Design and Analysis of security system in GSM Trans-receiver

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Abstract—*A global system for mobile communication is digital cellular communication system called as GSM. In wireless communication technology, quality of voice output at destination depends on the channel condition. Bad channel condition will produce distortion in the voice output and hence degrades the voice quality. Low bit rate modes is used in a bad channel condition to allow more bits for channel coding, while high bit rate modes on the contrary. Various speech coding techniques, such as a LPC (linear predicting coding) and RPE-LTP (regular pulse excitation) are used in different application. The report work addresses the challenges and opportunities starting from the basic issues. In speech coding techniques and standards, discussing current and future application outlining techniques for evaluating speech coder performance, and identify research direction. The most prominent speech coding standards are presented and their properties, such as performance, complexity, and coding delay, analyzed.*

Keywords—MATLAB & Simulink.

I. INTRODUCTION

The speech coding is a procedure to represent a digitized speech signal using as few bits as possible, maintaining at the same time a reasonable level of speech quality. Speech coding has matured to the point where it now constitutes an important application area of signal processing.

Figure 1.1 shows the block diagram of a speech coding system. The continuous time analog speech signal from a given source is digitized by a standard connection of filter using for eliminates aliasing, sampler which is use for discrete-time conversion, and analog-to digital converter using for uniform quantization is assumed. The output is a discrete time speech signal whose sample values are also discretized. This signal is referred to as the digital speech.

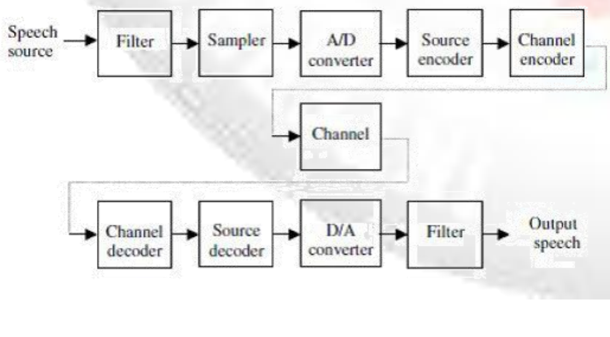


Figure 1.1: Block diagram of a speech coding system.

Traditionally, most speech coding systems were designed to support telecommunicationapplications, with the frequency contents limited between 300 and 3400 Hz. According to theNyquist theorem [10], the sampling frequency must be at least twice the bandwidth of thecontinuous-time signal in order to avoid aliasing. A value of 8kHz is commonly selected as thestandard sampling frequency for speech signals. To convert the analog samples to a digitalformat using uniform quantization, the digital speech will be roughly indistinguishable from theband-limited input more than 8 bits/sample is necessary. The use of 13 bits/sample provides aquality that is considered high. The following parameters are assumed for the digital speech signal:

- Sampling frequency = 8 kHz
- Number of bits per sample = 13
- This gives rise to
- Bit-rate = 8 kHz * 13 bits = 104 kbps

The above bit-rate, also known as input bit-rate, is what the source encoder attempts toreduce Figure 1.1.The output of the source encoder represents the encoded digital speech and ingeneral has substantially lower bit-rate than the input.The linear prediction coding algorithm[17], for instance, has an output rate of 2.4 kbps, a reduction of more than 53 times with respectto the input.The encoded digital speech data is further processed by the channel encoder, providingerror protection to the bit-stream before transmission to the communication channel, wherevarious noise and interference can sabotage the reliability of the transmitted data. Even though inFigure 1.1 the source encoder and channel encoder are separated, it is also possible to jointlyimplement them so that source and channel encoding are done in a single step.

The channel decoder processes the error-protected data to recover the encoded data,which is then passed to the source decoder to generate the output digital speech signal,having the original rate. This output digital speech signal is converted to continuoustimeanalog form through standard procedures:digital-to-analog conversion followed by antialiasingfiltering [10].

