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STUDY OF DIFFERENT TYPES CODERS FOR GSM

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Abstract— **Speech coders are very important devices in mobile communication. They determine the recovered speech and the capacity of the system. The original coder are used in the European digital cellular standard GSM rather by a grand-noise name of regular pulse excited long-term predication (RPE-LTP) codec. This codec has a net bit rate of 13 Kbps and was chosen after conducting exhausting subjective tests on various competing codec. The GSM Codec is relatively complex and power hungry. For that reason the Adaptive Multirate (AMR) codec is usually used in the GSM system. It is a more comprised natural codecs which produce lower bit rates and toll-quality speeches compared to other coder. These coders are multi-rate ACELP coders with 8 modes, operating at bit rates from 12.2 Kbps down to 4.75 Kbps. So in this article, we discuss about different types of parameters of the AMR codec, which make the GSM system more efficient.**

Index Terms— *LPC, AMR codec*

I. Introduction

Basically one can differ between the classification of lossless coding methods and lossy coding methods. In lossless a reconstruction of the speech signal is possible by regulating the decoder and gaining the same shape as the input speech signal. In the lossy coding the reconstructed speech signal is differs from the original speech signal waveform [18] [12]. Most of the speech coding techniques are based on the lossy coding techniques, in which irrelevant information is removed. In mobile communication systems, the design and subjective test of speech has been extremely difficult. The goal of all speech coding systems is to transmit speech with the highest possible equality using the least possible channel capacity. The hierarchy of speech coders is shown in Figure 1.1.

- **Attributes of speech coders**

Speech coding either enhances the quality of a speech signal at a particular bit-rate or minimizes the bit-rate at a given quality.

There are the following different properties for speech coders:

- Low bit-rate
- High speech quality
- Robustness to different speakers/languages
- Channel errors
- Low memory requirements
- Less computational complexity
- Low coding delay

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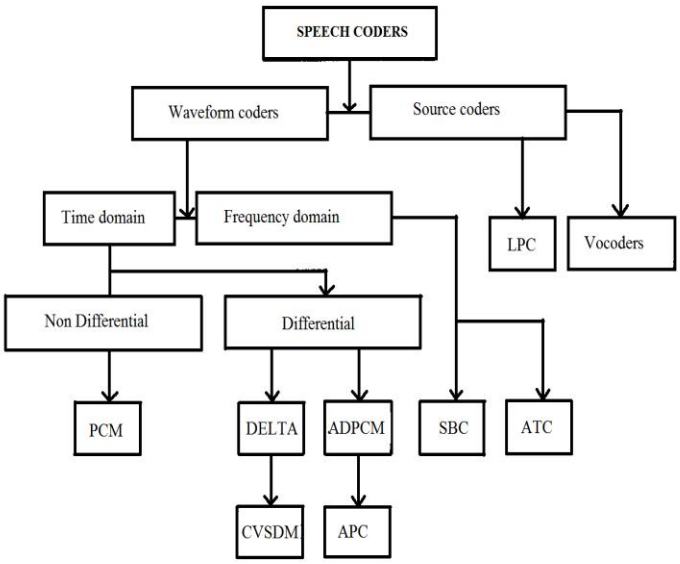


Figure 1.1 Hierarchy of speech coders [18]

II. Classification by Coding Technique

Speech coders differ widely in their approaches to achieving signal compression. Based on the means by which they achieve compression, speech coders are broadly classified into four categories [10] [12].

- Waveform coders
- Vocoders
- Parametric coders
- Hybrid coders

Waveform coders essentially try to reproduce the time waveform of the speech signal are closely as possible. They are, in principle, designed to be source independent and can hence code equally well a variety of signals [18] [4]. They have the advantage of being robust for a wide range of speech characteristics and for noisy environments.

Vocoder is an analysis/synthesis system, used to reproduce human speech. The vocoder was originally developed as a speech coder for telecommunications applications in the 1930s, the idea being to code speech for transmission. Transmitting the parameters of a speech model instead of a digitized representation of the speech waveform saves bandwidth in the communication channel; the parameters of the model change relatively slowly, compared to the changes in the speech waveform that they describe. Its primary use in this fashion is for secure radio communication, where voice has to be encrypted and then transmitted.

A variety of different forms of audio codec or vocoder are available for general use, and the GSM system supports a number of specific audio codecs. These include the RPE-LPC, half rate, and AMR codecs. The performance of each voice codec is different and they may be used under different conditions, although the AMR codec is now the most widely used [5, 6]. Also the newer AMR wideband (AMR-WB) codec is being introduced into many areas, including GSM.

In parametric coders the speech signal is assumed to be generated from a model controlled by some speech parameters. In these coders the speech signal is modeled using a limited number of parameters corresponding to the speech production mechanism. These parameters are obtained by analyzing the speech signal before transmission [2, 19].

Hybrid coders try to fill the gap between waveform coders and parametric coders. Hybrid coders operate at medium bit-rates between those of waveform coders and parametric coders and produce high quality speech than parametric coders. An example of hybrid coder is the Code Excited Linear Predictive (CELP) coders.

III. Frequency Domain Coding of Speech

Frequency domain coders are a class of speech coders which take advantage of speech perception and generation models without making the algorithm totally dependent on the models used.

The most common types of frequency domain coding include (i) Sub-band coding (SBC) (ii) Block transfer coding (BTC).

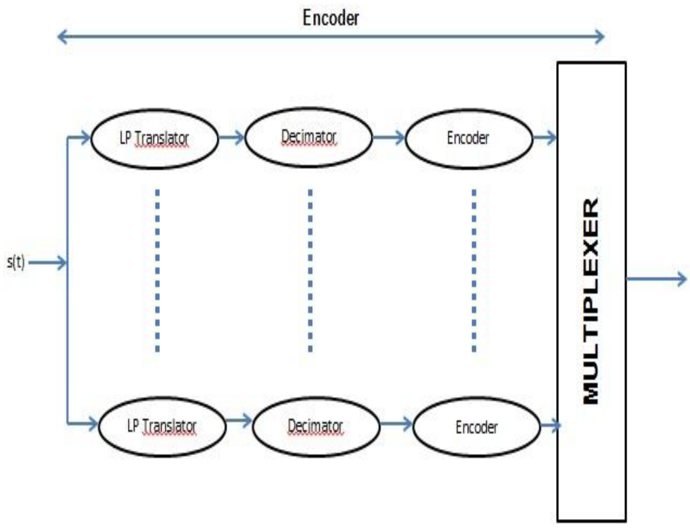
Sub-band coding: - It can be explain the method of controlling and distributing quantization noise across the signal spectrum. Quantization is a non linear operation which produces noise products that are typically broad in spectrum. The human ear does not detect the quantization distortion products at all frequencies equally well [9] [18] [12].

In sub-band coder, speech is typically divided into four or eight sub bands by a bank of filters, and each sub band is sampled at a band-pass Nyquist rate and encoded with different accuracy in accordance to a perceptual criteria.One partitioning of speech band according to this method as suggested by Crochiere et al [12, 15] is given below

Table 1.1: Speech band Partitioning

Sub-band Number	Frequency range
1	200-700 Hz
2	700-1310 Hz
3	1310-2020 Hz
4	2020-3200 Hz

Sub band coding can be used for coding speech at bit rates in the range of 9.6 Kbps to 32 Kbps. In this range speech quality is roughly equivalent to that of ADPCM at an equivalent bit rate. Its complexity and relative speech quality at low bit rates make it particularly advantageous for coding below about 16 Kbps. The CD-900 cellular telephone system uses sub-band coding for speech compression. The block diagram of sub-band coder and decoder as shown in Figure 1.3 below.



