Optimal IP Packet Size for Efficient Data Transmission in DVB-H

Vinod Kumar Malamal Vadakital[1†], *Miska M. Hannuksela*[2], *Mehdi Razaei*[1] *and Moncef Gabbouj*[1]

[1]Tampere University of Technology
Institute of Signal Processing
P.O. Box 553, FI-33101, Tampere
FINLAND
[†]Tel. +358-331154966, Fax: +358-3-3115 4989
[†]E-mail: vinod.malamalvadakital@tut.fi

[2]Nokia
Nokia Research Center
Tampere
FINLAND

ABSTRACT

This paper provides an experimental analysis on optimal internet protocol (IP) packet size in error prone DVB-H channel. Efficient transmission of data requires reliability of the data and bandwidth efficiency of the data itself. Overheads in the form of IP packet header data reduce bandwidth efficiency. However, larger packet sizes minimize header data but have a high probability of being corrupted during transmission thus reducing the reliability. Obtaining a balance between optimal packet sizes that can reduce overheads while still providing reliability is essential. Simulation conducted show that IP packet sizes between the range of 1024 to 2048 bytes show good efficiency in terms of bandwidth and reliability.

1. INTRODUCTION

Digital Video Broadcasting-Handhelds (DVB-H) [1] is the newest protocol from the Digital Video Broadcasting (DVB) Organization, specified to transmit data to mobile handheld terminals. Its standardization follows the successful testing and deployment of Digital Video Broadcasting-Terrestrial (DVB-T) [2] in many countries. DVB-T was initially designed to broadcast data to fixed reception terminals using roof-top directive antennas in large single frequency networks (SFN).

Field tests to evaluate the performance of DVB-T in a mobile environment proved that it worked well. However, a few concerns where identified when it came to reception by handheld mobile terminals. The most important of these concerns was the power constraint imposed by handheld mobile devices. The second concern was the ability to receive data reliably in highly error prone mobile wireless transmission environments.

In response to these concerns the experts at DVB formed a working group and began the specification of DVB-H. The power constraint problem was addressed by the introducing time-sliced reception. Additional link layer protection in the form of Reed-Solomon (RS) forward error correcting codes (FEC) computed over internet protocol (IP) [3] packets and packed into multi-protocol sections (MPE-FEC) was used to provide reliability to the transmitted data. The design of DVB-H was conceived such that its transmission can be done along with DVB-T signals without any modifications to the physical network. This meant that DVB-H transmission could use the same DVB-T infrastructure as long as some mandatory signaling of DVB-H was handled correctly.

DVB-H channel is prone to errors due to physical characteristics of the radio channel. MPE-FEC provides better reliability to the DVB-H channel. Uncorrectable errors are in the form of packet errors. A packet error in DVB-H can occur when even a single bit in the IP packet is uncorrectable and the cyclic redundancy code (CRC) in the packet fails. The packet error probability of smaller packets is lower than that of larger packet. However, smaller packets have additional header overloads and its coding becomes bandwidth inefficient beyond some point. Hence, it is important to identifying the packet sizes that provides error resiliency while maximizing bandwidth efficiency.

This paper aims at providing some insight on the behavior of error prone DVB-H channel to varying IP packet sizes. Section 2 introduces errors in the wireless channel and the methods used to counter the errors. A brief introduction to DVB-H data casting is given in Section 3. Simulation conducted to identify the best IP packet sizes is presented in Section 4. Section 5 presents the obtained results and limitations of the simulations are stated in Section 6. Conclusions are drawn in Section 7.

2. ERRORS IN WIRELESS CHANNEL

Wireless channels are faced with information loss due to the physical characteristics of a radio channel. Physical manmade and natural barriers, interference from other radio transmissions, and multi-path propagation and errors due to fading are some of the causes of transmission errors in the wireless channel.

There are many mechanisms to overcome the erroneous conditions in the wireless channel. The popular ones include additional protection in the form of forward error correcting (FEC) codes and automatic repeat request

1-4244-0413-4/06/$20.00 ©2006 IEEE

(ARQ) techniques. FEC techniques add additional parity information along with the actual data which can then be used to correct the data errors if any. ARQ techniques use retransmission request from the receiver to the sender when lost data is detected. When a repeat request is made to the sender, it makes and identical copy of the data that was lost and retransmits it to the receiver.

