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LOW LATENCY IN MULTIMEDIA COMMUNICATION THROUGH WIRELESS NETWORK

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ABSTRACT

Multimedia services over mobile networks pose several challenges, such as the efficient management of radio resources or the latency induced by network delays and buffering requirements on the multimedia players. In Long Term Evolution (OFDM) networks, the definition of multimedia broadcast services over a common radio channel addresses the shortage of radio resources but introduces the problem of network error recovery. In order to address network errors on OFDM multimedia broadcast services, the current standards propose the combined use of forward error correction and unicast recovery techniques at the application level. This project shows how to efficiently synchronize the broadcasting server and the multimedia players and how to reduce service latency by limiting the multimedia player buffer length. This is accomplished by analyzing the relation between the different parameters of the OFDM multimedia broadcast service, the multimedia player buffer length, and service interruptions. It is simulated to confirm how the quality of the multimedia service is improved by applying our proposals.

I . INTRODUCTION

Wireless communication is without a doubt a very desirable service as emphasized by the tremendous growth in both cellular and wireless local area networks (WLANs) (primarily, the ones that are compliant with the IEEE 802.11 family of standards, popularly known as Wi-Fi). However, these two radically different technologies address only a narrow range of connectivity needs, and there are numerous other applications that can benefit from wireless connectivity. The cellular networks offer wide area coverage, but the service is relatively expensive and offers low data rates: even the third generation of cellular networks (3G) offers (at best) low data rates (_2Mbps) compared to WLANs (>50Mbps for IEEE 802.11a and 802.11g and _100Mbps for proprietary solutions at the time of this writing). On the other hand, the WLANs have rather limited coverage (and the associated reduced mobility). Furthermore, in order to increase the coverage of WLANs, a wired backbone connecting multiple access points is required.

Wireless metropolitan area networks (WMANs) (e.g., the family of IEEE 802.16 standards), partially bridges this gap, offering high data rates with guaranteed quality of service to a potentially large customer base (up to tens of miles from the base station). The main drawback of WMANs is their (current) lack of mobility support and the line of sight (LOS) requirement: if a customer does not have a clear LOS to the WMAN base station, it is unlikely that he can receive service. In communities with a high density of obstructions (high-rise buildings or trees), more than half of the customers cannot be served due to the LOS requirement. Furthermore, the base stations tend to be complex and expensive.

Wireless mesh networks (WMNs) have the potential to eliminate many of these disadvantages by offering low cost, wireless broadband Internet access both for fixed and mobile users.

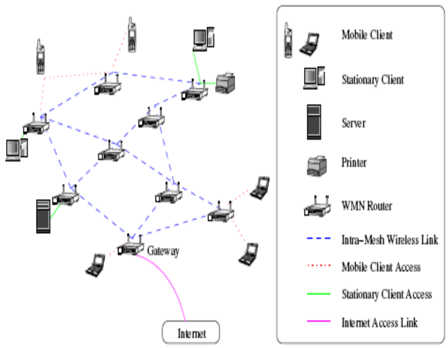

Fig 1. A wireless mesh network interconnecting stationary and mobile clients

In its most general form (see Fig. 1), a wireless mesh network (WMN) interconnects stationary and/or mobile clients and optionally provides access to the Internet. The defining characteristic of a WMN is that the nodes at the core of the network are forwarding the data to and from the clients in a multihop fashion, thus forming a (mobile) ad hoc network (MANET).

Beyond the multihop requirement, there are no other restrictions on the design of a WMN, resulting in considerable flexibility and versatility. This versatility allowed many players to enter the mesh networking arena with different products and applications. For example, the Internet access link in Fig. 1 can be wired (e.g., T1, Ethernet, etc.), wireless (point to point or point to multipoint), or be absent. Some WMN technologies are designed for high speed mobility (100mph), some for casual roaming in a building, while others are only meant to be used by stationary clients.

The wireless links used to connect the mobile clients can be of the same type as the intra-mesh wireless links or can be a completely different technology. (They can also be missing altogether.) Many implementations allow mobile nodes to connect to the WMN while in its range; their packets are forwarded in the same multihop manner as the ones of the stationary nodes (and in their turn, although not always preferable, the mobile nodes can forward packets on behalf of other nodes). Not all nodes have to support client nodes; the service provider can employ several relay nodes to increase the coverage of the network (or to improve its performance, as the relays can allow some clients to reach their destinations in fewer hops). Figure 1 is intentionally vague on the application scenario; both indoor and outdoor (even mixed scenarios) are specifically targeted by different companies.

