# DEPLOYING IPTV (MOBILE TV) OVER MOBILE WIMAX NETWORKS: A Comparative and Simulation Study

Jamil Hamodi[1, 2], Anwar Alsagaf[2], Ravindra Thool[1], Yousef Holba[2]

**Abstract**— Deployment IPTV (Mobile TV) by telecommunication companies around the world becomes a host of unique operational challenges, and one's of the intense interest subjects in the research these days, is anticipated to be the main revenue generators in the near future and the efficiency of video streaming over next generation 4G. IEEE specifies different modulation techniques for WiMAX; namely, BPSK, QPSK, 16 QAM, 64 QAM, and AMC. In this paper, a simulation performance study of Mobile TV over mobile WiMAX is conducted with different types of adaptive modulation and coding taking into account key system and environment parameters which include real time video coded by different video codecs (MPEG-4, H.264/AVC, and SVC), variation in the speed of the mobile, random mobility of mobile users, path-loss, scheduling service classes with the adaptive and fixed type of modulations. Simulation has been conducted using OPNET simulation. Simulation results show that higher modulation and coding, and dynamic adaptation of modulation and coding schemes based on channel conditions can offer considerably more enhanced QoS and at the same time reduce the overall bandwidth of the system.

**Index Terms**— Mobile TV, QoS, adaptive modulation and coding, WiMAX, OPNET, Random Mobility, Video Codec, Performance Study.

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## 1 INTRODUCTION

Internet Protocol Television (IPTV) is a method of delivering digital video and audio content across an IP broadband network, commercial deployments of IPTV services by telecommunication companies around the world continue to increase, and becomes a host of unique operational challenges for telecoms, cable, and satellite TV providers [1]. Worldwide Interoperability for Microwave Access (WiMAX) technology is one of the future communications capable for offering high QoS at high data rates for IP networks. The high data rate and Quality of Service (QoS) assurance provided by this standard has made it commercially viable to support multimedia applications such as video telephony, video gaming, and mobile TV broadcasting. System architecture to support high definition video broadcasting (like MPEG-X, H.264/AVC and SVC) that provides a mobility of 30 kmph in an urban and suburban environment has been developed [2], [3].

The International Telecommunication Union focus group on IPTV (ITU-T FG IPTV) defined IPTV as multimedia services such as television/ video/ audio/text/graphics/data delivered over IP based networks (which also sometimes triple play) managed to provide the required level of quality of service and experience, security, interactivity and reliability. The type of service providers' involved in deploying IPTV services range from cable and satellite TV carriers to the large telephone companies and private network operators in different parts of the world. IPTV has a numerous number of features [1] including: Support for interactive TV, The interactive means the two-way capabilities of IPTV systems which allow service providers to deliver a whole portfolio of interactive TV applications. The types of services delivered via an IPTV service can include standard live TV, high definition TV (HDTV), interactive games, and high speed Internet browsing; Time shifting: The time shifting is a mechanism for recording and storing IPTV content for later viewing, and IPTV in combination with a digital video recorder permits the time shifting of programming content; Personalization: an end-to-end IPTV system supports bidirectional communications and allows end users personalize their TV viewing habits by allowing them to decide what they want to watch and when they want to watch it; Low bandwidth requirements: Instead of delivering every channel to every end user, IPTV technologies allows service providers to only stream the channel that the end

1 *Information Technology Department, Shri Guru Gobind Singhji Institute of Engineering and Technology (SGGS), Nanded, India, E-mail: {jamil_hamodi, rcthool} @yahoo.com*
2 *Computer Engineering Department, Hodeidah University, Hodeidah, Yemen, E-mail: Anwar_alsaqaf@hotmail.com, yousef_holba2007@yahoo.com*

user has requested. This attractive feature allows network operators to conserve bandwidth on their networks; Accessible on multiple devices: Viewing of IPTV content is not limited to televisions. Consumers often use their PCs and mobile devices to access IPTV services.

Mobile TV is a technology that enables users to transmit and receive TV program data through IP-based wired and wireless networks. Users can enjoy IPTV services anywhere and anytime with mobile devices. Four kinds of mobile TV technology approaches, namely: mobile TV over IP that was discussed in this work, IPTV over mobile device, cellular IPTV, internet IPTV. However, with rapid adaptation to the user's requirements, Mobile TV may be preeminent in the future. It is convenient for users who can access an IPTV service through various wireless networks with mobile devices.

In cellular communication, the quality of a signal received by user equipment depends on a number of factors, namely: the distance between the desired user and interfering base stations, path loss exponent, log-normal shadowing, short term Rayleigh fading and noise. In order to improve system capacity, peak data rate and coverage reliability, the signal transmitted to and by a particular user is modified to account for the signal quality variation through a process commonly referred to as link adaptation. Traditionally, CDMA systems have used fast power control as the preferred method for link adaptation. Recently, Adaptation Modulation and Coding (AMC) have offered an alternative link adaptation method that promises to raise the overall system capacity. Based on channel conditions, the scheme can be changed on per frame and per user basis. In order to maximize throughput in a channel varying in time, Adaptive and Modulation Coding can be used [4]. AMC provides the flexibility to match the modulation-coding scheme to the average channel conditions for each user. With AMC, the power of the transmitted signal is held constant over a frame interval, and the modulation and coding format are changed to match the current received signal quality or channel conditions. In a system with AMC, users close to the BS are typically assigned higher order modulation with higher code rates (e.g. 64 QAM with R=3/4 turbo codes), but the modulation-order and/or code rate will decrease as the distance from BS increases [5].

Due to the rich-information characteristic of video, raw video needs a large amount of bandwidth for transmission over networks. Therefore, video compression is inevitable for video transmission in various networks to save bandwidth and faster transmission. Video compression could be adopted in various applications, such as VoD, TV broadcasting, Mobile TV, video over Internet, and video conferencing. Due to the emergence and high demand of HD videos, video compression techniques have become even more important and need further development. The most popular and dominant lossy compression methods used by commercial IPTV providers are MPEG and VC-1 technologies [1]. MPEG technology is a compression standard, which is widely used by satellite, cable, and terrestrial TV systems. MPEG is an acronym for Moving Pictures Experts Group and represents an industry association that was formed to help develop compression techniques suitable for video transmission. The group was originally created by the International Organization Standardization (ISO) with the International Engineering Consortium (IEC) to work on developing video and audio compression standards. Since its foundation, the group has produced a family of major compression standards—MPEG-1, MPEG-2, MPEG-4 (Parts 2 and 10), MPEG-7, and MPEG-21. MPEG-2 and MPEG-4 Part 10 (H.264) are the most widely deployed by IPTV service providers. MPEG-4 Part 10 AVC/H.264 is the explosion of next generation networks is driving demand for advanced video services, it supports a wide range of next generation of applications of digital compressed video from relative low bit-rate Internet streaming to high bit-rate HDTV broadcast. It is the most recent standard which is designed with the aim to compress raw video to provide good video quality at substantially lower bit rates compared with previous standards. The Scalable Video Coding (SVC) extension of H.264/AVC has achieved significant improvements in coding efficiency with an increased degree of supported scalability relative to the scalable profiles of prior video coding standards.

The performance evaluation of any new emerging technology with different applications is imperative in order to understand and improvise the system to the desired level. Emulation is a high fidelity model of the physical system which could be a better alternative for simulation. This paper goes significantly further, providing a comprehensive study of deploying IPTV (Mobile TV) over Mobile WiMAX, including insights into the characteristics and requirements of the new services (IPTV) needed to be supported in our networks. In this paper, we seek to answer the following questions about a large-scale IPTV (Mobile TV) deployment. What are the QoS requirements for the new service we need to deploy it? Will existing network support this new service and satisfy the standardized QoS requirements? Which is the best Modulation and Coding scheme to guaranty the best users view? This work is geared towards investigating the performance study of IPTV (Mobile TV) over Mobile WiMAX networks when considering different adaptive and fixed modulation and coding schemes using simulation software OPNET Modeler. OPNET Modeler provides comprehensive development of network models including all the necessary parameters that need to be reflected in the design procedure of PHY and/or MAC layers. A series of simulation scenarios under OPNET for broadband wireless communication is developed. In sharp contrast to prior work [27], the research work and results presented in this paper focus mainly on the use of real-time audio/video movie coded by different video codec (MPEG-4. H.264/AVC, and SVC) for modeling and simulation Mobile TV deployment over Mobile WiMAX. This paper, in the main, aims to establish a comparative study of performance for Mobile TV (VoD) over Mobile WiMAX under varying speeds of the mobile nodes, random mobility of mobile nodes, using different path-loss models under both AMC and fixed types of modulation techniques, and to identify the factors affecting the video performance.

The remainder of this paper is organized as follows. Section 2 gives relevant background and preliminaries on WiMAX PHY and MAC layers. Section 3 provides a brief state of the art. The architecture of IPTV over WiMAX networks is highlighted in Section 4. Section 5 provides video traffic characteristics and requirements. Section 6 describes the practical steps to be taken prior to simulation. Simulation results and analysis obtained are provided in section 7. Section 8 is naturally the conclusion of the findings as a whole and a summation of this modest research endeavour.

## 2 WiMAX BACKGROUND AND PRELIMINARIES

This section provides relevant WiMAX background and preliminaries pertaining to QoS support and the physical layer on WiMAX modulation and coding schemes. The WiMAX physical layer is based on orthogonal frequency division multiplexing (OFDM), which gives immunity to frequency selective fading and inter-symbol interference. OFDM is the transmission scheme of choice to enable high-speed data, video, and multimedia communications and is used by a variety of commercial broadband systems, including DSL, Wi-Fi, Digital Video Broadcast-Handheld (DVB-H), and MediaFLO, besides WiMAX. The standard defined within OFDM is an elegant and efficient scheme for high data rate transmission in a non-line-of-sight or multipath radio environment. Five PHY layers interfaces are in the IEEE 802.16 standard. These physicals interfaces are Wireless MAN-SC known as SC operating in LOS condition at a frequency between 10-66 GHz, Wireless MAN-SC known as SCa operating in NLOS condition at a frequency below 11GHz, Wireless MAN-OFDM known as OFDM operating in NLOS condition at a frequency below 11GHz, Wireless MAN-OFDMA known as OFDMA operating in NLOS condition at a frequency below 11GHz, and last PHY interfaces is a Wireless HUMAN with frequency below 11 GHz. Among these PHY interface standards Wireless-MAN OFDM, a 2.5 GHz base frequency using 5 MHz has been used in our simulation that provides a 512-point TDD-based OFDM PHY for point-to-multipoint operations in NLOS conditions on frequencies between 2GHz and 11GHz [6].

