Aim: To implement GSM data interleaving

Software: Octave

Theory

During a call speech signal has to be digitized, coded, formatted , modulated and then transmitted on wireless media. The entire signal processing flow is discussed as follows:

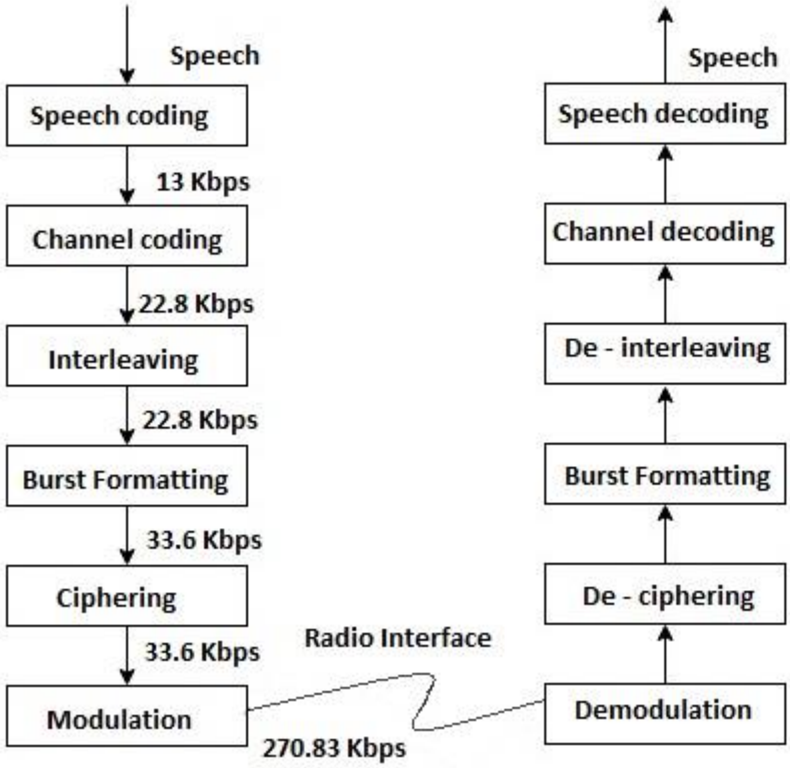

Figure : GSM speech processing

i) **Speech Coding:** The users voice signal is fed to a speech coder.The GSM speech coder is based on the Residually Excited Linear Predictive Coder (RELP), which is enhanced by including a Long-Term Predictor (LTP). The coder provides 260 bits for each 20 ms blocks of speech, which yields a bit rate of 13Kbps. Scope for using half-rate coders are included in the specifications. The GSM speech coder takes advantage of the fact that in a normal conversation,each person speaks on average for less than 40%% of the time, By incorporating a voice activity detector (VAD) in the speech coder, GSM systems operate in a Discontinuous Transmission mode (DTX) which provides a longer subscriber battery life and reduces instantaneous radio interference since the GSM transmitter is not active during silent periods. A comfort noise subsystem (CNS) at the receiving end introduces a background acoustic noise to compensate for the annoying switched muting which occurs due to DTX.

ii) **Interleaving:** Interleaving is done to ensure that successive and continuous data is not sent on consecutive time slots, in order to minimize the effect of sudden fades on the received data. Diagonal interleaving is used to send GSM speech signals. In this process, the total of 456 encoded bits within each 20 ms speech frame or are broken into eight 57 bit sub-blocks. These eight sub-blocks which make up a single speech frame are spread over eight consecutive eight consecutive frames for a specific TS).

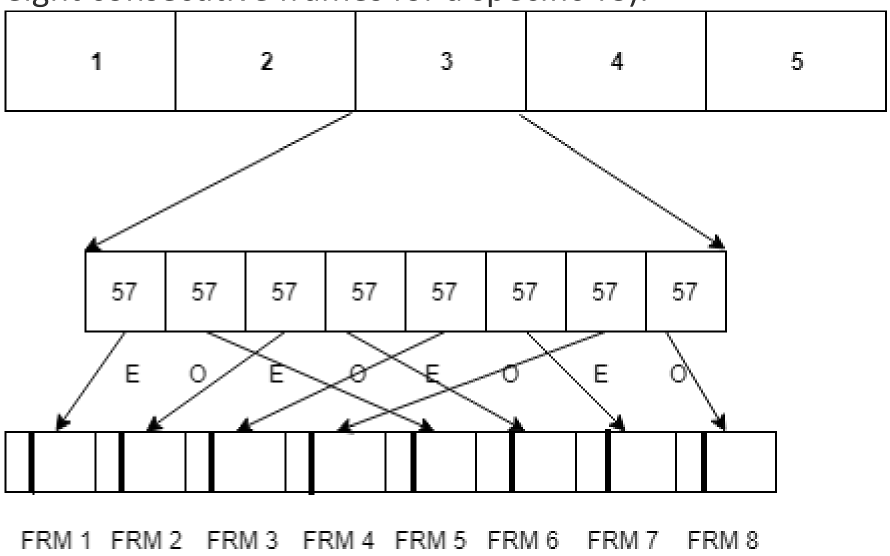

Figure : Diagonal interleaving for GSM speech signal

The 57 bits sub-blocks made out of 1 speech frame are termed as ‘even’ and ‘odd’ parts. Suppose TS 5 has been allotted to the user, then the even parts of the sub-blocks are mapped to the TS5 of first 4 frames.Thus the even parts of the same speech frame are transmitted first. The odd parts are then mapped to TS5 of the next 4 frames and transmitted. In a TS structure, there are two fields for payload. The first field contains even part of current speech frame (57 bits) and second field contains odd part of previous speech frame Thus, if a burst is lost due to interference or fading, channel coding ensures that enough bits will still be received correctly to allow the error correction to work.

Conclusion: