ABSTRACT

Today, the primary constrain in wireless communication system is limited bandwidth and power. Wireless systems involved in transmission of speech envisage that efficient and effective methods be developed (bandwidth usage & power) to transmit and receive the same while maintaining quality-of-speech, especially at the receiving end. Speech coding is a technique, since the era of digitization (digital) and computerization (computational and processing horsepower - DSP) that has been a material-of-research for quite sometime amongst the scientific and academic community. Amongst all elements of the communication system (transmitter, channel and receiver), transmission channel (carrier of information/data, also called the medium) is the most critical and plays a key role in the transmission and reception of information/data. Channel conditions decide the quality of speech at receiver. Modeling a channel is a complex task. Many techniques are adopted to mitigate the effect of the channel. AMR (Adaptive Multi Rate) is one such technique that counteracts the deleterious effect of the channel on speech. This technique employs variable bit rate that dynamically switches to specific modes of operation (switching bit rates – called modes of operation) depending upon the channel conditions. For example, Low bit rate mode of operation is selected in adverse channel conditions, this helps to provide more error protection bits for channel coding and vice versa. AMR shares advantages like moderately high Mean Opinion Scores and moderate compression ratios but suffers with high computational complexity and coding delay in comparison with other standard speech coders [1]. Here, in this paper, application of Code Excited Linear Prediction (CELP) source coder on speech followed by AMR codec is investigated and studied. An e-test bench using MATLAB is created to implement the CELP based AMR Codec scheme, and the same studied and investigated through a series of simulation. The results of the simulations are recorded and compared in various graph, this includes SNR Vs Bit rates, Absolute Error Vs Bit rates, Mean Square Error Vs Bit rates and Perceptual Evaluation of Speech Quality [10]. Simulation results clearly advocate that, it is possible to produce variable bit-rates (tuning to channel conditions) in CELP coder by affecting coefficients of the coder while still maintaining a good quality of speech. Further, higher the bit-rate used, the better is the quality of speech.

1. INTRODUCTION

The world-wide coverage of GSM-like networks led to explosion of the number of subscribers, causing the saturation of some networks. Besides, the old technology used in GSM (Global System for Mobile Communication) systems provided the customer with an unsatisfactory speech quality in many common situations. At the same time advances in DSP (Digital Signal Processing), computing capabilities and the state of the art in speech and channel coding have improved significantly. Therefore, to increase the capacity and the quality of GSM networks, ETSI (European Telecommunication Standards Institution) decided to standardize by bringing in a new speech transmission system called AMR. AMR is Adaptive Multi Rate system, that is a combination of speech and channel Codecs triggered and controlled by signaling means aimed at providing the best speech quality under background noise and transmission errors. Research in the area of source coding techniques, that include CELP (Code Excited Linear Prediction), ACELP (Algebraic CELP), RPE-LTP (Regular Pulse Excited linear predictive coder with a Long Term Predictor) still continue worldwide. The application of source and channel coding help mitigate the dependence of voice quality on channel condition, AMR is one remedy to this channel problem. In this paper, GSM AMR 06.90 Coder is implemented and simulated in MATLAB environment using CELP Algorithm. Further various objective evaluations are carried out on the proposed system to judge its performance. This paper is organized as follows. In section 2, AMR Coder is introduced. In section 3, CELP coder is described whose quantization bits are affected to be adjusted to AMR Codec mode bitrates. In section 4, various objective evaluation measures have been touched upon. In section 5, we describe MATLAB simulation e-test bench for the proposed coder. In section 6, performance evaluation of proposed coder is computed and demonstrated using set of graphs. Finally the concluding remarks are given in section 7.
2. ADAPTIVE MULTI RATE CODER

AMR is a technique that is utilized to maintain good speech quality under varying channel conditions. AMR has several operating codec modes which are switched adaptively in accordance to the dynamics of the channel (good or bad channel). The process of dynamically switching due to varying channel conditions is known as AMR adaptation [6]. AMR has several codec modes, for example; AMR full-rate and AMR half-rate. AMR full-rate works with the highest bit rate of 22.8 kb/s; whereas AMR half-rate works with the highest bit rate of 11.4 kb/s. As shown in Fig. 1, AMR full-rate has 8 codec modes operated in kbps, i.e.: 12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15, and 4.75; whereas AMR half-rate has 6 codec modes operated in kb/s, i.e.: 7.95, 7.4, 6.7, 5.9, 5.15, and 4.75. The above codec modes are a precipitate of the ETSI standardization in 1999.

