

Testing and Verification of Automatic Conditional Switching (ACS) between TCP Reno, and RTP

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Abstract— in this article the Automatic Conditional Switching (ACS) [7] by encapsulating TCP Reno [21] [41] [48] into RTP [7] [71] was tested, verified and analysed using FreeBSD [62] [78] compared to USAGI [74]. ACS was also tested using WIRESHARK, R, Packet Tracer, and OSI Soft PI this is as an important pre-deployment step.

Keywords— TCP Reno and RTP, VoIPv6 Performance Enhancement, Automatic Conditional Switching (ACS).

standardizations including [24] [34], G.114, RFC 1631, International Telecommunication Union (ITU), and RFC 2993 [79].

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I. Introduction

The Automatic Conditional Switching Protocol (ACSP) according to [7] is a Delay Avoidance mechanism, Two Pass network switching [28] [54] [63] [69] method between TCP Reno [17] [20] and RTP [40] [50] compared to [84] who developed new Congestion Control Algorithm [18] for TCP/UDP which is different from [70] as it is a Network Routing Based on two dimensional stability constrains. ACS deployment is recommended in the routers, and switches in the protocol layer of the IPv6 network [26] [42] [72] [85] utilizing Delay based Congestion Avoidance (DCA) [60] as a driver. The deployable [13] performance enhancement [29] [39] [86] techniques must start with network performance testing and measurements [44] [46] and must be in accordance with the Requests for Comments (RFCs) and the international standardizations such as [25], [34], [58] including RFC 1631, and FC 791[59], International Telecommunication Union (ITU), and RFC 2993 [79]. TCP Reno was selected among the other TCP flavours [45] [57] because it is a delay-based technique. Packet delay in the network is the main negative factor affecting the packets transmission from source to destination. According to this work findings, ACS is successfully reduced the cumulative delay amount for the packetized voice keeping the VoIPv6 delay under the less possible delay which is 150 milliseconds as standardized by the standard track recommendations G.114, G.711, and G.744 of the ITU.

II. The Automatic Conditional Switching Protocol (ACSP)

Automatic Conditional Switching Protocol (ACSP) according to [7] utilizes Delay Avoidance mechanism, Two Pass switching method to be applied between TCP Reno, and RTP. ACS deployment is recommended in the routers, and switches in the protocol layer of the network utilizing DCA (Delay based Congestion Avoidance) as a driver. The deployable performance enhancement techniques must follow carefully the Requests for Comments (RFCs) international

III. Selected review works

[31] confirmed the fact that in June 2003 Internet structure the attaining of heavy deployment of new TCP/IP [10] [64] enhancements such as ACS [7] requires substantial enough encouraging economic factors that must be present in order to adopt such new enhancements, yet these encouraging economic factors to deploy new enhancements are not attractive enough or may cause temporary sacrifice of the available network resources for users at least for a considerable length of time before these new deployments become practically popular in the network. ACS [7] is of high economic impact in the dramatic reduction of the cost value for long distance and international voice calls, E business transactions in addition to the real-time packetized voice and the social media are also of high economic value.

The research work carried out by [35] and ACS [7] utilizes the benefit of Delay based Congestion Avoidance (DCA), additionally [35] studied the performance of Primal Dual Schemes for Congestion Controls focused by [67] in congested networks with dynamic stream [2] such as the packetized voice network [12], [14], [15], [16].

IV. ACS description

It is important hear to clarify the process of how to do the switching between TCP Reno and RTP equipped with testing details [5] [6] [7], and configurations to validate ACS by incrementing it to the IPv6 [23] [61] protocol structure. ACS is between TCP Reno and RTP to be activated at the starting of the hard time for the travelling voice packets stream when packet arrival time is greater than the value of 150 milliseconds [24] utilizing the normality in RTP as a refined form of UDP conditionally in order to attain VoIPv6 performance [8] [32] [43] [51] enhancement breakthrough [38] in transmitting packetized voice and network stream transmission [65] [66] and managing VoIPv6 traffic [80]. RTP [1] takes data from higher-layer protocols and place it in UDP messages forwarded by passing the packets down to the

Internet Protocol (IP) layer for transmission. These steps for packet transmission using RTP which get reversed at the destination are:

1. An application sends a message to the UDP software resides at the Higher-Layer Data Transfer.
2. RTP Message Encapsulation: In this step the higher-layer message is encapsulated into the Data-field of the RTP message, after which the RTP Header messages are filled in including the *Source Port* of IPv6 to RTP and the Destination Port of the receiving end by calculating the checksum value.
3. Forwarding the UDP packets to IP for transmission, this is due to the simplicity of UDP as it does not establish connections before sending data though it's refined form RTP which requires obtaining the communication socket.