Wireless errors are burstier in nature than wire-lined networks. Research on the nature of errors in wireless transmission systems have revealed that bit errors in these systems occur as clustered bursts rather than isolated uniformly distributed bit errors. The characteristics of the errors vary depending on the location of the receivers, the motion characteristics of the receiver, and the interference levels of signals from other radio signals at the point of reception. When there are long burst errors FEC cannot always recover the erroneous data. Interleaving is a technique that helps in transforming long burst errors into smaller burst error or individual bit errors so that FEC can have better chance of correcting the errors. It must be noted that interleaving does not solve the problem totally but just tries to ameliorate the problem. In a very bad case of long clustered burst errors even FEC with interleaving can fail.

ARQ techniques are good when there is a feedback channel from the receiver to the sender. When the receiver detects an error, it sends a request to the sender to resend the lost data again. However, ARQ techniques have a delay associated with it. Hence it cannot be used for multimedia data when the delay for resending the data from the sender to the receiver is high. ARQ techniques are also a challenge for point-to-multipoint type communications like multicasts and broadcasts. This is due to a possibility of request implosions, when the sender is flooded with ARQ requests from multiple receivers for retransmission of lost data.

In packet oriented networks bit errors are translated into packet errors due to CRC failures at various layers in the protocol stack. For example, a single bit error in an IP packet can result in a whole IP packet being lost. This is illustrated in Figure 1 where red indicates a lost packet and blue indicates a correctly received packet. The receiver consumes the application layer packet only when all component bits are received correctly. This can lead to large amount of correctly received bits to be discarded by the upper layers because some amount of bits is lost in the packet. The amount of discarded data depends on the packet sizes of the application layer. In general, in networks where segmentation is involved, smaller application layer packets are more reliably received than larger ones. However the disadvantage of using smaller application layer packets is that more header overheads are incurred hence leads to inefficient use of bandwidth.

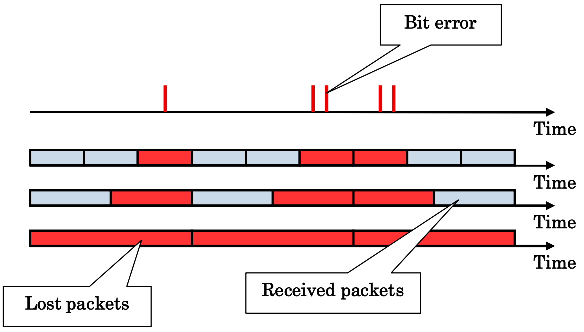

Figure 1: Effect of bit errors on different packet sizes.

3. DATACASTING IN DVB NETWORKS

DVB standards were initially designed to deliver high bandwidth multimedia contents to consumers directly to their homes. However it was later realized that the same protocols, with limited extensions, can be used to deliver other kinds of data to consumers. This broadcasting of data was called Datacasting [4]. Depending on the type of data and the applications producing and consuming the data, datacasting methods were classified into six profiles [5]. There profiles are (a) Data piping (b) Data streaming (c) Multi-Protocol Encapsulation (MPE) (d) Data Carousals (e) Object Carousals and (f) Service specific protocols. For addressable data MPE is the preferred profile.

3.1 Protocols used in DVB-H

DVB uses MPEG-2 systems standard [6] for communication of data over its transmission network. The MPEG-2 system standard uses 188 byte transport stream (TS) packets for transmission, of which 4 bytes are allocated for header information. Depending on the data that a TS packet carries, it can be broadly classified into data TS packets and information TS packets. Data TS packets carry the data to be transmitted as payload while information TS packets carry system information (SI) and program specific information (PSI) as payload. SI and PSI tables are signaling tables that describes and identifies the elementary streams forming any particular program in a multi-program TS stream. Every TS packet has a unique identifier in the form of a packet identifier (PID). TS packets with the same PID belong to the same data source or a particular signaling table. In other words a sequence of TS packets of the same PID can be considered as one data stream.

Addressable data, such as IP data, requires an intermediary translator that can understand the addressable data protocol and transform it into a form that can be understood by a DVB network. MPE is used as this intermediary translator in DVB networks. The encapsulation is done using Digital Storage Media – Command and Control (DSM-CC) protocol for private data. MPE closely resembles the IEEE local area network/ metropolitan area network (LAN/MAN) standards.

IP is a popular protocol and is the backbone of communications in the Internet. From a modest beginning in the early 1970’s, the Internet today has grown into an immense world wide network. This growth was, at least in part, fueled by the variety and number of applications using the internet. Today there is a huge amount and variety of data being produced, transmitted and stored across the world. Convergence of networks has become the new network technology buzzword largely due to the Internet revolution.

DVB-H too has moved the convergence way. It uses the IP/UDP/RTP protocols to encapsulate the data into MPE sections before transmission over the DVB-H channel. This convergence means that Internet contents can now be used with out any substantial complexity to be broadcasted over DVB-H. The network providers have a huge content base and the content providers have a high bandwidth network for delivery of their contents.

