Performance Assessment of IP Traffic over ATSC Interactive Datacasting Systems

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Abstract — *The analysis of IP (UDP and TCP) traffic performance over terrestrial ATSC DTV channels is carried out in this work. A testbed to study the performance of a DTV datacasting system was designed and developed for this purpose. The system characterization measurements are defined to suit the performance assessment of IP data transmission over ATSC DTV channels. Different scenarios of IP data transmission subject to channel noise and traffic load are investigated. Results obtained from these tests offer valuable information for broadcasters who intend to implement IP data broadcasting services over ATSC DTV systems, such as in the planning process of data services over ATSC DTV infrastructures.*[1]

Index Terms — ATSC Datacasting system, IP over DTV, performance analysis.

I. INTRODUCTION

The Advanced Television Systems Committee (ATSC) Digital Television (DTV) standard [1] has been adopted for over-the-air delivery of digital television programs to the homes. The ATSC system was designed to carry multiple standard definition television (SDTV) programs or a single high definition television (HDTV) program with the associated sound and ancillary data information in a single 6 MHz terrestrial broadcasting channel. The system can also carry IP data [2] encapsulated onto an ATSC transport stream (TS) [3~5]. The data transmission rate for a single ATSC terrestrial channel which shares the same spectrum as the terrestrial analog broadcasts is 19.4Mbps, resulting in a very powerful system for the delivery of IP data to a large audience.

The ATSC standard relies on sophisticated channel coding as well as error concealment techniques implemented in the receiver to ensure high-quality reception of the audio and video signals in the presence of transmission errors. Error protections are carried out by using trellis coding, interleaving and Reed-Solomon coding. Although some tests have been conducted [8~10] to characterize the system performance for television broadcasting, little has been done to characterize the ATSC system for IP data transmission. The performance assessment of IP traffic over ATSC datacasting systems is considered as a very valuable work in the following contexts:

1. Understanding the behavior of IP traffic over DTV datacasting networks.
2. Providing valuable information for broadcasters, including those who intend to implement IP data broadcasting services over ATSC DTV infrastructures.
3. Planning the network development and expansion such as WiFi reception of DTV programs.
4. Assessing the quality of network services, and attribution of network usage to end-users.

In this contribution we attempt to fill this gap by presenting a laboratory testbed for quantitative performance assessment of interactive datacasting systems as well as investigating the performance of the ATSC standard for the transmission of IP data.

In order to carry out this work, an ATSC datacasting testbed needs to be developed. The testbed consists of two essential components: one is a real ATSC interactive datacasting system which serves as the IP data carrier over terrestrial DTV channels, the other is a system which is able to inject various IP traffic into the datacasting system, subject the RF transmission path to noise and record traffic information for later analysis.

The organization of the rest of this paper is as follows. In Section 2 we briefly describe the interactive multimedia datacasting system developed at the Communication Research Centre (CRC). In Section 3, the testing system and its setup for the performance assessment of IP traffic over terrestrial ATSC DTV channels is described. Test results and discussions are presented in Section 4. The conclusions and future work are summarized in the final section.

II. INTERACTIVE MULTIMEDIA DATACASTING SYSTEM

In a typical digital multimedia terrestrial broadcast system, programs such as video, audio and data are digitally encoded and compressed either at the source or within the DTV service center. The important components of the service center include servers for stored content, network management control blocks, monitoring equipment, subscriber and billing management. All available programs are multiplexed for transportation over the distribution network. System Information (SI) tables of channel allocation and program information are added to the broadcast stream to allow tuning and selecting programs on a subscriber's premise by an electronic program guide. An antenna transmits the broadcast over-the-air to the receivers. End-user receiving equipment includes set-top box, tuner etc. Data broadcast technologies enable DTV service subscribers to interactively send requests to the service provider or broadcaster via a return channel. The return channel is usually implemented via a narrowband communication link such as a public switched telephone network (PSTN) or an integrated services digital network

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Manuscript received January 15, 2005

(ISDN). Recent implementations of interactive multimedia broadcasting systems for specific applications can be found in [3~5] and [13].

At the Communication Research Centre (CRC) an ATSC interactive multimedia datacasting system was developed [6~7]. One of the goals of the implementation of this system is to investigate, develop and demonstrate means of utilizing the DTV infrastructure to provide interactive broadband multimedia services to remote and rural areas in Canada. This system is composed of a broadcast channel (downlink) subsystem and a return channel (uplink) subsystem. It is illustrated in figure 1.

