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Empirical Mode Decomposition for BER Improvement in Cellular Network

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Abstract: Call drop out is one of the most annoying problems in mobile communications. Over the years, many strategies have been proposed to solve the problem of call drop out, but it is still prevalent. One of the important reasons for call drop outs is high Bit Error Rate (BER). In this study, our intent is to reduce the call drop out by decreasing the BER based on Empirical Mode Decomposition (EMD). Thereafter, we introduce a new signal processing subsystem at the receiver section to decrease BER and thereby improve the end-to-end performance of the system. Our simulation is valid specifically for Code Division Multiple Access (CDMA) with QPSK modulation, although it can be extended to any cellular network. Our simulation proves that the new signal processing subsystem improves the BER performance.

Key words: EMD, cellular network, BER, CDMA, QPSK

INTRODUCTION

Cell phones keep getting fancier, but the old problems never seem to go away. Today we can get mobile phones that let you browse the Web, locate the nearest restaurant or even watch live TV. However, the old problems of call drop outs remain.

Call drop out is a measure of the ability of a mobile network to maintain a call until it is terminated. The call drop our rate is measured as the amount of call drop outs as a percentage of calls generated in whole network. The lower the percentage, the better is the QoS.

The main reasons for call drop out related to BER are:

- **Transmission problem:** If transmission is not perfect i.e., high BER is observed, then QoS provided is not good. When the service quality drops below a threshold the call is dropped (Liu and Zhang, 2003)
- **Hand-over:** Hand-over or Handoff is the process of changing the channel associated with the current connection while a call is in progress. Hand over is initiated when the BER of serving base station increases above a certain threshold (Kumar and Jack, 1994). If hand over between two sectors is not well defined, it leads to call drop outs
- **Large hysteresis value during handoff:** The Received Signal Strength (RSS) measurement is one of the most common criteria to initiate a handoff. Handoff is initiated if the RSS of the new Base Station (BS) is sufficiently stronger by a hysteresis value than that of the serving BS. If hysteresis is too large, the long handoff delay can cause the signal strength to be weak which result in a dropped-call (Zhu and Kwak, 2006)

To reduce BER, several signal processing techniques such as Adaptive Noise Cancellation (ANC) are commonly used when enhancing speech sequences, as opposed to fixed linear filters such as the Wiener filter (Moon and Stirling, 2000). This is the case because the adaptive filter does not require prior knowledge of either the signal or the contaminating noise source. Alternative approaches exits (Hamacher *et al*., 2005), however in the case of non-stationary signals such as speech, a multi-resolution approach which incorporates the Empirical Mode Decomposition (EMD) may be effective. By performing a sifting process, the EMD decomposes the desired signal into Intrinsic Mode Functions (IMFs) which are data-adaptive as opposed to other transforms which use predefined basis functions. But EMD is a relatively new, data-driven adaptive technique for analyzing multi-component signals. Although, it has many interesting features and often exhibits an ability to decompose nonlinear and non-stationary signals, it lacks a strong theoretical basis which would allow a performance analysis and hence the enhancement and optimization of the method in a systematic way. Some theoretical analysis can be seen in Tanaka and Mandic (2007) and Kopsinis and McLaughlin (2008).

In this study, a novel signal processing subsystem is developed that could be placed at the appropriate point within the receiver block diagram sequence. As we take

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CDMA network as our experimental subject, the data which consists of a sequence of 1's and 0's is modulated using QPSK. The result can be extended to other cellular network such as Global System for Mobile Communications (GSM).

