

REDUCING CALL-DROP IN GLOBAL SYSTEM FOR MOBILE COMMUNICATION

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Abstract

Call Drop is one of the most disturbing problems in Mobile Communications. It is one of the most important Quality of Service (QoS) indicators for a mobile carrier, hence over the years, many strategies have been proposed to solve the problem but without much success. One of the important factors for call drops is high Bit Error Rate (BER). This paper proposes the reduction of call drop due to high BER, by using a new signal processing subsystem at the receiver section of a wireless system. In particular, the calculation of un-mixing matrix and automated identification for distinguishing separated signals were adopted. To explore the effects of this subsystem on the performance of the wireless system, the overall block diagram for Global Systems for Mobile (GSM) communication system was simulated and its BER measured. It was found that this subsystem helped to improve the BER and hence the end-to-end performance of the system. The effect of this signal processing block was explored by considering Additive White Gaussian Noise (AWGN), Ricean Fading and Channel Coding. The implication of this, is that the existing network will see a reduction in call drops. The results showed a coding gain of 3dB.

Keywords: *Bit Error Rate (BER), Quality of Services (QoS), Channel Coding, Additive White Gaussian Noise (AWAN), Global Mobile System (GSM), Channel Equalization, Call Drop.*

1.0 Introduction

The power of the mobile handset has gone beyond the imaginations of those who found it. It has gone from simple signal communication to detection, surveillance, etc, functions. In the developing countries its welcome has been frustrated by the almost common existence of the problem of call drops. The mobile operators have several messages to pacify the user like network congestion, line busy, please dial x, user not available etc. Most of these are due to call drops.

Call drop in essence is a measure of the ability of a mobile network to maintain a call until it is terminated. A call is said to drop when the active phone call is unexpectedly terminated. The call drop rate is commonly measured as the amount of call drops by a percentage of the calls. The Quality of Service (QoS) has been found to be better, when the above percentage is lower. One of the important

reasons for call drop is high Bit Error Rate (BER). It is a parameter used in assessing systems that transmit digital data. As signal is transmitted from one location to the other, errors due occur and so BER can be said to be the percentage of bits with errors divided by the total number of bits that have been transmitted, received or processed over a given period of time.

Most service providers have a certain threshold set for the BER. Its increase above the threshold leads to call drop.

Three common reasons for call drop related BER are

- (i) Poor transmission (Liu et al, 2003)
- (ii) Hand-over or Handoff; and
- (iii) Large hysteresis during hand off

Handoff is the process of changing the channel associated with the current connection while is a call in progress Kumar et al (1994).

2.0 Existing Methods For Preventing Call Drops

Quality of Service (QoS) provisioning in wireless networks is a challenging problem due to the scarcity of wireless resources and to support various integrated services with a certain QoS requirement in wireless networks, resource provisioning is a major issue.

Call Admission Control (CAC) is one of such resource provisioning strategies. It is a technique to provide QoS in a network by restricting the access to network resources. Simply stated, an admission control mechanism accepts a new call request provided there are adequate free resources to meet the QoS requirements of the new request without violating the committed QoS of already accepted calls. It limits the number of call connections into the networks in order to reduce the network congestion and call dropping (Ghaderi et al, 2006), (Fang et al, 2002).

Various algorithms and schemes have been developed for CAC. The purpose of an admission control algorithm is to decide at the time of call arrival, whether or not a new call should be admitted into the network. Some of CAC schemes are features are:

- Guard Channel Schemes: (Fang et al, 2002): In this scheme, each cell in a wireless network reserves a number of channels (or bandwidth units) exclusively for handoff calls and whenever a channel is released, it is returned to the common pool of channels.
- Whenever a mobile user travels from one area of coverage or cell to another cell within a call's duration the call should be transferred to the new cell's base station. Otherwise, the call will be **dropped** because the link with the **current** base station becomes too weak as the mobile recedes. This seamless **ability** of transference from one base station to another during the duration of a call is known as a handoff procedure, (Ekiz et al, 2005).
- During handoff, the user needs to release the currently allocated channel and acquire a channel from another base station without the call being dropped. The base station without Guard Channel Schemes may not have channels available, leading to call drops. Guard Channel Schemes make sure that a portion of channels are reserved for handoff calls and call drops do not occur during handoff requests.
- The main drawback in this strategy is that it increases the new call blocking probability and under-utilizes the radio resources, far out-weighting the reductions it achieved in **handover** blocking probabilities. In other words, the cost of the classical method of employing guard channels to decrease the call dropping rate is the increase in call blocking rate. Since subscriber mobility changes in time, the number of handoff attempts in each cell is subject to fluctuations, making static assignments (or periodical update) of a given number of guard channels inefficient. Therefore, only slight gains in system utilization can be achieved by this scheme. Furthermore it does not give any thought as to whether any handover channels are required at any given moment or if so, by how many.
- Queuing Priority Schemes, (Chang et al, 2001): In this scheme, calls are accepted whenever there are free channels. When all channels are busy, instead of dropping a call, new calls are queued.
- Queuing of handoff requests, when there is no channel available, can reduce the dropping probability but at the expense of higher new call blocking. If the handoff attempt finds all the channels in the target cell occupied it can be queued. If any channel is released it is assigned to the next handoff waiting in the queue and not to a new call.

- Some dynamic channel assignment schemes using power control have also been proposed. It is shown (Grace et al, 2000) that these schemes reduce call drops and increase capacity compared to those using fixed transmitter power.

3.0 Connectivity between the BER and Call Drops

The methods discussed so far speak about the call drops and channel unavailability. These methods aim at improving call drop by controlling the channel assignments, assuming that there is availability of channels, are all the new calls always admitted? Or is the quality of call always good enough? This is because new call is admitted if and only if its QoS constraints are satisfied without jeopardizing the QoS constraints of existing calls in the network. QoS is usually defined according to the BER in digital transmission. For example, the QoS requirement for voice users is usually expressed as a BER less than 10^{-3} in order to guarantee quality of communication (Liu et al, 2003). Thus, if we improve the BER for the system, the problem of call drops can be reduced. In general, the BER is the percentage of bits that have errors relative to the total number of bits received in a transmission, usually expressed as ten to a negative power. Hence BER is expressed as in equation (1).

$$\text{BER} = e/N \quad \dots(1)$$

Where, e = number of errors
 N = total number of bits received

With a strong signal and an unperturbed signal path, this number is small and can be considered insignificant. Bit-error-rate performance is usually depicted on a two dimensional graph. The ordinate is the normalized signal-to-noise ratio (SNR) expressed as E_b/N_0 : the energy-per-bit (E_b) divided by the power spectral density of the noise (N_0), expressed in decibels (dB). The abscissa is the BER, a dimensionless

quantity, usually expressed in powers of ten.

To create a graph of BER versus E_b/N_0 , a series of points are plotted. Each of these points requires that a simulation be run at a specific value of SNR or E_b/N_0 . A typical plot of BER vs. E_b/N_0 is shown in fig 1. For the BER curve to be improved, that is for the same E_b/N_0 , and less BER can be obtained, then the end-to-end performance of the system would have been said to be improved. This in turn will help in the reduction of call drop.

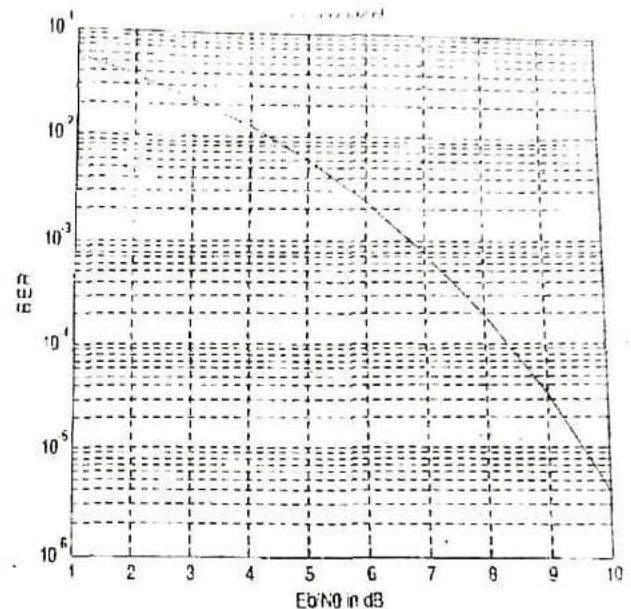


Fig. 1: Typical plot for BER vs. E_b/N_0 (Sharma, 2008)

