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BROADBAND CABLE ACCESS NETWORKS FOR TRIPLE PLAY SERVICES: SOURCE-DESTENATION

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ABSTRACT: This paper evaluates the performance of sending Triple play Services over hybrid networks. The network performance factors will be considered by observing the network`s availability, packet loss, delay and throughput. These evaluations will be tested over different network scenario, where the last mile media services suggested to be delivered over telephone twisted pair by means of using ADSL. The attention will be concerned for services over cable, where a proposal of using bandwidth efficient turbo trellis coded modulation (TTCM) in ADSL DMT systems instead of the multidimensional 16-state Trellis Coded Modulation (MTCM) that is given as an option in Asymmetric Digital Subscriber Lines (ADSL) standard based on discrete multitone (DMT) techniques. The results show that by using turbo codes, it can obtain 6 dB coding gain for a bit error rate (BER) of 10^{-6} in AWGN channels and more than 6.8 dB coding gain for a BER of 10^{-7} using a concatenated coding scheme.

Keywords: *Triple Play, ADSL2+, IPTV, VoIP, QoS, and Network Availability*

I. INTRODUCTION

Generally triple play architecture consists of the Service Provider Network, the core network, the access network where end users reside and the equipments at the subscriber's home. A Triple Play solution can distribute 50 to 150 TV channels over an IP network with voice over IP and high-speed Internet. Services of video, voice, and data can be sent from the IP head-end using an IP core network over an optical backbone network to the central office (CO) [Michael 10]. The CO relies the data to the access network (AN) in which digital subscriber line access multiplexers (DSLAMs) will be proposed to home`s services requirements. From the technoeconomic evaluation of telecommunications market studies, there are an addressing of a wide range usage of telecommunication networks due to major business cases [Borgar 06]. Asymmetrical Digital Subscriber Lines (ADSL), as an access technology over the existing nonloaded copper loop plant, are intended to provide up to *8 Mbps* downstream digital transport from central office to customers and up to *640kbps* upstream transmission [Nasser 09]. Such an asymmetric transmission has potential usage in services like advanced videotext, compressed TV quality video and distant education applications, where most of the information goes from the service providers to the customers. Forward error correcting codes (FEC) are employed in communication systems achieve coding gain to increase the system margin and the maximum achievable transmission rate [Neubauer 06]. A four dimensional version of MTCM concatenated with RS code was proposed previously for ADSL DMT systems to provide about *5 dB* coding gain without bandwidth expansion [Moreira 06]. Further improvement is very difficult to obtain because of the complexity of the Viterbi decoding (VA) for the MTCM. A new powerful coding scheme, turbo coding, has the potential of providing near Shannon limit performance with reasonable complexity in AWGN channels. Therefore, applying turbo codes in ADSL DMT systems is now a challenging practical problem.

II. TRIPLE PLAY NETWORK TOPOLOGY

The major components are the backbone and aggregation networks, which consist mainly of server head end, external Internet peers, IP core, Broadband Remote Access Server (BRAS) edge, and ethernet aggregation network, like DSLAMs, local loop, and the customer household [Michael 10], see Figure (1). The next sections will define the important components that are related to this work.

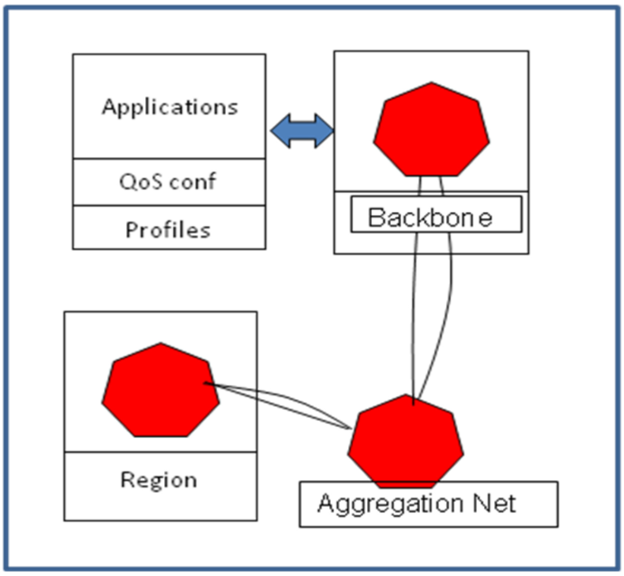


Figure 1 Network topology

2.1 Backbone Subnet:

In Figure (2) the "Backbone" subnet is designed with 4 servers configured to stream stored audio and video contents, HTTP and FTP. It contains a 100Mbps IP network and access routers for both IP Multicast traffic load (R3, RP, and R4) and IP Unicast traffic load (DSR) these routers are connected to the switches (Source and PPP) which are divided the traffic into VLANs. Then these routers are connected to the BRAS router at Aggregation subnet through a PPP_SONET_OC24 with data rate (1244.16 Mbps) wide area network (WAN) link. The approximate distance between the backbone subnet and Aggregation subnet is 5.44 km, which corresponds to approximately 1.813ms propagation delay. The multicast and unicast techniques are configured with VLAN parameter to have been efficiency triple play services. The network topology of this work will be explained in the next paragraphs.

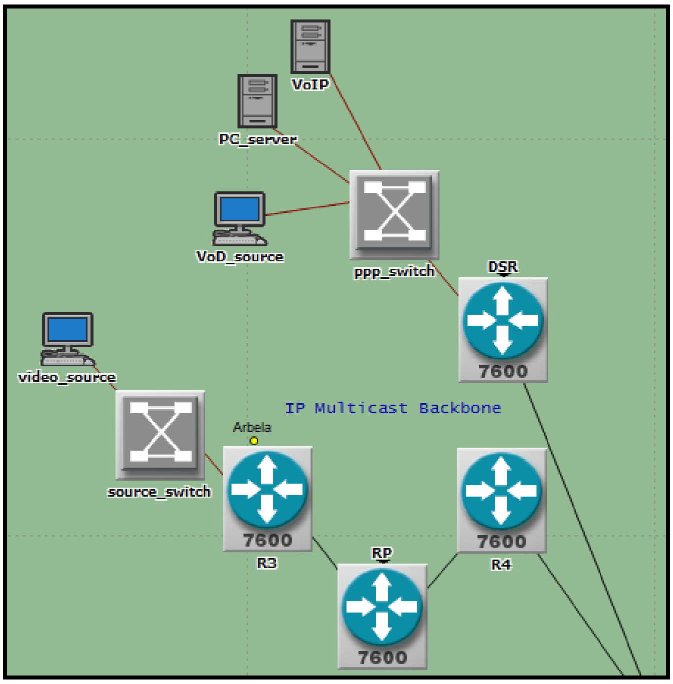


Figure 2 Backbone Subnet

Multicast technology protocols must be presented in the first IP network at the core, edge, and access layers of an IP ADSL triple-play network for supporting IPTV services. Where multicast is enabled in routers by configuring it using the attribute "IP multicast -> IP multicast parameter -> multicast routing"

2.2 Aggregation network Subnet:

The "Aggregation" subnet, that receives all traffic from backbone network by using PPP_SONET_OC24 with data rate (1244.16 Mpbs), delivery it to access network at a customer side by using Metro Ethernet Network [Gallant 06]. This subnet has several components as show in Figure (3); these components are discussed next subsections.

