A Modern Approach for Improving Quality of Service in 4g Wireless Networks

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Abstract

The aim of this paper is to enhance the Quality of Service (QoS) in 4Gwireless networks. The various QoS factors in wireless communications are Bit Error Rate, access delay, Throughput and network life time. A new QoS factor namely User Satisfaction Factor (USF) has been proposed which is a function of both the number of received bits and the delay sensitivity. This USF not only predicts the final delivered QoS, but also takes advantages of the fact that different packets can be decoded at different time in the receivers. Based on this USF, three types of scheduling schemes are considered such as the maximum approach, the overall performance approach, and the proportional approach. These scheduling schemes achieve different tradeoffs between system performance and individual fairness.

Keywords: *Quality of Service, Scheduling Schemes, User Satisfaction Factor, 4G*

1.0 INTRODUCTION

Fourth-Generation (4G) is a term used to describe the next complete evolution in wireless communications. A 4G system will be able to provide a comprehensive IP solution where voice, data and streamed multimedia can be given to users on an "Anytime, Anywhere" basis, and at higher data rates than previous generations. The fourth generation is the total replacement of the current 3G networks. The objectives of 4G include: that 4G will be a fully IP-based integrated system. 4G will be capable of providing between 100 Mbit/s and 1 Gbit/s speeds both indoors and outdoors, with premium quality and high security. The infrastructure and the terminals of 4G will have almost all the standards from 2G to 4G implemented. Although legacy systems are in place to adopt existing users, the infrastructure for 4G will be only packet-based (all-IP). Some proposals suggest having an open platform where the new innovations and evolutions can fit. The technologies which are being considered as pre-4G are the following: Flash-OFDM, WiMax, WiBro, iBurst, 3GPP Long Term Evolution and 3GPP2 Ultra Mobile Broadband. Existing cellular networks, based on circuit switching, consist of base stations (or base transceiver stations), base station controllers, switching centers, gateways, and so on. The base station (BS) plays the role of physical transmission with fast power control and wireless scheduling. The base station controller (BSC) executes the largest part of the radio resource management. Whenever a mobile terminal (MT) moves into another cell, it requires handoff to another base station. In contrast, the 4G network has a simple structure where each BS must function intelligently to perform radio resource management as well as physical transmission. This makes the BS act the role of an access router (AR). This architecture is shown in Fig. 1. It incurs high overhead, however, especially when an MT configures mobile IP (MIP) addresses for handoff. In addition, the 4G network is expected to have a small cell radius due to its use of high frequency band, which possibly results in short cell residence time [1].

Providing multimedia services involving digital audio and video with QoS guarantee in such an environment presents more challenges due to the limited bandwidth resource and highly variable environment and user mobility.

