Performance of Inconsistent Networks

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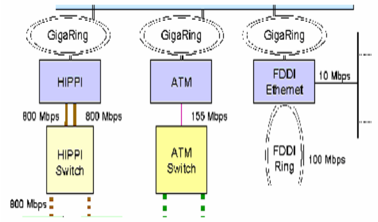
Abstract- *The interconnected networks have many policies and traffic environments. The inconsistency between networks results in set of conflictions that debase the entire performance. The convergence process takes sometime to adapt the sub-networks, this lags the service time at all levels. The inconsistency between the sub-networks during this paper concerns on ten elements: Packet Length (PL) and format, Time assigned for signal To Life (**TTL**), throughput, bandwidth (**BW**) of bottlenecks, error manipulation, damping parameters, routing algorithms, Sub-networks Address Translation (**SAT**) protocols, routing polices and management protocols. This paper discusses also the mutual performance and performs a set of simulation experiments to estimate the interoperability and performance of the entire network. The entire results show that increasing the inconsistency level results in debasing the entire performance.*

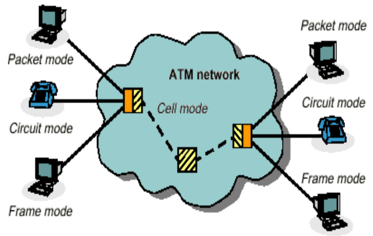
Keywords: inconsistency; channel occupancy; Rendezvous Point (**RP**); Network Elements (**NE**'s); Shortest Path(**SP**);

1.0 Introduction

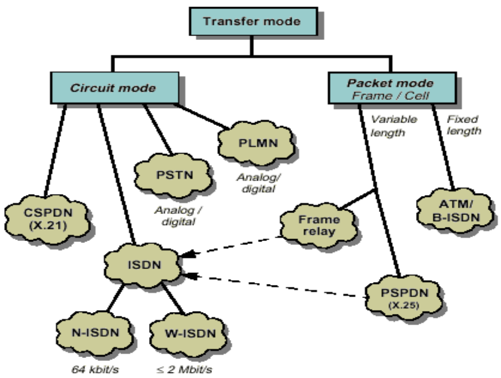
The individual networks have different PL's and formats. Moving from one format to another consumes an extra time delay for segmentation and re-assembly (**SaR**). This delay depends on traffic load and formats on both networks' sites. TTL, damping parameters, QoS parameters and error management tools vary from one network to another at all OSI management layers. Traffic intensities ; bottlenecks like hubs, switches and routers; BW of NE's ; congestion and erroneous traffic cause damping that debases the network serviceability. Network serviceability vary according to the capabilities of infra-structures. The manner that each network deals errors varies depending on the Available Bandwidth (**AB**), Class of Service (**CoS**) and congestion level. The main schemes are: fixing the faulty signals, requesting another correct copy or use a combination of both schemes. The increase of congestion level results in violation of damping factors[1]. The performance metrics like delay, QoS and reliability vary among most of the internetworking layers. The variety in routing schemes, transfer rate, transfer mode and admission control protocols result in inter-networking conflictions. ATM infra-structures match many inconsistent networks that handle variable and unspecified bit rates at real and non-real time [1]. Traffic control metrics in ATM fulfill the least TTL using tools like Cell Loss Priority (**CLP**), Available Bit Rate (**ABR**), Allowed Cell Rate (**ACR**), Peak Cell Rate (**PCR**) and Maximum Cell Delay (**MCD**). Routing have a main effect on network performance. Routers select the proper routes and deliver the messages through them. Tools for selecting the proper paths and tools of delivering messages, differ from one network to another. Inter-domain Routers (**IR**) perform Address translation Tables (**AT**) to guide traffic between sub-domains[2]. Tools to handle and update AT vary according to protocols of the gateway boarders (**GB**). Routers filter and omit the bad routes to reduce the size of Route Access-Tables (RATs) using different tools. Routing Information Protocols (**RIPs**) associate with each source-derived-message, a sequence of hops' addresses [3]. RIPs send Routing-update-Signals (**RS**) to all the adjacent routers to update the spatial configurations and RATs. RIPs define the default interval between the periodic RS = 30 seconds. RIPs use hop-count as a metric to count number-of-hops between sources and destinations. Number of hops in each path is limited to be ≤ 16 hops[3]. Sakr [4] showed that increasing the average number-of-hops per path results in a longer time for traversing messages via that path. RIPs assign a default value for route-time-out = 180 seconds if no reply is reflected after sending a signal[5]. The time-out and number-of-hops differ from network to another. RIPs use the split-horizon or hold-down mechanisms to prevent looping and propagation of faulty information[6]. RIPs use link-status flags to test the reachability to the next hop using reliability and preference of each route. Routers use various time-stamps to discard the time-out traffic. Each router assign a route-timeout timer for the connected routes. When the route-timeout expires, the route is omitted from RATs. When routes are repaired, the routers reconfigure RATs. Accordingly, RIPs authorize or deny routes[7]. Routing Algorithms (**RA**'s) have different schemes: static or dynamic routing; single-path or multipath routing; flat or hierarchical routing; internal or external routing; flood or source-derived routing and hop-by-hop or end-to-end routing. Routers identify the SP regarding five metrics [5]: ToS, delay, reliability, bandwidth and load. Routing processes have seven axes[6]: configuration of routers; level of routing ;domain ; routing schemes; direction; casting mode "unicast, multicast, fuzzycast or broadcast" and multicasting type "simple or hierarchical". Fig-1 shows a wide variety of interconnected networks.



