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Study and Performance of AMR Codecs for GSM

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Abstract: In wireless communication system, limited bandwidth and power is the primary restriction. The existing wireless systems involved in transmission of speech visualized that efficient and effective methods be developed to transmit and receive the same while maintaining quality of speech, especially at the receiving end. Speech coding technique is a material of research for the scientific and academic community since the era of digitization (digital). Amongst all elements of the communication systems (transmitter, channel and receiver), transmission channel is the most critical and plays a key role in the transmission and reception of information. The quality of speech at receiver end decides by channel conditions. Modelling a channel is a multifarious task. A number of techniques are adopted to alleviate the effect of the channel. Adaptive Multi Rate is one of the techniques that neutralize the deleterious effect of the channel on speech. This technique utilizes variable bit rate that dynamically switches to specific modes of operation depending upon the channel conditions. For example, Low bit rate mode of operation is selected in adverse channel conditions, this helps to provide more error protection bits for channel coding and vice versa. Therefore, in this paper, application of Code Excited Linear Prediction (CELP) source codec on speech followed by AMR codec is studied. Further, higher the bit rate used, the better is the quality of speech. In this paper apart from speech codec about AMR is also studied that why the AMR is proposed for the GSM, how the bits rates are reduced in AMR, operation of AMR and other applications of AMR.

Keywords: AMR, LPC, CELP, GSM

I. INTRODUCTION

Speech coding still is a major issue in the area of digital speech processing. Speech coding is transforming the speech signal to a more compacted form, which can then be transmitted with a significantly smaller memory. It is not possible to access infinite bandwidth. Hence, there is a need to code and compact speech signals. Speech compression is necessary in long-distance communication, high-quality speech storage, and message encryption.

For instance, in digital cellular technology many consumers require to share the same frequency bandwidth. Utilizing speech compression makes it potential for more consumers to share the available system. Another example where speech compression is desirable is in digital voice storage. For a set quantity of available memory, compression makes it possible to store longer messages. Speech coding is a lossy type of coding, which means that the output signal does not closely sound like the input. The input and the output signal may perhaps distinguish to be different. Several techniques of speech coding such as Linear predictive Coding (LPC), Waveform Coding and Sub-bands Coding exist. The speech signals that require to be coded are wideband signals with frequencies from 0 to 8 kHz.

Speech coding could be explained as the conversion of an analog speech signal into a digital signal, but such an explanation better suits the usual meaning of an analog to digital (AD) converter. Speech coding means the conversion of a speech signal which has been already digitalized, into another digital signal featuring a lower bit rate than the original signal. it is also referred to as transcoding and also as compression or bit rate decrement.

Time and again, speech coding does not for only mean encoding a speech signal, but somewhat the complete process including decoding. The words speech codec by and large refers in a similar way to the decoder as well. At times, also the word “codec" is used which is a concatenation of the words coder and decoder, and not of the words encoder and decoder, as one would anticipate. A codec encodes a data stream/signal for transmission, storage, or decodes it for playback or editing [2][3].

Speech coding is essential to the operation of the public switched telephone network (PSTN), digital cellular communications, videoconferencing systems and emerging voice over Internet protocol (VoIP) applications. The objective of speech coding is to represent speech in digital form with as few bits as possible while retaining the clearness and quality required for the particular application. Interest in speech coding is motivated by the evolution to digital communications and the need to minimize bit rate, and therefore, conserve bandwidth.

There is always an exchange between lowering the bit rate and preserving the delivered voice quality and clearness, however, depending on the application, many other restraints also must be considered, such as intricacy, delay, and performance with bit errors or packet losses.

Two networks that have been developed primarily with voice communications in mind are the public switched telephone network (PSTN) and digital cellular networks. Furthermore, with the popularity of the Internet, voice over the Internet Protocol (VoIP) is growing swiftly and is anticipated to do so for the near future [1].