EMPIRICAL MODE DECOMPOSITION

Empirical Mode Decomposition is a non-linear technique for analyzing and representing non-stationary signals. Detailed analysis can be found (Norden *et al*., 1998; Rilling *et al*., 2003). The EMD is data-driven and decomposes a signal, in the time domain, into a complete and finite set of adaptive basis functions which are defined as Intrinsic Mode Functions (IMFs). Although, these IMFs are not predefined as is the case with the Fourier and the Wavelet Transforms, the IMFs that are extracted are oscillatory and have no DC component. The EMD algorithm examines the signal between two consecutive extrema (e.g., minima) and picks out the high frequency component that exists between these two points. The remaining local, low frequency component can then be found. The motivation behind the EMD is to perform this procedure on the entire signal and then to iterate on the residual low frequency parts. This allows identification of the different oscillatory modes that exist in the signal. The IMFs found must be symmetric with respect to local zero means and have the same number of zero crossings and extrema. These criteria lead to the development of the EMD (Norden *et al*., 1998) in forming IMFs as described by the stages in the following algorithm:

- Identify all extrema of the signal x(t)
- Interpolate between minima/maxima producing envelope $e_{min}(t)/e_{max}(t)$
- Compute the mean:

$$m(t) = \frac{e_{min}(t) + e_{max}(t)}{2}$$

- Extract the detail signal, d(t) = x(t)-m(t)
- Sift detail signal, d(t), by performing steps (i) to (iv) until it has zero mean based on some stopping criteria (Rilling *et al*., 2003), thus generating an IMF
- Iterate (finitely) on the residual m(t) until subsequent IMFs have been extracted

By use of the EMD, the frequency information is embedded in the IMFs.

Since voice is the main payload of cellular network, we consider the model described by Eq. 1:

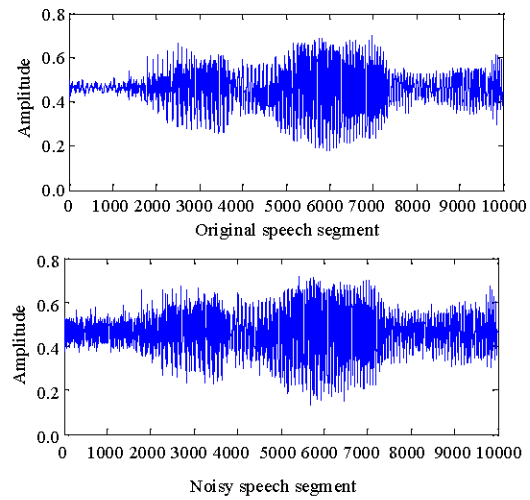

Fig. 1: Segment of original speech utterance

$$x[n] = s[n] + t[n] \tag{1}$$

where, x[n] is the noisy speech signal, s[n] is the original noise-free speech and t[n] is the noise source. Figure 1 shows a clean speech segment and a noisy segment (t being zero mean, AWGN with SNR = 25). The EMD of this utterance is plotted in Fig. 2 and 3. In both cases, the first six IMFs are illustrated since most of the signal energy exists within these.

Comparison of the IMF plots reveals that when noise is added to the clean speech, the first few IMFs contain most of the noise energy as well as some of the speech. What we need to do is to distinguish which IMF contain the speech or noise, that would determine which IMF should be reserved or canceled for the de-noise purpose.

However, it can also be seen that the EMD decomposition pushes a significant amount of the speech energy to latter IMFs along with some residual noise. The reconstruction process is given in Eq. 2, which involves combining the N IMFs and the residual r[n]:

$$x[n] = \sum_{i=1}^{N} IMF[n] + r[n] \tag{2}$$

EMD BASED SIGNAL PROCESSING SUBSYSTEM FOR CELLULAR NETWORK

The signal flow is shown in Fig. 4. A decision boundary needs to be defined for the mapping purpose at the demodulator. Based on the decision boundaries, the detector decides what the received symbol is and accordingly, converts that symbol into bits. The detection