Fig. 2 is a block diagram of AMR codec operation. AMR working is a two step process; uplink channel and downlink channel. In uplink channel, MS sends information to BTS which contains speech data, codec mode indicator uplink (MIu), and codec mode request downlink (MRd). Speech data is the main information sent from MS via BTS with bit rate determined by its speech coder. The speech data is sent along with its codec mode indicator uplink (MIu) and codec mode request downlink (MRd). This process is repeated continuously for the next speech data sent. MIu comes from codec mode command uplink (MCu) to determine codec mode which will be used for uplink transmission from MS to BTS. MRd comes from computation of error condition at downlink channel from BTS to MS (downlink quality measurement). After MRd arrives at BTS, the process of downlink mode control is carried out to produce codec mode. Uplink channel performance process is very dependent on downlink channel performance process, and vice versa. In downlink channel, BTS send information to MS which contains codec mode command uplink (MCu), speech data and codec mode indicator downlink (MId). At first, MCu comes from the computation of error condition in uplink transmission (uplink quality measurement), which is then processed through uplink mode control to produce codec mode command uplink (MCu). Arriving at MS, MCu is then used to give codec mode order which will be used for uplink transmission from MS to BTS. MId comes from codec mode command downlink (MCd) which is used to determine codec mode to be used for downlink transmission from BTS to MS.

The computation of the error condition quality in uplink and downlink channel (uplink quality measurement and downlink quality measurement on Fig. 2) is determined by the ratio of signal carrier (C) to interference (I), which expressed in dB. Here, the quality of channel condition has a value from 2 to 22 dB. A particular codec mode will be selected in accordance with its respective channel condition. The quality of channel condition is computed on the parts of channel measurement (Fig. 2), as well as on the mobile station and BTS. The working process of the uplink and downlink channel in AMR is going on repeatedly. This means that if there is a sudden change in the channel condition, then there will also be a change in the codec mode in accordance with the channel condition at that moment. Codec mode is determined by the quality of the channel condition. If the channel condition is good, then there will be no significant error correction, so that the bit rate speech coding sent is higher than the bit rate channel coding (error correction). On the contrary, if the channel condition is bad, then a significant error correction will be needed. In this case, the information sent will have much correction, so that it will produce good voice quality [6].

![Figure 1 AMR full-rate and half-rate codec](image-url)
3. CELP SPEECH CODER

The CELP block diagram is shown in Fig. 3. Voice signal input is the conversion result of human voice – an analog signal to digital signal which is carried out by ADC (Analog to Digital Converter). The buffer process and LP (Linear Prediction) analysis is used to estimate the vocal impulse response system at each frame, which then produces pitch delay and LP coefficient. The pitch delay will be used in the pitch synthesis filter and the LP coefficient \(a_i\) at LP synthesis filter. Before the process of pitch synthesis filter, pitch filter coefficient \(b\) will be produced first from the computation using the pitch delay \(P\). The process of LP synthesis filter before the processing of LP coefficient in the block is carried out by converting LP coefficient \(a_i\) to become the reflection coefficient of LP [5]. To obtain gain parameter \(\theta_0\) and codebook index \(k\), perceptual weighting filter process will be done and then error minimization process is carried out.
In the error minimization block, gain and codebook index which will be used in the next block will be determined [5]. After all the parameters are obtained, voice compression process can be carried out. Each block will work as shown in Fig. 3. Therefore, the compressed voice signal can be formed and adapted to the AMR codec mode.