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II. SPEECH CODING SYSTEM

When Based on these developments, it is feasible today, and it is likely in the near future, that our day-to-day voice communications will involve multiple skips including heterogeneous networks. This is a significant departure from the plain old telephone service (POTS) on the PSTN, and certainly, these future voice connections will vary to a great extent even from the digital cellular calls connected through the PSTN today. As the networks supporting our voice calls become less homogeneous and include more wireless links, many new challenges and opportunities emerge. [2]

In 1990's, there was almost an exponential growth of speech coding standards for a wide range of networks and applications, digital cellular, PSTN, multimedia streaming and standard in another network or application. According to the bandwidth occupied by the input and the reproduced source, speech and audio coding can be classified. Narrowband or telephone bandwidth speech occupies the band from 200 to 3400 Hz, while wideband speech is restricted in the range of 50 Hz to 7 kHz. The high quality audio is normally occupies the range of 20 Hz to 20 kHz. Since a higher bit rate implies a greater bandwidth requirement, the goal is always to minimize the rate required to satisfy the distortion constraint. For speech coding, we are interested in achieving a quality as close to the original speech as possible.

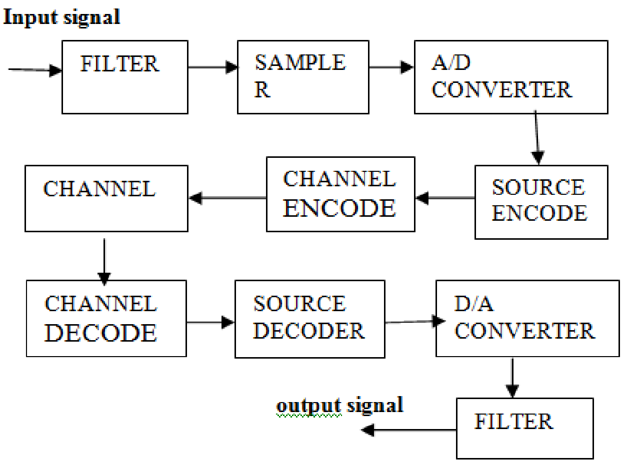

Fig.1 Speech coding system

The most common approaches to narrowband speech coding center are around two paradigms, i.e. waveform-following coders and analysis-by-synthesis methods. Waveform-following coders attempt to reproduce the time domain speech waveform as perfectly as possible, whereas analysis-by-synthesis methods utilize the linear prediction model and a perceptual distortion measure to reproduce only those characteristics of the input speech that is determined to be most important. Another approach to speech coding breaks the speech into separate frequency bands, called subbands, and then codes these subbands separately, possibly using a waveform coder or analysis-by-synthesis coding, for reconstruction and recombination at the receiver. Extending the resolution of the frequency domain decomposition leads to transform coding, wherein a transform is performed on a frame of input speech and the resulting transform coefficients are quantized and transmitted to reconstruct the speech from the inverse transform.

III. PROPERTIES OF SPEECH CODERS

The aim of speech coding is to enhance the quality of a speech signal at a particular bit-rate or to minimize the bit-rate at a given quality. The bit-rate at which the speech is to be transmitted or stored depends on the rate of transmission or storage, the computation of coding the digital speech signal and the quality of the speech signal required. Therefore the desirable properties of the speech coder include [1] [4] [12].

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

Low Bit-Rate - The lower the bit-rate of an encoded bit stream the lesser is the bandwidth required for transmission. But any decrease in the bit-rate results in a loss in the quality of the speech signal which is unwanted.

High Speech Quality - The decoded speech signal must have high quality and must be suitable for the intended application. Quality of the speech signal depends on factors like clearness, impulsiveness, pleasantness and speaker recognisability.

Robustness to Different Speakers and Languages - The speech coder must be general enough, so that it is used for any type of speakers like male, female and children, it can also be used for any type of language. But it is not a small task because every voice signal has its own characteristics. Robustness to Channel Errors - This is a vital for digital communication systems where channel errors have a negative impact on the quality of the speech signal.

Low Memory Size and Computational Complexity - In order to have good marketability for the speech coder, the costs linked with its implementation must be as low as possible. The cost of a product depends on the memory required to support its operations and the computational intricacy. Therefore, the speech coder must have lower memory requirements and computational intricacy to have better marketability.