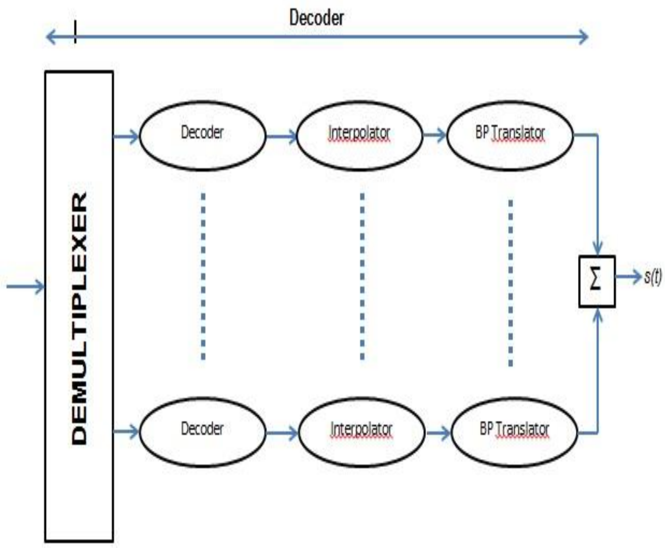


Figure 1.2: Block diagram of sub-band (a) encoder (b) decoder **[18] [19]**

Adaptive Transform Coding (ATC) is another frequency domain technique that has successfully used to encode speech at bit rates in the range 9.6 Kbps to 20 Kbps. This a more complex technique which involves block transformation of windowed input segments of the speech waveform. Each segment is represented by a set of transform coefficients, which are separately quantized and transmitted. At the receiver the quantized coefficient are inverse transformed to produce a replica of the original input segment.

One the most attractive and frequently used transforms for speech coding is the discrete cosine transform (DCT). The DCT of a N-point sequence x(n) is defined as [18, 8]

$$Xc(k) = \sum_{n=0}^{N-1} x(n)h(k)\cos\left[\frac{(2n+1)k\pi}{2N}\right]$$

k= 0, 1, 2…N-1…….. (1.1)

Where $h(0) = 1$ and $g(k) = \sqrt{2}$, k= 1, 2,…., N-1. The inverse DCT is defined as:

$$x(n) = \frac{1}{N}\sum_{k=0}^{N-1} Xc(k)h(k)\cos\left[\frac{(2n+1)k\pi}{2N}\right] \quad n = 0, 1, 2, ..N-1$$

……………………………………………….(1.2)

In practical situations the DCT and IDCT are not evaluate directly using the above equation developed for computing the DCT in a computationally efficient manner are used.

IV. VOCODERS

Vocoders are class of speech coding system that analyze the voice signal at the transmitter, transmit parameters derived from the analysis, then synthesize the voice at the receiver using those parameters. Vocoders or speech codecs are used within many areas of voice communications. Obviously the focus here is on GSM audio codecs or vocoders, but the same principles apply to any form of codec [7] [6].

Audio codecs or vocoders are universally used within the GSM system. They reduce the bit rate of speech that has been converted from its analogue for into a digital format to enable it to be carried within the available bandwidth for the channel. Without the use of a speech codec, the digitized speech would occupy a much wider bandwidth then would be available. Accordingly GSM codecs are a particularly important element in the overall system [14] [17]. Vocoders are, in general, much more complex than the waveform coders and achieve very high economy in transmission bit rate. However they are less robust and their performance tends to be talker dependent. The most popular among the vocoding systems is the linear prediction coder (LPC). Figure 1.4 shows the traditional speech generation model that is the basis of all vocoding system.

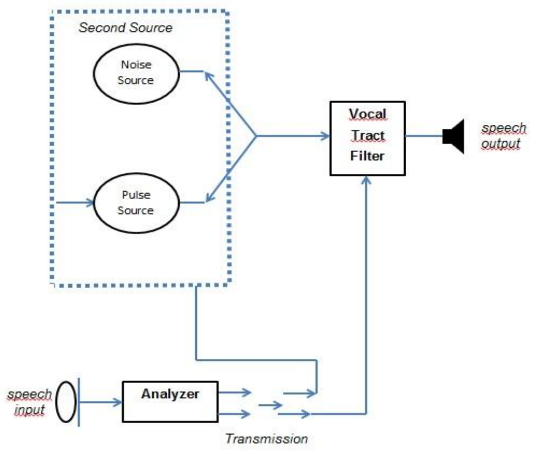


Figure 1.3: Speech Generation Model

V. DIFFERENT TYPES OF CODERS

A. Liner Predictive Coders

LPC Vocoders- LPC belong to the time domain class of vocoders. This class of vocoders attempts to extract the significant features of speech from the time waveform. With LPC, it is possible to transmit good quality voice at 4.8 Kbps and poor quality at even lower rates.

