Adaptive Admission Control Strategy for Multimedia Mobile Packet Networks

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***Abstract* – In this paper we propose a strategy for adaptive admission control in multimedia cellular networks. The proposed strategy uses adaptive adjustment of threshold values for voice and video traffic by monitoring the network performance, which is measured through performance parameters such as packet loss probability on a packet-level, as well as new call blocking and call dropping probability on a call-level basis.**

***Keywords* – Admission control, Mobile, Quality of Service, Video, Traffic**

I. Introduction

Mobile networks, due to their cellular structure, are characterized with different traffic characteristics than wired networks. The coverage area is divided in smaller areas called cells. In our analysis we consider packet-based traffic in the mobile network. In such case, during a single ongoing call a subscriber is allowed to handover from its current cell to another neighboring cell. In the handover process the user is releasing the certain amount of bandwidth in the old cell and occupies another same amount in the target cell.

Third generation (3G) mobile networks [1] and beyond (e.g. 4G), as well as Wireless LANs, have many times higher bandwidth than cellular networks in the past (e.g., up to 2 Mbps in 3G, and up to 54 Mbps in 802.11 family of WLANs). Higher bandwidth provides possibility for service providers to offer multimedia services (e.g., video streaming) besides voice service. In such case, we are facing new challenges considering dimensioning and efficient resource utilization in multimedia mobile networks. To gain from statistical multiplexing we should use the same wireless channels for voice and video traffic. However, there are different bandwidth and Quality of Service (QoS) requirements from each of these two traffic types [2, 3]. For example, a voice call demands smaller bandwidth compared to a video call. On the other side, voice is conversational bidirectional traffic which is very sensitive to delay, while video streaming is basically unidirectional and therefore can tolerate higher packet delays (they can be compensated by buffering at the receiving end).

In this paper we propose a novel strategy for adaptive call admission control in mobile multimedia networks, considering voice and video traffic, based on active monitoring of the network performances, defined through packet loss probability, call-dropping probability (P_d) and call-blocking probability (P_b).

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The paper is organized as follows. In the next section we define the network model that we use in simulations. Section III presents the principles of the network operation. Section IV gives the simulation results. Finally, Section V concludes the paper.

II. Network model

We created a simulation environment in Matlab. Here, we briefly report on our simulation model.

The coverage area of the mobile network is divided into hexagonal cells. Every cell is assigned a certain bandwidth, which presents the capacity of that cell for call handling. The subscribers can be served only by one cell at a moment.

Speed of each subscriber in mobile state is modeled with normal (Gaussian) distribution truncated at 0. Position of each subscriber is represented with 2 coordinates (x and y), and may be in two possible states: moving or stationary. Subscriber movement is defined in the following manner: in every step of the simulation, each subscriber is allowed to move in either x and y directions or in both. The length of the trajectory is dependent upon the subscriber speed.

In our work we consider two call types, voice calls, randomly generated by given traffic parameters, and video calls, intentionally implemented as downlink video streams in a monitored cell. Voice calls, and voice packets have higher priority, but only when voice traffic intensity is lower than a given threshold. One of the issues of this paper is the choice of a criterion for assigning that threshold, and the influence of its value over the network performance.

We model voice call arrival process with Poisson distribution [4]. Each traffic class is given call arrival rate λ (calls/hour/user). Furthermore, call duration time is modeled with exponential distribution, and we denote with t the mean call duration.

In a packet cellular network we may distinguish among three main types of losses:

- Blocking of a new voice call: it occurs when subscriber attempts to make a new call, but there are no free resources in the target cell, and the network rejects that call.
- Dropping of a voice call: it happens when there are no free resources in the target cell at handover events.
- Lost packets (for both voice and video flows): they occur when packets are lost due to full buffers. We assume that the wireless link is error-free.

We denote new call blocking probability and call dropping probability for voice calls with P_b and P_d, respectively. Furthermore, we use Pl_{VI} and Pl_{VO} to denote packet loss probability for video and voice, respectively.

In our network model, we use the policy “blocked calls cleared”, that is, an already blocked new call, dropped handover or lost packet is cleared from the system.

