CHAPTER 2

METHODOLOGY

2.1 UWB TECHNOLOGY

Wireless data transfer capacity is a returning issue in current wireless communications. To make a rough simplification there are two variables to work with information density per frequency unit and the occupied frequency bandwidth. To use a wide bandwidth enables more data transfer capacity but imposes great challenges to the transceiver front-end. (Park et al. 2007) The analog front-end including the antenna has been an important part ever since the first wireless communication system was invented. It has been subjects for a lot of research throughout the years, with various parameters in focus in order to further enhance system performance. When the idea of ultra wideband radio entered the scene, the interest for radio frequency front-ends with very large bandwidth has got a rebirth.

Ultra-Wideband radio is a power-limited, and rapidly evolving technology, which employs short pulses with ultra low power for communication and ranging. A UWB impulse radio system is found to be very useful and consists of various satisfying features such as high data rate, high precision ranging, fading robustness, and low cost transceiver implementation. UWB is very promising for low-cost networks. UWB is a fast rising technology with uniquely attractive features inviting major advances in wireless communications, networking, radar, imaging, and positioning systems (Chia-Chin Chong et al. 2006).
Ultra wideband has actually experienced over 40 years of technological developments. In fact, UWB has its origin in the spark-gap transmission design of Marconi and Hertz in the late 1890s. Owing to technical limitations, narrowband communications were preferred to UWB. Originally, this concept was called “carrier less or impulse technology” due to its nature. UWB was used for applications such as radar, sensing, military communication and localization. A substantial change occurred in February 2002, when the federal communication commission issued a report allowing the commercial and unlicensed deployment of UWB with a given spectral mask for both indoor and outdoor applications in USA. This wide frequency allocation initiated a lot of research activities from both industry and academia. In recent years, UWB technology has mostly focused on consumer electronics and wireless communications.

2.1.1 Overview of Ultra-Wideband Communication

Ultra Wideband radio communication is used for small to medium range communications and positioning applications. UWB technology has been around since 1960s, when it was mainly used for radar and military applications (R. C. Qiu et al. 2005). The American Federal Communication Commission has published a first report on this subject where guidelines are given as to what can be expected from a regulatory point of view and it is expected that the European regulatory body issue similar restrictions. The key limitations for wireless communication using UWB are

- Maximum average -41.3 dBm/MHz or 75 nW/MHz Effective Isotropic Radiated Power (EIRP) in the frequency range 3.1 GHz – 10.6 GHz for indoor applications.
- Even lower maximum EIRP for other frequency bands, especially for GPS bands
• Peak-to-Average Ratio (PAR) of maximum 20 dB

The total emitted power is therefore upper limited by 0.55mW and even this low transmit power is not realistic, as it would require the entire bandwidth of 7.5GHz. The upper boundary is designated $f_H$, and lower boundary is designated $f_L$. The fractional bandwidth $B_f$ is defined as

$$B_f = 2 \cdot \frac{(f_H - f_L)}{f_H + f_L} \quad (2.1)$$

FCC Part 15 regulations limit the emitted Power Spectral Density from a UWB source measured in a 1 MHz bandwidth at the output of an isotropic transmit antenna at a reference distance. For indoor systems, the average output power spectral density is limited to -41.3dBm/MHz and with which compares the spectral occupation and emitted power of different radio systems (Zhang et al. 2008).

### 2.1.2 Advantages of UWB

**High data rate wireless transmission:** Due to the ultra-wide bandwidth of several GHz, UWB systems can support more than 500 Mb/s data transmission rate within the range of 10 m, which enables various new services and applications.

**High precision ranging:** Due to the nanosecond duration of typical UWB pulses, UWB systems have good time-domain resolution and can provide centimeter accuracy for location and tracking applications.

**Low loss penetration:** UWB systems can penetrate obstacles and thus operate under both line-of-sight and non-line-of-sight environments.

**Fading robustness:** UWB systems are immune to multipath fading and capable of resolving multipath components even in dense multipath
environments. The transceiver complexity can be reduced by taking the advantages of the fading robustness. The resolvable paths can be combined to enhance system performance.

**Security:** For UWB signal, the power spectral density is very low. Since UWB systems operate below the noise floor, it is extremely difficult for unintended users to detect UWB signals. Probability of intercept is low in UWB. The UWB system is also difficult to be interfered with because of its huge bandwidth.

**Coexistence:** The unique character of low power spectral density allows UWB system to coexist with other services such as cellular systems, wireless local area networks, global positioning systems, etc.

**Low cost transceiver implementation:** Because of low power of UWB signals, the RF chip and baseband chip can be integrated into a single chip using CMOS technology. The up-converter, downconverter, and power amplifier commonly used in a narrowband system are not necessary for UWB systems.