Objective of this project, to efficiently synchronize the broadcasting server and the multimedia players. To reduce service latency by limiting the multimedia player buffer length. It's analyzed the relation between the different parameters of the LTE multimedia broadcast service, the multimedia player buffer length, and service interruptions. To improve the quality of the multimedia service.

II .RELATED WORK

Lohmar and Einarsson, et al.,[1] propose the new DASH specification with a special focus on Live streaming services. In order to re-use existing web content distribution schemes, the new streaming technique provides the live stream as a sequence of files, which are continuously downloaded by the streaming client. This way of streaming introduces new delay components into the system streaming. Live Streaming technology is often used for events like sport events to allow other users to virtually participate. It is generally preferred to minimize the end-to-end delay for live services.

Truong, et al.,[2] present the use of MPEG DASH standard to stream audiovisual content. We first describe a novel estimation method for connection throughput. Then, employing the extensibility feature of DASH syntax, we present a systematic method for selecting the best audio and video alternatives given the estimated throughput.

M. Seufert et al.,[3] to derive the QoE influence factors that emerge as a result of adaptation. The main contribution is a comprehensive survey of QoE related works fromhuman computer interaction and networking domains, which are structured according to the QoE impact of video adaptation. To be more precise, subjective studies that cover QoE aspects of adaptation dimensions and strategies are revisited.

D. Lecompteet al., [4] describes the relevant use cases for eMBMS in terms of service. It then gives a tutorial on eMBMS, in particular highlighting the evolution over MBMS.

Garcia, and Dytko, et al., [5] investigate the quality impact due to initial loading, stalling, and video bitrate for High Definition (HD, 1920×1080 pixels) audiovisual sequences. The analysis is based on the results of three audiovisual subjective tests conducted in a laboratory environment.

III .SYSTEM IMPLEMENTATION

OFDM (Orthogonal Frequency Division Multiplexing)**:**Popular technique for transmission of signals over wireless channels. Converts a frequency-selective channel into a parallel collection of frequency-flat sub channels. Subcarriers have minimum frequency separation required to maintain orthogonally of their time domain waveforms. Signal spectra of the different subcarriers overlap in frequency.

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The Transmitter can adapt its signaling to match the channel if knowledge of channel condition is available at transmitter. Adaptive strategies in OFDM can approach water pouring capacity of frequency-selective channels. In practice this is achieved by using adaptive bit loading techniques on N subcarriers. Time duration of n OFDM symbol is N times larger than that would correspond to a single-carrier system. OFDM modulator can be implemented as an inverse fast Fourier transform (IFFT) followed by a DAC. Each block of N IFFT coefficients is preceded by a cyclic prefix (CP) to mitigate ISI caused by channel time spread. The receiver can use fast signaling processing transforms such as FFT for OFDM implementations

With the ever growing demand of this generation, need for high speed communication has become an utmost priority. Various multicarrier modulation techniques have evolved in order to meet these demands, few notable among them being Code Division Multiple Access (CDMA) and Orthogonal Frequency Division Multiplexing (OFDM). Orthogonal Frequency Division Multiplexing is a frequency – division multiplexing (FDM) scheme utilized as a digital multi – carrier modulation method. A large number of closely spaced orthogonal sub – carriers is used to carry data. The data is divided into several parallel streams of channels, one for each sub – carriers. Each sub – carrier is modulated with a conventional modulation scheme (such as QPSK) at a low symbol rate, maintaining total data rates similar to the conventional single carrier modulation schemes in the same bandwidth. Orthogonal Frequency Division Multiplexing is a special form of multicarrier modulation which is particularly suited for transmission over a dispersive channel. Here the different carriers are orthogonal to each other, that is, they are totally independent of one another. This is achieved by placing the carrier exactly at the nulls in the modulation spectra of each other.

