Distortion Optimized Multi-Service Scheduling for Next-Generation Wireless Mesh Networks

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***Abstract*—Distributing multimedia content over wireless networks is challenging due to the limited resource availability and the unpredictability of wireless links. As more and more users demand wireless access to (real-time) multimedia services, the impact of constrained resources is different for different media types. Therefore, understanding this impact and developing mechanisms to optimize content delivery under resource constraints according to user perception will be key in improving user satisfaction. In this paper, we develop a novel scheduling algorithm for multi-hop wireless networks, which optimizes packet delivery for multiple audio, video and data flows according to user perceivable quality metrics. We formulate a multidimensional optimization problem to minimize the overall distortion while satisfying resource constraints for the wireless links. Our Quality-of-Experience (QoE)-optimized scheduler makes use of models to determine the user's perception of quality that are specific to the type of service being provided. Our experimental results, obtained with the NS-2 IEEE 802.16 MESH-mode simulator, show that distortion-aware scheduling can significantly increase the perceived quality of multimedia streaming under bandwidth constraints. As the scheduler allows the modeling of fairness constraints among multiple competing flows, we also demonstrate an improvement in fairness across different flows.**

***Index Terms*—Quality of Experience, Distortion, Mean Opinion Score, IEEE 802.16, WiMAX, PSNR, E-model**

I. Introduction

Research in network performance has generally aimed to provide improvement in objective, clearly-measurable traffic parameters, such as throughput, link utilization, delay and packet loss. The optimization of objective network parameters does not, however, translate linearly to optimal satisfaction for the end-user, and the impact of resource constraints on the perceivable quality varies from service to service. Specific characteristics and traffic patterns of services such as audio and video streaming show that common network metrics are hardly indicative of the end-users' perception of the quality of the service being provided. Likewise, network mechanisms for obtaining the best objective-quality parameters (i.e., throughput, delay) may be unsuitablefor the next-generation content-centric, multimedia-oriented networks, where a focus on the end-user's subjective perception, the Quality of Experience (QoE), is expected to yield better results.

Recently, mesh networks have attracted substantial research interest due to their ability to provide broadband wireless access to large areas at reasonable costs. While research on QoS support for meshed networks is very active, so far there has been a limited effort on providing support for QoE management in the meshed environment. Instead, most research work concentrates on resource management for one-hop communications; moreover, it does not cope with multi-hop and multi-service communications.

In the present paper, we present a novel approach for maximizing the QoE for multiple flows having multiple media types in wireless mesh networks. The key contributions are the design of a packet scheduler which operates at each network node and maximizes the user perceived QoE of multiple flows, including audio, video and data, in resource constrained multi-hop wireless networks. For this purpose, we develop a mathematical model which determines the impact of packet dropping on different media types. Based on utility models, we develop an optimization algorithm that maximizes overall flow utility under given resource constraints. A performance evaluation on the NS-2 simulator shows that our approach significantly increases the QoE of multiple flows when the bandwidth of the mesh network is fully utilized.

The remainder of this paper is organized as follows. Section II gives a synopsis of related work on distortion-based packet scheduling, uni- and multi-services, and their support in mesh networks. Section III describes the models selected for the estimation of subjective quality of video, audio, and file transfer services. Then, Section IV describes the scheduler's algorithms and mathematical models for the optimization of multi-service MOS-based mesh scheduling. The implementation of the scheduler in *ns2mesh* and its performance evaluation are presented in Section V. Finally, we summarize our research and include topics for future work in Section VI.

II. Related Work

Most of the prior work investigates service differentiation and packet scheduling while ignoring the packet contents and the implications of packet drops on user perception. On the other hand, the existing research on content-aware scheduling focuses on optimizing a single service over various network designs. In particular, a rate-distortion optimized frame dropping technique for video streaming is presented in [1]. The proposed approach accounts for the impact of packet drops on the transmission rate, a method which we also adopt in our work. Additional related works that focus on single service video streaming networks are [2] that explores rate allocation across multiple wireless networks, [3] that presents a cross-layer approach for video delivery over multi-hop wireless networks, and [4] which proposes multipath transport of video over ad-hoc networks. Finally, the work in [5] optimizes audio streams over WiMAX networks.