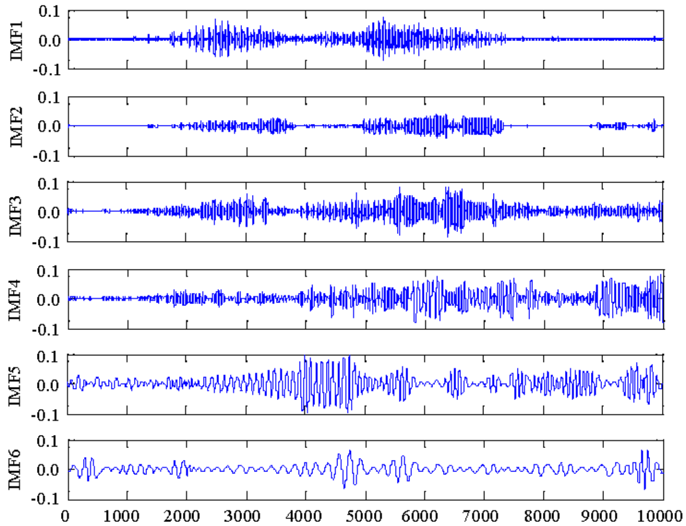

Fig. 2: IMF plot of clean speech segment

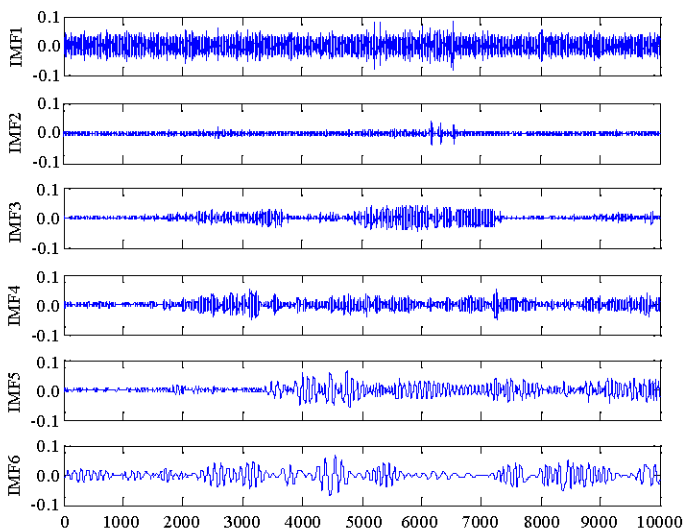

Fig. 3: IMF plot of noisy speech segment

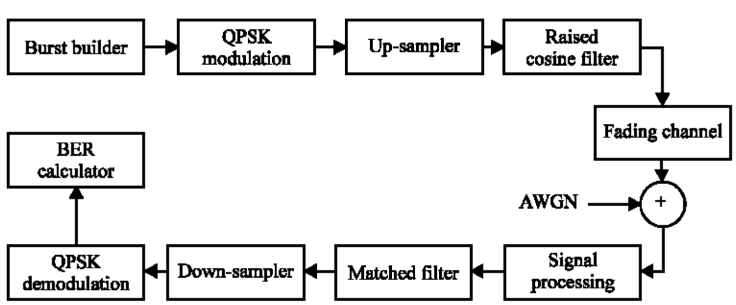

Fig. 4: Physical layer wireless communication system with the signal processing subsystem

process can go wrong in the presence of noise. Hence, the noise reduction blocks are always placed before the demodulator of the receiver section of a wireless system, preferably at the beginning of the receiver. Digital modulation maps input bits to symbols. Phase Shift Keying (PSK) is a popular digital modulating technique that conveys data by changing the phase of a reference signal. The general analytic expression for PSK (Proakis, 1995) is given below:

$$s_i(t) = \sqrt{\frac{2E}{T}}\cos[\omega_0 t + \phi_i(t)] \quad 0 \le t \le T, i = 1,\ldots,M \tag{3}$$

where, the phase term $\phi_i(t)$ will have M discrete values, typically given by:

$$\phi_i(t) = \frac{2\pi i}{M}, \qquad i = 1,\ldots,M \tag{4}$$

E is the symbol energy and T is symbol time duration for $0 \le t \le T$. The most popular and widely used form of PSK is Quadrature Phase Shift Keying (QPSK). QPSK changes the phase of transmitted waveform. Each finite phase change represents unique digital data. A QPSK modulated carrier undergoes four distinct changes in phase that are represented as symbols and can take on the values of 0, 0, $\pi/2$, π and $3\pi/2$.

