{P^{-1} w_0}{w_0^T P^{-1} w_0} \tag{11}$$

In the above equation $P = \beta_{aci}\ A^2\ R_{aci}$. The bit energy at the receiver input is given as E_b=$A^2\ h_{TC}^T\ h_{TC}$, therefore P can be denoted as

$$P = N_0 \frac{E_b/N_0}{h_{TC}^T h_{TC}} [\beta_{aci} R_{aci}] \tag{12}$$

We see from the derivations that in order to design this filter we need to know about the autocorrelation function of the interference and the transmit filter response. The autocorrelation function can be found by using the approach mentioned in [1].

In our work we use the above equation to design the matched filter response for the receiver of a M-QAM system.

B. Adaptive Linear Filter Design

Here we design a linear filter as mentioned in [11] for linear signal estimation when discrete time data consists of a signal plus additive independent noise. This filter can be used when signal probability distributions are completely unknown but the noise mean and covariance properties are known. This particular filter can be designed using results similar to the ones derived from our scheme to measure Adjacent Channel Interference in [1]. Let s_k and n_k be independent sequences corresponding to signal and noise respectively; the received signal is given by x_k=s_k + n_k. The noise sequence is assumed to have a zero mean, and for zero mean the autocorrelation function and covariance function are the same. The covariance function is given by $R_n(k,j)$=E$\{n_k\ n_j\}$.

This filter design is based on the principle of averaging. Using the averaging operation, the effect of noise can be reduced over the sample space, similarly here an estimate of the signal is obtained by linear weighting of M points preceding the estimated point and L points after the estimated point.

$$\hat{s_k} = \sum_{j=-M}^{L} \beta_j(N) x_{k+j} \tag{13}$$

where k varies from 1 to N, here N is the total number of samples. The above Eq. (13) can be represented in vector form as follows

$$\hat{s_k} = B_N^{'} X_k \tag{14}$$

where $B_N^{'}$ and X_k are column vectors of size $M + L + 1$. Now the weight vector needs to be determined from the sample points. The ideal filter would be one which will make Eq. (13) close to s_k. Therefore we need to find value of $B_N^{'}$ which would