Mobile WiMAX encompasses a number of key features that are necessary to transport IPTV services and applications, these features includes [1]: The technology supports peak data speeds of around 32-46 Mbps, these types of speeds if deployed correctly allow delivery of compressed IP based high definition content to mobile handsets; It exploits technologies such as OFDMA and optimized handoffs to allow IPTV viewers access multicast broadcast TV channels in geographical areas that are susceptible to the effects of multipath transmission paths; It integrates with multimedia subsystem (IMS), which simplifies interworking between IPTV applications and other IP based services such as high speed Internet access and VoIP; Mobile WiMAX provides support for advanced quality of services (QoS) mechanisms, which are beneficial to real-time applications such as IPTV.

WiMAX PHY layer uses adaptive modulation based on Orthogonal Frequency Multiplexing (OFDM), which enables it to serve many users in a time division manner in a round robin fashion. This adaptive modulation is used to create the highest data rate for a certain link class. Modulation is a process by which carrier waves are used to carry a digital signal or message. The most commonly modulation used are: Amplitude shift keying (ASK), frequency shift keying (FSK), and phase shift keying (PSK). The choice of one of these modulations depends on the technology, the channel quality, the space, and the transmitted power. The last modulation which called quadrature amplitude modulation (QAM) in WiMAX is a combination of combined ASK and PSK.

WiMAX supports a variety of modulation and coding schemes and allows the scheme to change on a burst-by-burst basis per link, depending on channel conditions, these modulation and coding schemes are: QPSK, 16 QAM, and 64 QAM. WiMAX BS, working from site conditions, distance, channel interference, the user’s need, and the QoS desired, dynamically selects a modulation scheme. High-order modulation, such as 64 QAM, provides higher throughput but a lower coverage range. Low-order modulation, such as QPSK, provides lower throughput but a higher range from the same BS. Using the channel quality feedback indicator helps the mobile provide the base station with feedback on the downlink channel quality. For the uplink, the base station can estimate the channel quality based on the received signal quality. The base station scheduler can take into account the channel quality of each user's uplink and downlink and assign a modulation and coding scheme that maximizes the throughput for the available signal-to-noise ratio. Adaptive modulation and coding significantly increases the overall system capacity, as it allows real-time trade-off between throughput and robustness on each link. The modulation used in WiMAX is the orthogonal frequency division multiplexing (OFDM). WiMAX OFDM features multiple subcarriers ranging from a minimum of 256 up to 2048, each modulated with QPSK, 16 QAM, or 64 QAM modulations, where 64 QAM is optional on the uplink channel. The advantage of orthogonality is that it minimizes self-interference, a major source of error in receiving signals in wireless communications.

Channel coding schemes are used to help reduce the SNR requirements by recovering corrupted packets that may have been lost due to burst errors or frequency selecting fading. These schemes generate redundant bits to accompany information bits when transmitting over a channel. Coding schemes include convolution coding (CC) at various coding rates (1/2, 2/3 and 3/4) as well as conventional turbo codes (CTC) as various coding rates (1/2, 2/3, 3/4, and 5/6). The coding rate is the ratio of the encoded block size to the coded block size. The available coding rates for a given modulation scheme with the minimum signal to noise rate and the peak UL and DL data rates for 5 MHz channel mobile WiMAX with different information bits/symbol are listed in Table 1 [7], [8].

**Table 1. Mobile WiMAX PHY data rate and SINR for 5 MHz channel**

| Modulation Scheme | Coding | Information bits/symbol | Minimum SINR (dB) | Average Received SINR (dB) | Dwon-link rate (Mbps) | Up-Link rate (Mbps) |
|---|---|---|---|---|---|---|
| QPSK | ½ | 1 | 5 | 9.4 | 3.17 | 2.28 |
| | ¾ | 1.5 | 8 | 11.2 | 4.75 | 3.43 |
| 16 QAM | ½ | 2 | 10.5 | 16.4 | 6.34 | 4.57 |
| | ¾ | 3 | 14 | 18.2 | 9.5 | 6.85 |
| 64 QAM | ½ | 3 | 16 | 22.7 | 9.5 | 6.85 |
| | 2/3 | 4 | 18 | 23.2 | 12.6 | 9.14 |
| | ¾ | 4 | 20 | 24.4 | 14.26 | 10.28 |

The key feature of adaptive modulation is that it increases the range over which a higher modulation scheme can be used, since the system can flex to the actual fading conditions, as opposed to having a fixed scheme that is budgeted for the worst case conditions [9]. Two sets of AMC schemes, AMC-1 and AMC-2, are considered to be used in DL in our simulation. These AMC profiles are same as in [10], [11]. Each AMC is basically characterized by two threshold parameters, one is mandatory and the other is the minimum entry threshold for different modulation schemes. The mandatory exit threshold is the SINR at or below where this burst profile can no longer be used and where a change to a more robust (but also less frequency-use efficient) burst profile is required and the minimum entry threshold is the minimum SINR required to start using this burst profile when changing from a more robust burst profile [12]. Table 2 shows the AMC profile used in our simulation. Here, AMC-2 is conservative AMC as it is using lower order MCS most of the time.

**Table 2. AMC profiles**

| Modulation and coding schemes | AMC-1 | | AMC-2 | |
|---|---|---|---|---|
| | Mandatory exit threshold (dB) | Minimum entry threshold (dB) | Mandatory exit threshold (dB) | Minimum entry threshold (dB) |
| QPSK ½ | -20 | 2.0 | -20 | 2.0 |
| QPSK ¾ | 5.0 | 5.9 | 11 | 11.9 |
| 16 QAM ½ | 8.0 | 8.9 | 14 | 14.9 |
| 16 QAM ¾ | 11 | 11.9 | 17 | 17.9 |
| 64 QAM ½ | 14 | 14.9 | 20 | 20.9 |
| 64 QAM 2/3 | 17 | 17.9 | 23 | 23.9 |
| 64 QAM ¾ | 19 | 19.9 | 25 | 25.9 |

WiMAX supports different signal bandwidths ranging from 1.25 to 20 MHz to facilitate transmission over longer ranges in different multipath environments. In wireless communication systems, information is transmitted between the transmitter and the receiver antenna by electromagnetic waves. During propagation, electromagnetic waves interact with the environment, thereby causing a reduction of signal strength. Another limiting factor for higher sustained throughput in wireless communications, especially when the terminal nodes have the mobility, is caused by reflections between a transmitter and receiver, viz, a propagation path between the transmitter and the receiver is regarded. The propagation path between the transmitter and the receiver may vary from simple line-of-sight (LOS) to very complex one due to diffraction, reflecting and scattering [13]. To estimate the performance of mobile TV over WiMAX channels, propagation models in [14] are often used. Path loss models describe the signal attenuation between a transmitting and a receiving antenna as a function of the propagation distance and other parameters which provide details of the terrain profile required to estimate the attenuating signals. Path loss models represent a set of mathematical equations and algorithms which are applied to radio signal propagation prediction in certain environments [15]. Path-loss is highly dependent on the propagation model, the common propagation models namely Free Space, Suburban Fixed (Erceg), Outdoor to Indoor and Pedestrian Environment and Vehicular Environment are given in Table 3. These models are used in mobile WiMAX performance evaluation through OPNET simulation.

**Table 3. Path-loss models**

| Propagation Model | Mathematical formulation | Description |
|---|---|---|
| Free Space model | $P_{rx}(r) = P_{tx}G_{tx}G_{rx}/((4\pi)^2 d^2 L)$, where Ptx is the transmitted power, Prx(r) is the received power, Gtx is the transmitter antenna gain, Grx is the receiver antenna gain, d is the Tx-Rx separation and L is the system loss factor depended upon line attenuation, filter losses and antenna losses and not related to propagation. | It is a mathematical model hardly applicable without considering the fading effect due to multi-path propagation, and Free space model predicts that the received power decays as negative square root of the distance. |
| Erceg's suburban fixed model | $PL = H + 10\gamma log_{10}(d/d_0) + X_f + X_h + s$, where, PL is the instantaneous attenuation in dB, H is the intercept and is given by free space path-loss at the desired frequency over a distance of d0= 100m. γ is a Gaussian random variable over the population of macro cells within each terrain category. Xf and Xh are the correlation factors of the model for the operating frequency and for the MS antenna height, respectively.<br>The Erceg pathloss models are classified in three categories (Category A, Category C and Category B) were calculated by the following Equation:<br>PL modified = PL + ΔPLf + ΔPLh<br>Here, PL is the pathloss given earlier, ΔPLf is the frequency term, and ΔPLh his the receive antenna height correction terms given as follows:<br>ΔPLf = 6log10 (f/2000)<br>ΔPLh = -10.8log10 (h/2) for categories A and B<br>= -20log10 (h/2) for categories C | It is based on extensive experimental data collected at 1.9 GHz in 95 macro cells of suburban areas across the United States. This model has a very large cell size, base stations with high transmission power and higher antenna height. Subscriber stations are of very low mobility.<br>The Terrain Type A is a hilly terrain with moderate to heavy tree density, representing rural environments and has highest path loss. Terrain Type B is Characterized by either a mostly flat terrain with moderate to heavy tree density or a hilly terrain with light tree density. Terrain Type C is a flat terrain with light tree density and is associated with the lowest path loss for rural environments. |
| Outdoor-to-indoor and pedestrian pathloss environment | $PL = 40log_{10}R + 30log_{10}f + 49.$ Where PL is the instantaneous attenuation in dB, R is the distance between the base station and the mobile station in kilometers and f is the carrier frequency. | This environment is characterized by small cell size, base stations with low antenna heights and low transmission power are located outdoors while pedestrian users are located on streets and inside buildings and residences |
| Vehicular environment | $PL = 40(1-4*10^{-3}*\Delta h_b)log_{10}R - 18log_{10}\Delta h_b - 21log_{10}f + 80dB.$ where,<br>R is the distance between the base station and the mobile station, f is the carrier frequency and $\Delta h_b$ is the base station antenna height in meters | This environment is characterized by larger cells and higher transmitter power. All subscriber stations have a high mobility |