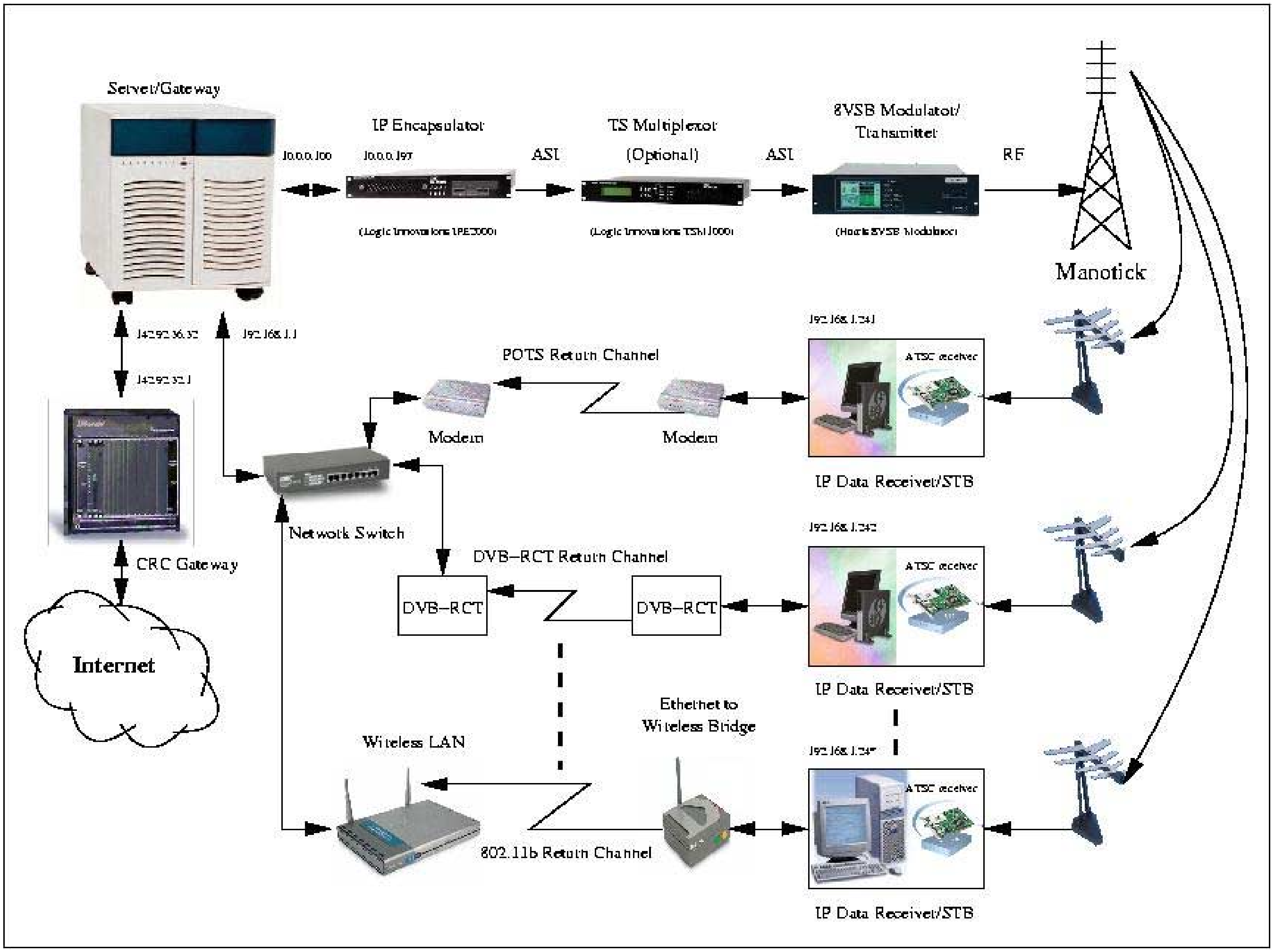

Fig. 1. The interactive multimedia datacasting system.

A. Downlink Subsystem

The downlink subsystem shown in Fig. 1 serves to combine various data, compressed video/audio, other multimedia information onto a MPEG-2 transport stream (MPEG2-TS) which is then modulated to radio frequencies (RF) suitable for terrestrial transmission. The IP Encapsulator (IPE) encapsulates the IP data coming from the server/gateway onto an 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 MPEG2-TS, it is sent to a transport stream multiplexer (TSM) where other MPEG2-TS can be added. Several services and programs can be multiplexed together as long as the 19.4Mbps limit imposed by the system is respected. The signal from the multiplexer is then fed to the 8VSB modulator/transmitter and sent over-the-air for broadcasting to the receivers. The receiver/STB demodulates the received RF signal to recover the MPEG-2 TS, extracts the media types and dispatches the demultiplexed media packets to clients accordingly. This subsystem has the advantage of being able to deliver Quality of Services (QoS) transmission by setting a maximum as well as a guaranteed data rate for each individual program identification (PID) at the encapsulator. It is important to note that each PID can serve a number of IP addresses thus allowing classification of services according to clients' requirements.

B. Uplink Subsystem

For the uplink subsystem, various options such as IEEE 802.11b wireless LAN or the Public Service Telephone Network (PSTN, POTS) can be adopted depending on required reach and available budget. Investigations about how to provide a return channel using available channels in the television UHF band is ongoing since the current North American DTV infrastructure does not provide a return channel to support interactive applications. In Europe, a return channel is part of the DVB system with currently two possible return channel sub-systems being defined: DVB-RCT (Return Channel Terrestrial) and DVB-RCS (Return Channel Satellite).

III. TESTING SYSTEM, TOOLS AND EXPERIMENTAL METHODOLOGY

Previous researches on performance assessment of the ATSC DTV systems were mainly focused on non-interactive television broadcasting [8~9], DTV reception characterization [10] [14~15], channel behavior modeling and analysis [16] etc. With more demands for interactive data services, IP data traffic analysis and characterization becomes more and more important in the context of the ATSC DTV systems. As of this writing, little work has been carried out in this field.

In order to fill this gap, a test system has been designed and implemented specifically for assessing IP traffic over ATSC datacasting systems. This testbed has a downlink subsystem used as a transmission chain and RF path for sending IP data to the receiver and, an uplink subsystem for service or data requests and control. Supporting software and hardware are needed for IP traffic generation, data collection, status analysis and RF channel noise generation. This testbed is depicted in Figure 2 and each component is described in detail in the rest of this section.

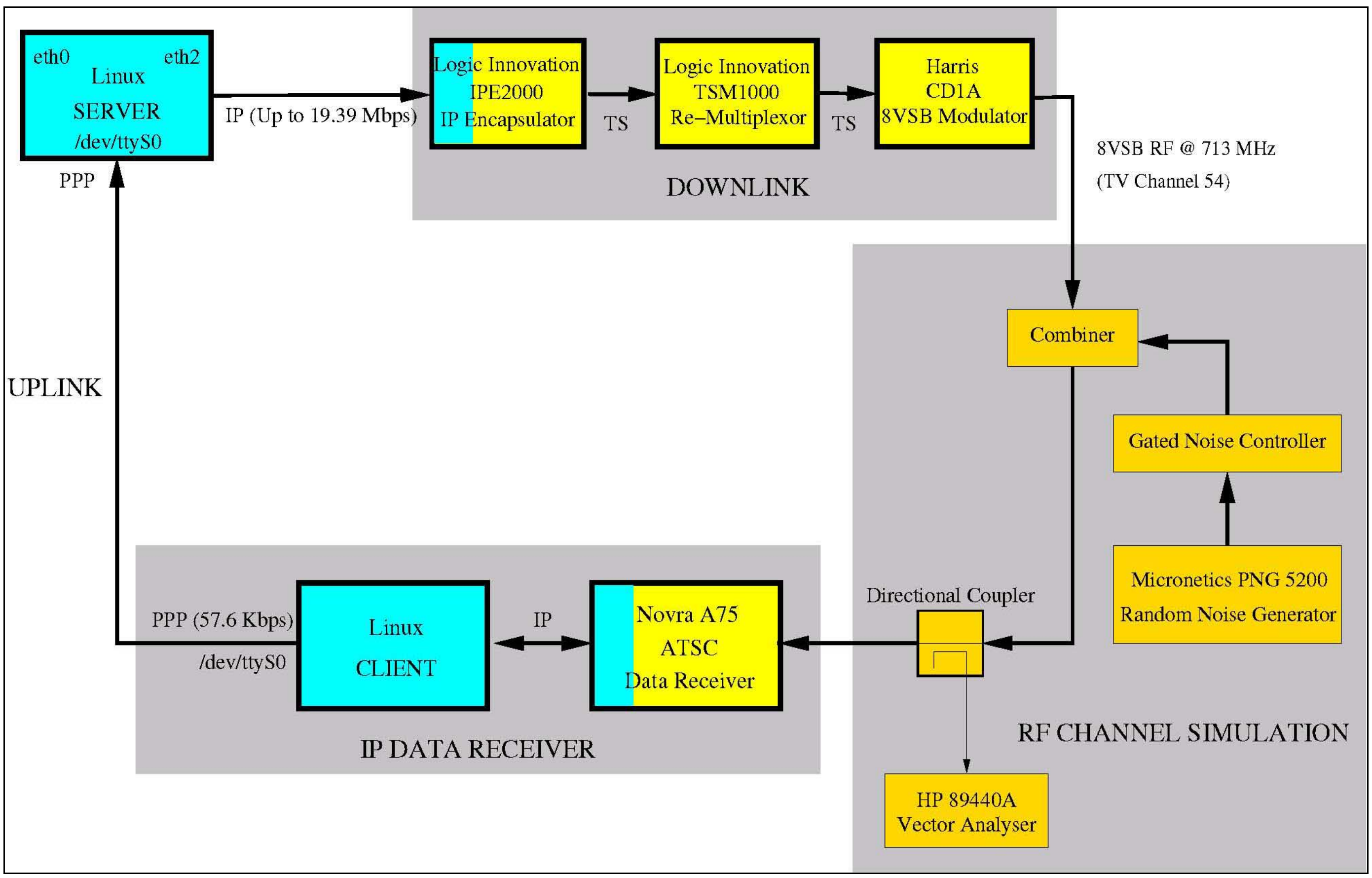

Fig. 2. Block diagram of the testing system.