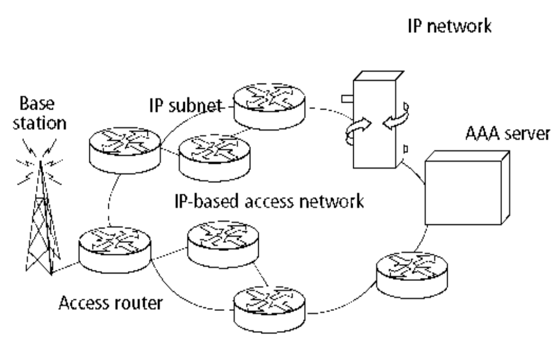

Fig.1: The pure all-IP 4G network

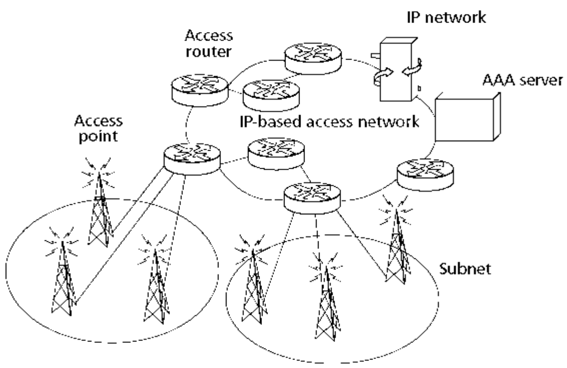

Fig.2: The subnet-based 4G network

To solve the fundamental problem of all-IP cellular networks, we can separate the functionality of an Access Router (AR) from that of an access point (AP) so that each undertakes L3 and L2 protocols, respectively. Figure 2 shows an example of a simple network where an AR manages several AP. This relation is similar to that between base station controller (BSC) and base station (BS) in existing cellular networks. A subnet consisting of an AR and several AP can be configured by Gigabit Ethernet. Then, an MT moving within the subnet (i.e., changing AP) performs L2 handoff without changing MIP attachment. When the MT moves into another AR area, it experiences L3 handoff.

A main difference is that the former is decentralized, and the latter is centralized. Since the pure all-IP network incurs L3 protocol in the end access link, it requires long handoff latency and high signaling overhead. However, the architecture is simple and cost-efficient for implementation. On the other hand, the subnetbased all-IP network implements hierarchical architecture, so it is possible to do efficient resource management in spite of its inflexibility.

With the increasing demand and the penetration of wireless services, the emerging innovations in radio technology and the evolution toward wireless/wire line integration uses of wireless network now except QoS and performance. The QoS in wireless network is considered at two levels, i.e. the application level and connection level.Application level QoS is related to perceived quality at the user end and I commonly considered in packet-switched networks. A set of parameters such as delay/delay jitter, error/loss & throughput etc are used to describe application level QoS [1]. Since packet switched networks take advantage of higher degree of multiplexing among services, packets of certain service flow may experience varying delay, delay jitter and loss. Efficient packet access protocols and scheduling schemes play key role in solving these QoS problems. Connection level QoS is related to connection establishment and management.

The QoS can be measured by means of certain parameters like bandwidth, delay, jitter, throughput, reliability and cost. QoS policies for wired networks are not directly applicable to wireless networks as the fundamental characteristics of the underlying link layers differ significantly and some layers of the IP stack are also affected However, in a wireless environment packet loss is mainly due to bit errors on the wireless link and not congestion. Therefore, if congestion control mechanisms are invoked when there is no congestion, it will lead to a considerable drop in throughput. Hence, the issue with TCP in a mobile environment is that congestion control mechanisms are invoked when they are not required. Mobile nodes get frequently disconnected during handoffs and some packets are lost as a result, this is another instance when the congestion control mechanisms are invoked. there are some mechanisms that were designed for wired networks but are being adapted for providing QoS in wireless. The most common examples are the IntServ (Integrated Services) and DiffServ (Differentiated Services) mechanisms. However, both IntServ and DiffServ are not directly applicable to wireless networks. Certain characteristics of wireless networks like limited bandwidth, lossy wireless medium and mobility have to be considered before these techniques can be adapted for use in wireless networks [2].

Quality of Service (QoS) and multimedia support are critical to wireless home networks where voice, video and audio will be delivered. It has become very significant to enhance QoS capabilities of real-time data transmission in the service provider networks.

With the growing demand for heterogeneous applications in wireless networks, it is critical to deliver flexible, variable rate services with high spectral efficiencies to the end users. Different Quality of Service (QoS) provisioning over time varying channel conditions is a key challenge for system design.

Resource allocation provides a strategic means to address this challenge. Current wireless systems use Signal to Noise Ratio (SNR) as a QoS measure. However with the increasing demand for multimedia transmissions, their delay sensitive QoS cannot be satisfied by the SNR measure itself. So it is necessary to define new QoS measures for multimedia transmission services. Opportunistic scheduling has attracted a lot of research attentions as a special kind of resource allocation over time. The basic idea of “opportunistic scheduling” is to allocate resources to links experiencing good channel conditions while avoiding allocating resources to links experiencing bad channel conditions, thus efficiently utilizing radio resources. This is also referred as channel-aware scheduling which explores time/multi- user diversity. On the other hand, opportunistic scheduling introduces an important tradeoff between system performance and fairness among users. In this paper our goals are to define a new QoS measure for multimedia transmissions and to develop new scheduling scheme, so as to improve the system performance and maintain the individual fairness among users.

