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ENERGY EFFICIENT NODE DEPLOYMENT WITH ROUTING TABLE ESTIMATION FOR SECURED 3G/4G MOBILE DATA NETWORKS

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Abstract: Data transmission in the 3G/4G Mobile Data Networks works in increased data rate Due to the collision in the network the delay is increased while data transfer and the energy efficiency reduced. So a new method called Node deployment with routing table estimation is used to reduce the delay in the network. Ant-net algorithm helps to enhance the performance parameters of Adhoc On-Demand Vector to find the shortest path. Probability Density functions are created to establish the node deployment and the routing table generation is done by the Routing table generation method. Queue length and the buffer length calculation are done in the NDRT estimation. Hash key is generated to increase the security of the network.

Keywords: Node Deployment, routing table, hash key, queue length, buffer length.

1. INTRODUCTION

Wireless communication technology has been making significant progress in the recent past and will be playing a more and more important role in access networks, using widespread adoption of wireless local area networks (WLANs), wireless home networks, and cellular networks. These wireless access networks are usually interconnected using wired backbone networks, and many applications on the networks run on top of the transmission control protocol/Internet protocol (TCP/IP). Inferring the unused capacity or *available bandwidth* is of great importance for various network applications [3]. Obtaining useful estimates of the available bandwidth from routers is often not possible due to various technical and privacy issues or due to an insufficient level of measurement resolution or accuracy. The key to the successful integration of a new or enhanced transport protocol is to use standard based and well-structured software components that have predictable behaviors under a large number of scenarios[7]. We believe it is a good practice to have a TCP protocol that has both predictable performance and nice social behaviours under diverse scenarios. The TCP protocol should becapable of high throughput when available bandwidth permits, and it should have good network behaviours such as maintaining stable and small queueing delay and not forcing heavy packet losses. The long-term evolution (LTE) as defined by the 3rd Generation Partnership Project (3GPP) is a highly flexible radio interface LTE supports both frequency-division duplex (FDD) and time-division duplex (TDD), as well as a wide range of system bandwidths in order to operate in a large number of different spectrum allocations[8].

We find that nearly half of the paths measured have a non-access bottleneck link with available capacity less than 50 Mbps. Moreover, the percentage of observed paths with bottlenecks grows as lower-tier destinations. The bottlenecks identified are roughly equally split are roughly equally split between intra-ISP links and peering links between ISPs [1]. Our observations provide key insights into the location and nature of performance bottlenecks in the Internet, and in some cases, address common impressions about constraints in the network. To obtain a robust estimate, it is necessary to develop an estimator that allows the identification and elimination of noise due to cross traffic along the network path [6].

Based on all the above references and concepts, we developed an energy enhanced 3G/4G mobile data network by providing high security using Hash key method.

2. EXISTING SYSTEM

The emerging mobile data networks fuelled by the world-wide deployment of 3G, and LTE networks created new challenges for the development of Internet applications. In mobile data networks it is essential to be able to accurately characterize two key network properties: queue length and buffer length in bottleneck. Existing work tackles the challenge in these two network properties in modern mobile data networks.

The 3G/LTE radio consumes significant amounts of energy; Battery hog is well-known to most users anecdotally and from experience, and much advice on the web and on blogs is available on how to extend the battery life of your mobile device. Unfortunately, essentially all such advice says to "disable your 3G data radio" and "change your fetch data settings of network to reduce usage. We show the measured values of 3G energy consumption for multiple Android applications. For most of these applications (which are all background applications that can generate traffic without user input, except for Facebook), less than 30% of the energy consumed was during the actual transmission or reception of data Previous research arrived at a similar conclusion: about 60% of the energy consumed by the 3G inter face is spent when the radio is not transmitting or receiving data. In principle, one might imagine that simply turning the radio off or switching it to a low-power idle state is all it takes to reduce energy consumption. This approach does not work for switching between the active and the different idle states takes a few seconds because it involves communication with the base station, energy consumption of switching states, switching incurs signalling overhead on the wireless network, which means that it should be done only if the benefits are substantial relative to the cost on the network. This paper tackles these challenges and develops a solution to reduce 3G/LTE energy consumption without appreciably degrading application performance or introducing a significant amount of signalling overhead on the network. Unlike currently deployed methods that simply switch between radio states after fixed time intervals an approach known to be rather crude and sub-optimal, our approach is to observe network traffic activity on the mobile device and switch between the different radio states by adapting to the workload. The description given above captures the salient features of the 3GPP standard. Another popular 3G standard is 3GPP2.