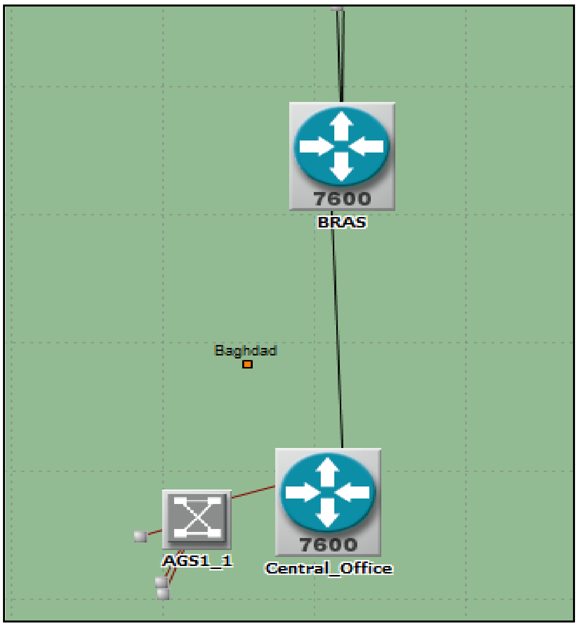


Figure 3 Aggregation_network Subnet

2.2.1 BRAS

The "BRAS" is a Broadband Remote Access Server router that forwards packets between the core and customer. It is a complex router that implements dynamic per-subscriber IP policies, Quality of Service (QoS) profiles, rate limiters, packet manipulation, address assignment, session termination and forwarding.

2.2.2 ***Center Offices (CO)***

The CO router relies the data to the access network (AN) which consists of digital subscriber line access multiplexers (DSLAMs) and broadband digital loop carriers (DLCs). There it has the so-called last mile distribution of the service (i.e. video, voice or data) which afterwards enters the subscriber’s home through the modem. Central Office router and Remote DSLAMs at the region subnet are supplied with Gigabit Ethernet links.

2.2.3 Metro Ethernet Network:

DSLAM are aggregated by Aggregation Switches also called (metro Ethernet network). A metro Ethernet network is useful when a switched layer between the DSLAM at Region subnet and the CO router in Aggregation subnet provides cost-effective aggregation capacity. Such a scenario would arise if the bandwidth utilization per-DSL-port is not enough to justify connecting DSLAMs directly to the CO. This benefit needs to be weighed against the expected traffic loads to and from DSLAMs. Also consider whether a switch can continue to offer enough statistical traffic multiplexing gains in the future. Metro Ethernet is replacing SONET links to Gigabit Ethernet links (1000Basex_adv) to flexibly and cost effectively scale the network to support increased video traffic. Another drawback to SONET in the access network is that the available feeder bandwidth has only recently upgraded to OC-3 (155 Mbps). These networks require other costly upgrades to get to OC-12 (622 Mbps), and even newly planned networks at OC-48 (2.4 Gbps) will be limited if VoD and higher speed data services achieve their expected growth. The available bandwidth divided among all of the triple-play services and video services alone so that Ethernet link 1000Basex_adv with data rate 1Gpbs in this network is used.

2.2.4 Access Network (Region Subnet)

This network shows an example for the delivery of Triple-Play services (Data, Voice and Video) over Asymmetric Digital Subscriber Line (ADSL) with data rate (downstream 12Mpbs/ upstream 1.3Mpbs). It simulates end-to-end communications between residential customers and backbone network.

III. ADSL DMT TRANSMISSION MODEL

Figure (4) shows a typical configuration of ADSL DMT system. The proposed FEC structures of the transmitter and receiver of ADSL DMT system is shown in Figure (5) and figure (6) respectively. The input bits are first RS coded. With the bit allocation table, the transmitter then allocates the corresponding number of bits to the subchannels. The turbo encoding/decoding operates across all the subchannels, followed by the constellation mapping. TTCM is thus performed by proper combination of the turbo coding and the constellation mapping.

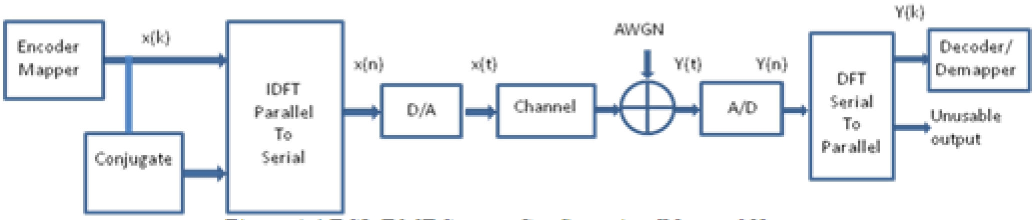


Figure 4 ADSL DMT System Configuration [Nasser 09]

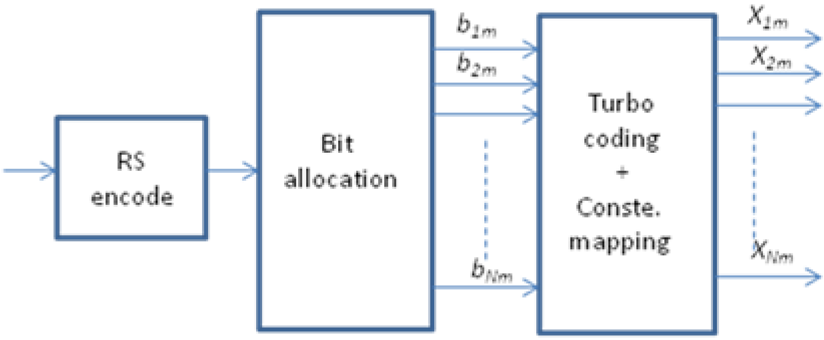


Figure 5 System encoder

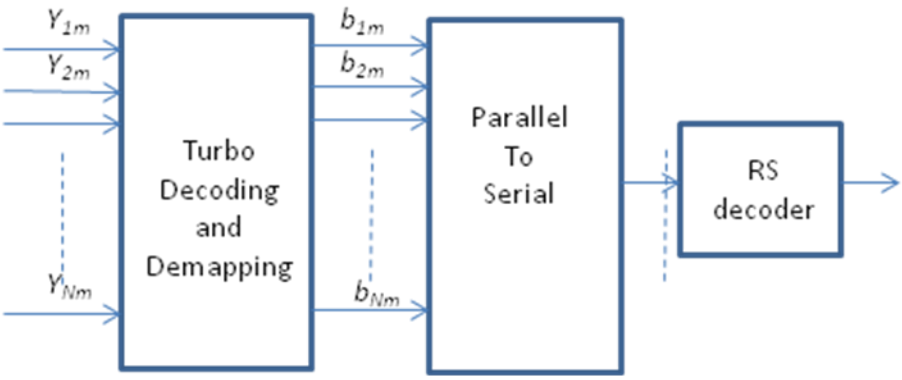


Figure 6 System decoder

3.1 Encoding Operation layer design:

In this section a method of providing forward error correction for data services uses a parallel concatenated convolutional code which is Turbo Code is used. An encoder of a parallel concatenation (PCCC) of typically two systematic is used [Moreira 06]. Which consists of recursive convolutional codes ("constituent codes") separated by an interleaver that randomizes the order of presentation of information bits to the second constituent encoder with respect to the first constituent encoder, see Figure (7). The two encoders are identical and built based on the RSC encoder of Figure (8). The performance of a Turbo Code depends on the choice of constituent codes, interleaver, information block size (which generally increase with higher data rates), and number of decoder iterations. For a particular Turbo Code, in which the constituent codes are fixed, one can ideally adjust the block size and number of decoder iterations to trade-off performance, latency, and implementation complexity requirements. As the block size changes, however, a new interleaver matched to that block size is required.

