IP Over Terrestrial ATSC DTV Channels: Performance Evaluations on Data Transmission Throughput

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Abstract—**The throughput of IP (UDP and TCP) data transmission over terrestrial ATSC DTV channels was evaluated. A testbed was developed for this purpose. Various issues affecting the throughput of IP data transmission over DTV channels were analyzed, such as channel noise, traffic load, IP packet size and return link capacity. Results obtained from these tests offer valuable information for broadcasters who plan to implement datacasting services over ATSC DTV systems.**

Index Terms—**ATSC, datacasting, IP over DTV, performance evaluation, throughput analysis.**

I. Introduction

THE ATSC digital television (DTV) system [1] was designed to carry a mixture of standard definition television (SDTV) and/or high definition television (HDTV) programs and their associated sound and ancillary data information in a single 6 MHz terrestrial broadcasting channel. It can also carry IP data encapsulated into an ATSC transport stream (TS) [2], [6]. This feature, combined with a high data rate of 19.39 Mbps, a flexible transport mechanism and wide area coverage, results in a very powerful system useable for the delivery of a wide range of one- and two-way IP-based services to fixed and portable devices.

Considering the rapid convergence of IP data with a large number of new and existing communication systems such as the ATSC DTV system, extensive studies will be required to understand the behavior of IP data transmission over a multitude of communication channels having different channel coding and propagation characteristics. To this effect, the Communications Research Centre has initiated a valuable study characterizing IP data transmission over an ATSC DTV infrastructure in order to complement earlier tests performed to characterize the performance of the ATSC system in broadcasting television programs [4], [5], [19]–[21].

In order to carry out this work, we developed an ATSC datacasting testbed consisting of two main components: (1) a laboratory ATSC downlink and return channel infrastructure which serves as the IP data carrier, and (2) an IP traffic generator and recorder sub-system. The study covers a number of key issues such as network protocol parameters affecting the throughput of IP data transmission, IP and MPEG encapsulation overhead, the influence of the return channel capacity, Program Specific Information (PSI) and NULL packet insertion in the transport stream multiplexer, and errors caused by channel impairments such as noise.

The organization of the rest of this paper is as follows. In Section II we describe the ATSC datacasting test system and its configuration. In Section III, issues limiting the throughput of IP data transmission are analyzed. Test results and discussions are presented in Section IV. Conclusions and future work are summarized in the final section.

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II. IP Over Terrestrial ATSC DTV Channels: Test System

Previous studies on the performance assessment of the ATSC DTV systems have focused mainly on television broadcasting [4], DTV reception characterization [5], [9], [22], [23] and channel behavior modeling and analysis [10], [20]. With an increasing demand for fixed and mobile interactive data services, IP data traffic analysis and characterization are becoming more and more important in the context of the ATSC DTV systems. To date, little work has been carried out in this field.

In order to fill this gap, we developed a testbed to evaluate IP data transmission over ATSC. This testbed consists of a transmission subsystem for sending IP data to the receiver (client PC); a reception subsystem for data reception and interactive requests; and a channel simulation/analysis sub-system for simulating IP traffic and transmission channel impairments. Supporting software and hardware have been developed or acquired for IP traffic generation, data collection, status analysis and RF channel noise simulation. The testbed is depicted in Fig. 1 and each component is described in detail in the remainder of this section.

A. Transmission Subsystem

The transmission subsystem combines various data, compressed video and audio, and other multimedia information onto a MPEG2 transport stream (MPEG2-TS), which is then modulated to radio frequencies (RF) for terrestrial transmission. In this testbed, we employ the same transmission subsystem previously described in [3]. This subsystem consists of a Linux-based data server, an IP encapsulator (IPE), a transport stream multiplexer (TSM) and an 8-VSB modulator/transmitter. The IPE encapsulates the IP data coming from the server/gateway into MPEG2-TS according to the Multi-Protocol Encapsulation (MPE) method of the ATSC Data Broadcast Standard A/90 [2]. Once the IP data is converted to

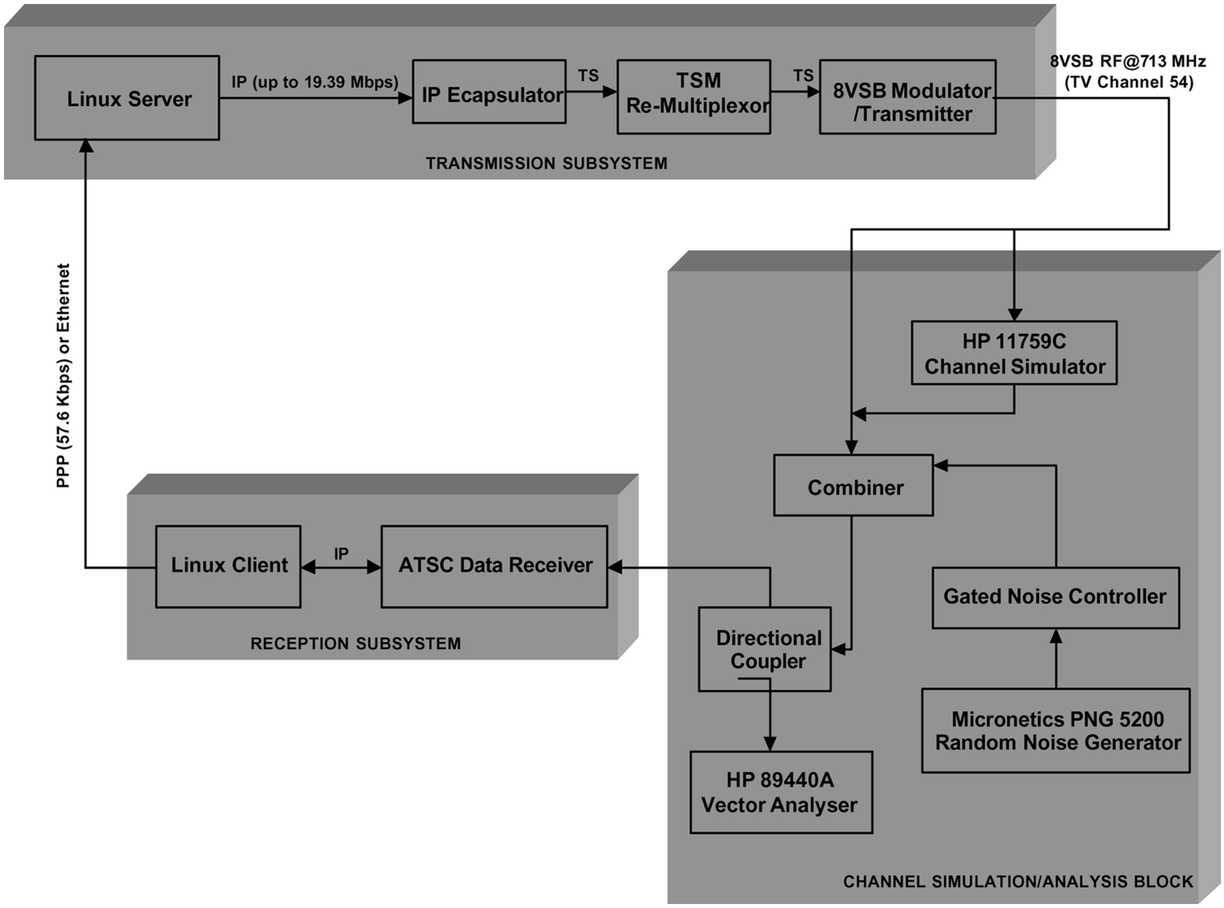

Fig. 1. Block diagram of the test system.

MPEG2-TS, it is sent to the TSM where other MPEG2 transport streams can be multiplexed. Several services and programs can be multiplexed together as long as the total data rate does not exceed 19.39Mbps. The signal from the multiplexer is then fed to the 8VSB modulator/transmitter and broadcasted to the receivers.

B. Reception Subsystem

Data reception and interactivity is realized via the reception subsystem. The ATSC data receiver demodulates the received RF signal, recovers the MPEG2-TS, extracts the data types and dispatches the packets corresponding to the selected program identification numbers (PID's) to the client applications. The interactivity between the client PC and the Linux server can be accomplished via a number of different technologies such as the IEEE 802 family of wireless LAN, Public Switched Telephone Network (PSTN) or cellular phone, depending on the application. We are currently investigating the use of the DVB-RCT (Return Channel Terrestrial) system [15] as a possible return channel option for the ATSC downlink system.

For the purpose of these tests, we have used two wired return links: one using the Point-to-Point Protocol (PPP) simulating a telephone line return link, and the other a 100 Mbps Ethernet connection simulating a return link with no throughput restriction. Two components serve the interactive connection: the Linux client and the data server. The Linux client sends requests to the data server through the PPP connection at a rate of 57.6 Kbps over a serial line or through the Ethernet connection at a rate of 100 Mbps.

C. Channel Simulation Block and Software Tools

To construct a complete test system, supporting hardware components and software tools were added. The following tools were integrated into this testbed:

1) Channel Simulation/Analysis Block: One function of the RF channel simulator is to allow the addition of a controlled level of noise to the transmitted signal in the downlink subsystem. The simulator consists of a Random Noise Generator, a Gated Noise Controller (attenuator), a Combiner and a Channel Simulator, the latter of which is not used in our study. The RF channel simulator is capable of generating and injecting Additive White Gaussian Noise (AWGN) at different levels onto the transmission channel.

2) Software Tools: The network traffic generation software running on the server generates TCP, UDP, or HTTP traffic of different inter-departure time intervals, sizes and distributions. The IP traffic generator software tools that were available for this research include: the Distributed Internet Traffic Generator (D-ITG) [11], the Multi-Generator Toolset (MGEN) [12], the Real-Time Data Emitter (RUDE & CRUDE) [13]. We adopted the D-ITG software for the throughput analysis tests.

The D-ITG software also needs to be installed in the client PC in order to capture and analyze the incoming packet status, as well as acquire network information relating to the throughput of IP data transmission.

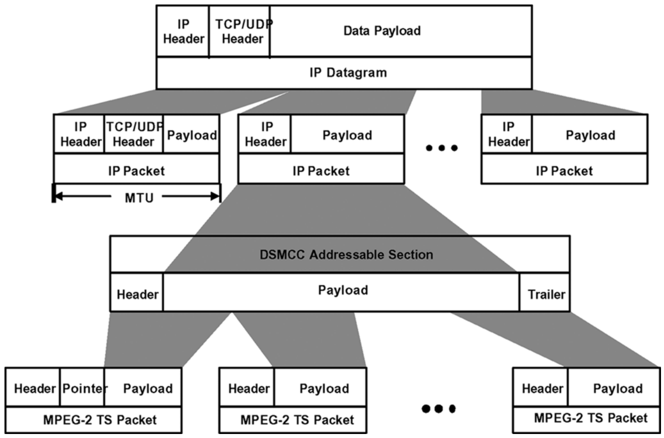

Fig. 2. Encapsulation of IP datagram into TS packets with the addition of multi-tier overhead.

III. IP Data Transmission Throughput Over ATSC DTV Channels: Restriction Issues

Several factors limit the throughput of IP data transmission over terrestrial ATSC DTV channels. These include network and system overhead, the multiplexer's PSI and NULL packet insertion, the return link speed limitations, etc. We discuss these issues in the following sub-sections.

A. MPEG2 Overhead for Transporting IP Packets

A TS packet consists of a header and payload portions. The overhead reduces the throughput, defined as the maximum error-free transmission rate for the data payload [7]. From the protocol stack of IP data transmission over ATSC channels illustrated in Fig. 2, we can see the multi-tier overhead added to the to-be-transmitted data payload. A quantitative assessment of the overhead occupancy over the total throughput can provide us with a better understanding of the efficiency of the ATSC system to transport IP data.

As we know, the Internet Protocol adds 20 bytes of overhead to a TCP/UDP packet. TCP adds 20 bytes and UDP adds 8 bytes to data packets. If the data payload size (in bytes) is denoted by D, the TCP/IP, UDP/IP overhead H equations are:

$$H = \frac{20+20}{20+20+D} \text{ for TCP} \tag{1}$$

$$H = \frac{8+20}{8+20+D} \text{ for UDP} \tag{2}$$

In the context of MPEG2 data transmission using the Digital Storage Media-Command and Control (DSM-CC) protocol, a two-layer structure is adopted, consisting of a DSM-CC section layer and a transport layer [18]. The overhead of the DSM-CC section layer in transporting IP data is as follows [8]:

- An ATSC DSM-CC addressable section header, which includes: a section header of 3 bytes and a Multi-Protocol Encapsulation (MPE) header of 9 bytes;
- A CRC checksum of 4 bytes;

resulting in a DSM-CC section overhead of 16 bytes.

For the transport layer, each MPEG2 transport packet contains a 4-byte header and a 184-byte payload. When a new section commences (i.e. a new IP packet arrives), the first byte of the MPEG2 TS packet's payload field contains a pointer field of one byte, and so will have only 183 bytes of payload.

When section packing is not used, if the data of an IP packet do not completely fill the payload part of an MPEG2 TS packet, the remaining bytes of the TS packet are filled with NULL bytes (0xFF, also referred to as padding bytes). The following IP packet then starts with the next TS packet. Let I stands for the IP packet size, n is defined as $n = \|(I+16+1/184)\|$, where $\|x\|$ denotes the smallest integer larger than x. The number of padding bytes will then be $184-[I+16+1-(n-1)\times 184] = 184-[I+17-(n-1)\times 184]$.

Thus, the overhead will be:

$$H = \frac{17+n\times 4+184-[I+17-(n-1)\times 184]}{n\times 188} \tag{3}$$

Based on the above equation, the overhead for data transmission with TCP/IP is:

$$\begin{aligned} H &= \frac{17+n\times 4+184-[D+40+17-(n-1)\times 184]}{n\times 188} \\ &= \frac{201+n\times 4-[D+57-(n-1)\times 184]}{n\times 188} \end{aligned} \tag{4}$$

Similarly, we have the overhead for data transmission with UDP/IP:

$$H = \frac{201+n\times 4-[D+45-(n-1)\times 184]}{n\times 188} \tag{5}$$

With section packing enabled, the new IP packet immediately follows the last data byte of the previous IP packet, within the same TS packet. The section layer overhead becomes independent of the transport layer overhead, except for the pointer field, which occurs at most once on each transport packet. We then have two cases:

a. If $I \geq (183-16)$, there will be one pointer per IP packet, so we can add the one-byte overhead to the section overhead. The transport overhead will be 4 bytes per transport packet, i.e. the transport layer adds an overhead of $(I+16+1/184)\times 4$ per IP packet. The total overhead is then:

$$H = \frac{17+\left(\frac{I+17}{184}\times 4\right)}{I+17+\left(\frac{I+17}{184}\times 4\right)} \tag{6}$$

b. If $I < (183-16)$, there will be one pointer per transport packet, and we can add the one-byte overhead to the transport overhead, making it 5 bytes per transport packet. Therefore the total overhead is:

$$H = \frac{16+\left(\frac{I+16}{184}\times 5\right)}{I+16+\left(\frac{I+16}{184}\times 5\right)} \tag{7}$$

With these equations, the throughput occupancy rate of IP encapsulations over MPEG2-TS can be quantitatively evaluated. The overhead occupancy rate (H) varies with the data size (D or I). Table I lists the overhead occupancy rate (over the total throughput) of IP data over MPEG-TS with section packing disabled.