To the best of our knowledge, there are very few works that consider jointly the scheduling optimization problem for multiple services, while also taking into account the user perception, or subjective quality, of such services. For instance, the authors in [6] present a multi-user multi-service cross layer optimization which, however, does not consider the specifics of wireless mesh networks in its analysis.

III. Mean Opinion Score and Distortion Models

First, we present an overview of mathematical models that relate certain network and/or content properties to values which express user satisfaction for a given service. The goal is to find models that relate the impact of resource constraints on user perceivable quality. Such resource constraints are typically the result of wireless link characteristics, traffic demands, etc. and manifest themselves as delay, packet loss or bandwidth constraints. Such models will be later used in our framework in order to derive the optimization problem to be solved by the packet scheduler.

We start with probably the most common metric to capture user satisfaction: Mean Opinion Score (MOS). While MOS was originally proposed for subjectively estimating voice quality, it provides a numerical measure of the quality of human speech at the destination. Table I shows the MOS scores and their meaning in terms of user satisfaction.

Table I
MOS Scores and User Perception

Score	User Satisfaction
5	Excellent
4	Good
3	Fair
2	Poor
1	Bad

In our work, we redefine the use of MOS as a measure of user satisfaction with a given service, to other services such as video streaming or downloading. This will enable us to develop an optimization framework based on such MOS metrics that is capable of maximizing overall MOS subject to fairness constraints, or average MOS for all flows subject to resource constraints. Since the MOS based models will be used by our proposed scheduler to estimate the distortion as a result of packet drop decisions under resource constraint, their accuracy will directly reflect on how well the scheduler is capable of keeping distortion across all flows to a minimum.

A. Real-Time Model for Video

A video distortion model should evaluate the impact of packet losses on video quality and account for application-specific parameters such as video codec, loss recovery techniques, bit rate, packetization and content characteristics. It should be applicable to any motion-compensated video scheme, in particular H.264 which is the current *de facto* standard for encoding video.

In this work, we employ the distortion model from [7] that first estimates the impact caused by the loss of a single slice of a frame from the video stream. Then, given the video frame structure and the probability of the occurrence of such a loss, an overall distortion value for the whole stream is obtained.

In particular, the distortion caused by the loss of a single slice in a frame is given by

$$D_1 = \frac{\gamma^{T+1} - (T+1) \cdot \gamma + T}{T \cdot (1-\gamma)^2} \cdot \sigma_S^2, \qquad (1)$$

where σ_S^2 is the distortion caused by the loss of that slice, $T-1$ is the number of P-frames in each group of pictures (GoP), and γ is an attenuation factor ($\gamma < 1$) to account for spatial filtering. To extend this model to multiple losses of different frames, it is necessary to average σ_S^2 over all lost frames.

The overall distortion $\overline{D}$ is then calculated from D_1:

$$\overline{D} = s \cdot \overline{n} \cdot P_e \cdot L \cdot D_1, \qquad (2)$$

where s is the number of slices per video frame, $\overline{n}$ represents the loss burstiness ($\overline{n} \approx 1$ for Bernoulli losses, $\overline{n} > 1$ for Bursty losses, depending on the aggressiveness of the burst errors), P_e is the packet error rate, and L is the number of frames per packet. This model thus includes the effects of the packet loss patterns as expressed by $\overline{n}$ and P_e, the transmission bit rate expressed via the number of slices per frame with the ratio $s \cdot L$, the packetization strategy as represented by L, and the distortion caused by lost slices from D_1.

The Mean Square Error (MSE) distortion $\overline{D}$ can be mapped to a Peak Signal-to-Noise Ratio (PSNR) using

$$PSNR = 10 \cdot log_{10} \frac{255^2}{\overline{D}}, \qquad (3)$$

which we can then map to a MOS score using a non-linear relation, as suggested in [8]:

$$MOS = 4.5 - \frac{3.5}{1 + exp(b_1 \cdot (PSNR - b_2))}, \qquad (4)$$

where b_1 determines the steepness of the mapping curve, and b_2 represents its central point.