The UWB can provide a low cost transceiver solution for high data rate transmission. UWB systems communicate by modulating a train of pulses instead of a carrier. The carrierless nature of UWB results in simple, lowpower transceiver circuitry, which does not require intermediate mixers and oscillators. These benefits allow UWB radio to become a very attractive solution for future wireless communications and many other applications including logistics, security applications, medical applications, control of home appliances, search-and-rescue, supervision of children, and military applications (Qiu et al. 2002).
2.1.3 UWB Importance

The design objectives of UWB communication systems can be seen from the AWGN channel capacity as given by Shannon’s theorem

\[ C = B \log_2 (1 + \text{SNR}) \]  

(2.2)

where \( C \) is the divert limit in bits/s, \( B \) is the direct transfer speed in Hz, and \( \text{SNR} \) is the flag to-clamor proportion.

In conventional narrowband frameworks a given data transfer capacity is distributed to the administration and utilized just by this administration. As recurrence range is a rare asset, the data transmission will ordinarily be chosen as little as could be allowed. The main parameter of the channel limit that can be balanced is subsequently the SNR, which is the outline parameter that chooses the execution of the framework. One evident issue with the channel limit is it increments by the logarithm of the SNR. This implies just a little pick up can be accomplished from enhancing the SNR, which is a major issue when high piece rate remote associations are wanted.

UWB framework isn't band-constrained as in the narrowband case, yet rather power limited. Channel limit builds relative with the transmission capacity, exchanging data transmission for SNR is worthwhile i.e SNR is significant. Another advantage is SNR turns out to be low to the point that the channel limit increments practically corresponding with the SNR, attempting endeavors into enhancing the SNR more helpful than NB frameworks. Require over the top coding to satisfy given Nature of-benefit requests.

UWB should in this way have the capacity to coincide in an indistinguishable recurrence range from as of now allotted administrations without aggravating these and managing the impedance from these
administrations. This puts a furthest point of confinement on the transmitted energy of a UWB motion and its radiated range. Another intriguing element is the innate low power required in the transmitter, as the yield control is restricted to a small amount of a milliwatt. The aggregate power required by a UWB framework is in this manner not extremely constrained by the transmitted power, which here and there refers to the situation with narrowband frameworks, is making UWB appealing for battery-controlled gear like cell phones.

### 2.1.4 UWB Channel Model

A UWB channel show is gotten from the altered Saleh-Valenzuela display appropriate for home condition with both viewable pathway and non-line-of-sight conditions. (Foerster 2012) In the SV models, both the group and the beam landing times are demonstrated autonomously by Poisson forms. The numerical portrayal of the multipath channel motivation reaction can be exhibited as

\[
h_i(t) = X_i \sum_{l=0}^{L_i} \sum_{k=0}^{K_{il}} \sigma_{ikl}^i \delta(t - T_i^l - \tau_{ikl}^i)
\]

where, \( \sigma_{ikl}^i \) represents the multipath gain coefficient, \( T_i^l \) is the delay of \( l \) th cluster, \( \tau_{ikl}^i \) is the delay of the \( k \) th multipath component relative to the \( l \) th cluster arrival time (\( T_i^l \)), \( X_i \) represents the log-normal shadowing, and \( i \) refers to the \( i \)th realization. The modified Saleh-Valenzuela model uses the following definition, and assumes that \( \tau_{0,0} = 0 \).

- \( T_i \) = the arrival time of the first path of the \( l \)th cluster
- \( T_{ikl} \) = the delay of the \( k \)th path within the \( l \)th cluster relative to the first path arrival time, \( T_i \)
\( \Lambda = \) cluster arrival rate

- \( \lambda = \) ray arrival rate, i.e., the arrival rate of path within each cluster. The distribution of cluster arrival time can be presented by

\[
p(T_i / T_{i-1}) = \Lambda e^{-\Lambda (T_i - T_{i-1})} t > 0
\] (2.4)

And the ray arrival time by

\[
p(T_{k,l} / T_{(k-1,l)}) = \lambda e^{-\lambda (T_{k,l} - (k-1,l))} k > 0
\] (2.5)

where, the channel coefficients are defined by

\[
\alpha_{k,l} = p_k l \xi l \beta_{k,l}
\] (2.6)