The linear predictive coding system models the vocal tract as an all pole linear filter with a described by [18, 19]:

$$H(z) = \frac{G}{1 + \sum_{k=1}^{M} b_k z^{-k}} \quad \dots\dots\dots\dots\dots\dots\dots\dots (1.3)$$

Where G is a gain of the filter and z^{-1} represents a unit delay operation. The prediction principles used are similar to those in ADPCM coders.

However instead of transmitting quantized values of the error signal representing the difference between the predicted and actual waveform, the LPC system transmits only selected characteristics of the error signal. The parameters include the gain factor, pitch information, and Voice and unvoiced decision information, which allow approximation of the correct error signal. At the receiver, the received information about the error signal is used to determine the appropriate excitation for the synthesis filter.

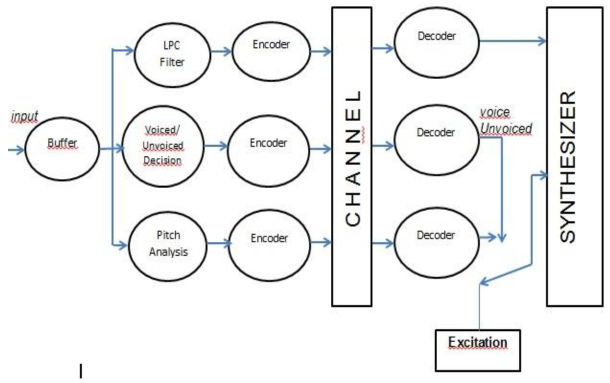


Figure 1.4: Block diagram of LPC [18]

This technique requires that the transmitter extract pitch frequency information which is often very difficult. More ever the phase coherence between the harmonic components of the excitation pulse tends to produce a buzzy beats in the synthesized speech. These problems minimized in other two methods.

- Multipulse Excited LPC
- Code-Excited LPC

In multi-pulse LPC no matter how well the pulse is positioned, excitation by a single pulse per pitch period produces audible distortion. Therefore using more than one pulse typically eight per period, and adjusting the individual pulse positions and amplitude sequentially to minimize a spectrally weighted mean square error. This technique is called the multiple excited LPC (MPE-LPC) and result in better speech quality, not only because the prediction error is better approximated by several pulses per pitch period, but also because the multipulse algorithm does not require pitch detection. The number of pulse can be reduced. A variety of different codec methodologies are used for GSM codecs**.**

Code- Excited LPC, the coder and decoder have predetermined book of stochastic excitation signals. For each speech signal the transmitter searches through its code book of stochastic signal for the one that gives the best perceptual match to the sound when used as an excitation to the LPC filter [17] [18] [19]. The code excited LPC (CELPC) coders are extremely complex and can require more than 500 million multiply and add operations per second. They can provide high quality even when the excitation is coded at only 0.25 bits per sample. These coders can achieve transmission bit rates as low as 4.8 kbps. The main principle behind the CELP codec is that is uses a principle known as "Analysis by Synthesis". In this process, the encoding is performed by perceptually optimizing the decoded signal in a closed loop system. One way in which this could be achieved is to compare a variety of generated bit streams and choose the one that produces the best sounding signal.

VSELP (Vector Sum Excitation Linear Prediction) codec The Vector Sum Excitation Linear Prediction codec one of the major drawbacks of the VSELP codec is its limited ability to code non-speech sounds. This means that it performs poorly in the presence of noise. As a result this voice codec is not now as widely used, other newer speech codecs being preferred and offering far superior performance. Figure 2.3

shows the method for selecting the minimum excitation signal. The procedure is best illustrated through an example.