Source bit stream encoded by FEC encoder. Coded bitstream mapped to a constellation by digital modulator, and encoded by MIMO encoder. Each of parallel output symbol stream corresponding to a certain Transmitter antenna follows the same Transmission process: Insertion of pilot symbols. Modulation by inverse FFT. Attachment of CP and Preamble. Finally, data frame is transferred to IF/RF stage for Transmission. The received symbol stream from different Rx antennas are fist synchronized. Preambles and CPs are extracted from Rx symbol stream. Remaining OFDM symbols demodulated by FFT. Frequency pilots are extracted from the demodulated OFDM symbols, and are used for channel estimation. Estimated channel matrix aids the MIMO decoder. Estimated Transmitted symbols are demodulated and decoded.

In the method of generating an OFDM symbol, initially, N input complex symbols are padded with zero's to get Ns symbols which are used to calculate the IFFT.The output of the IFFT is the basic OFDM symbol. Based on the delay spread on the multi-path channel, a specific guard-time is chosen .A number of samples corresponding to this guard time is taken from the beginning of the OFDM symbol and then added at the end of the symbol. Interleaver arranges the input data in a random order so the consecutive data's are spaced apart. It is then given to IFFT and cyclic prefix is added to eliminate the intersymbol interference and it also acts as a guard interval. Now the parallel data is converted into serial data and the cyclic prefix is removed and is sent to FFT where the values are computed easily. It is again sent to deinterleaver and demapper to perform the reverse operation of mapper and interleaver. Interleaving is a technique that is used in communication systems to overcome correlated channel noise such as burst error or fading. The Interleaver arranges the input data in a random manner so that the consecutive data that is pre- sent is spaced apart. Finally the parallely received data is converted into serial data and achieves its original form that is the original signal is obtained similarly, the same number of samples is taken from the end of the each OFDM symbol and inserted at the beginning. The OFDM symbol is multiplied with the raised cosine window to remove the power of the out-of-band sub-carriers. The windowed OFDM symbol is then added to the output of the previous OFDM symbol with a delay of Transmitter, so that there is an overlap region of Transmitter between each symbol which causes inter carrier interference

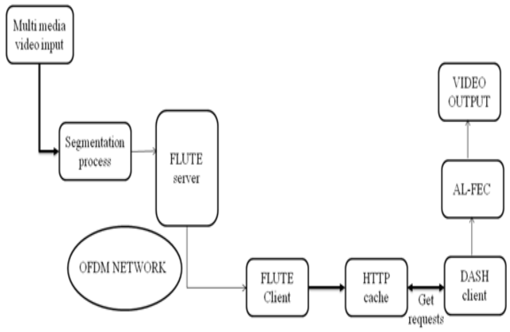

Fig 3.1. Hybrid FLUTE/DASH architecture. The figure shows the components involved in the delivery ofDASH segments over OFDM in combination with unicast error recovery using HTTP.

Fig. 3.1 shows the architecture proposed by the 3GPP to support OFDM multimedia broadcast services. DASH, initially designed to work as a multimedia streaming solution over HTTP, has been adapted for multicast streaming services over OFDM using the eMBMS channel. However, even if the use of eMBMS allows sending the same multimedia segment to multiple receivers using a common channel, the eMBMS channel is unidirectional and unreliable, so recovery mechanisms must be used to ensure a correct delivery of the DASH segments over eMBMS.

The 3GPP has proposed both application and physical level mechanisms to overcome the lack of reliability in eMBMS. At the physical level, the 3GPP proposes the use of Physical Layer Forward Error Correction (PHY-FEC) techniques to protect data against errors during its transmission over the wireless OFDM channel. Errors are also mitigated by the use of Multicast/Broadcast over Single Frequency Network (MBSFN). MBSFN is a transmission mode that improves the Signal to Interference plus Noise Ratio (SINR) of the multicast receivers by defining how several base stations cooperate to transmit the same signal with very precise time/frequency synchronization.

However, the use of MBSFN and PHY-FEC techniques is not enough to provide a reliable delivery of multimedia segments (files) over OFDM. This is the reason why these techniques at the physical level are combined with two additional mechanisms at the application level, AL-FEC and unicast error recovery. AL-FEC techniques consist of the generation of redundancy data that are sent together with the file to be transmitted. A receiver is able to recover a file if enough data have been received, either from redundancy or from the original object. Otherwise, the data is discarded and the file cannot be delivered to the application.