And

\[
20 \log_{10}(\xi_{k,l}) \propto N(\mu_j, \sigma_1^2 + \sigma_2^2)
\]

\[
\left| \frac{\xi_{k,l}}{\beta_{k,l}} \right| = 10^{\left(\frac{\mu_j + \nu_j}{20}\right)}
\] (2.7)

where, \( n_1 \propto N(0, \sigma_1^2) \) and \( n_2 \propto N(0, \sigma_2^2) \) are independent and correspond to the fading on each cluster and ray, respectively. The behavior of the (average) power delay profile is

\[
E\left[ \xi_{k,l}^2 \beta_{k,l}^2 \right] = \Omega_0 e^{-T_k / T} e^{-T_l / T}
\] (2.8)

where, \( \Omega_0 \) is the mean energy of the first path of the first cluster, and \( p_{k,l} \) is equiprobable +/-1 to account for signal inversion due to reflections. The \( \mu_{k,l} \) is given by
where, $\xi_l$ reflects the fading associated with the $l^{th}$ cluster, and $\beta_{k,l}$ corresponds to the fading associated with the $k^{th}$ ray of the $l^{th}$ cluster. Finally, since the lognormal shadowing of the total multipath energy is captured by the term, $\xi_l$, the total energy contained in the terms $\alpha_{i,k,l}$ is normalized to unity for each realization. This shadowing term is characterized by

$$20\log_{10}(X_i) \sim N \left( 0, \sigma_{x}^2 \right)$$

The model derives the following channel parameters as an output

- Mean and root mean square delay
- Number of multipath components

Power decay profile The model inputs the following parameters

- $\Lambda$ = cluster arrival rate
- $\lambda$ = ray arrival rate i.e., the arrival rate of the multipath components within each cluster
- $\Gamma$ = cluster decay factor
- $\gamma$ = ray decay factor
- $\sigma_1$ = standard deviation of cluster lognormal fading term (dB)
- $\sigma_2$ = standard deviation of ray lognormal fading term (dB)
- $\sigma_x$ = standard deviation of lognormal shadowing term for total multipath realization (dB)
Four unique situations are characterized for the indoor UWB channel by Intel, specifically CM1, CM2, CM3, and CM4. CM1 speaks to a LOS situation, which is an unhindered way from transmitter to recipient with broadcasting range under 4 m. CM2 and CM3 relate to NLOS situation, where there are various checks made by human, furniture, material of room dividers, roof, between the transmitter and the beneficiary, with transmitting range under 4 m and 4-10 m individually. At long last, CM4 remains for a situation with extraordinary NLOS multipath channel.

2.2 H.264 /AVC ENCODER

H.264/AVC coder is collected of two main parts video encoder and video decoder. The encoder part performs prediction, transforming and encoding operations to generate a compressed bit stream. In H.264, while predicting the current blocks, previously coded pixels are used. For this reason, in an encoder inverse quantization, inverse transformation and reconstruction operations are also applied to quantized transform coefficients in order to produce the previously coded pixels. The video decoder part performs the opposite operations of encoding, inverse transforming and reconstruction to generate a decoded video sequence. The compressed H.264/AVC bit stream can be transmitted or stored in different mediums such as the internet or DVD disks. Because H.264/AVC is a lossy compression standard, in general, there will be some differences between the original video and the decoded one.
2.2.1 H.264/AVC Encoding Technique

Figure 2.1 demonstrates the H.264/AVC encoder chart. Each progression of the encoding procedure will be point by point in this segment. The Shading Space Change piece is the transformation from RGB space to YUV space.

Color Space Conversion

There are different ways to represent a picture, these way are called color space. The most known is the RGB space which codes each pixel with a grayscale value of the three colors composing the pixel. Red Green and Blue. As our visual perception is based on the intensity of the light which penetrates our eyes, we are more sensitive to the brightness of the picture than to the color itself. Therefore there is another way of coding a picture which is actually more accurate the YUV space. The luminance (Y) is the intensity of the light on the picture (its average of grey).
Transform

The H.264/AVC uses $4 \times 4$ block size while the prior standard based their transform on $8 \times 8$ block size which prevents from ringing artifacts and blocking artifacts which therefore provides a better video quality. This choice was motivated by the importance of prediction in H.264/AVC standard: smaller block size allow better prediction as they can provide more matches. As a result, an encoder based on $4 \times 4$ blocks will provide better data compression than an $8 \times 8$ DCT based encoder. However, the $8 \times 8$ block can be use for the Transform step. There are several transforms available depending on the profile the $4 \times 4$ transform, the $8 \times 8$ transform and the Hadamard transform (for $4 \times 4$ and $2 \times 2$). The H.264/AVC provides a new $4 \times 4$ or $8 \times 8$ spacial based transform, unlike the Discrete Cosine Transform, the new transform does not switch from spacial space to frequency space.