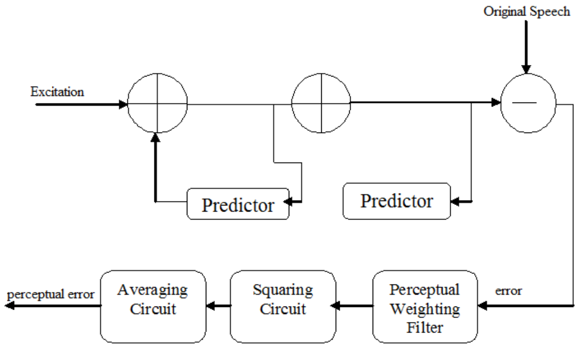


Figure 1.5: Block diagram illustrating the ACCELP Codec [17] [18]

Consider the coding of a short 4 ms block of speech signal. At a sampling frequency of 5 KHz, each block consists of 20 speech samples. A bit rate of ½ bit per sample corresponds to 10 bits per block. Therefore, there are 2^{10} = 1024 possible sequences of length 40 for each block.

Residual (Error) Excited LPC The rationale behind the residual excited LPC (RELP) is related to that of the DPCM technique in waveform coding. In this class of LPC coder, after estimating the model parameters (LP coefficients or related parameters) and excitation parameter (voiced/unvoiced decision, pitch, gain) from a speech frame, the speech is synthesized at the transmitter and subtracted from the original speech signal to from a residual signal. The residual signal is quantized, coded, and transmitted to the receiver along with the LPC model parameters. At the receiver the residual error signal is added to the signal generated using the model parameters to synthesize an approximation of the original speech signal. The quality of the synthesized speech is improved due to the addition of the residual error. Figure 2.4 shows a block diagram of a simple RELP codec.

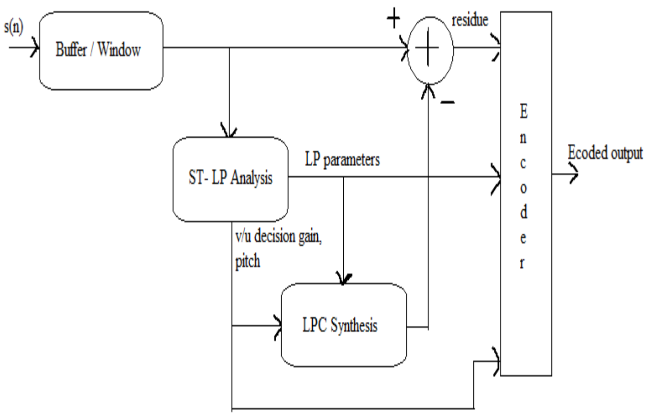


Figure 1.6: Block diagram of RELP encoder [18]

Table 1.2: Speech Coders used in various first and second generation wireless systems [8, 10]

Standard	Service type	Speech Coder Type Used	Bit Rate (Kbps)
GSM	Cellular	RPE-LTP	13
CD-900	Cellular	SBC	16
USDC (IS-54)	Cellular	VSELP	8
IS-95	Cellular	CELP	1.2,2.4, 4.8, 9.6
IS-95 PCS	PCS	CELP	14.4
PDC	Cellular	VSELP	4.5, 6.7, 11.2
CT2	Cordless	ADPCM	32
DECT	Cordless	ADPCM	32
PHS	Cordless	ADPCM	32
DCS-1800	PCS	RPE-LTP	13
PACS	PCS	ADPCM	32