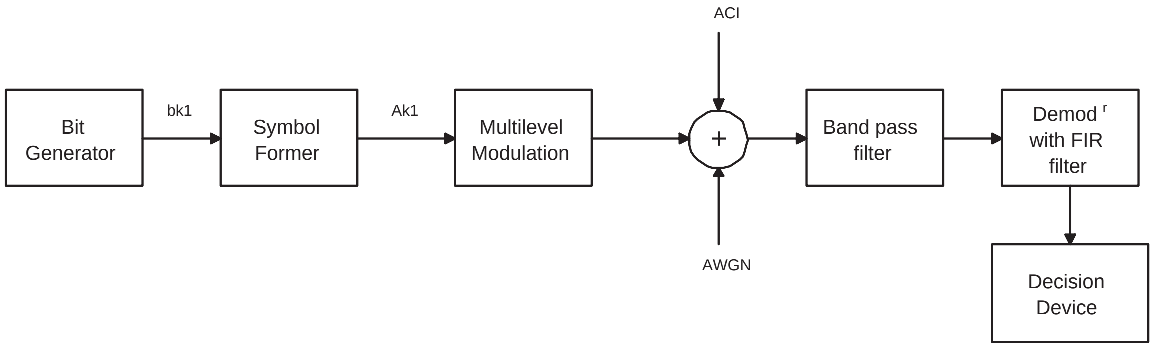

Fig. 1. M-QAM Emergency System

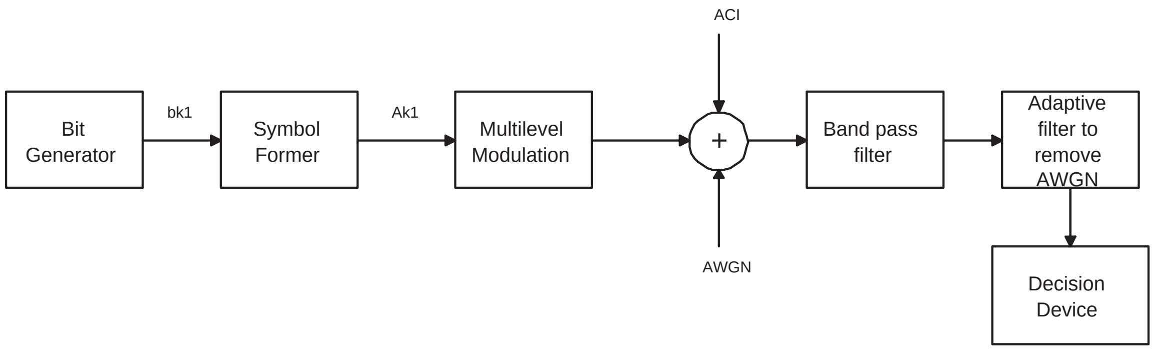

Fig. 2. M-PSK Emergency System

make it possible, hence following the mean square error approach

$$E\{s_k - B_N^{'} X_k\}^2 = E\{x_k - n_k - B_N^{'} X_k\}^2 \tag{15a}$$

$$= E\{((X_k - B_N^{'} X_k) - n_k)^2\} \tag{15b}$$

$$= E\{(x_k - B_N^{'} X_K)^2\} - E\{2nk(X_k - B_N^{'} X_K)\} + E\{n_k^2\} \tag{15c}$$

As mentioned above, we need to minimize the error. The minimum of Eq. (15a) is obtained when

$$\frac{1}{N}\sum_{k=1}^{N}[x_k x_{k+i} - R_n(k, k+i)] \tag{16a}$$

$$= \sum_{j=-M}^{L} \beta_j(N)[1/N \sum_{k=1}^{N} x_{k+j} x_{k+i}] \tag{16b}$$

where i varies from $-M$ to L. In matrix form we have B_N=$r^{-1}g$, where

$$r = 1/N \sum_{k=1}^{N} x_{k+j} x_{k+i} \tag{17}$$

$$g = 1/N \sum_{k=1}^{N}[x_k x_{k+i} - R_n(k, k+i)] \tag{18}$$

The above mentioned design procedure assumes that noise has zero mean.

IV. Emergency System Design

From [1], the absolute and actual Adjacent Channel Interference in a M-QAM system follows gamma distribution. For this case we use the FIR filter to combat the Adjacent Channel Interference. The FIR filter is used as a matched filter and is therefore incorporated in the demodulator circuit as shown in the Fig. 1.

The results in [1] show that the adjacent channel interference in a M-PSK system follows exponential distribution, when the absolute value of the deviation is considered whereas it follows normal distribution when actual value of error is considered. For the purpose of calculating probability of error, the one-sided distribution suffices, but for the purpose of filter design we need to consider the two sided distribution of error. Hence we can use the adaptive filter design in the M-PSK system after the demodulation stage as shown in Fig. 2 to combat ACI .

V. Software Defined Radio

SDR is a technology which facilitates implementation of some of the functional modules in a radio system such as modulation, demodulation, signal generation, coding and link-layer protocols in software. SDR helps in building reconfigurable software radio systems where dynamic selection of parameters for each of the above-mentioned functional modules is possible. For example, SDR can be used to download coefficients for a filter block or equation of a generation polynomial for a coding block. Since the filter coefficient and generation polynomial determine the performance of the filter and coding block respectively, a desired system can be obtained at run time by downloading the external parameters using the air interface [12]. SDR systems implemented in [13], [14] and [15] support multi-band and multimode radio standards for such applications as wireless LANs and cellular phone systems, whose carrier frequency ranges from 500 MHz to 9 GHz, applicable modulation schemes of BPSK, QPSK, 8-PSK, 16-QAM, and 64-QAM, and bandwidth of 15MHz.

For the emergency system, we propose an SDR platform which supports multiple modulation schemes of GMSK and either M-PSK or M-QAM. SDR would download and implement the corresponding filter coefficients during emergencies to minimize the effect of Adjacent Channel Interference.

VI. Results and Discussion

In this section, we study three sets of results for M-PSK and M-QAM systems. We incorporate the FIR filter in the M-QAM systems and the adaptive filter in M-PSK systems. The performance of the various systems is evaluated in the presence of Adjacent Channel Interference. While simulating the systems, it has been ensured that system parameters like average power, amount of interference, bit timings and carrier frequency are the same for all systems.