A. Downlink Subsystem

The downlink subsystem consists of a data server based on a Linux PC, an IP encapsulator (IPE), a transport stream re-multiplexer (RE-MUX), an 8-VSB modulator/transmitter (MOD), an ATSC data receiver and a Linux client PC. This subsystem is used to generate, transmit and receive IP data encapsulated in an MPEG-2 transport stream [11] over an off-air television channel through an 8VSB modulator /transmitter. The receiving end demodulates the received RF signal, recovers the MPEG-2 TS, extracts the data types and dispatches the packets to the client applications.

B. Uplink Subsystem

The uplink subsystem can be based on a number of different technologies, such as wireless LAN or DVB-RCT (Digital Video Broadcasting - Return Channel Terrestrial) as shown in figure 1. For the current datacasting performance assessments, we configure a wired PPP to serve as the uplink connection in order to simulate a return channel using telephone lines (PSTN). Thus, only two components form the uplink subsystem: the Linux client and the data server /gateway. The Linux client sends requests to the data server/gateway through the PPP connection at a rate of 57.6 Kbps over a serial line.

C. Experimental Tools

To construct a complete test system, we need to add supporting hardware components and software tools. The following tools are integrated into our testbed:

1) Hardware components

a) RF Channel Simulation

The RF channel simulator serves as adding controlled channel noise to the transmitted signal in the downlink subsystem. This function block consists of a Random Noise Generator, a Gated Noise Controller (noise attenuator) and a Combiner. The RF channel simulator is capable of generating and injecting Additive White Gaussian Noise (AWGN) at different levels onto the transmission channel. With this block, controlled channel noise perturbation and propagation attenuation can be simulated.

b) Vector Analysis

A vector analyzer is required for the accurate calibration of noise attenuators and the measurement of carrier/noise levels.

2) Software tools

The network traffic generation software running on the Server is used to generate TCP/UDP/HTTP traffic of different inter-departure time intervals, sizes and distributions. For our test purposes, we made use of a number of IP traffic generating softwares and applied them appropriately for different testing scenarios. The frequently used IP traffic generators include: Distributed Internet Traffic Generator (D-ITG) [17], the Multi-Generator Toolset (MGEN) [18], the Real-Time Data Emitter (RUDE & CRUDE) [19] etc.

The client computer is used to capture and analyze the incoming packet status as well as network information such as network throughput, packet loss rate, latency and delay variation etc. In addition to using the above mentioned software, we developed specific data processing software for data collection and analysis.

D. Experimental Methodology

Although intensive researches have been carried out on Internet traffic measurements and IP network modeling [20 ~ 22], little work has been done in the context of IP data transmission over ATSC DTV datacasting systems. Some measurement methodologies for Internet traffic analysis and characterizations such as those proposed in [12] [22] were adapted to our research work.

Our analysis is based on the collection of TCP and UDP packet traces. These traces are captured as log files and saved in the server or client for later analysis. For two-way analysis such as TCP related tests, both server logs and client logs are needed. For one-way analysis such as UDP related forwarding performance tests, client logs usually are sufficient. Data traffic flows from server to client through the downlink channel. Control traffic such as requests, acknowledgement signals are from client to server via uplink or return channel. Figure 3 illustrates the traffic flow abstraction chart.

To simulate the real ATSC DTV transmission environment, the RF Modulator is tuned to 713MHz which equivalents to TV broadcasting Channel 54. The Attenuator in the RF Simulator is calibrated to have a power attenuation of approximately –53dBm, which is representative of the power level of a real TV channel.

All experiments are focused on characterizing and evaluating the system performance. This characterization can be performed by analyzing the packet latency, latency variation, packet loss rate, throughput etc. by changing channel and transmission conditions such as the addition of RF channel noise, the variation of packet size and packet inter-departure time etc. In general, one can execute the experiments in any order or skip the ones that do not apply to specific needs.

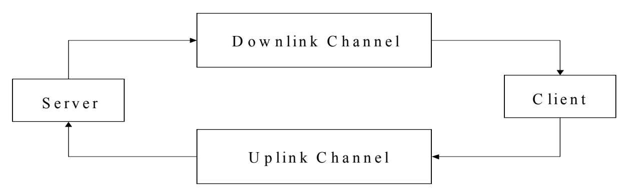

Fig. 3. Traffic flow abstraction.

IV. TEST RESULTS AND DISCUSSION

The results and discussions of performance assessment tests of TCP and UDP traffic over our interactive ATSC datacasting system are presented in this section. We start with the analysis of TCP and UDP throughputs, follow by measuring the TCP round-trip-time (RTT), then by characterizing the UDP packet loss behavior, and finally the examination of UDP packet delay and delay variance.

A. Throughput Analysis

The throughput at sender side is calculated as follows:

$$throughput = packet\ size \times sending\ rate$$

Because of the packet loss, retransmission, channel bandwidth limitation and eventually transmitter buffer overflow, the throughput at receiver side will not be the same as formulated at the sending side.

