IJE TRANSACTIONS C: Aspects Vol. 26, No. 3, (March 2013) 229-240

International Journal of Engineering

Journal Homepage: www.ije.ir

Analysis of Packet Loss and Latency Control for Robust IPTV over Mobile WiMAX and LTE Assessment

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PAPER INFO

Paper history:
Received 7 February 2012
Received in revised form 5 September 2012
Accepted 15 November 2012

Keywords:
IPTV
WiMAX
LTE
IPv6
MPEG
FEC
Coding
Streaming
Signal Brust

ABSTRACT

There are different schemes for streamed audio video (AV) IPTV content across mobile WiMAX to reduce packet loss and latency. The objective of this paper is to verify the effectiveness of forward error correction (FEC) techniques and to suggest the techniques for robustness problems and to analyze the issues either due to AV coding encoding or due to access network. The paper concentrates on handoff and handover delay, mobile WiMAX downlink burst mapping and the reliability of IPTV over LTE system.

doi: *10.5829/idosi.ije.2013.26.03c.02*

NOMENCLATURE

S	Slice composed of original bursts	WS	The width of stripe
ʙ	Burst	tω	The number of wasted aditional or extra slots in stripe allocation
bí	Original bursts	Q	The number of network elements
sí	Slots, time slice		
n	Number of burst	**Greek Symbols**	
H	The height of the frame	O	The complexity of algorithm
ŵS	Temperary width of slice	∑	Summation sum over …sum from
hí	The temporary height of each burst	ε	Element of
đ	Delay processing, propagation, queuing	∀	All of, for any, for each
ẅí	Adaptive width of each burst	⌈·⌋	Rounds to closest integger

1. INTRODUCTION

Internet Protocol Television (IPTV) provides digital information and audio video contents using high speed internet [1]. This IP based network is managed to provide quality of service (QoS), quality of experience (QoE), security, reliability and interactivity. Before transmission, the IPTV contents must be digitized as MPEG format. IPTV has an intelligent content management to deliver the programs by encoded video stream, application layer coding schemes and retransmission of extra redundant data. There are several services of IPTV like interactivity (two way

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communication between consumers and service providers), downloading, electronic program guide, personal video recorder, time shifted TV and video on demand (VOD). It has high-speed access channels via set-top boxes or receiving equipment. The IPTV setup for fixed and mobile subscribers is shown in Figure 1. Service Provider Core/Edge IP Network is the core network of the service provider and includes hardware. The access network connects the service provider to the subscriber's home as a fixed subscriber of IPTV via set top box. The IP based core network is connected with mobile WiMAX for mobile subscribers.

In IPTV over mobile WiMAX setup the AV content rate adapter is used for adapting the IPTV service bandwidth.This adapter is used for available WiMAX bandwidth and the IP encapsulation.

The two primary protocols used for IPTV are internet group multicast protocol (IGMP) version 2 for channel change signalling for live TV and real time streaming protocol (RTSP) for video on demand. Figure 2a shows the serialised AV signal for IPTV transmitting through DVB-H. Figure 2b shows the receiving end of IPTV.

The protocol stack to transmit the broadcasting content for IPTV services is shown in Figure 3a. IPTV sends MPEG type of AV contents in sequence to network side after RTP, UDP and layer 2 encapsulated. Figure 3b shows a protocol stack in each entity to manage the channel registration and release to provide IPTV service [2]. IPTV head-end captures audio video and then AV is processed by encoding each individual channel into a digital video format MPEG-2-4 before being sent over the IP network.

The AV encoded streams performance is inherently a function of available link bandwidth and delay characteristics. So, analysis of performance including packet loss, latency, jitter and minimum peak throughput is discussed to quantify IPTV over the core mobile WiMAX infrastructure.