One of the attractive features of the Medium Access Control (MAC) layer of WiMAX that it is able to construct and transmit fixed and variable-sized frames. The MAC layer has three sublayers, these

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sublayers interact with each other via service access point (SAPs). The service-specific convergence sublayer achieves the mapping or transformation of external network data by the help of the SAP. The common part of the SAP receives information from the MAC Service data units (MSDUs), which are packed into the payload field to form MAC protocol data units (MPDUs). IEEE 802.16 MAC defines up to five separate service classes to provide QoS for various types of applications. The service classes include: Unsolicited Grant Scheme (UGS), Extended Real Time Polling Service (ertPS), Real Time Polling Service (rtPS), Non Real Time Polling Service (nrtPS) and Best Effort Service (BE). Each service class has its own QoS parameters such as the way to request bandwidth, minimum throughput requirement and delay/jitter constraints. These service classes are [16] [17] described as follows:

Unsolicited Grant Service (UGS) is designed for constant bit rate (CBR) real-time traffic flows generating fixed-size data packets such as E1/T1 circuit emulation. It provides a fixed periodic bandwidth allocation. Once the connection is setup, there is no need to send any other requests. The main QoS parameters are maximum sustained rate, maximum latency and tolerated jitter (the maximum delay variation).

Real-time Polling Service (rtPS) is appropriate for variable bit rate (VBR) real-time traffic such as MPEG compressed video. Unlike UGS, rtPS bandwidth requirements vary and so the BS needs to regularly poll each MS to determine what allocations needed to be made. The QoS parameters are similar to the UGS, but minimum reserved traffic rate and maximum sustained traffic rate need to be specified separately.

Extended real-time Polling Service (ertPS) is similar in nature to rtPS service, and is designed to support VoIP with silence suppression. No traffic is sent during the silent periods. ertPS service is similar to UGS in that the BS allocates the maximum sustained rate in active mode, but no bandwidth is allocated during the silent period. There is a need to have the BS poll the MS during the silent period to determine if the silent period has ended. The QoS parameters are the same as those in UGS.

Nonreal-time Polling Services (nrtPS) is appropriate for non-real-time VBR traffic with no delay guarantees. Only the delay minimum rate is guaranteed. File Transfer Protocol (FTP) traffic is an example of applications using this service class.

Best Effort (BE) is appropriate to traffic with weak QoS requests, so most of data traffic falls into this category guarantees neither delay nor throughput. The bandwidth will be granted to the MS if and only if there is a leftover bandwidth from other classes. In practice most implementations allow specifying minimum reserved traffic rate and maximum sustained traffic rate even for this class.

## 3 LITERATURE REVIEW

There have been some works based on performance studies of video streaming over WiMAX networks. Many of these research workers have explored WiMAX in the context of real-time and stored video applications. For example, Pandey et al. [18] developed a model to dimension the network for IPTV service providers that offer VoD services to their customers in heterogeneous environments. The proposed modelling and simulation technique allows us to determine the optimum deployment conditions for a given number of potential IPTV users while satisfying predefined QoE measures. On other hand, Shehu et al. [19] discussed issues regarding challenges for delivering IPTV over WiMAX. These issues include the challenges of QoS requirements. Also, they describe the transmission of IPTV services over WiMAX technology, and the impact of different parameters in WiMAX network when deploying this service. An intelligent controller has been designed based on fuzzy logic to analyze QoS requirements for delivering IPTV over WiMAX in [20] is used to analyze three parameters: jitter, losses and delays that affect the QoS for delivering IPTV services. The aim is to define a maximum value of link utilization among links of the network.

Hrudey et al. [21] used OPNET Simulation to design, characterize, and compare the performance of video streaming to WiMAX and ADSL. The simulation results indicate that ADSL exhibits behaviour approached the ideal values for the performance metrics while WiMAX demonstrates promising behaviour within the bounds of the defined metrics. The work in [22] is extending the work in [21] to include generation and integration of a streaming audio component, also enhances the protocol stack to include the real time protocol (RTP) layer. Network topology is redesigned to incorporate WiMAX mobility. They also include characterization of WiMAX media access control (MAC) and physical (PHY) layer. Simulation scenarios are used to observe the impact on the four performance metrics. Gill et al. [23] used OPNET Simulation to compare the performance metrics between ADSL and WiMAX by varying the attributes of network objects such as traffic load and by customizing the physical characteristics to vary BLER, packet loss, delay, jitter, and throughput. Simulation results demonstrate considerable packet loss. ADSL exhibits considerably better performance than the WiMAX client stations.

Hamodi et al. [24] used OPNET Simulation to design, characterize, and deployment the performance of video streaming to WiMAX under different video codec (SVC, and AVC). The simulation results indicate that SVC video codec is an appropriate video codec for video streaming over WiMAX. The work in [25] is extending the work in [24] to investigate the performance of video streaming over WiMAX under two different terrain environments, namely Free Space, Outdoor to Indoor and Pedestrian. The simulation results indicate that, free space path loss model is a basic path loss model with all other parameters related to terrain and building density set as constant.

Hamodi et al. [26] used OPNET Simulation to investigate the performance of Internet Protocol Television (IPTV) over Fixed Mobile WiMAX system considering different combinations of digital modulation. The performance is studied taking into account a number of key system parameters which include the variation in the video coding, path-loss, scheduling service classes different rated codes in FEC channel coding. The performance was studied in terms of packet lost, packet jitter delay, end-to-end delay, and network throughput. Simulation results show that higher order modulation and coding schemes (namely, 16 QAM and 64 QAM) yield better performance than that of QPSK. However in [27] simulation performance study of Mobile TV over Mobile WiMAX is conducted with different types of adaptive modulation and coding taking into account key system and environment parameters which include the variation in the speed of the mobile, path-loss, scheduling service classes with the adaptive and fixed type of modulations. Simulation results show that dynamic adaptation of modulation and coding schemes based on channel conditions can offer considerably more enhanced QoS and at the same time reduce the overall bandwidth of the system. On the other hand, work in this paper is different than work in [26], [27] whereas this work considering single layer real-time video codecs for deploy Mobile TV over Mobile WiMAX, also this work considering SVC codec to be used in different mobile speeds, and random mobility for mobile users which is be as in reality with respect all fixed and adaptive modulation and coding schemes.

However, many of recent works explore the performance studies of WiMAX under different modulation and coding schemes. For example, Telagarapu et al. [28] analysed the physical layer of WiMAX with different modulation techniques like BPSK, QPSK, QAM and comparison of QPSK modulation with and without Forward Error Correction methods. Singh et al. [29] provided an analysis of the Bit error rate Vs SNR and Block error rate Vs SNR curves using QPSK, 16 QAM and 64 QAM in an AWGN channel with parameters analysed for different code rates. It was found that the greater number of symbols transmitted per block, the transmission quality decreases.

Islam et al. [30] evaluated WiMAX system under different combinations of digital modulation (BPSK, QPSK, 64-QAM and 16-QAM) and different communication channels AWGN and fading channels (Rayleigh and Rician), and the WiMAX system incorporates Reed-Solomon (RS) encoder with Convolutional encoder with ½ and 2⁄3 rated codes in FEC channel coding. Also the work in [31] evaluated and analysed the bit error rate performance of WiMAX Physical layer with the implementation of different concatenated channel coding schemes under various digital modulations over realistic channel conditions. Simulation results demonstrate the outperformance of the concatenated Cyclic Redundancy Check and Convolutional (CRC-CC) code when compared to Reed-Solomon and Convolutional (RS-CC) code and the proposed WiMAX system achieves good error rate performance under QAM modulation technique in AWGN, Rayleigh and Rician fading channels. Whereas Rehman et al. [32] used the Simulink tool to evaluate WiMAX system under different combinations of digital modulation and different communication channels AWGN and fading channels. Both Reed Solomon (RS) encoder with Convolutional encoder with ½ and 2⁄3 rated codes in FEC channel coding are incorporated in WiMAX system.

Bhunia et al. [14] presented an in-depth performance evaluation of mobile WiMAX is carried out using adaptive modulation and coding under the real-like simulation environment of OPNET. They have evaluated the performance parameters of mobile WiMAX with respect to different modulation and coding schemes. Their performance has been evaluated in terms of average throughput, average data-dropped, the MOS value of voice application and the BW usage in terms of UL data burst usage when deployed VoIP on WiMAX Networks. It has been observed that using lower order modulation and coding schemes, the system provides better performance in terms of throughput, data dropped and MOS at the cost of higher BW usage.

## 4 WIRELESS IPTV OVER WIMAX ARCHITECTURE

To provide a proper IPTV (Mobile TV) service to end users, this section describes the architecture of IPTV over Mobile WiMAX. IPTV providers must have an appropriate IP network to guarantee QoS at the services level. The QoS for delivering IPTV services depends especially on network performance and bandwidth. The generic network topology of the IPTV application over WiMAX is given in Figure 1.