VI. The GSM Codec

The Original speech coder used in the pan-European digital cellular standard GSM goes by rather most popular name of regular pulse excited long term prediction (RPE-LTP) codec. This codec has bit rate of 13 Kbps and was chosen after conducting exhaustive subjective tests on various competing codecs. More recent GSM upgrades have improved upon the original codec specification. The RPE-LTP codec combines the advantages of the earlier French proposed base band RELP codec with those of the multi-pulse excited long term prediction (MPE-LTP) codec proposed by Germany. The most advantage of the base band RELP codec is that it provides good quality speech at low complexity. The speech quality of a RELP codec is however, limited due to tonal noise introduced by the process of high frequency regeneration and by the bit errors introduced during transmission. The MPE-LTP technique produces best speech or excellent speech quality at high complexity and is not much affected by bit errors in the channel. By modifying or enhancing the RELP codec to incorporate certain features of the MPE-LTP codec, the net bit rate was reduced from 1.477 Kbps to 13.0 Kbps without loss of quality. The most important modification was the additional of a long term prediction loop.

The GSM codec is relatively complex and power hungry. The Figure 2.5 shows a block diagram of the speech encoder. The encoder is comprised of four major processing blocks. The speech sequence is first pre-emphasized, ordered in to segment of 20 ms duration and then Hamming- windowed. This is followed by a short tem prediction (STP) filtering analysis where the logarithmic area ration (LARs) of the reflection coefficient $r_n(k)$ (eight in number) are computed. The eight LAR parameters have different dynamic ranges and probability distribution functions, and hence all of them are not encoded with the same number of bits for

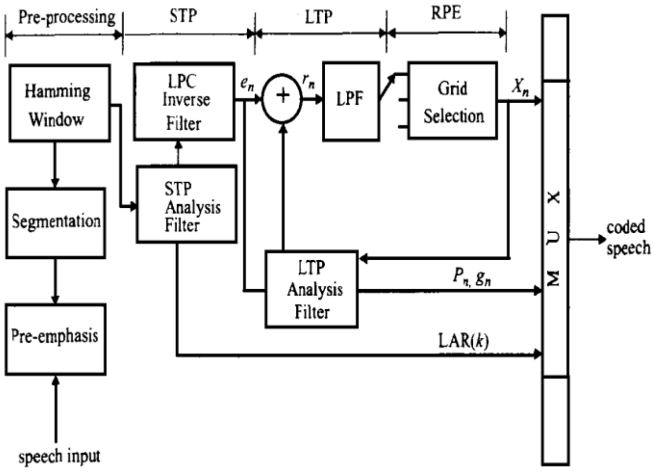


Figure 1.7: Block diagram GSM speech encoder [18]

transmission. The LAR parameters are also decoded by the LPC inverse filter so as to minimize the error e_n.

LTP analysis which involves finding the pitch period p_n and gain factor g_n is then carried out such that LTP residual r_n is minimized. To minimize r_n, pitch extraction is done by the LTP by determining that value of delay, D, which maximizes the cross correlation between the current STP error sample, e_n, and a previous error sample, e_{n-D}. The extracted pitch p_n and gain g_n are transmitted and encoded at a rate of 3.6 Kbps. The LTP residual, r_n, is weighted and decomposed into three candidate excitation sequences. The energies of these sequences are pulse indentified, and the one with the highest energy is selected to represent the LTP residual [12] [18]. The pulse in the excitation sequence are normalized to the highest amplitude, quantized, and transmitted at a rate of 9.6 Kbps.

Figure 2.6 shows a block diagram of the GSM speech decoder. It consists of four blocks which perform operations complementary to those of the encoder. The received excitation parameters are RPE decoded and passed to the LTP synthesis filter which uses the pitch and gain parameter to synthesis the long term signal. Short term synthesis is carried out using the received reflection coefficient to recreate the original speech signal.

Every 260 bits of the coder i.e. 20 ms blocks of speech are ordered depending on their importance, into groups of 50, 132, and 78 bits each. The bits in the first group are very important bits called I_a bits. The next 132 bits are important bits called I_b bits [4] [5] [18] and the last 78 bits are called type II bits. The least type II bits have no error correction or detection.

