8. RESULTS

8.1 RESULTS OBTAINED AT THE SOURCE CODER

The comparative results obtained for the input speech after passing through the source encoder, where the speech is compressed and at the receiver it is decompressed is as given below. For different sampling rates the processing time results obtained are as given below. The processing time required is very much less than the CCITT [1] requirement of 2 ms. Initially the order of the STP chosen was 10 and the MOS obtained for both the male and child were good but the quality of the female voice degraded because of high pitch. Hence to overcome this drawback we increased the order of the STP to 50 to retain the female pitch.

In plate 8.1, the red graph represents the input speech before compression at the transmitter and the green signal represents the output decompressed signal. The speech signal chosen was a female voice where in ‘y’ axis represents the amplitude and the ‘x’ axis represents the speech data. The female voice considered is sampled at 8 kHz. The order of the STP was varied from 10, 30 and 50. For 10th order STP the female voice was heard as male voice and for 30th order the transitions were not smooth. Only at 50th order we got desired output with good MOS.
Plate: 8.1 Comparative input and output of female speech signal at source coder

The plate 8.2 represents the input speech signal before compression and after decompression. The speech signal considered was male voice. The signal was sampled at 8 kHz. The obtained decompressed signal had good MOS and had smooth transition both at 10th order STP and also at 50th order STP.

Plate: 8.2 Comparative input and output of male voice at source coder

The plate 8.3 represents the graph obtained by considering a child’s voice as input signal. The MOS of decompressed child’s speech was good at 50th order STP.

Plate: 8.3 Comparative input and output of child speech signal at source coder
The results obtained by the speech coder were presented at IETE conference.

The table 8.1 below gives the comparative processing time required to compress the speech with different sample rates. The different sampling frequencies considered are 6 kHz, 8 kHz, 12 kHz and 16 kHz. The input data is compressed by 4:1 ratio. The quality of the signal slightly degrades at the sampling frequency of 4 kHz and below. These comparative results were published in the international journal [3]. According to the CCITT the source coding delay must be less than 2ms and the obtained results comparatively have very less processing time hence preventing the use of echo cancellers.

Table: 8.1 Comparative results obtained at the source coder

<table>
<thead>
<tr>
<th>I/P SPEECH SAMPLE RATE</th>
<th>RATE OF COMPRESSED O/P SPEECH</th>
<th>PROCESSING TIME REQUIRED</th>
</tr>
</thead>
<tbody>
<tr>
<td>48Kb/sec</td>
<td>12Kb/sec</td>
<td>0.000216sec</td>
</tr>
<tr>
<td>64Kb/sec</td>
<td>16Kb/sec</td>
<td>0.000242sec</td>
</tr>
<tr>
<td>96Kb/sec</td>
<td>24Kb/sec</td>
<td>0.000248sec</td>
</tr>
<tr>
<td>128Kb/sec</td>
<td>32Kb/sec</td>
<td>0.0003sec</td>
</tr>
</tbody>
</table>

8.2 THE RESULTS OBTAINED BY THE COMBINED SOURCE AND CHANNEL CODING TECHNIQUE:

By combining the source coding technique and the adaptive channel coding technique we were able to get high quality speech over a wide range of channel
conditions. The joint source and channel coders designed are robust for wide range of noise. This proposed algorithm supports the transmission of signal over the Gaussian channel and also over the Rayleigh fading channel maintaining the bit error rate. This algorithm keeps the continuous track of the channel condition and adaptively selects the channel coding technique for the same. This also supports the full-rate data transmission or half-rate data transmission depending on the type of coding technique used. The overall performance of the channel coded signal when passed through the Gaussian channel and also through the Rayleigh fading channel is as shown in the photographs below. In the photographs for BER vs. $E_b/N_0$ shown below the red plot is drawn for the Gaussian channel and the green plot for the Rayleigh channel. Hence by using the selected channel coding techniques we were able get the desired O/P signal to noise ratio irrespective the channel quality. The first plot is for the animal’s voice, second plot for the child voice and the third and fourth plots are for the female and male voice.

Plate: 8.4 Source coder plot and its BER plot in Gaussian channel and Rayleigh channel for voiced and unvoiced signal
The processing time required by the source encoder and the source decoder using LD-CELP algorithm is 0.8msec and the total processing time required by the source coder, channel coder and the channel to process the voiced and unvoiced signal was 3.6sec. The channel encoder selected by the adaptive channel coding algorithm is convolutional coder. All the graphs are drawn by considering $E_b/N_0$ in db as ‘x’-axis and BER value in ‘y’-axis. From the above graph we can see the BER reduces as the $E_b/N_0$ increases irrespective of the channel. If the channel is noisy the designed algorithm automatically shifts to full rate but maintains the BER using the convolutional coder.

Plate: 8.5 Source coder plot and its BER plot for Gaussian channel and Rayleigh channel for child’s voice

The processing time required by the source encoder and the source decoder using LD-CELP algorithm is 0.8msec and the total processing time required by the source coder, channel coder and the channel to process the speech was 3.08sec. The channel coder selected by the adaptive channel coding algorithm is convolutional coder as the S/N ratio is less than the threshold. The results were published in the international journal 2.
The processing time required by the source encoder and the source decoder using LD-CELP algorithm is 0.2msec and the total processing time required to process the data through the source coder, channel coder and the channel was 3.07sec. The channel coder selected by the adaptive channel coding algorithm is cyclic coder with sampling frequency of 8 kHz.

Plate: 8.7 Source coder plot and its BER plot for Gaussian channel and Rayleigh channel for male voice
The processing time required by the source encoder and the source decoder using LD-CELP algorithm is 0.22msec and the total processing time required to process the data through the source coder, channel coder and the channel was 3.17sec. The channel coder used is the cyclic coder with sampling frequency of 6 kHz. If the channel is less erroneous then adaptive coder chooses the cyclic coder and uses half-rate by accommodating two users within the packet. This increases the traffic rate of the channel without losing the quality of the signal.

The processing time of the channel coder mainly varies with the noise level in the channel. The obtained results due to joint source and channel coder were published in the international journal 3 and 4.