Based on these equations, the data transmission throughput can also be deduced, since the ATSC TS data rate is 19.39 Mbps.

TABLE I
OVERHEAD OCCUPANCY RATE (%) OF IP OVER MPEG-TS

data payload size (byte)	64	192	384	512	640	768	1024
UDP	44.7	38.3	24.8	26.6	9.6	14.0	5.7
TCP	51.0	41.5	27.0	28.2	11.2	15.3	6.7

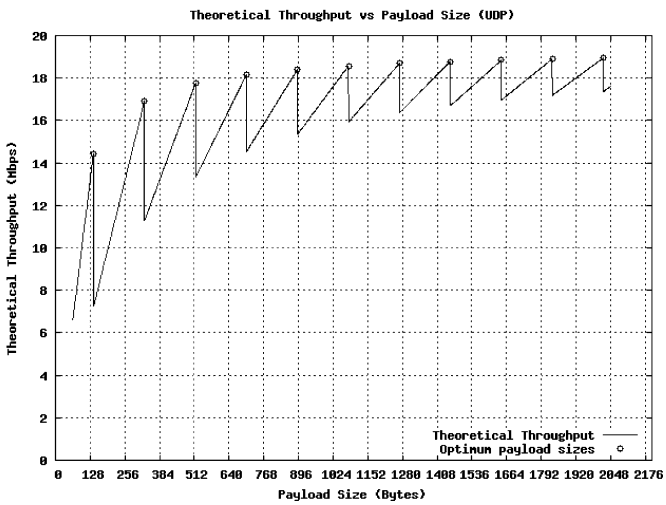

Fig. 3. Theoretical analysis of throughput vs. payload size.

A theoretical analysis of UDP data transmission throughput vs. payload size in the case without section packing is drawn in Fig. 3. The periodic drop in throughput that we observe in Fig. 3 is the price to pay for adding an extra byte to the payload, which results in the transmission of an extra TS packet carrying only one payload byte.

B. Return Channel Capacity for TCP-Like Data Transmission

For TCP-like data transmission, such as TCP/HTTP/FTP applications where received packets must be acknowledged, the overall throughput is affected by the return channel speed. For example, when packets are lost during transmission, the receiver issues retransmission requests to the server through the return channel. A low-speed return channel, such as PPP, will certainly be disadvantageous compared to a higher speed one (e.g. a DVB-RCT or a wireless LAN return channel). The throughput constraint resulting from the rate limitations of the return channel will be investigated in the next section describing our test results.

C. Packets Added by the Transport Stream Multiplexer

The TSM will normally add two types of data packets to the MPEG2-TS by (1) periodically inserting MPEG-2 Program Specific Information (PSI) packets into the transmission stream, including the Program Association Table (PAT) and Program Map Table (PMT), and by (2) inserting NULL packets into the transmission channel to maintain a stable output bit rate of 19.39 Mbps at the output of the TSM. Due to the addition of these packets by the TSM, the input data rate to the TSM must be less than the prescribed data rate of the system. How much additional data is being generated at the Transport Stream Multiplexor will be investigated in the following section.

D. Channel Noise

The ATSC standard relies on sophisticated channel coding as well as error concealment techniques implemented in the receiver to ensure high-quality audio and video reception in the presence of transmission errors. The packet loss rate can increase from 0 to 100% with a variation of only 1 dB of the carrier noise ratio (C/N) [14]. Packet losses result in throughput reduction at the receiver. The effect of channel noise on throughput will be studied in the following section.

IV. TEST RESULTS AND DISCUSSIONS

In this section, we present the results of various throughput tests with TCP and UDP traffic. Following a brief introduction of the experimental environment, we start with the throughput comparison with different return channel capacity; followed by an analysis of the effect of TSM packet stuffing; and finally with the system throughput analysis with and without the presence of channel noise. For the tests that were performed, the transmission TCP packet sizes are 32, 64, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536 and 1664 bytes.

A. Experimental Environment

Our analysis is based on the collection of TCP and UDP packet traces. These traces are saved as log files in the server or client computer for later analysis. In analysing TCP data transmission throughput, both server and client logs are required. For UDP related throughput tests, client logs are sufficient. Data traffic flows from server to client through the downlink channel. Control traffic such as requests and acknowledgment signals are from client to server via the uplink or return channel.

To simulate the real ATSC DTV transmission environment, the RF Modulator is tuned to 713 MHz corresponding to DTV channel 54. The attenuator in the RF simulator is calibrated to have a power attenuation of approximately -53 dBm, which is typical of the power level of TV channels used in laboratory simulations. To maximize the data transmission throughput, the IPE is allocated a data rate of 19.39 Mbps, which equals the maximum ATSC data rate.

Data throughput is one of the important metrics in the performance characterization and evaluation of data transmission over the DTV system. Throughput analysis can be performed by changing parameters such as RF channel noise, packet size and packet inter-departure time.