B. ITU-T E-Model for audio

The quality of VoIP calls is adversely affected by loss and delay of its audio packets. The ITU-T E-Model [9] provides a parametric estimation and defines an R-factor that accounts for various aspects which impair voice quality. This rating factor R, for VoIP applications [10], [11], is given by:

$$R_f = 94.2 - I_e - I_d, \qquad (5)$$

where I_d represents the impairments caused by delay, and I_e the impairments caused by network and codec losses.

$$MOS = 1 + 0.035 R_f + 7 \cdot 10^{-6} \cdot R_f (R_f - 60)(100 - R_f) \qquad (6)$$

An R-factor higher than 70 translates to a high MOS (an audio stream of good quality).

Delay impairments: The total mouth-to-ear delay d in a VoIP call can be written as

$$d = d_{codec} + d_{de-jitter\_buffer} + d_{playout} \tag{7}$$

with codec delay (d_{codec}), playout delay ($d_{de-jitter\_buffer}$) and network delay ($d_{network}$) [9]. Once d is obtained, its impact on the R-factor is given by:

$$I_d = 0.024 \cdot d + 0.11 \cdot (d - 177.3) \cdot \mathbf{I}(d - 177.3) \tag{8}$$

where $\mathbf{I}(x)$ is the indicator function ('0' if $d < 177.3$, '1' otherwise). The mouth-to-ear delay determines the interactivity of a voice conversation, and has a critical value of 177.3 ms. It is considered that, if the total delay exceeds this value, the interactivity of the conversation is significantly compromised.

Loss impairments: Different codecs react in different ways to packet losses in a VoIP call. The impact I_e of packet losses in the R-factor is given by:

$$I_e = \gamma_1 + \gamma_2 \cdot ln(1 + \gamma_3 \cdot e), \tag{9}$$

where e is the total loss probability (including network and playout buffer losses), and the γ_i's are fitting parameters. Specifically, γ_1 determines the impairment caused by coding, while γ_2 and γ_3 represent the impact due to packet loss for a given codec.

C. General model for file transfer services

The user satisfaction of a file transfer service is solely dependent on the provided data rate. If we consider that each user has a given rate expectation which corresponds to the best user satisfaction, the model in [12] derives the following logarithmic relationship between MOS and throughput:

$$MOS = a \cdot log_{10}[b \cdot R \cdot (1 - P_e)], \tag{10}$$

where R is the data rate of the service and P_e is the packet error rate. The parameters a and b in (10) are obtained from the upper and lower rate expectations for the service.

IV. Multi-Service Distortion-Optimized Packet Scheduler for Mesh Networks

The goal of our packet scheduler, which runs at every network node, is to maximize the overall achievable user perceived quality and fairness among competing flows under given resource constraints. Intelligent scheduling of the packet transmissions at a node is necessary in mesh networks in order to deal with bandwidth bottlenecks that can occur anywhere along the network path of a flow.

To this end, we formulate a minimization problem of a cost function which denotes the impact of packet drop decisions on the MOS of individual flows, given the utility functions as defined in the previous sections. Such packet drop decisions will be triggered by an active queue management algorithm in case resource limitations are discovered, e.g., once the buffer utilization reaches a definable threshold. The scheduler then determines, based on the outgoing data rate and buffer fill rate, how much data it needs to drop from the buffer so as to keep it from overflowing.

Figure 1 illustrates the scheduling process. The scheduler locates sets of packet combinations that would satisfy the required buffer reduction. For each of these combinations, an estimation of the MOS decrease for each flow is calculated. The scheduler then drops the packets whose combination results in the smallest MOS decrease.