a- Inconsistent Interactive Rates



b- Inconsistent Signaling Modes



c- Inconsistent Transfer Modes

Fig. 1. Samples for Inconsistent Networks.

2.0 Compatiblizing the Inconsistent Sub-Networks (A Survey for Related Works)

BW on Demand (**BoD**) and BW Allocation Tools (**BATs**) reduce the blocking rates between sub-networks and improve their compatibility. BATs collect the infinitesimal fragments to form new windows may be enough to handle more traffic.

The excess in AB = Σ partial fragments……(1)

BATs improve the AB and results in reducing the blocking rate within the network[8,9]. This section explores the inconsistent routing protocols and algorithms and protocols to converge sub-networks. RAs handle Multiple Protocols Routing (**MPR**). Multiple Protocols over ATM (**MPoA**) can manage up to 32 different protocols[10]. MPoA use multicasting address resolution and Dynamic Hop Configuration Protocols (**DHCP**) to minimize the entire SAT. Routes aggregation reduces the size of RAT by hiding the unnecessary routing information. Time to select a route from RAT $O\log_2 n$, where n is number of nodes[8]. MPR causes imbalanced loads on NE's and necessitates more times to reconfigure the network mesh[11,12]. MPR use the SP algorithms of Dijkstra[13] regarding metrics like: delay, utility, reliability, CLR, ABR, ACR, PCR and MCD. Many of the multicast protocols shown in table-1, are inconsistent.

Table-1[14,15]. **A Collection for the Inconsistent Routing Protocols.**

Distance Vector Multicast Route Protocols DVMRP
Hierarchical DVMRP (HDVMRP)
Multicast Open Shortest Path First (MOSPF)
Link State Routing Protocols (LSP)
Fuzzy Routing Protocol (FRP)
Reverse Path& Forwarding Protocols (RPF)
Protocols Independent Multicast (PIM)
Multicast Internet Protocols (MIP)