A multimedia content encoded as defined by DASH is composed of several segments. Each segment is independently protected using the AL-FEC and then it is sent to the multiple receivers using eMBMS. For the

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delivery of DASH segments over eMBMS, the 3GPP proposes the use of the File Delivery over Unidirectional Transport (FLUTE) protocol. FLUTE is a protocol particularly suited for the unidirectional delivery of files in multicast networks, since it works over UDP and it can be used together with AL-FEC techniques.

However, the AL-FEC decoder might not be able to recover a multimedia segment when there is a high error rate in the eMBMS channel. In this case, the 3GPP proposes to recover the lost segments using HTTP. The whole system works as follows (Fig. 1). Using FLUTE, the DASH multimedia segments are pushed over eMBMS to the multiple receivers connected to the eMBMS channel. When a FLUTE client receives a DASH segment, it copies it to a local HTTP cache. A DASH client request goes first through the local HTTP cache, so a segment that is received correctly over FLUTE is delivered from the cache to the DASH client. In case a segment is not available in the cache, it is retrieved from the content server via HTTP using the regular unicast access channel.

3.1 DELAY ANALYSIS OF A LIVE STREAMING SERVICE OVER OFDM

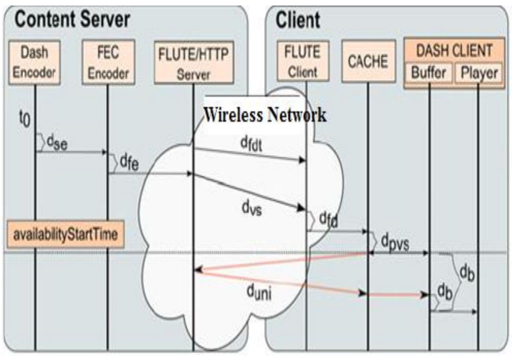

Fig. 3.2. Delay analysis. This figure shows the delays suffered by a DASH segment that is sent using eMBMS and the earliest availability time of the segment on the mobile device

In an OFDM multimedia broadcast service, a multimedia segment suffers a series of different delays from its generation in the content server until its playback in the multimedia player, as shown in Fig. 3.2.

A. Generation of DASH Video Segments

The delay analysis starts at instant t0, representing the instant the first byte of a live video signal is received by the content server from the source. In order to prepare the video data to be sent over the eMBMS channel, a DASH encoder located in the content server is in charge of encoding the input stream and splitting it into segments of a predefined duration. Often, it is necessary to convert the video, applying a video compression format such as H.264. In the particular case of an OFDM multimedia broadcast service, only one representation is needed to be sent over the multicast channel, so video segments are coded with a single bitrate. In conclusion, both segmentation and coding are needed to generate a DASH video segment. In Fig.3.2, the time taken to generate a DASH video segment is denoted as d_{se}.

B. Protecting Video Segments Using AL-FEC Techniques

At present, the AL-FEC technique standardized by the 3GPP to provide a reliable transmission over eMBMS is Raptor. Raptor is a block code that allows splitting a data stream into several files, also known as source blocks, so each source block can be independently coded. In an OFDM multimedia broadcast service, each multimedia segment generated is a different source block.In Fig. 2, the time taken to encode a video segment together with the resulting AL-FEC code is denoted as d_{fe}, whereas the time needed to decode a video segment is denoted as d_{fd}.

C. Delivery of Live Video Using FLUTE

In a multicast transmission over eMBMS, video segments are delivered within a FLUTE session. Before receiving a file, a FLUTE receiver needs to receive the attributes of the FLUTE session and a description of the properties of the file. This information is sent in-band within the file delivery session through an XML file called File Delivery Table (FDT). In a live streaming service, the video segments are progressively generated, so it is necessary to notify, also progressively, the properties of each of the video segments that are sent using FLUTE over the multicast channel. If an AL-FEC technique is used, the FEC Object Transmission Information must be sent within the FLUTE session. This paper considers using the FDT to transmit the FEC Object Transmission Information, because it requires a lower overhead compared to sending this information in-band. This information depends on the FEC schema used. With a fixed transmission rate, the time taken to transmit an instance of the FDT (*d*fdt) is considerable shorter than the time of transmitting a video segment (*d*vs), so that *d*fdt can be ignored. Note that the FDT size can be a few hundreds of bytes, whereas the size of a video segment is larger by more than one order of magnitude.