Disadvantages:

- Low efficiency and low reliability.
- Delay is occurred in many areas. Collision is also present in areas due to the increase of the congestion in the network.
- Network Security is very low in this network

3. PROPOSED SYSTEM

A new method called NDRT estimation (Node deployment with routing table) is used to reduce the delay in the network. Ant-net algorithm helps to enhance the performance parameters of AODV to find the shortest path. PDF functions are created to establish the node deployment and the routing table generation is done by the RTG method Queue length and the buffer length calculation are done in the NDRT estimation. Hash key is generated to increase the security of the network. The performance evaluation is done with the parameters called Packet delivery ratio, Threshold value, network throughput, Data loss, initial energy, remaining energy and the simulation time period.

This work offers a first look into the impact of TCP protocol efficiency on mobile network capacity. Given the extremely high cost of mobile network infrastructure, the loss of even a few percent of the network capacity can be very costly. Yet our analysis revealed that unless the cell operates at high traffic load, both NDRT and LTE networks would suffer from significant capacity losses. This calls for the need to further optimize TCP for use over mobile data networks. In addition, our analysis of channel-limited capacity loss revealed that upgrading the mobile base station to higher-speed mobile standards is an effective way to improve cell bandwidth utilization, even if the underlying cell capacity is kept unchanged. In addition, the analysis also revealed an inter-play between cell capacity and protocol/channel bandwidth limit. In particular, too large a cell capacity may not necessary improve service quality significantly as the bottleneck would be shifted to the protocol/channel bandwidth limit. This opens up a new problem/opportunity in the allocation of bandwidth to mobile cells, as a cost-effective allocation will need to incorporate the impact of protocol efficiencies and channel bandwidth limits, in addition to traffic load and other network parameters. These have substantial economic significance to mobile operators and thus warrant further investigations.

This problem is not completely unnoticed by today's smart phone vendors. Our investigation into the open source Android platform reveals that a small untold trick has been applied to mitigate the issue: the maximum TCP receive buffer size parameter has been set to a relatively small value although the physical buffer size is much larger. Since the advertised receive window cannot exceed the receive buffer size and the sender cannot send more than what is allowed by the advertised receive window, the limit on tcprmem max effectively prevents TCP congestion window (cwnd) from excessive growth and controls the RTT (round trip time) of the flow within a reason- able range. However, since the limit is statically configured, it is sub-optimal in many scenarios, especially considering the dynamic nature of the wireless mobile environment. In high speed long distance networks (e.g., downloading from an oversea server over 4G LTE (Long Term Evolution) network), the static value could be too small to saturate the link and results in throughput degradation. On the other hand, in small bandwidth-delay product (BDP) networks, the static value may be too large and the flow may experience excessively long RTT

Advantages:

- By using the Ant-net Algorithm the shortest path between the nodes are identified.
- By increasing the life time, the energy efficiency is also increased.
- On behalf of the shortest path the Delay is reduced.
- Security of the network is increased by the hash keys in the network.

- Increased packet delivery ratio.

3.1 Architecture diagram:

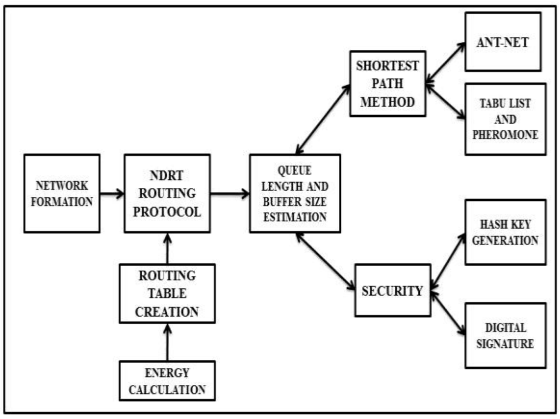

Fig1: Architecture diagram for proposed system

3.2 List of Modules:

- Network formation
- NDRT Routing protocol
- Queue length and buffer length estimation
- Shortest path method
- Hash Key Generation
- Performance Analysis

3.2.1 Network formation:

Mobile data network forms the network using four steps (Route request, Route reply, Route error, Hello message)

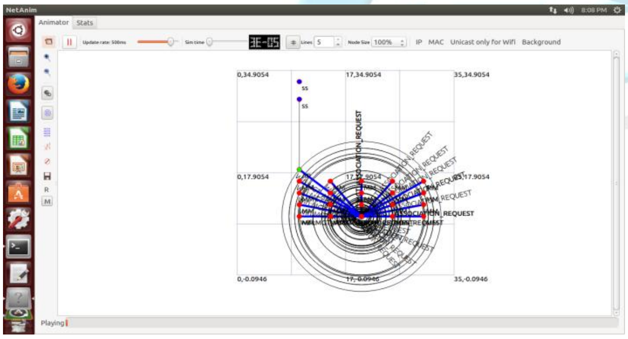

Fig2: Network Formation

3.2.2 NDRT routing protocol:

Probability Density Functions(PDF) are used to establish the node deployment and rounting table is generated using RTG method.

