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Video QoS/QoE over IEEE802.11n/ac: A Contemporary Survey

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DEDICATION

I dedicate this thesis to my parents (Kazhal and Rebwar) for their support and encouragement. To professor (Bruce Hartpence) for his support and guidance through the entire process. To my friends at RIT, especially (Swagata D. Chaudhury) for her help and support.

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ABSTRACT

Video QoS/QoE over IEEE802.11n/ac: A Contemporary Survey

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The demand for video applications over wireless networks has tremendously increased, and IEEE 802.11 standards have provided higher support for video transmission. However, providing Quality of Service (QoS) and Quality of Experience (QoE) for video over WLAN is still a challenge due to the error sensitivity of compressed video and dynamic channels. This thesis presents a contemporary survey study on video QoS/QoE over WLAN issues and solutions. The objective of the study is to provide an overview of the issues by conducting a background study on the video codecs and their features and characteristics, followed by studying QoS and QoE support in IEEE 802.11 standards. Since IEEE 802.11n is the current standard that is mostly deployed worldwide and IEEE 802.11ac is the upcoming standard, this survey study aims to investigate the most recent video QoS/QoE solutions based on these two standards. The solutions are divided into two broad categories, academic solutions, and vendor solutions. Academic solutions are mostly based on three main layers, namely Application, Media Access Control (MAC) and Physical (PHY) which are further divided into two major categories, single-layer solutions, and cross-layer solutions. Single-layer solutions are those which focus on a single layer to enhance the video transmission performance over WLAN. Cross-layer solutions involve two or more layers to provide a single QoS solution for video over WLAN. This thesis has also presented and technically analyzed QoS solutions by three popular vendors. This thesis concludes that single-layer solutions are not directly related to video QoS/QoE, and cross-layer solutions are performing better than single-layer solutions, but they are much more complicated and not easy to be implemented. Most vendors rely on their network infrastructure to provide QoS for multimedia applications. They have their techniques and mechanisms, but the concept of providing QoS/QoE for video is almost the same because they are using the same standards and rely on Wi-Fi Multimedia (WMM) to provide QoS.

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1. INTRODUCTION

Wireless networks have evolved extensively over the last decade, and this growth is expected to continue. The primary reason behind this development is the demand for mobility, high-bandwidth and support for the variety of content available to transmit over heterogeneous networks. Various applications are flourishing to enable developing multimedia delivery services over wireless networks. WLANs are designed to provide support for multimedia traffic such as voice, video streaming, online gaming and video conferencing. However, streaming high-quality videos over WLAN with limited bandwidth is still a challenge [1] [2]. By 2018 more than 66% of the traffic over WLAN will be videos [3]. Transmitting video over the wireless channels is challenging because wireless channels are dynamic and error-prone same as compressed video content which is error sensitive and time critical as well [4].

To address video transmission issues, two approaches should be taken into consideration, reducing video size and enhancing the channel bandwidth. Video compression is the technique that has been used to reduce video size and remove redundant video data. Various video codecs have been developed to reduce video size and enhance quality. The most used video codec is H.264, which is the most efficient compression technique in use for video delivery. On the other hand, wireless bandwidth has advanced, and IEEE 802.11 standards are continuously enhanced. IEEE 802.11n is the standard that currently in use which was first introduced in 2009. It can provide up to 600 Mb/s using multiple-input–multiple-output (MIMO) technology. IEEE 802.11ac was approved in 2014. This standard aims to reach maximum aggregate system throughput of at least 1 Gigabit per second on a 5-GHz band [5]. However, Improvements have been made in both areas (video codecs and wireless bandwidth). Video transmission over WLAN is still a challenge because of the nature of video and wireless channels. This thesis aims to survey most recent and relevant proposed solutions for this issue over the last six years and evaluate those solutions based on the two standards mentioned above.

1.2 Problem Statement

Wireless bandwidth improved in both 802.11n/ac but providing Quality of Service (QoS) for video, is still not satisfied. As mentioned above, compressed video is error sensitive. If any transmission error happens and a frame corrupts it will not only affect that frame but also the following frames. Moreover, latency threshold needs to be maintained within a certain range for video data packet transmission. The decoding time at the receiver frequently decides on the threshold delay for the next frame when the current video frame is being displayed. Any packet that exceeds the delay threshold will be unusable. Wireless technology has advanced and many new features have been added to address this issue but still video transmission over WLAN is a challenge. Much research has been done to find solutions. Each research project targeted a particular layer or two combined layers to investigate and propose a solution based on the nature of the layer(s). Three major layers are involved in video transmission issues over the WLAN. These are the Physical (PHY), Media Access Control (MAC) and application layers. With 802.11ac enough bandwidth is provided for video transmission, which means PHY layer solutions alone are less likely to be considered. PHY layer solutions will be effective when used as part of a cross-layer solution with another layer. In the past few years, several single-layer and cross-layer solutions have been proposed. This thesis proposes a survey investigation on many possible solutions considering video codecs which are at the application layer, channel bandwidth at the PHY layer and error detection and correction at the MAC layer, focusing only on the two standards 802.11n/ac. 802.11n is the standard that is in use today, and 802.11ac has started to come to the market recently. Moreover, solutions by three popular wireless vendors will be presented and evaluated based on their effectiveness to solve video QoS/QoE issues.

The rest of the thesis is organized as the following: Chapter 2 provides a brief summary of the related works. Chapter 3 presents a technical overview of both video codecs and IEEE 802.11n/ac standards. Chapter 4 overviews QoS/QoE metrics and implementation techniques in IEEE 802.11 standards. Chapter 5 presents a technical analysis of the most recent proposed solutions for video over WLAN issues. Chapter 6 presents the findings and open key problems. Chapter 7 presents the general conclusions and the future works.

2. RELATED WORKS

Several surveys similar to this one have been done before. One focused on different innovations that have been developed to improve video transmission over 802.11. The primary focus was on three network layers, (Application, MAC and Physical) by categorizing solutions into two main layers, single-layer and cross-layer. Introducing 802.11 standards and then analyzing video QoS over WLAN and differentiated between several proposed 802.11 solutions depending on those three layers. Furthermore, researchers went beyond QoS performance and discussed non-QoS measures such as power-efficient and security solutions for video over 802.11 [4]. Providing QoS for real-time multimedia transmission were studied, by examining various proposed PHY and MAC layer solutions. Aiming to study the scheduling mechanisms, the call admission control algorithms, and the anticipated MAC improvement that are proposed for 802.11 to satisfy QoS for multimedia transmission [6].

QoS satisfaction over WLAN has been studied focusing on MAC protocols in 802.11e, investigating in QoS requirements for voice and video. The study was divided into two sections: how 802.11 works and categorizing MAC protocols. The main goal of the research was to provide pros and cons of different techniques that have been used to provide QoS over WLAN [7]. Wireless video surveillance requirement and challenges are analyzed, focusing on PHY layer, and the main goal was to survey all available solutions that maximize video quality over 802.11 regarding recourse limitation. The research strategy was surveying video capturing and video coding mechanisms, then analyzing video transmission and presenting issues related to video encryption and security [8]. On the other hand, different video codecs have different quality, existing video coding standards and video coding approaches have been studied to provide a better understanding of video, and it is features and characteristics [9]. In addition, channel assignment is considered one of the mechanisms that enhance the performance of 802.11 standards. Channel assignment schemes have been studied, and different schemes were compared in a qualitative manner regarding complexity, scalability, and algorithm execution behaviors [10].

3. VIDEO OVER WIRELESS OVERVIEW

The research strategy is to analyze the most recent innovations and solutions to solve the video over WLAN transmission issues. Some of the solutions are applied to a single layer, while some others require two layers to function. Solutions that involve video compression techniques and video codecs will be application layer solutions. Physical layer solutions will be on enhancing throughput, while MAC layer solutions will be error detection and correction solutions. However, video transmission over WLAN is a single problem with many different solutions. Investigating in this problem will require categorization and classification. Therefore, this chapter reviews video coding standards and network architecture of both 802.11n/ac standards.

3.1 Video codecs

Video codecs have been advanced and evolved for telecommunication applications through the ITU-T coding standard developments, starting with H.261. Several video coding standards have been designed to compress the video and provide better quality. Each video codec has different characteristics and different quality. This section reviews those video codecs that mostly used in video transmission over WLAN. For example, H.264 is the most used video codec so far. The previous version was H.263 and HEVC (H.265) is the most recent compression technique that introduced in 2013, with the expectation to provide higher compression efficiency than H.264. Despite the ITU-T coding standards, there are other video codecs that worth mentioning such as Google VP8 coding standard. However, many research projects were made proposing solutions for the video QoS issues based on the video codecs. This analysis of video codecs will lead to a conclusion that can explain how compression technique will affect the transmission process over both standards 802.11/n/ac.

3.1.1 H.263

H.263 is an ITU-T low bit-rate video coding standard developed in 1995/1996. This codec was originally designed to utilize H.324 video conferencing and video telephony based systems such as Public Switched Telephone Network (PSTN). Later on, it was found to be used in H.323 (RTP/IP based systems), Secure Real-time Transport Protocol (SRTP) and Session Initiation Protocol (SIP). This video coding standard was developed based on the previous H.261 standard, and it is backward compatible with it. Techniques such half-pixel, full-pixel precision and variable length coding are used to improve video quality [11]. In addition, with the introduction of this video codec, four optional improvement modes were introduced:

- **Unrestricted Motion Vectors Mode:** Restriction of the vectors will keep the pixels within the picture, while some edge pixels are required to predict non-existing pixels. Therefore unrestricted mode will be useful in case of motion around the edge, best result of this mode will be found with small pictures.
- **Syntax-based Arithmetic Coding Mode:** Instead of variable length coding**,** arithmetic coding is used in this mode to generate less bit and keep the same reconstructed frames and Signal-to-noise ratio (SNR) [12].
- **Advanced Prediction Mode:** This mode has the most impact on the resolution and considerable imprudent in which it reduces the blocking artifacts. This mode uses Overlapped Block Motion Compensation to luminance the elements of the picture.
- **PB-frames Mode:** The primary advantage of PB-frames is to increase frame rate without raising the bit-rate significantly. The PB-frame technique is to combine two pictures in one unit. This frame has two pictures, the P-picture is a predicted picture from the previous decoded P-picture, and the B-picture which is a predicted picture from both decoded and currently decoding P-pictures.

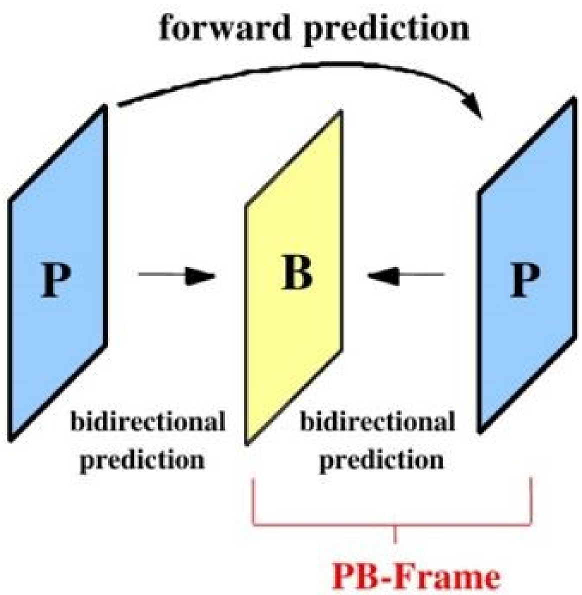

Fig. 3.1 H.263 PB-Frame Mode [13].

H.263 provided remarkable improvement in video coding, in 1998 the second version (H.263v2) or as known as (H.263+) has been introduced. Later on the newer enhanced version of the same standard introduced in 2000, which was H.263v3 or as known as (H.263++).

3.1.2 H.264 / AVC

H.264 is the following ITU-T standard of H.263, which was introduced in 2003. This video coding standard was designed to meet video streaming over wireless and mobile networks requirements. This coding standard has two layers that describe the functionality and the state of the art of this standard which is explained in the following subsections [14]. Also, it has different characteristics, features and tools that made this video coding standard quite popular and successful innovation.

3.1.2.1 Network Abstraction Layer

NAL layer is designed to provide network friendliness for facilitation and efficient customization of video coding layer for variety of platforms and different systems. The ability to map video content data to transport layer is one of the main reasons behind the design of NAL. This layer enables the RTP/IP transportation for any real-time streaming and video conferencing over wired and wireless networks [15]. It also supports different file formats and different video conferencing codecs. NAL has several key concepts which are described below:

- **NAL Units:** This unit holds the video codec data that divided into packets. Each packets contains integer number of bytes. The first byte represents the header in which indicates the type of data held in the NAL unit, and the remaining bytes are the payload data. As a support for transport systems, NAL unit has a generic format to be used for both bitstream-oriented and packet-oriented [16].
- **NAL Units in Byte-Stream Format Use:** NAL uses this format when some systems use a different codec that requires the whole NAL unit or a partial NAL unit stream to be delivered in order. The NAL unit boundaries should be recognized and identified from the coded data. For these type of systems, H.264 uses byte stream format, in which every NAL unit has a specified prefix of three bytes that called start code prefix. This technique will guarantee emulation prevention byte, because, with the existence of these three bytes at the start of the new NAL unit, it will uniquely identify the NAL unit. Additional bytes maybe added to allow decoders recover important alignment information. Moreover, for the expansion purposes, additional data can rapid the recovery process of byte alignment.
- **NAL Units in Packet-Transport System Use:** In the RTP/IP system, the coded data will be delivered in packets and it will be encapsulated into frames in which it has all the required identifications. In this case, there is no need for start code prefix and the NAL unit will be delivered in packets.
- **VCL and Non-VCL NAL Units:** NAL unit is divided into two unit types, VCL, and non-VCL. VCL NAL units represent those units that contain video pictures value data. Non-

VCL NAL units represent additional data that are not related to the video pictures value and not important for the video decoding process.

- **Parameter Sets:** the set that contains information about the decoding process of a large number of VCL NAL units. It has two types of parameter sets, sequence parameter set also called video coded sequence in which applied to sets of sequentially coded video pictures. The second type is picture parameter sets which apply to the decoding of specific pictures in a coded video sequence. Both types help to reduce the frequent changes in the information during the decoding process. Every VCL NAL unit has an identifier that refers to the picture parameter set and each picture parameter set has an identifier that contains information related to the sequence parameter set. This sequencing technique helps to avoid the repetition of the information in every VCL NAL units. These parameter sets can be transmitted before the actual data in the VCL NAL units to prevent data loss. In some cases, parameter set may be sent in the same channel with VCL NAL units, in other cases parameter set may require a dedicated channel that is more reliable than the video channel.