II. PRINCIPLE DESIGN

- *Traffic Channel for GSM System*

The sequence of operation of a GSM sender and receiver is depicted in Figure 2.1

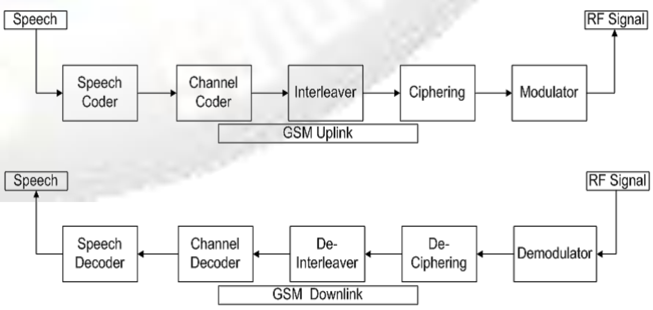


Figure 2.1: Block diagram of a GSM sender and receiver BTS

The signal created by a microphone is an analog signal. Since GSM is an all digitalsystem, this analog signal is not suitable for use on a GSM network.The analog signal must be converted into digital form. This is done by using an

Analog toDigital Converter (ADC).In order to reduce the amount of data needed to represent the sound wave, the analogsignal is rst imputed into a band pass filter.

A. Speech Coding and Decoding:

1) Analog to Digital Conversion:

In order to reduce theamount of data needed to represent the sound wave, the analog signal is burst imputed into aband pass filter.The Band Pass_lter only allows frequencies between 300Hz and 3.4 kHz to passthrough it as shown in Figure This limits the amount of data that the Analog Digital Converter isrequired to process.The filtered signal is imputed into the analog to digital converter. The analog to digital converterperforms two tasks. It converts an analog signal into a digital signal and it does the opposite,converts a digital signal into an analog signal.

The A/D converter measures the analog signal, or samples it 8000 times per second. This meansthat the ADC takes a sample of the analog signal every .125 sec. Each sample is quantified witha 13-bit data block. If we calculate 13 bits per sample at 8000 ,we determine a data rate of104,000 bits per second, or 104 Kb/s as shown in samples per-second ,we determine a data rateof 104,000 bits per second, or 104 Kb/s as shown in figure 2.2

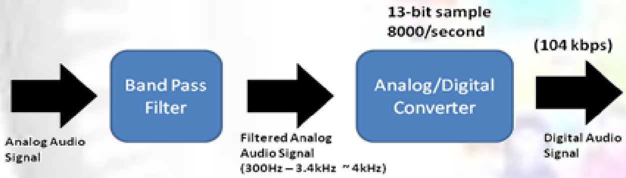


Figure. 2. 2 :ADC: Quantization Process

- *Speech Encoder :*

The speech encoder used in GSM is called Linear Predictive Coding (LPC) and Regular PulseExcitation (RPE).Speech coder maps speech into digital blocks. Coder used in GSM phase 1compresses the speech signal to 13 kbps rate using the Rectangular Pulse Excited LinearPredictable coding with Long Term Prediction (RPE-LTP) technique as per GSM 06-10specification. And in the GSM phase 2 half rate (HR) Enhance Full Rate (EFR), scheme achievesa rate of 5.6 kbps. EFR provides same quality and better performance than RPE-LTP . Thesealgorithms produce a speech block of 260 bits every 20 ms as shown in Figure 2.3

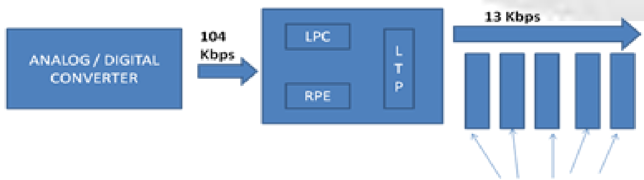


Figure 2.3: RPE – LTP speech coder

Decription	Formula	Result
Convert ms to sec	20ms=1000	0.02 sec
Calculate bits per sec	260bits=0.02 sec	13 kbps

B. Channel Coding and Decoding

Channel Coding is itself consist of two phase they are as follows.

