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Title of Thesis: "Development of Signal Processing Scheme to Enhance the Overall Performance of EDGE and GPRS Enabled GSM Receiver."

Abstract

Global System for Mobile (GSM) System is a 2nd Generation cellular mobile communication standard that has encompassed almost the entire globe. The remarkable growth of this system encouraged us to work in this field. The primary goal of a GSM system is the transmission/reception of high quality speech through digital signal processing chain in its physical layer of OSI model. The aim of this work is to develop a signal processing scheme for GPRS/EDGE enabled GSM system to enhance the quality of speech and improvise its transmission/reception capabilities.

After literature survey related to GSM and its signal processing, we decided to concentrate on speech processing, channel coding and modulation sections. Since speech is the main entity of the system, more emphasis is given to speech quality and its enhancement. The complexity increases when an attempt is made to enhance the speech quality. Thus, we intend to develop a signal processing scheme that limits the overall complexity after enhancing the speech quality. The channel coding scheme is modified to limit the overall complexity of the system after enhancing the speech quality.

Modified Adaptive Multi Rate-Wide Band (AMR-WB) speech codec is chosen for the scheme to have the best possible speech quality. This is possible by using source-controlled (SC) rate adaptation for speech and channel coding instead of using the current channel conditions. In SC approach, the rate adaptation unit estimates the

quality of input speech on regular basis. If the quality of speech is good then the codec adapts a lower bit rate and channel coder adapts a higher bit rate. It reduces the complexity of rate adaptation block, lessens the time required for rate adaptation process and enhances the capacity of the network.

This scheme improves the quality of the system a bit and reduces the complexity of the rate adaptation unit but increases complexity of the speech coder. To counter the increase in the complexity of the overall system, the Classical Viterbi Algorithm (VA) channel decoding scheme is modified to have Reduce-complexity Viterbi Algorithm (RVA). This reduces the complexity of the system but at the cost of bit error rate (BER).

The proposed scheme is simulated and output data is plotted. To compare the performance of the proposed scheme, simulation is carried out with four different combinations of speech coders (standard AMR-WB (identified as AMR) and modified AMR-WB (identified as SC)) and channel decoders (VA and RVA). These combinations are: AMR and VA, AMR and RVA, SC and VA, SC and RVA.

The performance of these four processing schemes is compared to get the best possible performance. To judge the performance of the proposed signal processing schemes, the Objective Evaluation of speech is conducted for Absolute Error, Mean Square Error, Signal-to-Noise Ratio and Bit Error Rate for all nine data rates of AMR-WM codec using for all four combinations in Typical Urban channel environment with a mobile speed of 50 km/h. The results indicated that the performance of SC and VA combination is the best.