4. OBJECTIVE EVALUATION OF PROPOSED CODER

In this paper waveform based and perceptual based analysis of proposed codec has been carried out.

4.1 Waveform based analysis

The following parameters are evaluated in this category.

1. Absolute Error (ABS) is mathematically defined as

   \[ \text{ABS Error} = \sum | S_i - S_o | \]  

2. Mean Square Error (MSE) is mathematically expressed as

   \[ \text{MSE} = \frac{\sum (S_i - S_o)^2}{N} \]  

3. Signal to Noise Ratio is mathematically defined as

   \[ \text{SNR} = 10 \log_{10} \frac{\sum S_i^2}{\sum (S_i - S_o)^2} \]  

Where \( S_i \) = input signal, \( S_o \) = decoded signal and \( N \) = total no. of frames

4.2 Perceptual based analysis

The following is the important parameter for performing perceptual based analysis.

1. Perceptual Evaluation of Speech Quality (PESQ)

   In comparison with other objective measures, the PESQ measure is the most complex to compute and is the one recommended by ITU-T P.862 for speech quality assessment of 3.2 kHz (narrow-band) handset telephony and narrow-band speech Codecs [10]. PESQ score is computed as a linear combination of the average disturbance value \( D_{\text{rand}} \) and the average asymmetrical disturbance values \( A_{\text{rand}} \) as follows:

   \[ \text{PESQ} = \alpha_0 + \alpha_1 D_{\text{rand}} + \alpha_2 A_{\text{rand}} \]  

Where \( \alpha_0 = 4.5, \alpha_1 = -0.1 \), and \( \alpha_2 = -0.038 \) [10].

5. SIMULATION OF CELP BASED AMR CODER USING MATLAB

To compare the performance of each AMR codec mode of operation (defining different bit rates), simulation using MATLAB program is carried out with bit allocation as shown in Table I. The simulation is carried out using CELP speech coding technique and is in sync to the bit rate used in AMR as per ETSI. Partial programming and tweeting in MATLAB helped to construct an e-test bench, that will help simulate and produce results through tables and graphs. The simulations considered Standard GSM Cellular System architectural configuration, having a 20ms mother frame, consisting of four sub-frames within the mother frame, with each sub-frame having a length of 5ms. As can be seen in Fig. 4, Speech Ss (i) and C/I ratio are provided as inputs to the MATLAB e-test bench of CELP based AMR Codec.

Depending upon the provided C/I ratio, the mode of AMR Codec is selected. MATLAB program has provision to continuously monitor the change in C/I ratio so that an appropriate AMR full rate mode is selected for subsequent frames [9]. After selection of Codec mode LP analysis is done and the parameters like Frame length (N), Block length (L), Order of filter (M), LP parameter (c), Codebook index (Cb) and Pitch index (Pidx) are computed and provided to CELP analysis part of AMR Codec. Information parameters, as a result of CELP analysis, relating to the AMR codec include: linear prediction coefficient (a), pitch lag (p), codebook index (k), gain (\( \theta_0 \)) and pitch filter coefficient (b), been investigated, studied and recorded through the simulation e-test bench created using MATLAB. The bit allocations of these five information parameters, using scalar and vector quantization, are shown (horizontally) in Table I. As each AMR codec mode has different bit allocation, the total bits is in accordance with the bit allocation for AMR codec as per ETSI standards 1999. Recovered speech Ss (i) can then be reproduced from these coded data bits by passing them through CELP synthesis filter as shown in Fig. 4.