It is a purely integer transform which, as a result, provide an exact inverse transform. Consequently, there is no mismatch due to rounding coefficients as in DCT/IDCT transform. This means when transform and an inverse transform steps are processed, the matrix before and after remain the same. In addition, the new H.264/AVC transform suits more the low complexity platforms (such as DSP). Indeed, the former standards were based on the Discrete Cosine Transform which requires 32bits multiplication and 32 bits memory access. The new transform is based on simple integer arithmetic which is an approximation of the DCT transform and require only 16 bits for the arithmetic computation and 16 bits memory. As a conclusion, the new H.264/AVC Transform enhances the video quality avoiding mismatch between the Transform and the Inverse Transform steps, and reduces the complexity of both Transform and Inverse Transform steps.
**Intra Prediction:**

In intra prediction, the macroblock is coded from the before coded data from the current frame. The intra prediction is based on the principle that for a typical block of luma or chroma samples, there is relatively high correlation between samples in the block and samples that are immediately adjacent to the block. Therefore intra prediction uses samples from the previously coded data to predict the values in the current block. (Miguel Garcia et al. 2017) Data outside the current slice or frame is not considered. H.264 intra prediction offers nine prediction modes for 4*4 luminance (luma) blocks, nine prediction modes for 8*8 luminance blocks and four prediction modes for 16*16 luminance blocks. H.264 uses Rate Distortion Optimization technique to select the best mode among the nine prediction modes which gives a good quality video at lower bit rates. (NeuLion 2016).

The mode which gives less value for Rate Distortion cost is considered as the best prediction mode. The encoder should process all the nine prediction modes for a 4*4 luma block and has to select one among the prediction mode which gives less Rate Distortion cost. The encoder going through all nine prediction modes shows the computational complexity of the encoder and for every 4*4 luma block, an overhead bit has to be transmitted to the decoder to state which mode has been used in encoding the respective 4*4 luma block.

Hence Rate Distortion performances of an intra-frame coding are less than inter frame coding. Prediction efficiency and cost of signalling the prediction mode are related to the choice of intra prediction block size. Smaller intra prediction block size gives better prediction efficiency but increases the cost of signalling the prediction mode as more number of overhead bits are required to be transmitted to the decoder. Larger block sizes
give less accurate prediction but decreases the cost of signalling the prediction mode as less number of bits are needed to be transmitted.

**Inter Prediction:**

In inter prediction, the Macro Block (MB) is coded from the earlier coded data from previous or future frames (neighbouring frames). Inter prediction is based on the principle of predicting a block of luma and chroma samples from a picture that has been coded and transmitted, a reference picture. Inter prediction removes temporal redundancy from the neighbouring frames and therefore achieves high compression rate. 2.2.2 Transform and Quantization.

We know that visual distortion is introduced into the signal as a trade off for higher compression performance. Prediction stage in H.264 is lossless. Hence distortion occurs in the transform/quantization process. Mismatch between the encoders and the decoders are quite common in many codec’s. In H.264, the transform and quantization processes are done 19 to minimize the computational complexity and to avoid mismatch between encoders and decoders.

Previous standards for image and video compression used two dimensional Discrete Cosine Transform (DCT). A two dimensional discrete cosine transform is given by:

\[
X_{ij} = \sum_{x=0}^{N-1} \sum_{y=0}^{N-1} C_x C_y Y_{xy} \cos \left( \frac{(2j+1)y\pi}{2N} \right) COS \left( \frac{(2j+1)x\pi}{2N} \right)
\]  

(2.11)

Here, Yxy are the input coefficients and Xij are output image or residual samples. During the computation of the DCT, some irrational factors, \( \cos \frac{a\pi}{2N} \) have to be approximated which significantly bring mismatch between the -
encoder and decoder implementations. Hence H.264 uses a Core Transform, an integer transform which is based on integers and hence the drift between the encoder and decoder implementations is eliminated.

**Figure 2.2 Macro Block partitioning for Inter Prediction**

Henceforth in H.264, pieces of leftover examples are changed utilizing a 4*4 or 8*8 whole number change, a rough type of Discrete Cosine Change. The yield of the change, piece of change coefficients, is quantized (i.e. adjusting off by separating with whole number esteem). H.264 utilizes a scalar quantizer. The whole number which is isolated is known as the Quantization Parameter. A higher estimation of Quantization Parameter implies that more coefficients are zero bringing about high pressure to the detriment of poor picture quality.

**Quantization**

Quantization generally refers to the quantization step and the ZigZag Scan. The quantization part scales the matrices of coefficients from the Transform bloc in order to reduce the difference between the coefficients.
This reduces the amount of bits needed to code the matrix. Each coefficient is quantized using a Scalar quantization, and the parameter of the quantization can take 52 values (the H.263 only provided 31 values). As a result, we observe a better data compression which provides a 12% bit rate reduction. The ZigZag scan reads the quantized matrix and reorders the coefficient according to a zigzag pattern. The aim of rearranging the coefficients is to gather the zero values. As most of the non-zero coefficients are in the top-left part of the matrix, the ZigZag pattern starts with these coefficients. The purpose of reordering is to facilitate the next encoding step: the entropy coding.