B. Throughput Analysis With Different Return Channels

The return channel capacity is expected to affect the reception throughput of TCP-like data transmission. This is explored in the following test: TCP data is transmitted from the server to the receiver through the downlink subsystem. The uplink subsystem is configured to use either a slow speed link such as a wired PPP link or 3 G wireless phone with a data rate of 57.6 Kbps, or an Ethernet LAN with a data rate of 100 Mbps. The results are depicted in Fig. 4. In this figure, we observe a much higher throughput (around 16.5 Mbps) with the Ethernet LAN return link than with the PPP return link (around 7.2 Mbps). The throughput of both return links are limited by the fact that

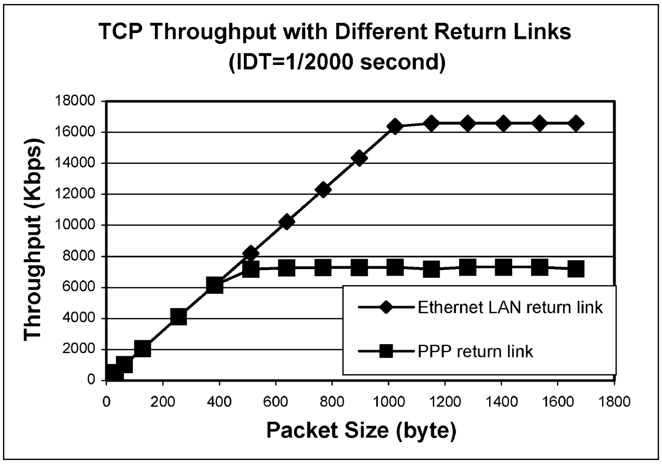

Fig. 4. Throughput comparison with different return links.

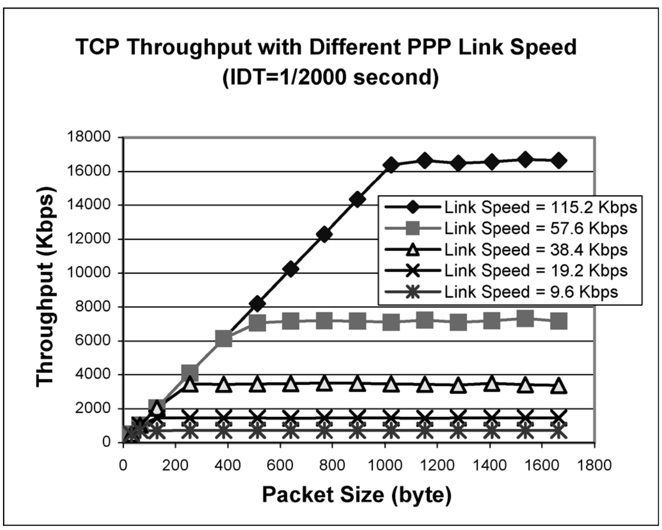

Fig. 5. Throughput comparison with different PPP link speeds.

the server must wait for acknowledgment (ACK) of packet reception from the receiver before sending more data. The rate at which the return channel can transmit acknowledgment will thus affect the downlink throughput. In the case of packet loss, the receiver has to issue retransmission requests in addition to the ACK signaling to the server through the return channel.

It is beneficial to characterize the minimum return channel capacity required to support full ATSC downlink capacity of 19.39 Mbps. For this purpose, tests on TCP downlink throughput with respect to different PPP return link speeds were carried out and the results are depicted in Fig. 5. As can be seen, higher return link speed leads to higher downlink throughput. With a return link speed of 115.2 Kbps, the downlink throughput reaches 16.5 Mbps, which is virtually equivalent to the Ethernet return link.

C. Throughput Affected by TSM Packet Stuffing

Tests to study the impact on the throughput of the PSI and NULL packet insertion in the TSM are described in this section. These tests aim at determining how much overhead is introduced by the TSM packet stuffing mechanism. Throughput

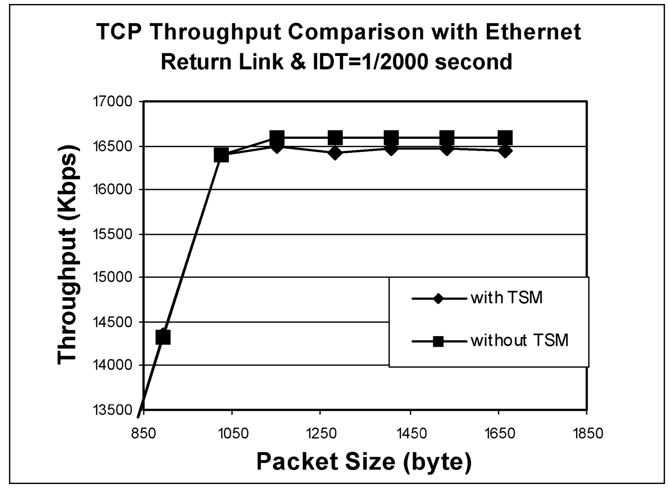

Fig. 6. TCP throughput comparison with and without TSM.