Figure 4 shows the receiver side TCP throughput versus packet size variation. The sending TCP packet sizes are respectively 32, 64, 128, 256, 384, 512, 640, 768, 896, 1024, 1152, 1280, 1408, 1536, 1664, 1792 bytes. 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, throughput measured at the receiver side increases linearly for small packet sizes. When the sending packet size reaches a certain level, the throughput does not increase linearly anymore. For the system under test, the throughput limit for TCP data transmission saturates at about 7.25 Mbps.

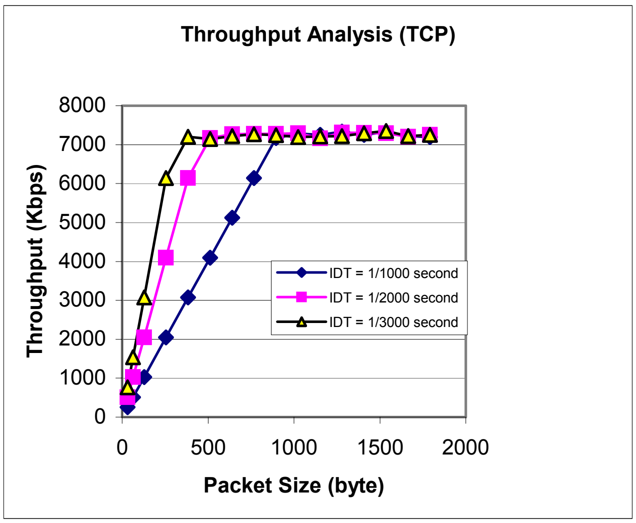

Fig. 4. Receiving throughput with TCP traffic.

We examined the throughput in the case of UDP traffic. These tests were carried out under conditions similar to those for TCP. The results obtained are illustrated in figure 5. Similar conclusion can be drawn from the results: a linear relationship exists between the throughput and packet size for small packets. But more throughput fluctuations can be seen with larger packet sizes. Nevertheless it shows increased throughput limit in comparison with the case of TCP: an average of 14Mbps when the packet size is less than 1500 bytes.

Digging into more details, as far as throughput analysis in the case of IP data transmission over the DTV datacasting system under test, three major factors affect system's performance: (1) the IPE bandwidth allocation, (2) carrier channel bandwidth controlled by the Transport Stream Multiplexer (TSM), (3) return channel bandwidth. In all tests, we have allocated up to 20Mbps as of the bandwidth for IPE, thus the throughput bottleneck turns to the TSM whose bandwidth is preset to 19.39Mbps, which is the DTV channel bandwidth.

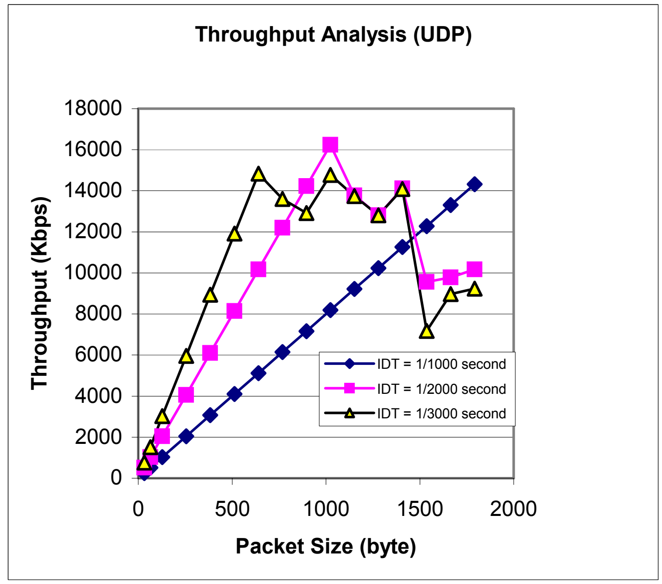

Fig. 5. Receiving throughput with UDP traffic.

For UDP data transmission, when total sending bit rate is over 19.39Mbps, TSM starts dropping packets. In this case, the receiving throughput saturates at its highest level. Considering the overhead added by the system, such as IP, UDP, TS headers, the receiving throughput at this time is limited at around 14Mbps. After saturation begins, the packet dropping rate varies with the packet size, this causes the throughput variation along its average limit of 14Mbps. The interesting phenomenon happens when packet sizes are over 1500bytes when IDT is respectively 1/2000 and 1/3000 seconds. We see a steep drop in throughput. This can be explained as the addition of IP header when transmitted packet size exceeds the MTU (Maximum Transfer Unit). Whereas for the case of IDT equals to 1/1000 second, no packet dropping starts yet when the packet size exceeds 1500 bytes.

In the case of TCP data transmission, when packet loss begins, the receiver will start issuing retransmission requests to the server through the uplink. Since we use low speed PPP in the return channel, the retransmission requests speed is caped by this channel's bandwidth. At this time, the TCP throughput is predominantly restricted by the return channel bandwidth. This explains the results drawn in Figure 4.