The MOSPF protocols apply the MPR to multicast traffic within the SP [14]. Routes are discriminated by five metrics[16]: shortest delay, longest delay, capacity, availability and quality. Routers forward periodic Link-State Advertisements (**LSA**) to all the adjacent routers to update their RAT and avoid the congested portions[17]. GB look ahead on the NATs to discover the shortest and loop-free paths. IR translate the Domain Name Servers (**DNS**) into IP addresses. PIM identify the Loop-free Spanning Trees (**LST**) to accomplish the minimum delays and handle two routing modes [18]: Dense Mode (**DM**) or Sparse Mode (**SM**). DM is source-rooted tree, fulfills low delay and is efficient for networks with extensive dynamic multicast. SM is a shared tree, uses RP as networks' backbones [19]. SM suits the non- widely deployed multicast and uses Hierarchical-Tree-Routing (**HTR**) to decide the proper route. HTR divides the routing domains into areas connected via hierarchical RP. HTR provides the fastest loop-free routing scheme. DVMRP uses RPF to forward packets to all routers except the one they come from[20]. DVMRP verify preferences of routes, list of next hops and the nested routes. RP may be informative like RIP or operative like OSPF. HDVMRP use hierarchical DVMRP to minimize the entire delay. MIP uses both SM and DM spatially to aggregate and partition routing domains, which results in minimum routing time [21,22]. Time to update the RAT $O(n.\log_2 n)$ where n is number of nodes in the network [23]. DVMRP apply the RPF to prevent looping[23]. Routing policies permit, prefer or deny routes for the egress and ingress traffic. Routers consider both policy constraints and performance constraints to select the proper routes. Quality of a Route (**QoR**) refers to the set of metrics like: AB, delay of route, BW, TTL and jitters clipping tools. Robustness debases the QoR [24]. Routing Operation and Maintenance protocols (**ROMs**) reduce the entire routing time; install and refresh the routes; reconfigure RAT; diagnose congestion and loops; and detect the bad routes and blackholes [8]. ROM's adapt timestamps, traffic violations and authentications of the used routes.

3.0 Forming Models for the Problems

There are many problems arise when converging the inconsistent networks like: avoiding the congested bottlenecks; avoiding the blackholes; specifying the dynamic routes; and specifying the route-timeout limitations. Trouble shooting reasons like blackholes and congested bottlenecks consume throughput and block network admission. This section presents a set of mathematical models. First, paper suggest a dynamic function for the Time necessary for a signal To Life during its trip in the network (**TTL**). This suggestion aims to omit the messages after a specific time using a linear function (for simplicity). The TTL time function is load dependent. This results in a less risky state and makes load more stable.

TTL = int (180 *(1- load / capacity))…..(2)

Where signals have a default life time through the route = 180 sec [6], "int" is the function that return an integer value for the expression. This value tends to zero when load on NE has full capacity. Hold Down Algorithms (HDA)are applied to clear the duplicated signals after a default time= 30 seconds from the time in which the signal is received correctly[6]. TTL is standardized for sub-networks, the elapsed time stamp is tested for each signal. If elapsed time = TTL, the signal is canceled. This equation could be embedded in the HDA to filter the traffic. TTL affects negative reachability signals that inform routers that the signal failed to reach its destination, if it couldn't be heard within the adapted TTL time.

The priority of traffic could be adapted also in a similar function to allow the early traffic of low priority to get a higher one depending on the elapsed time, which improves the serviceability of the network. The next relation presents a linear function to update the priority:

priority = priority + int(4*elapsed time/180)….(3)

Where number of priority levels is assumed to be 5, depending on classes of ATM. The new priority has a ceiling limit=5. The elapsed time has a ceiling value = TTL. During this section the paper presents some mathematical models that concern consistency of sub-networks to be used for simulation during the next section, these models concern with the five axes that affect network performance: throughput, delay, utility, reliability and interoperability.