Adhoc On Demand Vector(AODV) a reactive routing protocol is used to generate the routing table for nodes which are in demand.

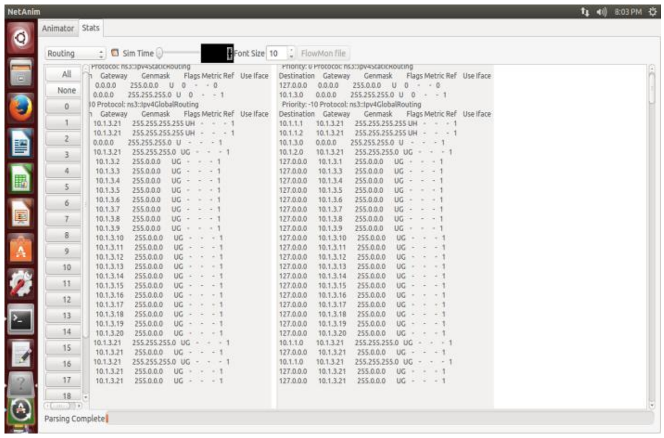

Fig3: Routing table

3.2.3Estimation of Link buffer size and Queue length:

Crovella [2] proposed the method called max-min which is used for the estimation of link buffer size from the estimated transmission delay and the differences between maximum and minimum round-trip times (RTTs). Consequently a frame loss will translate into longer transmission time, resulting in a sudden bandwidth drop. Therefore sum-of-delays(SoD) algorithm to estimate link buffer size of the network and queue length of the underlying network connection. RTT is used in both the principles of max-min [2] and Loss-pair [3] methods to estimate the link buffer size. We show that loss pairs can be used to estimate bottleneck buffer sizes of drop tail routers[3]. By increasing the link buffer size in the network, the packet delivery ratio and the packet send ratio is increased. And overall performance of the network is also increased. RTT is calculated to achieve the link buffer size as [1]

$q_{max} = \max\{(L-1)T+\delta\} = LT$
$rtt_{max} = P+T+U+q_{max}$
$rtt_{max} = rtt_{min}+LT$

By rearranging terms, we can determine the link buffer size from [4],

$$L = \frac{rtt_{max} - rtt_{min}}{T}$$

With knowledge of measured link capacity (C) and known packet size (S), we could compute the estimated transmission delay (T) from

$T = S/C$

and determine the link buffer size accordingly:

$$L = \frac{(rtt_{max} - rtt_{min})C}{S}$$

Queue length can vary from time to time depending on many parameters, including offered traffic load, traffic characteristic, link capacity, and so on[1]. Therefore queue length measurement is meaningful only in the context of the actual data flow generated by the transport and/or application protocols. If the Queue length of the network is increased means the coverage area of the network is also increased. Queue length estimation is implemented by two queue-length estimation algorithms, namely the Vegas algorithm in TCP Vegas and the FAST algorithm in FAST TCP, both using passive estimation [1].Timing of Actual packets and ACK packets are used for the estimation of the queue length. Sender continuously measures the RTT, denoted by *rtti*, and records the congestion window size, denoted by *cwndi*, at the time

ACK packet *I* is received[1]. It then keeps track of the minimum *rttmin* by[1]

$rttmin = \min \{rtti \mid \forall i\}$

and then computes the estimated queue length is calculated as

$$n'i = \frac{cwndi\ (rtti - rttmin)}{Rtti}$$

where*cwndi/rtti* is the estimated bandwidth and *(rtti −rttmin)* is the queuing time. Hence their product is the estimated queue length. If the Queue length of the network is increased means the coverage area of the network is also increased.

3.2.4 Shortest path method

Shortest path is identified using the method called Ant Net which is inherited from the Ant Colony Optimization (ACO).Used to find optimal paths inside of a graph and give approximate solutions to optimization problems based on ants method of finding food.