Low Coding Delay - Coding delay is the time elapse from the time, the speech sample arrives at the encoder input to the time the speech sample appears at the decoder output. Hence the coding delay must be as low as possible.

A. **Different codec methodologies are used for GSM codecs**:

CELP*:* The Code Excited Linear Prediction Codec (CELP) is a vocoder algorithm was originally proposed in 1985 and gave a significant improvement over other voice

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codecs of the days. The basic principle of the CELP codec has been developed and used as the basis of other voice codecs including ACELP, RCELP, VSELP, etc. Hence the CELP codec methodology is now the most widely used speech coding algorithm. Accordingly CELP is now used as a generic term for a particular class of vocoders or speech codecs and not a particular codec. 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 optimising 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.

These coders can achieve transmission bit rates as low as 4.8 kbps. Basis for the AMR is CELP, a hydride procedure for data compression. The result of this process is a good voice quality, which communicate with the pulse -code modulation even at low bit rates. The basis for most hybrid coding processes is the coding by the help of linear predictive coding (LPC). The compression is more time-consuming than the decompression. The drawback of CELP is a signal delay of approximately 50 ml, which has been reduced in 1992 to 2 ms. The Algebraic Code Excited Linear Prediction (ACELP) is a patented advanced development of this process.

ACELP codec - The Algebraic Code Excited Linear Prediction (ACELP) codec or vocoder algorithm is a development of the CELP model. But the ACELP codec codebooks have a specific algebraic structure as indicated by the name. In the ACELP, signal is processed and separated into 20 ms frames. The coefficients of the linear prediction-filters are calculated by the help of the Levinson –Durbin-Algorithm. The signal is separated into 4 sub frames. Based on this the Pitch Delay Gain of the Pitch-syntheses filters are defined. After that the fixed codebook is defined. The LP-coefficients, indices and gains are quantized, sent to the recipient and re-synthesized [12][4].

VSELP codec - The VSELP or 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 executes poorly in the presence of noise. Thus, this voice codec is not now as extensively used, other newer speech codecs being preferred and offering far superior performance [3][12][10].

Residual (Error) Excited LPC -The justification 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. [1]

IV. AMR PRPOSED FOR GSM

The Adaptive Multi-rate codec (AMR), is now the most widely used GSM codec. The AMR codec was adopted by 3GPP in year 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 [5][10].

The world-wide coverage of GSM-like networks led to explosion of the number of subscribers, causing the saturation of some networks. Besides, the old technology used in GSM (Global System for Mobile Communication) systems provided the customer with an unsatisfactory speech quality in many common circumstances. To increase the capacity and the quality of GSM networks, ETSI (European Telecommunication Standards Institution) decided to standardize by bringing in a new speech transmission system called AMR. AMR is a combination of speech and channel Codecs activated and controlled by signaling means aimed at providing the best speech quality under background noise and transmission errors. The application of source and channel coding help mitigate the dependence of voice quality on channel condition, AMR is one remedy to this channel problem. The AMR codec would operate both in the full-rate (22.8 kbit/s) and half-rate (11.4 kbit/s) Channels of GSM. It would adapt to radio channel and traffic load conditions and select the optimum channel mode (full-rate or half-rate) and codec mode (bit-rate trade-off between speech and channel coding) to deliver the best combination of speech quality and system capacity [12] [9].

A. ADAPTIVE MULTI RATE CODER

First, AMR is a technique that is utilized to maintain good speech quality under varying channel conditions. This has several operating codec modes which are switched adaptively in accordance to the dynamics of the channel (good or bad channel). The process of dynamically switching due to varying channel conditions is known as AMR adaptation [3] [6]. The use of 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 [4]. However network operators are able to prioritize each station for either quality or capacity. Two variants of AMR exist, narrowband AMR and wideband AMR (AMR-WB). Narrowband AMR consists of eight codec modes with different source bit rates, from 12.2 kbps down to 4.75 kbps. It provides the

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traditional audio bandwidth of PSTN telephony of about 100–3500 Hz. AMR-WB contains nine different codec modes with source bit rates from 6.6 kbps up to 23.85 kbps, and with an audio bandwidth of 50–7000 Hz. The increased bandwidth improves the intelligibility and naturalness of speech significantly at the same time as the quality for music and mixed content material is improved. In the AMR NB 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.AMR full-rate works with the highest bit rate of 22.8 kb/s; whereas AMR half-rate works with the highest bit rate of 11.4 kb/s which is shown in the fig.1.2 [7][4].