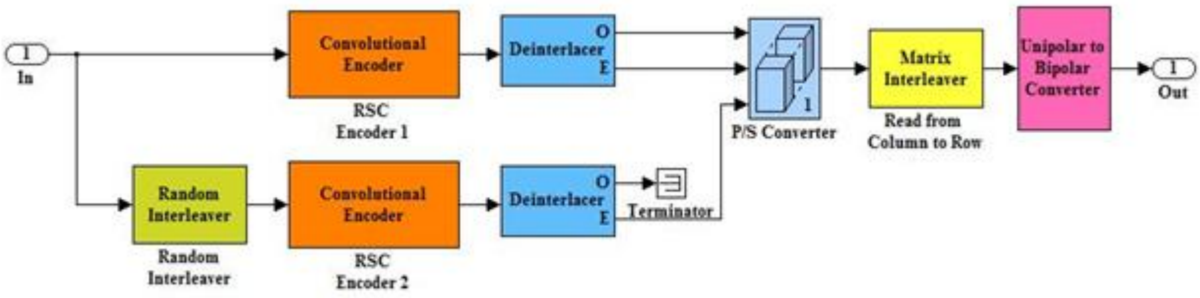


Figure 7 Turbo code encoder

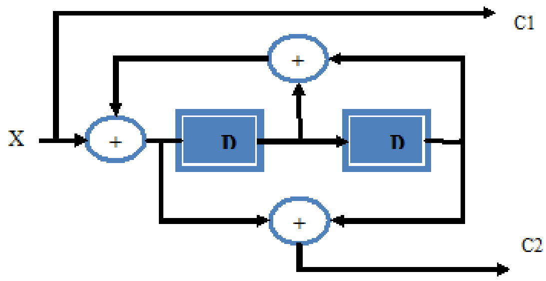


Figure 8 The RSC encoder with r=1/2 and K=3

In this work the code rate of 1/2 and 1/3 is considered to get the proper decision making of bandwidth efficiency and system performance. The information bits are always transmitted across the channel. Depending on the desired code rate, different code rates are achieved by puncturing the parity bit sequences from the two constituent encoders. As the code rate increases, the bandwidth efficiency will be improved and the performance is degraded since the decoder has less information to use in making a decision. Therefore, a tradeoff must be made between the code rate and the performance.

The deinterlacer block accepts the input vector that has an even number of elements and alternately places the elements in each of two output vectors. Therefore it is used to separate the systematic and the parity bit of each RSC encoder. As mentioned previously, the systematic bit of the second encoder is nothing more than a repetition code, thus a termination block is used on the odd output of the second encoder. The three streams: the systematic bits and the two parity bits are concatenated using vertical concatenation. A matrix interleaver is used to perform block interleaving by filling a matrix with the input symbols row by row and then sending the matrix contents to the output port column by column so as to avoid burst errors. The output then forwarded to the puncture block which will periodically remove bits from the encoded bit stream, thereby increasing the code rate.

3.2 Turbo Decoding

The truly unique aspect of Turbo codes is their iterative decoding process. The iterative decoding structure consists of two Soft-Input, Soft-Output (SISO) decoding modules that are separated by a pseudo-random interleaver/deinterleaver. The performance analysis of Turbo codes always assumes the usage of a Maximum Likelihood (ML) decoder at the receiver for efficient data recovery. The output of each encoder depends on the last input bit and the generator matrix, which enables the encoding process of the Turbo code to be represented by two joint Markov processes. It is possible to decode Turbo codes, first by independently estimating each process and then refining the estimates by iteratively sharing information between two decoders [Avril 07]. Since the two processes run on the same input data, it means that the output of one decoder can be used as *a priori* information by the other decoder.

It is necessary for each decoder to produce soft-bit decisions in order to take advantage of this iterative decoding scheme. The soft-bit decisions are usually in the form of Log Likelihood Ratios (LLRs). The LLR data serves as the *a priori* information and is defined as the likelihood of the received bit being a one rather than a zero where the decision 1 is made for a positive LLR and the decision 0 is made for the negative LLR.

A decoder that accepts input in the form of *a priori* information and produces output in the form of *posteriori* information is called a Soft Input Soft Output (SISO) decoder. The inputs to the decoder are systematic data, parity data and the *a priori* data from the previous decoder and the output of the decoder is the LLR data. The generic block diagram of a SISO decoder is shown in Figure (9).

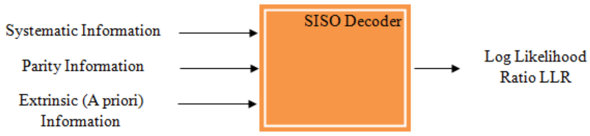


Figure 9 Block digram of SISO decoder

3.2.1 *Operation of Turbo Decoding*

A block diagram of a Turbo decoder is shown in Figure (10) which consists of two component decoders – decoder #1 to decode data sequences from encoder 1 and decoder #2 to decode sequences from encoder 2.

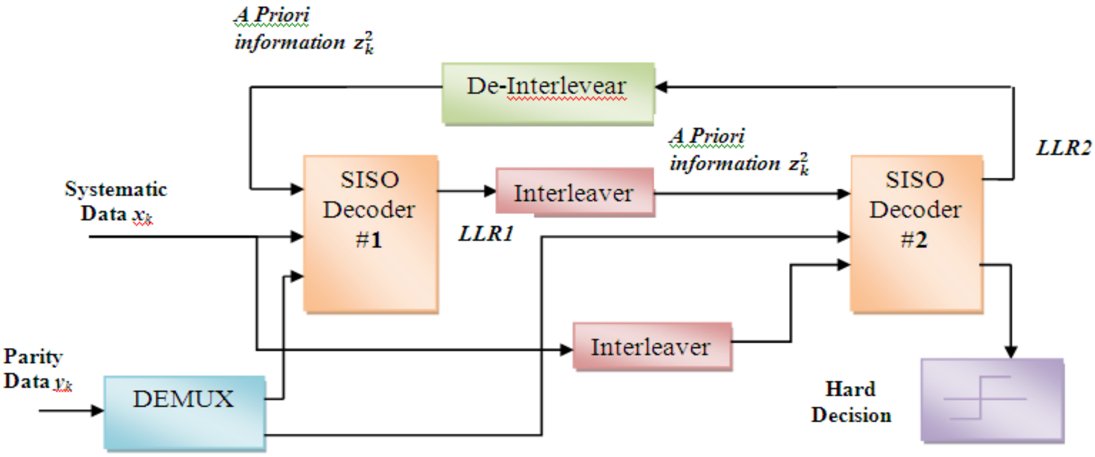


Figure 10 Turbo decoder

The first decoder operates on the systematic channel observation y_k^0 , the parity channel observation from the first RSC encoder y_k^1 and the *a priori* information z_k^1 . The *a priori* information for SISO (decoder #1) is initially set to all zeros, since the second decoder has not produced any information. This implies that each information bit is equally likely to be a 0 or a 1 initially. Both channel observations are multiplied by the channel reliability $Lc = 4aEs/No$ where the variable *a* is the fading amplitude, *Es* is the average symbol energy and *No* is the noise Power Spectral Density (PSD). The channel reliability places more emphasis on the channel observation when the SNR is high and there is no fading. Likewise, more emphasis is placed on the *a priori* information z_k when the SNR is poor or when there is a deep fade.