Several attempts have been made at improving BER performance. These includes;

- Using two dimensional Fourier Transform for speech enhancement, this involves the reduction of noise in the system.
- Using empirical mode decomposition. Here by a sifting process, the desired signal is decomposed into intrinsic mode functions (Kopsinis et al; 2008).
- Using independent component analysis. This is used to separate two signals from each other when two

mixtures of those signals are provided as inputs, a process known as blind source separation (Singh, 2008). (Proakis, 2008) has even modified this by adding noise before filtration of the mixed signal.

- (iv) Using channel Equalization technique. Equalization is used to extract the desired signals from the unwanted reflections. This is because as signal transmit, and fall on bodies there could be multiple signal reflections.

Probability of error is defined as the probability that the transmitted symbol is detected in error. This error is largely influenced by the channel conditions. Channel fading and AWGN causes this error. The BER calculation is thus used to compare the actual payload with the decoded bits. The error introduced depends on the value of the SNR. Higher value of the SNR indicates that less noise is added and fewer errors are introduced.

Since the study categorically deals with the improvement of BER, it is important that the BER for the existing QPSK modulation is known. The symbol error rate for the QPSK modulation with AWGN channel (Proakis, 2008) is given as:

$$P_M = 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \left[1 - 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right)\right] \dots (2)$$

The relation between the BER and symbol error rate (Proakis, 2008) is given by

$$P_b = \frac{1}{k} P_M \dots (3)$$

Where, P_b = Bit error rate
 P_M = Symbol error rate
 and k = Number of bits per symbol

Hence, for QPSK (4-ary PSK), $M = 4$.
 Therefore, $k = 2$.

$$\text{Bit Error Rate} = \frac{1}{2} P_M \dots (4)$$

Thus, the BER for QPSK with the AWGN channel is given as.

$$P_b = Q\left(\sqrt{\frac{2E_b}{N_0}}\right) \left[1 - 2Q\left(\sqrt{\frac{2E_b}{N_0}}\right)\right] \dots (5)$$

Hence, from equation (5) it can be seen that the probability of bit error or the BER is dependent on the ratio of energy per bit to noise spectral density.

4.0 Theoretical and Simulated BER Comparison for AWGN Channel

To validate the simulation of the physical layer of the wireless communication system, a baseline for the comparison purpose was required. The theoretical curve for QPSK with AWGN channel (fig. 2) served as this baseline.

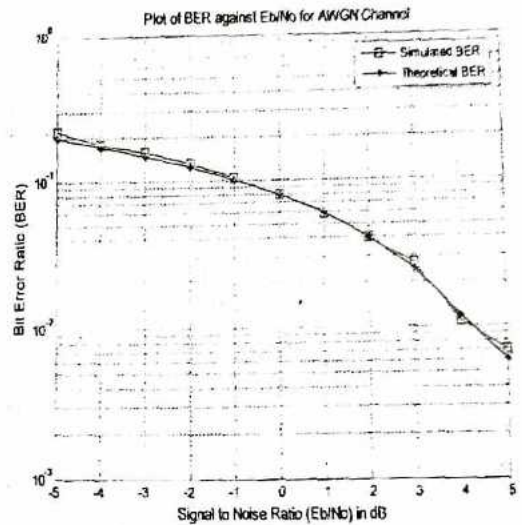


Fig 2: Theoretical and Simulated BER curve

Fig 2 was simulated for a burst length of 1000 bits, without fading, and the BER was calculated for different values of E_b/N_0 . Theoretical BER was also calculated using equation (5).