D. Protection against Delay Variations

In the case of an OFDM multimedia broadcast service, the impact of the *availability Start Time* is different, but a similar margin of security is needed. The time taken for a video segment to be pushed to the local cache of the clients depends on the delay components previously analyzed, which are variable. In this case, the possible consequence is that DASH clients request a video segment that is not yet available in the cache, but that will be received later via eMBMS. This is not desirable, because the video segment will then be unnecessarily requested and transmitted over HTTP, thus wasting communication resources. Therefore, for an OFDM multimedia broadcast service it is also necessary to introduce a margin of security that takes into account the delay variations of the previous components. This ensures that video segments are not requested before they are copied to the receiver cache. This margin of security is denoted as *d*pvs in Fig. 2.

E. Unicast Delivery

If a client requests a video segment that is not available in the cache, a session is established between the client and the server to deliver the segment over HTTP. Taking into account the margin of security introduced by *d*pvs, this can only happen in two different scenarios:

1) When a video segment has been incorrectly decoded by the FEC decoder, or,

2) When the video segment has been correctly decoded, but the instance of the FDT that describes the video segment has been lost.

The use of HTTP to deliver a video segment adds an additional delay, denoted as *d*uniin Fig. 2. This delay depends on several factors.

F. Client Buffer

In a multicast streaming service over OFDM, buffer requirements are different. Only one representation is sent over eMBMS, so the buffer level cannot be used to switch between different representations. On the other hand, a video segment that is lost in the eMBMS channel needs to be recovered via HTTP, so a long enough buffer is needed to avoid buffer starvation during the recovery of lost segments. The buffer level that is required to start the playback of a video is specified by the *min Buffer Time* field of the MPD. If a low value of *min Buffer Time* is set, and there is a high error rate that implies the loss of several video segments in the multicast channel, buffer starvation can happen. To solve this problem a strategy to calculate the minimum buffer level that is needed to avoid possible video freezes during the playback is proposed in the next section. This buffer level will be characterized by its value in seconds, denoted as *db*in Fig. 2.

G. Playback Deadline

Based on the delay analysis presented above, one of the proposals of this paper is to set the *availability Start Time* field of the MPD to a value that guarantees that a video segment correctly delivered over eMBMS is available on the local cache of the clients before it is requested over HTTP. This avoids the synchronization problems that would lead to unnecessary transmissions over HTTP. Taking into account the different components of the delay analyzed above, the proposed value for the *availability Start Time* field can be calculated as

Availability Start Time = $d_{se} + d_{fe} + d_{vs} + d_{fd} + d_{pvs}$_____ (1)

This time is depicted in Fig. 2 as a dashed line together with the tag *availability Start Time*. The playback deadline, defined as the instant in which a video segment content is started to be played by the DASH client is then calculated as

Playback $_{\text{deadline}}$ =Availability Start Time +d_b______ (2)

Where *db*is the buffer level. In an OFDM multimedia broadcast service, the buffer level determines the seconds of video that must be stored in the buffer before starting the playback. In the following section, we are going to analyze how to calculate the minimum buffer level needed to avoid service disruptions when video segments are lost in the multicast channel and the multimedia player recovers the segments using HTTP.

3.2. BUFFER REQUIREMENTS OF MULTIMEDIA PLAYER

The use of AL-FEC techniques is not always sufficient to ensure a reliable delivery of DASH video segments, so video segments lost in the multicast channel are recovered via HTTP. However, there is no reservation of bandwidth for this communication.

To calculate the minimum buffer level needed to avoid buffer starvation, the worst possible case in term of losses is considered, i.e., a case in which a consecutive sequence of video segments are lost. The probability of losing a video segment is calculated as the probability of the FEC decoder not recovering a segment due to errors in the packets received through the eMBMS channel.

This probability is denoted as*P*(*f*RC), and is given as

$P(f_{RC}) = \sum_{n=0}^{k+r} P(f_{RC}|n).P(N = n)$______ (3)

Where $P(f_{RC}| n)$ is the failure probability of the Raptor decoder in case of receiving n encoding symbols and where $P(N = n)$ is the probability of receiving n symbols.