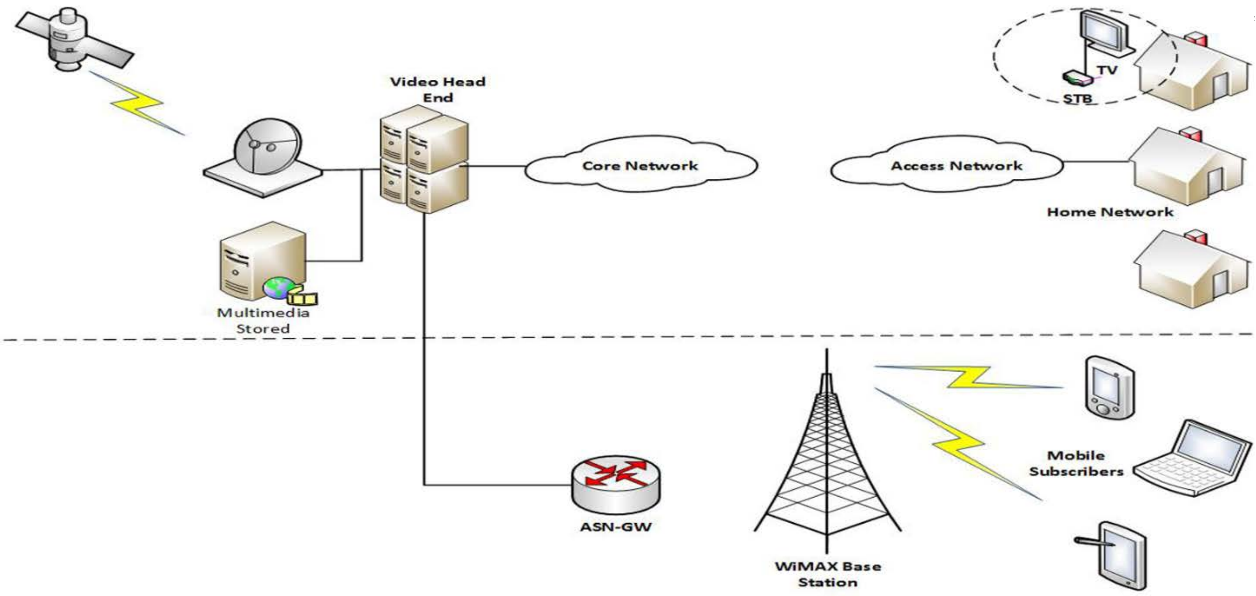



**Figure 1. Deploying IPTV Services to both Fixed and Mobile Subscribers'**

Whereas, the Access Service Network- Gateway (ASN-GW) which is owned by a Network Access Provider (NAP) and comprises one or more base stations and one or more ASN gateway that form the radio access network. The ASN performs the following functions [7]: IEEE 802.16e-based layer 2 connectivity with the Mobile Station (MS); Network discovery and selection of the subscriber's preferred CSN/NSP; AAA proxy: transfer of device, user and service credentials to selected NSP; Relay functionality for establishing IP connectivity between the MS and Connectivity Service Network (CSN); Radio Resource Management (RRM) and allocation based on the QoS policy and/or request from the NSP or the Application Service Provider (ASP); Mobility-related functions, such as handover, location management, and paging with the ASN, including support for mobile IP with foreign agent functionality.

A functional block diagram of an IPTV protocol stack is illustrated in Figure 2. At the video head end in Figure 2 the video streams are encoded and compressed (e.g., MPEG4) from live and stored programs. The MPEG channels are encapsulated as real time transport protocol (RTP) and transported as a UDP or TCP streams to the IP layer. The IP packets are encapsulated into Ethernet frames and then sent over the network through the physical layer. The WiMAX BS receives this data and decapsulates them up the IP layer and re-encapsulates them into specific MAC and PHY PDUs. The physical layer performs FEC, symbol mapping and modulation while the radio transceiver transmits the resulted signals to the mobile nodes. Here, the video streams are sent to the STB or PC decoder and regenerated into video content [3].

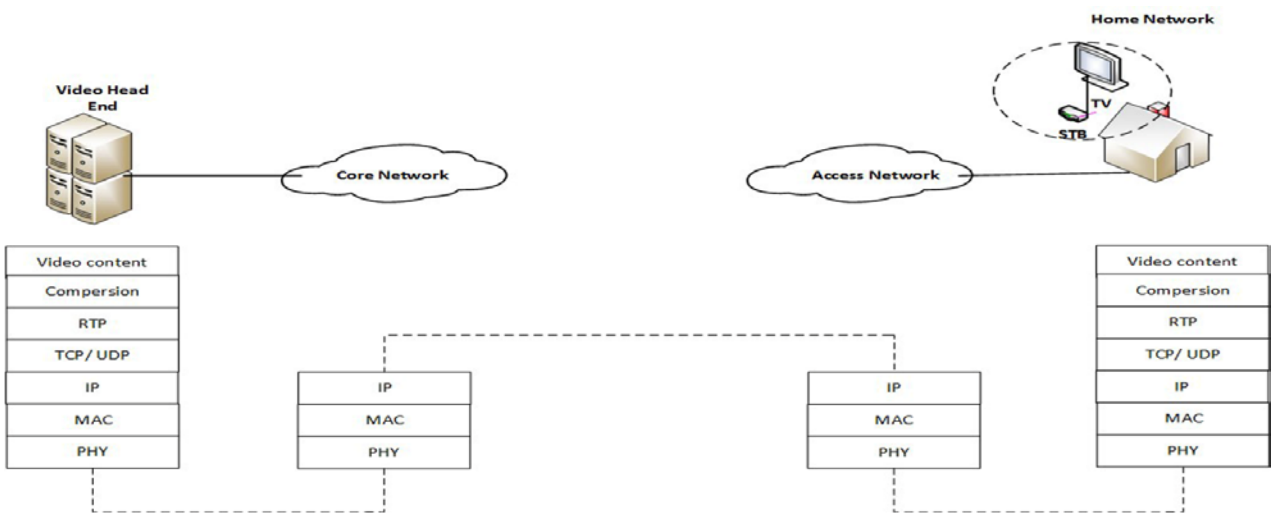



**Figure 2. IP Protocol Stack for IPTV**

IPTV (Mobile TV) architecture over Mobile WiMAX network split into five subsystems, as can be seen in Figure 1: Video Head End is the first subsystem of our model, in this subsystem the servers store video content of any type of movies and audio content source. Different video types stores the video source as national TV broadcasters, local broadcasters, Internet TV operations and any other future video broadcast service. Our video content held in this part has been variable bit rate videos encoded. The video content is delivered to the WiMAX network through the long distance, high capacity content distribution core network. The core network distributes the video flows from the header to the distribution network of the service provider. The distribution network goes from the end of the core network to the aggregation router, where the access network starts. The access network lets the user connect to the service provider and allows access to the multimedia content. The first requirement of an access network is to have enough bandwidth to support multiple IPTV channels for each subscriber, while it allows other services (telephony and data). Finally, the customer network (mobile subscribers) enables communication and information exchange between the subscribers and devices connected to the services offered from service provider.

## 5 VIDEO PERFORMANCE METRICS

Measuring visual quality is a difficult challenging because there are so many factors that can affect the results, that is to specify evaluate and compare video communication. This section discusses some issues related to performance metrics of video transmission. Performance metrics can be classified as objective and subjective quality measures. Objective measures that observe packet transmissions include packet loss, packet delay, packet jitter, and traffic load throughput rates. Other objective metrics that attempt to quantify video quality perceptions include the ITU video quality metric (VQM) and peak signal to noise ratio (PSNR), which measure the codec's quality of reconstruction. Subjective video quality is a subjective characteristic of video quality. It is concerned with how video is perceived by a viewer and designates his or her opinion on a particular video sequence. The main idea of measuring subjective video quality is the same as in the Mean Opinion Score (MOS). QoS requirement is very important for deploying IPTV and VoD as real time services over WiMAX networks. In order to assess the performance of video transmission systems, a suite of relevant performance metrics was identified to appropriately benchmark the system. Video on demand (VoD) deployments over WiMAX is affected by time varying bandwidth, packet delays, and losses. Since users expect high service quality regardless of the underlying network infrastructure, a number of metrics were collectively used to measure the video content streaming performance to ensure compliance and user's quality of experience (QoE) [22]. The following objective measures, widely used in the video content performance analysis employed, are packet loss ratio (PLR), packet delay (PD), packet jitter, and minimum throughput. The performance parameters affecting video have been shown in Table 4 as in [33].

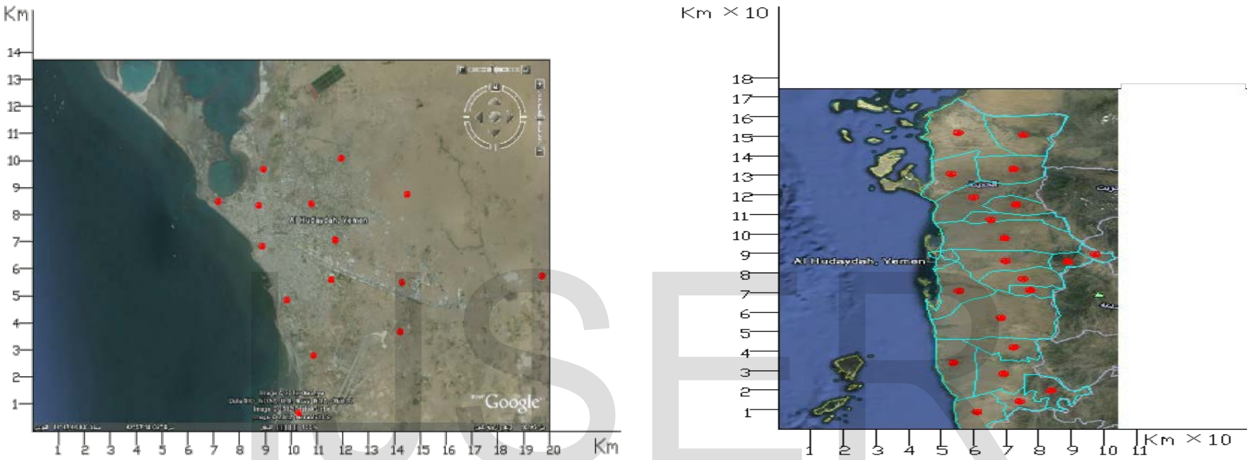
**Table 4. Performance parameters for deploying IPTV**

| Metrics | Mathematical formulation | Description | Acceptable |
|---|---|---|---|
| Packet loss ratio (PLR) | $PLR = \left(\frac{lost_{packet}}{lost_{packet} - received_{packet}}\right)$ | PLR is the corrupted, lost, or excessively delayed packets divided by the total number of packets expected at the video client station. | 10-3 |
| Packet End-to-End Delay (E2E) (ms) | $D_{E2E} = Q\ (d_{proc} + d_{queue} + d_{trans} + d_{prop})$, where:<br>Q is the number of network elements between the media server and mobile station .<br>$d_{proc}$ is the processing delay at a given network element.<br>$d_{queue}$ is the queuing delay at a given network element.<br>$d_{trans}$ is the transmission time of a packet on a given communication link between two network elements.<br>$d_{prop}$ is the propagation delay across a given network link | Packet delay is the average packet transit time between the media server and the video client station. | <400 |

| | | | |
|---|---|---|---|
| Packet delay variation (PDV) or Packet jitter (ms) | $j_{pkt} = t_{actual} - t_{expected}$ , where: $t_{actual}$ is the actual packet reception time. $t_{expected}$ is the expected packet reception time. | Packet jitter is defined as the variability in packet delay within a given media at video client station. | <50 |
| Throughput (bps) | The throughput for variable bit rate (VBR) traffic loading is dynamic in nature and it is a function of the scene complexity and associated audio content. Variable bit rate (VBR) traffic loads is typically quoted as peak throughput ranges. | Throughput is defined as the traffic load that the media stream will add to the network. It can be measured in bits/sec | 221-5311 |