3.1 Concerning Throughput

Throughput of different networks differ according to the effective BW of the backbones of these networks. The Effective Throughput (ET) is the crossing traffic after filtering the faulty, time-out and the low-priority portions. The suggested relation could be:

ET= throughput (1- ε), ………..(4)

where ε is the percentage of filtered traffic, it is a function of TTL, error ratio and traffic intensity. ET is debased when congestion level increases. Congestion is tested periodically or at risky conditions. When ET increases, it results in increasing the average delay per message which results in turns in more loss ratio[16]. Kuo [8] showed that for each network, there is an optimal delay that fulfills a maximum throughput. This delay depends on traffic intensity, traffic distribution and management tools. Jain[16] showed that throughput is related to the loss ratio, PL and Round Trip Time (**RTT**) by the relation:

$$\text{throughput} = \frac{\text{packet length}}{\text{round trip time} * \sqrt{\text{probability of packet loss}}}$$

This relation indicates that the shortest path results in a maximum throughput. For any packet:

TTL ≥ ΣRTT ……(5)

If the relation doesn't hold, packets will be deleted before they arrive to their destination.

ET/ throughput =1- percentage of dummy traffic.(6)

Where the ratio: ET/throughput, indicates the percentage of effective traffic. To improve the entire throughput, High Performance Parallel Interfaces (HiPPI) and terabit routers are used as bottlenecks. The effective bandwidth at bottlenecks is estimated by the relation[9]:

$$\text{BW} \geq \frac{\text{average load}}{\text{average waiting time + average transmission time}}$$

The ratio between effective BW of a NE and its actual BW indicates the utility of $\forall$ NE_i, i ε(1,n), where n is number of NE's forming the backbones. The mutual throughput between two sub-networks:

$\text{Throughput}_{i,j} \leq BW_{GB}$…………….(7)

Where GB forms the bottleneck between both sides. The mutual throughput is non-reflexive, i.e.:

$\text{Throughput}_{i\text{-}j} \neq \text{throughput}_{j\text{-}i}$ ……….(8)

The throughput of the entire network is a function of BW's of all GB's in mesh. The inconsistency in throughput debases the entire performance.

3.2 Concerning Delay Time

The average delay time per packet is a function of waiting times on queues at different GB's, distribution of loads among bottlenecks and BW's of NE's.

The end-to-end delay per message =
Σ capsulate times + Σ peel times +
Σ times to deliver through the successive hops +
Σ times to select the VC to the successive hops +
Σ address translation times ………..(9)

The time to peel a message = the time to peel a cell
* average number of cells per message …(10)

Time to assemble messages at their destination depends on cells' delay variation (**CDV**) which is a function of the intermediate queues[25]. Number of intermediate hops affects the end-to-end delay, where:

transition time = number of hops per path *
average time to handle a cell at hop …..(11)

The average time to survive a path increases at the congestion states. This time depends on network topology, the delivering protocols, the scheduling discipline, time to respond the control signal and utilities of NE's. Time to reconfigure the RAT affects also the delay time. Therefore, the short time between testing signals save time of re-routing through alternative paths, but it increases the percentage of dummy traffic. The time taken to translate addresses in GB (for tunnels and brouters) depends on the rate of migration. Time to adapt protocols in GB consumes an additive time and increases the compatiblizing delay between the sub-networks. Kuo [8] showed also that for each network, there is a PL that fulfills a minimum delay per packet. Sakr[4] showed that the shorter PL results in a reduced congestion and more uniform traffic. Algorithm of Dijkstra for constructing a Minimum Spanning Tree (MST) has a complexity $On*\log_2(t)$, where n and t are number of nodes in RAT and number of tree branches respectively[13]. Gross[25] showed that number of waiting packets in a GB between two sub-networks =

$$.\ \rho + \frac{(\rho^2 + \lambda^2\sigma^2)}{2*(1-\rho)}$$

Where ρ is the route utility = arrival rate/service rate, λ is the arrival rate and σ^2 is the average variation in packets service time. This relation indicates that the traffic which has more variation in service time results in more waiting time. The increase in inconsistency value (σ^2) results in an increase in waiting delay. The waiting time is minimum for consistent networks (where $\sigma = 0$). Therefore, segmenting packets into a standard size fulfills the least delay.