**Entropy Coding**

Entropy coding is the last stage of video coding. It is a lossless coding technique that replaces the data element with coded representation, which reduces the data size significantly. In H.264/AVC, two modes of entropy coding - context adaptive variable length coding and context adaptive binary arithmetic coding are used. The context adaptive variable length coding mode offers approximately 10%-15% enhanced compression efficiency compared to the context adaptive variable length coding mode. The context adaptive variable length coding consists of two stages. First, the blocks produced during the core coding process, transformed and quantized motion compensated macroblocks, motion vectors, different flags, and parameters, are converted into a sequence of binary symbols, known as binarization.

**2.3 H.264/AVC DECODER**

The Entropy Interpreting is the turnaround operation of context adaptive variable length coding or context adaptive binary arithmetic coding,
it is tied in with recreating the "macro block coefficients" from the NAL bit stream utilizing the Query Tables.

**Figure 2.3 H.264/AVC Decoder**

**Inverse Quantization and Inverse Transform**

The Inverse Quantization and Inverse Transform are the inverse operation of the Transform and Quantization steps of the encoder. As mentioned before, the Inverse Transform is the exact inverse operation of the Transform so it avoids losses due to mismatches. As a result, the video quality in increased. The Inverse Quantization consists of restructuring the matrix from the entropy decoding step output and scaling it to get the matrix as it was before the Quantization step in the encoder.

**In-loop Deblocking Filter**

The In-loop Deblocking Filter is a new feature introduced in the H.264/AVC standard. Its reason is to reduce the video quality losses due to
macro blocks segmentation. Indeed, coding a video using macro block partition can create what is called "blocking artifacts". As the video is encoded macro block by macro block, (and mainly because of the Quantization step) the edges of the macro blocks become less accurate as they tend more to the "general value" of the macro block. As a result a strong difference between the edges of two neighboring macro blocks might be observed, and so the picture might look like "pixelized". The filter is called "In-loop" because it is performed right after the Inverse Transform and before the Frame Store Woon-Sung Park et al. (2016).

The filter is included in the process, it is not a post-processing step, which means that it filters the current macro blocks and not a reconstructed frame, so the filtering is taken into account for intra and inter prediction. The filter processes separately on luminance and chrominance components. It filters first vertically (from the top to the bottom) and then horizontally (from the right to the left). Sometimes, there is a real difference between the edge of the macro blocks not due to the encoding/decoding process but to the image itself. That is why a decision process is included in the In-loop Deblocking Filter. The process is the same for both vertical and horizontal filtering.

2.4 H.265/HEVC ENCODER

H.265/High Efficiency Video Coding (HEVC) is the latest Video Coding format. It challenges the state-of-the-art H.264/AVC Video Coding standard which is in the industry by being able to reduce the bit rate by 50%, retaining the same video quality.

HEVC is designed to address existing applications of H.264/MPEG-4 AVC and to focus on two key issues increased video resolution and increased use of parallel processing architectures.
Figure 2.4 H.265/HEVC Encoder

Motion vector signaling:

The HEVC standard uses a technique called advanced motion vector prediction to derive several most probable candidates based on data from adjacent PBs and the reference picture. A “merge” mode for MV coding can be also used, allowing the inheritance of MVs from neighboring PBs. Moreover, compared to H.264/MPEG-4 AVC, improved “skipped” and “direct” motion inference are also specified (G. J. Sullivan et al. 2012)

Motion compensation:

The HEVC standard uses quarter-sample precision for the MVs, and for interpolation of fractional-sample positions it uses 7-tap (filter coefficients: -1, 4, -10, 58, 17, -5, 1) or 8-tap filters (filter coefficients: -1, 4, -11, 40, 40, -11, 4, 1). In H.264/MPEG-4 AVC there is 6-tap filtering (filter coefficients: 2, -10, 40, 40, -10, 2) of half-sample positions followed by a bi-linear interpolation of quarter-sample positions. Each PB can transmit one or two motion vectors, resulting either in uni-predictive or bi-predictive coding, respectively. As in H.264/MPEG-4 AVC, a scaling and offset operation may
be applied to the prediction signals in a manner known as weighted
prediction. (G. J. Sullivan et al. 2012)

**Inter Prediction**

The improvements of HEVC over AVC in this prediction mode are
as follows. HEVC uses the so-called merge-mode, where, motion parameters
are not encoded, instead, a candidate list of motion parameters is created from
the corresponding PU. Generally, motion parameters of spatially neighboring
blocks and also temporally predicted motion parameters that are obtained
based on the motion data of a co-located block in a reference picture. These
chosen motion parameters is signaled through an index into the candidate list.
Advanced motion vector prediction algorithm is used for prediction. In
Advanced motion vector prediction algorithm, a candidate list is created for
each motion vector. The candidate list may consist of motion vectors of
neighboring blocks with the same reference index and also temporally
predicted motion vectors. These motion vectors are coded by signaling an
index to the candidate list for specifying the chosen predictor and coding a
dereference vector. These tools help in coding of motion parameters e ciently in
comparison to previous standards.