The output of the SISO decoders is expressed as a Log-Likelihood Ratio (LLR) Λ_k where the decoder's output at time *k* can be broken down into three distinct parts: the scaled systematic channel estimate $\frac{4aE_s}{N_0}y_k^0$, the *a* priori information z_k and the extrinsic information l_k .The k^{th} LLR is expressed as:

$$\Lambda_k = \frac{4aE_s}{N_o} y_k^o + z_k + l_k$$

The extrinsic information is the new information generated by the current stage of decoding. In Turbo decoder, the extrinsic information for the first decoder is determined by subtracting the systematic channel observation and the current stage's *a priori* information from the LLR Λ_k^1 , thereby preventing positive feedback. The extrinsic information is then permuted by pseudo-random interleaver and used as the weighted *a priori* information for the second decoder. The second decoding module operates on the weighted *a priori* information from first decoder z_k^2 , the permuted channel observation $\tilde{y}_k^0$ and the parity channel observation from the second RSC encoder y_k^1 ,that generate a new LLR Λ_k^2 .

The first decoder is presented with the result from the second decoder one can imagine that it might improve its performance, compared to its first decoding attempt. The two decoders iteratively exchange this extrinsic information and improve their estimates about the decoded bits. If all the decoding iterations have been completed, the final output Λ_k^2 is deinterleaved and hard-limited to produce the final decision. The iterative decoding process improves the BER performance of Turbo codes in a superior way [Neubauer 07]. As stated before, when Berrou and Glavieux achieved BER =10^{-5} at *Es/No* within 0.7 dB of the Shannon limit using a rate 1/*2* turbo code, they used 18 decoding iterations.

3.2.2 Parallel Concatenated Code

The receiver error handling mainly consists of two parts, which are the Receiver Front End and the Turbo Decoder block as shown in Figure(11).

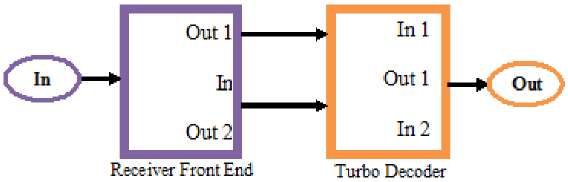


Figure 11 Turbo Decoder Main Blocks

As shown in Figure (12), the data is divided by the noise variance through the gain block, then sampled and held for a specified sample period by the zero-order hold block. After that, a matrix deinterleaver block is used which fills the input symbols into a matrix column by column and then sending the matrix contents to the output port row by row.

Two interlacer blocks are used to reconstruct the data as produced by the two de-interlacer in the transmitter side, then the output of the both interlacer is forwarded to the Turbo decoder block. Figure (13) shows the PCCC decoding process that consists of two APP decoder blocks, a random interleaver and a feedback loop. As in SCCC, these blocks form a loop and operate at a rate of six times faster than the encoding portion.

The error rate block is the same as that used in SCCC system. As shown in Figure (14), the data are sampled and held for the specified sample period by a zero-order hold block. The error rate is calculated for all iterations by comparing the received data with the transmitted data and the output is converted to six independent channel samples. Then a mean block is used to return the mean of the input elements over the time. Finally the display block shows error rates of the six iterations where the final BER is obtained from the last iteration.