Also considered in the decision process is the fairness of the packet drop selections. In particular, we associate a penalty factor with each packet drop combination based on the standard deviation of the incurred MOS reductions. In this way, the scheduling mechanism favors packet drop combinations that contribute to the smallest variation of MOS changes across the various flows.

One issue in mesh networks is that the nodes on a network path are unaware of what packets were dropped at upstream nodes, which is important for accurate estimation of a flow's quality. To this end, we propose to keep track of dropped packets at a node, and then broadcast this information to its neighbours. For audio and data flows, the impact of packet loss can be inferred by tracking sequence numbers. On the other hand, for video flows, each node must attach a header to the packets it is forwarding, containing the distortion of the frames it has dropped.

A. Rate-Distortion Cost Function

The optimal packet drop combination can be found by minimizing a cost function, $Q(p)$, calculated for each packet drop combination p. Specifically, for n flows, $Q(p)$ is given by:

$$Q(p) = \sum_{i=1}^{n} k_i \cdot \Delta MOS_p^i - \lambda \cdot \sum_{i=1}^{n} \Delta R_p^i + \mu \cdot n \cdot \sigma(\Delta MOS)\,, \tag{11}$$

where ΔMOS_p^i is the MOS reduction for flow i due to the dropping of p, and k_i is a weighting coefficient for preferential packet dropping from specific flows. Through k_i, the scheduler can be configured to drop more packets from certain classes of traffic, such as from data streams. Furthermore, ΔR_p^i is the rate reduction of flow i due to the dropping of p, while λ is a weighting factor that controls how aggressively the scheduler drops packets. Finally, $\sigma(\Delta MOS)$ is the standard deviation of the MOS reduction over all flows, while μ determines the weight of the standard deviation in (11). ΔMOS values take into account the distortion information sent by the other nodes in the flow's path, in order to determine the flow's real MOS (as opposed to the MOS seen locally at the node). Knowing the accumulated distortion is important as the MOS mappings are non-linear, and different emphasis is given to each drop depending on the status of the associated flow.

The generic model in (11) can be applied directly to all packets present in the buffer. However, to reduce the processing time, a pre-selection of packet combinations can be done based on a target buffer size. The rate reduction component $\lambda \sum_{i=1}^{n} \Delta R_p^i$ is thus unnecessary, resulting in the following

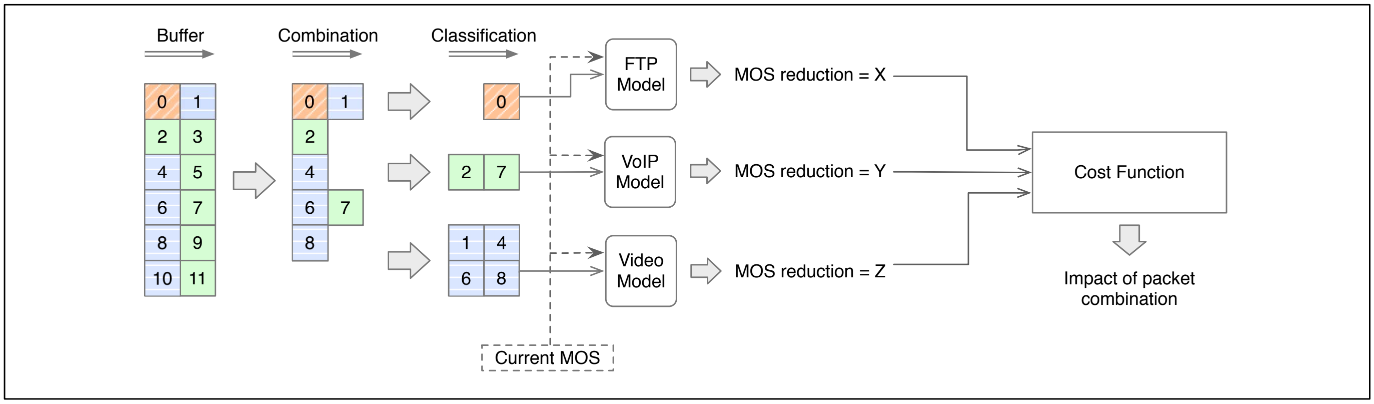

Figure 1. Determining the impact of a packet combination

minimization function:

$$Q'(p) = \sum_{i=1}^{n} k_i \cdot \Delta MOS_p^i + \mu \cdot n \cdot \sigma(\Delta MOS) \qquad (12)$$