The simulated and theoretical plots of the BER vs. E_b/N_0 were plotted from Table 1 and as seen from the plots, the theoretical and simulated curves were almost coinciding, thus providing some validation that the physical layer simulation of a basic wireless communication system is correct.

Table 1: Theoretical and Simulated BER

E_b/N_0 in Db	-5	-4	-3	-2	-1	0	1	2	3	4	5
BER theory	0.201	0.173	0.150	0.125	0.101	0.080	0.060	0.040	0.025	0.012	0.006
BER Simulated	0.221	0.181	0.160	0.132	0.105	0.080	0.059	0.040	0.027	0.011	0.007

The physical layer block (PLB) of a wireless system with the Signal Processing Block (SPB) at the receiver end, was then simulated in the presence of

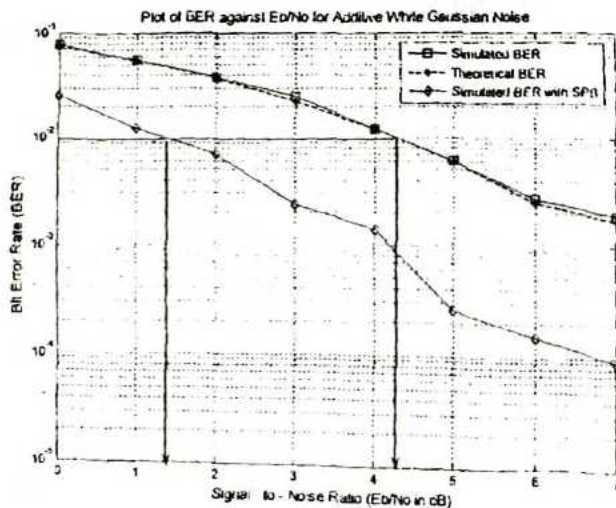
- Additive White Gaussian channel (AWGN)
- Rician fading channel and
- Channel coding

The signal Processing Block in this system acted as both an amplifying and filtering element. It ensured good signal output and hence allowed an improved BER. It allowed simulation of different layer configurations resulting in fast comparison and output processing.

Since the work is mainly based on the improvement in the BER, the results are plots of BER vs. E_b/N_0 . The received information bits were compared with the transmitted bits, to calculate the bit error rate for different values of the signal to noise ratio.

Table 2: BER and E_b/N_0 for AWGN channel

E_b/N_0 in dB	0	1	2	3	4	5	6	7
Theoretical BER	0.0755	0.0546	0.0370	0.0225	0.0122	0.0060	0.0025	0.0017
Simulated BER without SPB	0.0783	0.0555	0.0380	0.0252	0.0124	0.0064	0.0028	0.0020
Simulated BER with SPB	0.0266	0.0128	0.0078	0.0028	0.0014	0.0003	0.0001	0.0001



Simulations were carried out using MATLAB version 7.0.4. The following sections would be used to present the BER obtained with the SPB for different configurations of the physical layer.

5.0 Physical Layer Simulation of a Wireless System with the SPB for AWGN Channel

The SPB incorporated in a wireless communication system, was simulated (without fading) for a burst length of 1000 bits and the results for the BER for different values of E_b/N_0 were calculated. These are shown in tables 2 – 5 and figs 3. In the plot (fig 3), 'Simulated BER with SPB' was obtained in presence of the SPB and 'Simulated BER without SPB' was obtained in the absence of the SPB. 'Theoretical' is the plot of theoretical BER with AWGN which as depicted, served as the baseline to compare the performance. The values of BER for different values of E_b/N_0 for this simulation are shown in the table 2.

Fig 3: BER vs. E_b/N_0 for additive White Gaussian channel

With the SPB incorporated at the receiver section, the BER improved as shown in fig 3. The difference in SNR observed between the Simulated BER and that with incorporation of SPB for Bit Error Rate of 10^{-2} is $4.25 - 1.4 = 2.85$ dB. It was thus observed that a coding gain of approximately 3dB could be obtained.

6.0 Physical Layer Simulation of a Wireless System with the SPB for Rician Channel

Here the effect of fading was considered. Rician fading was modeled. The BER curves were plotted for different values of Rician factor (K).