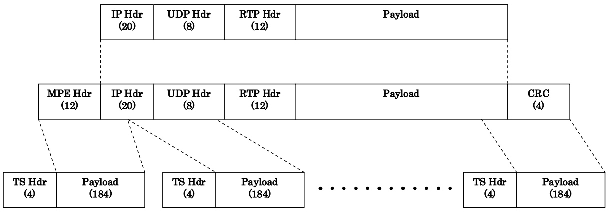

Figure 2: Encapsulation of IP/UDP/RTP packets into MPE sections and its fragmentation into TS packets.

Figure 2 illustrates how an IP packet using User Datagram Protocol (UDP) and Real time protocol (RTP) is encapsulated in an MPE packet and then fragmented into TS packets. The length of IP packets are such that they do not exceed the maximum transfer unit (MTU) for the payload portion of the datagram section. The MTU used in DVB data casting is 4096 bytes. When encapsulating an IP packet into MPE section, an overhead of 16 bytes is incurred for the section header and a further 16 bytes for the cyclic redundancy code (CRC).

3.2 MPE-FEC

To provide better data transmission reliability additional protection in the form of RS FEC computed over IP packets was included in the DVB-H standard. This optional multiplexer layer FEC is called MPE-FEC.MPE-FEC is computed in a matrix of 255 columns and a flexible number of rows. Row sizes of 256, 512, 768, 1024 bytes are currently defined in the standard. Each cell in the matrix hosts one information byte. The first 191 columns are dedicated to IP datagrams and possible padding. This part of the MPE-FEC frame is called the application data table (ADT). The next 64 columns of the MPE-FEC frame are reserved for the RS parity information, and called the RS data table (RSDT). The ADT can be completely or partially filled with datagrams. The remaining columns, when the ADT is partially filled, are padded with zero bytes and are called padding columns. Padding is also done when there is no more space left in the MPE-FEC frame to fill the next complete datagram. The RSDT is computed across each row of the ADT using RS (255,191). It is not necessary to compute the entire 64 columns of the RSDT and some of its right-most columns could be completely discarded and this procedure is termed puncturing. The padded and punctured columns are not sent over the channel.

4. SIMULATIONS

Simulations to experiment with different IP packet sizes and evaluate the optimal packet size range that provides efficient transmission both in terms of bandwidth and error resiliency were carried out. The DVB-H channel simulation assumed a code rate of 3/4 (191 data columns and 64 RS columns).The simulations considered two row sizes: 512 and 1024. The row size of 1024 has a larger interleaving capability than 512 bytes. The maximum transfer unit (MTU) size in DVB-H is governed by the maximum size of an MPE section, which is 4096 bytes. Data bit rate was set to 128 Kbps. The different IP packet sizes simulated were 184, 200, 368, 512, 600, 1024, 1536, 2048, 2560, 3072, 3584, and 4096 bytes. The simulations assumed that all packets were of equal length.

4.1 Simulation of channel errors

Three TS error patterns E1, E2, and E3 were generated by approximating a typical urban (TU6) channel. Some statistics such as the TS packet error rate (PER), the average TS burst error length (ABEL) and the variance in TS burst error length (VBEL) of these error patterns are listed in Table 1.

Table 1: ABEL and VBEL for TS error patterns simulated.

Pattern	TS PER	TS ABEL	TS VBEL
E1	0.0610	12.6919	160.1943
E2	0.0718	13.2770	179.1752
E3	0.0949	14.6697	240.8363

16-QAM modulation was used for generating the TS error patterns. The guard interval was set to 1/4 and the Doppler frequencies were 79Hz, 80Hz, and 82 Hz respectively.

4.2 Efficiency of transmission

Assume that the application packet size P^{App} (IP/UDP/RTP packet) is segmented into TS packets of size P^{TS}. Let R be the packet-to-segment ratio i.e. $R = P^{App}/ P^{TS}$. If statistically independent TS packet loss is assumed, then the application packet loss before FEC decoding is given by $p_{app}= 1 – (1 –p_{ts})^R$, where p_{ts} is the loss rate of the TS packet. It can been seen from the definition of p_{app} that as R becomes larger p_{app} also increase. The paper's objective is to evaluate the optimal value for P^{App} while minimizing $p_{app.}$

If b_t is the total amount of IP data bytes (inclusive of IP/UDP/RTP headers) transmitted, b_r is the correctly received IP data bytes at the receiver, and b_o is the amount of overhead bytes incurred due to TS headers, MPE section header and MPE-FEC, then for efficient transmission, b_r has to maximized and b_o has to be minimized**.** Efficiency e can then be defined as the ratio $e=b_r/(b_t+b_o)$. A plot of the efficiency versus IP packet size will show the optimal IP packet size for any given input bit rate.