IJAET/Vol.II/ Issue II/April-June, 2011/52-59
Table 1 AMR bitrate selection according to parameters of CELP Coder

<table>
<thead>
<tr>
<th>AMR mode (Kbps)</th>
<th>a</th>
<th>p</th>
<th>K</th>
<th>Θ₀</th>
<th>B</th>
<th>Total Bits</th>
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<tr>
<td>12.2</td>
<td>24</td>
<td>8,8,8,8</td>
<td>17,17,17,17</td>
<td>15,15,15,15</td>
<td>15,15,15,15</td>
<td>244</td>
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<td>10.2</td>
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<td>11,11,11,11</td>
<td>11,11,11,11</td>
<td>204</td>
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<td>7.95</td>
<td>23</td>
<td>8,8,8,8</td>
<td>12,12,12,12</td>
<td>7,7,7,7</td>
<td>7,7,7,7</td>
<td>159</td>
</tr>
<tr>
<td>7.4</td>
<td>24</td>
<td>8,8,8,8</td>
<td>11,11,11,11</td>
<td>7,7,7,7</td>
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<td>6.7</td>
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<td>8,8,8,8</td>
<td>10,10,10,10</td>
<td>5,5,5,5</td>
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<td>4,4,4,4</td>
<td>3,3,3,3</td>
<td>118</td>
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<tr>
<td>4.75</td>
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<td>7,7,7,7</td>
<td>2,2,2,2</td>
<td>2,2,2,2</td>
<td>95</td>
</tr>
</tbody>
</table>

Figure 4 MATLAB implementation of CELP based AMR codec

6. OBJECTIVE EVALUATION OF PROPOSED CODER
The objective performance evaluation of speech files includes calculation of parameters like Absolute Error, Mean Square Error, Signal to Noise Ratio and Perceptual Evaluation of Speech Quality respectively. Three wave files are used here for the purpose of this analysis, they are: Voice.wav, Five.wav and Doormono.wav. The Voice.wav having 12000 samples, while Five.wav and Doormono.wav having samples 4329 and 20071 respectively. Equations utilized to calculate the above parameters are as inked in section 4. MATLAB simulated graphical resulting plots are shown in Fig. 5, 6, 7 & 8. Results obtained by the objective analysis are found to be satisfactory as can be judged from figures cited at below.
Figure 5 Calculation of ABS ERR for different wave files at various bit-rates of AMR codec

![Absolute Error vs Bitrate (KBPS)](image)

Figure 6 Calculation of MSE for different wave files at various bit-rates of AMR codec

![Mean Square Error vs Bitrate (KBPS)](image)
7. DISCUSSIONS AND CONCLUSIONS
The results of the simulator study reveal that integrating scattered blocks (single-monolithic whole) makes it interactive (via feedback) that eventually helps to provide optimal solution. The present AMR techniques and advancements reveal the recent trend to develop telecommunication signal processing algorithms as a one monolithic whole [7], effectively, eliminating old standards that consist of several scattered independent processing blocks. Our simulation study is a step in this direction, which clear definition of integration of various functional blocks, as depicted in Figure 4. In stark contrast, the different blocks are now developed together as one.
monolithic whole, and interact (via feedback) with each other. This enables the designers to provide optimal solutions. Besides, AMR is a huge system that supplies a multiplicity of Codec’s to enable the GSM standard to adapt to the numerous conditions and applications in wireless communications. AMR increases the robustness under channel errors (due to changing channel conditions) and limits the degradation of speech quality under background noise as compared to the other GSM coders like Full Rate, Half Rate and Enhanced Full Rate Coders.

As can be seen from the obtained results and graphs, it is possible to produce variable bit-rates in CELP coder by changing coefficients of the coder. Despite the fact that insufficient total bit allocation has occurred due to the prediction signal, under the circumstances, the resulting quality of each codec mode is still good and can be heard clearly. The higher the bit-rate used, the better the speech quality. As seen in Fig.5, 6, 7 & 8 CELP based AMR Codec provides acceptable values for various parameters in the objective analysis part when implemented in MATLAB with different wave files. SNR and PESQ improve with increase in bit-rate from 4.75Kbps to 12.2Kbps. Also, reduction in Absolute Error and Mean Square Error is clearly visible with increase in bit-rates.

8. REFERENCES

4. L. Besacier, “GSM Speech Coding and Speaker Recognition”, University of Neuchatel, A.L.Breguet, Switzerland