

# Traffic Control and Distributed Optimization Routing Problems in ATM Networks

P. Rajan, Dr. Arul Lawrence Selvakumar

Research Scholar, Bharathiyar University, Coimbatore-641046, Tamil Nadu, India

Professor and Head, Department of Computer Science & Engineering, Rajiv Gandhi Institute of Technology, RT Nagar, Bangalore, Karnataka 560032, India

## Abstract

Aggressive research as gigabit network has led to dramatic improvements in network transmission speeds. One result of these improvements has been to put pressure on router technology to keep pace. This paper describes a router nearly completed. This is more than fast enough to keep up with the latest transmission technology. This router has a back plane speed of 50 gigabit and can forward tens of millions packet. Scheduling algorithm can be implemented on CVAR applications but in this research scheduling is implemented on CBR applications and the performance on WLAN network is enclosed by delivering different traffic load. QOS parameters [5] will be considered as the performance metrics on this study. The comparative study of various algorithms can show the best scheduling algorithm in WLAN with CBR applications. ATM was the focus of action research and significant investment in the early to mid 1990's. This paper discuss several visions for ATM prevalent at the time and analyses how ATM evolved during this period this paper also consider the amplifications of this history for current connection oriented technologies such as optical transport network and MPLS.

## I. Introduction

The satellite payload can be transparent providing only layer 1 connectivity or it can regenerate thus providing layer 2 packet connectivity. The connectivity can be static or quasi static in the case of a layer 1 transparent satellite and more dynamic in the case of a layer 2 packet switching satellites. Adaptations of satellite parameters to the different application needs. Symmetry, Delay, Sensitivity, Jitter tolerance and QOS classes. Adaptations of efficient management and resources strategies able to optimize the throughput especially as the TCP layer. Introduction of QOS management for Multimedia applications. The group key is updated regularly to reduce the probability of successful cryptanalysis of the encrypted traffic. The group key may also need to be changed on demand if it is determined that the key has been compromised. Rekeying may be required when a new user joins the multi cast group. This ensure that the user cannot decrypt and the enclosed decrypt that was sent prior to their joining. Rekeying may be required when an existing user departs from the multicast group. This ensures that the user cannot decrypt enclosed traffic i.e. sent after they leave. Rekey when a key user departs from the multi cast group. Although it is the most secure alternative it has the disadvantage that when there are a large number of group members changing the key on each departure may be a heavy processing load on the key server and is unlikely to scale. Periodically rekeying i.e. different here is the intention to handle together a number of departure user and efficiently rekey them simultaneously. This reduces the total rekey workload and increases the scalability of the multicast group especially large dynamic groups.

The multiple QOS requirements and the complex tradeoffs among them to make it difficult to define multi routing metrics. The multi constraint routing problem has been proven NP complete and which is impossible to solve in polynomial time. Further more in case of multiple routing metrics the selected metrics can be orthogonal to each other to avoid redundant information. ATM networks where initially expressed to replace the current router based internet. Although this change did not happen ATM switches are widely used as the core network and backbone technology. Hence an ATM routing algorithm must be scalable to the size of today's and the future internet.

## II. Related Works

Application Layer applications invoke TCP/IP services sending and receiving messages or streams with other host. Delivery can be intermittent and continuous. Transport layer provides host to host packetized communications between applications using either reliable delivery connection oriented TCP or un reliable delivery connectionless UDP. Exchanges packets end to end with other hosts. Network layer encapsulates packets with an IP datagram which contains routing information receiver or ignores incoming datagram's as approximate from other hosts. Choices datagram validity handles network error and control messages. Physical layer includes physical media signaling and lowest level how functions exchanges network specific data frames with other devices. Includes capability to screen multicast packets by port number at the low level. Messages composed of state event and control information as used in DIS entity state PDU. Implemented using multicast complete messages semantics is involved in a single packet encapsulation without fragmentation. Lightweight interactions [5] are received completely or not at all. Light weight network resources references multicast

receiving groups can be cached so that repeated queries are answered by group member instead of servers. Pointers do not contain a complete object as light weight interactions do instead containing only a reference to an object. Large data objects requiring reliable connections oriented transmission. Typically provided as a www query response to a network pointer request. Live audio video DIS 3D graphics images or other continuous stream traffic that require real time delivery sequencing and synchronization implemented using multicast channels.

### III. Simulation and Performance results

The total number of packet dropped due to deadline miss during the measurement interval. Missed deadline divided by missed deadline divided by the total number of packets acknowledged during the measurement interval. The total number of packets dropped due to buffer overflow during the measurement interval. The total delay of all packets acknowledged during the measurement interval divided by the number of packets acknowledged during the measurement interval. The data service rate divided by the link bandwidth . Improved network throughput and the lower packet loss achieved by using the proposed scheme indicate of better utilization than the other scheme.