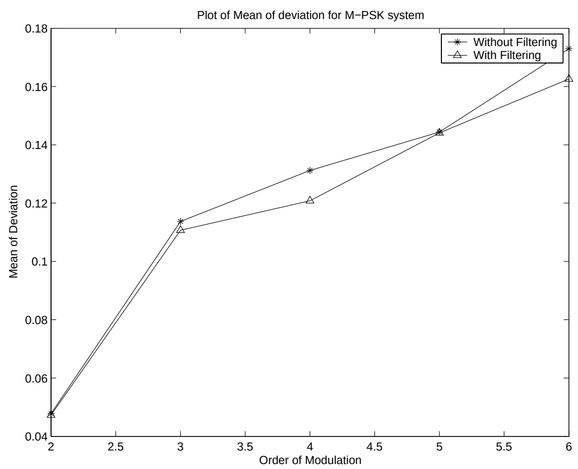

Fig. 3. Mean of the Deviation for M-PSK System

The first set of results shows the amount of deviation of the received signal from the transmitted signal in the presence of Adjacent Channel Interference M-PSK and M-QAM systems. The value of mean of the deviation is obtained by averaging the values of deviation for all symbols. This value of the deviation is calculated for the various communication systems, with and without the presence of filters. The resulting value of the mean of the deviation is plotted against the order of the modulation. Fig. 3 shows the values of deviation for the M-PSK system. The plot in Fig. 4 shows the deviation for a M-QAM system. Note that these plots show a reduction in the mean of the deviation, which will result in significant changes in the Bit Error Rate. As can be seen from the above plots, the use of filters helps in reducing the mean of the error. Also the amount of reduction obtained for M-QAM systems using an FIR filter is far more than that obtained for M-PSK systems using adaptive filters.

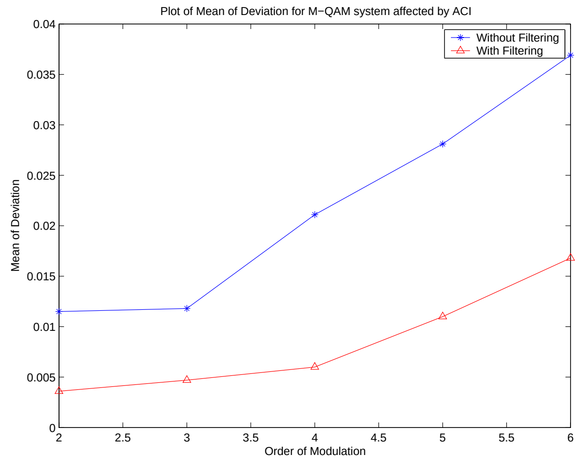

Fig. 4. Mean of the Deviation for M-QAM System

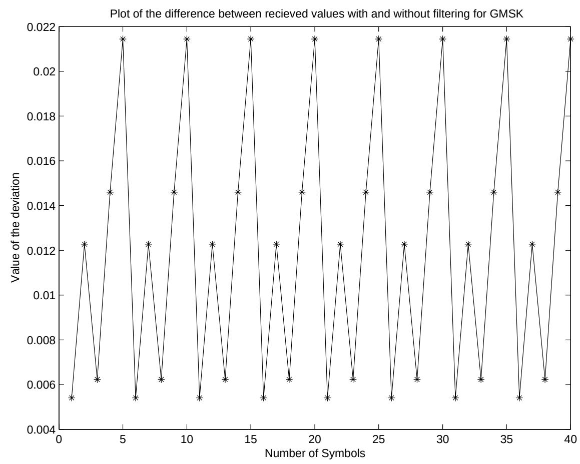

Fig. 5. Difference between the Transmitted and Received Signal Value for GMSK System

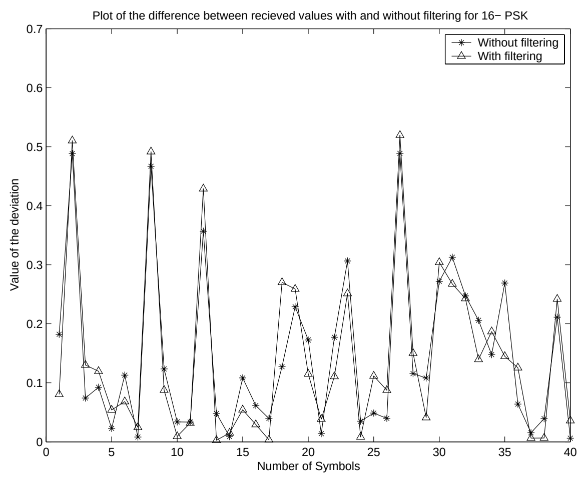

Fig. 6. Difference between the Transmitted and Received Signal Value for 16-PSK System