$$\text{The minimum waiting time} = \frac{\rho(1-\rho/2)}{1-\rho} \ldots.(12)$$

The last relation has a minimum value=0 at ρ=1(*using the mean value theory*). i.e. the minimum waiting time per cell is accomplished at full utility (ρ=1) not 0.75 as shown by Chen[9]. The waiting time[25] for the unlimited services "M/M/∞ discipline" tends to 1/ μ where μ is the average service rate. Chen[9] indicated that multicasting, broadcasting and fuzycasting reduce the ET. Each type of casting has its own influence on throughput and delays. Chen proved that random walk of messages reduce the efficiency of delivering, while the predefined routing result in a faster delivery. He estimated the average delay per packet t_i =

$$\frac{1}{packet\ arrival\ rate} * \sum_{\forall channels\ i} \frac{flow\ in\ channel_i}{capacity\ of\ channel_i - flow\ in\ channel_i}$$

Where the flow in channel i must be ≤ its capacity. Chen estimated an optimal occupancy for NE to be ≈ 0.75 , that results in minimum delay.

The mean delay variation = **MDV** =

average $_{i,k}$ (time to pass through the alternative route - time to pass through the shortest route)$_{i,k}$…(13)

This time causes inconsistent flow and increases the waiting time to assemble messages at the destination. The rate of increase in number of waiter in network

$$= \frac{\Delta waiters}{\Delta t} = arrival\ rate - service\ rate \ ....(14)$$

Where the incremental number of waiting cells, arrival rates and service rates are all statistical and time dependent. Cumulative number of waiting cells :

$$w(t+\Delta t) = w(t) + \Delta\, w(t) = \rho + \frac{(\rho^2 + \lambda^2\sigma^2)}{2*(1-\rho)} + (\lambda-\mu)*\Delta\tau \ldots\ldots(15)$$

This number is a decreasing function if $\lambda<\mu$, constant if $\lambda=\mu$, and explosive if $\lambda>\mu$. This number is minimum for uniform service rate where σ=0 , the case of consistent traffic. Increasing the Threshold Admission Delay (**TAD**) results in more admissible calls and less call-blocking rate[6]. TAD varies from network to network. Channel BW affects permanently the blocking ratio especially at bottlenecks.

3.3 Concerning utility

Utility (ρ) of a NE specifies the percentage of its occupancy, it is strongly related to congestion level. This metric is a dynamic function of bandwidth and spatial loads. It is tested periodically or occasionally "at congestions". Utility of a NE has a maximum value at the optimal load[9] and decay if load increased. The load on a NE is not the same at forward and reverse directions, therefore, the shortest route from x to y may be not the shortest one from y to x. The utility of a router is not the same at all ports, some port are congested than others as shown in fig-2. This statistical study was done on a router of Saudi-telecom , Buraidah, KSA.

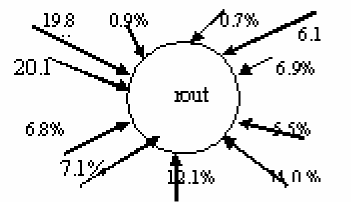


Fig.2. Inconsistent Statistical Relative Weights for Loads on a Random Router.

High utility results in saturation of NE and increasing the waiting delay exponentially [4]. This enforces network sponsors to expand BW of the current NE's. Utility of a NE could be enhanced by clipping traffic intensity using dynamic TTL or dynamic priority. This filters the timeout messages and increase the priority of early messages to hurry up their processing. Congestion may cause a deadlock-utility that reduce the benefits and usability of networks [7]. The expected ρ of the NE can be estimated to a normal distribution N(μ,σ), where μ,σ are the mean and standard deviation of load at the NE, they both are dependent on dynamic load.

$$\mu = \Sigma \text{ percentage weight of traffic in channel}_i * \text{average } \rho \text{ of channel}_i \ \ldots.(16)$$

$$\sigma = \sqrt{\Sigma \text{ Percentage weight of traffic in channel i * variance of utility of channel i}} \ \ldots(17)$$

ρ tends to "1" at the peak requesting periods. Network utility depends mainly on the reliabilities of GB's.