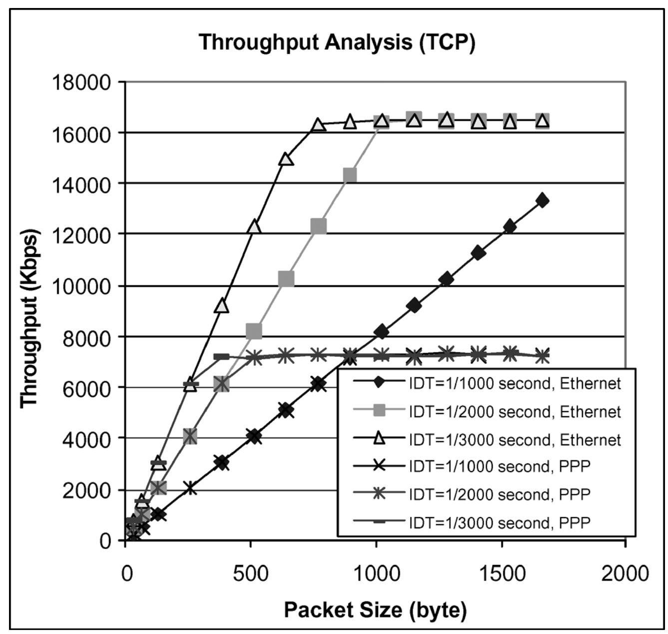

Fig. 7. Throughput at the receiver for TCP traffic.

comparisons with and without the presence of the TSM are carried out in these tests. Fig. 6 shows a comparison of TCP throughput (with Ethernet as the return link and inter-departure time of sending packets equals to 5 milliseconds) with and without the TSM. It can be seen that without the TSM, the TCP throughput is slightly higher than that with the TSM.

The average difference in throughput between these two cases is calculated to be approximately 110 Kbps, which is equivalent to a 0.56% increase in rata rate. These results are considered negligible in comparison to the system throughput. Similar differences were observed for UDP.

D. Throughput Analysis Without Channel Noise

This section analyzes issues such as overhead, return channel capacity and TSM packet stuffing on IP data transmission throughput over the ATSC DTV system. Fig. 7 shows the TCP throughput at the receiver end for a range of packet size, with

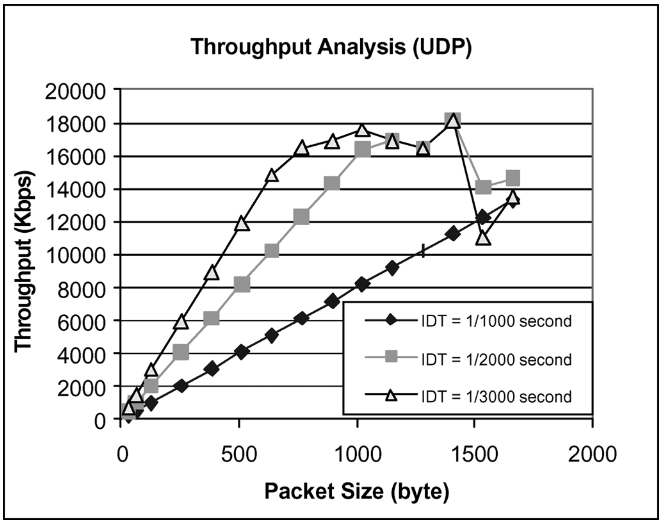

Fig. 8. Throughput at the receiver for UDP traffic.

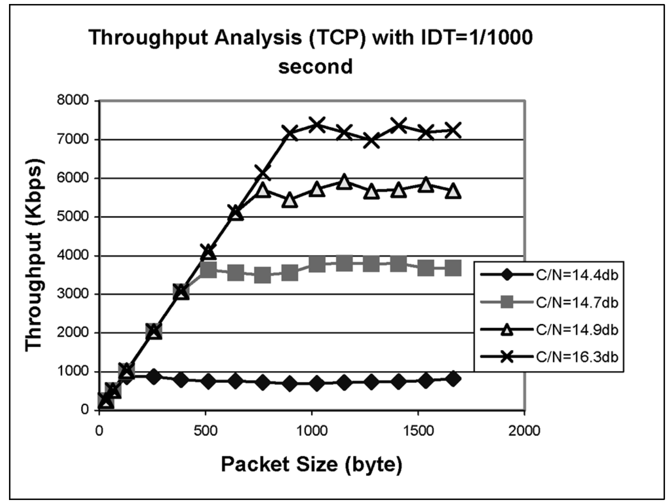

Fig. 9. TCP throughput in the presence of noise.