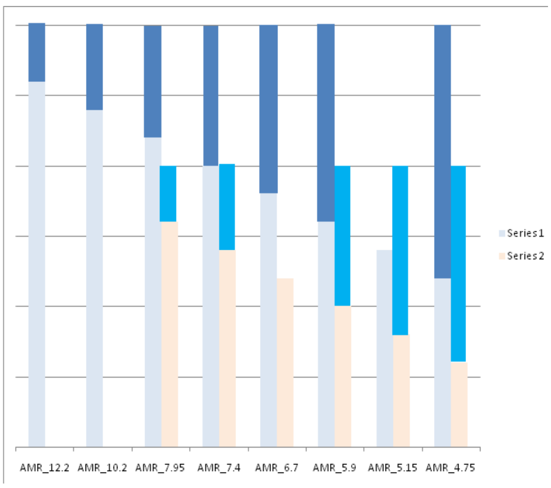
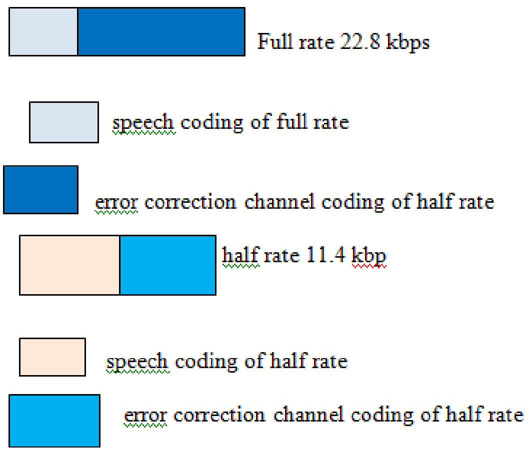


Figure 1.2 AMR full-rate and half-rate codec



The above codec modes are a precipitate of the ETSI standardization in 1999.

V. BLOCK DIAGRAM OF THE AMR CODEC SYSTEM

AMR working is a two step process; uplink channel and downlink channel. In uplink channel, MS sends information to BTS which contains speech data, codec mode indicator uplink (MIu), and codec mode request downlink (MRd). Speech data is the main information sent from MS via BTS with bit rate determined by its speech coder. The speech data is sent along with its codec mode indicator uplink (MIu) and codec mode request downlink (MRd). This process is repeated continuously for the next speech data sent. MIu comes from codec mode command uplink (MCu) to decide codec mode which will be used for uplink transmission from MS to BTS. MRd comes from computation of error condition at downlink channel from BTS to MS (downlink quality measurement). After MRd arrives at BTS, the process of downlink mode control is carried out to produce codec mode. Uplink channel performance process is very reliant on downlink channel performance process, and vice versa. In downlink channel, BTS send information to MS which contains codec mode command uplink (MCu), speech data and codec mode indicator downlink (MId). At first, MCu comes from the computation of error condition in uplink transmission (uplink quality measurement), which is then processed through uplink mode control to produce codec mode command uplink (MCu). Arriving at MS, MCu is then used to give codec mode order which will be used for uplink transmission from MS to BTS. MId comes from codec mode command downlink (MCd) which is used to decide codec mode to be used for downlink transmission from BTS to MS. The computation of the error condition quality in uplink and downlink channel is determined by the ratio of signal carrier (C) to interference (I) which expressed