To reconstruct streamed video across an IEEE 802.16E mobile WiMAX channel, the different schemes are discussed in this paper. The paper concentrates on the redundant data to verify the effectiveness of forward error correction (FEC) bit rate and appropriate compression rate. To analyse the tradeoff between FEC and degradation of video quality for given packet loss ratio, the encoders are used as test sequences for the channel error models.

For IPTV services the mobile WiMAX is discussed as a medium which has handover latency components same as in mobile IPv6 and latency jitter in mobile WiMAX which is the cause of delay in sensitive applications. There are three latency components of mobile IPv6.

There are different distribution technologies for IPTV network like delivery over fibre access network, ADSL network, next generation cable TV network, internet, mobile WiMAX and over LTE. This paper concentrates on IPTV delivery over mobile WiMAX. The IPTV transmission system over LTE has also some issues as reliability, packet loss rate and delay time. The cellular networks like WiMAX and LTE are not only for conventional voice service but also for data service, for the readability of services the different techniques are required to bring new generation on the mobile market.

This study is based on following sections; related work will be discussed in section 2, proposed system will be described in section 3, comparison and performance evolution will be described in section 4 and conclusion will be discussed in section 5.

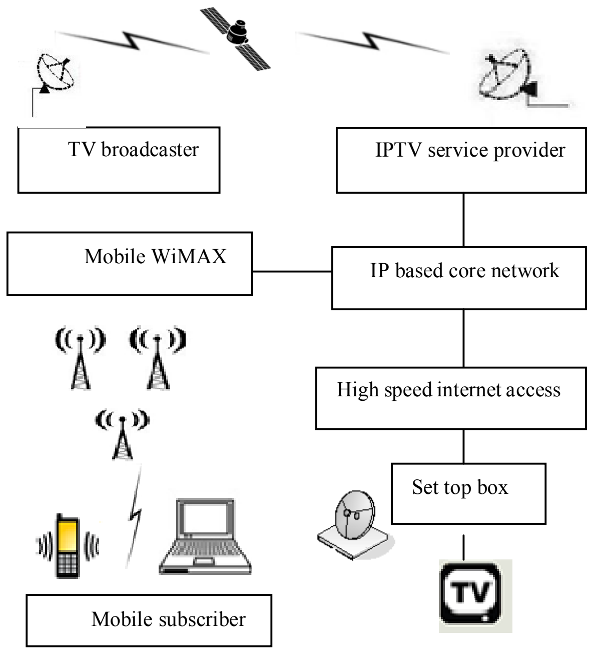

Figure 1. IPTV setup for fixed and mobile subscribers

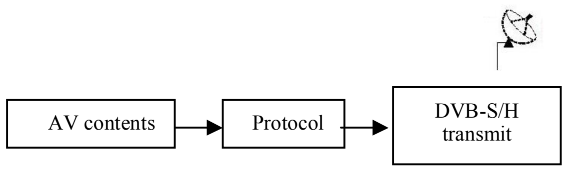

Figure 2a. Transmitting AV signals

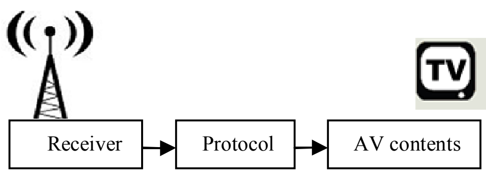

Figure 2b. Receving AV signals

2. 1. 3. Intrarefrsh Macroblocks The insertion of intra-refresh (IR) and macroblocks (MB) are encoded in to pictures through motion compensated prediction which allows temporal error propagation to be arrested if matching in a previous picture.

In the H.264/AVC JM implementation, various IR schemes exist such as random, which sets a maximum percentage of MBs, or cyclic, which replaces each line of the picture in turn in cyclic order. Notice that naturally encoded IR MBs are also inserted into predicatively coded p-pictures when inter coding brings limited or no advantage. (e.g., this may occur during rapid motion or when a new object that is not present in a previous picture is uncovered) [5].

