GSM
Communication systems project

General rules

Submission deadline:

All MATLAB assignments should be submitted due to 26/12/2016 at 23:59, any mail after this date will be directly excluded without exceptions.

Submission mail:

communicationsystems2016ssp@gmail.com

Notes:

(1) This mail is for submission only and will be checked only when it is announced that there is a submission.

(2) For questions, queries use my conventional mail

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remonadly2704@gmail.com

General rules:

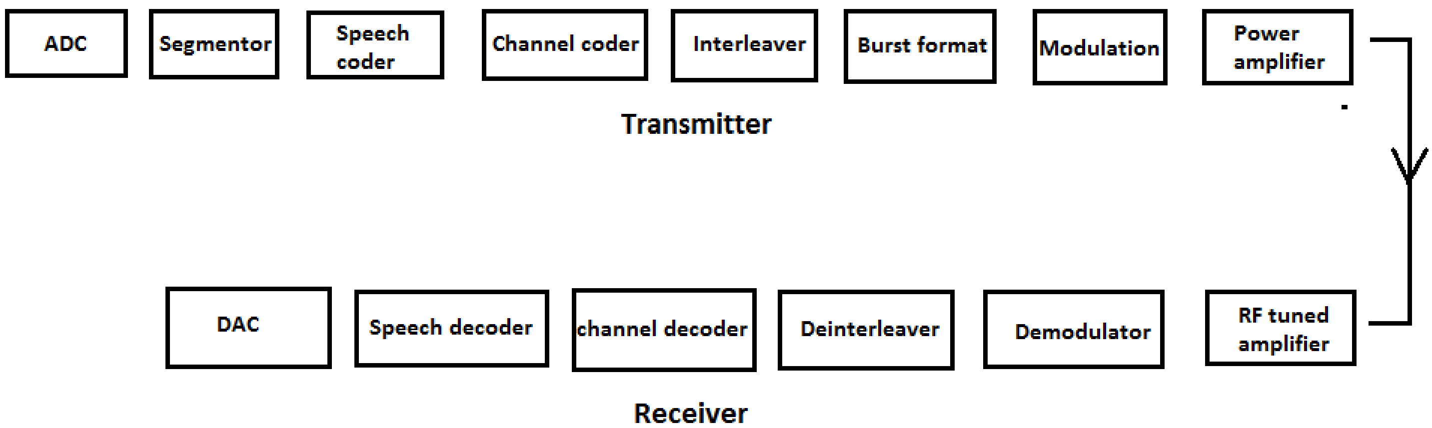
(1) Students are to be divided into groups of 6-7 students each.

(2) Any copied codes will be awarded “zero” without notification, however doing your best and trying to follow the shown procedure will usually awarded above 50% of total mark.

(3) The mail attachment should include:

1) One softcopy report with all the figures, comments and other requirements
2) One m-file for the whole project. The m-file should be well commented, indented, and properly organized. Variable naming should be self-explanatory and clearly indicating which variable/physical quantity it represents in order to ease the tracing of the code. Please DO NOT include names such as: “toto”, “tete”, “brbr”, etc...!

Project description



Simplified GSM physical layer block diagram (Transmitter and Receiver)

Objective:

Understand and implement the simplified GSM block diagram.

Theoretical background:

Analog to Digital converter (ADC):

First of all, Voice is converted into Electrical Analog signal with the help of Microphone. Then Analog Signal Need to be converted into Digital Form by PCM. PCM stands for Pulse Code Modulation it involves Following Steps:-

- Sampling
- Quantization
- Encoding

Sampling: - Sampling Means Converting Analog Signals into Discrete Sample Value. Analog Signal is sampled at Nyquist Rate. "*To reproduce an analog signal without distortion, the signal must be sampled with at least twice the frequency of the highest*
Frequency component in the analog signal". So, Sampling is carried out at 8Khz Because the Human can speak till 3.4Khz.

Quantization :- Quantization Means Dividing Total amplitude Range into Number of level & Rounding of Each sample value to Nearest Level. In PCM we select level as 256 but in GSM we used 8192 levels. Since GSM is wireless so in order to get Speech Quality Equivalent to PCM we need to take 8192 levels. i.e. k=13 bits/sample.

Encoding : - Encoding means assigning Digital Values to Each Quantization Levels. Thus Each Sample Value is assigned with Digital values at the output of Encoding.