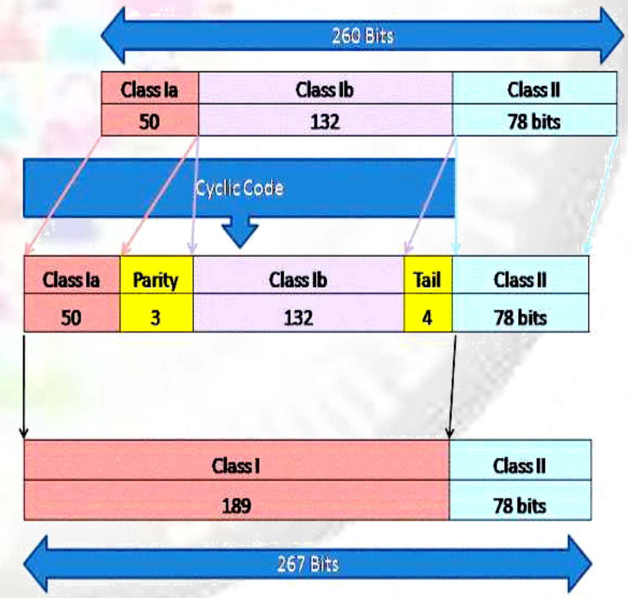
1) Block Coding

A single 260-bit (20ms) audio block is delivered to the block-coder. The 260 bits are divided upinto classes according to their importance in reconstructing the audio as shown in Figure 2.4.

Class Ia - 50 bits (most sensitive to bit errors)
Class Ib - 132 bits (moderately sensitive to bit errors)
Class II - 78 bits (least sensitive to error)

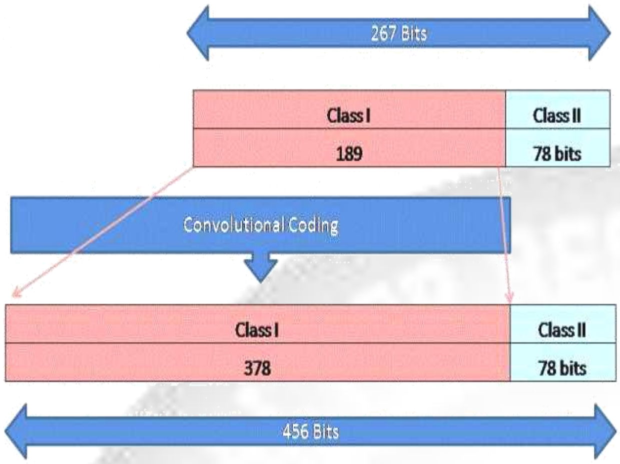
Figure. 2.4: classes of bits

The Class Ia bits are protected by a Cyclic Code(CRCCoding). The cyclic code is run on the 50Ia bits and calculates 3 parity bits which are then appended to the end of the Ia bits as shown inFigure . Only the class Ia bits are protected by this cyclic code. **The Class Ia and Ib**bits are thencombined and anadditional 4 bits are added to the tail of the class I bits (Ia and Ib together). All four bits are zeros (0000) and are needed for the next step which is "Convolutional Coding".There is no protection for **Class II** bits. As you can see, block codin adds seven bits to the audio block, 3 parity bits and 4 tail bits, therefore, a 260-bit block becomes a 267-bit block.



2) Convolution Coding :

This 189-bit block is then feeded into a convolutional coder.Convolutional coding allows errorsto be detected and to be corrected to a limited degree. The Convolution encoder is used forinternal error correction.For every k input bits the Convolution encoder produces n outputbits.The output bits depends not only on the current input bit but also on previous input bits. Thenumber of previous input bits, which govern the output, is termed the Constraint length. Thiscoding uses 5 consecutive bits to calculate the redundancy bit, this is why there are 4 bits addedto the Class I bitswhen the cyclic code was calculated.The number of class I bits is doubled from189 to 378 bits as shown in Figure.