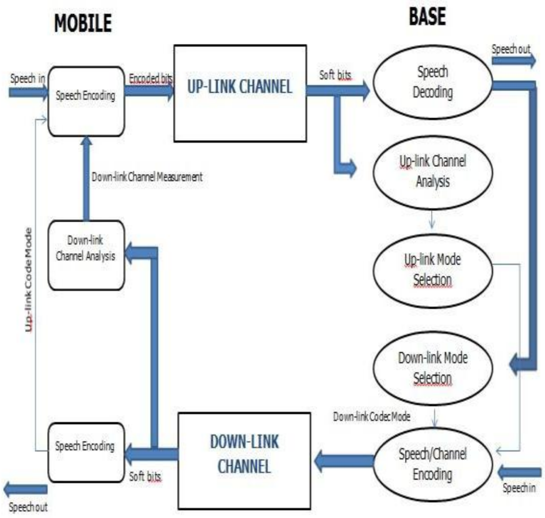


Fig 1.3 AMR block diagram [9]

in dB. The working process of the uplink and downlink channel in AMR is going on repeatedly [3] [6]. This means that if there is a sudden change in the channel condition, then there will also be a change in the codec mode in accordance with the channel condition at that moment. Codec mode is determined by the quality of the channel condition. If the channel condition is good, then there will be no significant error correction, so that the bit rate speech coding sent is higher than the bit rate channel coding (error correction). On the contrary, if the channel condition is bad, then a significant error correction will be needed. In this case, the information sent will have much correction, so that it will produce good voice quality [5].

The AMR speech codec is based on the ACELP algorithm, and consists of the multirate speech codec, a source controlled rate (SCR) scheme including a voice activity detector (VAD) and a comfort noise generation system, and an error camouflage mechanism to combat effects of transmission errors and lost packets. Voice Activity Detection (VAD), Comfort Noise Generation (CNG) and Discontinuous Transmission (DTX) are some important features of AMR. VAD discriminates speech from silence to aid in speech processing and the VAD in AMR helps improve the quality of speech coding. CNG is low-level background static added on purpose to help reduce some negative effects of silence such as sudden swings in sound levels from voice to silence. DTX technology controls the transmitter switch during wireless conversations so that the battery or amplifiers are not used unnecessarily during silent periods when there is no speech. The multi-rate encoding (i.e., multi-mode) capability of AMR and AMR-WB are designed for protecting high speech quality under a wide range of transmission conditions. Every AMR or AMR-WB codec implementation is required to support all the respective speech coding modes defined by the codec and must be able to handle mode switching to any of the modes at any time. Both codecs support voice activity detection (VAD) and generation of comfort noise (CN) parameters during silence periods. Hence, the codecs have the alternative to reduce the number of transmitted bits and packets during silence periods to a minimum. The operation of sending CN parameters at regular intervals during silence periods is generally called discontinuous transmission (DTX). The AMR or AMR-WB frames containing CN parameters are called Silence Indicator (SID) frames.

1. AMR-WB codec –

Adaptive Multi-Rate Wideband(AMR-WB codec) is known under its ITU designation of G.722.2. It is also based on the earlier popular Adaptive Multi-Rate, AMR codec. AMR-WB also utilizes an ACELP basis for operation, but it has been further developed and it provides improved speech quality therefore, it encodes the wider speech bandwidth. AMR-WB has a bandwidth extending from 50 - 7000 Hz which is radically wider than the 300 - 3400 Hz bandwidths used by standard telephones. Though this comes at the cost of additional processing, but due to advances in IC technology this is perfectly acceptable. The AMR-WB codec contains a number of functional areas i.e. it primarily includes a set of fixed rate speech and channel codec modes and its 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 used i.e. 50-6400 Hz and 6400-7000 Hz. These bands are coded separately to reduce the codec intricacy. This split also serves to focus the bit allocation into the subjectively most important frequency range [12] [10].