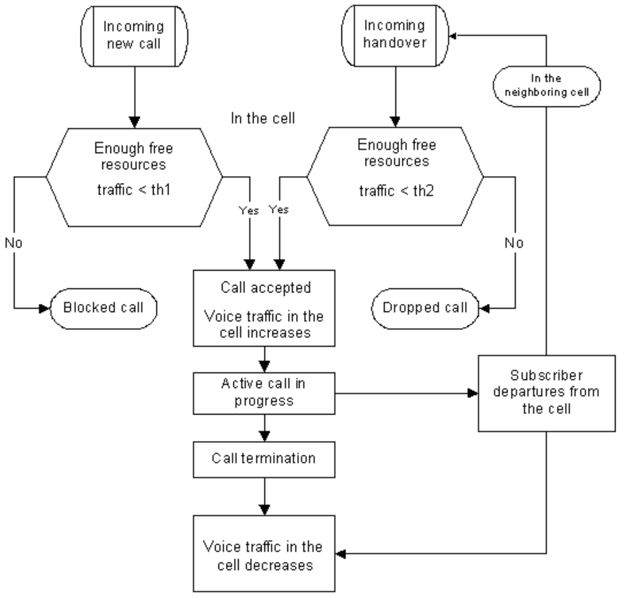

Fig. 1. General principle of network operation

III. The principles of network operation

As written above, in our simulated network we implemented randomly generated voice calls, and few deterministic video streams (1, 2 or 3 video streams). Voice calls bandwidth requirements are fixed to 10 kbps (enough for 3G wireless networks), with exponentially distributed ON-OFF periods. Mean value for these periods is equal to 0.35 s (ON period), and 0.65 s (OFF period). Video streams are PAL (25 fps) video sequences with resolution 316x288 pixels, encoded with MPEG-4 encoder engine. The average bit rate of each of these sequences is approximately 520 kbps. The capacity of a single cell is set to 2048 kbps (2Mbps). From the cell bandwidth certain fractions are assigned to both types of calls (voice and video).

The estimation of the assigned fractions of bandwidth is following few rules:

- Voice calls, once accepted, are of highest priority in the network, and fractions are estimated according to voice bandwidth requirements.
- Video streams, if any, are of lower priority, but have guaranteed minimal fraction of bandwidth, which varies in different simulation scenarios.
- Guaranteed minimal fraction for video is reached with a definition of so-called “threshold” level for voice traffic intensity. This threshold is compared with voice traffic intensity and has the same dimension (bytes/second, or bits/second).
- The control of overwhelming voice traffic (above the “threshold” level) is obtained rather with rejecting new calls, than with terminating active ones. That problem is solved with assigning two different thresholds, one slightly higher than the other. The higher one is valid for the incoming voice handovers, and the lower is for the new calls.

Explained behavior of the system is presented in Fig. 1, where conditions and situations in which one call may be lost (dropped or blocked) are shown. Threshold levels are denoted with *th*1, for new calls, and *th*2 for handovers.

In this work we analyzed three different criteria for determining the threshold levels, as follows:

- *No threshold* (NT): Voice traffic is absolutely of higher priority and can occupy the whole cell bandwidth, if necessary.
- *Fixed threshold* (FT): The fixed fractions of bandwidth are assigned to voice and video traffic (e.g., half of the bandwidth to each one). Voice traffic can occupy the whole bandwidth only when video traffic is not present in that cell. Otherwise, it will use only its portion of the bandwidth.
- *Adaptive threshold* (AT): For each video stream, the system reserves fraction of the bandwidth equal to average bit rate of that video stream. Voice traffic occupies the rest of the bandwidth. Calculation of the video average bit rate is performed on the last few frames of the stream. The system in this case involves prediction, assuming that the next frame will have an average frame size. Concerning the number of frames taken into account for the prediction of the bitrate in near future, we tried several experiments, and concluded that the optimal number is around 100 (10 is too small for getting real average, and 1000 is too large: requires more computational power, and there are problems at the start and at the end of a certain video stream)

By the 3rd criterion, it may seem that video traffic has the priority over the voice, but these reservations are not “hard” reservations, i.e. with reserved bandwidth. It is a “soft” reservation, more like recommendation to the system not to allow more voice calls to enter the system when voice traffic reaches the threshold. However, the fractions of bandwidth are still estimated according to voice traffic, and once accepted, the voice call still has the highest priority.

IV. Simulation results

We performed several simulation experiments in which we obtained dependence of mobile network performances upon various traffic parameters and upon the three defined threshold estimation criteria.

Because there are many traffic parameters that influence in the same manner traffic intensity, we defined so called “traffic coefficient” T. The bigger this traffic coefficient is, the higher is the traffic intensity. We obtained the value of $T = U\lambda\ t$ by multiplying three traffic parameters: Number of subscribers per cell U, average call arrival rate λ [calls/hour/subscriber], and mean call duration (holding time) t [s].

In this simulation experiment, we performed two groups of simulations, using different threshold policies. In each group, we worked with two different scenarios, i.e. with two and three MPEG-4 video streams activated in the monitored cell. In each scenario, for various traffic parameters (various T) we performed several simulations, examining the influence of traffic intensity on the network performance.

