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A QoS centric layered transmission topology for multimedia data in wireless cellular networks

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ABSTRACT - Here in this paper, we propose a routing topology for audio visual data transmission over Cellular networks, which is Quality of Service Centric. A novel scheduling mechanism introduced to distinguish the arbitrary loss of the data packet and transmission delay. The aim of the proposal is to achieve a layered approach in the tasks involved in audio and visual data routing. The expected layered approach separates the routing functionalities such as scheduling and buffering process. Simulation studies are performed to demonstrate the efficiency of the proposed model, in contrast with other known model.

Index Terms - Quality of Service (QoS), Cellular IP, Multimedia applications.

1. INTRODUCTION

Real-time multimedia systems are suitably more and more popular in a range of applications. The wide range of applications includes group collaboration, remote medical diagnosis/treatment, conferencing systems, on-demand video services, and distance/remote sensing and monitoring [1]. Recently, there has been an increasing demand for a huge variety of multimedia over wireless networks. Applications such as Internet Cellular Phone and access to real-time data through the web need that the network offers quality of service to moving users. There has been unusual demand for IP- based wireless networks in which all traffic is distributed in packet-based form. Mobility of hosts has an important impact on QoS parameters for real-time application. The internet is rising exponentially; every day millions of new users are associated and the internet becomes the spine to provide a variety of data services. The 4th wireless communications generation proposes a fully IP network with packet switching so that traffic processing could be made easy. It is clear that the Internet is integrating a Global Mobile system and different classes of systems into a big standard IP network [11]. There are a lot of benefits achieved by rising mobile IP-based networks:

- Compatible with, and autonomy from, radio access technology. This means that we can acquire rid of the lock-in among the core network protocol and the link layer, the radio protocol.
- Reliability is achieved with modest bandwidth.
- The capacity to build global mobile networks.
- IP networks allow the user to control the quality of Multimedia applications.
- They can be cost-effective suitable for the use of relatively inexpensive existing equipment.

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Based on the above reasons, we choose mobile IP to be our technology to take multimedia applications. Mobile Internet Protocol (MIP), an Internet protocol intended to support the mobility of a user, provides the capability for hosts to stay associated despite of their location. MIP tracks a mobile host's long-term IP address. Since MIP has problems and problem such as like overhead in terms of improved delay, packet loss and signaling, we chose Cellular IP (CIP). CIP is a micro-mobility protocol proposal built-up by researchers from Columbia University and Ericsson as a solution to the performance and scalability issues of MIP [5].

Multimedia applications have very unusual necessities from applications for which the internet was initially designed. A primary issue is the require for performance assurance. The Datagram model, on which the Internet is based, has little resource management ability inside the network, and hence, cannot provide resource guarantees to users. Another issue is the need for separation. Because an IP network treats all packets the same way, it can offer only one service. Multimedia applications are real-time applications that need a certain amount of bandwidth to make sure the bit-rate wanted by each media stream and harsh delay difference wants to keep away from buffer underflow at the receivers.

In addition to the necessities of the multimedia applications, there are other properties linked with wireless networks. Wireless channels have exclusive characteristics not found in wire line channels, namely limited bandwidth, bursty channel errors and location- dependent channel errors. Also, the fact that a mobile node can potentially change its point of attachment to the network a lot of times during a session makes multimedia application over wireless networks more challenging. To address these concerns, we require a mechanism to monitor the network non-intrusively to notice any change. Moreover, for supporting multimedia applications, we have a specific means for supporting mixed quality of service (QoS) requirements. In Section

2, We briefly discuss the earlier related work. The proposed model is introduced in Section 3. The replication environment and the results are reported in Section 4. Concluding remarks are included in Section 5.

2. RELATED WORK

Many researchers have addressed the issue of supporting QoS for mobile IP by using scheduling algorithms. Simple scheduling algorithms such as First- Come-First-Served (FCFS) were first recommended. In [10], the early deadline-first (EDF) algorithm is extensive for scheduling real-time traffic in an IP-based cellular network. In each time slot, the scheduler calculates the priority for each packet and schedules packets in decreasing order. This approach does not include any mechanism for distinguishing traffic and it is relatively costly since the priority most be calculated fore each arriving packet.

In the Token Bank Leaky Bucket algorithm (TBLB), admission control and device policing have been used for servicing real-time multimedia traffic streams in packet-switched networks [6]. The drawback of this approach is that the typification overhead rise linearly with the number of flows present in the network.

Fair queuing is another approach that has been used for real-time traffic [3]. Increasing the weight for specific classes may usually result in better performance with respect to delay. However, it is a complex task to find suitable values for the weight, even in an ideal generalized processor sharing (GPS) scheduler.