5. SIMULATION RESULTS

Graphs showing the efficiency plots are shown in Figure 3 & Figure 4. Figure 3 plots the efficiency graph for a 512 row MPE-FEC matrix, while Figure 4 plots it for 1024 row MPE-FEC frame.

From the plots it can be easily seen that packet sizes in the range of 1024 to 2048 bytes show the most efficient transmission of data to the receivers. It can also be noticed that as the error rate of the channel increases, the optimal packet sizes tend to get smaller.

Another fact that can be noticed from the graphs is that the transmission efficiency for a 1024 row MPE-FEC matrix is better than that for 512 row MPE-FEC frame. This result can be explained by the fact that a 1024 row MPE-FEC frame has a better interleaving capability than a 512 row MPE-FEC frame. Since interleaving helps in spreading burst errors, converting them into smaller burst errors or isolated errors, the MPE-FEC protection has a better chance at protecting the data.

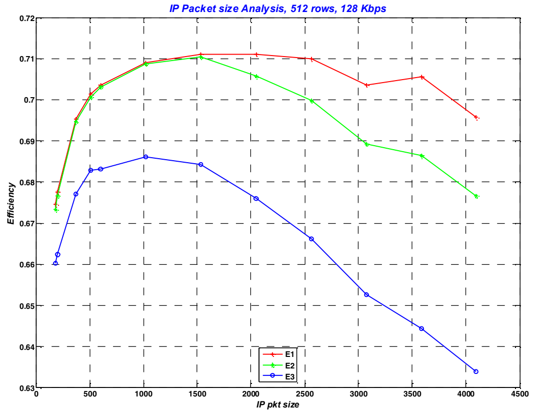

Figure 3: Efficiency plots for different IP packet sizes protected with a 512 row MPE-FEC frame.

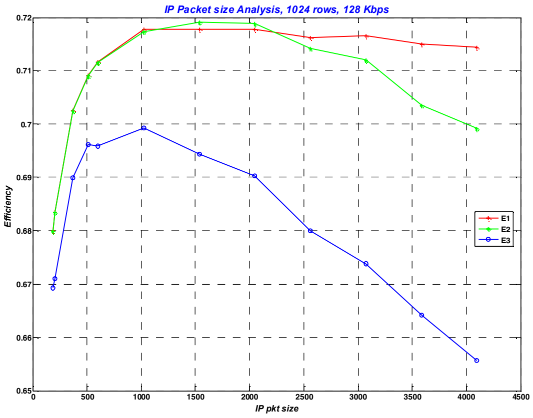

Figure 4: Efficiency plots for different IP packet sizes protected with a 1024 row MPE-FEC frame.

6. LIMITATIONS

This study did not depend on the type of data being transmitted and the quality of reception (measured as efficiency) depended only on the amount of data received at the receiver. However, for multimedia data, where quality is defined by other criteria, such as the subjective quality perception of the user, the optimal packet sizes could vary. Furthermore, the simulations also assumed that all packets in the stream were of equal length which is normally not true for multimedia data.

7. CONCLUSIONS

This paper provided a study of the effect of different internet protocol (IP) packet sizes on the bandwidth efficiency and the reliability of the channel. Even though the maximum transfer unit (MTU) of an Multi-protocol encapsulated (MPE) section is 4096 bytes, it is found that under error conditions as simulated in this paper, a packet size ranging from 1024 bytes to 2048 bytes provide optimal efficiency in terms of data bandwidth and error resiliency. Comparative graphs between the two row sizes simulated also show that a 1024 byte row size is better compared to a 512 byte row size. This is due to the bigger interleaving capability of larger row sizes.

REFERENCES

[1] ETSI, “Digital Video Broadcasting (DVB): Transmission systems for handheld terminals,” ETSI standard, EN 302 304 V1.1.1, 2004.
[2] ETSI, “Digital Video Broadcasting (DVB): Framing structure, channel coding and modulation for digital terrestrial television.” ETSI standard, EN 300 744, 2001.
[3] J. Postel, “RFC 791: Internet protocol,” Sept. 1981.
[4] ETSI, “Digital Video Broadcasting (DVB): Specification for Data Broadcasting,” ETSI standard, EN 301 192, 2001.
[5] ETSI, “Digital Video Broadcasting (DVB): Implementation guidelines for data broadcasting,” ETSI standard, TR 101 202 V1.2.1, 2003.
[6] ISO/IEC 13818-1: “Information technology – Generic coding of moving pictures and associated audio information: Systems,” Nov. 1994.