**Algorithm used:**

Find the lowest priority call on this uplink beam.

Make a list of alternate beams which have the required uplink coverage for this call stored in most capacity first order.

For each beam in this list of alternate beams try to migrate this lowest priority call this to alternate beam.

Now do the same thing for the down load link of this beam if it is ever capacity.

QOS parameter table

Parameter	Priority	RR	WF	WRR
Throughput	4137	4140	4148	4157
Jitter	0.013	0.13	0.011	0.011
Delay	0.014	0.014	0.013	0.012



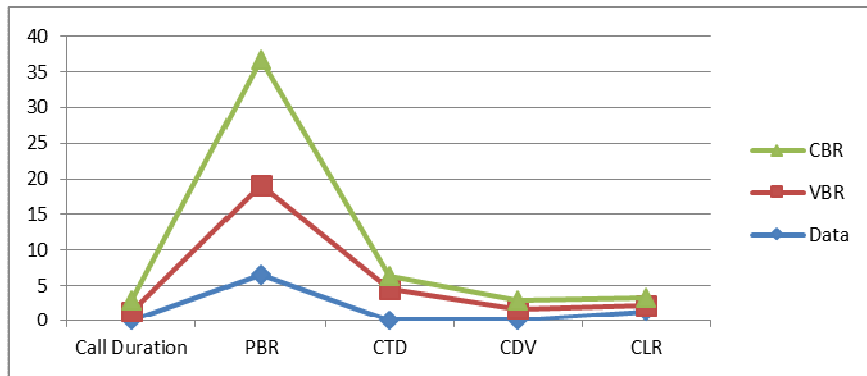
Average Jitter = Total Packet of Jitter of all Received Packets / Number of packets used – 1

Throughput = Total byte sent / Time last packet read – time of first packet received

Call blocking Ratio = Number of rejected Connection Setup Calls / Number of arriving connection setup calls

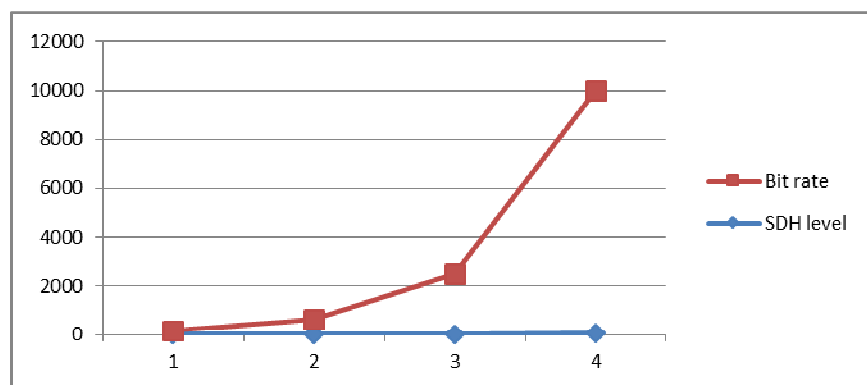
Traffic character table

Traffic	Call Duration	PBR	CTD	CDV	CLR
Data	0.15	6.5	0	0	1.25
VBR	1.25	12.7	4.5	1.75	0.95
CBR	1.5	17.53	1.85	1.27	1.2