Other recent trends of achieving QoS scalability are, Li and Chong in recent times proposed clustered multihop cellular networks (cMCNs) and considered a fixed channel assignment (FCA) scheme in [15], and a dynamic channel assignment (DCA) scheme in [14] for cMCNs. The key idea behind cMCN is to relate the MANET clustering in SCNs so that we can attain the characteristics of micro-cell/macro-cell hierarchically overlaid structure. The BSs in traditional SCNs will cover the whole macro-cell with a radius of rM. In the projected cMCN architecture [15], the original macrocell area is divided into seven micro-cells (or clusters) with a radius of rm; with one center micro-cell and six surrounding virtual micro-cells. The transmission range of the BS and MS is reduced from rM to rm in order to increase the spectral efficiency. The proposed cMCN uses a DIP acting as a cluster head in the center of each virtual micro-cell. The DIPs are installed to extend the coverage of the BS in terms of control information exchange and coordinate the peer-to-peer transmissions. In a gist the cMCNs and DCA can be concluded as these are an effective solution to achieve QoS scalability, but these solutions are highly demanding the infrastructure refurbishing in this process.

Recently, it seems to have been agreed upon that per- class architectures will be a viable solution for providing service guarantees in the Internet. Compared with per-flow architectures, per-class architectures they the benefit of working with simpler algorithms for implementing QoS guarantees than, and hence, they can be deployed with minor changes to the network architecture.

3. LAYERED TRANSMISSION AND PRIORITY RANKING OF SERVICES

A cellular network is one access network formed by a set of Base Stations (BSs) through which mobile hosts get wireless service and the gateway routers are used to connect this access network with the internet. In an access network the mobile host recognized by its actual address and the Care-Of Address remains same in local mobility controlling, hence cellular networks are away from the overhead of location updates. Under QoS factors considerations, we aimed to materialize the layered approach to transmit multimedia data. The proposed layered approach buffers the multimedia data by classifying that data into different classes based on the data properties and for each class a different buffer layer will be allotted.

The scheduling strategy of gateway routers fixes the order of egress transmission packets, in contrast the buffering strategy of gateway routers fixes the order of ingress transmission packets. Until very recently, scheduling and buffer management were held separately, even though both mechanisms address the issue of managing.

The model we use relies on using adaptive realtime scheduling method that take into account the characteristics of the wireless channel as well as the QoS requirements for each traffic stream. There are different parameters we consider in developing this model:

Packet classification: Each class of traffic is connected to the relevant layer of the buffer that buffers the packets in the form of FIFO. The packet that streams into a network through a gateway router will be moved to the relevant buffer layer that selected from a class to which that packet belongs to. The packet classification process not aware of the stream to which that ingress packet belongs to.

Class level ingress Buffering: In layered buffering each layer is having a finite size, which can be differ from the sizes of other layers, but the total size of all layers together must be equal to the actual buffer.

Class level egress schedule: Upon an ingress packet moved to its class level buffer layer the service priority ranks will be normalized according to the load at individual class level buffer layers in order to achieve the service agreement. In this regard, often packets can be discarded from some of buffer layers.

As a matter of generalizing, enhanced delay tolerance and utmost packet drop possibility and ability of detecting fair changes in target network can be considered as QoS aspects for multimedia data transmission.

Service priority ranking under service scheduler must be done under QoS aspects. This process can be elaborated with an example that follows

Let us consider multimedia traffic at a gateway router. Here the ingress is classified into three classes and the same number of buffer layers are used to store this three class of ingress. If the classes are being prioritized in the same order such that class one get top priority, class 2 is in second order and class 3 follows class 1 and class 2 with rank 3. Then few QoS aspects can be defined as follows :

The delay in class 2 data must be double to the delay at class 1 data: This QoS aspect helps to achieve the better throughput at less packet overhead. This QoS aspect can be generalized as delay in a class A data must be double to the delay in a class B data, if class A rank follows class B rank in order.

The packet loss in class 3 must be equal to packet loss in class 2: This QoS aspect helps to achieve the better bandwidth utilization with fewer packet overhead. This QoS aspect can be generalized such that the packet loss ratio of two classes that are sequenced in given priority ranks must be same.

The enhanced delay of packets at class 3 must be less than the max delay threshold: This QoS aspect fixes the finite state for delay enhanced by a buffer layer during the transmission, hence the packets that can't survive beyond the max delay time can be dropped from the buffer to accommodate for other capable packets in the stream. This QOS aspect helps to achieve the effective buffer management. This QoS aspect can be generalized such that the max delay threshold of the class that stands lost in the priority rank order must set to finite state.