B. TCP RTT Analysis

The TCP round-trip-time (RTT) is calculated from the sender log as:

$$RTT = TimeR - TimeS$$

where *TimeR* represents the time when the replying ACK signal from the receiver is received, whereas *TimeS* records the sending time of a TCP packet from the sender.

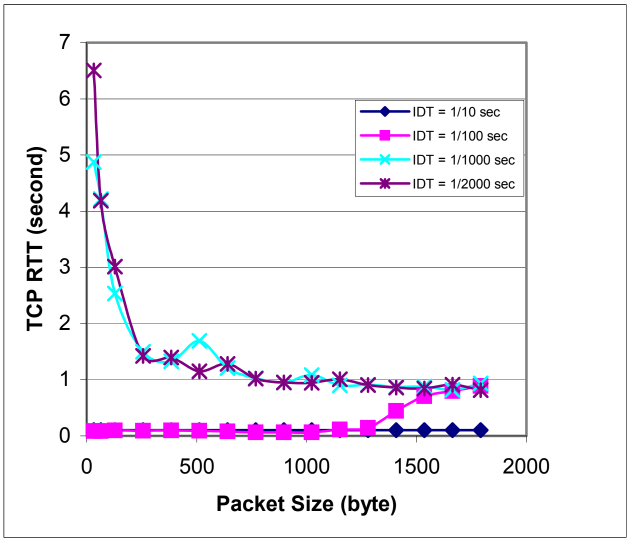

Fig. 6. Round-trip-time (RTT) delay of TCP packets.

Figure 6 depicts the variation of the TCP RTT in relation to the packet size. Different transmission inter-departure times (IDT) are taken into account.

From this figure, we see very stable RTTs when the IDT equals to 0.1 second. These RTTs are all close to 100 milliseconds for packet sizes ranging from 32 to 1792 bytes. Things change slightly when IDT decreases to 10 milliseconds, we can notice the increase of RTT when the packet size exceeds 1280 bytes. In this case the RTT keeps increasing to 800 milliseconds when the packet size grows to 1792 bytes. Dramatic results can be observed when the transmitting IDTs are 1 and 0.5 milliseconds. In these two cases, big RTT delays take place when the packet sizes are small, such as 32, 64, 128 bytes. For larger packet size, sharp decline in the RTT delay can be observed. Starting with packet size of 512 bytes, the RTT delay decreases gradually toward 800 milliseconds with increasing packet sizes.

As known, TCP uses transmission buffer to collect packets before handing them to the network. For small packet sizes, a larger number of packets are in the buffer. This results in more demanding processing to both sender and receiver, such as hand-shaking, overhead processing etc. Furthermore, in the context of channel error protection, small packets tend to be more susceptible to errors, thus more packet retransmissions are required. This can explain the increase in RTT delays when small packets are transmitted at higher sending rates.

C. UDP Packet Loss Analysis

The packet loss rate is expressed the same way as defined in RFC2544 [12]:

packet loss rate = ((input_count – output_count) × 100) / input_count

Different from networking tests, in our experiments, both the effects of packet size and channel noise need to be taken into account. Channel noise is generated and calibrated with the RF Channel Simulation block (Figure 2).

We first investigated the relation between packet loss rate and channel noise. Figure 7 shows this relation for two payload sizes: 512 bytes and 1024 bytes. It can be seen that noise plays an important role in packet loss during transmission. From packet lost rate 0 to 100%, there is only about 1db difference in noise level. This proves the theoretical analysis of packet loss characteristics of a typical DTV system under noisy environment which does not follow a graceful degradation pattern. These results are similar to what is obtained from Threshold of Visibility (ToV) tests [23].

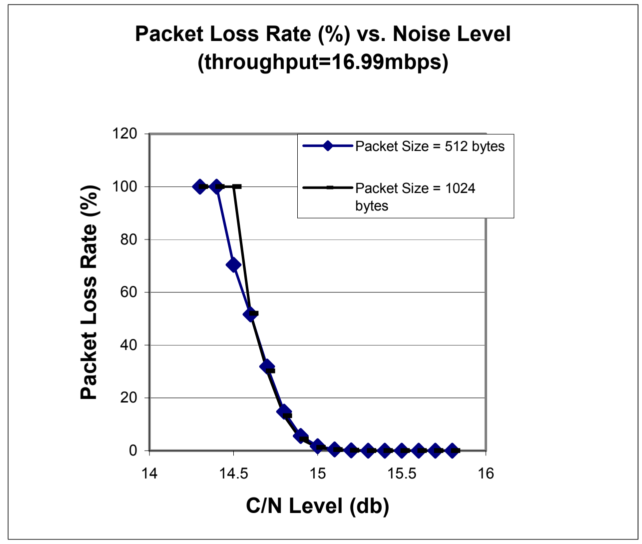

Fig. 7. Packet loss versus channel noise level for transmitting packet size of 512 and 1024 bytes.

