Cairo University
Faculty of Computers and Information
Information Technology Department

Subjective QoS of Voice over IP

By

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A Thesis Submitted to the

Faculty of Computers and Information

Cairo University

in Partial Fulfillment of the

Requirements for the Degree of

Master of Science

in

INFORMATION TECHNOLOGY

Under the supervision of

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Information Technology Department
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CAIRO
March 2009

ADMISSION

I certify that this work has not been accepted in substance for any academic degree and is not being concurrently submitted in candidature for any other degree.

Any portion of this thesis for which I am indebted to other sources are mentioned and explicit reference are given.

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Cairo

March 2009

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List of Abbreviations

AQM:	Active Queue Management
AF:	Assured Forwarding
BA:	Behavier Aggregate
BE:	Best Effort
CBS:	Committed Burst Rate
CIR:	Committed Information Rate
CBR:	Constant Bit Rate
Diffserv:	Differentiated Services
DS:	Differentiated Services
DSCP:	Differentiated Services Code Point
EF:	Expedited Forwarding
ITU:	International Telecommunication Union
IETF:	Internet Engineering Task Force
IP:	Internet Protocol
LAN:	Local Area Network
NS2:	Network Simulator 2
OSI:	Open System Interconnection
PBS:	Peak Burst Rate
PIR:	Peak Information Rate
PHB:	Per-hop Behavior
PSTN:	Public Switched Telephone Network
QoS:	Quality of Service
RED:	Random Early Detection
SLA:	Service Level Agreement
SRTCM:	Single Rate Three Color Marking
SQL:	Structured Query Language
TCA:	Traffic Conditioning Agreement
TRTCM:	Three Rate Three Color Marking
TSW3CM	Time Sliding Window Three Color Marker
TSW2CM	Time Sliding Window Two Color Marker
TCL	Tool Command Language
TCP:	Transmission Control Protocol
UDP:	User Datagram Protocol
VoIP:	Voice over IP

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Abstract

The demand for Voice over IP (VoIP) applications has increased tremendously through the last two decades. This great demand is due to the great decrease in VoIP installation and operating costs. In addition to that, VoIP offers a great variety of services, such as Voice/Data Integration. These features make the VoIP more important in the business field. On the other hand, VoIP applications face some challenges. These challenges are speech quality, delay and packet loss. The speech quality may decrease as the voice stream is very sensitive to the packet loss and packet delay. The delay may occur due to different routes that are available for the voice stream. Packet loss is considered the most serious problem, as it occurs due to the congestion or queue overflow.

Due to these challenges, Quality of Service (QoS) concept appears in the VoIP applications to ensure the quality of transmitted and received voice between customers. This is done through specifying a simple, scalable and coarse-grained mechanism for classifying, managing network traffic and providing QoS guarantees on modern IP networks. This concept is applied through the Differentiated Service (Diffserv) architecture. Traffic management is done through the Diffserv traffic conditioner component. This component focuses on metering and marking the stream of bits according to some network parameters. The Single Rate Three Color Marking (SRTCM) and Two Rate Three Color Marking (TRTCM) are examples of the traffic conditioner components of the Diffserv. The traffic conditioners marks packets in the traffic based on some traffic rates. The packets may be marked as red or yellow or green depending on the network condition and traffic rates. Depending on the packet color, packets are enqueued into three virtual queues and treated in different ways. The Random Early Detection (RED) Algorithm is used to drop packets from virtual queues depending on the queues sizes and dropping probabilities.

In this thesis, we develop a comparison is done between the TRTCM and SRTCM. In order to perform this comparison a model is build using the Network Simulator (NS2). Through this model we change the dropping probability, packet size and number of VoIP nodes in order to find the most suitable traffic conditioner, dropping probability and packet size. We find the effect of the packet size and dropping probability on the number of lost bits and the fairness index. A comparsion is done between the best effort serveice and the differentiated service when applying the TRTCM or SRTCM as traffic marking and RED as an active queue management algorithm. A new measuring technique is introduced in order to fasten the simulation analysis and decrease the required time and effort for performing the analysis. This method uses the database and open source concept in performing the required analysis.

The TRTCM achieves better fairness and decreases the packet loss against the SRTCM. On the other hand, the best effort achieves better packet loss than the SRTCM and TRTCM. This happens because the queue configuration in the best effort network does not apply any kind of active queue management algorithm (AQM).

Keywords: Voice over IP, Differentiated Service, Traffic conditioner, Fairness Index, Single Rate Three Color Marking, Two Rate Three Color Marking.