Because we desire to maintain complete guarantees and do not use admission control, a set of service guarantees may be infeasible at certain times. For example, it may be impossible to meet both a delay bound and a loss rate bounds at the time of a burst of traffic. In case the system of service guarantees is infeasible, some guarantees may need to be relaxed. For instance, proportional guarantees may be relaxed in favor of absolute bounds, or loss guarantees may be relaxed in favor of delay guarantees. We assume that all QoS guarantees are given a precedence order, which is used to determine which constraints are relaxed in case of an infeasible system. There are two requirements that a relative separation model should meet:

Controllability: the network operator should be able to regulate the class spacing between classes based on their criteria.

Predictability: the separation is consistent (a higher class is better or at least no worse than a lower class) and the relative ordering between classes should be met autonomous of load condition and time scales.

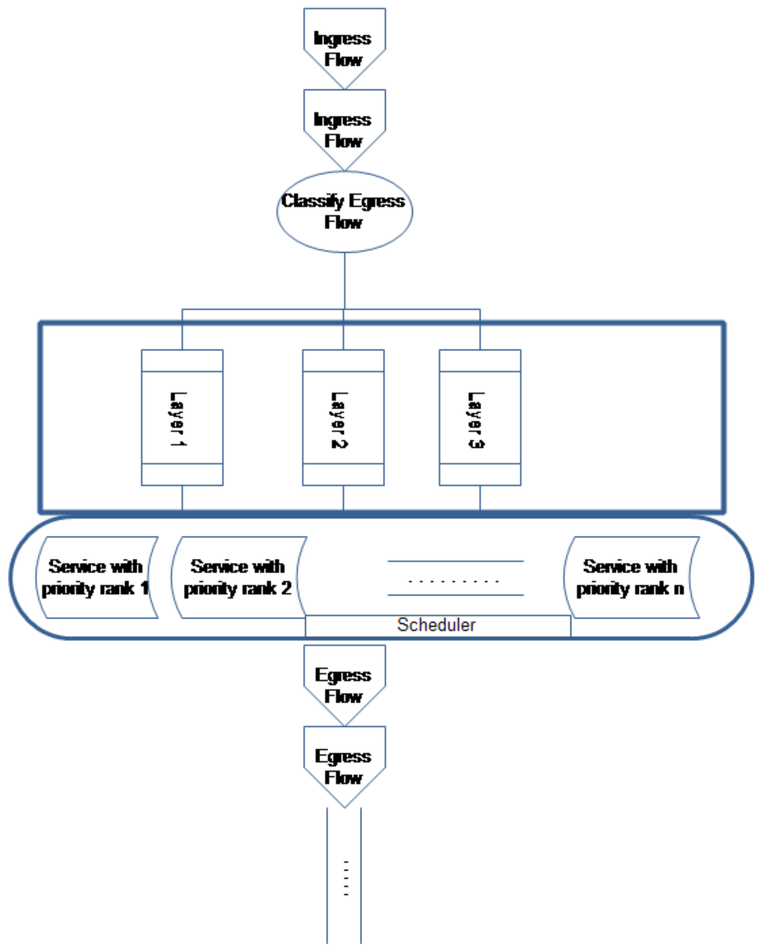

Figure 1: Architecture of layered transmission in wireless cellular networks

The scheduling algorithm operates as follows. For each arrival, the service rate allocation of traffic classes is modified so that all QoS service guarantees are met. If there exists no possible rate allocation that meets all service guarantees, traffic is dropped, either from a new arrival or from the current backlog.

In the model we propose, the number of the queues is identical to the number of the separation classes. The state history is maintained at a classification stage, not on a flow state as with others future separation algorithms. Thus, the state record of a class then is to consider to the transmission of a packet from a class or falling of a packet from a class. For class priority, our scheduler will grantee class C_i will obtain improved or at least no worse service than class C_{i+1}.

4. SIMULATION RESULTS AND ANALYSIS

We present an estimate of the model discussed in section (3) using NS-2 network simulator. Our aims are (1) to conclude if and how well the preferred QoS is can be achieved by differentiation; (2) that the scheduling algorithm can reach controllable and predictable interruption and packet loss separation; and (3) to compare our algorithm with existing models.

We test the algorithm for combining TCP and UDP traffic; we also want to study the level of quality of service and end-to-end flow that can be received with the proposed per-node guarantees. The relationship between the CIP nodes is 45 Mbps, and sources relate to the Gateway by 100 Mbps links. Each 45 Mbps link has a propagation delay of 3 ms; each 100 Mbps association has a propagation delay of 1 ms. Class 1, 2 and 3 only consist of TCP traffic, and Class 4 only consists of UDP traffic. All flows consist of packets with a fixed range of 500 Bytes, and the experiment lasts 70 seconds of simulated time. The offered load is asymmetric, since primarily Class 1 contributes 10% of the aggregate traffic, Class 2 contributes 20%, Class 3 contributes 30% and Class 4 contributes 40%. The complete delay and packet loss constraints for class 1 are 2ms and 1% respectively. The relative separation factors are set of four for the ratio of delays of two successive classes, and of four for the ratio of failure rates of two consecutive classes. We compared the performance of buffering approach along with standard service scheduling currently in use and layered buffering and dynamic service scheduling with priority ranks that we proposed.