The QoS measure, User Satisfaction Factor (USF) for different delay constraints and different applications such as voice, video, and data is proposed. The proposed USF is a function of both the number of received bits and the delay sensitivity profiles. Since some layered decoders can decode the layers such as base layer earlier than the other layers, the USF measure can take this advantage for better resource allocation.

Moreover, the USF predicts the final delivered QoS during the transmission. Then based on this QoS measure, we develop four types of scheduling schemes, namely the Maximin approach, the overall performance approach, and the proportional approach [3].

The Admission control algorithm is working in such a way that based on the available network resources, it estimates the impact of adding or dropping a new session request. In both 2G and 3G systems, admission control refers to a single network. As we are moving towards heterogeneous wireless networks referred to as systems beyond 3G or 4G, admission control will need to deal with many heterogeneous networks and admit new sessions to a network that is most appropriate to supply the requested QoS[4].

One of the most important radio resource management (RRM) mechanisms used in wireless networks is admission control (AC). AC algorithms are utilized to ensure that admittance of a new flow into a resource constrained network does not violate the service level agreements guaranteed by the network to already admitted flows. The goal of an efficient admission control algorithm is to ensure the quality of service of the ongoing connections, while at the same time, to care for the optimal utilization of the available radio spectrum. Admission control schemes are the decision making part of the networks aiming at providing users with services of guaranteed quality, something that leads also to reduced network congestion and call dropping probabilities and thus to more efficient resource utilization [5].

2.0 ADMISSION CONTROL IN THE CONTEXT OF FUTURE HETERO -GENEOUS WIRELESS SYSTEMS

In order to ensure a high degree of satisfaction of user QoS requirements, a form of cooperation between WINNER and legacy RANs is required. A universal admission control mechanism across all networks that operate in the same area is a key element in achieving their efficient cooperation. The main goal of the AC algorithm is to control the admission of new or handover sessions while maintaining the load of the network within some boundaries that do not disturb the QoS of any other sessions. The main function of an efficient AC algorithm for heterogeneous networks is to decide at a specific point in time if there is a network that has the available resources to serve (to satisfy the QoS requirements of) a new session [6]. The design of the AC algorithm must be made very carefully to minimize the following: False rejections, false admissions.

3.0 USER SATISFACTION FACTOR (USF)

There is a need to define USF for QoS measurement such that the system can adapt its resource allocation strategy by predicting the final delivered QoS. Suppose there is an estimator g that estimates the number of transmitted bits at time n' , based on the transmission history and channel conditions using the information no later than time n. The most straight-forward method is the linear prediction. At the current time n, the resulting prediction of the number of bits received at a future time n' is

$$g(n, n') = \frac{n' \sum_{j=1}^{n-1} T_i(j)}{n-1}, \; n' \geq N,$$

where T*i* (*j*) is the *i*th user’s transmitted bits at time j. Other types of function g can also be explored, such as moving window, Kalman filter, etc.We define USF at time n as:

$$USF_i(n) = \sum_{n'=N}^{\infty} P(n') f(g(n, n'))$$

3.1 Four Scheduling Schemes with Usf

In this paper, based on USF, I develop four types of scheduling schemes to improve both individual fairness and system overall performance. Then, to demonstrate the performance of the proposed scheduling schemes, a simple analysis regarding how different parameters affect the value of the USF change is given.

3.1.1 Scheduling Scheme 1: Maximum Approach:

The first scheduler selects the user with the minimal USF for transmissions so as to improve the minimal performance of the system.
The scheduling policy is expressed as:

$$\arg\min_{n} w_i.USF_i(n)$$

where w_i is the weight value for user i . In this paper, we assume $w_i = 1, \forall i$ .This approach is fair among users but will generate inferior system performance because the users in the bad channel conditions might consume most of scheduled transmission times.