Specifically, to obtain a set of suitable packet drop combinations from the buffer, we first determine the amount L of bits to be dropped from the queue based on the buffer fill rate R_{buff}, the outgoing data rate R_{tx}, and the buffer fullness λ_{buff}:

$$L = (R_{buff} - R_{tx}) \cdot \lambda_{buff} \qquad (13)$$

Then, a range of $[C_{min} \cdot L, C_{max} \cdot L]$ bits is chosen and a set S of n_S packets is selected so that

$$C_{min} \cdot L < \sum_{i \in S} size(S_i) < C_{max} \cdot L, \qquad (14)$$

i.e., the total size of all packets in the set must be within a range of C_{min} and C_{max} around L.

V. Performance Evaluation

In order to evaluate its performance, we implemented our scheduler, the associated MOS models and the distortion tracking agents in the *ns2mesh* simulator [13], an extension to the NS-2 simulator[14] that adds support for 802.16 mesh topologies.

We simulate video traffic with video traces found at [15]. To carry distortion information, a hint track is added to the video streams by attaching a header to the first packet of each video frame, containing the frame's associated Mean Square Error. Further information on the structure of the employed video traces and how to process and packetize their video frame information can be found in [16].

A. Implementation

The implemented architecture acts on top of the existing packet scheduler in *ns2mesh*, which is responsible for pushing packets from the buffers to bandwidth reservation slots. We set the queue management algorithm to be active on the node's buffer in the case of either of the following two events:

(a) - the buffer size exceeds a user-defined threshold
(b) - a packet is to be dropped due to insufficient buffer space

Once the queue management algorithm is invoked, it first begins by profiling the packets in the buffer, to determine their size, distortion information (for video packets), and to make a list of which flows are passing through the node. Then, equation (13) is applied to determine the amount of bytes that are to be dropped from the buffer. Next, the scheduler determines those packet combinations that satisfy (14). The number of combinations, in a full-lookup method, is equal to 2^n, where n is the number of packets in the buffer. Each combination ID is a base-10 integer and its base-2 equivalent represents an array of boolean values. An example of this mapping can be seen in Figure 2.

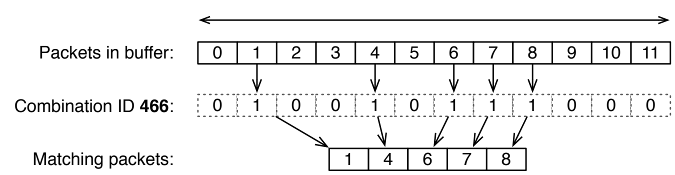

Figure 2. Mapping a base-10 combination ID to a list of packets

With the list of packet drop combinations satisfying (14), the MOS reduction per combination is calculated for each flow. It is also at this point that the MOS of the flows going through the node is updated. The overall total and mean reduction in MOS is calculated from these MOS values, as well as their standard deviation (σ^2). Finally, (12) is applied to find the packet combination with the lowest impact on quality. The scheduler then drops this combination from the buffer.

B. Methodology

We ran a series of simulations in which we compared the performance of our MOS-based scheduler to that of a standard (MOS-unaware) mesh scheduler. Each simulation was run for 10 seconds and the first second of all measurements was subsequently discarded in order to remove the effect of network transients. We used the following simulation parameters:

- Burst profile *QPSK 1/2*, max link bandwidth 5.82 Mbps.
- Scheduler is triggered on buffer overflow, locate combinations of 15% the buffer size, with a 1% margin.