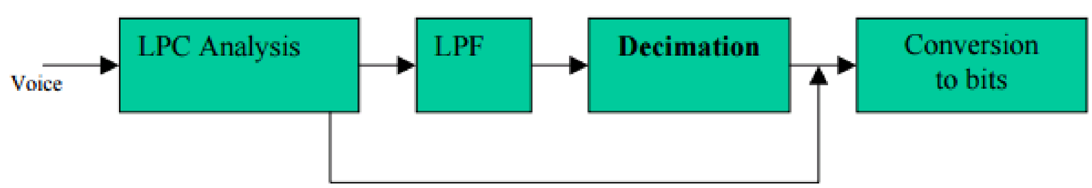
So, The Output of A/D Conversion Includes samples of 13 bits sampled at 8Khz.
Note :- Data rate at the output of A/D converter is 104kbps for one user, Taking data rates of 8 users we get 832kbps.But, Carrier Bandwidth is 200khz. So 832 kbps is not compitable for 200 kHz bandwidth. So we need to somehow reduce Data rate which is done by Segmentation & Speech coding

Segmentation and speech coding:

In this process 20 ms of digital speech signal is block uniformly & coded .This speech signal is coded at rate of 50 times/sec.
ie 1/20ms = 50 times /sec .
The goal of all speech coding systems is to transmit speech with the highest possible quality using the least possible channel bandwidth. The lower the bit rate at which the coder can deliver quality speech, the more speech channels can be implemented within a given bandwidth. Many types of speech coders are available. Some offer better speech quality, at the expense of a higher bit rate (waveform coders). Others use lower bit rates, at the expense of lower speech quality (vocoders).

Instead of using 13 bits per sample as in A/D conversion, GSM speech coding uses 260 bits to encode one segment. This calculates as 260 bits / 20 ms = 13 kb/s. This provides a speech quality which is acceptable for mobile telephony. The most popular vocoding system is Linear Predictive Coding, LPC.

Standard LPC analysis yields the spectral coefficients. The number of spectral coefficients, L, is predetermined. These spectral coefficients are then used to synthesize the original speech. The synthesized speech is then subtracted from the original speech to form a residual signal. The residual signal is then low pass filtered to extract a baseband signal from the residual signal. The baseband residual sequence is then decimated. The decimation factor controls the amount of compression in the encoding process. The baseband residual sequence, the LPC coefficients and the first L original speech values are then coded to bits and transmitted.



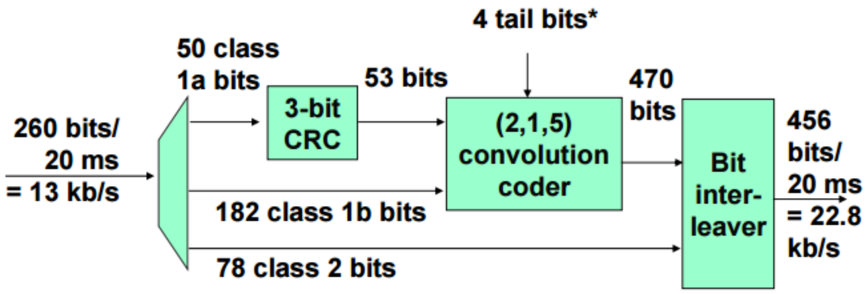
Speech coding

At the receiver, using the spectral coefficients that were transmitted, the L original values and the residual sequence the original speech is reconstructed

Channel coding:

Speech coding does not consider the problems which may be encountered on the radio transmission path. The next stages in the transmission process, channel coding and interleaving, help to overcome these problems. Divides the incoming 260 information bits into three different classes, i.e. class 1a, class 1b, class 2, depending on the importance of the bits. For instance, any transmission errors in the class 1a bits affect the overall speech quality more severely than errors in class 2 bits. Due to this variation in bit importance the different classes of bits are encoded accordingly.

Channel encoder



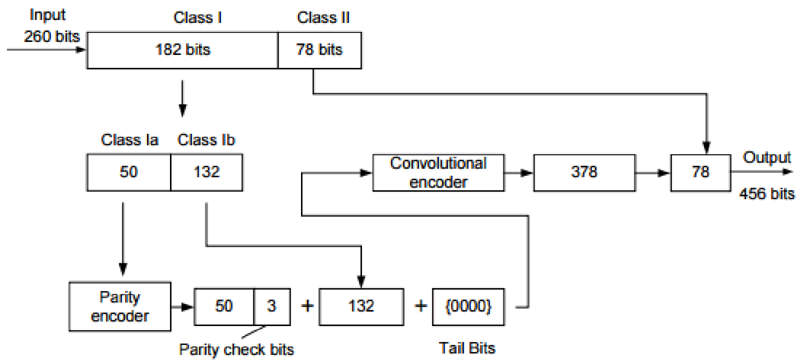
Class 1a: CRC (3-bit error detection) and convolutional coding (error correction)
Class 1b: convolutional coding
Class 2: no error protection
*tail bits to periodically reset convolutional coder