3.1.2 Scheduling Scheme 2: Overall Performance Approach:

This type of scheduler tries to maximize the overall USF. Define $\Delta USF_i(n)$ as the USF change if the scheduler assigns the current time slot to user i . Since each time only one user occupies the channel, the scheduling policy is

$$\arg\max_{n} w_i.\ \Delta USF_i(n)$$

By doing this, a user with a good channel condition has a good chance for transmissions. Consequently the scheduler can take advantages of multi-user diversity and channel diversity to optimize the system performance. However the fairness among users is not considered. Notice that this type of scheduling scheme is very unfair for the users with no delay constraint, since ΔUSF is always a small number for them. In implementation, when the other users have USF's larger than 1, the users with no delay constraint are selected.

3.1.3 Scheduling Scheme 4: Proportion Approach:

In [4], the proportional fair scheduling is proposed, where the scheduler selects the user with the largest ratio of current possible maximal rate over the sum of rates in his/her transmission history. By doing this, the scheduler takes into account of the current channel condition and maintains the long-term proportional fairness. In this paper, we employ the similar idea. Compared with the second approach, the scheduler weights the user with the inverse of his/her current USF value plus a small positive constant ε . This constant is to prevent the error of "divided by zero". If the user has a better channel, the rate adaptation can transmit more information to improve his/her USF and the user has the advantage to transmit. On the other hand, if the USF is small, the user also has the advantage for transmissions. The scheduler can be expressed as:

$$\arg\max_{i} \frac{w_i.\Delta USF_i(n)}{USF_i(n)+\varepsilon}$$

This scheduler provides another tradeoff between the performance and fairness.
By using some assumptions to simplify the problem, there is a provision of some theoretical analysis on how different parameters influence the value of ΔUSF. By using Rayleigh fading assumption as an example, the channel of any link is modeled as a zero mean circularly symmetric complex Gaussian random variable with unit variance. Assume a linear assumption of f as $f = \mu g(n, n^1)$ and $\alpha > 0$. The larger μ, the higher demand for transmission rate.

The estimated ΔUSFi is large, when the demand of transmission rate μ is large, the delay requirement is strict (small α), or the channel gain Gi is large. On the other hand, when the current USF is large, the user has less advantage for ΔUSFi. Notice that the above equation is obtained based on several assumptions such as linear assumption of f and Rayleigh Fading. In practice, it is very difficulty to obtain the close form solution of ΔUSFi. In the next section, we will conduct numerical study to evaluate the proposed schemes.
In [3], the proposed four schemes give more proportions of time to the first type of users. Consequently, the first types of users have higher throughput and higher USF than those of the second type of users. This is because the proposed scheduling schemes take consideration of the strict short term delay requirement of the first type of users. In addition, for data users, USF is always equal to 1 for all schedulers, because there is almost no delay requirement. Scheme1has the least differences of USF among type 1 and type 2 payloads. This is because scheme 1 always tries to improve the performance of user with the least value of USF. However this scheme has the lowest throughput. Scheme 2 has the highest overall USF and the overall rough put is also high. However the difference of USF for two types is large. This is because scheme 2 only tries to maximize the overall USF without considering the fairness among users. Scheme 3 and scheme 4 provide certain tradeoffs between the

system performance and individual fairness. Scheme 3finds the tradeoff closer to the user fairness while scheme 4 finds the tradeoff more close to the system performance.

4.0 CONCLUSION

In this paper, we define a new QoS measurement for Multimedia and data transmissions considering heterogeneous and delay sensitive applications. Based on this measurement, four scheduling schemes are proposed. From the simulation results, compared with the weighted round-robin and modified proportional fair schemes, the proposed schemes find the different tradeoffs between individual fairness and system Performance, while the heterogeneous nature and delay sensitivity of payloads are considered. The schemes can be further generalized to facilitate the design of the future wireless networks.

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