3. 4 Concerning Reliability

Reliability of the entire network is a function of the reliabilities of NE's forming the bottlenecks. The improve in reliability of NE results in improving admissibility to the networks. The increase of number of requests for a NE increases its utility and may results in blocking service. This debases the spatial reliability. The reliability is estimated by:

1- expected failure rate ……….(18)

which is a statistical time dependent function, it depends on the request and service ratios.

Frankline[8] estimated the expected failure rate =

$$\frac{\text{number of connection request failures} + \alpha * \text{number of transnetworking failures}}{\text{total number of connection request}}$$

where α is a fraction depends on the congestion level. Many other factors affect the reliability of routing such as desirability to use or avoid a specific path, authenticating or denying the use of a specific NE, and the real time usability of NE's. Cohen[7] showed that the increase of connect request ratio results in more blocking ratio. The mutual reliability depends on the blocking rates between sub-networks. It is not a reflective function, i.e.: $R_{ab} \neq R_{ba}$

3.5 Concerning the Interoperability

Interoperability tends to match processing between sub-networks. This metric is closely related to the consistency between the adjacent modules. The isolated sub-networks are non-interoperable, so the mutual interoperability=0. The improvement of consistency increases the interoperability between sub-networks. Interoperability between two sub-networks =1 iff they are completely consistent in BW's, protocols and traffic flow. In the completely consistent sub-networks, the GB is considered a mirror to both sides and no congestion occurs. This means that a cell merge between the two sub-networks as if it merges through one module.

Certainty that two sub-networks are interoperable = percentage of their consistency. ……(19)

The likelihood between two sub-networks could be estimated by the linear relation:

Prob. of likelihood = 1- inconsistency……(20)

The inconsistency is measured by the percentage of variety between two adjacent sub-networks. The inconsistency ,as well , is not a reflexive function, i.e. $\text{inconsistency}_{i\text{-}k} \neq \text{inconsistency}_{k\text{-}i}$

The next section indicates a set of simulation experiments that support the upper relations concerning the interoperability and consistency.

5.0 Simulation Experiments

This section focuses on metrics of consistency. For 1Gb/s router, the time to translate address at GB and select the proper VC is shown in fig-3, regarding number of GB. Number of GB within a path affects the average rate of path selection.

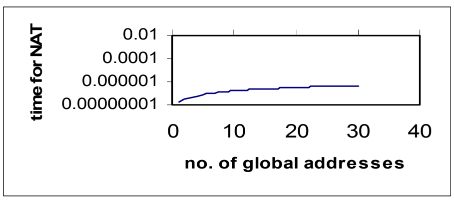


Fig. 3 . Relation Between Time of Translating NAT and Number of GBs.

The next simulation experiment indicates the relation between number of hops (handovers) per path and the expected time to decide the path. If "T" is time to select the next hop, "n" is average number of hops. Then time to decide the path O(n * T), T O $\log_2 n$, this gives that the expected time to decide a path $On*\log_2 n$ [23]. Analyzing the simulation results for servicing rate "=1G instructions/s", assuming that each IP address has 3 cycles before deciding the VC (fetch, compare and decide the next hop) and applying Dijkstra's algorithm for selecting the shortest path, the next simulation results show the relation between time to select a path and number of hops, considering max hops/path=16.

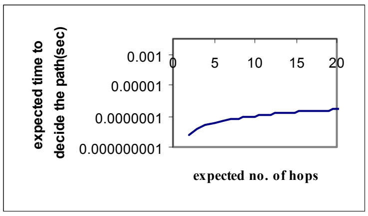


Fig- 4. Expected Time to Select a Path versus Number of Hops .