Since, we are dealing with the improvement of the BER, it is important that we know the BER for the existing QPSK modulation. The symbol error rate for the QPSK modulation with AWGN channel is given as in Eq. 5:

$$P_M = 2Q(\sqrt{\frac{2Eb}{N_0}})[1 - 2Q(\sqrt{\frac{2Eb}{N_0}})] \tag{5}$$

According to the relation between the BER and symbol error rate we have:

$$P_b = \frac{1}{k}P_M \tag{6}$$

where, P_b is bit error rate, P_M is symbol error rate and k is number of bits per symbol.

Hence, for QPSK, M=4, k=2. Thus, the BER for QPSK with the AWGN channel is given as in Eq. 7:

$$P_b = Q(\sqrt{\frac{2Eb}{N_0}})[1 - 2Q(\sqrt{\frac{2Eb}{N_0}})] \tag{7}$$

The QPSK modulated signal is the input of our signal processing subsystem based on EMD algorithm. The subsystem yields several IMFs including signal and noise. We must decide which IMF is the signal one. For all IMFs, transform them with FFT (Fast Fourier Transform) to find the most mach one with QPSK signal and this IMF is what we expect.

The Eb/N_0 and SNR (signal-to-noise ratio) is defined as in Eq. 8:

$$SNR = \frac{Eb}{N_0} \times \frac{R}{W} \tag{8}$$

where, R denotes bit rate and W system bandwidth. For CDMA system, R = 307.2 kbit sec^{-1}, W = 1.25 MHz.

SNR is defined as in Eq. 9:

$$SNR = 10\log_{10}\frac{\sum_{n=1}^{N} s_k^2(n)}{\sum_{n=1}^{N}(x_k(n) - s_k(n))^2} \tag{9}$$

SIMULATION RESULT AND ANALYSIS

We take pure and noise-added voice data to validate our proposed method compare. As shown in Fig. 1, the upper subfigure is original voice without noise and the lower one is noise pollution voice segment. All data for experiment is collected from real world. The QPSK modulated signal, with AWGN QPSK noisy signal and the FFT transform of the noisy signal is shown in Fig. 5. We plot first six IMFs and their FFT as shown in Fig. 6 and 7.

From Fig. 5 and 7, we can see IMF1 contains most of the noise energy and IMF2 would be the main component which has the most likely FFT as input QPSK signal. With various SNR, the main component would not be fixed, but always in first 3 IMFs.

Now that we distinct the signal and noise, it is easy to improve SNR. BER is related with SNR, the improved BER is shown in Fig. 8.