Fig 4 was plotted for $K = 3\text{dB}$, a Doppler shift (f_d) of 200 Hz and a burst length of

1000 bits. Fig 6 was plotted for $K = 12\text{dB}$, $f_d = 200\text{ Hz}$ and a burst length of 1000 bits. The values of the BER were tabulated for both cases.

Table 3: BER and E_b/N_0 for Rician fading with $K = 3\text{dB}$

E_b/N_0 in dB	0	1	2	3	4	5	6	7
Simulated BER without SPB	0.1412	0.1152	0.0989	0.0787	0.0659	0.0501	0.0369	0.0280
Simulated BER with SPB	0.0842	0.0629	0.0467	0.0377	0.0336	0.0234	0.0222	0.0142

The difference in SNR for Rician fading of 3dB observed between the simulated BER and that with the incorporation of SPB in fig 4 is $3.00 - 0.22 = 2.78\text{dB}$.

Table 4: BER and E_b/N_0 for Rician fading of $K = 12\text{dB}$

E_b/N_0 in dB	0	1	2	3	4	5	6	7
Simulated BER without SPB	0.0879	0.0721	0.0479	0.0312	0.0194	0.0113	0.0064	0.0031
Simulated BER with SPB	0.0335	0.0221	0.0140	0.0062	0.0040	0.0021	0.0009	0.0004

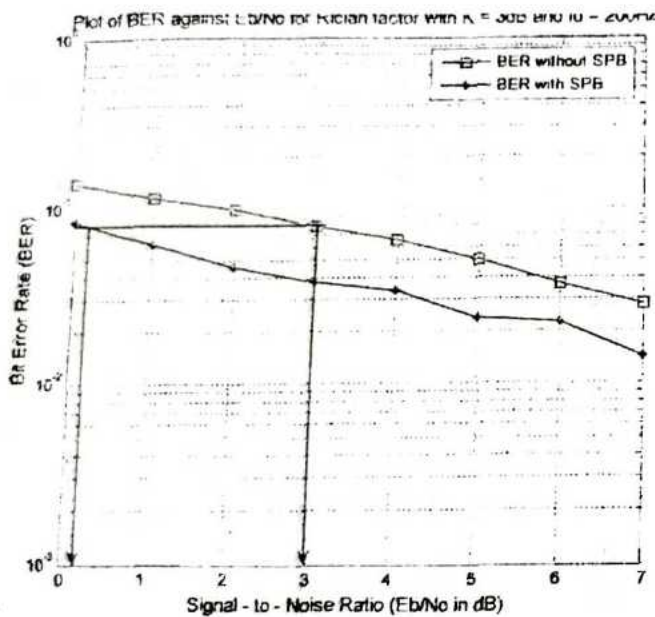


Fig 4: BER vs. E_b/N_0 for Rician Channel with $K = 3\text{dB}$

The difference in SNR for Rician fading of 12dB observed between the simulated BER and that with the incorporation of SPB in figure 5 is $5.28 - 2.45 = 2.83\text{dB}$.

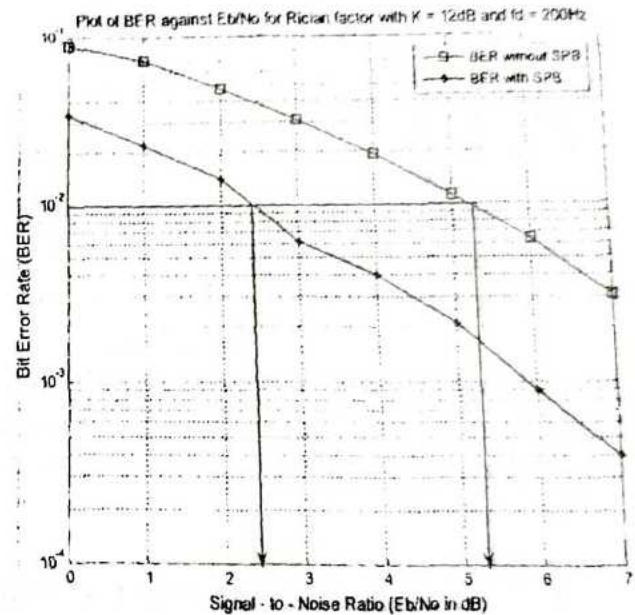


Fig 5: BER vs. E_b/N_0 for Rician channel with $K = 12\text{dB}$

Fig. 4 and 5 are plots of BER v/s signal to noise ratio (E_b/N_0) obtained by simulating the SPB in the presence of channel fading and AWGN. It was thus observed that a coding gain of approximately 3dB could also be obtained