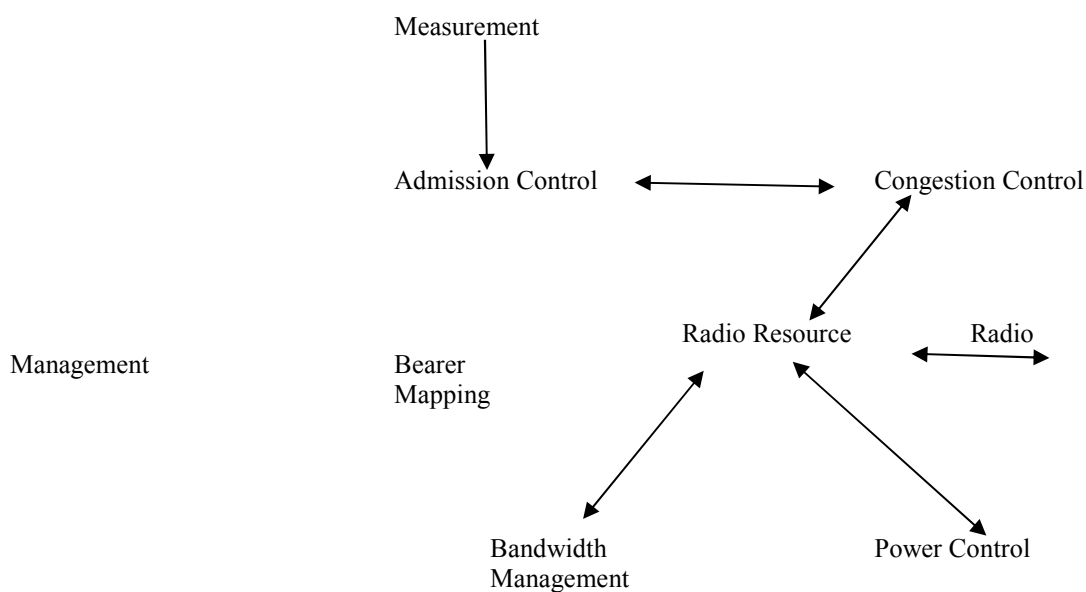


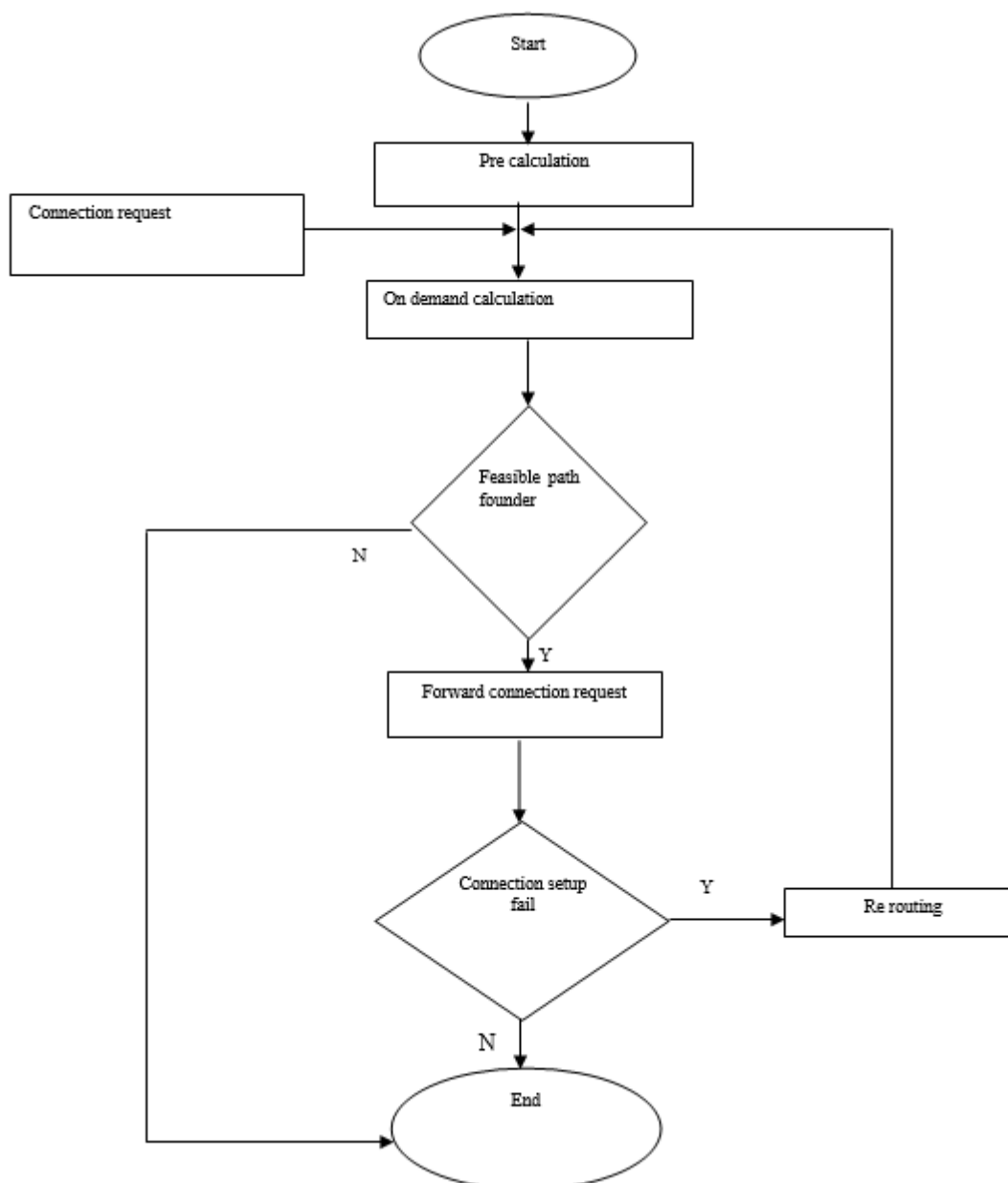
SDH hierarchy table

SDH level	Bit rate
1	155.5
4	622.1
16	2488.23
64	9953.82



Rating in accuracy = Number of incorrect route selections / total number of sessions request





They use either UDP or TCP as a transport mechanism. Remember that UDP is unreliable and offer no flow control so in this case the application has to provide its own error recovery, Flow control, and congestion control functionality. It is often easier to build application on top of TCP because it is reliable stream, connection oriented, congestion friendly, flow control enabled protocol. As a result most application protocol will use TCP but there are applications built on UDP to achieve better performance through increased protocol efficiencies.