Video streams in the system are not always active, but they are introduced one by one. For example, if we have three video streams, the simulation will run for a certain time only with voice, and then sequentially we introduce the three video streams, one after other, as shown in Fig. 2. In the scenario with two video streams we use the first two streams.

We present the simulation results in a particular order with aim to notice the dependence of QoS on our three threshold policies. The results for one video stream (light video traffic), because of the small network load, and small losses, were not illustrative enough, and they will not be shown in this paper. Also, it is important to state that in FT (Fixed Threshold) policy, the threshold level was optimized for medium video traffic intensity, i.e. for two video streams.

In Fig. 3 to 8 we show the packet losses in the system in the case with two video streams.

From Fig. 3 we can observe highest losses of voice packets with NT (No Threshold) strategy, which is odd at first sight because the NT policy gives the highest priority to voice traffic. But, these losses are consequence exactly of that behavior. Uncontrolled acceptance of new voice calls and incoming handovers overwhelms the server buffers in the system, and probability of appearance of lost packets increases. In other words, voice traffic is jamming itself, especially at high traffic intensity in the system, because in that case even the whole bandwidth of the cell is not enough to transfer all the voice calls.

On the other hand, with NT policy we have very few blocked and dropped calls (unrestrained acceptance), which can be seen in the Fig. 7 and 8. The NT policy gives highest packet loss probabilities for video packets too. This results also from high acceptance rate of voice calls. As written above, the bandwidth quotas are estimated according to voice requirements, so in this case we note high packet loss probability for video at higher network load, as shown in the Fig. 4.

Concerning AT and FT policies, the FT policy gives lower losses for both voice and video traffic. This is due to higher acceptance rate in system with AT. If the threshold is fixed (FT), when there is only one active video stream, the system still rejects new voice calls when the traffic is above the fixed threshold, although the rest of the bandwidth may be more than enough for one video stream. As shown in Fig. 5, after introducing the first video stream, the voice traffic is reduced to its value defined by the threshold level. With AT, that problem is solved, and system adaptively recalculates the threshold allowing higher voice traffic intensity, but this also means higher probability of buffer overload and lost packets. As we can see from Fig. 6, with AT scheme voice traffic reduction occurs only after the acceptance of the second video stream in the system. Thus, with AT we have slightly more lost packets, but we avoid unnecessary blocking of new or handover voice calls, which can be seen from Fig. 7 and 8, where both P_b and P_d have smaller values for AT strategy compared to FT.

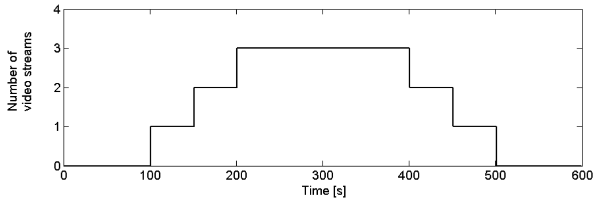

Fig. 2. Introducing video streams in the simulation

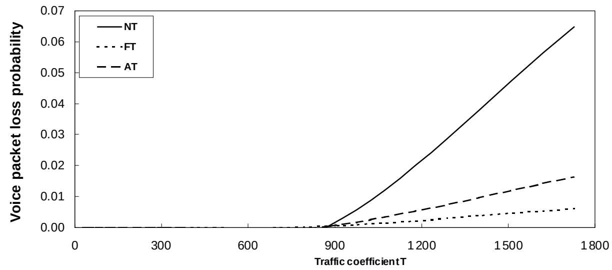

Fig. 3. Voice packet loss probability vs. traffic intensity (2 video streams)

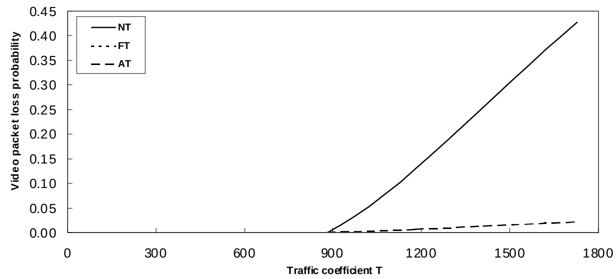

Fig. 4. Video packet loss probability vs. traffic intensity (2 video streams)

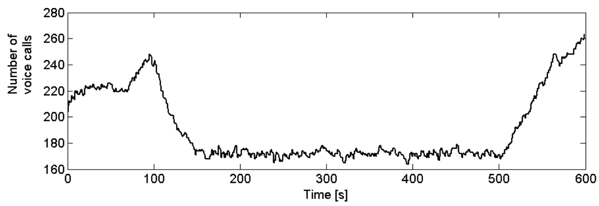

Fig. 5. Number of active voice calls over time (FT)