Table 5: BER and E_b/N_0 for Rician fading and channel coding

E_b/N_0 in dB	0	1	2	3	4	5	6	7
Simulated BER without SPB	0.1403	0.1047	0.0702	0.0495	0.0137	0.0078	0.0049	0.0024
Simulated BER with SPB	0.0294	0.0208	0.0084	0.0037	0.0020	0.0010	0.0007	0.0004

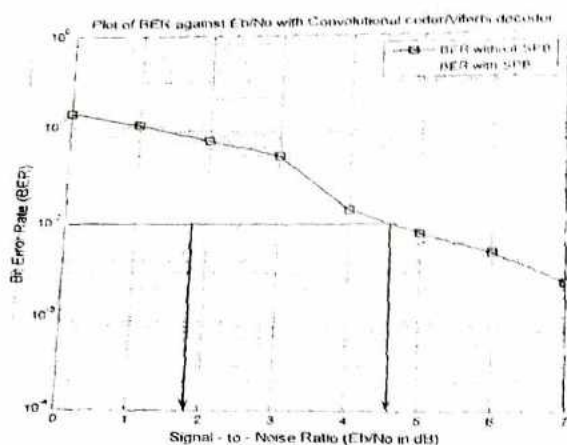


Fig 6: BER vs. E_b/N_0 for Rician channel and channel coding

Table 6: Summary Table for BER for different values of E_b/N_0 with the SPB incorporated

E_b/N_0 in dB	0	1	2	3	4	5	6	7
BER AWGN	0.0266	0.0128	0.0078	0.0028	0.0014	0.0003	0.0001	0.0001
BER Fading	0.0842	0.0629	0.0467	0.0377	0.0336	0.0234	0.0222	0.0142
BER Coding	0.0294	0.0208	0.0084	0.0037	0.0020	0.0010	0.0007	0.0004

7.0 Conclusion

In this work, the Signal Processing Block (SPB) was introduced at the receiver end of the physical layer of a wireless system and then simulated. The simulation results proved that an improvement in the BER is achieved over many typical existing configurations of the physical layer a GSM wireless system.

This improvement in BER will:

- Help in improving the GSM system performance by reducing the BER and thereby potentially reducing call drops due to high bit error rate.
- Improve coding gain over an existing GSM system.
- Potentially help to extend the range of coverage area.
- Improve the overall QoS.

This work tried to give an overview of a technique for improving bit error rate performance of a wireless communication system. It is a standard that would ensure interoperability without stifling competition and innovation among

The difference in SNR observed between the simulated BER and that with the incorporation of SPB for Bit Error Rate of 10^{-2} is $4.65 - 1.85 = 2.80\text{dB}$.

Even though the Rician factor of 3 and Doppler shift 200 Hz were considered, it is seen that with the simulation of channel coding/decoding, the BER was significantly improved. The use of the SPB with channel coding further reduces the BER.

In summary, the BER for different values of E_b/N_0 with our SPB for the different configurations of the physical layer of a wireless communication system can be tabulated as shown in Table 6.

suppliers, to the benefit of the public both in terms of cost and service quality.

Even though using simulation results, the use of the SPB was verified in noise reduction, the real time implications of the SPB still needs to be investigated.

The simulation of the physical layer of a wireless communication system (WCS) with the SPB incorporated in it, took approximately 5 minutes in MATLAB. Moreover, the SPB utilized iterative 'gradient ascent method' for noise reduction. Hence, verification should be performed to ensure that this method can be implemented in real time.

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