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Priority Based Call Admission Control Protocol for Videoconference Traffic in Wireless/Cellular Networks

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Abstract

In this paper, we propose a new CAC protocol which provides videoconference traffic taking multiple services like voice, text and multimedia at a time for multiclass users. In this proposed protocol we use priority on basis of account balance of customers and serve the calls with higher priority first. The higher the account balance higher is the priority. The call continues until the user closes the call or the balance is zero or the user is taking considerable amount of resources causing starvation. Video packet delay requirements are strict because delays are annoying to a viewer. An adaptive scheduling scheme to allocate optimum rate for each traffic queue is proposed to minimize the scheduling delay. By simulation experiments, we show that the proposed protocol achieves optimum rate with reduced delay, maximum use of bandwidth and maximum Quality of Service (QoS).

Keywords: CAC; Bandwidth utilization; Call waiting time; QoS

1. Introduction

Call Admission Control (CAC) protocol limits the number of calls allowed through the cell. CAC protocol deny new calls attempted if the cell is getting overloaded. If the cell is being overloaded it is better to deny further calls and thus maintain quality of the ones currently in progress, rather than allowing too many calls that would overload the cell and reduce quality of all calls in progress. The decision to admit or reject a call is made by CAC protocol. New calls are allowed only if there is enough bandwidth and processing power left to be able to handle the resulting media streams effectively.
Buffer provisioning for new and /or handoff calls can reduce the blocking probability of new and/or handoff call attempts which is shown in paper [1]. In order to provide enough bandwidth to accommodate broadband services to multiple mobile users the size of the wireless cells is decreasing toward the pico cell architecture [2]. In this environment, due to user mobility the traffic conditions in the cells can change very quickly. Also, when mobile users change their point of attachment (handoff), the end-to-end path may be changed, whereas they still expect to receive the same QoS. An efficient CAC mechanism should be able to cope with this strict user requirement. Another method involves the regulation of calls according to defined characteristics such as priority descriptors [3]. In First Come First Serve (FCFS)

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scheme if a request arrives and if there is enough bandwidth to accommodate it, the call is admitted, otherwise it is rejected. FCFS produces a good utilization of the channel with requirements of high bandwidth. FCFS does not support priority. The paper [4] analyze a class of partitioning and threshold based admission control algorithms that make acceptance/rejection decision not to satisfy QoS requirements but also to optimize the revenue of the system taking into account prices and arrival/departure information of service calls. It is a ‘charge-by- time pricing scheme. In our proposed protocol we use priority on basis of account balance of customers and serve the calls with higher priority first. We keep a limited flavor of FCFS to avoid starvation for a low priority call taking from a primary buffer.

2. Proposed Call Admission Control Protocol

In this protocol new calls are taken dynamically and store them in a primary buffer. If the primary buffer is full then the system will not accept any new call. It has to be blocked for lack of availability of resources. In the mean time a thread is monitoring whether the secondary buffer is empty or not, if empty then it takes the calls from the primary buffer, stores them and sorts them according to their priority. The higher the account balance higher is the priority.

The account balance amount of the customer decreases at a pre-defined rate after a fixed interval of time. While a certain customer is being served, if it utilize the resource for a considerable amount of time then it has to be dropped for the sake of not jeopardizing the services of other calls waiting for a long time. This validation helps to prevent starvation. When the balance of a certain customer becomes zero then his/her service has to be discontinued. Monitoring is needed for the remaining balance of the customer.

This protocol is based on parallel execution of videoconference traffic [5] (i.e voice, text and multimedia) to improve the utilization of channel. Here three types of services (i.e voice, text and multimedia) are served synchronously. This protocol implements a call waiting threshold which indicates the maximum allowable limit to a requesting call in the buffer until either the admission request is granted or the call is dropped. In order to do this extra intelligence to monitor the rates and number of packet arrival to all the queues is needed. The system should then be able to predict future arrivals based on the previous data obtained. This system should also be able to determine the optimal queue size needed as increasing the size too much would lead to an increase in the mean waiting time of packets. We use multithreaded environment for the implementation of this protocol. Text, voice, multimedia calls are treated as separate thread. All the three thread of a given customer needs to run simultaneously. We implement it with a single CPU so we need to synchronize the execution of the threads. The delay caused can be neglected. The block diagram of the proposed protocol is shown in Figure 1.