**Intra-picture prediction**

Intra prediction in HEVC is quite similar to H.264/AVC. Samples
are predicted from reconstructed samples of neighboring blocks. The mode
categories remain identical DC, plane, horizontal/vertical, and directional
although the nomenclature for H.264’s plane and directional modes has
changed to planar and angular modes, respectively. For intra prediction,
previously decoded boundary samples from adjacent PUs must be used.
Directional intra prediction is applied in HEVC, which supports 17 modes for
4x4 block and 34 modes for larger blocks, inclusive of DC mode (Athina
Directional intra prediction is based on the assumption that the texture in a region is directional, which means the pixel values will be smooth along a specific direction (Athina Kalampogia et al. 2016). The increased number of directions improves the accuracy of intra prediction. However, it increases the complexity and increased overhead to signal the mode (Athina Kalampogia et al. 2016). With the flexible structure of the HEVC standard, more accurate prediction, and other coding tools, a significant improvement in coding efficiency is achieved over H.264/AVC. HEVC supports various intra coding methods referred to as Intra_Angular, Intra_Planar and Intra_DC. An evaluation of HEVC coding efficiency compared with H.264/AVC is provided. It shows that the average bit rate saving for random access high efficiency (RA HE) case is 39%, while for all intra high efficiency (Intra HE) case this bit rate saving is 25%, which is also considerable. It seems that further improvement of intra coding efficiency is still desirable.

**Quantization control**

As in H.264/MPEG-4 AVC, uniform remaking quantization is utilized as a part of HEVC, with quantization scaling grids upheld for the different change piece sizes.

**Entropy Coding**

HEVC uses context adaptive binary arithmetic coding for entropy coding which is similar to the one used in H.264/MPEG-4 AVC. It has some changes to improve its throughput speed. These improvements can be used for parallel processing architectures and its compression performance, and to reduce its context memory requirements.
In-Loop Deblocking Filter

The HEVC standard uses a deblocking filter in the inter-picture prediction loop as used in H.264/MPEG-4 AVC. But design has been simplified in regard to its decision-making and filtering processes, and is made more friendly to parallel processing.

Sample adaptive offset (SAO)

A non-linear amplitude mapping is introduced in the inter-picture prediction loop after the deblocking filter. The goal is to better reconstruct the original signal amplitudes by using a look up table that is described by a few additional parameters that can be determined by histogram analysis at the encoder side.

PROBABILISTIC NEURAL NETWORK (PNN) FOR IN-LOOP FILTERING

PNN is utilized as a classifier. Info edge and reference outline is given as a contribution to the PNN. A solitary "predisposition unit" is associated with every unit aside from the info unit. Net actuation is

\[
net_j = \sum_{i=1}^{d} a_i w_{ji} + w_{j0} = \sum_{i=0}^{d} a_i w_{ji} \equiv w_j^t a,
\]

(2.12)

Here \(a\) is the input, \(w\) is a weight, the subscript indexes units in the input layer, \(j\) in the hidden layer; \(w_{ji}\) represents the input-to-hidden layer weights at the hidden unit \(j\). For each hidden unit there is an output which is a nonlinear function of its activation. PNN detaily explain in chapter 4.

\[
x_j = F(net_j)
\]

(2.13)
FEATURES H.264/AVC Vs H.265/HEVC

Table 2.1 Describe the various features of H.264/AVC and H.265/HEVC encoder.

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<th>H.264/ AVC</th>
<th>H.265/ HEVC</th>
</tr>
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<td>Coding Unit</td>
<td>16x16, Intra or inter-coded MB</td>
<td>64x64, 32x32, or 16x16, hybrid intra/inter in a LCU, quad-tree signaling of CU partitioning</td>
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<tr>
<td>Intra prediction</td>
<td>Luma-8x8: angular(8 modes), DC Luma 16x16/Chroma 4x4: Ver, Hor, DC, plane</td>
<td>8/7-tap for luma, 4-tap for chroma, WP</td>
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<tr>
<td>Inter prediction partitions</td>
<td>16x16, 16x8,8x16,8x8,8x4,4x8,4x4</td>
<td>2Nx2N,2NxN,Nx2N,NxN (N=4,8,16,32)</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2NxnU,2NxnD,nLx2N,nRx2N (N=8,16,32)</td>
</tr>
<tr>
<td>Skip/direct/merge mode</td>
<td>Skip/direct, single MVP candidate</td>
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2.5 ANFIS RATE CONTROLLER

The Neural Network and the Fuzzy Logic Neural networks are two integral technologies. The Fuzzy inference system implemented in the framework of adaptive networks. By using a hybrid learning procedure, the ANFIS can construct an input-output mapping based on both human knowledge (in the form of fuzzy if-then rules) and stipulated input-output data pairs. It is viewed as black box approach. The Neuro-fuzzy program can be categorized into three groups (Jang et al 2013)

a) The rule basis fuzzy design is build up by a supervised NXN learning technique.

b) The fuzzy rule basis design is build up by reinforcement-based learning.

c) The rule wise fuzzy design is build up by NXN to construct its fuzzy partition of the input space 1.2.