2. 1. 4. Adaptive Scheme Application layer, adaptive rateless channel coding scheme is used to reconstruct streamed video across an IEEE 802.16e (mobile WiMAX) channel, rateless code decoding is reliant upon the identification of clean symbols. In the adaptive scheme [6], the probability of channel loss (PL) serves to predict the amount of redundant data to be added to the payload. Assume that "bursty" error conditions are generated by the widely used Gilbert-Elliott model, which is a form of hidden Markov model with a good and bad state.

In the Gilbert-Elliott model *pgb* is probability of the transition from the good state to the bad state, and *pbg* is the probability of transition from the bad state to the good state. Then *p*gg and *pbb* are the probabilities of staying in the good state and the bad state respectively as shown in Figure 5.

This latter function is performed by PHY-layer FEC which passes up correctly received blocks of data (through a cyclic redundancy check) but suppresses erroneous data. For example, in IEEE 802.16e [7], a binary, nonrecursive convolution encoder with a constraint length and a native rate operates at the physical layer.

The scheme reduces the FEC overhead. Whichever scheme reduces, the number of corrupted packets reduces the overall delay introduced into the video stream. However, if packets are simply not received then the adaptive rateless coding scheme cannot help. The main disadvantage of using model generated loss traces is that statistical properties may not fit to the statistical properties of a measured trace as they are likely to be biased by model limitations.

2. 2. Reducing Handoff Handover Latency Hand-off and handover have become a crucial factor in quality of service responsive networks. In WiMAX the handoff latency ratio is based on its hierarchical model and it is required to use centralized control approach to reduce end to end packet delay. For this purpose, previously proposed techniques and studies are given.

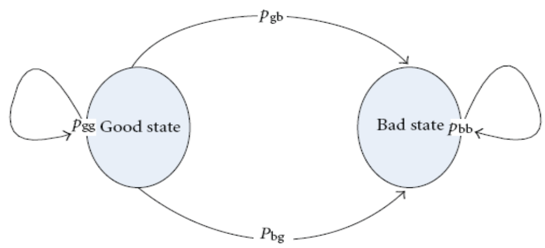

Figure 5. Gilbert-Elliott two-state model

2. 2. 1. A Cross Layer Approach To reduce delay, jitter, and packet loss innovative cross-layer content, delivery architecture is capable of receiving information from the network and adaptively tune transport parameters, bit rates, and QoS mechanisms according to the underlying network conditions [8]. The proposed adaptive cross layer video streaming system is composed of a video server and a video client. The server streams the audiovisual object to the client via an IP Diff-Serv network using the RTP protocol. The client decodes and composes the original MPEG-4 scene. When network congestion occurs, less important AV streams are dropped automatically by the active queue implemented in the Diff-Serv router. The IP Diff-Serv marker tags each video data packet belonging to one AV object with one of the supported Diff-Serv class of service to reflect object priority. Hence, important objects will be marked with a low drop precedence to guarantee a minimum loss, and so on. Object data-packets within the same class are then transmitted over the selected transport layer with the corresponding bearer capability. The disadvantage of AV object-based methods is that all packets of objects are given the same importance regardless of distortion caused by each one. The major drawback of existing cross-layer designs is that the simplification occurs at the system modelling phase rather than the problem solving phase.

2. 2. 2. WiMAX with MIPv6 Handover Mobile IPV6 is an important factor to study the latency components.There are three latency components. The first one is D1 (delay 1) which is the time it takes at link-layer to switch to the new access medium. The second one is D2 which is the time to detect the new IPv6 network, complete DAD (duplicate address detection) and configure the new CoA (care of address). The third latency component is D3 which is finally the delay to update the home agent and all other communication and parameters.