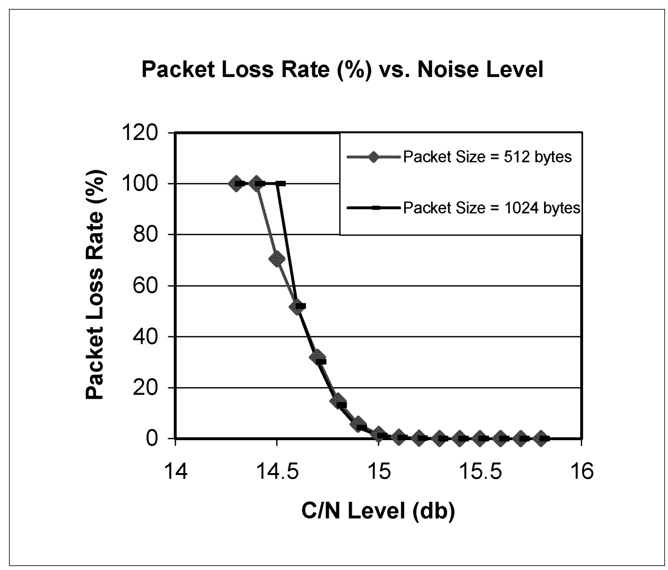

Fig. 10. UDP packet loss rate in the presence of noise.

the Ethernet and PPP return links. Three different inter-departure times (IDT) were used to schedule packet transmission: 1/1000 second, 1/2000 second and 1/3000 second.

As can be seen from this figure, the throughput increases linearly with packet size until the packet size reaches a certain level and then remains constant. For the system under test, the throughput limit for TCP data transmission is about 16.5 Mbps for the Ethernet return link and 7.25 Mbps for the PPP return link.

This is explained from the fact that in TCP data transmission, the receiver sends regular acknowledgment packets to the server via the uplink. The server waits for reception of these acknowledgment before sending new packets. A slow speed return channel can thus seriously limit the TCP throughput. This explains the results of Fig. 7, which show that the throughput is much lower with a PPP return link than with an Ethernet return link.

We also examined the throughput in the case of UDP traffic, under conditions similar to those for TCP. The results obtained are illustrated in Fig. 8. Similar conclusions can be drawn from the results: a linear relationship exists between the throughput and packet size for small packets. However, more throughput fluctuations are observed for larger packet sizes.

An interesting phenomenon is observed when packet sizes are over 1500 bytes and IDT is 1/2000 or 1/3000 seconds: we note an important drop in throughput. This can be explained by the fact that IP packet fragmentation occurs when the transmitted packet size exceeds the Maximum Transfer Unit (MTU) parameter of the network, resulting in extra packets being generated for each packet containing more than MTU bytes thus exceeding the maximum throughput prescribed by the system and causing more packets to be dropped. These dropped packets will ultimately cause the throughput to be lower than expected. For the case where IDT equals 1/1000 second, a drop in throughput is not observed when the packet size exceeds 1500 bytes because there is still enough capacity to transmit data even though packet fragmentation occurs.

A key issue that applies for UDP data transmission is that when the total transmission rate reaches levels higher than 19.39 Mbps, system components (IPE and/or TSM) will drop packets to respect the throughput limit imposed by the system. In this case, the throughput measured at the receiver will show saturation at the maximum achievable rate and packets will be lost resulting in the degradation of the service being rendered. Taking that into consideration, a system designer has to be careful when selecting system parameters such as service packet size and throughput in order to avoid or minimize service degradation.

Given the insignificant contribution of the TSM packet stuffing (Section IV-C) and the throughput results obtained in this section, it can be stated that the throughput limitation of the system is mainly caused by the MPEG2 encapsulation overhead.

E. Throughput Analysis in the Presence of Channel Noise

Figs. 9 and 10 respectively illustrate the TCP throughputs and UDP packet loss rate in the presence of channel noise. As can be seen from these figures, the throughput or packet loss rate measured at the receiver is very sensitive to the variation in the carrier to noise ratio (C/N). For example, in the case of TCP data transmission (Fig. 9) the throughput increases from less than 1 Mbps to almost 6 Mbps within a C/N variation of 0.5 dB, i.e. from 14.4 to 14.9 dB. For UDP data transmission at 17 Mbps as seen in Fig. 10, the impact of C/N on the packet loss rate is very dramatic, following what is termed a nongraceful degradation. A C/N variation of 0.5 dB is all that is required to go from full reception (no packet loss) to no reception at all (100% packet loss). The packet loss rate versus C/N test that was performed for UDP is in accordance with the analysis given in [14] regarding the performance characteristics of terrestrial broadcast.

V. Conclusions and Future Work

We presented a testbed designed to characterize the performance of IP data transmission over terrestrial ATSC DTV channels. The testbed consists of a transmission subsystem for downlink data broadcasting, a reception subsystem for data reception and uplink control, and a channel simulation/analysis part for channel noise simulation.

Issues affecting the throughput of IP data transmissions were analyzed. TCP and UDP data transmission throughputs subject to RF channel noise and traffic load were investigated in various scenarios. We noted three major factors affecting the throughput of data transmission in the ATSC DTV datacasting systems: (1) the IPE bandwidth allocation, limited by the ATSC data rate; (2) the return channel capacity; and (3) the packet sizes and interdeparture times. We analyzed the theoretical MPEG2 transport stream overhead for transporting IP packets.

We determined the throughput limits of TCP and UDP data transmission over DTV channels under normal conditions: a 57.6 Kbps PPP return channel imposes a downlink TCP throughput limit of approximately 7.25 Mbps. Using a 100 Mbps Ethernet LAN connection as the return link, the TCP throughput can increase to more than 16.5 Mbps. The UDP throughput has a limit of 18.1 Mbps with an average value close to 17 Mbps. We also determined that the TSM has a negligible impact on the throughput. We also confirmed that the channel noise has a significant, nongraceful degradation impact on data transmission throughput or packet loss rate.