2. CHANNEL CODING

The AMR contains eight channel codecs for the FR channel and six for the HR channel. Channel coding executes error correction and bad-frame detection. The error correction in all modes is based on Recursive Systematic Convolutional (RSC) coding with puncturing to obtain the required bit-rates. Each codec mode uses a 6 bit CRC (Cyclic Redundancy Check) for detecting bad frames. All channel codecs use convolution polynomials specified for the previous GSM traffic channels (either for speech or data traffic channels) to maximize commonality with the existing GSM system. The frames contain 160 samples and are 20 milliseconds long. AMR uses various techniques i.e. ACELP, DTX, VAD & CNG. The usage of AMR requires optimized link adaptation that selects the best codec mode to meet the local radio channel and capacity requirements. If the radio conditions are bad, source coding is reduced and channel coding is increased. This improves the quality and heftiness of the network connection while sacrificing some voice clarity. In the particular case of AMR, this improvement is somewhere around S/N = 4-6 dB for usable communication. The new intelligent system allows the network operator to prioritize capacity or quality per base station. Link adaptation consists of channel quality measurement and codec/channel mode adaptation algorithms. Link adaptation in AMR is double: it adapts the bit-partitioning between speech and channel coding within a transmission channel (codec mode), and the operation in the full and half-rate channels (channel mode). Depending on the channel quality and possible network constraints (e.g., network load), link adaptation selects the optimal codec and channel mode.

VI. APPLICATION OF AMR

Use of AMR is a Multirate speech codec that can change the bit-rate from 12.2 Kbits/s to 4.75 Kbits/s every frame, i.e., on a packet basis. AMR speech codec is required; a trans-coding procedure in the Gateway between two networks can be removed. It could reduce the overall transmission delay and intricacy, and improve the speech quality [12] [13].

The AMR codec consists of the multi-rate speech coder, a source controlled rate scheme including voice activity detection and a comfort noise generation system, and an error camouflage mechanism to combat the effects of transmission errors and lost packets. The AMR speech coder is a single integrated speech codec with eight source rates from 4.75 Kbit/s to 12.2 Kbit/s, and a low rate background noise-encoding mode.

The AMR codec is capable of switching its bit-rate ever 20 ms speech frame upon command. The AMR codec selects the best source-coding rate (or codec mode) and channel-coding rate to deliver the best combination of speech quality and system capacity. The network controls the overall operation of AMR codec, in terms of used codec modes as well as general adaption behavior.

VII. COMPARISION DIFFERENT AMR CODEC

In above tables there are different types of AMR half and full rate. So this table compare the all parameters of AMR file according its bit rate. So according its bit rate we can use the AMR codec for the GSM technology.

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Table 1.1: AMR Codec Bit rate [7][8][12]

MODE	BITRATES (KBIT/S)	CHANNEL	FRAME TYPE	NO. OF BITS IN BIT SUFFERING	FRAME LENGTH	COMPRESSION
AMR 12.20	12.20	FR	4	0	244	8
AMR 10.20	10.20	FR	4	0	204	10.2
AMR 7.95	7.95	FR/HR	4	5	159	13.1
AMR 7.40	7.40	FR/HR	4	0	148	14.1
AMR 6.70	6.70	FR/HR	4	6	134	15.5
AMR 5.90	5.90	FR/HR	4	6	118	17.6
AMR 5.15	5.15	FR/HR	4	5	103	20.2
AMR 4.75	4.75	FR/HR	4	5	95	21.9

Table 1.2: Comparison table for AMR Codec

Codec Name m	BIT RATE (Kbps)	COMPRESSION TECHNOLOGY	Band-width	Sampling rate
AMR	12.2-4.75	ACELP	100-3500Hz	8000Hz
AMR-WB	23.85-6.60	ACELP	50-7000Hz	8K-16KHz

VIII. CONCLUSION

The As the frequency in the GSM technology is limited i.e. 900 MHz Therefore it is essential to save frequency, power, and to increase the channel capacity and in wav file bit rates are high so it consumes more frequency and more power. So AMR is adopted in GSM as the AMR codec operate both in the full-rate (22.8 Kbit/s) and half-rate (11.4 Kbit/s) Channels of GSM. It adapt to radio channel and traffic load conditions and select the optimum channel mode and codec mode to deliver the best combination of speech quality and system capacity. So mean while it consumes less frequency i.e. 300-3400 Hz. So overall the study of and performance of AMR codec is very useful and necessity for the cellular system where we can save the more frequency according to the voice recogination.

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