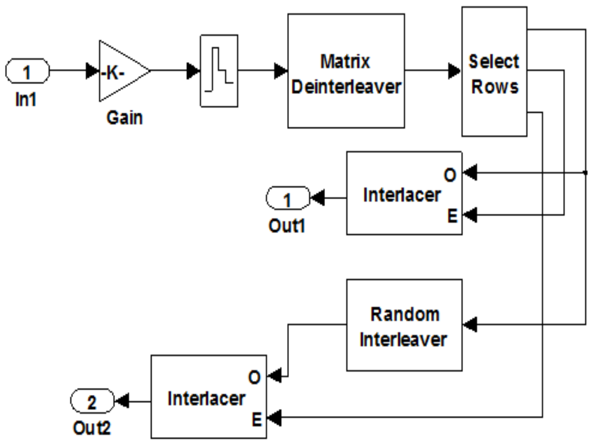


Figure 12 Receiver front end

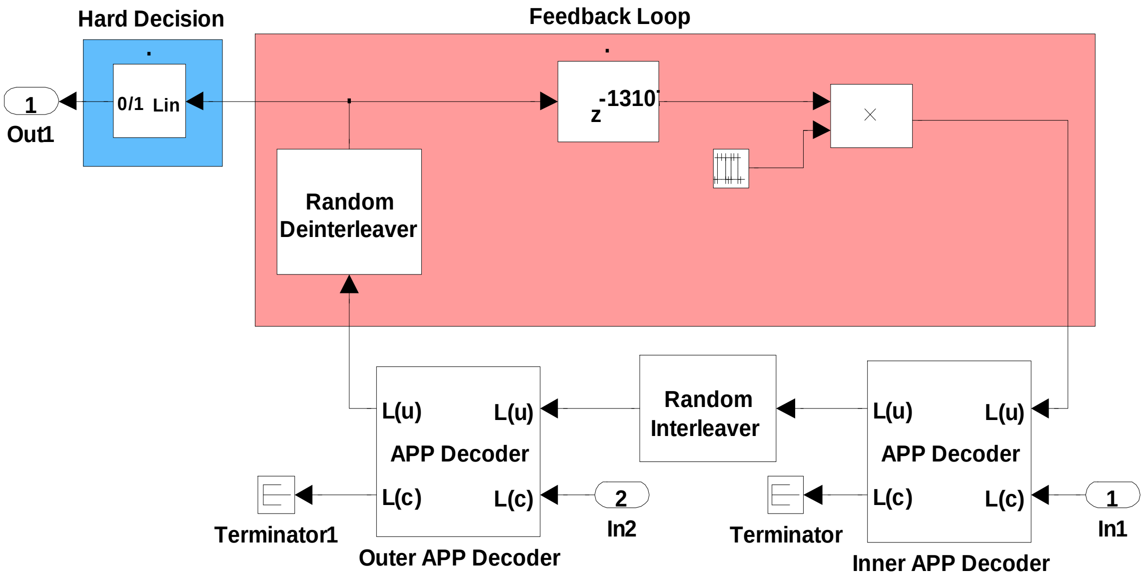


Figure 13 Turbo code decoder

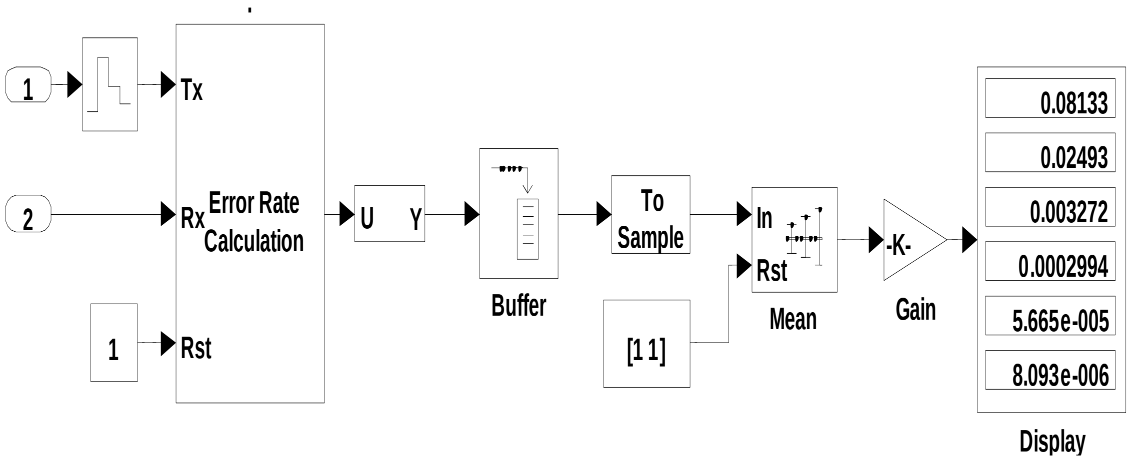


Figure 14 Multiple iterations error rate calculation

IV. SYSTEM TEST AND RESULTS

The system test will be expressed as ADSL DMT part to show the last mile network quality improvement, while the other part of the system test will give the overall system QoS that includes the evaluation of system QoS factors.

4.1 Simulation Results of using TTCM in QAM Transmission in ADSL DMT Systems

These simulations of modem apply DSL based on DMT system and turbo code. In this system four different services will be supported for a user by using modem with different code rate. The results of data bits performance are shown in Figure (15) and Figure (16) by using code rate 1/2 and 1/3 respectively. System with rate R=1/2 can be achieved by puncturing the parity bits of the RSC encoders. The performance of each service depends on the channel characteristics in the presence of the turbo decoder on correcting the errors. A code gain of 6 dB for system using code rate 1/3 is achieved as shown in Figure (17). This figure also shows the code gain will decrease as the number of iteration is increase. The Figures (18) and (19) show the effect of turbo coding on the data bits, where the BER performance of the system will increase as the iteration increases. As noticed the increasing in coding rate achieves better performance than that of using lowest coding rate, when using code rate 1/2 data rate reaches to 8.8 Mbps (2.4 for each service), while when code rate 1/3 is used the transmitted data rate reached to 14.8 Mbps (3.7 for each service) in this system.

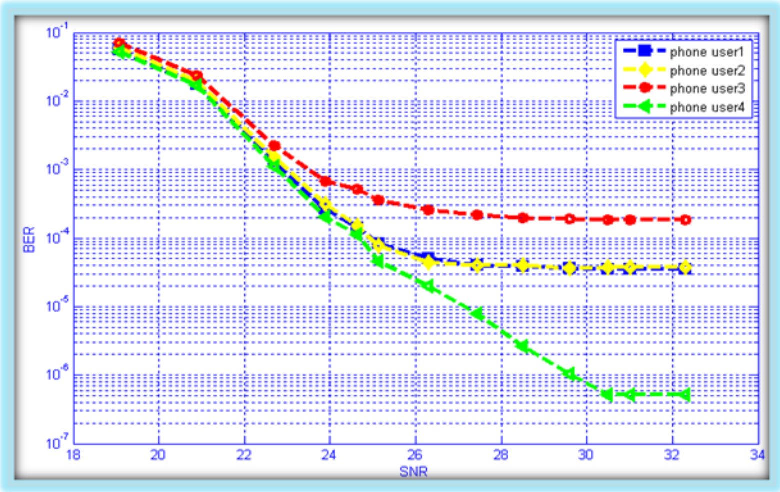


Figure (15): BER of the four service line users using turbo code rate ½

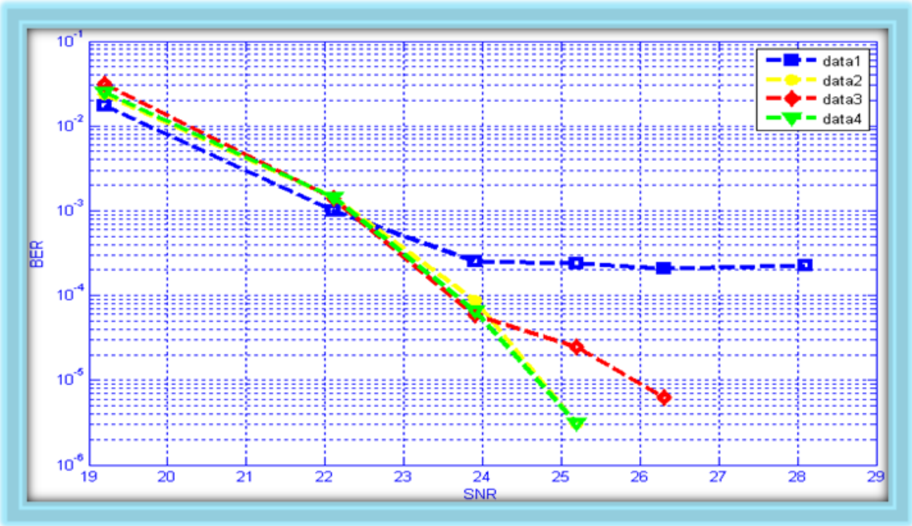


Figure (16): BER of the four service line users with code rate 1/3

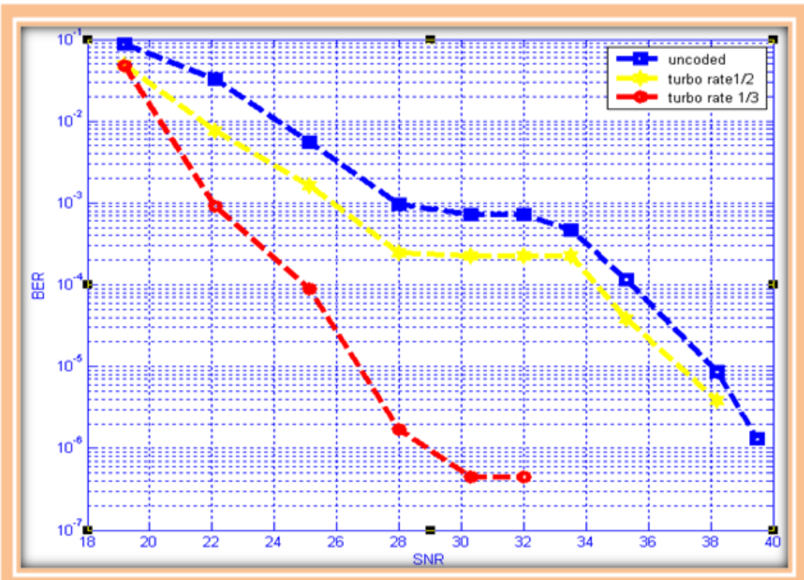


Fig (17) BER comparison of un-coded with different Turbo code rate (1/2 and 1/3)