ANFIS is one of the Neuro-fuzzy models. Neural networks and fuzzy systems both are stand-alone systems. With the increase in the complexity of the process being modeled, the difficulty in developing dependable fuzzy rules and membership functions increases. This has led to the development of another approach which is mostly known as ANFIS approach. It has the benefits of both neural networks and fuzzy logic. One of the advantages of fuzzy systems is that they describe fuzzy rules, which fit the description of real-world processes to a greater extent. Another advantage of fuzzy systems is their interpretability it means that it is possible to explain why a particular value appeared at the output of a fuzzy system. In turn, some of the main disadvantages of fuzzy systems are that expert’s knowledge or instructions are needed in order to define fuzzy rules, and that the process of tuning of the parameters of the fuzzy system (e.g. parameters of the
membership functions) often requires a relatively long time. Both these disadvantages are related to the fact that it is not possible to train fuzzy systems. (Johan De Praeter et al. 2003)

A diametrically opposite situation can be observed in the field of neural networks. It is known that neural networks are trained, but it is extremely difficult to use a prior knowledge about the considered system and it is almost impossible to explain the behavior of the neural network system in a particular situation. In order to compensate the disadvantages of one system with the advantages of another system, several researchers tried to combine fuzzy systems with neural networks. A hybrid system named ANFIS (Adaptive-Network Based Fuzzy Inference System) has been proposed by Jang. Fuzzy inference in this system is realized with the aid of a training algorithm, which enables to tune the parameters of the fuzzy system

2.5.1 ANFIS Systems

Adaptive Neuro-Fuzzy Inference Systems a type of flexible systems that is functionally comparative to fuzzy inference systems. The architecture of ANFIS is representing the Sugeno fuzzy model and the Tsukamoto fuzzy model both. The (Figure 2.5) is showing a complete ANFIS with its layers.

![Figure 2.5 ANFIS Block Diagram](image)
Layer 1: Contains the membership functions with adaptive parameters.

Layer 2: The Relationship between membership functions with fixed parameters (calculates the firing strength).

Layer 3: Normalizes the strength of firing.

Layer 4: Calculates individual output based on the normalized firing strength with adaptive parameters.

Layer 5: Parameters are assessed with the smallest squares method of forward pass. Here, the condition for the individual yield is considered as the capacity of information. Parameters gauge the back spread and the angle descent. The parameters are utilizing for participation capacities.

2.5.2 FIS Technique

The procedure of developing the leveling from a given feedback to an outcome using the fuzzy logic is known as FIS technique (Koushik, et al. 2003). The leveling then provides a support from which choices can be made, or styles discovered. The procedure of the fuzzy inference system includes all parts that are described in the If-Then Rules Logical Operations, and Membership Functions

- The Fuzzy Inference System (FIS) is a way of applying a feedback area to an outcome area using fuzzy logic
- FIS uses a compilation of fuzzy account features and guidelines, instead of Boolean reasoning, to consider the data value.
The rules in FIS are called fuzzy expert system. These systems are the rules of fuzzy production. The form of the rules is: If p, then q, where p and q are fuzzy statements.

For example, in a fuzzy rule If x is low and y is high then z is the medium. Here x is low, y is high; z is the medium. Those are fuzzy statements. The value of x and y are input variables. The value of z is an output variable.

The antecedent explains to what level the concept is applicable, while the summary designates a fuzzy operator to produce the outcome factors.

Most resources for dealing with fuzzy expert systems consider more than one summary per concept.

The set of rules in a fuzzy expert system is known as the knowledge base.

The functional operations in the fuzzy expert system proceed in the following steps, in this 1, 2, 3 and 4 are shows in below (Figure 2.6).

1. Fuzzification
2. Fuzzy Inferencing (It applies implication method)
3. Aggregation of all output
4. Defuzzification
Figure 2.6 The Basic Architecture of a Fuzzy Expert System

**Fuzzification**

In Above (figure 2.6) the procedure of this method, association functions explained on feedback factors are enforced to their definite values so that the level of fact for each concept assumption can be driven. In the antecedent part, the announcement of fuzzy logic resolve to a level of account or membership between the value of 0 and 1. If antecedent method has only one part, then this is the degree of approval for the rule. If the antecedent method has several parts, use the operators of fuzzy logic and determine the antecedent to a single value. The value should be between 0 and 1. Antecedent may be joined by OR, AND operators. 0 For OR  max 1 For AND min.