The 50 most significant bits (Class 1a bits) are parity encoded. The parity encoder used in GSM is a systematic cyclic encoder based on three check bits. Systematic means that the parity bits are added to the original class 1a bit sequence. This way the class 1a bits are left unchanged 3 extra parity check bits are added to the end of the Class 1a bits. The generator polynomial used in the parity encoder has a length of 4 bits and is given as $G(x) = x^3 + x + 1$. After parity encoding of Class 1a bits, the resulting 53 bits are combined with Class 1b bits and a tail sequence of four zeros are added to the block. The total number of bits is 189, and this block is sent to convolutional encoder. The convolution encoder takes a block of k bits as input and returns a block of n bits as output. The rate of the encoder, defined as the ratio k/n=n, is in the GSM system specified to be 1/2. In the convolutional encoding scheme each output bit, c_n is depending not only on the input bit presently being encoded, b_k, but also on some of the previous input bits. The number of input bits required in the processing of encoded output bit is called the constraint length of the encoder. GSM specifies a constraint length of 5 in its encoding scheme defined as:

$$C_{2k} = b_k \oplus b_{k-3} \oplus b_{k-4}$$
$$C_{2k+1} = b_k \oplus b_{k-1} \oplus b_{k-3} \oplus b_{k-4}$$
$$\oplus \quad = \text{Mod 2 addition}$$
$$k \in \{0,1,2,3,\ldots..189\} \quad \text{and } b_k = 0 \text{ for } -\infty < k < 0$$

As the convolution encoder is defined as a rate ½ encoder two output bits are generated for every input bit, hence the two expressions. After convolutional encoding, 378 bits are generated out of 189 input bits, and they are combined with Class2 bits to produce the output of the channel encoding. The input to the channel encoder has a block length of 260 bits, and the output of the encoder is 456 bits long

Convolutional decoding can be performed using a Viterbi algorithm. A Viterbi decoder logically explores in parallel every possible user data in sequence. It encodes and compares each one against the received sequence and picks up the closest match



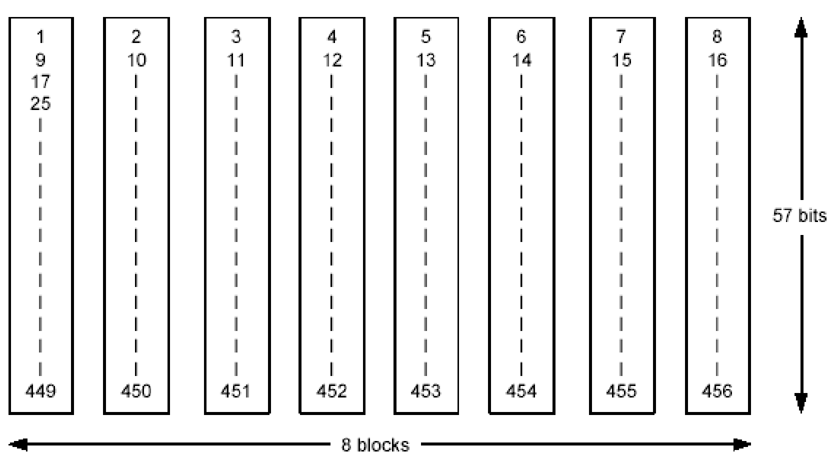
Channel Encoding in GSM

Interleaving

Interleaving is a process of dispersing the bits of a data burst over multiple bursts in a systematic way. Benefit of this technique: when a data-burst is lost (due to burst error in the radio interface) it does not mean a 100% loss of a single burst rather a partial loss of many bursts

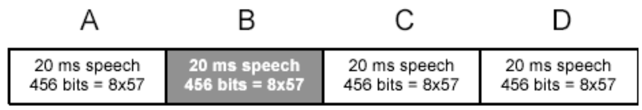
First level of interleaving

The channel coder provides 456 bits for every 20 ms of speech which are interleaved in eight blocks of 57 bits shown below.
The first block contains the 1^{st} , 9^{th} ,17^{th},25^{th},…………… & 449^{th} bit
This is called block interleaving