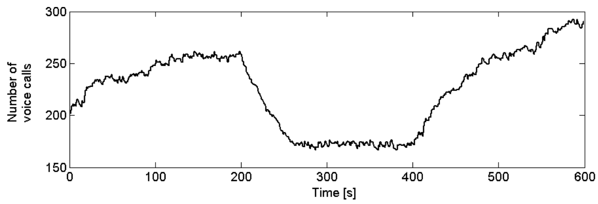

Fig. 6. Number of active voice calls vs. time (AT scheme)

Fig. 9 and 10 presents the losses with three video streams (heavy video traffic) in the system. From these Figures we can derive the same conclusions about high packet losses, and low blocking and dropping probabilities with NT policy. Higher video packet loss probability for FT is the result of the fact that fixed threshold is optimized for medium video traffic intensity (2 video streams), and at heavy video traffic the video bandwidth requirements are not satisfied. At expense of video packets, the voice packets have lower loss probability with FT, but that is negligible comparing to difference in video packet losses.

V. CONCLUSIONS

In this paper we introduced different strategies for handling two types of traffic (voice and video) in multimedia mobile networks. The strategies include various methods of estimating the "threshold", parameter which determines the bandwidth quotas for each of the call types. The performance of the network was considered through its Quality of Service (QoS), defined by new call blocking and call dropping probabilities, and by probability of packet losses. We investigated different strategies upon traffic conditions in the cell, i.e. voice traffic intensity, and different number of video streams.

Using simulation analysis we showed that network performance is highly dependent on chosen threshold strategy for call admission. But, which one is the most optimal with lowest level of losses? The answer to that question is dependent upon the priorities established in a certain system. For example, if we create a system with constraints on call blocking and dropping probabilities, regardless of the lost packets, we could use the NT strategy (with lower quality of accepted connections). On the other hand, if the constraint is packet loss, the use of FT strategy is recommended because of its simplicity and time independence. However, the most flexible solution appears to be AT strategy.

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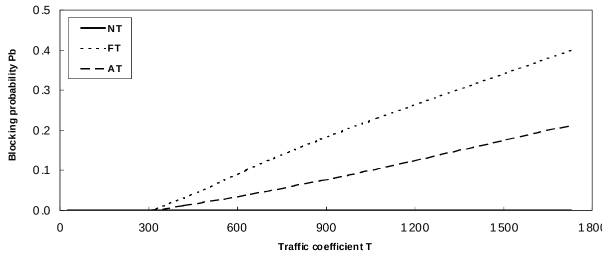

Fig. 7. Call blocking probability vs. traffic intensity (2 video streams)

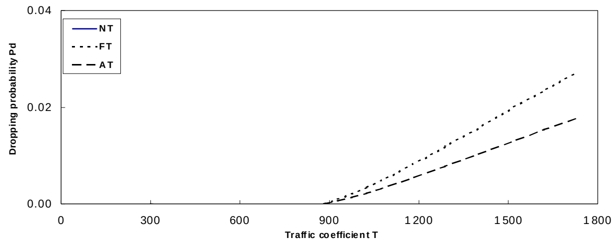

Fig. 8. Call dropping probability vs. traffic intensity (2 video streams)

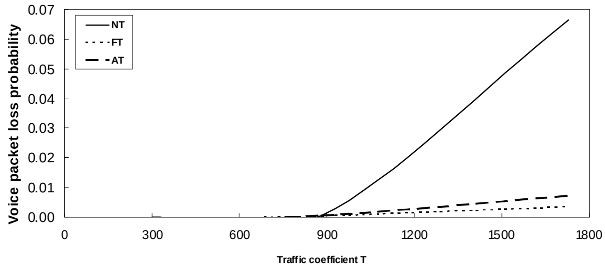

Fig. 9. Voice packet loss probability vs. traffic intensity (3 video streams)

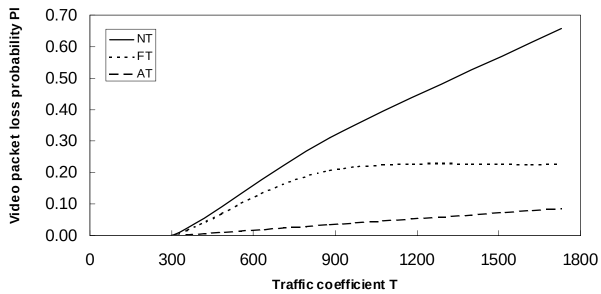

Fig. 10. Video packet loss probability vs. traffic intensity (3 video streams)