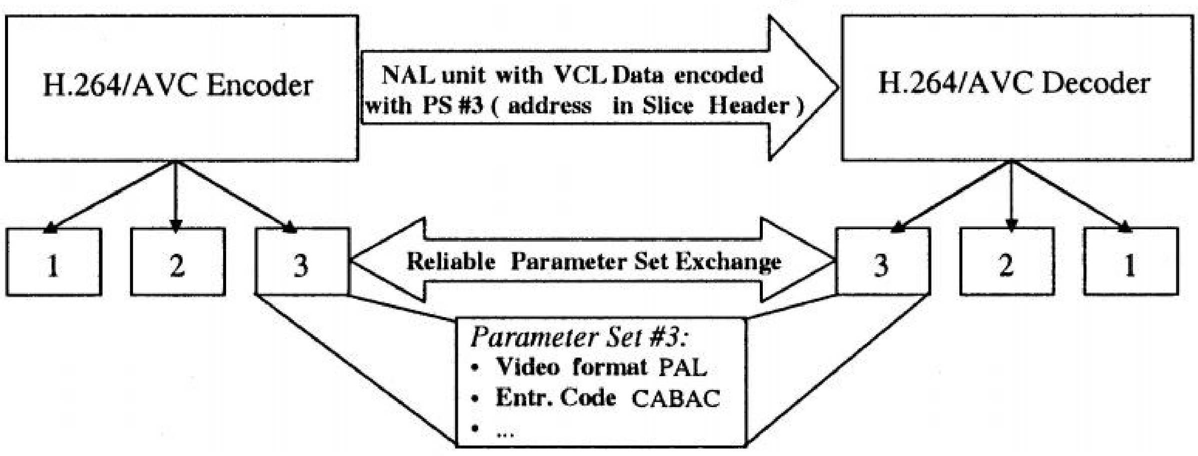

Fig. 3.2. Parameter Set Exchange [16].

- **Access Units:** is a set of specified NAL units, decoding of each access unit will produce a single coded picture. In some cases, the access unit is prefixed with a delimiter to locate the start of the access unit. Some other information may be included with the primary coded picture such as timing information. The following VCL NAL units may be the redundant information of the same coded picture to help the decoder recover lost or corrupted data. Moreover, if the decoded picture is the last picture in the whole NAL unit stream, NAL unit will send end of the stream indicator.

- **Coded Video Sequences:** is a series of access units that are consecutive in the NAL unit streams using only one sequence parameter set. Decoding a coded video sequence can be individual without referring to any other video sequence in the presence of accurate parameter set information. Each coded video sequence starts with an Instantaneous Decoding Refresh (IDR) access unit which contains a picture that can be decoded without interfering with the decoding process of any other pictures in the NAL unit stream. With IDR there is no need for the sequential picture in the stream.

3.1.2.2 Video Coding Layer

Starting from H.261 coded pictures presented in block-shaped units. Same representation design has been used in H.264 and this technique called block-based video coding. The block units are associated with smaller units that called macroblock which contains both luma and chroma. The structure of VCL is formed in a way that every single element has an impact on the coding improvements, and there is not any major element that helps to improve coding efficiently. Therefore, the improvements that can be made in this layer are divided into smaller improvements.

- **Frames and Fields:** coded picture could be represented in either frame or field. Each frame contains two fields, top and bottom. Type of the frame varies depending on the capturing time. When the fields are captured at a different time, the frame will be interlaced. If they were captured at the same time, the frame will be progressive. Therefore, improvements are required by the video coding based on the geometrics not just on the capturing time.

- **Color Space and Sampling:** The visual system of humans is sensitive to brightness more than colors. Designers of H.264 took advantage of this and used YCbCr color space technique to increase brightness and reduce sampling. The Y represents brightness, and it also called Luma. Both Cb and Cr are representing color deviation extent, and both referred to as Chroma.
- **Macroblocks:** Almost all ITU-T video coding standard has adopted this partitioning technique since the design of H.261. The idea of macroblock is to divide the coded picture into fixed-size macroblocks. Each macroblock contains 16x16 rectangular samples of Luma and 8x8 samples of both Cr and Cb Chroma.
- **Slices:** a coded picture is a collection of one slice or slices. Each slice has its information such as sequence and picture parameter set. During the decoding process, there is no need for information from other slices. But maybe some information will be required during the deblocking process.
- **Macroblocks Coding:** Each macroblock samples will be predicted, and the result of the prediction will be encoded using transform coding technique which is dividing each block into smaller 4x4 blocks. This subdivision process will break the video signal into macroblocks and associate them with slices. It is possible to process macroblocks simultaneously if there are multiple different slices in the picture.
- **Adaptive Frame coding:** As described above there are two type of frames interlaced and progressive. It is recommended that the two frame fields should be encoded separately to provide more efficiency. H.264 has a feature which enables the encoders to have three option to choose during the frame coding process. The first option is to combine the two fields and encode them together. The second option is to encode them separately, and the third is to compress the two fields together in a single frame and decompress them into the two fields before the decoding process.

- **Intra-Frame Prediction:** There are three intra coding types that supported in H.264 which are Intra_4x4, Intra_16x16 and I_PCM predictions**.** The first two are for Luma. For coding detailed picture is recommended to use 4x4. While the 16x16 is more suitable for coding pictures that have very smooth areas. Chroma has I_PCM prediction which enables encoders to easily perform the coding prediction and transformation process. It also enables the encoder to accurately present the samples value.
- **Inter-Frame Prediction:** As described in H.263, PB-frame is one of the improvement options, H.264 also has P and B, but called slices. Each P merges to a particular partition of the macroblock into the block shapes that will be utilized for prediction. On the other hand, B slices are coded in a way that macroblocks use a weighted proportion of two separated motion-compensated prediction values to generate the prediction signal.
- **Transform, Scaling, and Quantization:** the transformation process in H.264 is quite similar to previous video coding standards. The transformation technique is to use small size block such as 4x4, because it improves both intra and inter prediction process. Moreover, using 4x4 blocks will reduce the noise around the edges and eliminate statistical correlation.
- **Entropy Coding:** H.264 supports two entropy coding methods single infinite-extent codeword and Context-Adaptive Variable Length Coding (CAVLC). Single codeword method will use only one table for all elements and map them into a single codeword table. In CAVLC scheme different VLC will be switched to match syntax element which is the way of improvement compared to the first method.
- **In-Loop Deblocking Filter:** H.264 has filtering characteristic that helps to improve the quality of the video. The idea of the filtering process is that when neighbor blocks have vast different in their samples around the edge it is blocking artifacts and need to be

declined. This reduction will not affect the video content and it will significantly improve the video quality.

- **Hypothetical Reference Decoder:** For real-time video compression it is very important to specify bitrates, input and output buffers. This can be achieved by a model called Hypothetical Reference Decoder (HRD) which specifies the bitrate at the decoder and will not allow the encoder to generate bit streams that are not supported by the decoder. H.264's HRD has two buffer operations, one at the encoder which is Coded Picture Buffer (CPB) and the other one at the decoder which is Decoded Picture Buffer (DPB). Compared to the previous standards this feature enables the encoder to send video at different bitrates without significant delay. In addition, H.264 can use multiple frames as reference in past or future and HDR buffer management will handle memory usage at the decoder.

3.1.3 H.265 / HEVC

High Efficiency Video Coding (HEVC) is the following standard of the famous H.264 video coding that introduced in 2013. This standard was designed with two main goal in mind: improve video resolution and use more parallel processing. HEVC supports all H.264 applications and has a generic syntax that should be suited for non-H.264 applications. The standardization of HEVC includes only syntax and bitstream structure [17].

3.1.3.1 HEVC Characteristic and Features

HEVC is designed to achieve several goals such as efficient coding, flexible transportation, better video quality and use of parallel processing architecture. Below subsections summarily represent the main elements of HEVC design.

A. Video Coding Layer

HEVC uses the same hybrid coding approach that has been used since H.261. HEVC's encoded videos are expected to be scanned as progressive imagery. HEVC does not support interlaced scanning, thus it has metadata syntax that enables the encoder to detect interlaced-scanned video and inform the decoder when it sends interlaced video. The following are hybrid video coding features in HEVC standard.

- **Coding tree units and coding tree block (CTB) structure:** HEVC has coding tree unit as a core of the coding layer instead of macroblocks that have been used in H.264 standard in which the size of the units are selected by the encoder, and it is usually larger than macroblocks. CTU includes both Luma and Chroma, the size of Luma can be 16, 32 or 64 samples. HEVC supports smaller CTBs in which CTB can be partitioned into a smaller block.
- **Coding units (CUs) and coding blocks (CBs):** CU consists of one Luma CB and usually two Chroma CBs, collectively with associated syntax. Each CU has a transform unit (TU) and prediction unit (PU). CTB may contain one or more CUs.
- **Prediction units and prediction blocks (PBs):** CU decides on the use of interpicture or intrapicture prediction to code a picture. The size of Luma and Chroma can be predicted based on the prediction-type decision. HEVC supports prediction block sizes from 4x4 to 64x64.
- **TUs and transform blocks:** During the transformation process Luma CBs and Chroma CBs can break into smaller transform blocks (TBs). TBs will be generated in square sizes 4x4, 8x8, 16x16 and 32x32.
- **Motion vector signaling:** HEVC uses Advanced Motion Vector Prediction (AMVP), which derive various feasible candidates based on the data from neighboring PBs and picture references. Motion vectors can be inherited from neighboring blocks using merge mode. These features will enables HEVC to specify direct motion inference which is an improvement compared to H.264.
- **Motion compensation:** This feature is quite similar to H.264, HEVC uses quarter –samples for MVs and it also uses multiple reference pictures. The prediction

process of PBs is either unipredictive or bipredictive depending on the number of the transmitted MVs.

- **Intrapicture prediction:** When interpicture prediction is not taking place, decoded edge samples of neighboring blocks will be used as reference data for the spatial prediction process. HEVC supports 33 intrapicture directional modes and these modes will be selected based on the decoded neighboring PBs.
- **Quantization control:** Similar to H.264, HEVC is also using Uniform Reconstruction Quantization (URQ) and this quantization feature enables HEVC to support different transform block sizes.
- **Entropy coding:** HEVC is also using Context Adaptive Binary Arithmetic Coding (CABAC), similar to H.264's CABAC but with improvement in throughput, compression, memory usage and using parallel processing architecture.
- **In-loop deblocking filtering:** this filtering characteristic of HEVC is quite similar to H.264 in which it operates inside the interpicture prediction loop but with improvement in filtering process, decision-making, parallel processing and more simplified design.
- **Sample adaptive offset (SAO)**: after the deblocking filter process inside the loop, a nonlinear amplitude mapping will be introduced which uses look-up table to reconstruct the original signal.

B. High-Level Syntax Architecture

The design of HEVC includes several new features that provide flexibility for various applications and improve data transportation, below are brief explanation of HEVC syntax architecture features.

- **Parameter set structure:** in HEVC despite the sequence and picture parameter sets, Video Parameter Set (VPS), is included in the structure as a new feature.

Parameter set structure provides reliable mechanism to transport data that are vital for the video decoding process.

- **NAL unit syntax structure**: this is quite similar to the H.264 NAL unit, the NAL unit header includes two byte that presents the purpose of the payload data.
- **Slices:** A slice can be a region or the entire picture, and it can be decoded separately from other slices of the same picture. The primary goal behind the use of slices is resynchronization when data loss occurs. The numbers of payload bits are limited in the slices, and the number of CTUs will differ from a slice to another to minimize the overhead during the transmission process.
- **SEI and VUI metadata:** HEVC syntax supports many different metadata types which known as Supplemental Enhancement Information (SEI) and Video Usability Information (VUI). The data carries video timing, color space interpretation, 3-D stereoscopic frame packing and so forth.

C. Parallel Decoding Syntax and Modified Slice Structuring

HEVC has four new feature that designed to improve and modify slice structure in term of packetization and enhance the standard's parallel processing capabilities to provide better performance. Both the encoder and the decoder can decide whether they use these features or not, each feature has benefits for a specific application.

- **Tiles:** the division of a picture into rectangular regions is called tiles, this technique will enhance parallel processing and the tiles can be decoded as separate regions. A tile may include an equal number of regions and CTUs.
- **Wavefront parallel processing (WPP):** When this feature is enabled, the slices will break into row of CTUs. Each row can process only two CTUs, the CTU will be processed in the second row after the process of the first row. In some cases WPP slice technique provides better video compression.

- **Dependent slice segments:** this is a structure that allows data of a tile or wavefront to be carried in a separate NAL unit which will be beneficial for the packetization process. The primary advantage of using dependent slice segment is to reduce latency.

3.1.4 VP8

VP8 is an open source video codec that developed by a research group at On2 Technologies and owned by Google. VP8 was released in 2010, it was extended from VP7, and it belongs to VPx family, then it extended to VP9. VP8 is well known for its less complexity and high compression efficiency. Since VP8 release, researchers were investigating on how its new features will affect video over wireless transmission. Therefore, a technical overview of VP8 is necessary to provide a background knowledge on this video codec and highlight new technical features [18].

3.1.4.1 VP8 Technical Features

VP8 was designed with web-based applications in mind, which indicates that VP8 requires low bandwidth because it will operate in a limited bandwidth environment. Moreover, VP8 designed in such a way that supports verity of hardware devices that use web-based video content application and efficiently implement the video compression in both low and high power devices. Below are the technical features of VP8.

- **Hybrid transform with adaptive quantization:** Similar to the other video compression techniques, VP8 is also using transform coding in which the video image is divided into macroblock of 16x16 Luma and 4x4 Chroma then as a new feature VP8 divides both of them into 4x4 during the transform and quantization process. In the quantization process, two transformation techniques will be applied to Luma and Chroma. First, Discrete Cosine Transform (DCT) which converts the signal into transform coefficient. Second, Walsh-Hadamard Transform (WHT) which is a secondary order transform that used to further reduce redundant DCT blocks.

- **Reference frames:** VP8 is using reference frames in a different way than other video codecs for inter prediction. VP8 employs three types of references to perform inter prediction on the video image frames and use much less memory during encoding and decoding process, the reference frames are last frame, golden frame and alternate frame.
- **Intra prediction:** for a single video frame VP8 uses intra predictions, which has three block types 4x4 and 16 Luma and 8x8 Chroma. In addition to the prediction modes that have been used in other video codecs, VP8 uses true motion prediction (TM_PRED), which is a new feature that used by most of the coded blocks. This new intra prediction mode helps VP8 to achieve high compression efficiency.
- **Inter prediction:** VP8 employs previously encoded frames for inter prediction. It allows the reusability of neighboring macroblock vectors to provide better motion vector coding. As prediction mode VP8 uses SPLITMV which was designed to partition macroblocks into sub-blocks in a flexible way, in which every sub-block has a motion vector.
- **Adaptive loop filtering:** loop filtering process will take place after the quantization process to remove blocking artifacts. VP8 provides new filtering technique that brings high efficiency to the decoding process.
- **Entropy coding:** most of the VP8 header data values are coded by Boolean arithmetic coder which helps the estimation process. VP8 uses Huffman Tree to binarize Boolean values, in which a single symbol maybe represented by a binary string.
- **Parallel processing:** VP8 support parallel processing by dividing the compressed data into two classes, first coding modes and motion vectors, second the quantized transform. VP8 supports up to eight cores which makes the decoder much more efficient.

3.2 IEEE 802.11n

In 2009, IEEE released 802.11n standard as a further amendment to the 802.11 standards. Compared to previous standards, 802.11n provided many improvements and new features. Improvements such as longer communication range, reliability and higher throughput. The primary goal of this standard was to provide high throughput up to 300 Mbps at MAC layer; these improvements were achieved by redesigning the PHY layer and enhancing the MAC layer [19]. 802.11n operates in both 2.4GHz and 5GHz bands by occupying 20MHz and 40MHz channels. It also supports MIMO technology that can be used concurrently with Orthogonal frequency-division multiplexing (OFDM) to reach the data rate of 600Mbps at the PHY layer by doubling the channel bandwidth [20].

3.2.1 PHY

The main improvement in 802.11n PHY layer is the use of multi-antenna at the transmitter and the receiver. 802.11n uses MIMO and OFDM together to boost the channel capacity using 40MHz spectrum. MIMO relies on spatial multiplexing and antenna diversity that allows both transmit and receive from multiple spatial channels at the same time [21]. Spatial diversity is transmitting a single stream of data over multiple transmit antennas to enhance signal diversity. Spatial multiplexing is transmitting independent, and separate data streams over multiple transmit antennas to improve performance, 802.11n supports up to 4 spatial streams [22]. In other words, MIMO creates a multipath transmission environment that improves throughput and reduces bit error rates.

3.2.2 MAC

The main enhancement in 802.11n MAC layer is the frame aggregation, in which multiple frames are transmitted in one large aggregated frame. This transmission mechanism will reduce transmitting overhead time and backoff period. The aggregation process can be classified into two forms of frame aggregation: Aggregated Mac Service Data Unit (A-MSDU), in which multiple MSDUs are transmitted to the same destination in a single integrated MPDU. Every MSDU subframe in the A-MSDU frame has subframe header, payload data and padding field. The header includes source and destination address and the length of the payload data. This form of aggregation is only tolerable for aggregated packets with the same source and destination address. A physical header will be added to the A-MSDU frame before the transmission process which includes FCS field and MAC header. The second form of frame aggregation is Aggregated Mac Protocol Data Unit (A-MPDU), in which multiple MPDU subframes will be transmitted with a unique PHY header to reduce PHY header overhead. Similar to A-MSDU aggregation, multiple MPDU subframes are integrated into a single A-MPDU frame. Every MPDU subframe within the A-MPDU should have the same destination address, but it is possible to have different source addresses. Figure 3.3 is a notion of the two frame aggregation forms.

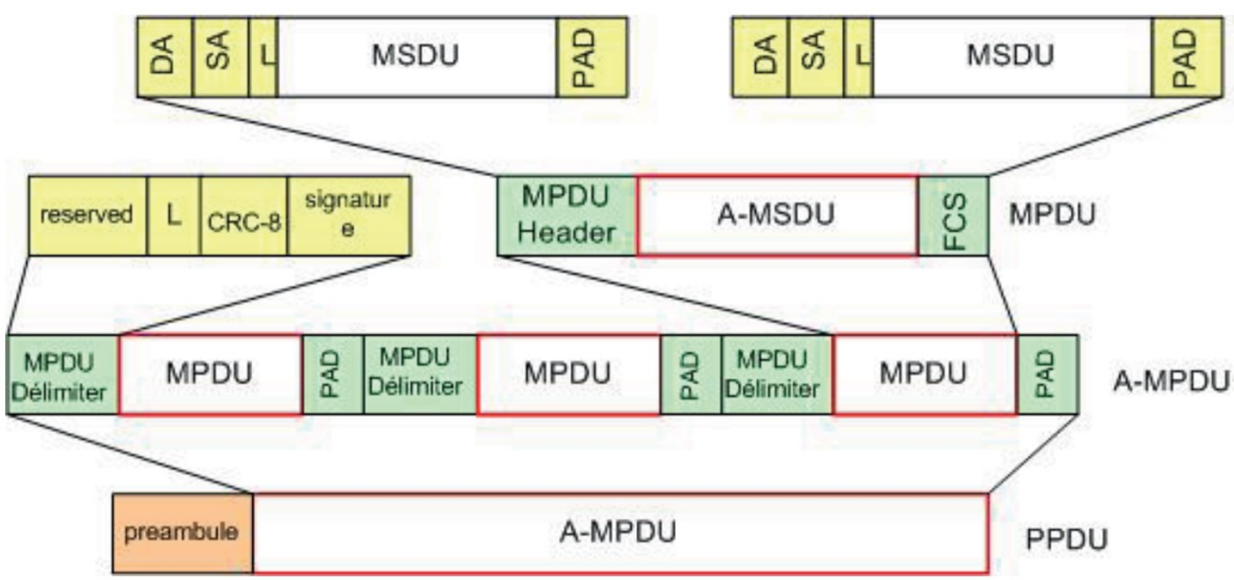

Fig. 3.3. 802.11n Frame Aggregation Forms [21].

In addition, Block Acknowledgment (BA) is another improvement scheme in 802.11n. Block acknowledgment was first introduced in 802.11e as an optional improvement scheme, later on, the support for block acknowledgment became mandatory in 802.11n. The idea behind BA is that the transmission of multiple MPDUs will be acknowledged using a single BA [21].

3.3 IEEE 802.11ac

802.11ac is the latest approved standard by IEEE and it is designed to be Gigabit WLAN. 802.11ac supports multi-user (MU-MIMO) which boost the throughput up to 1 Gbps at the MAC layer [23]. 802.11ac operates only on 5GHz band, and it is backward compatible with previous standards especially 802.11n. 802.11ac provides several improvements in both PHY and MAC layers such as increasing channel bandwidth, increasing the number of spatial streams and optionally supporting 256 QAM and MU-MIMO [24]. Below are the 802.11ac PHY and MAC layer features and improvements.

3.3.1 PHY

802.11ac is an evolution of 802.11n, and the design includes similarity in the use of MIMO and channels. 802.11ac supports 20MHz, 40MHz and 80MHz channels with the optional of using contiguous 160MHz. To modulate bits for transmission over WLAN, 802.11ac uses OFDM, and the modulation technique is similar to 802.11n which are 16 QAM and 64 QAM. In addition, 802.11ac optionally supports 256 QAM, which increases the PHY rate by 33% under perfect circumstances. Unequal Modulation (UEQM) is not supported in 802.11ac. Moreover, 802.11ac extends 802.11n special streams and supports 8x8 MIMO antennas. It also supports downlink MU-MIMO, which allows a single AP to transmit multiple frames to multiple destinations simultaneously using multiple spatial streams. Figure 3.4 shows single-user and multi-user MIMO beamforming.

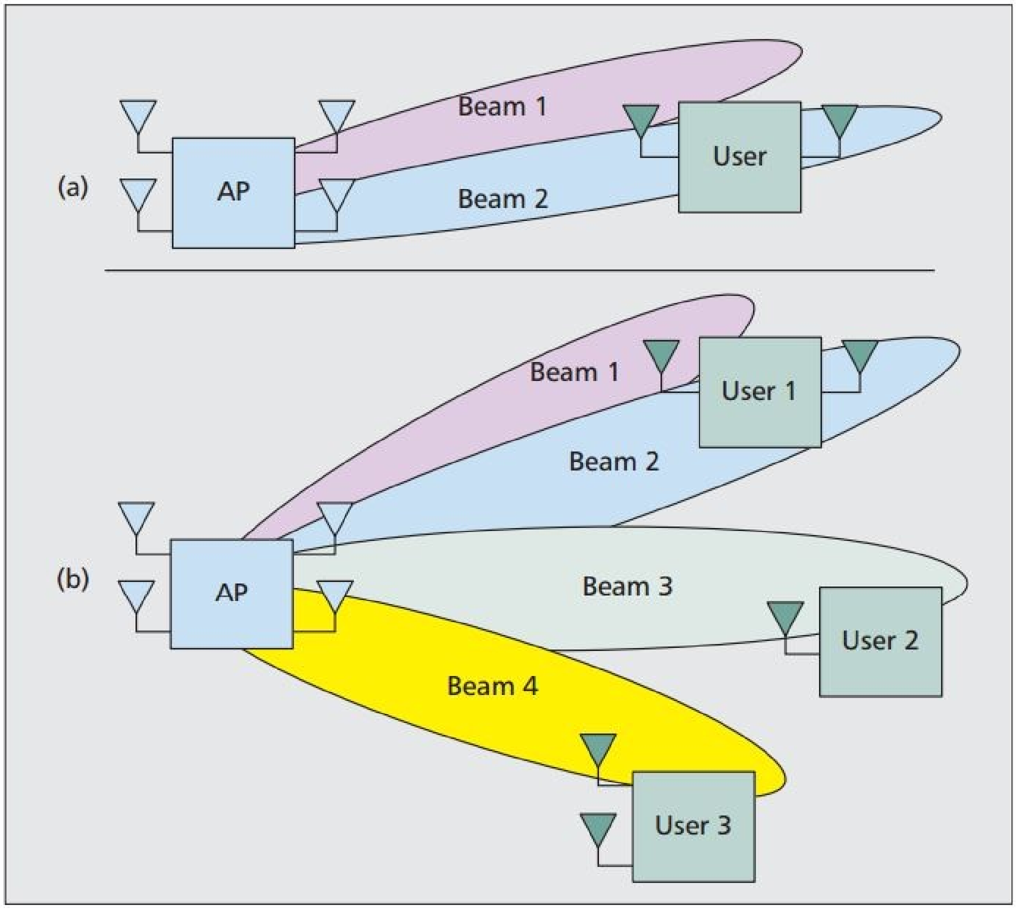

Fig. 3.4. SU-MIMO and MU-MIMO beamforming [25].

3.3.2 MAC

802.11ac provides new features and enhancements at MAC layer. Due to wider bandwidth and limited 80MHz channels, hidden nodes on non-primary channels will cause problems. 802.11ac upgraded Request to Send and Clear to Send (RTS/CTS) mechanism to detect different transmission process over secondary channels more efficiently [24]. Furthermore, 802.11ac provides improved aggregation scheme in which A-MSDU and A-MPDU frame size are expanded [21]. Transmission Opportunity (TXOP) sharing is a technique used by 802.11ac to allow multiple downlink streams to be transmitted to multiple destinations at the same time. The idea behind TXOP sharing is the group ID that used by APs when they transmit to multiple stations. Each AP has primary and secondary destinations. In addition, 802.11ac adds a new frame field that called

Very High Throughput (VHT) which includes two OFDM symbols: Binary phase-shift keying (BPSK) modulated symbols in order to allow 802.11n users. The second symbol is 90-degree rotated BPSK to enable VHT for 802.11ac users. Table.1 presents a comparison between 802.11n 802.11ac standards.

Table 3.1. A comparison between 802.11n and 802.11ac [21] [25].

	802.11n	**802.11ac**
Channel Width	20/40 MHz	20/40/80/160 MHz
Data Rate	600 Mbps	1.3 Gbps
Frequency band	2.4 GHz and 5 GHz	5 GHz
Maximum number of spatial streams	4	8
Modulation	BPSK,QPSK, 16/ 64 QAM	BPSK,QPSK, 16/ 64 256 QAM
Antenna technology	MIMO	MIMO/MU-MIMO
A-MSDU length (bytes)	7935	11426
A-MPDU length (bytes)	65535	1048579

4. QOS/QOE SUPPORT IN IEEE 802.11N/AC

4.1. QoS Overview

IP is a best-effort service which means it will try to transmit the packet to the right destination but with no guarantee of delivery. Due to widely deployment and multimedia support in IEEE802.11 standards providing QoS has become a necessity. Supporting multimedia and bandwidth hungry applications requires QoS provisioning to provide resource assurance and service differentiation.

4.1.1. Resource Allocation

A networked system has many different resources such as bandwidth, buffer size, routes and packet size. These resources need to be managed and allocated. Therefore, resources allocation is one of the major QoS issues. QoS supposed to allocate all these resources properly. Otherwise the network performance and service quality will face severe challenges due to packet loss and congestion. QoS has two main resource allocation approaches Integrated Service (IntServ) and Differentiated Service (DiffServ).

4.1.1.1. IntServ

IntServ reserve resources based on packet stream flow (per-flow), in which the packet stream has sources and destination address along with the port number. IntServ uses packet scheduler to allocate resources to individual packet flows along with prioritization support. The scheduler is used to obtain two type of delay bounds, statistical and deterministic. Moreover, IntServ has two key abstractions which are reserved resources and standards resources, in which reserved resource means that the router must be aware of the reserved resource by the on-going transmission process and the standard indicates the router must be aware of the link and buffer capacity.

4.1.1.1. DiffServ

DiffServ is a per-class resource reservation that classifies the traffic; it uses prioritization and able to forward traffic to multiple classes. DiffServ uses Per-hop packet-handling rule in which every hop along the path performs according to the priority values of a specific packet stream.

4.1.2. Service Differentiation

Service differentiation technique provides support for various services with different traffic requirements. For example, delay sensitive services will be distinguished from other traffic flows. But, providing this type of service provisioning is not easy especially in wireless networks. Different applications have different QoS parameter sets and different bandwidth requirements. Applications like email and audio require less bandwidth than video which is a bandwidth hungry application. Most often researchers use bandwidth, delay, jitter and loss rate as QoS metric. Providing QoS for video requires all four metric to meet their demand due to the nature of video in which any delay in live streaming will cause inconvenience. Jitter is the order of packet arrival time, if the packet arrives in a different time than it supposed to be, the video frames will be shown in a different sequence.

4.1.3. Admission control

Admission control is the technique of controlling new session allowance in which the new session will be allowed only if there is enough bandwidth to support it. The new session request will be permitted only if it has no negative effects on the ongoing traffic. However, admission control is a network layer QoS technique that is still a challenge to provide in wireless networks.

4.1.4. Congestion control

This technique is a functional QoS solution only with the use of Transmission Control Protocol (TCP) as transportation mechanism. Congestion occurs when the number of transmitted packets are higher than the number that the network can handle. The congestion control is the process of keeping the packet transmission load below the network capabilities.

4.1.5. Scheduling

This technique will share the network resources properly between the network hosts. The scheduler arranges the requests in order and manages the waiting queue. This will help to achieve service guarantees for time critical applications.

4.1.6. Traffic shaping and engineering

In a wired network architecture, boundary nodes will classify the data flow according to the service requirements in order to achieve average traffic rate and deduct congestion, this technique is called traffic shaping. While traffic engineering is the technique that distributes network resources in a proper way to boost network utilization.

4.2. QoS support in IEEE 802.11n/ac

Since IEEE 802.11 standards target PHY and MAC layer, the QoS support for multimedia application can be provided within the enhancement of those two layers. Due to the remarkable improvement at the PHY layer of both standards which boosted their throughput, PHY layer QoS enhancement is less likely to be considered. In addition, 802.11n and 802.11ac incorporate with admission control and scheduling algorithms.

4.2.1. MAC layer QoS features

Major MAC layer characteristics in WLAN are addressing, framing and frame aggregation along with access coordination and reliability. MAC layer QoS enhancements target overhead reduction, frame segregation based on priority and reducing collision. Several different MAC layer QoS enhancement techniques for wireless standards have been proposed, below are a brief explanation of those techniques.

4.2.1.1. Differentiated Services

Differentiating services for QoS enhancement at 802.11 MAC layer can be classified into two main approaches Distributed Coordination Function (DCF) and Point Coordination Function (PCF). Below are brief introductions to DCF and PCF and their techniques.

4.2.1.1.1. DCF

DCF is a basic 802.11 MAC layer technique that uses backoff algorithm and CSMA/CA to avoid collision and force station to extend their waiting time to access the occupied channel. DCF is a contention based service that can be deployed in both infrastructure and non-infrastructure based networks, due to its Ethernet type of communication DCF is widely deployed [27]. DCF has several deployment techniques which are described below:

- **Distributed Fair Scheduling (DFS):** is the technique of bandwidth assignment in which application with high priority will be assigned with more bandwidth than low priority application. It also uses backoff algorithm which makes stations with low priority traffic to stay longer in the backoff interval until the high priority traffic flows.
- **Varying DIFS:** varying Distributed Interframe Spacing (DIFS) is one of the techniques that used to differentiate among data flows. When data frames have different DIFS duration, backoff time will be maintained, and this will lead to collision avoidance. This technique helps in real-time applications in which delay will be greater than packet loss.
- **Differentiated Maximum Frame Length:** this is where the stations can transmit frames with different sizes. High priority stations can send frames greater than low priority stations. If the packet size exceeded the maximum limit, the packet will be fragmented and the fragmented packet will be sent with no RTS, the transmitter will expect ACK from the receiver.
- **Blackburst:** This technique allows every station to occupy the channel for a specific period of time. Based on their priority, stations can access the channel and transmit their frames during the time interval. As soon as the station occupies the channel, it will jams the medium for a period of

time. The station will send jamming signal which is so called blackburst. Each station has to wait for an amount of time, when the channel returns to idle, station with the longer waiting time will access the channel. Blackburst is quite similar to Time Division Multiplexing (TDM), which is used for real-time traffic and sharing the medium.

4.2.1.1.2 PCF

PCF is also an 802.11 MAC layer technique that coordinates the connection in such a way that the station waits for PIFS rather than DIFS to access the channel. In PCF accessing the channel is centralized and it's a contention-free service that restricted to infrastructure networks. Below are the two main PCF techniques to differentiate services:

- **Distributed TDM:** as a PCF technique, it employs polling method and defines time slots same as TDM then assign them to stations. When the station has a time slot assigned, it knows when to transmit frames. This can be done with a little effort from the AP.
- **Hybrid Coordination Function (HCF):** this coordination technique was first introduced in IEEE 802.11e standard in order to enhance DCF and PCF. HCF employs Enhanced Distributed Channel Access (EDCA) as a contention-based method and HCF-Controlled Channel Access (HCCA) as a contention-free method. In HCF environment APs act as traffic manager which known as Hybrid Coordinator (HC) and it is a centralized coordination mechanism. HC works with both contention-based and contention free and negotiates the transmission and handling process of the frames.

4.2.1.2. Priority queueing

Segregating data packets based on their priority is called priority queue, in which the transmitter will first send the packet with the highest priority. There are eight level of priority at MAC layer and this means that there are eight queues as well. Based on their status, applications will be assigned to priority levels.

4.2.1.3. Scheduling

The main goals of scheduling are obtaining better throughput and transmission time reduction along with delay which are QoS key metrics. IEEE 802.11 selects packets from different traffic and based on their requirements it distributes them over particular links. The idea behind scheduling is to queue packets based on their priority and select packets from high priority queues to be transmitted first. Scheduling has several algorithms to ensure QoS in wireless networks, namely: (i) weighted fair queueing (WFQ) in which a specific weight will be assigned to each queue, this scheme works properly with wireless networks. (ii) Weighted Round Robin (WRR) which is a frame-based version of WFQ. (iii) Strict priority in which the buffer will be divided into queues along with priority flow queues. (iv) Earliest Due Date (EDD) in this scheme deadlines will be assigned to data flows, it is designed for wired networks and implementing EDD in wireless networks will be a challenge.

4.2.1.4. Traffic shaping

Traffic shaping is the technique of limiting the number of packet per station, in which a traffic controller used to satisfy QoS requirements. This will control the data flows over the channels and enables the traffic shaper to distinguish the resources based on their requirements. Aggregation level and bursting level are among those traffic shaping parameter that have been used to provide QoS for 802.11 standards [26].

4.3 QoE Overview

As mentioned in the sections above bandwidth, delay, jitter and packet loss are quality of service metrics in which they are technical parameters to measure service quality. These parameters work best for application such as email, but for real-time application such as video streaming it is not possible to measure the user experience only with these parameters. User satisfaction is important for the firm and network administrators to maintain the network resources in such a way that satisfy their users. Quality of Experience (QoE), is the overall acceptability of a service from the user perspectives. Primarily it is the end-to-end performance measurement of a service from the user point of view [28].

4.3.1 QoE Measurement

To measure QoE for video over wireless networks, there are three different QoE assessment approaches which are subjective, objective and hybrid as described below:

4.3.1.1. Subjective Assessment

This approach is the most accurate one because it is based on the human perspective of the video quality. To apply this approach for QoE measurement, a panel need to be created to evaluate the video sequences as it is from the human point of view. The output of this assessment will be presented in mean opinion score which is one of the QoE metrics. Even though this approach is the most accurate to measure QoE because of the real human evaluation, but it needs a lot of time and administrative efforts. Therefore, it is less likely to be implemented in an automated measurement and monitoring system.

4.3.1.2. Objective Assessment

Due to it is expensive implementation, subjective approach is not the main focus of researchers and they are looking for alternative assessment techniques which employ formulas and algorithms. Objective approach uses algorithms to collect data from technical parameter to measure quality of experience. This approach has many different metrics, the simplest and the most common objective assessment metric for video is Peak Signal-to-Noise Ratio (PSNR), which described in section 4.3.2.

4.3.1.3. Hybrid Assessment

To provide more accurate QoE measurement, researchers created a hybrid assessment that combines subjective evaluation and objective parameters, and this assessment is called Pseudo Subjective Quality Assessment (PSQA). It uses Random Neural Network (RNN) statistics to work with real-time traffic [28].

4.3.2 QoE Metrics

Many different subjective and objective metrics have been proposed for QoE assessment. Metrics such as MOE and PSNR are used for audio and video QoE assessment only, while there are other applications which are also important to fulfill QoE scheme. Therefore, it is better to classify QoE metrics into two types: direct metrics which are directly related to video QoE measurements and indirect metrics which refers to factors that have no direct impact on the video such as service usability, interactivity and response time.

4.3.2.1. Direct Metrics

Direct metrics are those with direct impact on video perception and experience. Usually, it considers factors such delay, loss and codec information. Various direct metric have been used to measure QoE, below are the most relevant metrics that have been used for video QoE assessment.

- **Mean Opinion Score (MOS):** This metric was originally developed for audio QoS measurement which combines delay, jitter, and packet loss and codec information at the application layer. For video, MOS retrieves information from other metric to draw user viewpoint. Moreover, researchers proposed a mapping technique between MOS and PSNR to better assess video QoE. Table 2 shows the MOS quality evaluation.

Table 4.1. MOS [28].

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

- **Peak Signal to Noise Ratio (PSNR):** This metric measures the similarity between two different images in which images with high similarity will produce higher PSNR. In other words, PSNR is the ratio between maximum signal power and corrupted noise power. It employs Full Reference as measurement approach and processes Mean Square Error (MSE) of both original and received pixels and typically PSNR is measured in dB values. Table 3 shows the possible conversion of PSNR to MOS.

Table 4.2 PSNR to MOS [28].

PSNR (dB)	MOS
> 37	5 (Excellent)
31-37	4 (Good)
25-31	3 (Fair)
20-25	2 (Poor)
< 20	1 (Bad)

- **Structural Similarity (SSIM):** similar to PSNR, SSIM is also using Full Reference and it compares the correlation between the received and the original image by combining contrast, luminance and structural similarity.
- **Video Quality Metric (VQM):** This metric is also using Full Reference which considers block and color distortions, blurring and global noise to detect appreciable artifacts on the images.

4.3.2.2. Indirect Metrics

Indirect metrics consider those factors that impact the overall experience of multimedia transmission without direct impact on the content quality. These type of metrics are mostly based on the type of the service. The most relevant indirect metrics for video QoE assessment are start-up time which is the time since the user requests till delivery of a service. Response time is one of the indirect metrics which is the time that the system takes from the user action point till the acknowledgment arrives at the user end. Moreover, when multiple users request the same service, the content should be delivered to all users at the same time and this metric is called delivery synchronization which mostly used in video gaming [29].

5. PROPOSED SOLUTIONS

An enormous number of solution have been proposed during the last decade to provide QoS/QoE for video over wireless networks. Among these solutions many of them have been included in the standards or a vender included them in their wireless devices, while some other remained as an academic solution without any implementation. Therefore, proposed solutions can be categorized into two main categories academic solutions and vendor solutions.

5.1. Academic Solutions

Researchers have been enhancing video codecs, MAC protocols and transmission rates to fulfill QoS requirement for video over WLAN. Some of the enhancement require only one layer to operate, while some other need more than one layer to operate. Therefore, academic solutions can be categorized into single-layer solutions and cross-layer solutions.

5.1.1. Single-Layer Solutions

Since video transmission over WLAN is mostly depending on three layers: Application, MAC and PHY, the proposed solutions are based on those three layers. Single-layer solutions are limited to only one layer, therefore the solution is concentrating on developing or enhancing video codec, MAC protocols, frame aggregation scheme or transmission rates. Below are the most relevant solutions for video over WLAN based on those three layers.

5.1.1.1. Application Layer Solutions

Application layer solutions are mostly based on video codecs techniques and enhancements to generate better video streams. Video codec algorithms will have a direct impact on the QoS performance. Therefore, most of the solutions focus on a single codec algorithm. These type of solutions are not restricted to wireless networks it can be applied to wire networks as well. However, researchers are continuously working on video coding development and the most used video codec is H.264 with consideration of the latest standard by ITU-T which is HEVC and Google's VP8 which are described in Chapter 3. This section investigates latest proposed solutions that enhance video streaming quality. These solutions are mostly concentrating on three main factors that have the impact on the video codecs, and those factors are rate distortion, error resilience and joint source and channel coding. Furthermore, most recent studies conducted research on H.264/HEVC transcoders to speed up coding process.

Error resilience is one of the main factors that affects the coding process especially in H.264. Huong and Huu [30], proposed an explicit mapping approach using Flexible Macroblock Ordering (FMO) technique that generated frame-by-frame. The idea behind FMO is enabling the encoder to change the order of frame blocks. The proposed method will reduce error propagation impact between frames and macroblocks in slice groups using intra refresh. The result of the experiment shows that the proposed technique has lower macroblock loss rate and in some cases it has higher PSNR values. Although error resilience in H.264 mostly depends on FMO and coding refreshment, reducing the number of slice groups will provide better video quality [31]. The research has shown that five slices will be the optimal number of slice groups using FMO and this will increase video quality over the WLAN using clear channels. Depending on the channel state the slice size should vary to provide coding efficiency. Error free channels can transmit HD videos in bigger slice sizes and this will increase video quality because the coding process will be faster, but when the slice size is smaller than the line of macroblock coding process will slow down rapidly. The result of the conducted experiment has shown that the optimal number of macroblocks per slice for high complexity video will be 80. While this number can be raised to 1200 macroblock in case of low complexity videos. However, when it comes to PSNR and quality of experience, too large or too small slice will result in low PSNR rates and quality loss. Therefore maintaining the slice size will have a direct impact on error resiliency in H.264.

Rate distortion and frame rate are the two most important aspects of video quality. High Frame Rate (HFR), is a new approach to improve end-to-end distortion and enhance quality of experience [30]. HFR has become more popular and widely deployed in multimedia application due to it is high-quality in which frame rate reaches 60 fps. Transmitting video with high frame rates over wireless networks will pose a challenge due to limited bandwidth, demand on high throughput and delay. Hence, higher frame rates will lead to high traffic load, this means that it is possible for the end-user to experience delay because of the late arrivals of frames which eventually lead to higher end-to-end distortion. To analyze the tradeoff between frame rate and end-to-end distortion, researchers developed a framework called Frame Rate vs. Video Distortion (FRIED). This framework considers video distortion, FEC coding and wireless networks mathematical model. Figure 5.1 show the tradeoff between delay, frame rate and PSNR.

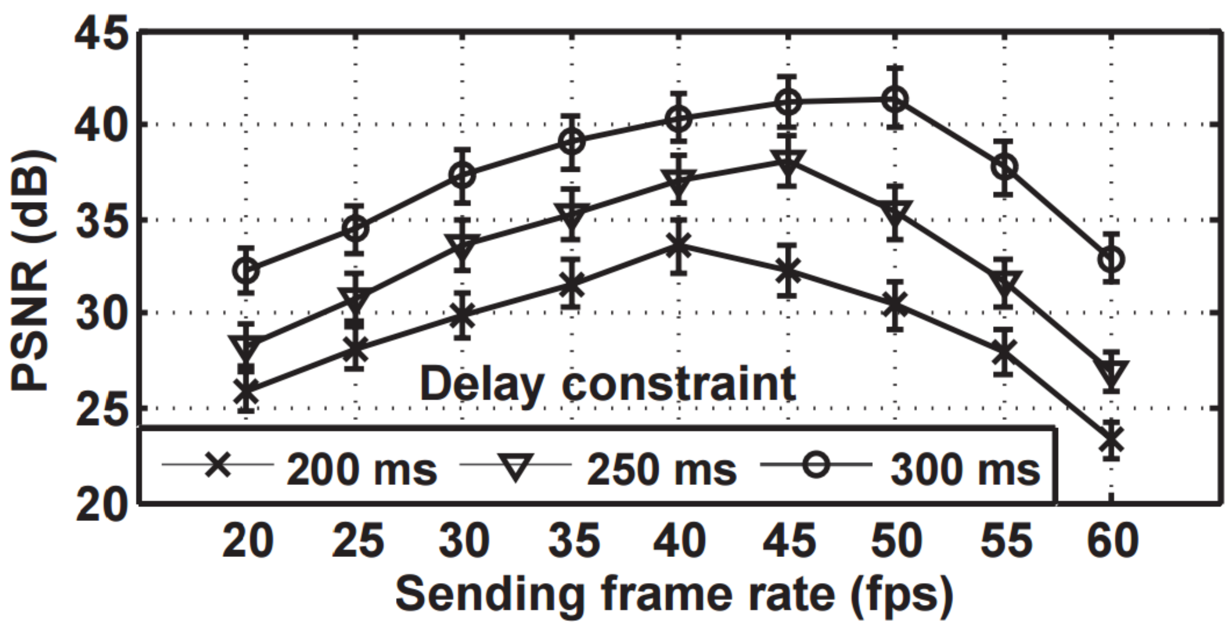

Fig. 5.1. Frame Rate, PSNR and Delay Tradeoff [30].

Different optimization model and coding scheme have been proposed to enhance HFR adaptation for wireless networks. Hybrid Automatic Request (HARQ) is the scheme that lessens the derived delay from the lost data and Joint Selection and Scheduling (JSS) algorithm which concentrates on delay and maximizes video quality based on frames subset selection without considering channel loss and bandwidth variation. Moreover, adaptive scheme is another approach that transmits all the video frames using Forward Error Correction (FEC) coding without any frame rate adjustments at the sender and filtering at the receiver. However, the most recent proposed model is Joint Frame Selection and FEC (JASCO) Coding, which is a model-driven optimization scheme that provides effective video frame selection algorithm to achieve ideal frame rate and lessen end-to-end distortion. This model also uses FEC coding to adjust packet size and redundancy value in order to meet quality requirement. An experimental study has been conducted to analyze and compare all four schemes mentioned above. The result shows that JASCO provides better PSNR, less end-to-end delay and higher frame rate compared to HARQ, JSS and Adaptive schemes. Figure 5.2 shows the result of the comparison study of the four scheme based on video frame rate to PSNR ratio.

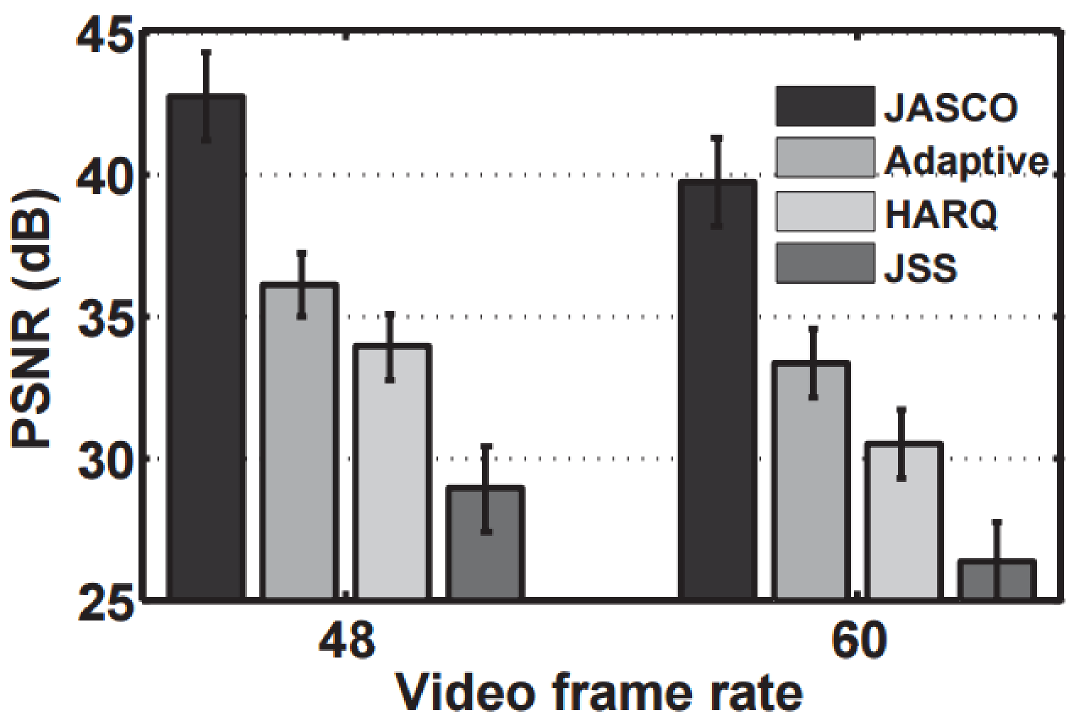

Fig. 5.2 Comparison between JASCO, HARQ, JSS and Adaptive based on Frame Rate to PSNR [30].

Another way to improve rate distortion is by modifying source Lagrangian and measuring channel loss by packet loss visibility. The proposed method is to optimize source rate distortion to obtain a balance between source loss and channel loss. The experiment was to compute SSIM of a 300 frame video against the uncompressed frames then considering packet loss effects on the overall quality. The experimental results have shown that the proposed method provides higher SSIM and better channel loss which reaches its peak 90.63% by using the visibility method and this will enhance the overall quality compared to the original standard [31].

In addition, there are other video codecs that have different feature and characteristic that can affect video transmission over WLAN as described in chapter 3. Yoon et al [32], have evaluated the performance of both H.264 and VP8 over the IEEE 802.11 standard. Both QoS and QoE for video metrics have been used to evaluate the video quality of both codecs. Their experiment involves H.264/AVC, H.264/SVC and VP8 and the measurement include PSNR, MOS, end-to-end delay and packet loss values. The experimental study is based on DCF, EDCA and Traffic Prioritization Algorithm (TPA). The result of the experiment has shown that in case of DCF it is hard to provide QoS with H.264/SVC due to the high fraction of data which makes the packet delivery a challenging process especially for real-time applications over WLAN. In term of packet loss both H.264/AVC and VP8 have shown similar results with no packet drops while H.264/SCV dropped ten packets. Although, H.264/AVC and VP8 have shown similar MOS, but the end-to-end delay of VP8 is higher than H.264/AVC under DCF scheme.

Since HEVC is the latest ITU-T video codec standard that based on the dominant H.264 codec and includes many legacy contents along with enhancement in the codec, many researchers have been proposing transcoding techniques to migrate from H.264 to HEVC. Furthermore, many new devices support the latest standard which means developing efficient transcoder have become a necessity. HEVC doubles the rate distortion performance of H.264, it also uses quadtree-based CU for block partitioning. However, HEVC provides better performance but with higher computational cost and complexity. Many transcoding mechanisms have been proposed to migrate and transcode H.264/HEVC videos with less complexity and faster CU splitting process, transcoders such as FQLD, PTDT-MVVD, PTCM-LDF, RT-FME and so forth [33] [34]. Diaz-Honrubia et al. [35], developed an algorithm for better decision making during the CU splitting process at the H.264 decoder. The analytical result has shown that applying the proposed algorithm will speed up the CU splitting process and avoids rate distortion optimization on every CU sizes. Moreover, the time reduction achieved by implementing this algorithm was 26%-56% depending on Bjontegaard Delta (BD) rates compared to other transcoders. In addition, despite the complexity reduction, this algorithm is suitable for both hardware and software of H.264/HEVC transcoding for real-time video.

5.1.1.2. MAC Layer Solutions

Channel access control, check sequence and frame retransmission are the goals of the IEEE 802.11's MAC layer. As described in section 4, MAC layer has two medium access coordination DCF and PCF. DCF as mandatory access coordination uses Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) for wireless channel access regulations and provides best effort service, while PCF was designed to support QoS for multimedia and delay-sensitive applications. Due to the bad performance of PCF, both DCF and PCF were replaced with EDCA and HCCA. EDCA as a contention-based mechanism to prioritize QoS by enhancing DCF and HCCA provides a contention-free mechanism that manages delay sensitive traffic better that PCF [4]. MAC layer of both IEEE 802.11n/ac has been enhanced and provide many new features that improve QoS/QoE for video. In IEEE 802.11n the enhancements were mostly focusing on frame aggregation and block acknowledgment. While in IEEE 802.11ac the enhancements were concentrating on TXOP sharing, frame format, and aggregation. With all these enhancements in both standards, QoS/QoE still not guaranteed for video traffic. Many MAC layer solutions have been proposed since the invention of both standards, below are the technical overview of the most relevant MAC layer solutions.

Frame aggregation has a direct impact on video streaming over WLAN [36]. Video data packets are small, therefore encapsulating and aggregating these packets in large frames will enhance throughput at MAC layer. IEEE 802.11n provides frame aggregation without specifying the exact implementation mechanism to enhance QoS for video. Researchers indicate that the size of subframes should be determined according to the channel condition and large frames may cause end-to-end delay for real-time video transmission. Therefore, the size of subframes should be maintained carefully during the aggregation process [37]. The frame aggregation mechanism that has been used in IEEE 802.11n is limited to unicast which means every MPDU in an A-MPDU should be addressed to the same destination. Lee et al [38], proposed a scheme in which A-MPDU carry MPDUs with different destination addresses.

The idea is to encapsulate video frames as MPDUs separately and put them into A-MPDU without any sorting or separate packing. This predestination framing technique will make the encapsulation process less complex and reduces the overhead. The proposed scheme is called Instantaneous Multi-Receiver Aggregation (IMA), that boost video streaming capacity over IEEE 802.11n. The experimental results have shown that single receiver scheme that has been used in 802.11n standard will suffer delay after three simultaneous video streams while the multi-receiver scheme suffers from capacity overrun at 8 video streams. This indicates that proposed scheme doubles the video streaming capacity of the 802.11n standard in which the number is 3 to 7 streams. Figure 5.2, shows the result of the experiment using both schemes in two scenarios 12 Mbps and 18 Mbps.

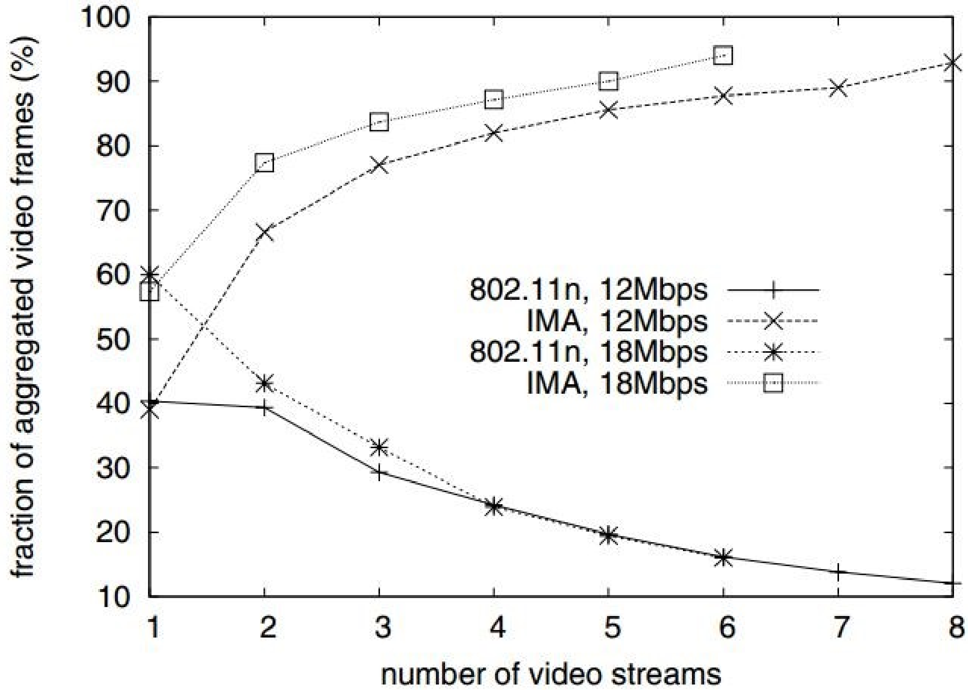

Fig. 5.3. Experimental result of single and multi-receiver frame aggregation schemes [38].

Zhu et al [39], proposed a frame aggregation scheme that based on IEEE 802.11ac A-MSDU. The aggregation mechanism is to divide the frame into several sub-MSDUs which called MAC blocks. Each MAC block contains 448 Bytes service data derived from packets along with interactive and control information. This technique will reduce MAC header and ACK payload along with collision and short inter-frame space (SIFS). The experimental result has shown that implanting this scheme will provide data rate over 1 Gbps at MAC layer. In addition, the QoS mechanism will prioritize the traffic for video provides enough bandwidth for the high-volume data as well. However, there are three types of frame aggregation which were introduced in IEEE 802.11n, and these aggregation types are A-MSDU, A-MPDU and both combined. Using Request-to-Send (RTS) and Clear-to-Send (CTS), with different aggregation types to fulfill QoS requirement has been studied [40]. RTS/CTS is an optional transmission mechanism in both IEEE 802.11n/ac in which the station that wants to transmit first send an RTS frame and waits for the receiver to reply with a CTS frame. Once the correct CTS frame received the sender will start the transmission process. Other stations can detect RTS/CTS frames and delay their transmission to those two station, and this will avoid collisions. The experiment is focusing on the use of RTS/CTS mechanism with both A-MPDU, and both A-MSDU and A-MPDU combined aggregation scheme along with the use of ARQ protocol which is used to enhance MAC layer ability to retransmit lost MSDUs while they are not reached their deadline. ARQ combination with aggregation schemes to enhance QoS over IEEE 802.11ac have been studied in [41]. However, the use of RTS/CTS with aggregation schemes has advantages in multiple states of the transmission. In case of high delay limit and collision in the network, it is better to use RTS/CTS due to their high recovery performance. While in low delay limit scenarios it is better not to use RTS/CTS because it will cause unnecessary overhead.

In addition, an assessment approach has been proposed to measure video QoE-based Random Neural Network (RNN), with consideration of MAC layer parameters such as frame aggregation, Bit Error Rate (BER), traffic load and a number of stations [42]. At first, the study conducted an experiment on how MAC parameters affect QoS parameters. The results have shown that increasing number of stations will increase MSDU drops along with higher jitter and more collision occurrence. Consequently, this will degrade video quality which will result in degrading QoE. However, the proposed QoE-RNN approach has been proven, and it can provide subjective QoE estimation considering MAC layer and QoS parameters.

5.1.1.3. PHY Layer Solutions

IEEE 802.11 PHY layer can provide different transmission rates using different modulation technique and coding schemes. IEEE 802.11n provides transmission rates up to 600 Mbps at PHY layer and IEEE 802.11ac provides up to 1.27 Gbps using MIMO technology. Both MIMO and rate adaptation have different modes that can be chosen by the receiver regarding QoS requirement and channel condition [4]. Most researchers are proposing algorithms to effectively select the MIMO and rate adaptation mode.

Rate adaptation is the process of selecting best data rate at PHY layer based on channel condition. Rate Adaptation and MIMO mode (spatial diversity and spatial multiplexing) selection are the two major factors that have been targeted by researchers to enhance QoS at PHY layer. Deek et al. [43], conducted research on rate adaptation and MIMO modes in IEEE 802.11n. First, they proposed a link metric that called diffSNR to captures channel condition; then they proposed a framework that adapts transmission rate and bandwidth based on the channel condition that retrieved from diffSNR. The primary goal of diffSNR (Differential SNR) is to measure the difference of SNR at the receiver's antennas. The simulation results have shown that diffSNR is 95% accurate in evaluating and predicting the channel condition. Moreover, to adopt diffSNR link metric in IEEE 802.11n rate adaptation, two approaches should be taken into consideration. Closed-loop and open-loop rate adaptation, in open-loop rate adaptation approach the sender will determine the transmission rate based on metrics used by the sender itself. While in closed-loop rate adaptation approach the receiver determines the transmission rate based on channel condition. Since IEEE 802.11n provides (32 rates two channel widths) which will be 64 rates combined and IEEE 802.11ac multiplies this number by 4. Consequently, implementing the open-loop rate adaptation has become less accurate and more complex due to the increasing number of variable rates that the transmitter cannot determine it accurately. Therefore, the contribution of the receiver is required at this stage to determine the rate adaptation accurately. The closed-loop rate adaptation will observe the channel condition from the receiver side.