**Fuzzy Inferencing**

In the process of the inference Fact value for the assumption of each concept is calculated and used to the summary aspect of each concept. The results in the fuzzy sets are to be allocated to each outcome varying for each concept. The use of the degree of support for the integrated guideline is to configure the fuzzy set outcome. The consequent of a fuzzy rule assigns an
entire fuzzy set to the output. If the antecedent is only partially true, (i.e., is
assigned a value less than 1), then the resultant fuzzy set is truncated
according to the implication method (Koushik 2003) . If the consequent of a
rule has multiple parts, then all values are impacted similarly by the output of
the antecedent. The consequent specifies a fuzzy set to be assigned to the
output. The implication function then modifies that fuzzy set to the degree
specified by the antecedent. The following functions are used in inference
rules (Rule defined in Chapter 3 and Chapter 4). The min and prod are
commonly used as the rules of interference. Min abbreviates the membership
function of the consequent.

**Aggregation of all outputs**

It is the process where the outcome of each concept is mixed with
only one the fuzzy set. The input of the aggregation process is for listing the
features of the outcome. It came back with the implication procedure for
every concept. The outcome of the process of aggregation is a fuzzy set for
each outcome volatile. Here, all fuzzy sets allocated to each outcome varying
are mixed together to type a single fuzzy set for each outcome varying using a
fuzzy aggregation operator. Some of the most commonly used aggregation
operators are The maximum an overall point basis maximization of the fuzzy
sets. The sum an overall point basis summation of the fuzzy.

**Defuzzification**

In De-fuzzification the output set of fuzzy logic is converted to a
fresh number. The centroid method and the maximum method are two
commonly used techniques. The centroid method is for searching the varying
value of the centre of severity of the membership function for the fuzzy value.
It also calculates the fresh value of the outcome. The maximum method is one
of the varying principles at which the fuzzy set has its highest possible fact
value. It selects the fresh value for the outcome value. Some other method for defuzzification is Bisector, which is the centre of the maximum value (the average of the maximum value of the output set), the largest of the maximum value, and smallest of the maximum value, and etc.

2.5.3 ANFIS Learning Method

Neural networks, the back propagation algorithm are used to learn, or adjust weights on connecting arrows between neurons from input-output training samples. In the ANFIS structure, the parameters of the premises and consequents play the role of weights. Specifically, the shape of membership functions in the “If” part of the rules is determined by a finite number of parameters. These parameters are called premise parameters, whereas the parameters in the “THEN” part of the rules are referred to as consequent parameters. The ANFIS learning algorithm consists of adjusting the above set of parameters. For ANFIS, a mixture of back propagation and least square estimation is used.

Back propagation is used to learn the premise parameters, and least square estimation is used to determine the parameters in the rules consequents. A step in the learning procedure has two passes. In the forward pass, node outputs go forward, and the consequent parameters are estimated by least squares method, while the premise parameters remain fixed.

In the backward pass the error signals are propagated backwards, and back propagation is used to modify the premise parameters while consequent parameters remain fixed. This combination of least-squares and back propagation methods are used for training FIS membership function parameters to model a given set of input/output data. The performance of this system will be evaluated using RMSE, root mean square errors (difference between the FIS output and the training/testing data output), defined as:
The Root Mean Square Error (RMSE) is used to measure the difference between desired output \( (y_k) \) and actual system output \( (o_k) \) and \( n \) is the number of training/testing samples.

### Membership Function and Rules Selection for ANFIS

In a conventional fuzzy inference system, the number of rules is decided by an expert who is familiar with the target system to be modeled. In ANFIS simulation, however, no expert is available and the number of membership functions assigned to each input variable is chosen empirically, that is, by plotting the data sets and examining them visually, or simply by trial and error. For data sets with more than three inputs, visualization techniques are not very effective and most of the time we have to rely on trial and error. This situation is similar to that of neural networks there is just no simple way to determine in advance the minimal number of hidden units needed to achieve a desired performance level. There are several other techniques for determining the numbers of membership functions and rules, such as clustering methods. In a fuzzy inference system, basically there are three types of input space partitioning

- Grid partitioning method
- Scatter partitioning method: It include Fuzzy-C means clustering method. Subtractive clustering method.
- Tree Partitioning method. In thesis FIS is generated by using Grid partitioning and scatter partitioning.

Rule of our proposed system, explain in chapter 3 and chapter 4.