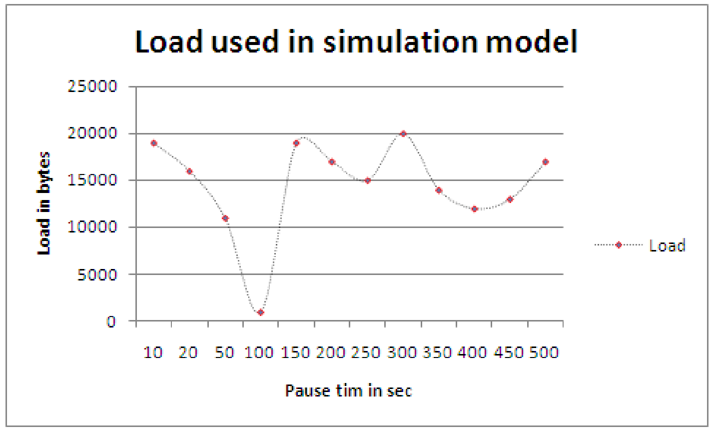

Fig 2: weight in bytes sent by the source node of the path [in regular interval of 10 Sec]

The same load is agreeing to all the paths with a regular period of 10 Sec. Consignment given in kilobytes is shown in fig 2. The fig 3 concludes the step up of proposed QoS centric layered buffering and dynamic service scheduling approach over buffering and service scheduling model such as EDF[10] in use. The packet delivery ratio evaluation between proposed "layer buffering with dynamic service scheduling" and existing model EDF[10] that is a single buffer strategy with standard service scheduling is compared in fig 4 that elevate the performance scalability as minimum packet overhead in the layered buffering and dynamic service scheduling.

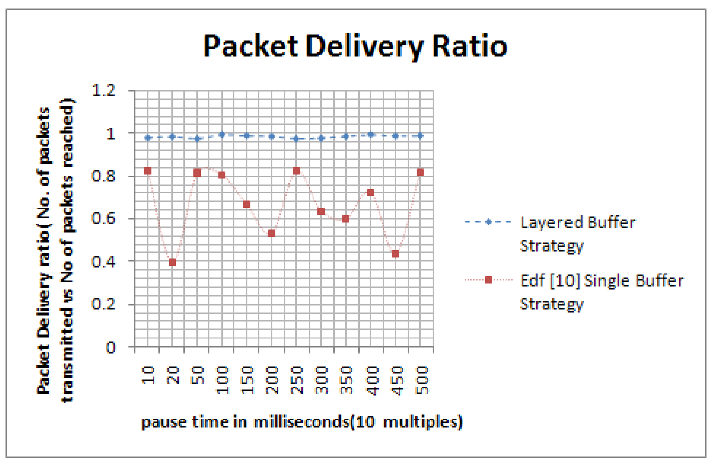

Fig 3: Packet delivery ratio: between layer buffering with dynamic service scheduling and EDF[10] single buffer strategy with standard service scheduling.

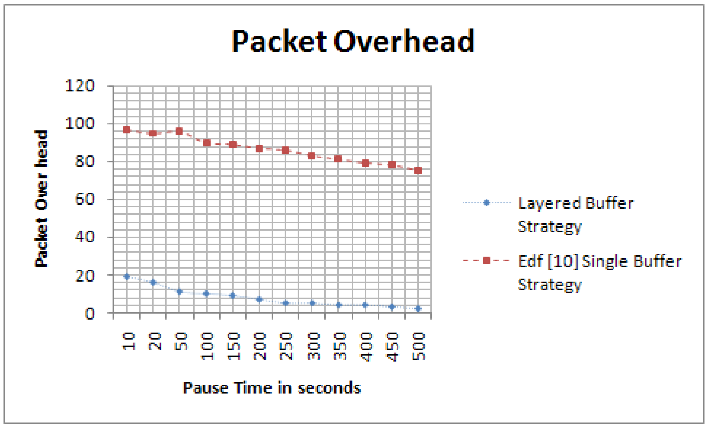

Fig 4: Packet overhead comparison between layer buffering with dynamic service scheduling and EDF[10] single buffer strategy with standard service scheduling.

5. CONCLUSIONS

In this paper, we proposed a QoS centric layered transmission of multimedia data in wireless cellular networks. We estimate the performance of the proposed model during wide simulation study with a traffic of audiovisual data combination. Our simulation study has revealed that the proposed topology attain considerable performance under QoS factors. The proposed layered buffering and dynamic service scheduling approach is extremely flexible in that it can offer a large range of QoS aspects. In future this work can be refined such that the proposed model is qualitative in mathematically prioritizing and scheduling services.

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