#### IV. Conclusion

Load balancing helps the network in many fields i.e. to remove congestion to minimize packet delay and packet loss to increase network reliability and efficiency. In this paper we surveyed various mechanism of load balancing is to find the optimum path to balance the load by calculating various traffic metrics. This mechanism can be deployed in MPLS traffic engineering to support different class of services as per the service level agreement. Routing problems increasingly have to be solved in environments where much of the detailed network state information is unavailable and/or various network technologies are in use. In connection oriented network such as ATM which provides[2] the infrastructure for making reservations and keeping to service guarantee for which the associated routing problem are far from solved and in connectionless network such as IP which also require new required resource management possible.

### Acronyms

CMIP - COMMON MANAGEMENT INFORMATION PROTOCOL  
CWDM – COURSE WAVELENGTH DIVISION MULTIPLEXING  
DWDM – DENSE WAVELENGTH DIVISION MULTIPLEXING  
GMPLS – GENERALIZED MULTIPROTOCOL LABEL SWITCHING  
MEMS – MICRO ELECTRO MECHANICAL SYSTEMS  
OAPM – OPTICAL ADDRESS PROBABILITY MULTIPLEXER  
VCAT – VIRTUAL CONCATENATION  
CELP – CODE EXITED LINEAR PREDICTION  
ETSI – EUROPEAN TELECOM STANDARD INSTITUTE  
DSCP – DIFFSERVE CODE POINT  
IPDC – INTERNET PROTOCOL DEVICE CONTROL  
GSTN – GENERAL SWITCHED TELEPHONE NETWORK  
LDAP – LIGHT WEIGHT DIRECTORY ACCESS PROTOCOL  
MGCP – MEDIA GATEWAY CONTROL PROTOCOL  
POTS - PLAIN OLD TELEPHONE SERVICE  
RTSP – REALTIME STREAMING PROTOCOL  
SGCP – SIMPLE GATEWAY CONTROL PROTOCOL  
VTOA – VOICE AND TELEPHONY OVER ATM  
AESA – ATM END SYSTEM ADDRESS  
GSTN – GENERAL SWITCHED TELEPHONE NETWORK  
MPOA – MULTI PROTOCOL OVER ATM  
NSAP – NETWORK SERVICE ACCESS POINT  
RTCP – REAL TIME CONTROL PROTOCOL  
BGCF – BRAK OUT GATEWAY CONTROL FUNCTION  
BICC – BEARER INDEPENDENT CALL CONTROL  
CSCF – CALL SESSION CONTROL FUNCTION  
DSCP – DS CODE POINT  
ECMP – EQUAL COST MULTI PATH ROUTING  
GGSN – GATEWAY GPRS SUPPORT NODE  
GPRS – GENERAL PACKET RADIO NETWORK  
GSTN – GENERAL SWITCHED TELEPHONE NETWORK  
MGCF – MEDIA GATEWAY CONTROL FUNCTION  
MRFC – MRF CONTROL PART  
MRFP – MRF PROCESSING PART  
PLMN – PUBLIC LAND MOBILE NETWORK  
RTSP – REALTIME STREAMING PROTOCOL  
SGSN – SAVING GPRS SUPPORT NODE  
UMTS – UNIVERSAL MOBILE TELECOM SYSTEM  
VLAN – VIRTUAL LAN

### References

- [1]. Alcatel Lucent, Internetworking LTE/EPC with WCDMA packet switched mobile, <http://www.lakatel-lucent.com>.
- [2]. Giamani Di Caro, 2004, Ant Colony Optimization and its Application to Adaptive Routing in Telecom Network, PP 131 and 89.
- [3]. Steven Will Molt, May 1999, AI for Network Routing Problems, Vol.1, pp 07-10.
- [4]. GioVani Neslia, December 2004, Ingress Traffic Control in Differentiated Services IP Networks, IEEE Trans. PP 2-15
- [5]. Gao Zihe , 2011, A Distributed Leo Satellite Network, ACM Trans. Vol 9, No. 2.