UDP does not provide acknowledgments in order to conform the receive of data and it exhibit best effort but provides less guarantees that the transmitted packets will reach to their destinations properly. In addition to that RTP does not detect lost packets and having no retransmission possibility. UDP does not ensure that data is received in the same order that they were sent, but RTP provides this criteria, besides that UDP does not provide data flow control management between network hops and nodes, also UDP doesn't handle packets congestion controls. The ACS main steps are:

- Testing the Switching in ACS Based on Domain Name System (DNS) [22] [75].
- Steps to perform ACS
- The client requests communication with the server.
- Server responds with specifying op-code to have client switch to TCP Reno.
- In case the server is not responding, then the client retransmit the voice packets over TCP Reno.
- The server reacts by opening TCP connection to client Network Address Translation (NAT).
- The client knows that given query should run prior over TCP in order to ensure that RTP will not be utilized first.
- The Domain Name System (DNS) tiny chunks converts RTP [50] into TCP Reno [4] [27] conditionally utilizing DNS and Name server searching trails.

The delay caused by adopting NAT is quantified by [82] counted as part of ACS overhead.

v. ACS testing and verification

As a platform for ACS verification and testing, a packetized voice Client/Server codes were developed using FreeBSD [5] [52] and the transactions were monitored and

capturing the inter-arrival time of the voice packets in a similar approach to [9] in studying Client/Server transactions for sensitive packetized voice traffic [47] [58] under debatable network efficiently subjective to enhancements leading the enhancement effort momentum in client-server network traffic [3] [68] which is under users competition to utilize the available network resources [53], [9] in a different approach from [7] and [68] who treated cloud computing developing a model in which software defined networking and compact encoding of important data in packet headers were in a gathering mode to make a fast, scalable, high capacity proxy server that can hide the server information while cope with the structure and design of the network infrastructure supporting network service providers.

The ACS tests conducted in this work were based on VoIP [33] captured data packets captured using FreeBSD compared to Microsoft [49] [83] based operating systems testing's. In case the intention is to run the ACS testing in real-time [30] then the possible two options would be:

- i. Software Virtual Machine in a new provided commercial software.
- ii. Developing hardware system which can be done by following the ACS implementation steps.

ACS switching was tested based on FreeBSD acquired packetized voice data (Packets Inter-arrivals) as illustrated in Figure 1.

If we would like to perform the tests in real time then the ACS steps [7] must be followed. The following equipment's are in need for ACS testing in real time:

- i. Analog to digital converter to produce the digitized voice in real time.
- ii. Traffic shaper to shape the packets in IPv6.
- iii. Perform ACS.

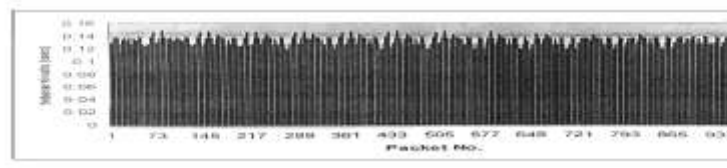


Figure 1. VoIPv6 Packets Inter arrival times Vs. Packet Numbers between 1 and 1000 (Bandwidth of 64Kbps) using FreeBSD.

VI. ACS testing and verification using wire shark, Packet Tracer, and OSI soft PI

The practical ACS testing approach is based on sending 300 voice packets stored in an audio file to be sent from source IPv6 address to a destination IPv6 addresses, the captured results are shown in Figure.2, Figure.3, and Figure.4. The Inter-arrival times VS Packet Numbers between 1 and 300 for the sent voice Packets captured using WIRESHARK, and traced using Packet Tracer software, the 300 VoIPv6 packets were sent under the following conditions:

- Packets sent through TCP Reno
- Packets sent through RTP
- Switch (by encapsulation) between TCP Reno, and RTP after 10 milliseconds waiting time.

WIRESHARK network traffic and clarity analysis [56] and applications software was utilized in the conducted tests in order to send, monitor, and capture the voice packets. The sent voice packets were traced and captured using Packet Tracer tool. The obtained test results using packet tracer as in Figure.2. Figure.3. and Figure.4. Showed evidence that ACS can reduce the packet delay [11] and latency [36] hence increase the network throughput.