According to the mathematical model presented by Stock hammer *et al.*, $P(f_{RC}|n)$ can be calculated as

$$P(f_{RC}|n) = \begin{cases} 1, & if\ n < k \\ 0.85 \times 0.567^{n-k}, & if\ n \geq k \end{cases}$$ ______ (4)

On the other hand, since each symbol is encapsulated in a packet, given a Packet Error Rate (PER), the probability of receiving n symbols can be calculated by using a binomial distribution, given as

$P(N = n) = \binom{k+r}{n}.(1 - PER)^n . PER^{k+r-n}$______ (5)

In order to calculate the minimum buffer level, a *threshold* is used to limit the probability of losing m video segments consecutively,

$P(f_{RC})^m$ <threshold. _____ (6)

The value of *threshold* represents the probability of a service disruption. In order to ensure a seamless playback for a long time, a low value of *threshold* is needed. For a given *threshold*, buffer starvation is avoided when the buffer is long enough to keep playing the video when m consecutive video segments are lost in the multicast channel.

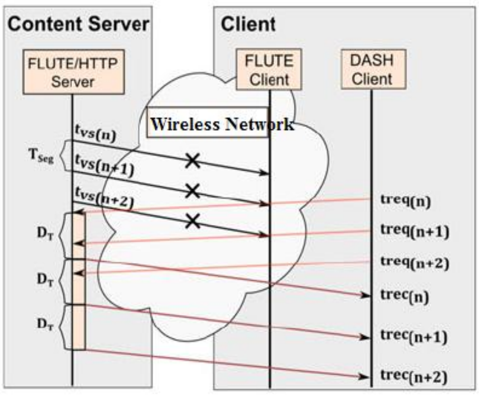

Fig. 3.3. Delay analysis. This figure shows the time that would be needed to recover, via HTTP, a burst of video segments lost in the eMBMS channel.

In order to calculate the minimum buffer level needed to avoid buffer starvation, Fig.3.3 depicts the case of losing m = 3 video segments consecutively. Fig.3.3 shows a FLUTE server sending three video segments over MBMS. However, these video segments are lost, i.e., the AL-FEC process at the FLUTE client is not able to recover the video segments due to packet errors. The DASH client requesting the video segments does not find them on the cache, so an HTTP connection is used to retrieve the lost video segments. It is assumed that a persistent HTTP connection, previously established, is used to recover the video segments. Otherwise, an additional delay due to the TCP three way handshake would need to be added to the analysis..

Fig.3.3 shows the case in which the transmission delay for each video segment is longer than the duration of the video segment, which is the worst-case scenario in regards to buffer starvation. Since the delay accumulated for the last segment is the longest one, the buffer level must be set accordingly. Therefore, the buffer level can be calculated as

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$d_b = trec_{(n+2)} - treq_{(n+2)} = RTT + 3D_T - 2T_{seg}$ _______ (7)

Where treq(n+2) is the time when the segment is requested, trec(n+2) is the time when the segment is received, DT is the transmission delay, and TSeg is the duration of the video segment, which is the same as the interval between requests. In general, for *m*video segments lost consecutively, when the transmission delay of the video segments is longer than the duration of the video segments ($DT > T$Seg), the minimum buffer level can be calculated as

$d_b = RTT + mD_T - (m-1)T_{seg}$ ________ (8)

If $DT < T_{Seg}$, the buffer is only needed to protect the playback during the time a unicast recovery is performed,

$d_b = RTT + D_T$ ________ (9)

To sum up, it is possible to calculate the probability of losing a DASH segment during its transmission over eMBMS (3). This probability is calculated for a given PER and by using a mathematical model that allows simulating a Raptor code (4). This probability is then used to calculate how many segments can be lost consecutively given a *threshold* (6), which represents the probability of service disruption. Finally, the minimum buffer level needed to avoid service disruptions (8-9) is calculated taking into account the transmission delay that is introduced due to the HTTP recovery of the DASH segments. This buffer level can then be notified to the DASH clients through the *min Buffer Time* field of the MPD, which is used by the DASH player to store enough data in the buffer before starting video playback.