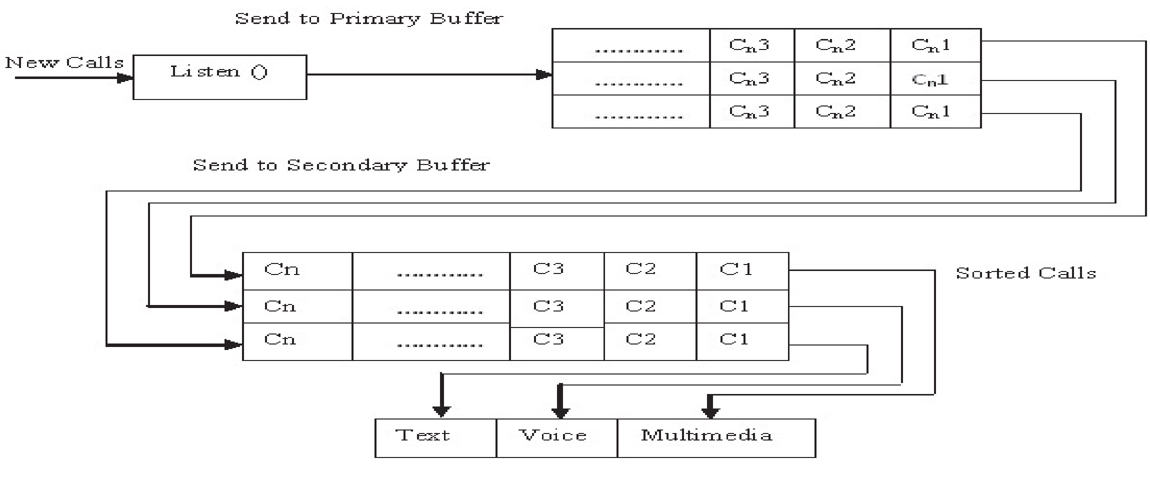

Figure 1. Block Diagram of Proposed Protocol

For each customer and for each type of call a separate class will be created containing customer name, account balance, call threshold time. 1st thread provides the service of a specific customer. 2nd thread is used for counting the primary and secondary buffer wait time for that specific call. 3rd thread is used for generation of random calls and send them to the primary buffer. 4th thread monitors whether the secondary buffer is empty or not. If empty, fills secondary buffer with the calls from primary buffer.

3. Numerical Results

Table 1: Sample Output for 1st set of Calls

Results for calls	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th
Initial Balance	350	220	250	280	150	99	171	310	300	9
Bill	198	159	228	144	27	99	171	300	213	9
Net Balance	152	61	22	136	123	0	0	10	87	0
Primary buffer wait time (ms)										
	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th
Text	1	0	1908	4032	1204	10782	7396	4193	2326	22408
voice	1	1	1914	4087	1205	10746	7433	4225	2298	22522
Multimedia	1	1	1909	4032	1205	10782	7398	4191	2326	22410
Secondary buffer wait time (ms)										
Text	0	4788	5297	7605	12461	24041	18187	991	11080	3339
voice	0	4780	5351	7626	12404	23934	18304	992	10997	3340
Multimedia	1	4784	5297	7604	12459	24038	18197	991	11080	3337
Call arrival time (ms)										
	0	3561	9177	15387	21068	26729	33279	39544	43259	50793

Table 2: Sample Output for 2nd set of Calls

Results for calls	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th
Initial Balance	350	220	250	280	150	99	171	310	300	9
Bill	198	159	228	144	27	99	171	300	213	9
Net Balance	152	61	22	136	123	0	0	10	87	0
Primary buffer wait time (ms)										
	1st	2nd	3rd	4th	5th	6th	7th	8th	9th	10th
Text	1	0	2279	4517	2645	11471	7784	4588	1195	23415
voice	1	1	2280	4345	2633	11453	7800	4590	1193	23337
Multimedia	1	1	2281	4515	2648	11470	7784	4588	1194	23414
Secondary buffer wait time (ms)										
Text	0	4533	5203	7605	12425	24018	18189	991	11007	3335
voice	0	4524	5201	7620	12406	23929	18253	996	10995	3335
Multimedia	1	4531	5201	7604	12423	24017	18189	991	11007	3334
Call arrival time (ms)										
	0	7887	12446	18255	25860	33249	40432	46740	53377	56895