## 6 SIMULATION MODEL

Figure 3 illustrates a typical network topology for WiMAX Base Station (BS) locations of Yemen mobile company in Hodeidah City and Hodeidah Government, Yemen, which are distributed and represented in the Geographical maps as shown as in the Figure 3(a, b). The network shown is realistic and used as a case study only; however, our work presented in this paper can be adopted easily for larger and general networks.




(a). Geographical map of Hodeidah city. (b). Geographical map of Hodeidah government

**Figure 2. Geographical maps of WBS of Yemen mobile company in Hodeidah City and Hodeidah Government, Yemen**

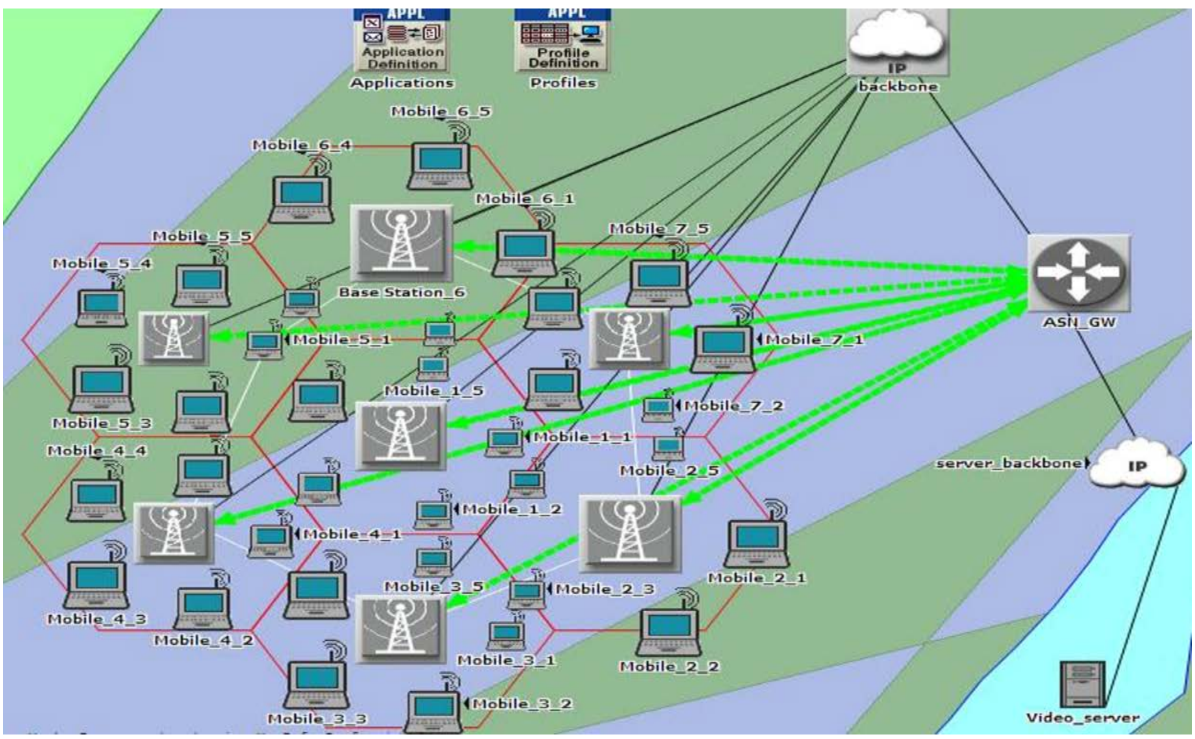



**Figure 4. Model of Mobile TV over WiMAX**

The network topology for our case study network is given in Figure 4 as a snapshot of the OPNET simulation model, which is deployed with 7-Hexagonal celled IEEE 802.16 system using the OFDMA - TDD operating mode with 0.2 Km radius. The SSs were placed in the range of a base station (BS) in circular distribution pattern, were all the stations were in a circle in the edge of the cell; the total bandwidth is 5 MHz, divided into 512 subcarriers. The BS maximum total transmission power is set to 5W with 15 dBi, which is evenly distributed among all subchannels; meanwhile the SS transmission power is 0.5W with -1 dBi. The base stations are connected to the core network by an IP backbone. The IP backbone is connected to the server backbone via an ASN-GW gateway to support the mobility in the WiMAX network. The ASN-GW gateway, IP backbone, and server backbone together represent the service provider company network. We have uses only one SS with varying speed. This node is (mobile 5_1) move along the trajectories indicated by white color around cells. The green bidirectional dotted lines represent the generic routing encapsulation (GRE) tunnels. The common attributes used for network configuration are highlighted in Table 5.

**Table 5: Network Configuration Details**

| Network | Mobile WiMAX Network |
|---|---|
| Cell Radius | 0.2 Km |
| No. of Base Stations | 7 |
| No. of Subscriber Stations | 5 |
| IP Backbone Model | IP32_cloud |
| Video Server Model | PPP_sever |
| Link Model (BS-Backbone) | PPP_DS3 |
| Link Model (Backbone-server Backbone) | PPP_SONET_OC12 |
| Physical Layer Model | OFDM 5 MHz |
| Traffic Type of Services | Streaming Video |
| Application | Real Video streaming |
| Scheduling | rtPS |

Video streaming over wireless networks is a challenging task. This is due to the high bandwidth required and the delay sensitive nature of video more than most other types of application. Variable bit rate (VBR) video traffic models which accurately present the traffic characteristics and statistical properties of real video, emerged as attractive alternative to overcome the drawbacks of CBR. Such as, it is costly, inefficient, and large delays [34]. As a result, a VBR video traces of 74 minutes Tokyo Olympics movie encoded by different codec: MPEG-4 part 2, H.264/AVC, and Scalable Video Coding (SVC) single layer is used in our simulation. This movie traces with different coding obtained from Arizona State [35], [36] with CIF $[352 * 288]$ frame resolution, Group of Picture (GOP) size is selected as 16 for this video for all codes, and encoded with 30 frames per seconds (fps). This work also adds audio frames, which is 21.6 as in [22]. Table 6 shows the mean and peak rates for these video codes of the Tokyo Olympics movie.

Two independent instances of the video conferencing application are used to stream the separate and distinct video and audio components of the Tokyo Olympics movie. These two applications configured to work simultaneously stream in the profile configuration. The key parameters of this application configuration are the frame inter-arrival time and frame size. The incoming inter-arrival times are configured to the video and audio frame rates of 30 and 21.6, respectively. It should be noted that the outgoing inter-arrival time remains set to ''none'' in order to achieve unidirectional streaming from the video server. Furthermore, the frame size parameters are configured to explicitly script the video and audio traces.

**Table 6. Video codecs traces characteristics**

| Parameters | Tokyo Olympics movie | | |
|---|---|---|---|
| Codec | MPEG-4 Part 2 | H.264 SVC | H.264 AVC |
| Frame Compression Ratio | 151.6 | 18.015 | 21.709 |
| Min Frame Size (Bytes) | 8 | 22 | 17 |
| Max Frame Size (Bytes) | 13992 | 58150 | 62269 |
| Mean Frame Size (Bytes) | 1003.01 | 8440 | 7004 |
| Peak Frame Rate (Mbps) | 3.36 | 13.956 | 14.945 |
| Mean Frame Rates (Mbps) | 0.24 | 2.0258 | 1.68 |
| Frame Rate (frames/seconds) | 30 | 30 | 30 |

## 7 RESULTS AND DISCUSSION

Series of simulation scenarios were simulated and the results are collected and summarized in different scenarios depending on varying speed of the mobile node, varying path loss models, and for different types of service classes. For each case, the types of modulation and coding schemes are chosen one at a time to obtain one set of simulation results for the different performance measures of video packet jitter, video packet end-2-end delay, SS node WiMAX packet drop, and traffic load throughput.

### 7.1 Scenario 1: Different video codec of video application

This subsection shows the simulation results of series simulation scenarios under this scenario. Each scenario used different video codec under different modulation and coding scheme in each cell. A path-loss model is chosen as free space and service class as rtPS are considered, and kept constant. This simulation is used to evaluate the performance parameters, namely: packet jitter, packet E2E delay, WiMAX packet drop of the mobile node, and throughput of the mobile node.

The average packet jitter, and average E2E delay with various modulation and coding schemes are shown in Figures 5(a) –5(b). Figure 5(a) shows the average variation of jitter for audio/video IPTV over Fixed WiMAX networks. For different coding, video quality is best if the jitter is zero. As shown in Figure 5(a), average audio/video jitter is approximately zero for higher modulation and coding scheme (MCS) (16 QAM, and 64 QAM) whereas QPSK has a worse average variation of jitter for Movie coded by AVC codec. From the results in Figure 5(a), it is observed that WiMAX using higher MCS (16 QAM, 64 QAM) as a modulation technique shows better jitter compared with other MCS (QPSK). It is also observed that video coded by SVC, and MPEG-4 has a better average jitter compared with the AVC codec. Therefore, video codec by SVC is the best for deploying IPTV. Average End-to-End delay for different video codec under MCS is shown in Figure 5(b), as it can be seen that the average E2E delay of different video codec gives lower packet E2E delay for audio/video IPTV when codec by SVC, and MPEG-4 that under all modulation and coding MCS.