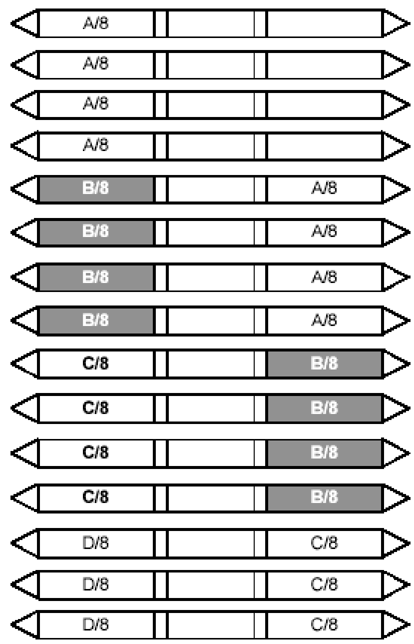


Second level of interleaving

If only one level of interleaving is used, a loss of this burst results in a total loss of 25%. This is too much for the channel decoder to correct. A second level of interleaving can be introduced to further reduce the possible BER to 12.5%.
Instead of sending two blocks of 57 bits from the same 20 ms of speech within one burst, a block from one 20 ms and a block from next sample of 20 ms are sent together. A delay is introduced in the system when the MS must wait for the next 20 ms of speech. However, the system can now afford to lose a whole burst, out of eight, as the loss is only 12.5% of the total bits from each 20ms speech frame. 12.5% is the maximum loss level that channel decoder can correct.



Speech Frame

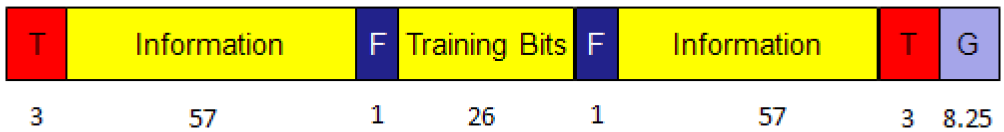


Burst format

This GSM burst is used for the standard communications between the basestation and the mobile, and typically transfers the digitized voice data.

The structure of the normal GSM burst is exactly defined and follows a common format. It contains data that provides a number of different functions:

1. **3 tail bits:**These tail bits at the start of the GSM burst give time for the transmitter to ramp up its power
2. **57 data bits:**This block of data is used to carry information, and most often contains the digitized voice data
3. **1 bit flag:** This bit within the GSM burst indicates the type of data in the previous field.
4. **26 bits training sequence:**This training sequence is used as a timing reference and for equalization.
5. **1 bit flag:**again this flag indicates the type of data in the data field.
6. **57 data bits:**again, this block of data within the GSM burst is used for carrying data.
7. **3 tail bits:** these final bits within the GSM burst are used to enable the transmitter power to ramp down. They are often called final tail bits, or just tail bits.
8. **8.25 bits guard time**: At the end of the GSM burst there is a guard period. This is introduced to prevent transmitted bursts from different mobiles overlapping. As a result of their differing distances from the base station.



Modulation

The bits must then be sent over the air using a carrier frequency. GSM uses the GMSK modulation technique with BT=0.3 the bits are modulated onto a carrier frequency and transmitted.

Here is a useful website that might help you:

http://rahulundegaonkar.blogspot.com.eg/2011/12/gsm-transmission-process.html.

Procedure and simulation parameters.

a. Read the attached audio file. You may use the matlab function (audioread).

b. Samling frequency: Fs=8000 sample/sec, you may use resample function.

c. Numbers of bits per sample k=13 bits/sample

d. Encode each segment to generate 260 bit using LPC vocoding (or **any** other speech coder of your choice) such that speech coder bit rate is 260 bit / 20 ms = 13 kb/s.

e. Apply the GSM channel encoding for the 260 bits long from the speech coder to produce 456 bits long using (3 bit CRC and convolutional code with rate ½)

f. Use viterbi decoder algorithm for decoding the 456 bits long from channel coding. (useful MATLAB functions: poly2trellis- convenc- vitdec)

g. Use the following values for each bit in the burst format

Tail bits	000
Flag bit	0
Training bits	11111001101010000011001010

h. carrier frequency = 890.1 MHz

2. Requirements:

a. Implement all mentioned blocks.
b. Plot the original speech (before speech coding) and after reconstruction (after speech decoding)
c. Repeat the plotting in (a) for the channel encoding and decoding