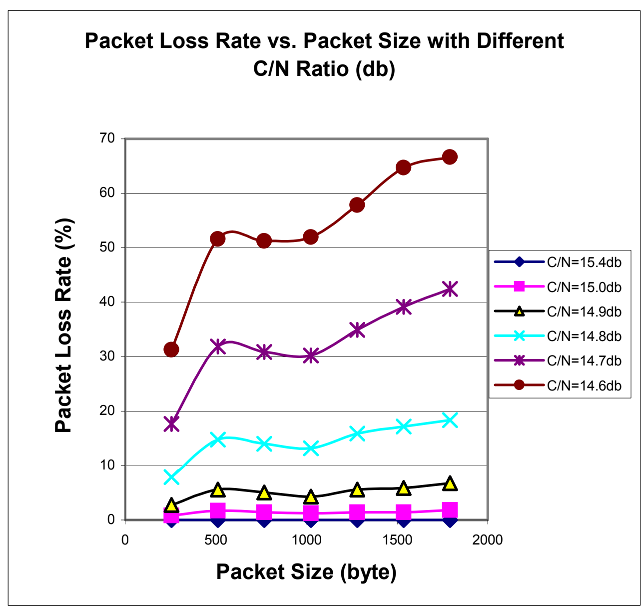

Fig. 8. Packet loss rate versus transmitting packet size.

We also compared packet loss rate vs. transmission packet size under different noise levels. The results are illustrated in figure 8. From this figure, we see that the packet loss rate and packet size do not bear a linear relation. A lower packet loss ratio is obtained for a payload of 1024 bytes for all the values of C/N. It can be seen that for the two lowest C/Ns the packet loss increases sharply for small packet sizes but reaches a plateau for packets sizes between 512 and 1024 byte with the lowest loss at 1024 bytes. For packet sizes above 1024 bytes the loss increases again but at a lesser rate than for small packets. This is an important finding that can help evaluate the optimum packet size to be used for data transmission over noisy ATSC DTV channels.

D. UDP Packet Delay and Delay Variance Analysis

The UDP packet delay or packet latency is defined as:

latency = receiving timestamp – sending timestamp

To investigate packet delay behavior affected by channel noise, we use the noise generator to create random noise ranging from 15.6 to 14.1 db, with a step size of 0.1 db. This range covers the packet loss rates from 0% to 100%.

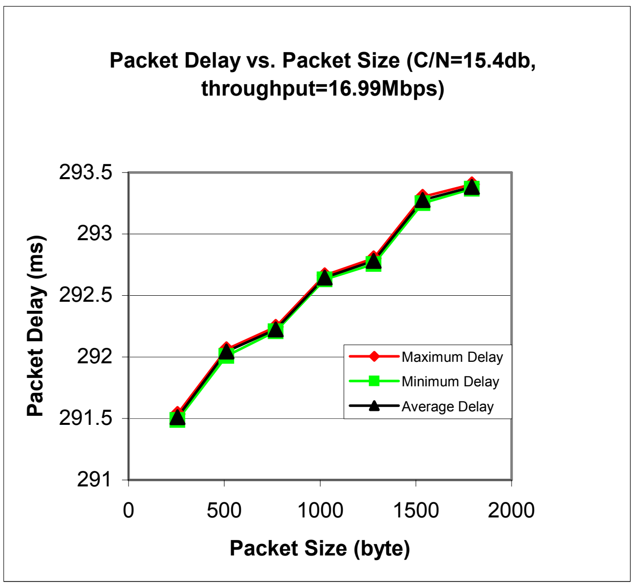

Fig. 9. Packet delay versus packet size variance.

Figure 9 depicts the average packet delay with the variation of packet size at a carrier/noise ratio of 15.4db and throughput of 16.99Mbps (based on the IPE reading). We also illustrate the maximum delay and minimum delay with the same packet size changes. Although this is a predictable behavior which could be deducted from packet size and bitrate, a direct measure is worthwhile to prove the calculation. It can be seen, as predicted, that the transmission delay tents to increase when the packet size increases. This also shows that the system does not introduce important non-linearities in the packet delay for different packet sizes.

To investigate packet delay variation (Jitter), two measurements are carried out: jitter vs. packet size variation and absolute average jitter vs. channel noise level. Jitter tests evaluate the jitter response of the system under test (SUT) to packet size changes at certain noise level. It also assesses jitter level of the SUT with the variation of channel noise at different transmitting packet sizes.

By definition, jitter can be expressed as:

jitter (N) = packet delay (N) – packet (N-1)

where N, N-1 represent indexes of receiving packets.

We show in the following (Figure 10) an example of jitter range with respect to packet size at carrier/noise ratio of 15.4db. Each vertical bar represents the minimum to maximum jitter range of a certain transmitted packet size. As can be seen from this figure, jitter variation does not seem to be influenced much by packet size changes.