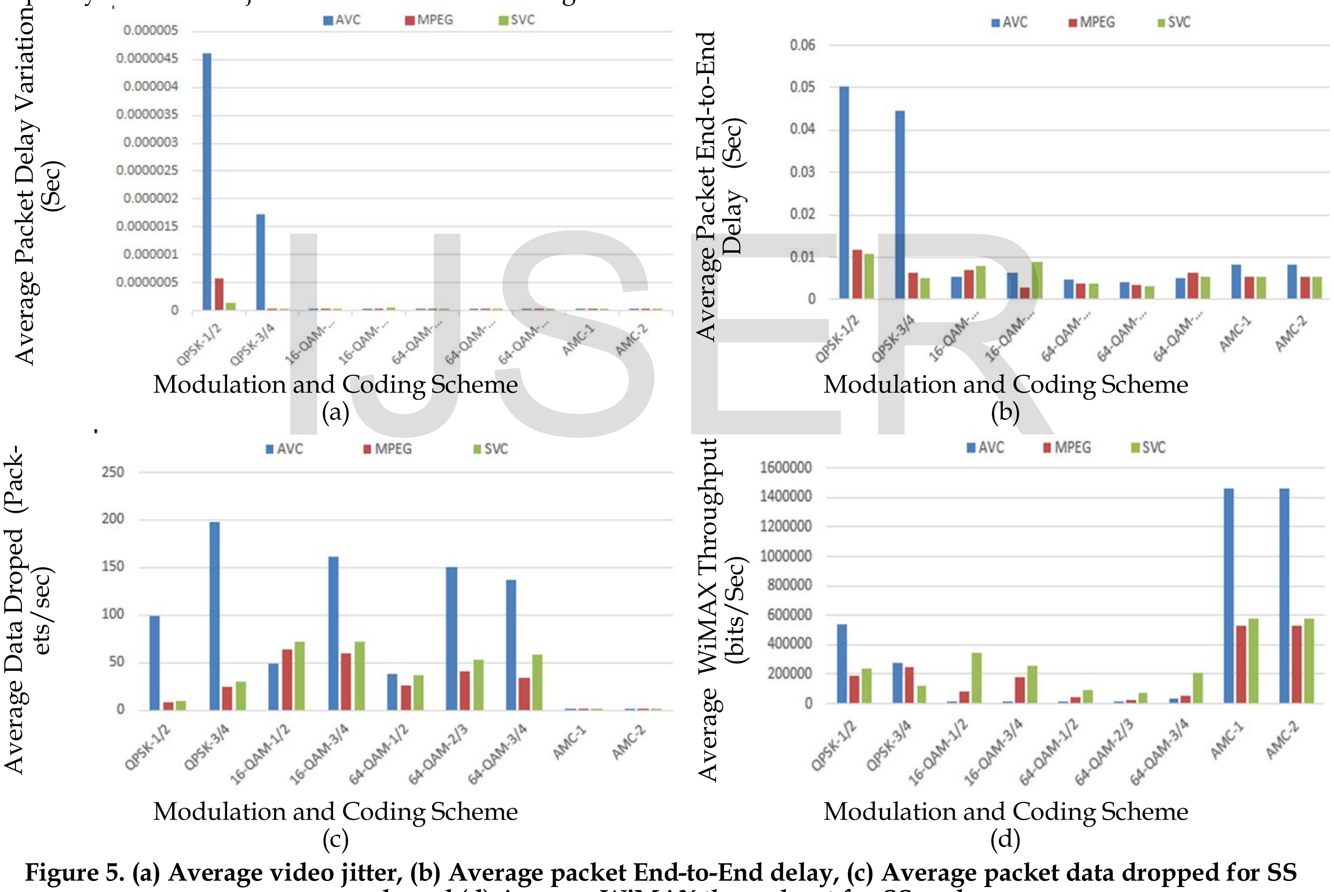



**Figure 5. (a) Average video jitter, (b) Average packet End-to-End delay, (c) Average packet data dropped for SS node and (d) Average WiMAX throughput for SS node**

As shown in Figure 5(c), the average data drop is significantly higher when video codec by AVC codec. The effect of data drop naturally decreases the average WiMAX throughput as shown in Figure 5(d). From Figure 5(c) it is observed that the data dropped is very low for SVC video codec for all modulation and coding schemes. Whereas, the other different video codec (AVC, and MPEG-4) has more data dropped. Figure 5(d), shows the average subscriber station (SS) WiMAX throughput. As we can see, the average throughput for SVC is higher compared with its data dropped as shown in Figure 5(c). Whereas, another codec has more throughput but also has more data dropped. According to the results as in Figures 5, it is observed that SVC codec is the best codec used to deploy IPTV over WiMAX, which has better performance (high throughput, low data dropped, and low E2E delay with low variation jitter

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delay) under all modulation techniques compared with the other video codec. In conclusion, transmitting SVC encoded videos over WiMAX networks is an effective solution for deploying IPTV.

### 7.2 Scenario 2: Mobile node with different speed

This scenario is investigated how mobile TV speeds perform using different modulation and coding schemes offered by Mobile WiMAX. Average throughput, drop, end to end delay, and jitter for SVC video are evaluated below when considering different modulation and coding schemes, and showed which the appropriate speed is the best for Mobile TV. The speed of the mobile is provided in kmph while path-loss model and scheduling service are kept constant. Path-loss model is chosen as free space and service class as rtPS is considered.

Average variation of jitter for audio/video Mobile TV over Mobile WiMAX networks for different speed has been shown in Figure 6(a). As in author's knowledge, video quality is best if the jitter is zero. Average audio/video variation of jitter is approximately zero for higher MCS and AMC under different speeds which is about 2.5753E-05 seconds, whereas other MCS as QPSK has a more average variation of jitter. From the results in Figure 6(a), it is observed that WiMAX using higher MCS (16 QAM, 64 QAM) and AMC (AMC-1, AMC-2) as a modulation technique shows better jitter compared with other MCS (QPSK). Average End-to-End delay for different speeds under MCS shows in Figure 6(b), as we can see the average E2E delay of different speeds give lower packet E2E delay for audio/video Mobile TV under higher modulation and coding MCS (16 QAM, 64 QAM) and AMC compared with QPSK.

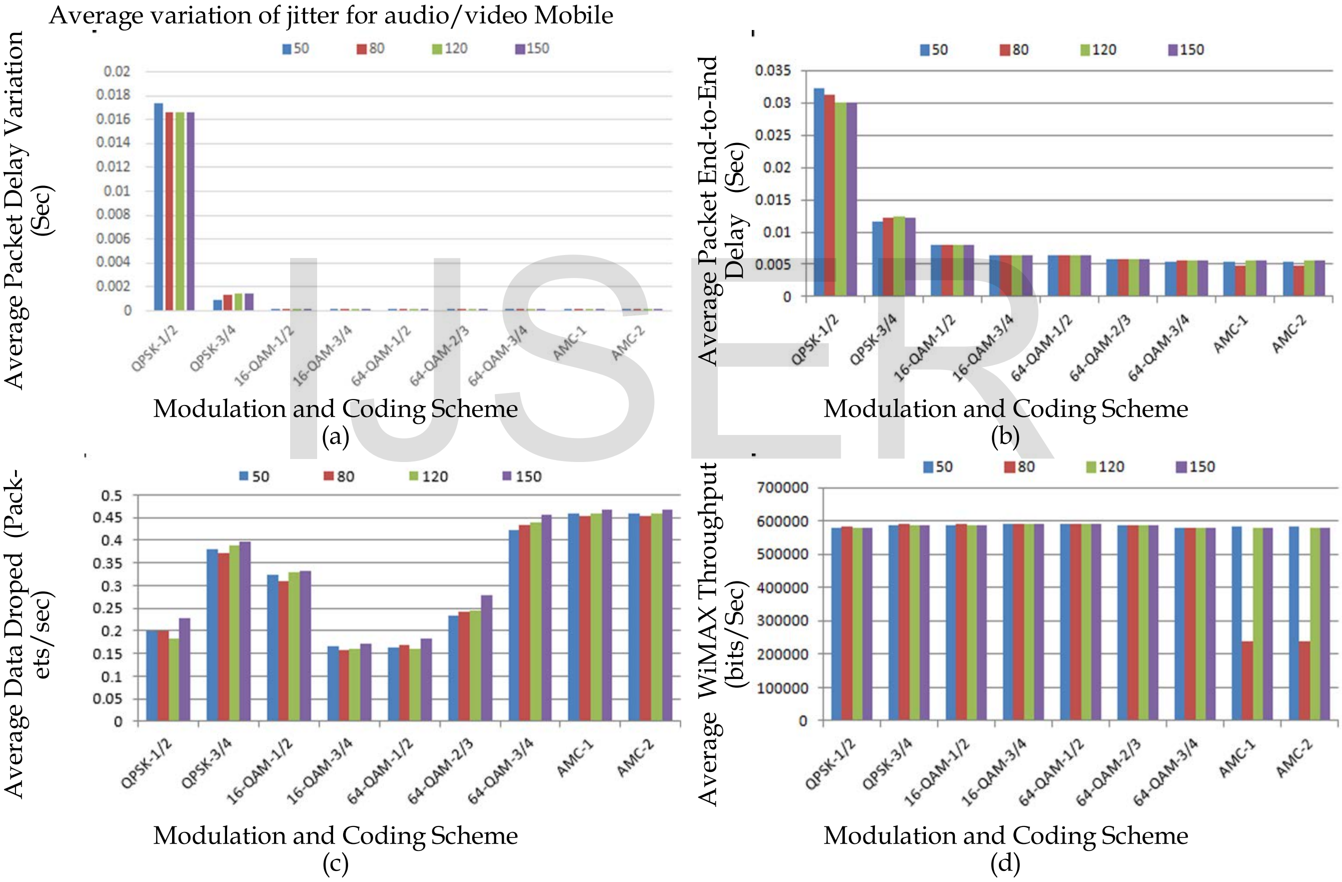


**Figure 6. (a) Average video jitter, (b) Average packet End-to-End delay, (c) Average packet data dropped for SS node and (d) Average WiMAX throughput for SS node**