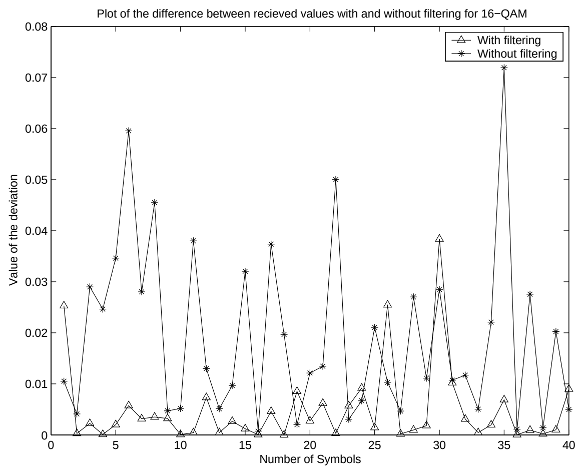

Fig. 7. Difference between the Transmitted and Received Signal Value for 16-QAM System Affected by only ACI

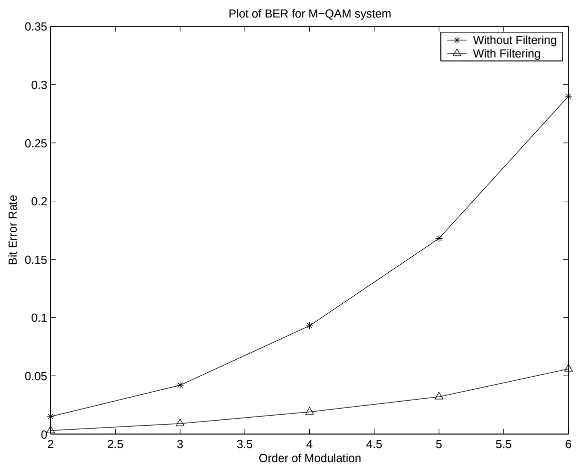

Fig. 9. Bit Error Rate of M-QAM System

The second set of plots show the absolute difference between the received and the transmitted symbol values for GMSK, 16-PSK and 16-QAM systems in the presence of ACI. Fig. 5 shows the absolute difference between the transmitted and received value versus the symbol number for a GMSK system. For 16-PSK and 16-QAM, two curves are shown within the plot, corresponding to the difference between the received and the transmitted symbol values with and without the presence of filters. Shown in Fig. 6 is the difference value versus the symbol number for a 16-PSK system. Similarly Fig. 7 shows the difference for M-QAM system. As evident in the plots, the difference between the transmitted signal and the received signal is lower for systems with FIR filters which results in a low mean.

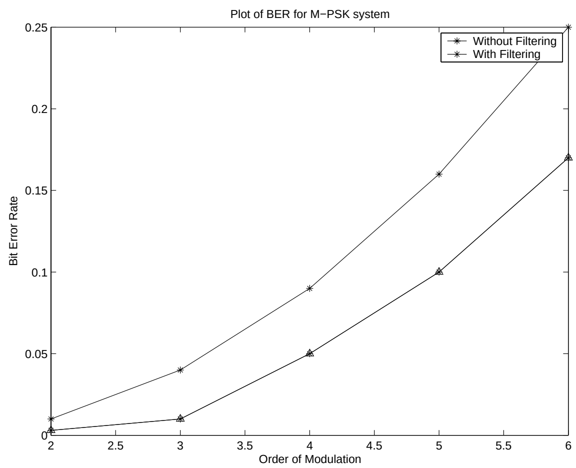

Fig. 8. Bit Error Rate of M-PSK System

The third set of results shows plots of bit error rate of the communication system versus the order of the modulation. As mentioned above, here again two curves are shown, one for the BER in the presence of filter and the other of BER without the filter. The plot in Fig. 8 shows the plot of BER versus the order of the modulation for an M-PSK system and Fig. 9 shows the plot of BER versus the order of the modulation for an M-QAM system. The BER of a GMSK system for the same set of parameters is 0.018. As expected in both the cases the BER increases with the order of the modulation. It can be observed from Fig. 7 and Fig. 4 that the FIR filter helps achieve significant reduction in the amount of Adjacent Channel Interference. The adaptive filter also helps achieve reduction in noise. Although this reduction might not seem quite significant, these small reductions are quite beneficial for an actual communication system where the noise might not be as high as we have modeled it to be.

VII. Conclusion

In this paper using results of previous work, we designed an FIR filter and adaptive filter to combat Adjacent Channel Interference for an M-QAM system and M-PSK system respectively. Results show that both filters help us achieve performance gains but the improvement obtained using the FIR filter is more as compared to that achieved with adaptive filter.

Using SDR we can design a system which can switch to an emergency system without major changes in the infrastructure. Hence the proposed system is a next step towards the development of a spectrally efficient Wireless Emergency Communication System.

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