IPv6 provides real-implementation results for significant parts of the handover process through measurements in a real MIPv6 implementation on a wireless test bed based on IEEE 802.11b [9]. IPv6 is the networking technology of choice to enable the internet

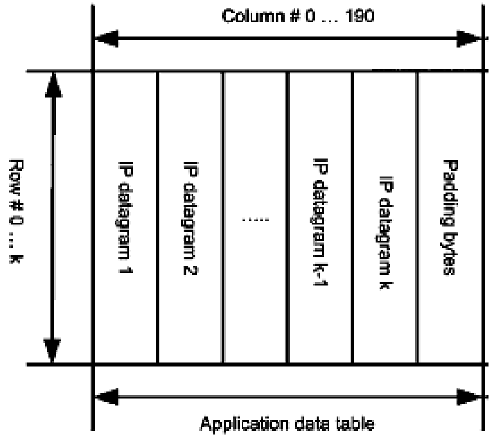

Figure 7a. MPE-FEC frame with application data table [13]

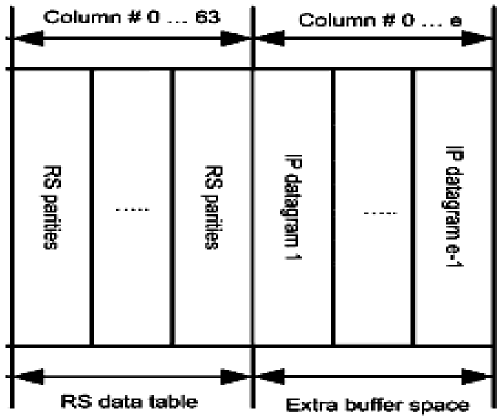

Figure 7b. MPE-FEC frame with extra buffer space

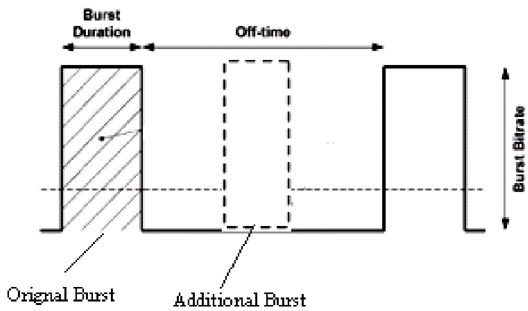

Figure 8. Example of additional burst

Figure 7b shows MPE-FEC frame with extra buffer space. The two parts of MPE-FEC are application data table (ADT) with IP data and RS data table (RSDT) with RS parity data. By applying a RS code on each row the RSDT is computed after filling the ADT column-wise. The proportion of transmitted columns in each table is a resulting coding rate. DVB-H standard allows padding to make the code stronger and perforating to make the code weaker. The use of MPE-FEC is always a trade-off between consumption of network capacity and robustness [14]. Application Layer Forward Error Correction (AL-FEC) is used to provide reliability in mobile broadcast systems. The Raptor codes adopted by DVB-H for file download services are systematic codes which are efficient implementation of fountain codes that operate at the application layer. To generate additional RTP streams, the FEC packets are used by the receiver to reconstruct lost packets.

There are two disadvantages for FEC techniques as:

- Additional bandwidth is needed to transmit FEC packets
- At the transmitter and the receiver stage the additional latency is introduced by FEC operations

3. 2. Proposed Technique Stategy This paper proposes the addition of an extra burst after the original burst with taking data from one protocol and translating it into another protocol using FEC. The extra burst containing parity data is transmitted several seconds after the informative burst. The informative burst indicates the initial time of next burst with new information and it indicates the time of the additional burst. Now, the coding (AL-FEC) rate for both burst depends on the size of the extra burst with parity data. The time of extra burst can be calculated with the sum of informative burst duration and off-time as shown in Figure 8.