Fig-5 indicates the simulation results for probability of likelihood between adjacent sub-networks and time to select the path. It shows that increasing number of hops, results in more inconsistency.

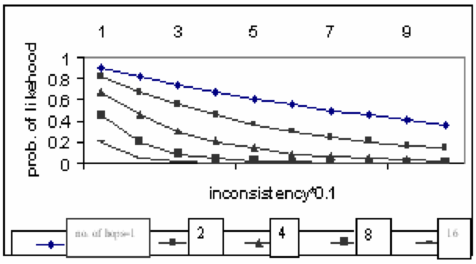


Fig-5: Probability of Likelihood Between Adjacent Sub-networks versus inconsistency.

The next simulation experiment indicates that improving the consistency level results in improving the interoperability. The inconsistency was defined in abstract has ten levels, each axis has a weight of 0.1 of the entire value.

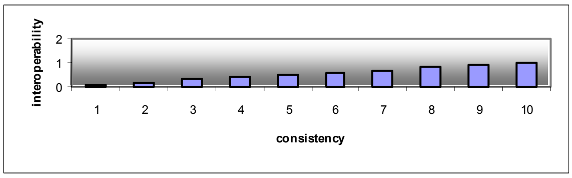


Fig. 6. Relationship Between Consistency and Interoperability.

The expected waiting time increases with the increase of load. The increase of load leads to more network utility and may lead to congestion. The next experiment shows the relation between the expected waiting time and utility of NE's.

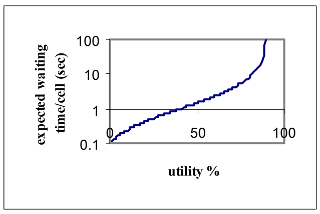


Fig .7. Expected Waiting Time versus Utility

Fig-8 indicates the simulation results for time needed for segmenting and assembly of packets that have different PL. Simulation environment assume that the PL varies between 50bytes and 10kbytes. The rate of routers is 1Gb/s. The packet overhead is assumed to be 32 bytes for CRC, VPI, VCI, next hop identifier, packet sequence number, synchronization and flags. This overhead is associated to each segment. The expected time in nsec. is: 2* int(PL/48)*32*8 , where 2 for SAR, PL/48 is number of cells in packet, 32*8 is the processed head time in nanoseconds.

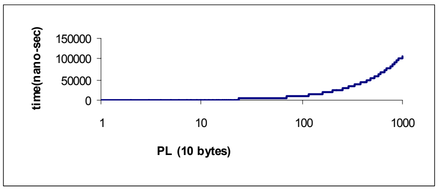


Fig.8. **Time for SAR Packets versus PL.**

Assuming the transition rate at the router =1Gb/s, the next results show the relation between transition delay and packet length (bytes).

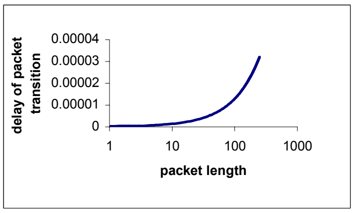


Fig. 9. Effect of PL on Delay of Transition.

The experiment in Fig-10, indicates the proper times for test as a function of congestion. The experiment assumes NE with capacity 10Gb/s, the load varies from 10% to 90%, a check signal is associated with a finite frame size (assumed 1Gb). Increasing the load results in reducing the time between control signals. This may increase the traffic burstness, but results in a proper reliability. These signals may reduce the NACK, and allow early fails discovery.