Figure.2. shows Sending 300 voice packets through TCP Reno

Figure.3. shows Sending 300 voice packets through RTP

Figure.4. shows Sending 300 voice packets under conditional switching between TCP Reno, and RTP after 10 milliseconds waiting time when the total delay exceeded the 150 milliseconds hit value.

An effective way to test the switching in ACS can be achieved by designing a programmable based routings/streaming/broadcasting method to be embedded in the network, which can be considered for future work. ACS was farther tested in this research work using "Packet Tracer" a software mainly used for network simulation [77], and useful for streaming in real-time other than network simulation in OPNET and NETSIM [37]. Packet Tracer can be downloaded freeware via torrents available at:

<http://www.computernetworkingnotes.com/cisco/download-packet-tracer.html>

In this ACS testing a test call was done using WIRESHARK software between 2 endpoints (SIP [19] [73] [76] phones), then the tracing and capturing of the packets was achieved using Packet Tracer software. In the conducted tests the call was kept running for 30 seconds, and repeat the test three times. Packet Tracer captured all the packets from the endpoints (Source, and destination IP addresses) under TCP Reno, RTP, and switching (by encapsulation) between TCP and RTP after 10 milliseconds waiting time conditionally following ACS. The steps below were followed in sending the

voice packets using WIRESHARK, and capturing them using Packet Tracer showing how to get the packet details:

- The call was kept running for 30 seconds,
- Repeat the test three time,

- WIRESHARK captured all the packets from the endpoint (source IP phone) under TCP, RTP, and encapsulating TCP into RTP after ten milliseconds waiting time.

The 300 voice packets were sent from source IP address to destination IP address as follows:

Source IP address: 10.1.130.244

Destination IP address: 10.1.131.247

After running WIRESHARK program the voice packets were filtered with the following command:

```
ip.src_host==10.1.130.244 && ip.dst_host==10.1.131.247
```

Packets details were captured using Packet Tracer using the following code:

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S: Facebook LinkedIn YouTube

Source IP: 10.1.130.244

Destination IP: 10.1.131.247

Running WIRESHARK program

Filter the packets with this command:

```
ip.src_host==10.1.130.244 && ip.dst_host==10.1.131.247
```

In order to find the packet details the following was used:

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Statistical analysis was tackled using R for the results from Sending 300 voice packets under TCP Reno using WIRESHARK, captured using PACKET TRACER bandwidth of 64 KBPS.

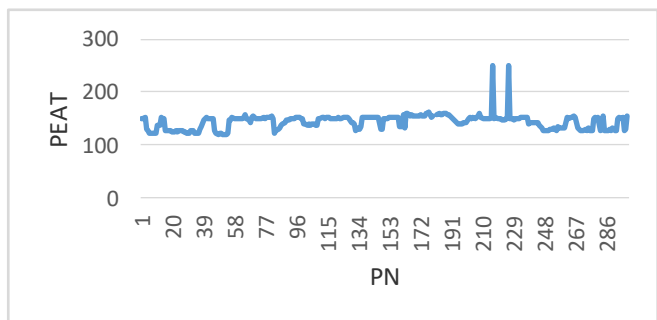


Figure 2. Sending 300 voice packets under TCP using WIRESHARK, captured using Packet Tracer bandwidth of 64 Kbps.

Mean = 143.98 milliseconds

Standard deviation = 14.5

The PEAT values are not normally distributed (Shapiro Wilk test shows P value < 0.05). Hence we calculated the median and range.

Median = 150

Range = 120 to 250 milliseconds

One-sample Wilcoxon-test was used to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

Figure.3. shows the process of Sending 300 voice packets under RTP using WIRESHARK bandwidth of 64 Kbps

Statistical analysis for the results from Sending 300 voice packets under RTP using WIRESHARK bandwidth of 64 Kbps was tackled using R.

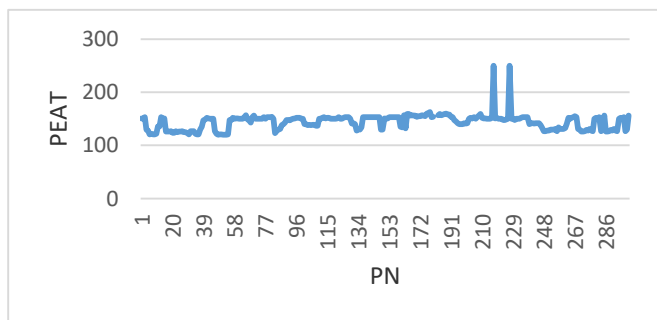


Figure 3. Sending 300 voice packets under RTP using WIRESHARK, captured using Packet Tracer bandwidth of 64 Kbps.