With the addition of an extra burst, the AV content can be decoded in three different ways. At the receiving end the original informative burst can be received correctly by neglecting the additional one. Another condition is possible that there is only part of the original burst received at receiving end and receiver will need a part or all the additional burst. Here is another case that when the receiver receives only the additional burst but cannot receive original burst then additional burst help as a redundant data to recover the information transmitted. The original burst and the additional burst has a statistical relationship which resolves the robustness, loss with respect to distance and time. The recovery or the improvement of the data is based on the speed and the time between both bursts. The correctly reception of bursts can be calculated with the average carrier to noise ratio (CNR), standard deviation of shadow fading in dB and distance. The proposed technique is used with MPE-FEC and AL-FEC therefore any modification of present implementation is not involved, because the raptor codes are already standardized for mobile WiMAX channel and all terminals. The proposed technique is useful by reducing the MPE-FEC and AL-FEC overhead, which may introduce extra delay. However, the delay can be adjusted by varying the AV content quality and, hence the burst size.

Performance metrics observe packet transmissions, include packet loss, packet delay, packet jitter and traffic load throughput rates. Video playback quality is strongly related to packet loss and end-to-end packet delays. The following objective measures, which are widely used in video content performance analysis, are employed in this paper.

4. 1. Packet Loss Ratio Packet loss ratio (PLR) is the number of corrupted, dropped, or excessively delayed packets in relation to the total number of packets expected at the video client station. PLR can be calculated as follows [18]:

$$PLR = \text{lost packets} \,/\, (\text{lost packets} + \text{received packets})$$

Another variation of this metric is the media loss rate (MLR) which track packet loss over time [19]:

$$MLR = \frac{(\text{Packets expected}) - (\text{Packets received})}{\text{Interval time in seconds}} \quad (7)$$

4. 2. Packet Delay Packet delay is the average packet transit time between the VoD server and the video client station. This metric can be calculated as follows [20]:

$$đ_{end} = Q\,(đ_{proc} + đ_{queue} + t_{trans} + đ_{prop})$$

where Q is the number of network elements (routers, switches and firewalls) between the sender and receiver, $đ_{proc}$ is the processing delay at a given network element, $đ_{queue}$ is the queuing delay at given network element, t_{trans} is the transmission time of a packet on a given link and $đ_{prop}$ is the propagation delay across a given network link.

4. 3. Packet Jitter Packet jitter is defined as the variability in packet delay within a given media stream at the video client station. This metric can be calculated as:

$$J_{pkt} = t_{actual} - t_{expected} \quad (8)$$

where t_{actual} is the actual packet reception time and $t_{expected}$ is the expected packet reception time.

4. 4. Throughput Throughput is defined as the traffic load that the media stream will impress upon the network. It can be measured in bytes/sec (Bps) or bits/sec (bps). For CBR content, the throughput is constant and it can be calculated as:

$$R_{min} = (\text{frame size in bytes})\,(\text{number of frames per second}) / \text{seconds} \quad (9)$$

However, with variable bit rate encoders, the traffic loading is dynamic in nature and a function of the scene complexity and associated audio content. Consequently, variable bit rate (VBR) traffic loads are typically quoted in throughput ranges.

4. 5. Comparison between IPTV over Mobile WiMAX and over LTE To analyse the performance of robust IPTV over mobile WiMAX, a comparison between WiMAX and LTE is required. This study chose LTE over WiMAX as the technological foundation for its 3G/ 4G wireless broadband network.

The throughput or peak network speed is an important factor to measure the performance. The peak network speed is usually quoted at layer 2 because of protocol overhead. LTE is a 4G wireless technology that Verizon Wireless and numerous leading wireless carriers have been chosen as their upgrade path beyond 3G technologies. Verizon Wireless will operate LTE in the 700 MHz spectrum, which translates to unprecedented performance and data access. The given table shows the evolution of LTE.