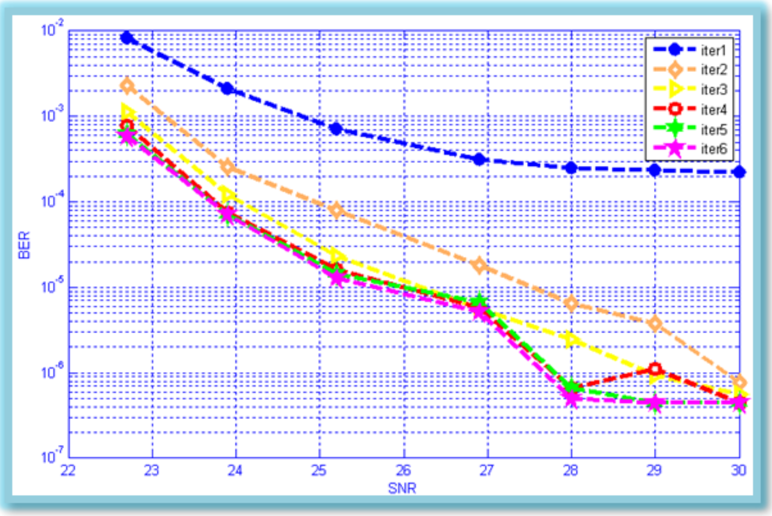
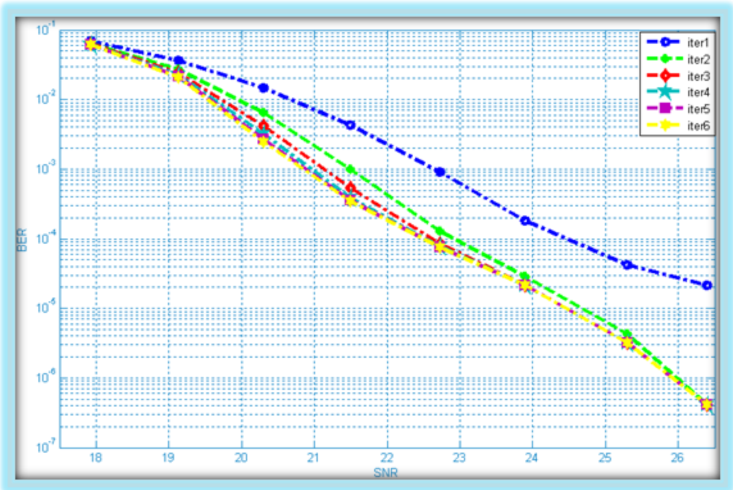


Fig (18) BER of turbo decoding process (code rate 1/2) on both data bits and phone users bits



Fig(19) BER of Turbo decoding process (code rate 1/3) on both data bits and phone users bits

4.2 Network Measurements

In this section the performance measurements factors [Kurose 10] of End to End Delay as given in Eq(1), Jitter as given in Eq(2), Packet Loss Ratio as given in Eq(3), and the Throughput as given in Eq(4) will be tested over an integrated network from the delivery source of triple play services to the destination user where the DSL is the mean media of last mile. At this test a service of VoD, IPTV, VoIP, HTTP, and FTP from source to destination will be demonstrated and measured corresponding to the above factors.

$$d_{endend} = Q(d_{proc} + d_{queue} + d_{trans} + d_{prop}) \qquad (1)$$

Q network number elements

d_{proc} processing delay at network element

d_{queue} queuing delay at network element

d_{trans} transmission time of a packet over link

d_{prop} propagation delay across network link

$$Jitter = (t4 - t3) - (t2 - t1) \qquad (2)$$

$t4 - t3$ is the expected packet reception time

$t2 - t1$ is the actual packet reception time

Negative jitter means that the packets where received in different time range i.e (t4 - t3) < (t2 - t1).

$$PLR = \frac{lost\_packets}{lost\_packets + received\_packets} \qquad (3)$$

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

$$PLR = \frac{excepted\_packets - received\_packets}{time\_unit} \tag{4}$$

$$R_{min} = \frac{(packets\_size\_in\_bytes)(number\_of\_packets\_in\_second)(8bit/bytes)}{second}$$

The QoS recommendations adopted by IPTV and voices service providers, Y.1541, will be considered in the term of system test. In additional to that the FTP downloads response time, HTTP page, Buffer Overflow Percentage response time, Free-space Pathloss, and the Erceg path loss model will be also observed. An Application Parameter of the type of services of voice Application, HTTP Application, FTP Application (File sharing /downloading), IPTV application, and VoD application are given in tables (1-5) respectively [PATRICK 07].

Table1: Residential VoIP application characteristics

Encoder Scheme	Voice Frame Per Packet	DSCP Value	Compression Delay(s)	Decompression Delay(s)
G.711	1	EF	0.02	0.02

Table2: Residential HTTP application characteristics

Http Specification	Object Size(Byte)	DSCP Value
Http 1.1	Constant(1) Media Image (0.5-2) Large Image (2-10)	BE

Table3: Residential FTP application characteristics

Download	Size File (MB)	DSCP Value
100%	5	AF13

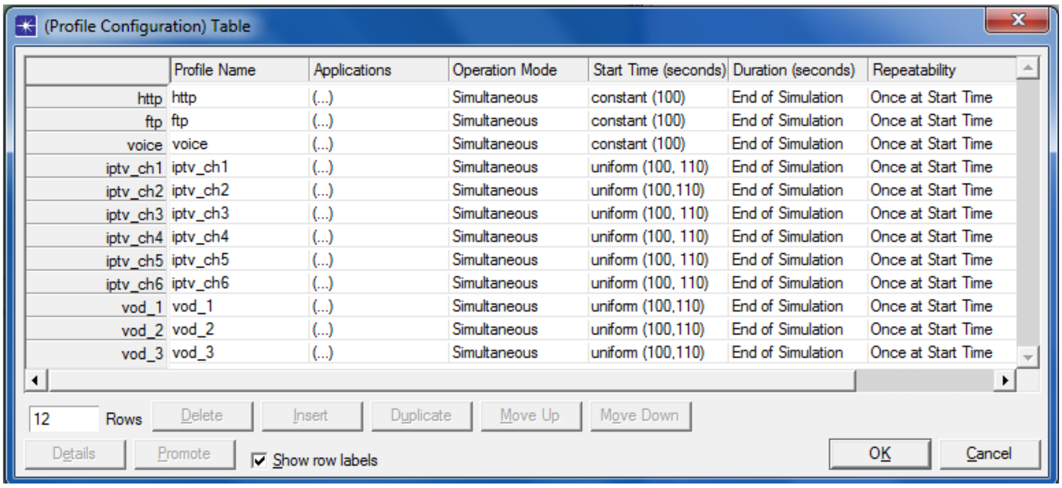
Table4: IPTV application characteristics

Frame Rate	Incoming Frame Size (Byte)	Outgoing Frame Size (Byte)	DSCP Value
15 Fps	17280	17280	AF33

Table 5: VoD application characteristics

Parameters	**T2**	**Matrix III**
Resolution	**1280x720**	**352x288**
Codec	MPEG-2	MPEG-4 Part 2
Frame Compression Ratio	58.001	47.682
Minimum Frame Size (bytes)	627	8
Maximum Frame Size (bytes)	127036	36450
Mean Frame Size (Bytes)	23833.792	3189.068
Display Pattern	IBBPBBPBBPBB	IBBPBBPBBPBB
Transmission Pattern	IPBBPBBPBBIB	IPBBPBBPBBIB
Group of Picture Size	12	12
Frame Rate (frames/sec)	**30**	**25**
Number of Frames	324,000	180,000
Peak Rate (Mbps)	30.488	7.290
Mean Rate (Mbps)	**5.720**	**0.637**
DCSP	**AF33**	**AF33**

For an IP network simulator based to OPNET Modeler 14.5, a Profile Parameters (ftp profile, http profile, voip profile, iptv_ch1 profile, iptv_ch2 profile, iptv_ch3 profile, iptv_ch4 profile, iptv_ch5 profile, iptv_ch6 profile, vod_1 profile, vod_2 profile, vod_3 profile) as given in Figure (20) will be created.