To accurately detect MIMO performance, a rate adaptation technique is required that should use closed-loop approach along with a feedback system to detect channel condition, in this case IEEE 802.11n supports MCS feedback. The proposed rate adaptation framework is called Agile Rate Adaptation for MIMO Systems (ARAMIS), which is per-packet and closed-loop rate adaptation that have designed to operate in MIMO environment to adopts both transmission rate and channel width at the same time. ARAMIS requires three main components to function properly which are: link metric which is the proposed diffSNR metric, link predictor that used to predict both Packet Reception Rate (PRR) and MSC, and the rate selector which sets the transmission rate based on the information gathered in the first two components then sends feedback to the transmitter. However, both the proposed framework and the link metric have been implemented and examined in a testbed with 15 IEEE 802.11n wireless nodes under different wireless condition scenarios. The experimental results have shown that the proposed framework provides better throughput compared to other rate adaptation solutions. In fact, ARAMIS provides an increase in throughput up to 10 times better than some other solutions.

5.1.2. Cross-Layer Solutions

Cross-layer solutions will require two or more layers to cooperate and interact to enhance QoS performance for video over WLAN. As mentioned in the previous section the three main layers that have direct impact on video data transmission over wireless networks are Application, MAC and PHY layers. Effective video coding at Application layer in cooperating with access mechanism at MAC layer or rate adaptation and MIMO at PHY layer will produce an efficient cross-layer solution. Below are the most recent and most relevant cross-layer solutions based on the three layers.

5.1.2.1. Application-PHY Solutions

The primary goal of this type of solutions is mapping video coding at Application layer to rate adaptation and MIMO at PHY layer. Most of the solutions are concentrating on selecting best modulation, transmission rate and MIMO mode at PHY along with selecting best coding algorithms at the Application layer. Wang and Liu [44], proposed an optimization scheme based on Adaptive Modulation and Coding (AMC) and video distortion to enhance video transmission over IEEE 802.11n. The aim of the scheme is to adjust Quantization Parameter (QP) at VCL and MCS at PHY to achieve best transmission rate and best video quality. Video distortion is usually under the influence of Mean Square Error (MSE) of compressed and uncompressed video, and Packet Loss Rate (PLR) of the channel that can be measured with SNR. The proposed scheme works based on two main modules, first to gather SNR and MSE information at the receiver, then compare the video distortion information regarding QP and MCS adjustment to find the best transmission mode. Figure 5.4 shows the system model of the proposed solution. The experimental results have shown that after applying the proposed algorithm to select QP and MSC, it is possible to decode video in large SNR ranges and keeping the PSNR range 30-50 which is the range of good to excellent.

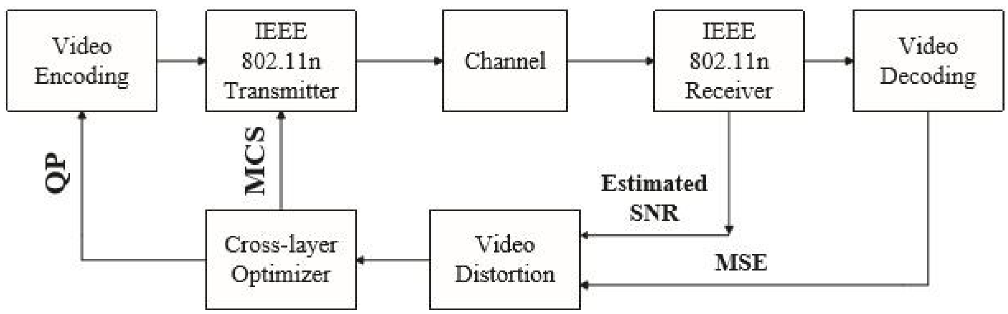

Fig. 5.4. System model used in [44].

Yun et al. [45], proposed Layered Integrated Video Encoding (LIVE) that ensures receivers with low antenna capability and worst channel condition can still achieve acceptable video quality and receivers with the strong signal will obtain better video quality. The proposed solution includes layered coding that considers the heterogeneity of the receivers that guarantees that every receiver will obtain reliable video information based on their bandwidth. It also includes a framework to optimize time and information of the transmission based on the channel condition along with using both soft and modulations. LIVE mostly uses soft modulation because it uses real numbers to represent the analog signals which makes it more efficient. It also uses an integrated soft and hard modulation in some cases to enhance the transmission performance. LIVE was implemented and examined then the simulation results were compared to other proposed frameworks and video codecs, LIVE have shown considerable improvements in PSNR for both unicast and multicast video transmission in which at most cases PSNR rage is 27-43 depending on the receiver's channel condition.

In addition, Bulut et al [46], proposed the use of Raptor code with Application Layer Forward Error Correction (AL-FEC) to improve multicast video quality over IEEE 802.11n. The study conducted on the performance analysis of SM and Space Time Block Coding (STBC) based on channel conditions. In case of high spatial correlation MCS needs higher SNR to meet the requirements, experimental results have shown that using Raptor code significantly increases SNR. Therefore, the use of Raptor code along with STBC will provide better performance under worst channel conditions because STBC performance relies only on SNR, while SM performance relies on both spatial correlation and SNR. However, in low correlation and good channel conditions scenarios, Raptor code will be less effective, but it can provide better video transmission, higher throughput along with faster MIMO switching using SM. In other words, using Raptor code with AL-FEC will provide better video transmission when the MIMO channels have low spatial correlation.

5.1.2.2. Application-MAC Solutions

This type of solutions can be classified into two main types of solutions based on the channel access mechanism at MAC layer. For the contention-based mechanism, the solutions mainly concentrate on mapping video streams into EDCA queue. While in contention-free, the solutions are focusing on bandwidth utilization. Bishnoi et al. [47], proposed a cross-layer approach to map video frames to IEEE 802.11n EDCA without aggregation. The proposed scheme is so called Traffic Queue Controlling Cross-layer Mechanism (TQCCLM), in which it prioritize H.264/SVC video packets at MAC layer based on two type of identification Temporal Identification (TId) Quality Identification (QId). The primary goal of the proposed scheme is to make decision dynamically and choose Access Categories (AC) for video packets based on queue length; figure 5.5 presents the block diagram of the proposed solution. The experiment was based on packet loss, throughput, and

PSNR. Simulation results have shown that the proposed solution provides better PSNR than the default method used in IEEE 802.11n.

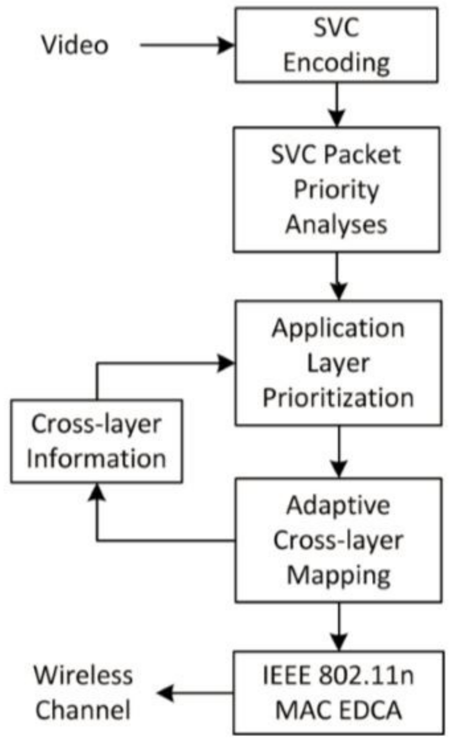

Fig. 5.5. Block Diagram of TQCCLM [47].

Chang et al [1], proposed multipolling controlled access (MPCA) which is an EDCA based contention-free channel access mechanism that is similar to TDMA and reduces overhead during control packet transmission process. Due to unpredictable latency in EDCA and low efficiency of HCCA, both will pose challenges for QoE satisfaction for video over WLAN. Therefore, the proposed MPCA aims to employ enhanced characteristics of both EDCA and HCCA and avoid their deficiency. Furthermore, to provide cooperation between video frames and MPCA, Quality Adjustment Strategy (QAS) has been proposed which is a cross-layer approach that breaks down the video frames into slices at VCL and a NAL unit header will be added to each slice then the slices will be reconstructed as a video frame. Eventually, these video frame will be transferred to

QoS Station (QSTA), figure 5.6 shows the QAS architecture. Moreover, QAS adopts resource allocation technique to differentiate between video and non-video traffic.

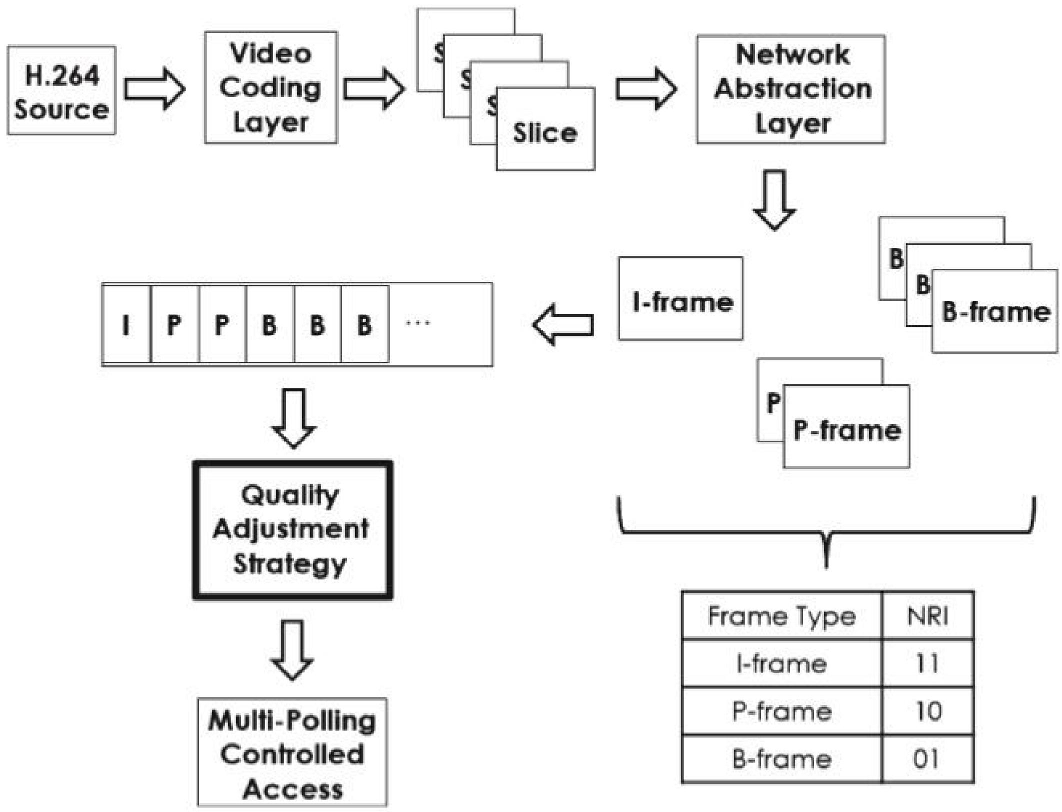

Fig. 5.6. QAS architecture [1].

An experimental study has been conducted to compare EDCA with the proposed MPCA based on packet loss, throughput, mean delay and PSNR over IEEE 802.11ac using eight antennas. The simulation results have shown that MPCA can provide better throughput for a higher number of QSTA than EDCA, which is 27 to 19 respectively. Regarding mean delay and packet loss, MPCA has also shown better performance than EDCA. However, when it comes to PSNR, in low traffic load scenarios both EDCA and MPCA provide similar PSNR rate, but when the traffic load increases, EDCA decreases PSNR ratio to 20 dB. While MPCA with/without QAS still provide

PSNR over 30 dB, figure 5.7 present the result of the comparison between EDCA and EMPCA based PSNR.

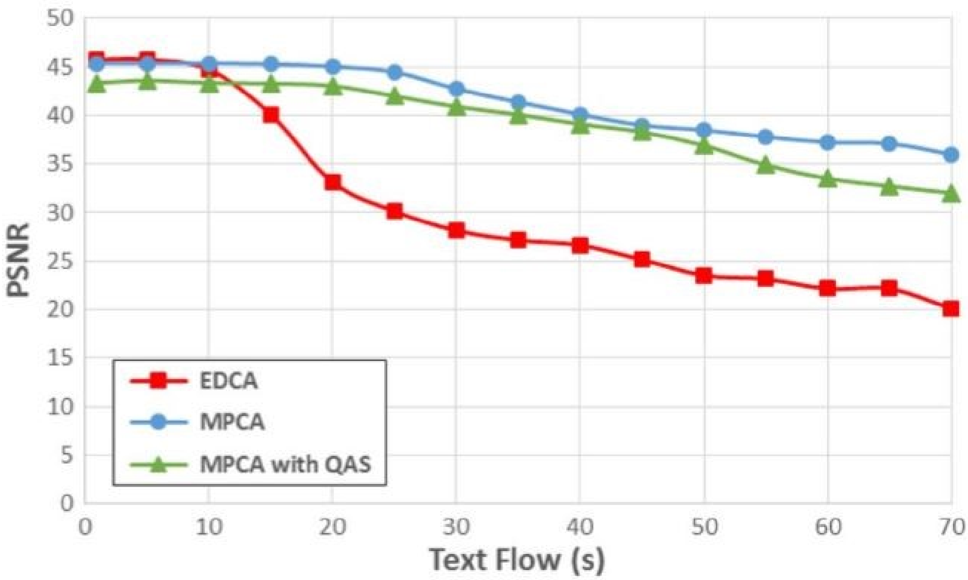

Fig. 5.7. EDCA vs. MPCA based on PSNR [1].

5.1.2.3. MAC-PHY Solutions

Since IEEE 802.11 operates on both MAC and PHY layer, the aim of this type of cross-layer solutions is to maximize the throughput. MAC-PHY cross-layer solutions are not required to be based on video characteristic. The solutions are mostly aiming to enhance the throughput based on MAC layer parameters and transmission rate at PHY layer. Therefore, the solutions will affect the overall traffic including video traffic as well. Tang et al. [48], proposed a rate adaptation algorithm to optimize the throughput based on both link quality and random channel access. The cross-layer solution is called MAC Throughput optimization based Rate Adaptation (MTRA) which is based on SNR link adaptation algorithm. The research study has three main objectives: first, evaluating the proposed MTRA based on performance gain for rate adaptation. Second, developing an analytical Markov chain based model for SNR to quantify MTRA's performance gain efficiently. Finally, the performance of the proposed rate adaptation algorithm has been evaluated and compared to the well-known link throughput optimization based rate adaptation (LTRA), based on parameters such as window size, RTS/CTS, and error rates. The experimental results have shown that MTRA performs better and improve throughput up to 20% at MAC layer. The study has also proven that the performance gain mainly comes from the link adaptation mechanism.

5.1.2.4. Application-MAC-PHY Solutions

This type of cross-layer solutions will involve all three layers to cooperate to provide better video quality over WLAN. As aforementioned, most of the solutions are targeting video compression and codecs at the application layer, aggregation and channel access mechanisms at MAC and transmission rate and MIMO at PHY. Therefore, Application-MAC-PHY cross-layer solutions should consider the characteristics and parameters of all three mentioned layers. Park et al. [49], proposed a cross-layer technique that involves error concealment unit and source significance information (SSI) at the application layer, MCS at MAC layer and Channel Quality Information (CQI) at PHY layer. Their implementation includes Unequal Error Protection (UEP), Rayleigh flat fading channels along with H.264 as video codec. The experimental setup is divided into two phases. First to improve video quality based on SNR, MCS, and error concealment unit, then the second phase will consider the use of SSI and CQI in MCS selection to enhance video quality. The simulation results have shown that the proposed solution provides better video quality but it worth mentioning that the data rate will be reduced slightly.