- 2^{20} combinations evaluated for size (1,048,576).
- $w\_ftp = 0.8$: FTP packets are 20% more likely to be dropped.
- Video parameters: attenuation factor $\gamma = 0$, P-frames per GoP $T - 1 = 10$, slices per frame $s = 1$, frames per packet $L = 1$, loss factor $\overline{n} = 1$ (bernoulli-type losses).
- Audio parameters: $\gamma_1 = 0, \gamma_2 = 30, \gamma_3 = 15$, codec G.711.
- FTP parameters: $a = 2.1, b = 0.3$. Maximum MOS for throughput $\geq$ 400 kbps, minimum for $<$ 10 kbps.

C. Video Service Scenario

Video streaming is the service with the most potential for MOS improvements due to proper scheduling. While in audio and data streams the packets are similar to each other, in video there are specific frames that contribute much more to the overall quality than some other frames. From highest to lowest importance, we have I-frames, P-frames, and B-frames. Losing a B-frame from the stream can only cause minor loss of detail, while losing an I-frame can cause skipping, blockiness, and other artifacts, thereby resulting into much larger distortion values. Here, we consider a scenario involving video streams only.

1) Video packet scheduling evaluation: The first simulation is run on a network grid topology with 9 nodes arranged in a square 3x3 pattern illustrated in Figure 3. This is a common topology for a mesh network, which can be mapped to a number of hypothetical real-life deployments. We introduce 9 flows in the network with random source and destination node pairs, and observe the behavior of these flows.

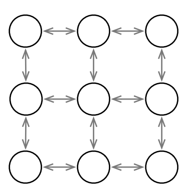

Figure 3. 9-node grid topology

We repeat the simulation with increasing number of flows, and the average MOS for all flows was plotted against that for the case of *ns2mesh*'s standard scheduler (see Figure 4). The gains in user perception are significant. As the network load is increasing, for more than three concurrent flows, the application of the MOS scheduler consistently returns improvements of more than 0.5 MOS points. By employing this scheduler, the network can support up to 7 video flows with a MOS rating of more than 4 points, a very good quality value for the end-user, and with 10 flows the resulting MOS is 2.5 points, which is about the average on the MOS scale.

D. Multi-Service Scenario

This simulation is meant to determine the performance of the scheduler when three different types of traffic are present in the network: video, voice, and file transfer services. Voice and file transfer services are less prone to quality degradation due to random packet dropping. The goal here is for the scheduler to be able to balance all three services concurrently, interacting with packets from all services at the same time.