(Profile Configuration) Table

	Profile Name	Applications	Operation Mode	Start Time (seconds)	Duration (seconds)	Repeatability
http	http	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
ftp	ftp	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
voice	voice	(...)	Simultaneous	constant (100)	End of Simulation	Once at Start Time
iptv_ch1	iptv_ch1	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch2	iptv_ch2	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
iptv_ch3	iptv_ch3	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch4	iptv_ch4	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch5	iptv_ch5	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
iptv_ch6	iptv_ch6	(...)	Simultaneous	uniform (100, 110)	End of Simulation	Once at Start Time
vod_1	vod_1	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
vod_2	vod_2	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time
vod_3	vod_3	(...)	Simultaneous	uniform (100,110)	End of Simulation	Once at Start Time

12 Rows Delete Insert Duplicate Move Up Move Down
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Figure (20) The profile attribute configuration

The QoS Attribute Configuration object defines the CAR, FIFO, WFQ, Custom Queuing, and Priority Queuing is given in Figure (21).
In order to configure the QoS queue priorities in the IP DiffServ the following three schemes are also considered for the purposes of evaluation and compression:

1. Priority Queuing (PQ): which guarantees absolute priority for the voice traffic flow.
2. Weight Fair Queue (WFQ) scheme: which assigns a portion of the total bandwidth to each traffic flow, according to its weight.
3. Low Latency Weight Fair Queuing (WFQ-LLQ): is a complex scheme, where the voice traffic flow has an absolute priority over the other two traffic flows (like in Q), but the video and data traffic flows are served by a WFQ algorithm.

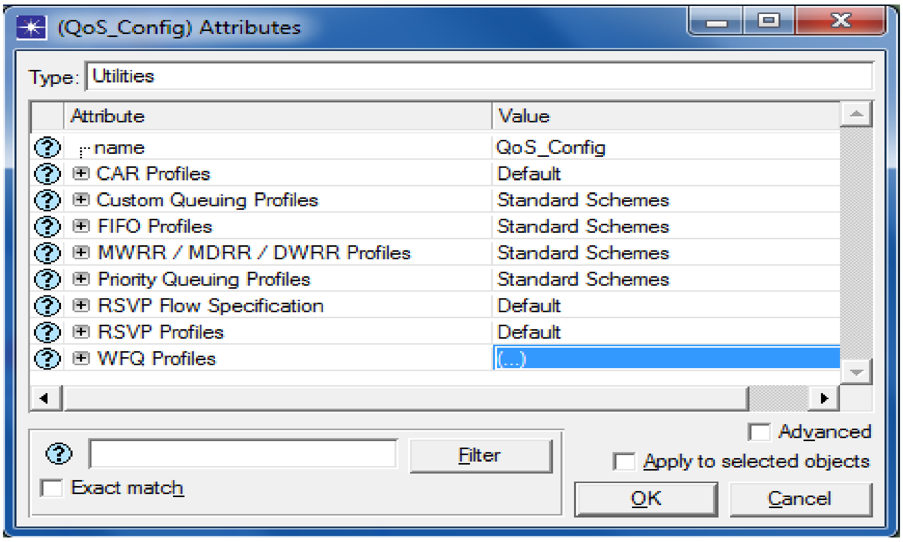

Figure (21) The QoS technologies

4.2.1 Required Network Simulation and test

The network simulation and test will be achieved corresponding to the above mentioned network applications parameters. A customer household of 30 houses all received full triple play services simultaneously, where video streams MPEG-2 and MPEG-4 at the source are available will be used to test the network QoS.

To extend the test of the network the backbone subnet of figure (2) is used to stream stored audio and video contents, HTTP and FTP. Then to forwarding Multicast traffic, IP multicast group addresses were used. A multicast group address is a single IP address taken from a reserved range (224.0.0.0/4 for IPv4, FF00: /8 for IPv6) to uniquely identify a group of hosts desiring to receive certain traffic. The video source server, see figure

(2), uses IP multicast address to setup a video conference (IPTV) session with all receivers. In multicast only a single session is setup for all receivers. The server sends only one copy of each video packet, while the router can listen to a particular group by setting the group address in the IGMP static membership information table.
The backbone, core and edge IP Multicast services must be supported by PIM-SM which is the protocol that distributes the routing information. The protocol is suitable for groups where a very low percentage of the nodes (and their routers) will subscribe to the multicast session. PIM-SM explicitly constructs a tree from each sender to the receivers in the multicast group.

At the edge where ADSL household an IP multicasting is presented that can be enabled or disabled for each IP interface. This can be configured using the attribute "IP Host Parameters->Multicast Mode".
Figures (22-24) show the performances metrics of packet loss ratio, packet jitter and end-to-end delay respectively to quantify the video streaming for VoD services which are compression by using MPEG-2 and MPEG-4.

The sender uses Unicast technology to setup a video conference session for VoD services with all receivers. In Unicast, separate sessions are setup for each receiver. Therefore, the sender sends copies of video packet, one for each receiver. While the sender uses IP Multicast technology to setup a video conference session for IPTV services with all receivers. In Multicast only a single session is setup for all receivers. Therefore, the sender sends only one copy of each video packet. Figure (25) shows Interface level example that highlighting link utilization with and without IP Multicasting.