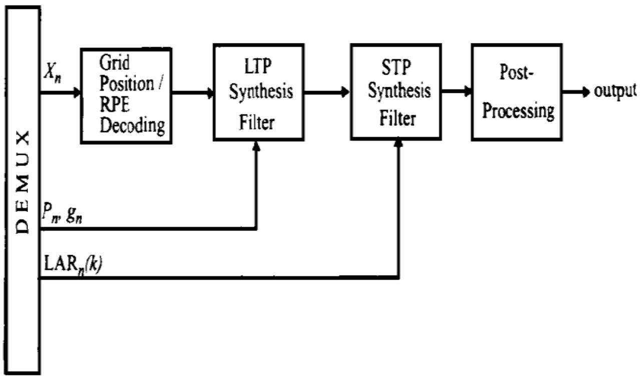


Figure 1.8: Block diagram of GSM speech decoder [18]

VII. GSM AUDIO CODECS / VOCODERS

A variety of GSM audio codecs / vocoders are supported. These have been introduced at different times, and have different levels of performance. Although some of the early audio codecs are not as widely used these days, they are still described here as they form part of the GSM system.

Table 2.2 b: Comparison of different technologies

CODEC NAME	BIT RATE (Kbps)	COMPRESSION TECHNOLOGY
Full rate	13	RTE-LPC
EFR	12.2	ACELP
Half rate	5.6	VSELP
AMR	12.2-4.75	ACELP
AMR-WB	23.85-6.60	ACELP

VIII. GSM AMR CODEC

The AMR, Adaptive Multi-rate codec is now the most widely used GSM codec. The AMR codec was adopted by 3GPP in October 1988 and it is used for both GSM and circuit switched UMTS / WCDMA voice calls.

The AMR codec provides a variety of options for one of eight different bit rates as described in the table below. The bit rates are based on frames that are 20 milliseconds long and contain 160 samples. The AMR codec uses a variety of different techniques to provide the data compression [16, 18]. The ACELP codec is used as the basis of the overall speech codec, but other techniques are used in addition to this. Discontinuous transmission is employed so that when there is no speech activity the transmission is cut. Additionally Voice Activity Detection (VAD) is used to indicate when there is only background noise and no speech. Additionally to provide the feedback for the user that the connection is still present, a Comfort Noise Generator (CNG) is used to provide some background noise, even when no speech data is being transmitted. This is added locally at the receiver.

The use of the AMR codec also requires that optimized link adaptation is used so that the optimum data rate is selected to meet the requirements of the current radio channel conditions including its signal to noise ratio and capacity. This is achieved by reducing the source coding and increasing the channel coding. Although there is a reduction in voice clarity, the network connection is more robust and the link is maintained without dropout. Improvement levels of between 4 and 6 dB may be experienced [11] [14]. However network operators are able to prioritize each station for either quality or capacity. The AMR codec has a total of eight rates: eight are available at full rate (FR), while six are available at half rate (HR). This gives a total of fourteen different modes.

Table 2.3: AMR codec data rates

MODE	BIT RATE (KBPS)	FULL RATE (FR) / HALF RATE (HR)
AMR 12.2	12.2	FR
AMR 10.2	10.2	FR
AMR 7.95	7.95	FR/HR
AMR 7.40	7.40	FR/HR
AMR 6.70	6.70	FR/HR
AMR 5.90	5.90	FR/HR
AMR 5.15	5.15	FR/HR
AMR 4.75	4.75	FR/HR

AMR-WB codec - Adaptive Multi-Rate Wideband, AMR-WB codec, also known under its ITU designation of G.722.2, is based on the earlier popular Adaptive Multi-Rate, AMR codec. AMR-WB also uses an ACELP basis for its operation, but it has been further developed and AMR-WB provides improved speech quality as a result of the wider speech bandwidth that it encodes. AMR-WB has a bandwidth extending from 50 - 7000 Hz which is significantly wider than the 300 - 3400 Hz bandwidths used by standard telephones. However this comes at the cost of additional processing, but with advances in IC technology in recent years, this is perfectly acceptable [9] [11] [14].