Increasing data drop is affected by the speed of SS. However, when speed of SS increases, the hand-off frequency increases, which results in increased data dropped and thereby decreased throughput. As can be viewed from Figure 6(c), the average data drop is significantly higher when SS moves with a greater speed as 150 kmph. The effect of data drop naturally decreases the average WiMAX throughput as shown in Figure 6(d). From Figure 6(c) it is observed that the data dropped is very low for 16 QAM 3/4, 64 QAM ½, and 64 QAM 2/3 modulation schemes and varies with speed. We know that a higher order modulation scheme is more sensible to SNR. As the SS is moving through the cell, it faces a different SNR value depending upon the distance from the BS and the propagation environment. With increasing distance, the SNR decreases and the higher order MCS gives more BLER than the lower order

MCS for the same SNR value. For that the higher order 64 QAM ¾ has more data dropped and AMC modulation schemes with varying speeds. The average throughput is taken as a measure that will give the average of observed throughput right through the simulation. Thus, the higher order of MCS 16 QAM ¾, 64 QAM ½, and 64 QAM 2/3 has the best performance (throughput) as observed in Figure 6(d). According to the results as in Figures 6, it is observed that WiMAX uses 64 QAM ½ and 16 QAM ¾ as modulation techniques show better performance (high throughput with low data drop) compared with other modulation techniques.

### 7.3 Scenario 3: Mobile node with random mobility

Average throughput, drop, end to end delay, and jitter for SVC video are compared below when considering different modulation and coding schemes. Average throughput in the MAC (Medium Access Control) layer of Base Station node (BS) is compared when considering different modulation and coding schemes in Figure 7(d), average drop packet in PHY (Physical Layer) is compared in Figure 7(c), average End-2-End delay is compared in Figure 7(b), and average of jitter delay also is compared in Figure 7(a). This scenario is investigated how mobile TV speeds perform using different modulation and coding schemes offered by Mobile WiMAX. A path-loss model is chosen as free space and service class as rtPS are considered, and kept constant.

Figure 7(a) shows the average variation of jitter for audio/video Mobile TV over Mobile WiMAX networks for random mobility speed, video quality is best if the jitter is zero. As shown in Figure 7(a), average audio/video jitter is approximately zero for all MCS and AMC unless QPSK ½ and 64 QAM ½ , for that maybe the speed and the path in MS which is changed randomly is be poor channel quality so the variation of jitter is be more. Average End-to-End delay for different random mobility under MCS shows in Figure 7(b), as we can see the average E2E delay of random mobility speeds give lower packet E2E delay for audio/video Mobile TV under higher modulation and coding MCS (16 QAM, 64 QAM) and AMC compared with QPSK unless in 64 QAM ½ and 16 QAM ½ and that for mobile randomly.

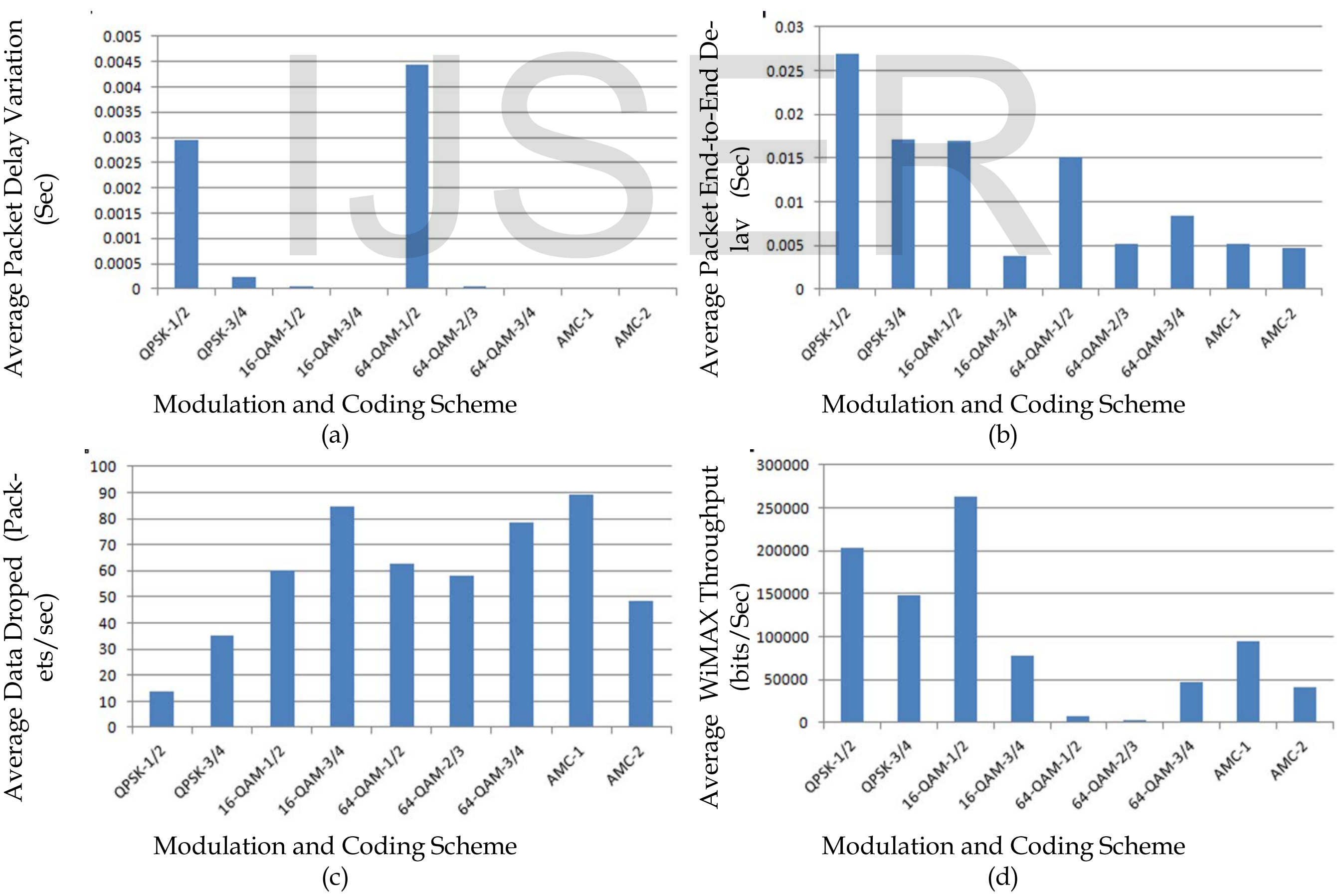


**Figure 7. (a) Average video jitter, (b) Average packet End-to-End delay, (c) Average packet data dropped for SS node and (d) Average WiMAX throughput for SS node**

Intuitively, increasing data drop is affected by the speed and randomly of SS. However, when speed of SS increases, the hand-off frequency increases, which results in increased data dropped and thereby decreased throughput. As can be viewed from Figure 7(c), the average data drop is significantly higher when SS moves randomly under higher modulation and coding schemes and adaptive modulation. The effect of data drop naturally decreases the average WiMAX throughput as shown in Figure 7(d). From Figure 7(c) it is observed that the data dropped is very low for QPSK modulation schemes and varies with mobile randomly speed. For that the higher modulation and coding schemes has more data dropped and AMC modulation schemes with varying random mobility speeds. So, the average throughput is taken as a measure that will give the average of observed throughput right through the simulation. Thus, the QPSK and 16 QAM ½ has the best performance (throughput) as observed in Figure 7(d). According to the results as in Figures 7(c) - 7(d), it is observed that WiMAX uses QPSK ¾ and 16 QAM ½ as modulation techniques show better performance (high throughput) with low variation jitter and E2E delay compared with other modulation techniques.

### 7.4 Scenario 4: Mobile node with different path loss

This scenario is investigated how mobile TV perform using different modulation and coding schemes offered by Mobile WiMAX with respect to various path-loss models. Average throughput, data drop, end to end delay, and jitter for SVC video are evaluated when considering different modulation and coding schemes, and showed which the appropriate path loss is the best for Mobile TV. The speed of the mobile is chosen as 50 Kmph while scheduling service are kept constant as rtPS is considered.