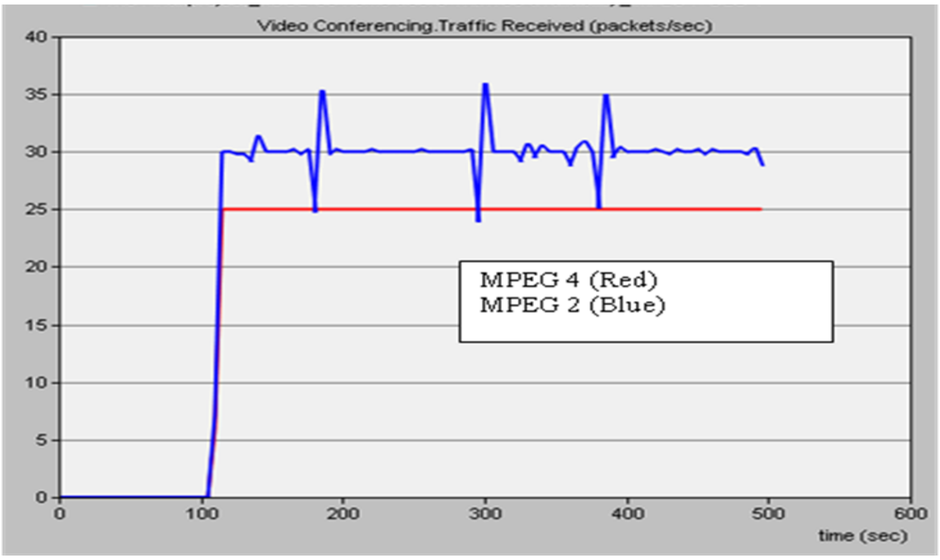

Figure 22: The MPEG-2 and MPEG-4 video PLR

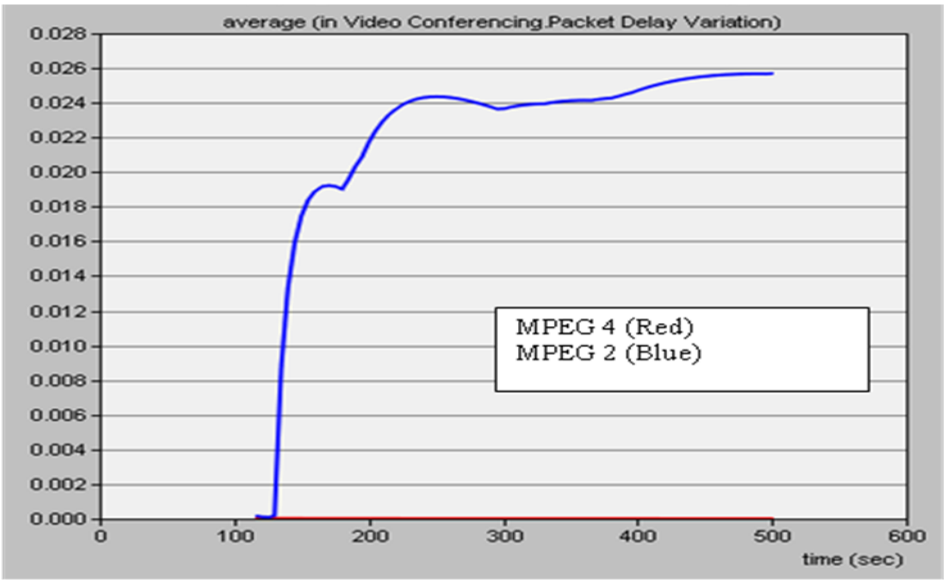

Figure 23: The MPEG-2 and MPEG-4 video packet jitter

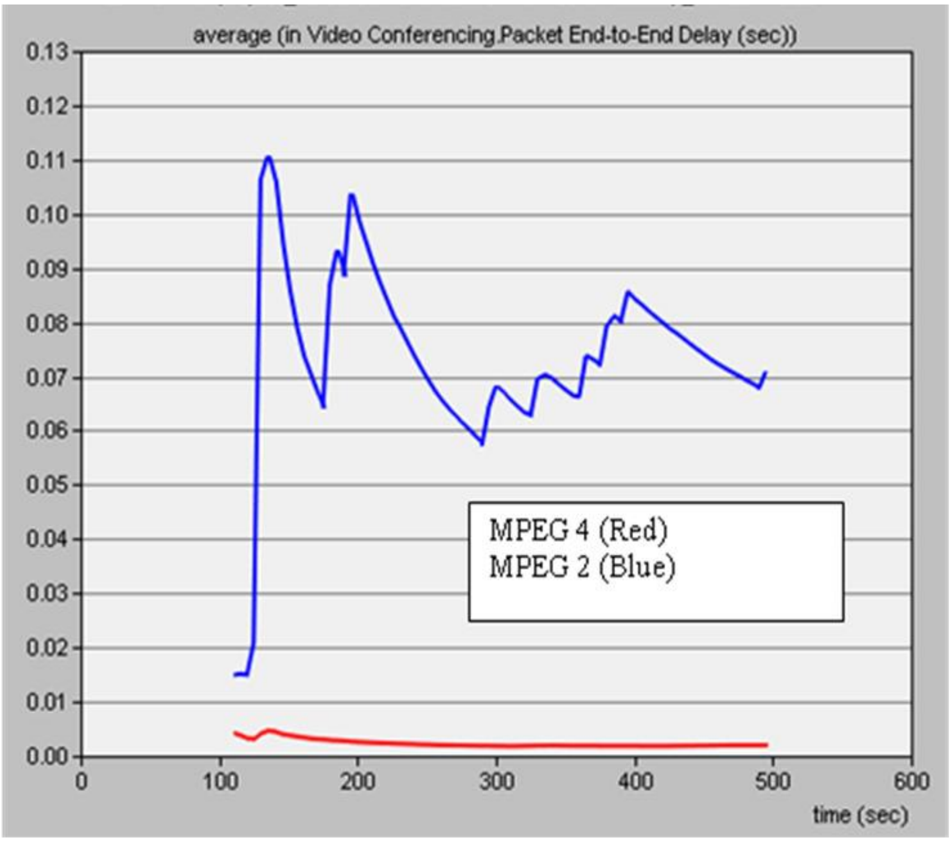

Figure 24: The MPEG-2 and MPEG-4 video end-to-end delay

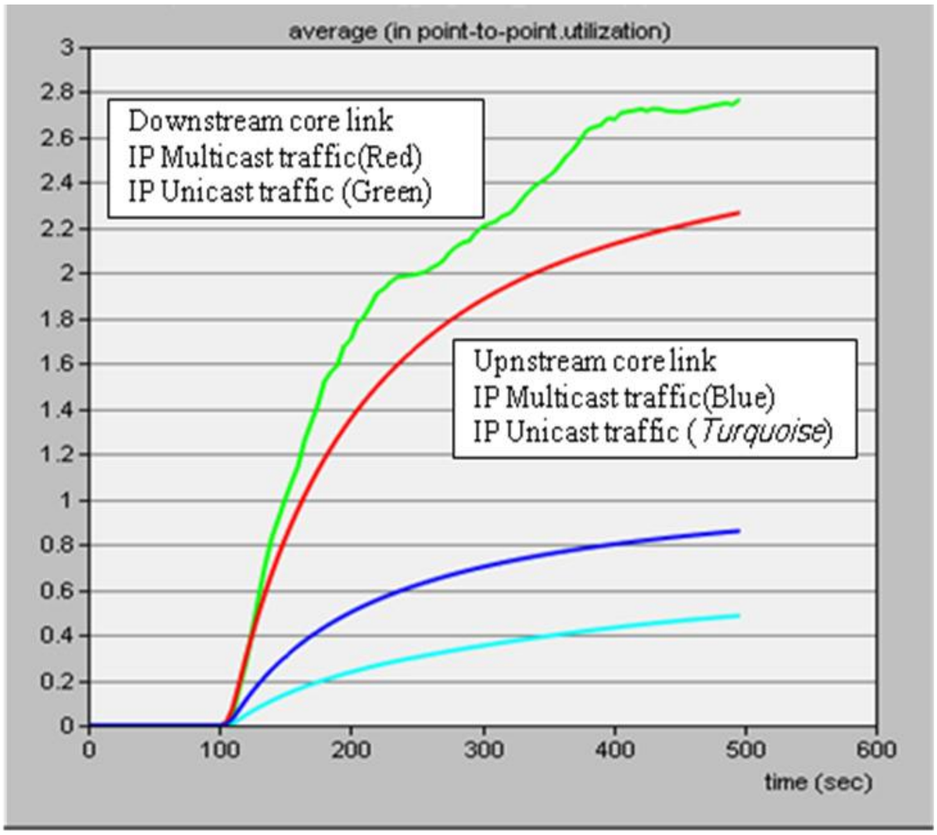

Figure 25: IP Multicast and IP Unicast Core link usage

V. DISCUSSION

This work presents results of an simulations triple play transmission over hybrid network where the last mile is an ADSL supported by turbo code as FEC. The use of such a network combination is a promising service to be provided over broadband access network. The utilization of the existing telephone lines infrastructure of the user residence decreases the cost of the service. For such applications where VoD services need high bandwidth, MPEG-4 codec is the proper solution against the problems of packet loss, jitter and reordering problems. Results show, when there is available bandwidth 100%, delay-variation for voice and video services are low. However, when the network availability decreases, delay-variation increases exponentially.

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