Koli et al. [50], proposed a cross-layer optimization approach to improve high-quality video transmission over WLAN. The optimization approach is so-called QoS-Optimized Adaptive Multi-layer (OQAM), which enhances Intra Prediction of H.264 at Application in which they call it (EnIP). QQAM uses the feature and characteristics of 802.11e MAC layer, and the goal is to improve adaptive retry limit that so-called Improved MAC Adaptive Retry Limit (IMALr). At PHY layer OQAM aims to enhance Adaptive Forward Error Correction in which they call it (EnAFEC). Figure 5.8 shows the OQAM architecture regarding the TCP/IP model structure.

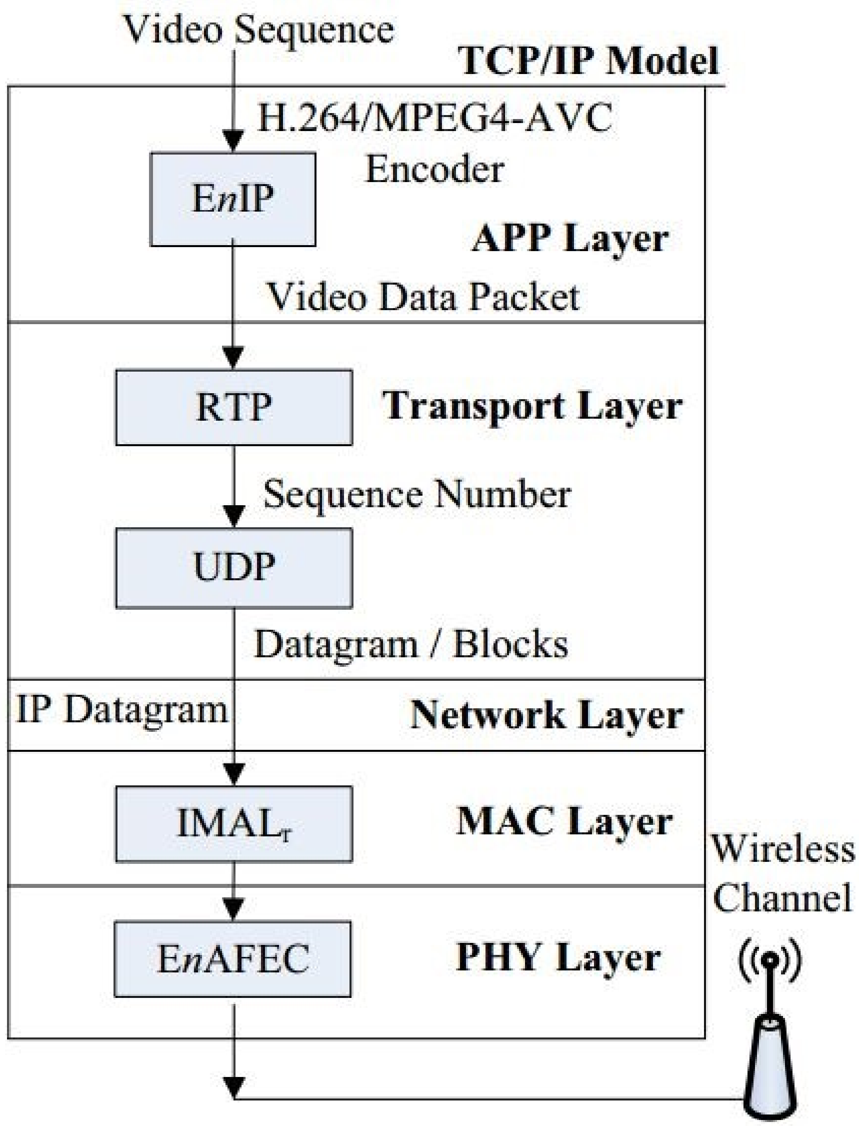

Fig. 5.8. OQAM architecture [50].

After applying the proposed scheme, the simulation results have compared to other proposed solutions based on the three layers. At the application layer, both edge PSNR (EPSNR) and bit rate values are considered to capture the performance of OQAM. The EPSNR value of OQAM is over

47 dB which is higher than previously proposed solutions and OQAM reduces the bit rate by 18% which improves compression efficiency. MAC parameters such as end-to-end delay, packet loss, packet delivery ratio, and throughput are considered to evaluate the OQAM at the MAC layer. The simulation results have shown that OQAM provides 7% lower loss rate, while the end-to-end delay is 16 ms which is acceptable but higher than previously proposed solutions. OQAM provides better throughput and packet delivery ratio compared to other solution in which the throughput is increased by 161%, and packet delivery ratio is 62%. At PHY layer with the use of EnAFEC, OQAM provides 0.3% loss rate, packet delivery ratio of 99.8 and better throughput.

5.2. Vendor Solutions

Most recent and relevant academic solution for video QoS/QoE over WLAN have been technically analyzed in the previous section. It can be observed from the above analytical outcomes that most of the proposed solutions are difficult to implement in the operational networks due to complexity and compatibility issues. Therefore, exploring vendor solutions for video over WLAN issues will provide a better understanding from a network administrator's perspectives and propose the use of features and mechanisms that have been included in the vendor's network devices. Most vendors rely on Wi-Fi Multimedia (WMM) standard to provide QoS for video. WMM provides priority queueing in four levels respectively: voice, video, best effort and background traffic. This feature will enable video packets to have higher priority than best effort packets in the queue. The idea behind WMM is to adjust two parameters at MAC layer which are the back-off timer and inter-frame space. This indicates that shorter back-off time will be assigned to high priority frames, and low priority frames will have to stay longer in the queue. Furthermore, WMM has a resource reservation mechanism that called Traffic Specification (TSPEC) [51], which allows wireless airtime reservation between the client and the access point and this will prevent resource exhausting by video clients. However, WMM implementation differs from one vendor to another and TSPEC specific implementation should be determined by the infrastructure to allow or deny clients.

As aforementioned both video and wireless data rate are variable. Therefore, admission control mechanisms are required to handle the flood of video streams.

Admission control algorithms have been used to detect and recognize different flows of traffic, then gather information about the flows to track client's behavior. This mechanism will allow network administrators to provide different priority for different sets of traffic based on the infrastructures requirements. For example, in the case of huge video traffic, the network administrator can set the bandwidth to 70% video and 30% other traffic (best effort and background). Moreover, admission control algorithm should also provide roaming and load balancing features for the clients. Most of the edge clients will suffer from the low signal. Therefore, the admission control mechanism should be able to roam them to a neighboring access points. In the case of heavy traffic load, the admission control should be able to provide a load balancing mechanism to divide the load among the neighboring access points. In addition, it is recommended that admission control for video be based on end-to-end delivery not only in the WLAN. To achieve this, clients should request both airtime through TSPEC and resources through Resource Reservation Protocol (RSVP).

Another challenge at the MAC layer is transmitting multicast video, this type of video transmission does not provide any delivery acknowledgments which will increase packet loss ratio. Furthermore, setting the transmission rate for the multicast group is another challenge because in multicast all the clients should have a chance to receive the transmission. However, to overcome this challenge most venders rely on their network infrastructure to change the multicast traffic into unicast only at MAC layer, in which packets at network layer remain multicast. This is an effective technique to solve multicast issues but it will pose another challenge due to airtime multiplication. Therefore, the number of client per multicast groups should be maintained by an admission control mechanism.

Since most vendors rely on their network infrastructure to provide QoS/QoE for video, their solutions involve layers other than PHY and MAC. The primary goal is to achieve end-to-end delivery for video through the entire infrastructure which requires detection and identification of video streams. In most cases, video clients will use WMM TSPEC requests to identify their video streams, but the infrastructures should have alternative identification mechanisms in case if the clients were not able to identify their streams. Differentiated Services Code Point (DSCP) is one of the most popular video identification techniques that have been used by the most vendor in which DSCP provides markings in the IP header to notify the infrastructure about the video content of the packet. Since DSCP is not providing the explicit request for admission, admission control will be a challenge.

Basic Service Set Identifiers (BSSIDs) is another technique to identify video flows in the infrastructure in which a dedicated BSSID used for video traffic. This identification technique is widely used in video conferencing solutions but for mixed traffic it is less likely to be effective. Furthermore, as an advanced detection method, the infrastructure can snoop on packets and look for video data. Snooping on protocols such as Session Initiation Protocol (SIP) and Internet Group Management Protocol (IGMP) can provide enough information for the infrastructure to recognize video traffic. In addition, Real Time Streaming Protocol (RTSP) and User Datagram Protocol (UDP) packets can be snooped by the infrastructure as well.

Vendors aim to provide end-to-end QoS for video, and the transmission protocol in use is Real-Time Transport Protocol (RTP) (RFC3550), which is an Internet protocol standard that used for real-time voice and video transmission. RTP operates with RTP Control Protocol (RTCP) during the transmission process to monitor QoS parameters and facilitates multiple stream synchronizations. It also works with Session Initiation Protocol (SIP) which is a signaling protocol that establishes connections from source to destination. RTP is used along with SIP to initiate, control and terminate sessions and it also works with voice and video codecs to prepare the encoded data for transmission. Furthermore, RTP supports multicast transmission to transfer data to multiple destinations.

As aforementioned many enhancements have been made at the application layer which involve video codecs and compression techniques. Video transmission over WLAN can be more enhanced if the infrastructure is aware of video frames and their impact on the transmission. There are two types of video frames that the infrastructure needs to be aware of, which are base frames and difference frames. The decoder will decode the video based on the base frame, which means decoding difference frames will require base frames existence, and this will pose a challenge when the base frames are lost. Therefore, the infrastructure needs to be aware of video frames and be able to distinguish between them so that when the congestion occurs it can drop difference frames instead of the base frame which will cause only one frame outage. Furthermore, setting priority for the base frames will be another solution, in which base frames are assigned low data rate and higher retry limits. Moreover, in case if the base needs to be dropped, the infrastructure should be able to assign the following difference frame as a new base frame.

In addition, there are other techniques that have been used to provide better video transmission based on video codecs. Transcoding is the technique that used to enhance video transmission because of the difference in error resiliency and packet loss rate from a codec to another. This technique will be effective if the infrastructure is aware of the codecs that have been used by both sides of the transmission. If one side is using an older version of the codec that has a higher rate of packet loss, the infrastructure should be able to transcode the video streams. Transrating is the process of setting data rate based on the wireless node capabilities which means nodes with lower signal will get low video quality and vice versa. Furthermore, synchronizing and caching are two additional feature that the infrastructure needs to be aware of. Synchronizing is required when voice and video are transmitted in separate streams, in this case, the priority for both voice and video should be at the same level. Caching technique is quite similar to other caching systems in which the infrastructure caches the requested video and make it available for other clients without processing the request to the original source [52].

Different wireless vendors have different techniques and feature to provide QoS/QoE for video. Below are the technical overview of three most popular vendors and their solution to overcome video transmission challenges.

5.2.1. Aruba Infrastructure

Aruba is one of the most popular wireless vendors in today's world. Aruba's architecture mostly relies on WMM to provide QoS for multimedia over WLAN. Aruba refers to two main areas as the major challenges in providing QoS over WLAN, which are packet prioritization and bandwidth usage by wireless nodes. To overcome these challenges Aruba proposes the use of Mobility Controller, which is similar to wireless controllers that provided by other venders. Mobility Controller acts as a stateful firewall that tracks and monitors packets to detect and distinguish different protocols and parameters. The controller first performs an in-depth packet inspection to detect protocols such as SIP and RTP; then the firewall sets the rules for both source and destination based on QoS parameters. When the multimedia packet identified by the firewall, it can change their priority and provide layer two and layer three tags for the packets to be transmitted across the infrastructure. Moreover, the firewall can match uplink streams with downlink streams, and if they were the same, it will apply the same QoS priority to the uplink stream. However, the QoS technique that has been used by Aruba is on a packet-by-packet basis, which means when the clients have multimedia streams alongside with background data the Mobility Controller can still provide QoS for multimedia based on packet prioritization.

In previous standards, admission control was one of the most important features in providing QoS for multimedia due to limited bandwidth and fixed codecs. But after the release of IEEE 802.11n standard, admission control is no longer a necessary feature and the admission issues are maintained by load balancing and bandwidth management techniques. Aruba provides spectrum load balancing and band steering to deal with admission issues. Spectrum load balancing is the technique of distributing wireless clients equally depending on the RF channels by moving clients between APs. Adaptive Radio Management (ARM) is a technology by Aruba that provides band steering which allocates clients based on their band capabilities, most of the time 2.4GHz band is crowded, and 5GHz is less occupied. Therefore, the band steering technique will allocate 5GHz capable clients to use 5GHz band so that the 2.4GHz client can utilize the 2.4GHz band. Furthermore, bandwidth management is also a necessary feature to prevent clients that are not capable of using new standards from being bandwidth siphoned by newer and faster clients. Aruba provides different bandwidth management techniques that are based on classes and types of the clients. The bandwidth allocation mechanism is a token-based mechanism that controls the transmission based on airtime requirements. Figure 5.9 presents Aruba's ARM architecture along with bandwidth management and WMM queuing.

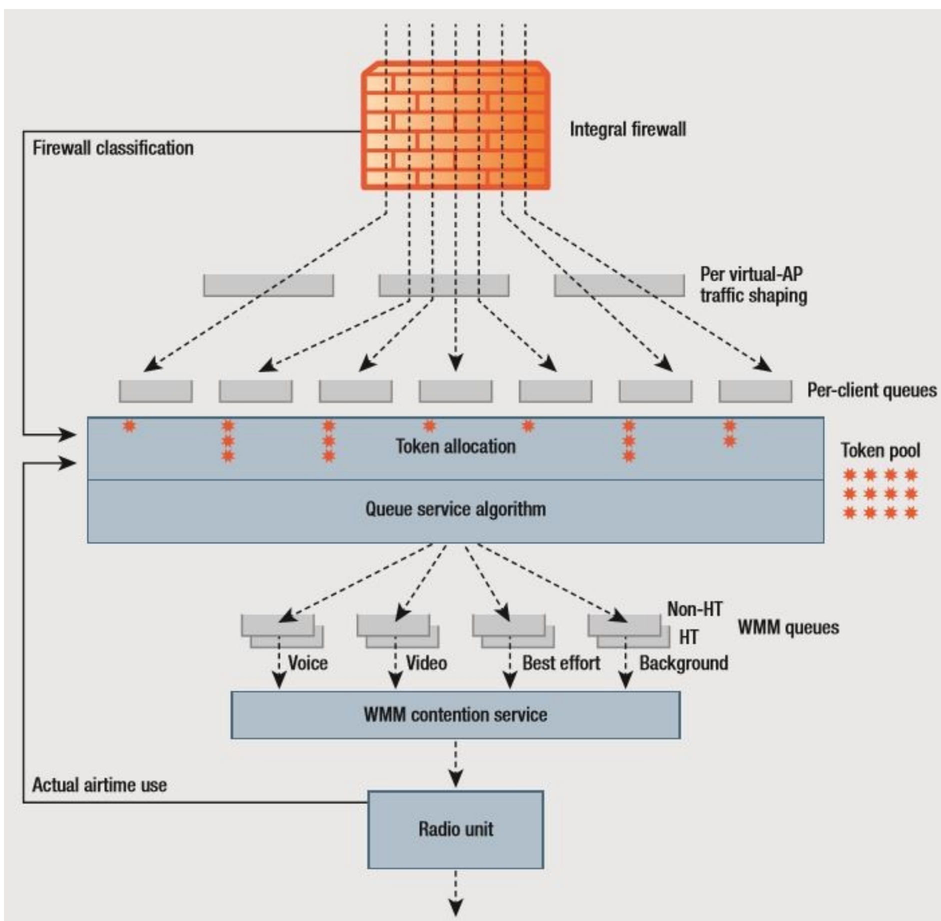

Fig. 5.9. ARM Architecture [53].

