PERFORMANCE IMPROVEMENT OF GSM TRAFFIC USING ADAPTIVE MULTI RATE (AMR)

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ABSTRACT

This paper describes the implementation of the Adaptive Multi Rate (AMR) speech codec for improving the performance of GSM traffic and the speech service. In other words, the AMR is used to reallocate the existing speech codec configuration to offer higher percentage of call success among mobile user with AMR capability.

In this paper, the trial of AMR implementation process in Cyberjaya area have been explained and discussed. An application software is developed using Visual Basic 6 to analyze the data obtained from the project and to assist traffic personnel to monitor traffic utilization in implementing AMR at any given area and date .

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CHAPTER 1

INTRODUCTION

1.1 Introduction

GSM is the way in which our mobile phone communicates with the cell antenna that is operated by the network provider. The signal then is being encoded and decoded at the base station, and is transmitted through wire to connect to regular phones or even another antenna. Meanwhile, mobile phones do use audio codecs and audio frames are decoded at the antenna then are sent back and forth to the mobile phone.

GSM has several types of codecs. The most common is called GSM-FR (GSM Full Rate) and GSM-HR (Half Rate). Later, a GSM-EFR (Enhanced Full Rate) was added with a bit-rate of 13 kbps (same as GSM-FR), but with a much better quality. The reason that EFR sounds better than FR is that FR uses too little redundancy for error detection. The bit-rate on the radio interface is 22.8 kbps and in FR 13 kbps is used for audio data and the rest for error correction and detection. So the speech quality improvement is due to improved transfer of audio data over the radio interface.

The next generation of audio codec for GSM is AMR (Adaptive Multi Rate) and uses 14 different codecs, 8 for Full Rate mode and 6 for Half Rate mode. The bit-rates of these codecs go from 4.75 kbps to 12.2 kbps. Depending on how 'bad' the radio channel is the amount of redundancy needed to assure 'secure' transfer of audio data is calculated. The word 'secure' here means that the bit error rate is kept below a certain level. If the radio channel is good then less redundancy is needed and more bits can be spent on audio data, hence a high bit-rate codec like 12.2 might be chosen. If the radio channel is bad then much redundancy is needed