TABLE 1. The evolution of GSM to LTE

	1xRTT	**1Xev-do Rel.0**	**1Xev-DO Rev.A**	**3GPP LTE**
Peak speed	153 Kbps (downlink) 153Kbps (uplink)	2.4Mbps (downlink) 153Kbps (uplink).	3.1Mbps (downlink) 1.8Mbps (uplink).	100Mbps (downlink) 50Mbps (uplink).
Average user throughput	60–80 Kbps (downlink) 60–80 Kbps (uplink)	400–700 Kbps (downlink) 60–80 Kbps (uplink)	600–1,400 Kbps (downlink) 500–800 Kbps (uplink)	5-12Mbps (downlink) 2–5Mbps (uplink)

TABLE 2. Technical difference between LTE and WiMAX

	LTE	**WiMAX 802.16E**
Technology	MIMO Downlink: OFDM Uplink:SC-FDMA	MIMO Downlink: OFDM Uplink: FDMA
Peak speeds	Downlink: 100 Mbps (20 MHz, 2x2 MIMO) Uplink: 50 Mbps (20 MHz, 1x2)	Downlink: 46 Mbps Uplink: 7 Mbps
Average user throughput	5 Mbps–12 Mbps (downlink) 2 Mbps–5 Mbps (uplink).	2 Mbps–4 Mbps (downlink) 500 Kbps–1.5 Mbps (uplink).
One-way airlink latency	15 ms	50 ms.
Bandwidth	20 MHz, 15 MHz, 10 MHz, 5 MHz, and <5 MHz.	3.5 MHz, 5 MHz, 7 MHz, 8.75 MHz, 10 MHz.
Spectrum	Verizon Wireless will use 700 MHz, but LTE can be deployed in various frequencies. Use of the 700 MHz frequency helps to increase in-building coverage for wireless signals.	2.3, 2.5, 3.5, 5.8 GHz.
Mobility	Targeted mobility up to 350 kmph.	Targeted mobility up to 120 kmph.

LTE and WiMAX are able to provide wide range coverage, high data rates, secured transmission and mobility supported at vehicular speeds. The quality of service (QoS) is required to transmit audio, video, voice, data services such as video gaming and mobile IPTV. The 3GPP LTE is the latest standard in the cellular network technology that previously realized the GSM/EDGE, UMTS/HSxPA network technologies of mobile subscribers. A radio access of LTE is named Evolved UMTS Terrestrial Radio Access Network (E-UTRAN) which improves end-user throughputs, sector capacity and reduces user plane latency to support IPTV services, with full mobility. For carrying all types of traffics, LTE is planned to provide IP-based traffic with end-to-end Quality of service (QoS).

TABLE 3. Techno-economic differences between WiMAX & LTE [21]

	WiMAX	LTE
Foundation	Based on IP	3G an evolution from a voice traffic design
Deployment	Yet to deploy in volume (2009) Deployed recently	3G has a substantial base station population (128 commercial HSPA networks rolled out and over 300 HSPA devices)
Maturity	Mobile WiMAX under ratification Mobile WiMAX has to win a customer base	3G technology is maturing 3G customer base is established by evolution from GSM
User devices	WiMAX will be part of existing hardware such as laptops and PDAs	3G uses dedicated hardware or plug in cards
Equipment cost	Owing to standards, cheaper	No standards available as on date, hence may be costlier (2009)
Deployment cost	Higher	Lower

TABLE 4. Similarities between LTE & WiMAX [21]

OFDMA	OFDMA supports advanced antenna technologies, such as MIMO, STC, and beamforming OFDMA is adopted as the basis of mobile WiMAX 3GPPP's Long Term Evolution (LTE) project plans to incorporate OFDMA7
IP	LTE is moving from a circuit-switched to all IP IP is built into mobile WiMAX based on the IEEE 802.16 air interface standard
Coustomer	Want the same experience when using mobile as they have when at home or in office