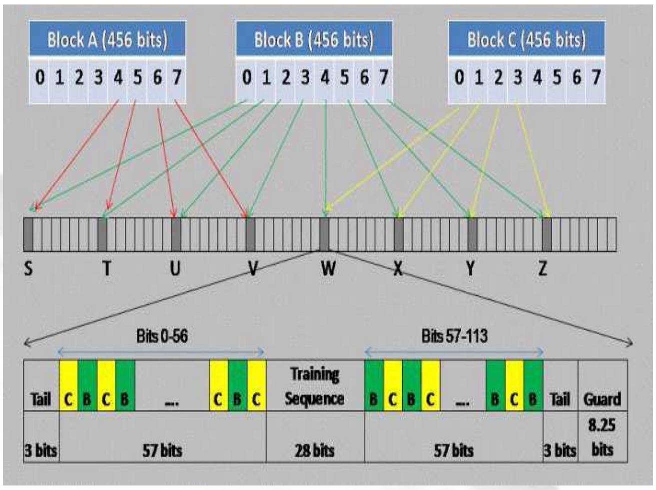
3) Reordering, Partitioning, and Interleaving

Now, one problem remains. All of this error detection and error correction coding will not do anygood if the entire 456-bit block is lost or garbled. In order to alleviate this, the bits are reorderedand partioned onto eight separate sub-blocks. If one sub-block is lost then only one-eighth of thedata for each audio block is lost and those bits can be recovered using the convolutional code onthe receiving end. This is known as *interleaving*. Each 456-bit block is reordered and partitionedinto 8 sub-blocks of 57 bits each. to see the ordering sequence.

These eight 57-bit sub-blocks are then interleaved onto 8 separate bursts. As you remember from the TDMA Tutorial,each burst is composed of two 57-bit data blocks, for a total data payload of114 bits.

The first four sub-blocks (0 through 3) are mapped onto the even bits of four consecutive bursts.The last four sub-blocks (4 through 7) are mapped onto the odd bits of the next 4 consecutivebursts. So, the entire block is spread out across 8 separate bursts.Taking a look at the diagram below we see three 456-bit blocks, labeled A, B, and C.Each block is subdivided into eight sub-blocks numbered 0-7. Let's take a look at Block B. Wecan see that each sub-block is mapped to a burst on a single time-slot. Block B is mapped onto 8separate bursts or time-slots. For illustrative purposes, the time-slots are labeled S through Z.Let's expand time-slot V for a close-up view. We can see how the bits are mapped onto aburst. The bits from Block B, sub-block 3 (B3) are mapped onto the even numbered bits of theburst (bits 0,2,4....108,110,112). You will also notice that the odd bits are being mapped fromdata from block A, sub-block 7 (bits 1,3,5....109,111,113). Each burst contains 57 bits of datafrom two separate 456-bit blocks. This process is known as *interleaving*.

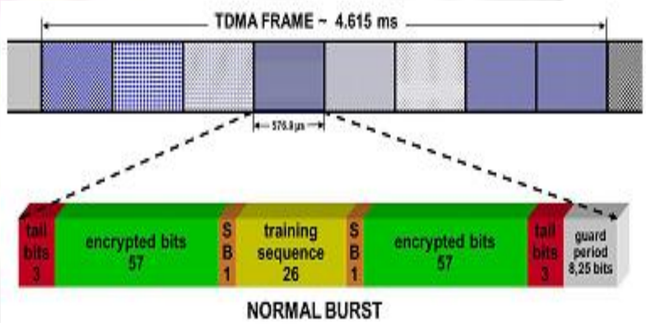
In the following diagram, we examine time-slot W. We see that bits from B4 are mappedonto the odd-number bits (bits 1,3,5....109,111,113) and we would see bits from C1 mapped ontothe even number bits (bits 0,2,4....108,110,112). This process continues indefinitely as data istransmitted. Time-slots W, X, Y, and Z would all be mapped identically. The next time-slotwould have data from Block C and Block D mapped onto it. This process continues for as longas there is data being generated.