Future work includes characterization of the system in the presence of multipath and impulse noise as well as investigations of DVB-RCT as a wireless return channel and its adaptation to the ATSC infrastructure.

References

[1] *ATSC Digital Television Standard*, ATSC Standard A/53, September 1995.

[2] *ATSC Data Broadcast Standard*, ATSC Standard A/90, July 2000.

[3] W. Li, H. Liu, and G. Gagnon, "Integration of an interactive multimedia datacasing system," in *17th International Conference on Industrial & Engineering Applications of Artificial Intelligence & Expert Systems*, Ottawa, Canada, May 17–20, 2004.

[4] Y. Wu, E. Pliszka, B. Caron, P. Bouchard, and G. Chouinard, "Comparison of terrestrial DTV transmission systems: the ATSC 8-VSB, the DVB-T COFDM, and the ISDB-T BST-OFDM," *IEEE Trans. on Broadcasting*, vol. 46, no. 2, pp. 101–113, June 2000.

[5] H. Kim, S. Park, and S. Kim, "Performance analysis of error propagation effects in the DFE for ATSC DTV receivers," *IEEE Trans. on Broadcasting*, vol. 49, no. 3, pp. 249–257, Sep. 2003.

[6] ISO 13 818-1, Information Technology –Generic Coding of Moving Pictures and Associated Audio: Systems, Nov. 13, 1994.

[7] S. Bradner and J. McQuaid, Benchmarking Methodology for Network Interconnect Devices, March 1999.

[8] *ATSC Recommended Practice: Implementation Guildlines for the ATSC Data Broadcast Standard (Doc. A/90)*, ATSC Standard A/91, June 2001.

[9] Y. Wu, X. Wang, R. Cita, B. Ledoux, S. Lafleche, and B. Caron, "An ATSC DTV receiver with improved robustness to multipath and distributed transmission environments," *IEEE Trans. on Broadcasting*, vol. 50, no. 1, pp. 32–41, March 2004.

[10] M. El-Tanany, Y. Wu, and L. Hazy, "Analytical modeling and simulation of phase noise interference in OFDM-based digital television terrestrial broadcasting systems," *IEEE-Transactions on Broadcasting*, vol. 47, no. 1, pp. 20–31, March 2001.

[11] S. Avallone, D. Emma, A. Pescapè, and G. Ventre, "A practical demonstration of network traffic generation," in *Proceedings of the 8th International Conference on Internet and Multimedia Systems and Applications*, Kauai, USA, August 16–18, 2004.

[12] MGEN—The Multi-Generator Toolset. [Online]. Available: http://mgen.pf.itd.nrl.navy. mil/.

[13] RUDE & CRUDE—The Real-Time Data Emitter. [Online] Available: http://rude. sourceforge.net/

[14] *Guide to the Use of the ATSC Digital Television Standard*, ATSC Standard A/54, Oct. 1995.

[15] "Digital Video Broadcasting (DVB); Interaction Channel for Digital Terrestrial Television (DVB-RCT) Incorporating Multiple Access OFDM,", ETSI EN 301 958 (v1.1.1).

[16] "Digital Video Broadcasting (DVB); Interaction Channel for Satellite Distribution Systems,", ETSI EN 301 790 (v1.2.2).

[17] T. Laud, M. Aitken, W. Bretl, and K. Kwak, "Performance of 5th Generation 8-VSB Receivers," *IEEE Trans. on Consumer Electronics*, vol. 50, no. 4, pp. 1076–1080, Nov. 2004.

[18] R. Chernock, R. Crinon, M. Dolan, and J. Mick, *Data Broadcasting: Understanding the ATSC Data Broadcast Standard*: McGraw-Hill, 2001.

[19] P. Angueira, M. Velez, D. de la Vega, G. Prieto, D. Guerra, J. Matias, and J. Ordiales, "DTV reception quality field tests for portable outdoor reception in a single frequency network," *IEEE Trans. on Broadcasting*, vol. 50, no. 1, pp. 42–48, March 2004.

[20] G. Guerra, P. Angueira, M. Velez, D. Guerra, G. Prieto, J. Ordiales, and A. Arrinda, "Field measurement based characterization of the wideband urban multipath channel for portable DTV reception in single frequency networks," *IEEE Trans. on Broadcasting*, vol. 50, no. 3, pp. 209–223, Sep. 2004.

[21] O. Bendov, Y. Wu, C. Rhodes, and J. Browne, "Planning factors for fixed and portable DTTV reception," *IEEE Trans. on Broadcasting*, vol. 51, no. 2, pp. 171–179, June 2005.

[22] J. Ong, H. Yan, S. Rao, and G. Shanmugan, "Indoor DTV reception: measurement techniques," *IEEE Trans. on Broadcasting*, vol. 50, no. 2, pp. 192–199, June 2004.

[23] J. Ahn, C. Lim, D. Han, and H. Oh, "Adaptive hybrid beamformer for mobile reception of ATSC DTV," *IEEE Trans. on Broadcasting*, vol. 51, no. 3, pp. 296–304, Sep. 2005.

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