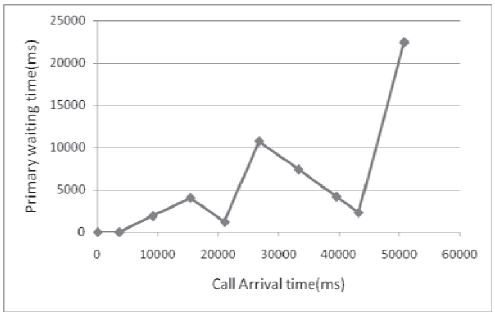

Figure 2. Primary wait time vs Call arrival time of each call for Table 1

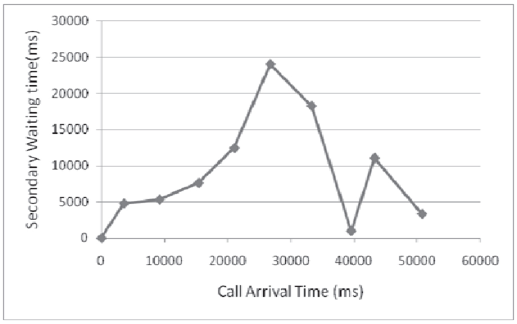

Figure 3. Secondary wait time vs Call arrival time of each call for Table 1

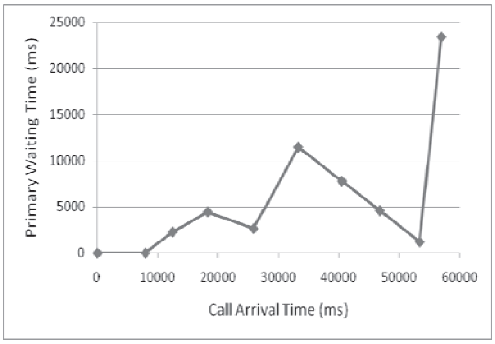

Figure 4. Primary wait time vs Call arrival time of each call for Table 2

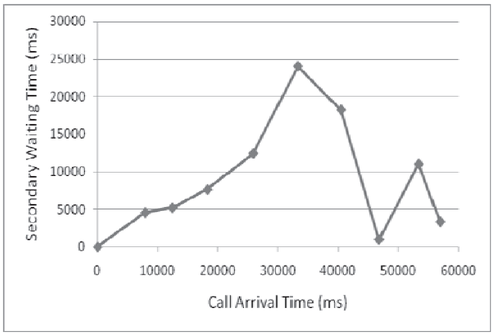

Figure 5. Secondary wait time vs Call arrival time of each call for Table 2

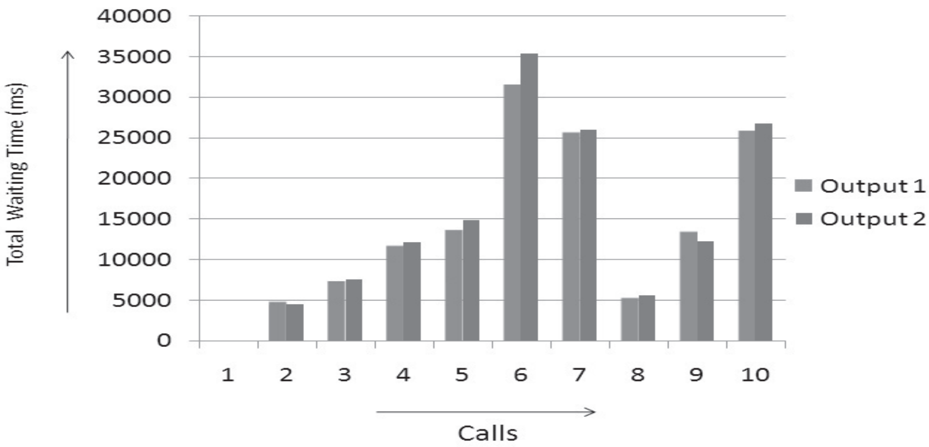

Figure 6. Comparison of total waiting time for the two given output

4. Discussions

The performance of our proposed protocol is based on computer simulation. The simulation were conducted with the use of the Java programming language in Windows, Linux, Mac, Solaris work stations. Also we use J2SE Software Development Kit (JDK*)* Standard Edition 1.5. The minimum required hardware setup to run this system is 256 MB RAM and Pentium IV/ hyper threading enabled Processor. The protocol uses swing components JScrollPane. Each simulation point is the result of ten independent runs (i.e ten customers) which are shown in table 1 and table 2. Each customer handles voice, text and multimedia calls synchronously. Each customer will be prioritized according to their account balance. It is clear from table 1 and table 2 that 1^{st} customer will get priority compared to 10^{th} customer due the maximum account balance of 1^{st} customer compared to 10^{th} customer. The primary wait time and secondary wait time will be less of those customers who hold more account balance compared to other customers which are shown in figures 2,3,4,5. In simulation the delay between calls (voice, text and multimedia) are so negligible that can be ignored easily. Here we have taken two sets of calls. Histogram which is shown in figure 6 gives the clear picture of comparison between two sets of calls. The account balance amount of the customers decrease at a pre-defined rate after a fixed interval of time.
This proposed CAC protocol is based on randomized algorithm of Monte Carlo type as it produces different output for same input..

5. Conclusions

The main problem we have faced while implementing is synchronization problem among threads. The concurrent activity may access and change common resources at the same time. However use of flags and 'synchronized' keyword in java helped to avoid this problem. Voice, text and multimedia these three types of calls execute synchronously. So this is a new efficient CAC protocol for multimedia traffic transmission over wireless cellular networks. The novelty of the protocol lies in the utilization of pre computed traffic scenarios combined with online simulation, for decision making on the acceptance or rejection of a new Videoconference call. The pre computation is based on the traffic parameters declared by the video source at call setup. We can only conclude that there is no single technique that could be regarded as the best. The best technique to minimize congestion depends on the situation. Here, there are two situations: the incoming traffic level and the rate at which packets arrive. To avoid congestions we take two buffers i.e primary and secondary buffer. This proposed protocol gives better performance for achieving optimum rate with reduced delay, maximum use of bandwidth and maximum Quality of Service (QoS) which are shown in simulation result.

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