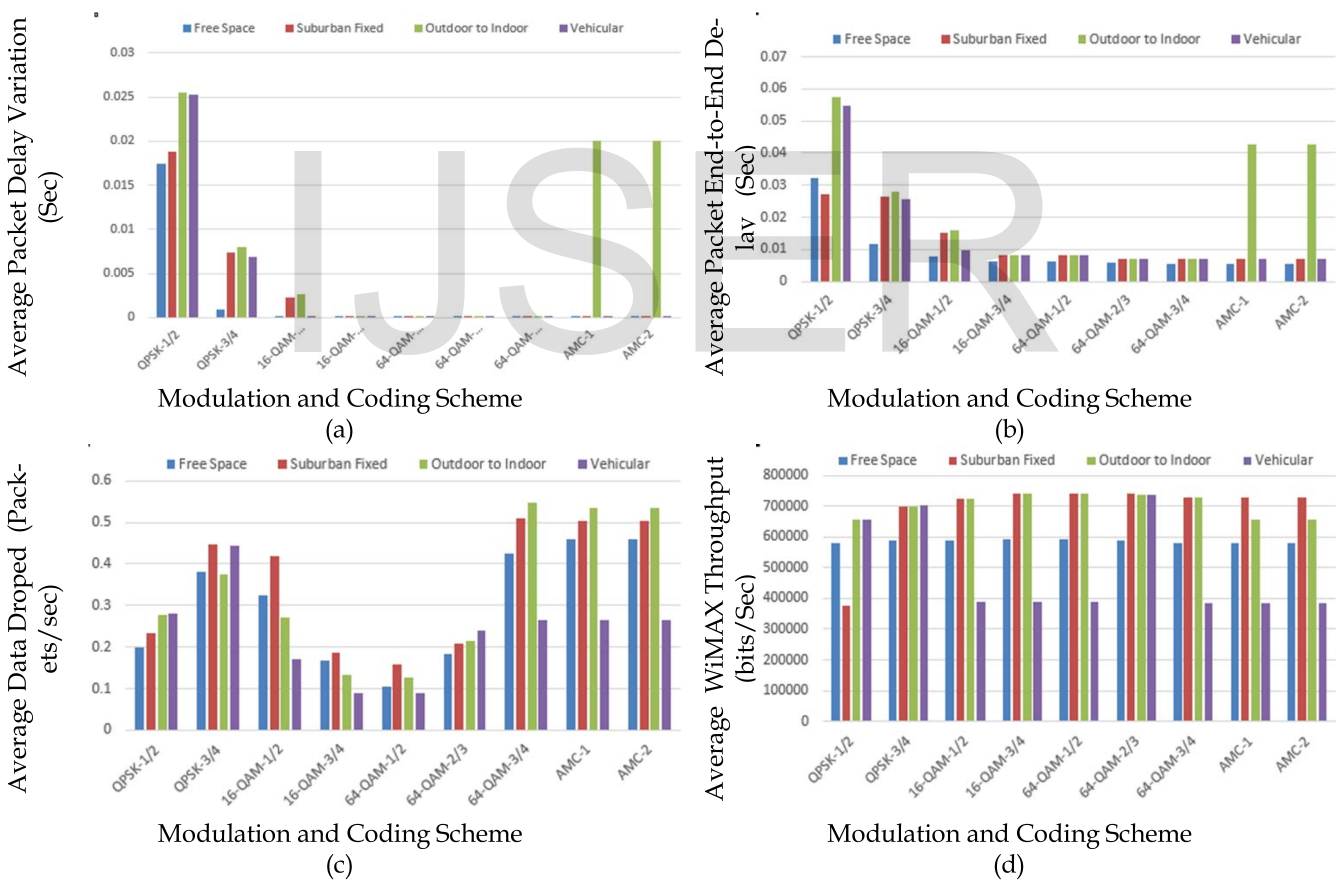


**Figure 8. (a) Average video jitter, (b) Average packet End-to-End delay, (c) Average packet data dropped for SS node and (d) Average WiMAX throughput for SS node**

In this study fixed radius WiMAX networks are considered for all the path loss. As in authors' knowledge, outdoor to indoor and pedestrian path-loss model is designed for small and micro cell WiMAX network. For outdoor to indoor and pedestrian propagation model, as SS node that moves away from BS will encounter a significant drop in SINR and as the higher order MCS (such as 64 QAM 3/4) requires high value of SINR to give a good throughput; higher order MCS will face a very large amount of data drop as revealed from Figure 8(c). However, outdoor to indoor and pedestrian has lower packet jitter and lower packet E2E delay as shown in Figures 8(a) -(b). Path-loss of free space is lowest; hence, the reduction of SINR with the distance from BS is less which leads to better throughput, lower packet jitter, lower packet E2E delay, and lower packet data dropped from all varies MCS as shown in Figures 8. Again, as the reduction of SINR with distance from BS is less, the SS has to change its modulation scheme less frequently which results in very high throughput for AMC as shown in Figure 8(d). Here, the suburban fixed model have been considered as hilly terrain with moderate to heavy tree density that representing rural environments and has highest path-loss. Also, same terrain is considered here for vehicular model. As outdoor to indoor and pedestrian propagation model experiences a very high packet drop compared with the others, it gives the lowest throughput compared with other propagation model as can be observed from Figure 8(d). For free space propagation model, 16 QAM, and 64 QAM has the same throughput under varied coding as in Figure 8(d).

### 7.5 Scenario 5: Mobile node with different classes

This scenario is investigated how mobile TV perform using different modulation and coding schemes offered by Mobile WiMAX with respect to various service classes; speed and path loss kept constant. The speed of SS is chosen as 50 kmph while path-loss model as free space is considered. Different service classes are used in this case under various modulations and coding scheme to obtain the performance metrics: packet delay variation, packet End-to-End delay, data drooped for a mobile node, and WiMAX throughput for the mobile node.

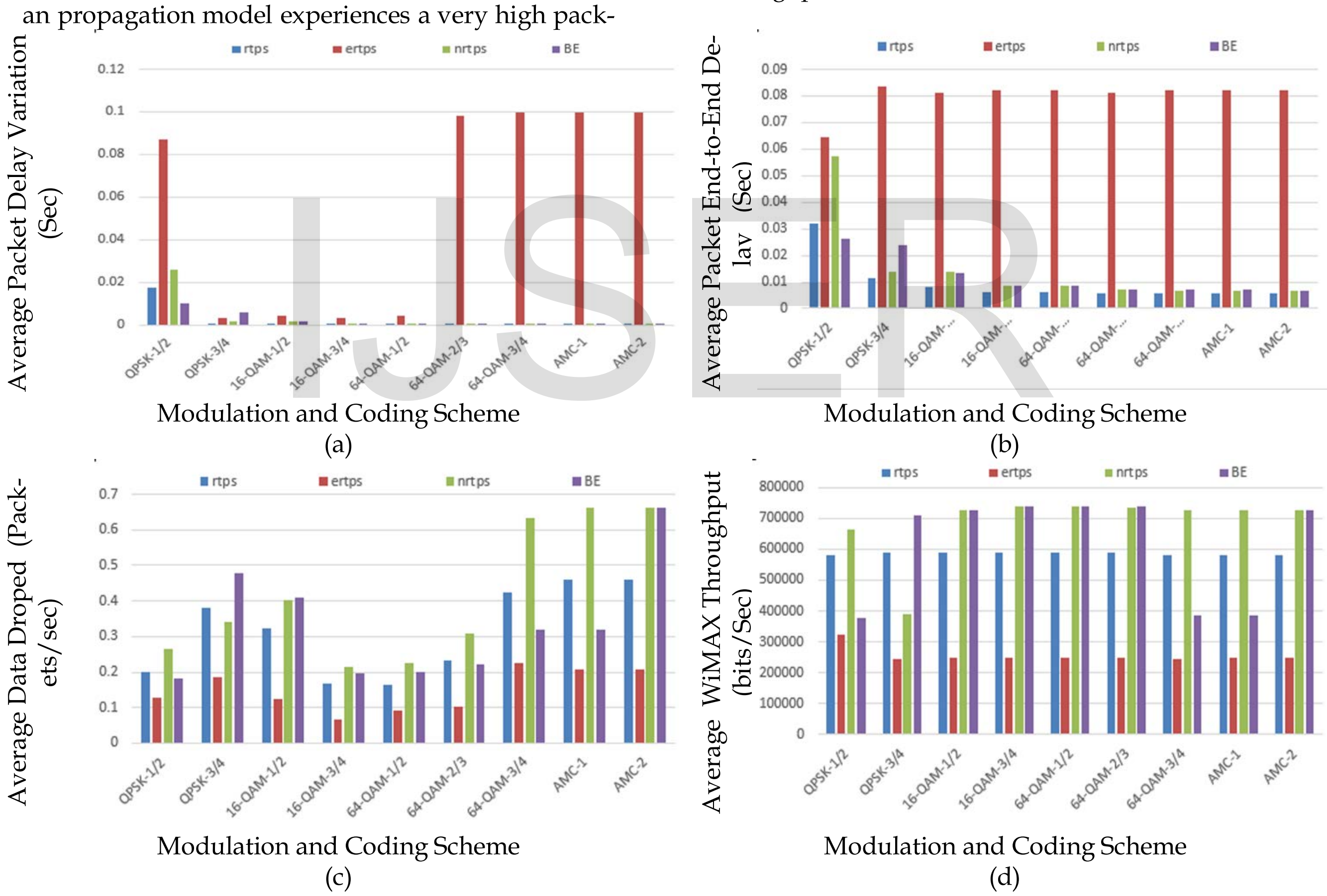


**Figure 9. (a) Average video jitter, (b) Average packet End-to-End delay, (c) Average packet data dropped for SS node and (d) Average WiMAX throughput for SS node**

To the best of the authors, UGS and ertps were designed to support VoIP [37]. However, UGS is used for CBR [38], whereas, A/V Mobile TV has VBR, it can't use UGS in our simulation. So, UGS classes are not supported in our case study. On the other hand, ertps are designed to support real-time applications generating variable bit rate traffic, and are designed for VoIP service with activity detection. Results obtained shows that this class has worse performance in all metrics as shown in Figures 9 it has lower throughput, higher jitter, and higher packet E2E delay. Figures 9(a) -(b) show packet delay jitter and packet E2E delay for different service classes: rtPS, nrtPS, ertPS, and BE. rtps, nrtps, and BE has given the best performance, which reveals that for higher modulation and coding, all service classes give equal throughput, equal packet jitter, and equal packet E2E delay. In Figure 9(d), it can be observed that rtps classes' providers have greater throughput than other classes, nrtPS, and BE. The reason is that rtPS is designed for streaming Audio or Video.

## 8 CONCLUSION

This paper presents a comparative performance study of Mobile TV over mobile WiMAX network with respect to different modulation and coding schemes under different parameters including speed of mobile node, path-loss models, and MAC service classes. The performance has been evaluated in terms of average packet jitter, average packet E2E delay, average throughput, and average data-dropped. It has been demonstrated, using OPNET simulation, the simulation results indicate that, the H.264/SVC video codec has been found to offer improved visual quality and appropriate codes for delivering video compared to the preceding standards. Furthermore, the higher order modulation and coding schemes (namely, 16 QAM and 64 QAM) provide better performance. Also, it has been shown that AMC schemes (namely, AMC-1 and AMC-2) give relatively comparable performances to 64 QAM ¾ scheme. In this work, simulation results show that the free space path loss is best for deploying A/V video application over different mobile node speeds, whereas outdoor to indoor is the worst case with the highest packet drop rate. Moreover, our simulation showed that rtPS scheduling service class is the most appropriate scheduling service for A/V video application. This work has limitations to certain assumptions like: Station transmit power, station antenna gain, carrier operating frequency and channel bandwidth.

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