4) Interleaving :

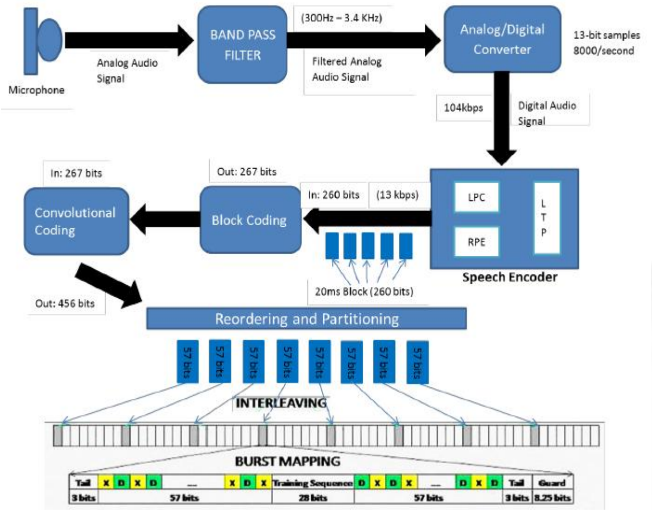
The process of interleaving effectively distributes a single 456 bit audio block over 8 separatebursts. If one burst is lost, only 1/8 of the data is lost, and the missing bits can be recovered usingthe convolutional code. Now, you might notice that the data it takes to represent a 20ms (456-bits) audio block is spread out across 8 timeslots. If you remember that each TDMA frame isapproximately 4.615ms, we can determine that it takes about 37ms to transmit one single 456-bitblock. It seems like transmitting 20ms worth of audio over a period of 37ms would not work.However, this is not what is truly happening. If you look at a series of blocks as they are mappedonto time-slots you will notice that one sub-block ends every four time-slots, which isapproximately 18ms. The only effect this has is that the audio stream is effectively delayed by 20ms, which is truly negligible.We also do for block c and block The following diagram illustrates the entire process, fromaudio sampling to partitioning and interleaving.

III. Brust Mapping



Tail Bits- All-zero bits to indicate the start and the end of the burst.**Data Bits** :- Speech data is to carry after interleaving and Burst mapping.**Training Bits** :- For channel adaptive equalization(is an equalizer thatautomatically adapts to time-varying properties of the communication channel.)**Guard Bits**-ramping time for transmitter ON/OFF, to avoid overlappingbetween adjacent time slots. Necessarily much longer for Access Burst.

– *Summary:*



IV. CIPHERING & DE-CIPHERING :

For security in cellular telecommunications systems are to secure conversation we use these terms "Ciphering & Deciphering". Ciphering is used to change the data patterns. It does not depend on the type of data to be transmitted, but is only applied to normal bursts.This is achieved by performing an XOR operation between a pseudorandom bit sequences and the 114 bits of each bursts Ciphering used at transmitter side and de-ciphering is used at receiver side.For Ciphering method I have pass my bit stream from X-OR gate

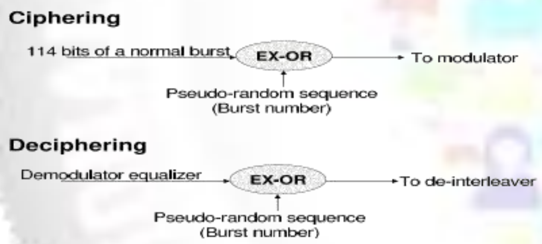


Figure GSM Ciphering / Deciphering

V. CONCLUSIONS

This a ongoing project ,In this paper comparative study of GSM transreceiver. In future work we are going to implementUplink modules basically contain the GSM speech coder, channel encoder, interleaver, cipher,and modulator; similarly downlink modules contains demodulator or equalizer, decipher, deinterleaver, channel decoder, and speech decoder. This chapter is organized for GSM uplink anddownlink, and their implementation in software using MATLAB Simulink also reduce the BER rate in GSM.

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