The AMR-WB codec contains a number of functional areas: it primarily includes a set of fixed rate speech and channel codec modes. It also includes other codec functions including: a Voice Activity Detector (VAD); Discontinuous Transmission (DTX) functionality for GSM; and Source Controlled Rate (SCR) functionality for UMTS applications [5] [7]. Further functionality includes in-band signaling for codec mode transmission, and link adaptation for control of the mode selection. The AMR-WB codec has a 16 KHz sampling rate and the coding is performed in blocks of 20 ms. there are two frequency bands that are used: 50-6400 Hz and 6400-7000 Hz. These are coded separately to reduce the codec complexity. This split also serves to focus the bit allocation into the subjectively most important frequency range.

The lower frequency band uses an ACELP codec algorithm, although a number of additional features have been included to improve the subjective quality of the audio. Linear prediction analysis is performed once per 20 ms frame. Also, fixed and adaptive excitation codebooks are searched every 5 ms for optimal codec parameter values [1, 3].

IX. AMR BASIC OPERATION

Figure 2.9 shows that there are fixed rate speech and channel codecs. Through a different level of error protection a different distribution of the available gross bit-rate between speech and channel coding is provided by each codec mode. The AMR codec contains a set of fixed rate speech and channel codecs, in-band signaling and link adaptation [16, 19].

Figure 2.9 shows a basic block diagram of the AMR codec in GSM. Each codec mode provides a different level of error protection through a different distribution of the available gross bit-rate between speech and channel coding. The link adaptation process bears responsibility for measuring the channel quality and selecting the optimal speech and channel codecs. In-band signaling transmits the measured channel quality and codec mode information over the air interface. The in-band signaling is transmitted along with the speech data. The Mobile Station (MS) and the Base Transceiver Station (BTS) both perform channel quality estimation for the receive signal path. Based on the channel quality measurements, a Codec Mode Command (over downlink to the MS) or Codec Mode Request (over uplink to network) is sent in-band over the air interface [15] [16].

The AMR codec contains a set of fixed rate speech and channel codecs, in-band signaling and link adaptation. Figure 1.9 shows a basic block diagram of the AMR codec in GSM.

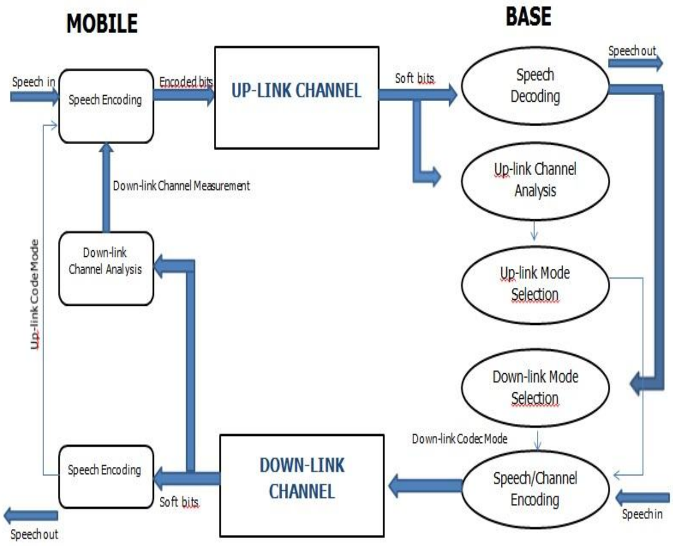


Figure 2.8: Simplified Block diagram of AMR speech coder

X. CONCLUSION

Above include the study on speech coder and vocoders. It also introduces and describes different types of codec technologies that are used in the GSM.

The frequency in the GSM technology is limited. Therefore it is essential to save frequency, power, and to increase the channel capacity. The codecs are an important factor in GSM, when it comes to a voice, which is to be transferred or recorded in a particular format because the voice or speech can also take the much frequency and power also. So above study the different types of coders which are used in GSM in that the AMR codec is very natural codec which having the low voice quality but taken less bandwidth and to improve the voice quality the AMRWB are used and very efficient manner.

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