With 300 trials.

Mean = 130.86 milliseconds

Standard deviation = 6.12

The PEAT values are not normally distributed (Shapiro Wilk test shows P value < 0.05). Hence we calculated the median and range.

Median = 130

Range = 108 to 159 milliseconds

One-sample Wilcoxon-test was used to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

Statistical analysis for the results under ACS:

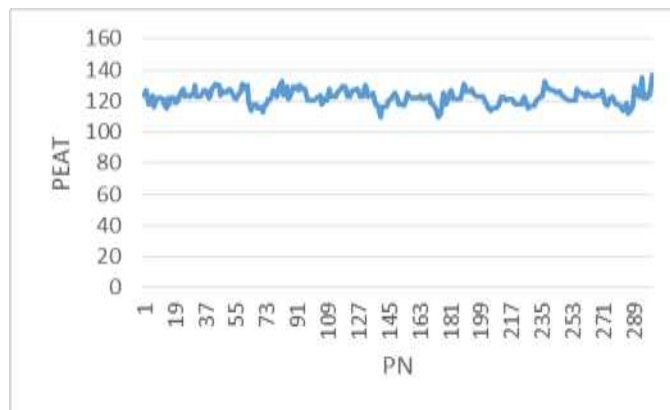


Figure. 4. Sending 300 voice packets under conditional switching (by encapsulation) between TCP, and RTP after 10 milliseconds waiting time when the total delay exceeded 150 milliseconds, bandwidth of 64 Kbps with 300 trials

Mean = 122.47 milliseconds

Standard deviation = 4.485

The PEAT values are not normally distributed (Shapiro Wilk test shows P value = 0.026). Hence we calculated the median and range.

Median = 123 milliseconds

Range = 110 to 137 milliseconds

One- sample Wilcoxon-test was used to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

VII. Testing ACS using OSI Soft PI

The following steps were followed:

- Click on dimples Topology Template
- Dimples Topology has 6 nodes
- Two of the nodes were used for the sender, and two for the receiver
- Rename the nodes as Server Reno
- All the links has the same bandwidth value as 3 megabits per seconds
- Propagation delay is 30 milliseconds
- Queue size is 100 packets
- Click the nodes and add an FTP application which runs over TCP
- Send the node to server Reno
- Topology design finished

Next step:

- Schedule events
- Transmit 50 packets using TCP Reno,
- Transmit 50 packets using RTP
- Transmit 50 packets under conditional switching between TCP Reno, and RTP

The results as shown in Fig.5. , Fig.6., and Fig.7. Showing evidence that ACS can improve

The performance by reducing the packet delay

Sending 50 voice packets under TCP Reno using OSI soft PI

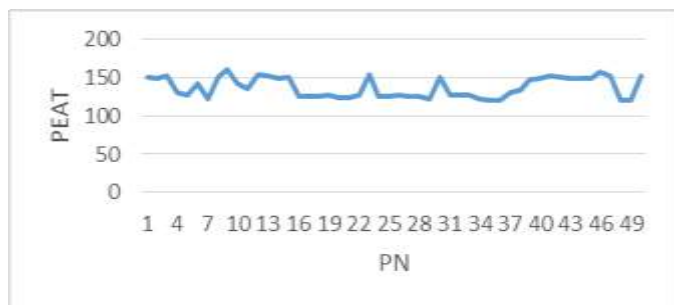


Figure. 5. Sending 50 voice packets under TCP Reno using OSI soft PI with 50 trials.

Mean = 137.48 milliseconds

Standard deviation = 13.27

The PEAT values are not normally distributed (Shapiro Wilk test shows P value < 0.0001). Hence we calculated the median and range.

Median = 132.5

Range = 120 to 161

One-sample Wilcoxon-test was done to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

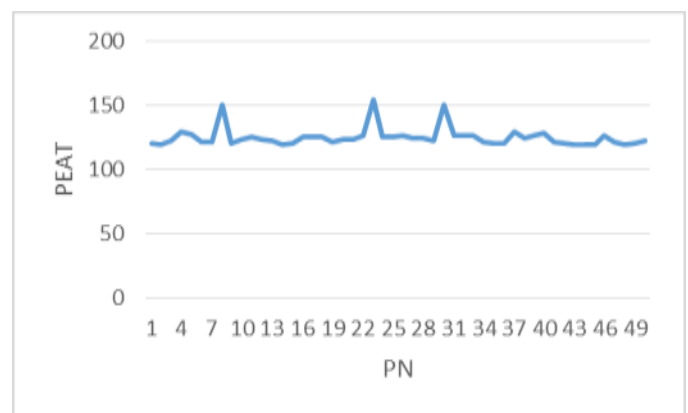


Figure. 6. Sending 50 voice packets under RTP using OSI soft PI.