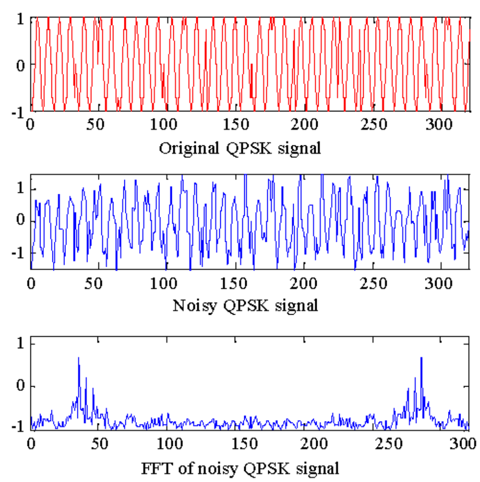

Fig. 5: QPSK modulated signal and FFT transform

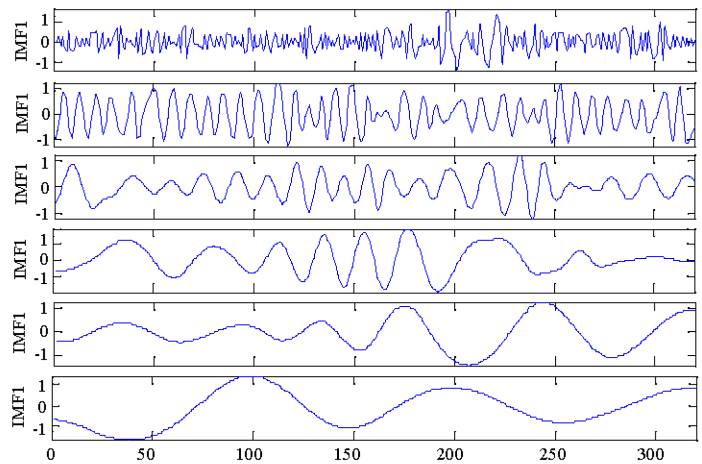

Fig. 6: IMFs of first six layers

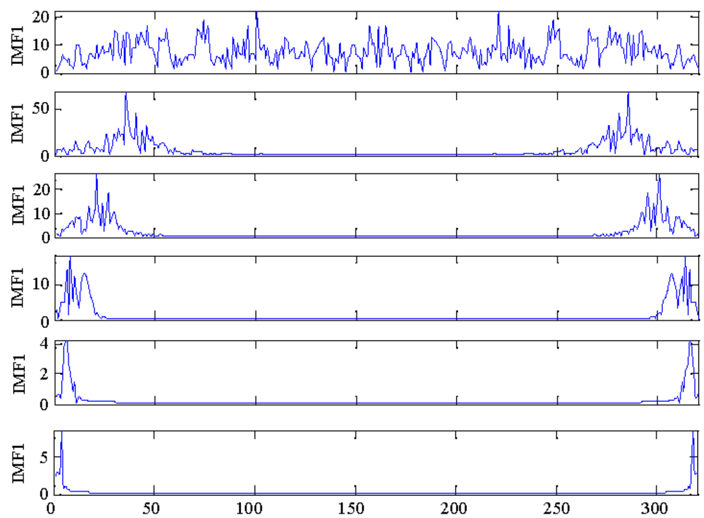

Fig. 7: FFT of IMFs

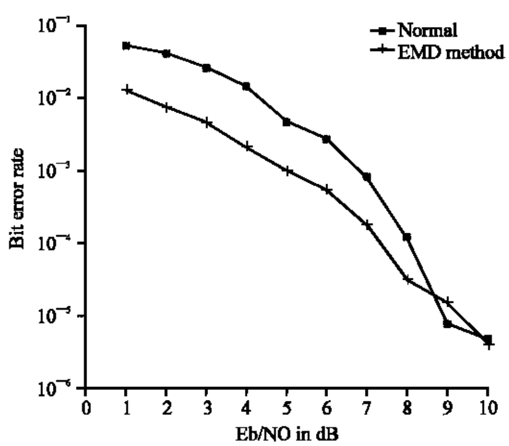

Fig. 8: BER vs. Eb/No

Figure 8 shows that in low Eb/No, which means signal under the background of strong noise, EMD method has obviously improved BER performance-approximately 3 dB gain. When Eb/No is more than 10dB, EMD method has no apparent improvement.

CONCLUSION AND FUTURE WORK

In this study, we introduce a novel signal processing subsystem based on EMD method. The subsystem can be used for noise reduction in a wireless communication system, especially for CDMA network. As seen from the analysis presented, after decomposing the signal into its IMFs using the EMD, the subsystem refines these IMFs that are formed. The main component of IMFs can then be used to reconstruct the signal thus offering superior results. Based on the simulation results it concludes that the addition of the subsystem at the receiver section improves the BER performance. The BER improvement will:

- Help in solving the problem related to call drop outs because of the high BER
- Improve the overall QoS
- Potentially help in the extension of the range of a base station
- Provide a coding gain

The receiver section of the mobile handsets could incorporate the subsystem and improve the call reception quality.

We have developed our simulation for a CDMA wireless system. This work could be extended to the GSM systems or other cellular network. And the EMD technique has a great deal of potential and it is not only limited to the domain of speech enhancement. The principles may be applied and integrated into a wide range of digital communication.

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