IV. RESULT AND DISCUSSION

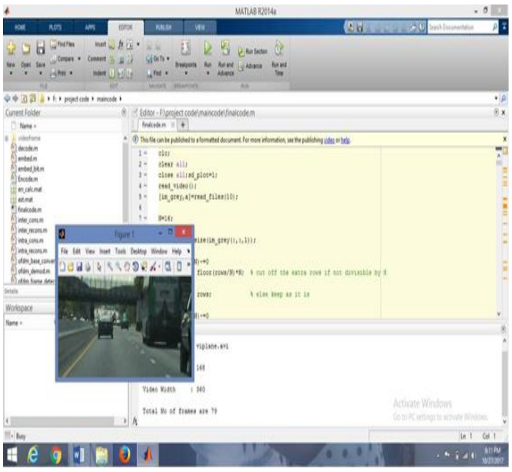

FIG 4.1 shows that the video is read by the program

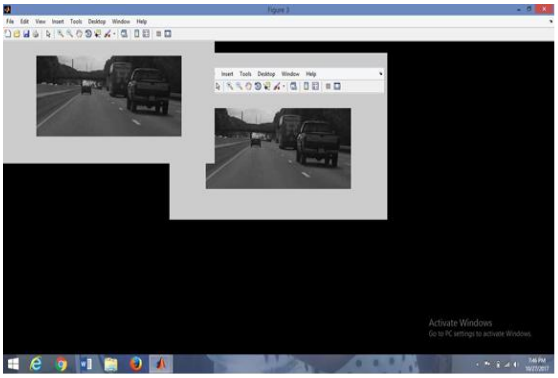

FIG 4.2 shows that the video is converting into frames

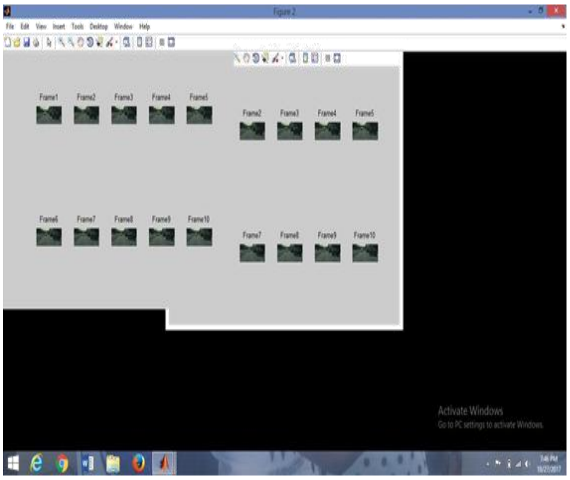

FIG 4.3 shows that the video is completely converted into frames

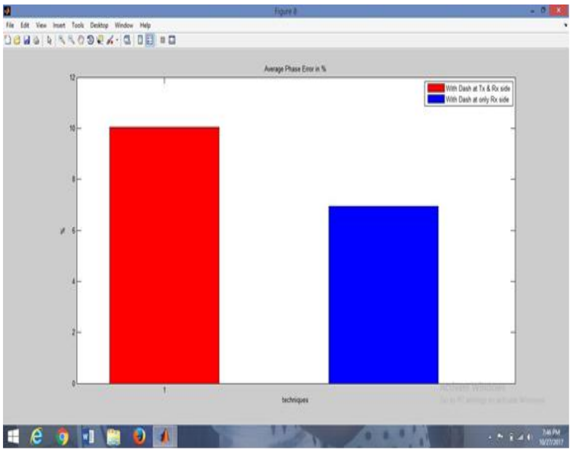

FIG 4.4 shows that the average phase error between the original image and the image received with DASH and AL-FEC processed

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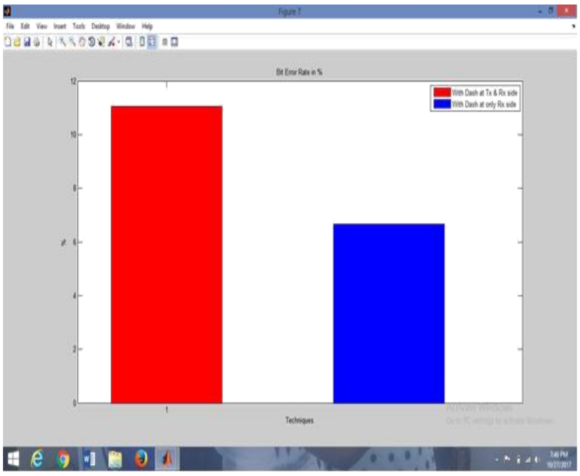


