PREFACE

Wireless or cellular mobile communication system has been evolving according to the advancement in the wireless technology and change in the user demand [2]. The ability to communicate with the people on move has evolved remarkably in the present days. The services offered by the mobile communication [3] system are enormous and hence the mobile subscribers are increasing at a larger rate. But the bandwidth allocated for the wireless communication is very much limited. The bandwidth is a precious commodity in wireless communication system. The demand for the improved services like data, video and internet has resulted in large spectral bandwidth which has forced the service provider to go for change in the cell architecture or change in the technology. The change in cellular architecture can be achieved by cell sectoring, cell splitting and by using overlaid cell schemes to meet the increased in demand of wireless services [2]. This improved the performance and gained remarkable acceptance by subscribers. The change in cell architecture alone however could not meet the demand for the improved services due to the large growth in the traffic density. This resulted in need for change in technology to meet the demand. One of the technological modifications is compression technique that can try to satisfy the demand to the possible extent. The compression technique can be either audio compression for speech & data or video compression for pictures. The present work is limited to audio compression in particular to speech compression. Different types of coders are used in speech compression. The basic requirement of coder is to maintain the toll quality of the speech and error robustness. One such coder with good toll quality and low bit rate is a hybrid coder [25]. Most frequently used hybrid coder is code excited linear predictor [CELP]. The CELP [11] exploits a perceptual quality criterion which enables it to offer superior quality speech signal but with a high processing delay. The CELP uses long-term predictors
which are very sensitive to the channel noise. Hence to overcome these drawbacks we are proposing speech compression technique using LD-CELP. This has low processing delay compared to the algorithms used in the GSM system or CDMA2000. The processing delay of LD-CELP is less than 2msec. The advantage of this algorithm is it’s

- Robustness to the channel error.
- Retains the perceptual quality of the signal.
- Low computational complexity.
- Low O/P bit rate.

The speech coded [25] data is forward error protected by using adaptive channel coding technique. The main objective of the adaptive channel [16, 17] coder is to select the channel encoder according to the noise level of the channel. In this thesis we propose a joint coding algorithm using which we can increase the capacity of the wireless system and improve the S/N ratio of the signal received at the receiver. The adaptive channel coder is used for error detection and correction and further it also protects the data. The type of the channel encoding scheme mainly depends on the signal to noise ratio of the channel to be used, signal to noise ratio at the input of the receiver and the data rate to be transmitted. In practical systems the maximal signal power and the bandwidth of the channel are restricted and noise power spectral density is also fixed for a particular operating environment. The only practical alternative to reduce the probability of the error is to use the channel coding technique [36]. The Channel coding [40] is a viable method to reduce information rate through the channel and increase reliability. This is to improve the capacity of a channel by adding carefully designed redundant information to the data being transmitted through the channel. In the present work adaptive channel coders used. The
adaptive channel coding technique designed in the present work will provide good speech quality over a wide range of channel conditions. The two types of the channel coder used in adaptive channel coder are Cyclic coder [39] and Convolutional coder [43]. The channel coding technique chosen mainly depends on the S/N of the channel through which we want to transmit the data. In this adaptive algorithm the first step is to check for the noise level in the channel. Depending on the noise level of the channel the algorithm designed automatically selects one of the channel coding techniques. The adaptation of the source coding rate and the channel coding rate can occur within a speech frame. For a good channel with less noise, less number of check bits are added and hence allowing large number of message bits to be transmitted. If the channel is less erroneous then we use Cyclic coding technique else we use Convolutional coding technique. By this methodology we can accommodate two users within one full rate traffic channel. Hence the channel efficiency is improved. The joint source and channel coder is robust to the noise present in the channel and supports variable coding rates to accommodate maximum number of users which is one of the important requirements for mobile communication. The proposed algorithm gives good S/N ratio at the receiver irrespective of the channel conditions and due to its low processing delay it prevents the use of echo cancellers which is mainly required for the real time application.