3.2.5 Hash Key Generation:

MAC algorithm is used to encrypt the message send from sender. Along with a key (K) is also generated. So a double encryption and decryption process takes place which will be more secured for sending a credential messages.

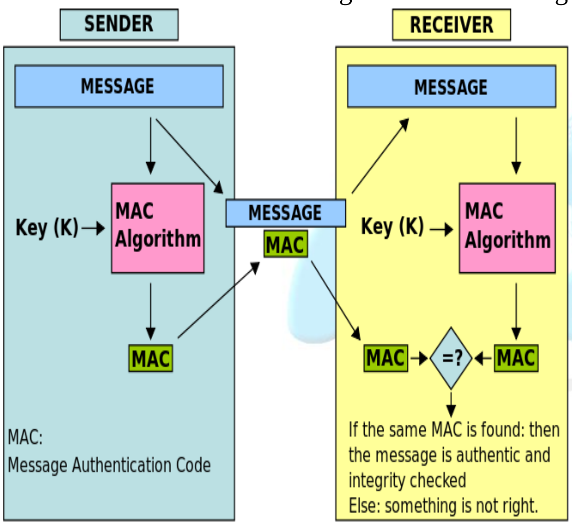

Fig4: Technique of generating hash key.

3.2.6 Performance Analysis:

The performance evaluation is done with the parameters called Packet delivery ratio, Threshold value, network throughput, Data loss, initial energy, remaining energy and the simulation time period.

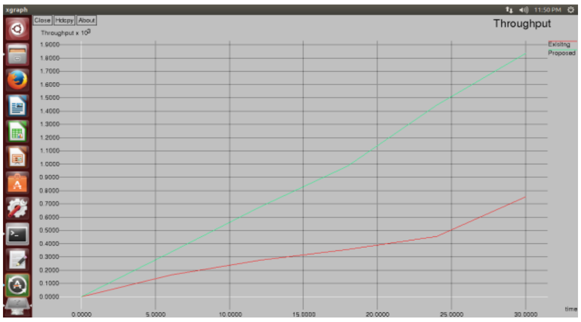

Fig5: Throughput analysis

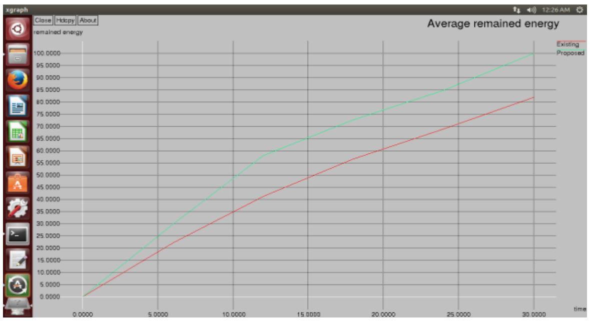

Fig6: Average End to End Delay

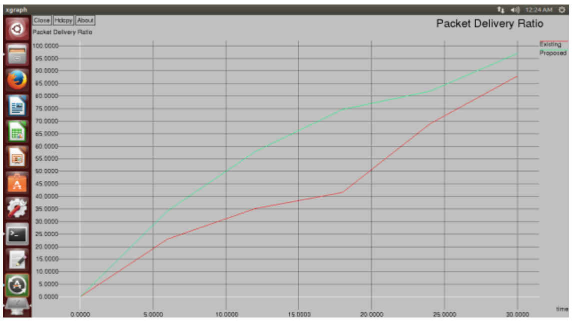

Fig7: Packet Delivery Ratio

IV. CONCLUSION

A new method called NDRT estimation (Node deployment with routing table) is used to reduce the delay in the network. The performance evaluation is done with the parameters called Packet delivery ratio, Threshold value, network throughput, Data loss, initial energy, remaining energy and the simulation time period.

In this paper, we thoroughly investigated TCP's behavior and performance over cellular networks. We reveal that the excessive buffers available in existing cellular networks void the loss-based congestion control algorithms and the naive solution adopted of setting a static tcprmem max is sub-optimal. Built on top of our observations, a dynamic receive window adjustment algorithm is proposed. This solution requires modifications only on the receiver side and is backward-compatible as well as incrementally deployable. We ran extensive experiments over various cellular networks to evaluate the performance of our proposal and compare it with the current implementation. Experiment results show that our scheme makes RTT 24 _ 49% lower than the current implementation of TCP while throughput is guaranteed to be the same in general cases or up to 51% higher in a high speed network with long latency.

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