Classifying the bandwidth based on the type of protocols and traffics will play a big role in providing QoS, because no traffic will be denied as long as there is enough bandwidth available. In other words, if the bandwidth is set to assign 60% to video and 40% to voice, background data can still be transmitted as long as the voice and video not reached their limits. Moreover, the bandwidth can be classified based on the client types. IEEE 802.11 standards are backward compatible and in case of having legacy clients the bandwidth can be assigned accordingly so that every client can have enough bandwidth and this prevent excessive bandwidth consuming by faster clients.

5.2.2. Cisco Infrastructure

Cisco is the dominant networking vendor in the market that manufactures many different network devices including the Cisco IOS along with the support for standards, protocols and providing several proprietary features. In the core of Cisco wireless infrastructure, there is Cisco WLAN Controller (WLC) which is a centralized management device that accelerates deployment process and controls the traffic between the wired and wireless networks. WLC can also assign security policies and provide QoS networkwide for multimedia. The Cisco wireless infrastructure also relies on WMM to provide QoS for multimedia, and it provides four levels of priority (Platinum, Gold, Silver, and Bronze), the prioritization process depends on the DSCP values in the packets in which the WLC can modify these values.

As a video QoS over WLAN solution, Cisco introduced VideoStream technology which operates across the three layers (PHY, MAC, and Application) to provide reliable and consistent transfer mechanism for the different type of video over wireless networks. VideoStream technology is a set of feature that includes admission, prioritization, resources reservation, multicast solutions and video scaling. Admission and prioritization mechanism allows system administrator to prioritize different video stream within the traffic flow. This will benefit those organizations that invests in video because video is very bandwidth intensive and this technique will help to manage the bandwidth properly. This feature can be applied to both 2.4 GHz and 5 GHz and provides several QoS options at the WLAN level.

Resource Reservation Control (RRC) is one the feature that Cisco VideoStream uses at the MAC layer to manage and control admissions and policies. RRC prevents the bandwidth from being exhausted by denying clients with oversubscribed requests. RRC’s technique is to determine admission control based on the channel utilization in which more clients with video traffic will scale the utilization. Furthermore, VideoStream provides multicast to unicast technique which enhances the video transmission process and provides reliable multicast video streaming. Wireless clients are using Internet Group Management Protocol (IGMP) messages to subscribe and join a specific multicast group then the infrastructure will snoop IGMP messages and collect information, metrics, and policies about that specific stream. After checking policies, the client will be notified about the transmission of reliable multicast streams, at the meantime the infrastructure measures if there are enough bandwidth and airtime available for the transmission and looks at the metrics to determine admission control. When all requirements are met, the infrastructure will notify the access point about the reliable multicast transmission then the access point converts the multicast frames into unicast frames.

In addition, VideoStream provides higher video scaling which is the technique that optimizes the traffic based on the number of supported clients per controller. In Cisco infrastructure, the frame replication process during the conversion of multicast to unicast happens at the access point (the edge) which reduces the load on the controllers and makes the network more efficient. Other vendors also use the multicast to unicast technique, but mostly it affects the entire infrastructure including wired and wireless.

5.2.3. Ruckus Infrastructure

As a wireless vendor, Ruckus produces many wireless devices and provides different QoS techniques for multimedia. Ruckus also relies on WMM and aims to provide features such as per client traffic queueing, prioritization, classification, airtime fairness, load balancing and band steering. Ruckus provides robust application-aware per client classification mechanism that provides better bandwidth management and better traffic shaping for multimedia. It also uses airtime fairness technique for better use of the available spectrum among clients. As a video QoS solution, Ruckus has SmartCast, which is an algorithm based technology that designed to enhance the performance of the delay-sensitive applications such as voice and video. SmartCast dynamically queues and schedules traffic per clients which means advanced classification and scheduling will be applied to each client. SmartCast per client queuing technique is an optimal solution for video because it prevents clients from being affected by disruptive clients. Moreover, SmartCast does not require any configuration, and it will automatically classify and prioritize traffic based on MAC or network layer.

The force behind SmartCast technology is the traffic inspection software that located in the core of Ruckus infrastructure which automatically inspects and classifies packets then based on the classification it will put the packets into designated queues. The inspection process results will be the information gathered from packets and frames headers (TCP, UDP, VLAN tags, IPv4, and IPv6). In the case of no tag available on the traffic, SmartCast uses heuristics analysis to classify that traffic flow. After classifying and queueing SmartCast uses weighted round robin to schedule the traffic based on airtime, throughput and prioritization. In addition, SmartCast converts multicast to unicast using IGMP snooping technique to guarantee delivery and provide highest data rate available.

SmartCast manages spectrum resources through load balancing and band steering techniques in which they distribute clients among the APs without the requirement of specialized software or configuration on the client side. SmartCast automatically steers dual-band devices to 5 GHz band in which 5 GHz band has 23 non-overlapping channels, and this will guarantee high availability and maximize spectrum resource availability. In addition, to prevent any disruption from legacy and underperforming clients, SmartCast employs an advanced scheduling technique that called airtime fairness in which the scheduling process is based on per station and clients will have equal airtime to transmit packets. This technique allows the faster clients to take advantage of their airtime to transmit and this will increase overall network performance [54].

6. FINDINGS AND OPEN RESEARCH ISSUES

Latest academic video QoS/QoE over IEEE 802.11n/ac solutions have been analyzed along with the QoS performance overview of three popular wireless vendors. Academic solutions are classified into two main categories single-layer and cross-layer solutions. Single-layer solutions are those solutions that targeted a specific protocol or algorithm to enhance at a specific layer. These type of solutions are improving the overall transmission performance at the targeted layer and they are not fixed to video transmission only. Application layer solutions are mostly focused on enhancing a particular video codec to increase PSNR and provide better video quality. At the application layer, the research studies were conducted on two main factors error resiliency and rate distortion. Since these type of solutions are mostly based on video codec characteristic they are not limited to wireless networks only. At the MAC layer, single layer solutions are mostly based on frame aggregation enhancements in which multiple frames are encapsulated into a single large frame to accelerate video transmission over WLAN. These type of solutions were designed to enhance video QoS/QoE, but they also have drawbacks that may cause collisions and higher jitter. Since IEEE 802.11ac provides up to 8 spatial streams and data rate of 1.3 Gbps it is hard to find research studies on enhancing video QoS/QoE at IEEE 802.11ac PHY layer. PHY layer solutions are mostly targeting rate adaptation and MIMO mode enhancement which may not be aimed for video transmission only.

Cross-layer solutions are more sophisticated and more effective than single-layer solution due to the cooperation of two or more layers in a single QoS/QoE implementation. Most of the cross-layer solutions are designed to enhance video QoS/QoE directly. Application-PHY solutions are aiming to adjust rate adaptation and MIMO mode at PHY layer and selecting proper coding algorithms at the Application layer. This type of solutions can be effective and have a direct impact on video QoS depending on correlation and channel conditions. Application-MAC solutions are quite an effective solution, and their primary goal is to enhance PSNR by using different channel access mechanism than EDCA. There are two main types of channel access mechanisms which are contention-based and contention-free, video QoS-based solutions using both channel access mechanisms have been analyzed, and the results show significant improvements in PSNR ratio. MAC-PHY solutions are not related to video transmission directly, and they aim to enhance the overall performance of the network by maximizing the throughput which eventually enhances video transmission. The solutions that involve all three layers are by far the most effective ones to enhance video QoS/QoE because they aim to select best coding algorithm at the application layer. Adjust QoS parameters at MAC layer then select proper rate adaptation and MIMO mode at PHY layer which means this type of solution will provide a pure QoS-based solution for video over WLAN. However, many academic solutions have been proposed to enhance video transmission, but most of them are not applicable and cannot be implemented due to the complexity. Despite the complexity many of the proposed solutions have drawbacks, and negative impacts on other factors especially power consuming.

In addition, since academic solutions are complex and not easy to implement, QoS solutions by three popular vendors have been technically overviewed. It can be noticed that most vendors are relying on WMM to provide QoS for multimedia which means their solutions are not designed specifically for video. Moreover, vendor’s solutions depend on their network infrastructure that includes specialized network management devices and specialized software to provide QoS. Since all wireless vendors are using the same IEEE standards and protocols their solutions are quite similar and in some cases are almost the same especially when it comes to multicast video challenges they are all using the multicast to unicast conversion technique. Figure 6.1 presents the enhancement techniques that have been the area of research to enhance QoS/QoE for video over wireless networks.

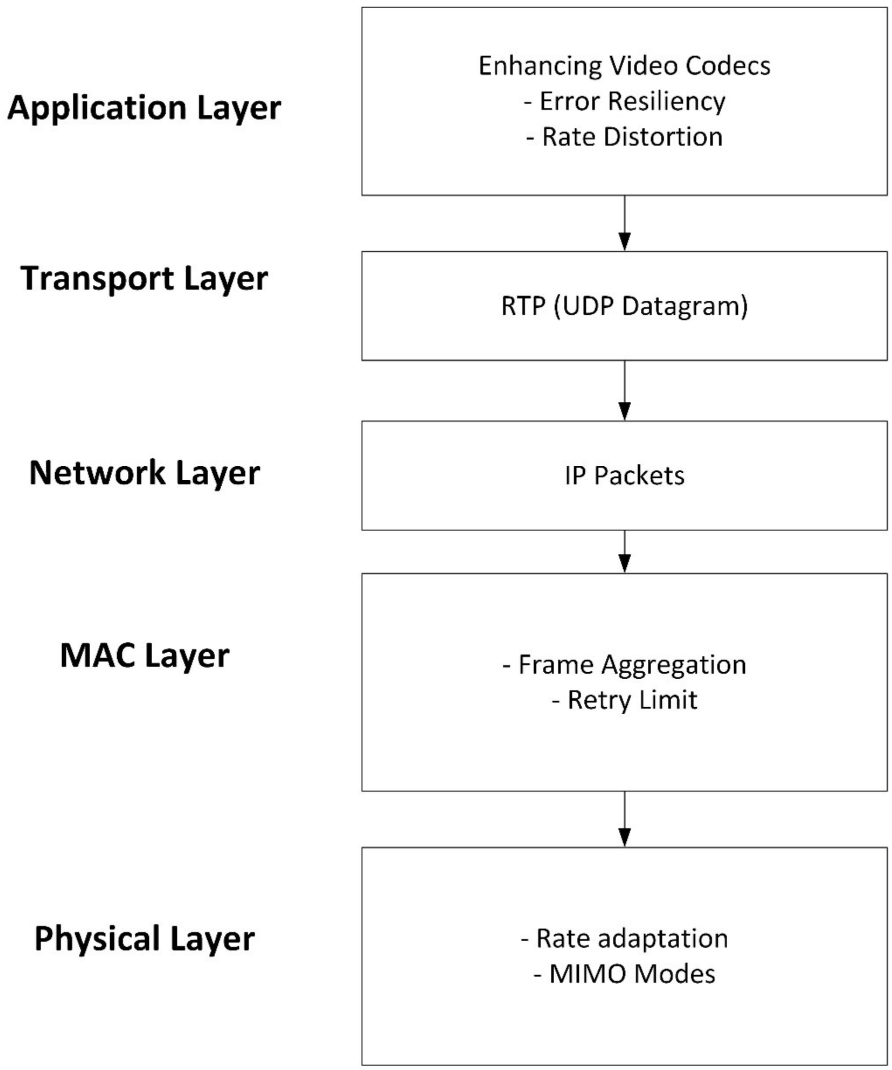

Fig. 6.1. QoS/QoE Enhancement Mechanisms Through OSI Model.

7. CONCLUSION AND FUTURE WORKS

This thesis provided a detailed technical overview of the most recent video QoS/QoE over IEEE 802.11n/ac solutions. The survey study started with highlighting technical characteristics and feature of video codecs. ITU-T provides several video coding standards which started with H.261, and the latest approved video codec is HEVC/H.265. H.264 is the most used codec for video transmission over WLAN, and its functionality is based on two layers NAL and VCL. NAL is designed to simplify and arrange video frames for transmission over the network. VCL is the coding layer that is responsible for blocking, slicing, quantization, filtering and predictions of the video frames. HEVC/H.265 is the latest video codec that based on H.264 and provides several new features that affect video transmission over WLAN. Furthermore, as an alternative to ITU-T video coding standards, Google's VP8 technical features have been presented. This thesis aimed to provide a technical analysis of possible video QoS/QoE over WLAN solutions. Therefore, both IEEE 802.11n/ac have been technically overviewed and briefly compared. IEEE 802.11n is the standard that is currently in use, and it provides the data rate of up to 600 Mbps and operates on both 2.4 GHz and 5 GHz. IEEE 802.11ac is the upcoming standard that operates only on 5 GHz and provides 1.3 Gbps data rate. After reviewing both video and wireless characteristics QoS and QoE strategies have been discussed, and their metrics and measurement techniques have been presented.

The thesis strategy was to review video QoS/QoE over WLAN issues and solutions from a network administrator's perspectives. Therefore, the survey study was divided into two main categories academic solutions and vendors' solutions. Since video over WLAN transmission involves three layers PHY, MAC, and application, the academic solutions are based on those three layers and they were divided into two major categories single-layer solutions and cross-layer solutions. Single-layer solutions are aiming to enhance the characteristic of a specific layer such as enhancing video codec or video coding algorithm at the application layer, frame aggregation at MAC and rate adaptation and MIMO mode at PHY. Single-layer solution mostly affects the entire traffic and they are not limited to video only. Cross-layer solutions are involving two or more layers in a single QoS/QoE implementation and most of them were designed specifically to solve video QoS/QoE issues. Cross-layer solutions perform better than single layer solutions due to the involvement of the three layers. However, single-layer solutions mostly affect the entire traffic and most of the cross-layer solutions cannot be implemented due to the complexity. Therefore, this thesis presented a technical overview of three popular wireless vendors' solutions for video QoS/QoE. Wireless vendors are relying on their network infrastructure and WMM features and characteristic to provide QoS for video. As future works developing a cross-layer solution that involves all three layers and modifies the video codec, MAC parameters, and rate adaptation according to the video characteristic along with less complexity in the implementation would provide better video quality and better user experience.

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