FIG 4.5 shows the bit error rate while sending original image and processed image

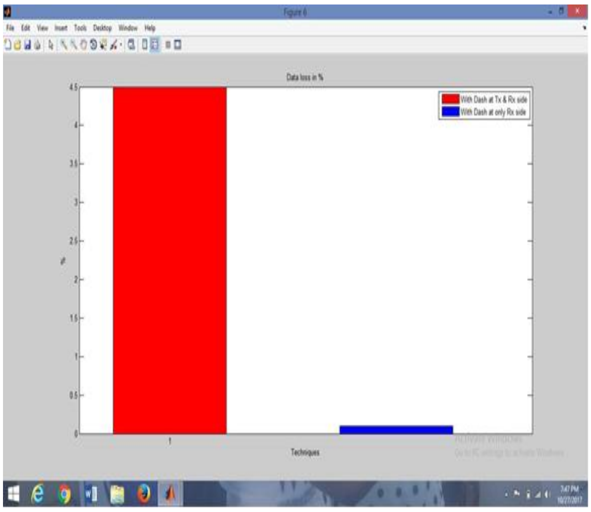
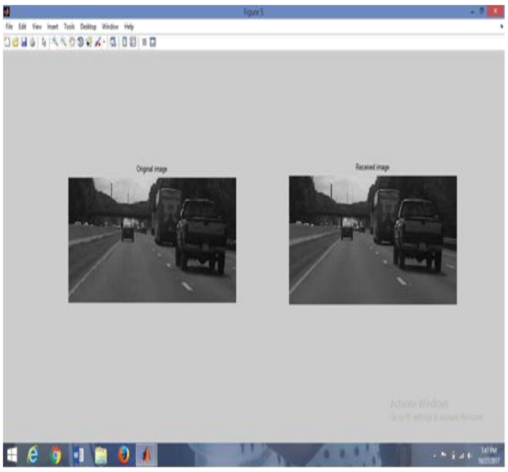


FIG 4.6 shows the data lose between the original image and processed image



4.7 FIG shows the quality improvement in the received image than the original image

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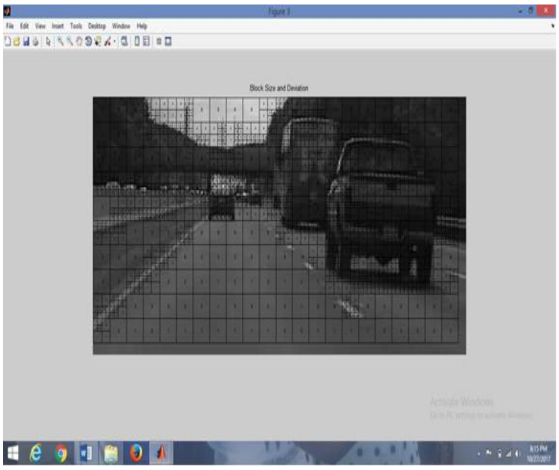

FIG 4.8 shows that the frames are converted to packets and sub division packets at equal size

V. CONCLUSION

The multimedia broadcast services over wireless OFDM networks integrates in a novel way elements borrowed from different standards and proposals, such as DASH to split the multimedia content in different segments, the channel to broadcast the segments to the wireless devices using the FLUTE file delivery protocol, AL-FEC techniques to recover from packet transmission errors, and DASH (HTTP) retrieval for segments that cannot be recovered using the AL-FEC.

This project has analyzed the different components of the latency, and has proposed a solution to allow a multimedia player running on a wireless device to play the multimedia content witha minimum latency. The key elements of the solution are a methodology for the calculation of an adequate value for error included in the DASH ,indicating the earliest instant a segment can be accessed, and a methodology for the calculation of the adequate segment duration and the minimum buffer required for a seamless multimedia stream reproduction.

The impact of this solution on the reduction of the latency can be significant, because normally the DASH component of a multimedia streaming service is configured for the retrievalof content over a unicast TCP connection. However, the impacton the design of wireless devices is minimal, requiring only the adequate definition of parameters on the MPD used to define the multimedia content that is broadcast over wireless. The Future work is planned to extend the present work with the help of an optimization technique

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