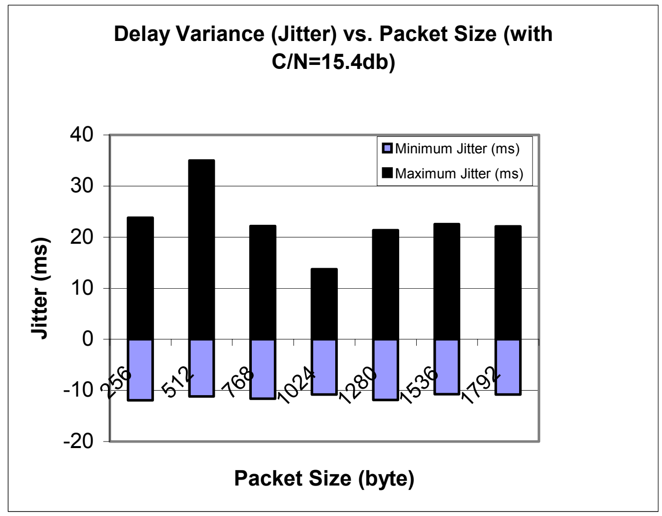

Fig 10. Jitter variation versus packet size at C/N level of 15.4db.

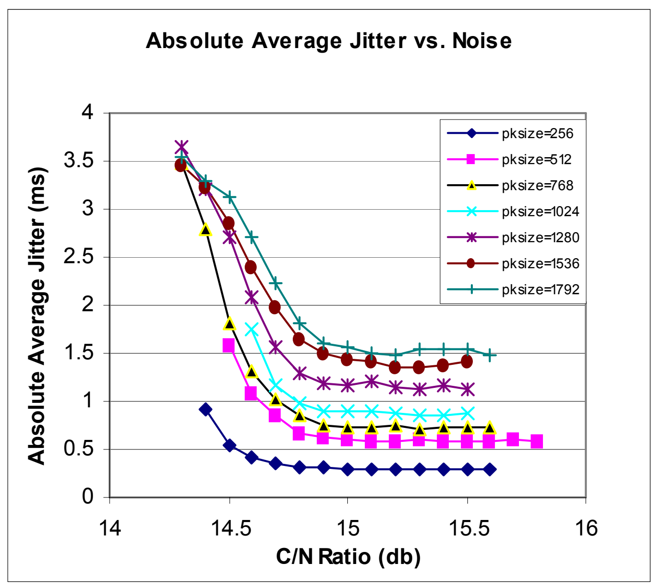

Fig. 11. Absolute average jitter vs. C/N rate variance.

The relation between the absolute average jitter and channel noise is illustrated in Figure 11. The absolute average jitter is defined as: $\Sigma|Jitter(N)|/M$, where M is the total number of jitters taken into account. From this figure, it can be seen that jitter level first decreases with the increase of C/N rate, and tends towards stable values when C/N continues to increase. This implicates that the channel noise has negative impacts towards IP data transmitting stability. We also observed that data traffic using smaller packet payload has in general lower jitter level. There is an increase of 1 millisecond in jitter level when the transmitted packet size increases from 256 to 1280 bytes. This means that smaller packets can be less affected by the jitter effect.

V. Conclusions and Future Work

A testbed for evaluating and characterizing the performance of datacasting systems is designed and developed. This testbed consists of a downlink subsystem and an uplink subsystem. The RF channel simulation block is added in the downlink subsystem to simulate channel noise perturbations to data transmitting from server to client. Supporting software is developed both for server and client in order for IP traffic generation, data collection, status analysis etc. TCP and UDP data transmission subject to RF channel noise and traffic load are investigated in various test scenarios. We noticed the three major factors restricting the throughput of data transmission in the ATSC DTV datacasting systems, i.e. the IPE bandwidth allocation, the TSM throughput setting and the return channel speed. We figured out the throughput limits of TCP and UDP data transmission over DTV channels under normal conditions: TCP throughput limits to 7.25 Mbps with the PPP return channel connection, and UDP throughput limits to be less than 16 Mbps with an average close to 14 Mbps. We discovered the interesting behavior of the TCP RTT with the variations in packet size and sending rate. We learnt that the UDP packet loss rate bears nonlinear relation with the transmitting packet size, which could lead us to in-depth research towards the usage of optimal packet size for data transmission over noisy ATSC DTV channels.

In conclusion, this work provides a better understanding of the behaviors of IP traffic over ATSC DTV datacasting systems, it gives valuable information for broadcasters who intend to implement IP data broadcasting services over ATSC DTV systems. It is also very helpful for planning the network development and expansion of DTV programs.

Next step work includes investigations towards the return channel selection and integration, such as the adoption of the DVB-RCT as the back channel, and its interference /restriction to the downlink subsystem. The development of behavioral traffic models that are consistent with empirical data obtained from our performance tests is another research topic in the future work.

Acknowledgment

The authors would like to thank Mr. Sebastien Lafleche and Mr. Benoit Ledoux of the Television Network and Transmission group (RTNT) in the Communications Research Centre Canada (CRC) for their help in setting up the testing equipment.

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