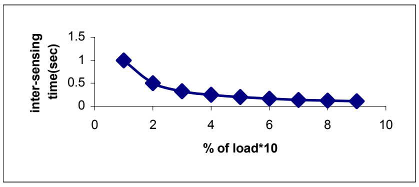


Fig.10. Time Between Sensing Activity of NE versus Load

Applying the dynamic priority for traffic reduces the blocking ratio. The next experiment indicates the admissibility versus load using fixed priority, dynamic priority and non priority schemes. The fixed priority traffic assumes five priorities for the five classes in ATM, where the high priority ToS is block free and the low priority traffic suffer from a specific blocking ratio. The dynamic priority assumes linear function that gives a higher priority for the earliest 10% of traffic, to speed up their processing.

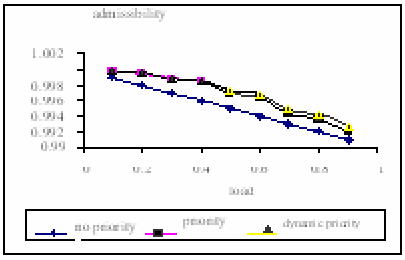


Fig.11. Admissibility versus Load Using Variable Priority Schemes.

The next experiment indicated the dynamic value of TTL as a linear function of load intensity as assumed in relation 1.

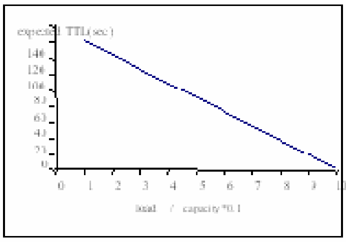


Fig.12. Expected TTL versus Load Intensity .

TTL varies regarding throughput and number of hops per path. The maximum TTL allowed in a path is 180 sec., therefore, the signal have to adapt its stamp time at congestions to reduce the omitted ratio due to time consumption. Sakr[4] showed that increasing the load results in increasing the average delay per cell, which may lead the filter protocols to cancel more cells when burst state arises. The paper suggest to increase the TTL for cell as a linear function of load intensity. Simulation experiment in fig-13 indicates an estimation for TTL that fulfills a minimum cell loss ratio. The experiment use a maximum hops=16.

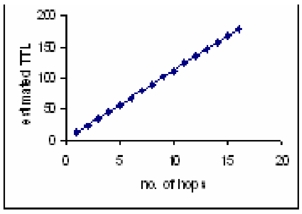


Fig.13. Expected Time Stamp Associated with Cells

The next experiment indicated the expected waiting time in a NE, as shown in relation 15. The suggested μ is 1 Gbps and the load is assumed to vary from 1kbps to 1Gbps.

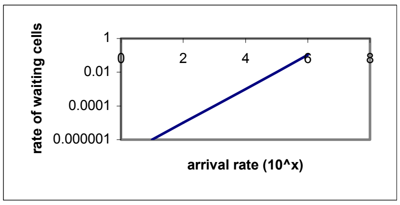


Fig.14. Expected Waiting Time in a NE

5.0 Conclusion

This paper explores many factors that affect the inconsistency and interoperability of sub-networks. The mutual inconsistency between sub-networks debases the entire performance. The paper concerned with the inconsistency factors like: routing schemes, traffic intensity, delay variation, admissibility and TTL. The paper presented set of mathematical models supported with simulation experiments. The simulation experiments result in: increasing number of GB results in more time to decide the proper path; the interoperability between adjacent networks improves when the software and hardware are more consistent; reducing TTL of signals at congestion helps in reducing the congestion level; increasing the TTL improves the serviceability of the network; increasing of TAD reduces the blocking ratio; and that the admissibility to the network at heavy loads could be enhanced using dynamic-priority traffic. The paper presented the interoperability as a new metric to measure the inter-networking consistency and mutual performance. The paper presented a stochastic study for a random SP that indicates the non-uniformity of uploading , the same non-uniformity is expected for downloading actions. The paper also developed set of experiments that estimated the admissibility, TTL and time between tests as a function of load. Unfortunately, increasing the load necessitates more testing rate to follow up the performance more precisely, this increases the level of burst and reduces ET.

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