Mean = 125.62 milliseconds

Standard deviation = 7.39

The PEAT values are not normally distributed (Shapiro Wilk test shows P value < 0.0001). Hence we calculated the median and range.

Median = 124

Range = 120 to 155

One-sample Wilcoxon-test was done to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

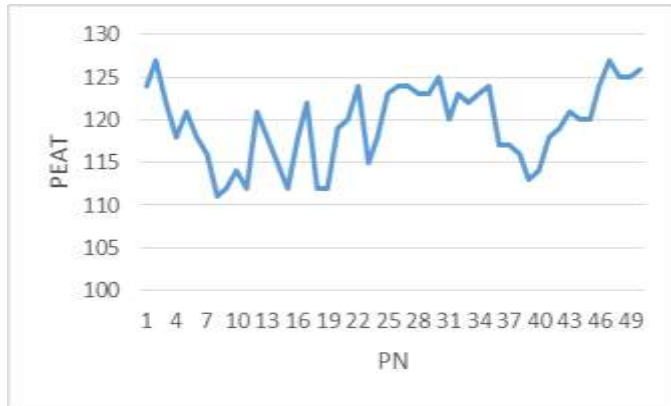


Figure. 7. Sending 50 voice packets under switching by encapsulation between TCP Reno and RTP using OSI soft PI.

Mean = 119.54 milliseconds

Standard deviation = 4.52

The PEAT values are not normally distributed (Shapiro Wilk test shows P value = 0.028). Hence we calculated the median and range.

Median = 120 milliseconds

Range = 111 to 127 milliseconds

One-sample Wilcoxon-test was done to assess if the median PEAT value is similar to 150 milliseconds which is the optimum value. The results shows that the true median is significantly different from 150. The p value for the test is less than 0.0001.

VIII. ACS and Non ACS

The comparison results for non ACS is present in Figure.1. Showing the voice packets arrival time without applying ACS, Figure.4. And Figure.7. Shows the voice packets arrival time after applying ACS using Packet Tracer. The injected data to Packet Tracer was voice packets arrival time captured using FreeBSD as in Figure.1. Showing the VoIPv6 Packets Inter arrival times VS Packet Numbers between 1 and 1000 (Bandwidth of 64Kbps). The result acquired from packet Tracer is shown in Figure.2. , Figure.3., and Figure.4. Showing the conditional switching, and total number of lost packets due to ACS on captured data of 1000 voice packets. The number of ACS switching's are 5 times, and the number of lost packets due to ACS are 5 packets in population of 1000 packets sample data values with a packet loss percentage of 0.5 percent.

IX. Conclusion and Future works on ACS Verification, and testing

ACS was tested, and validated using FreeBSD, R, WIRESHARK, Packet Tracer, and OSI Soft PI. The results were obtained supported by comprehensive literature review as a process that can lead to the ACS practical deployment.

Developing mathematical representation for the Conditional Switching in ACS is of concern hear as it can be a platform for farther practical work towards ACS practical implementation, in this context a proposal can be made to use Laplace Transform in order to represent the ACS switching process mathematically [81].

According to the obtained test results using FreeBSD, WIRESHARK, OSI SOFT PI, and Packet Tracer emphasizing on the time required to send the voce packet in each condition, ACS showed evidence that it can enhance the VoIPv6 packets transmission performance [55] as it is tacking the advantage of the difference in packet delay between TCP Reno, and RTP conditionally by utilizing the Real-Time monitoring for packet delay and take the switching decision conditionally.

As a future work, in order to perform ACS tests in real time the steps presented in the ACS descriptive diagram as in [7] must be followed carefully, and the following are required:

- 1- Analog-to-digital converter to produce the digitized voice in real time.
- 2- Traffic shaper to shape the VoIPv6 traffic in IPv6.
- 3- Perform the switching in real-time following ACS steps.

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[ACS verification and testing revealed and showed evidence that ACS can enhance the VoIPv6 packets transmission performance as it is tacking the advantage of the difference in packet delay between TCP Reno, and RTP conditionally by applying Real-Time monitoring for packet delay and take the switching decision back and Forth conditionally.]