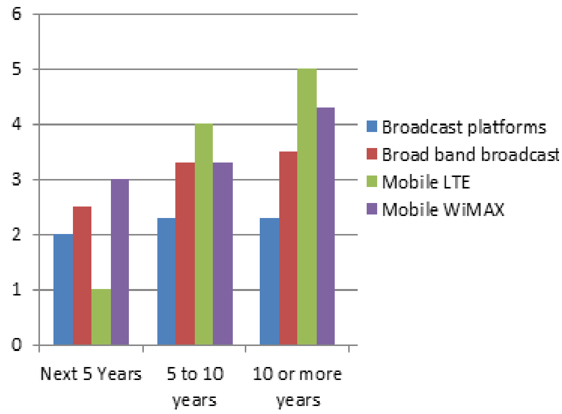

Figure 9. Technologies expected to complement TV broadcasting in the futur

LTE core network is the Evolved Packet Core (EPC) which allows for connections and hand-over to other fixed line and wireless access to deliver a seamless mobility experience. LTE minimizes the system and UE complexities and enable co-existence with other 3GPP Radio Access Technologies (RATs). Using evolved multimedia broadcast, multicast service (E-MBMS) 3GPP provides the IPTV services multi-cell synchronized transmission technology. In the viability of the LTE standard and its future potential 4G technologies determine the best fit for its network. IPTV delivery over WiMAX or LTE both have same issues as number of channels depends on viewer's waiting time (start up delay) and it does not depend on number of requests. A quality of transmission channel decreases due to a noisy environment. The symbol error rate (SER) will increase with respect to proliferation of fading frequency.

The overall radio performance is slightly equal with LTE faintly overtaking WiMAX. Strategic considerations on the future of TV broadcasting especially IPTV is shown in Figure 9 which gives an overview of the complementary technologies that are expected to emerge over the next 10 years and beyond. The height of the bars corresponds to the number of responses [22].

It can be seen that, terrestrial distribution will remain a significant delivery platform but will be complemented by mobile (LTE) networks. In the next five years the introduction of DVB-T2 alongside DVB-T is expected. There will still be a minimal use of non-broadcast technologies (mobile networks) for linear services. Furthermore, IPTV is expected to grow rapidly. Over the next 5-10 years, LTE will be available, but its capacity may be insufficient for significant delivery of linear services. Hence, it is expected that hybrid solutions will be adopted which will employ the terrestrial platform as one delivery mechanism. Clearly, there will still be an increasing importance for IP-based distribution over the internet.

5. CONCLUSION

In this work, an analysis of the effectiveness of AL-FEC and MPE-FEC schemes are discussed for IPTV delivery over mobile WiMAX and a technique is proposed to add an extra burst to resolve the issues. The mobile WiMAX downlink burst mapping algorithm is also discussed for the measurements of flooding of redundant slots. Based on the performed analysis, an optimization of the quality assessment is presented which analyses the adapted rateless channel coding. It is also described that how to transmit mobile IPTV service over cellular system specially, over LTE network. The delay time and packet loss rate are critical parameters for mobile IPTV service with highly QoS. IPTV has witnessed a rapid evolution moved from the STB to the phone, the PC and soon the game console. This paper reflects the status of techniques that are already deployed and researched to improve the IPTV issues. While IPTV of today still provides the lean back experience, where new researches are required, the future of IPTV is social and mobile and this future is already apparent.

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P A P E R I N F O

Paper history:
Received 7 February 2012
Received in revised form 5 September 2012
Accepted 15 November 2012

Keywords:
IPTV
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چکیده

طرح های مختلفی برای جریان های صوتی ویدئوی (AV) شامل IPTV در شبکه تلفن همراه وایمکس به منظور کاهش اتلاف سرمایه و زمان تاخیر وجود دارد. هدف از این مقاله، بررسی تاثیر تکنیک های ارسال تصحیح خطا (FEC) است و روش هایی را برای حل مشکلات، تجزیه و تحلیل ان را بر اساس برنامه نویسی AV پشتیبانی می کند ویا با توجه به دسترسی به شبکه نشان می دهد. این مقاله بر روی هندآف و هند اور، لینک دریافتی تلفن همراه وایمکس با نقشه برداری و اطمینان از IPTV در سیستم LTE تمرکز کرده است.

doi: *10.5829/idosi.ije.2013.26.03c.02*