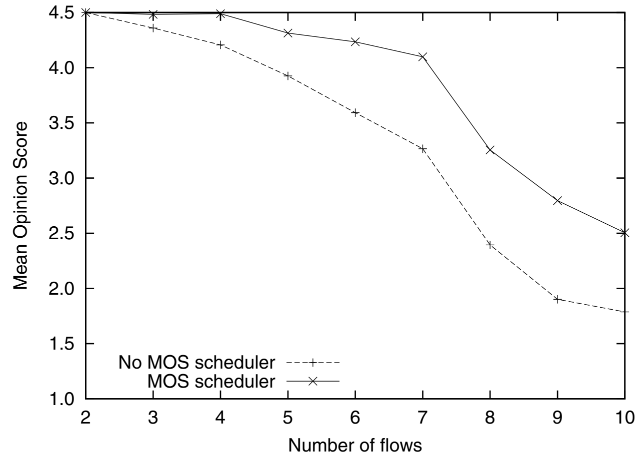

Figure 4. Average video MOS in a 9-node grid network for 'n' flows

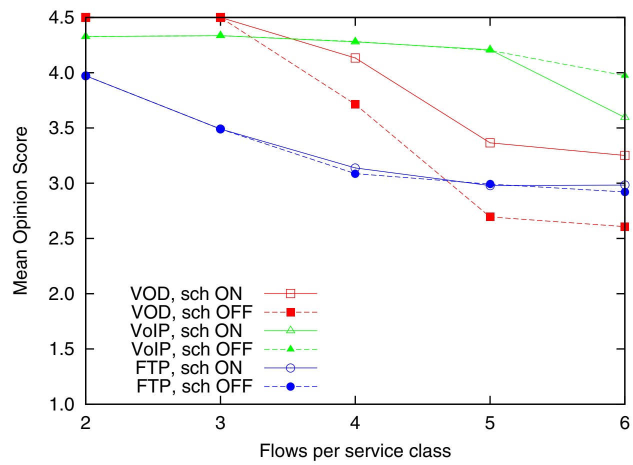

Figure 5. Average MOS in a 9-node grid network with 'n' flows per service

1) 9-Node Grid Topology: We now use three different service types on the same 9-node grid topology, where flows have random source and destination nodes. We increase the number of flows of each traffic type successively to saturate the network.

Figure 5 plots the average MOS of all flows per service type. The measurements with the MOS scheduler are plotted with a solid line, while those with the original *ns2mesh* scheduler are plotted with a dashed line. As expected, the gains are mostly seen on the video flows, with an average increase of 0.2 to 0.5 MOS points. The VoIP flows suffer a higher drop rate when network saturation is increased (6 flows per traffic type), due to being the flows with the higher MOS (relative to video). FTP flows already have the lowest MOS in the *ns2mesh* simulator, and thus, are relatively unaffected. It can be seen that in the highest saturation scenario, the flows under the MOS scheduler are spaced much closer together than the ones without this

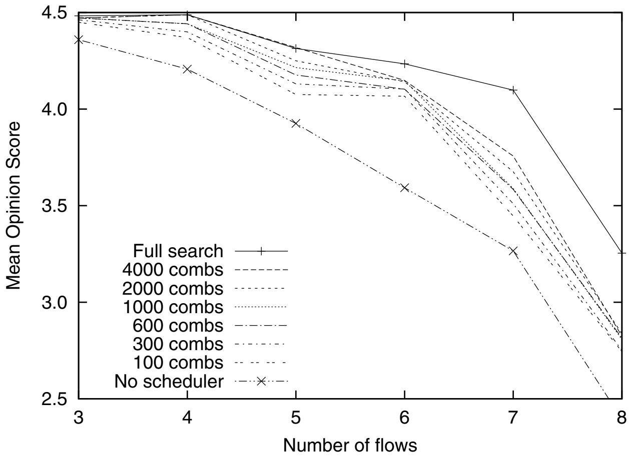

Figure 6. Average video MOS for varying number of drop combinations

scheduler.

E. Computational Complexity

The most resource-intensive processes in the implementation of our MOS scheduler are:

- Evaluation of packet combinations for their total bit size
- Computing each combination's MOS impact

In order to determine the influence of computational complexity on scheduling performance, we re-examine the 9-node grid scenario from Section V-C1, but now we progressively decrease the number of combinations (*maxcomb*) passed to the cost function. For each simulation run, we respectively set *maxcombs* to 4000, 2000, 1000, 600, 300, 100, and infinity. The combinations are chosen as they are found while matching for size. We also compare these results with those for the original *ns2mesh* scheduler.

The data shown in Figure 6 shows a predictable decrease in the MOS of all flows with the decrease in evaluated packet drop combinations. We can observe that limiting *maxcombs* to 4000 and 2000 does not significantly impact the final quality when less than 6 flows are present in the network at the same time. Also, setting the limit to a small value, such as 100 combinations, is generally sufficient to obtain a gain of 0.2 to 0.5 MOS points, which represents a good trade-off given the negligible computational effort that is invested.

VI. Conclusions & Future Work

We have developed a distortion optimized packet scheduler that maximizes the QoE in multi-service wireless mesh networks. The proposed approach is based on the minimization of a flexible cost function that is capable of evaluating not just the effects of dropping a single packet from a stream, but the joint effect of dropping packet combinations over multiple flows to achieve an efficient utilization of the available network resources. The optimization takes advantage of the fact that different media types as well as the data units comprising a single media presentation typically exhibit varying degrees of sensitivity to limited network resources. By taking into account the computational demands of our modeling, the optimization framework is also effective for actual deployments, where constraints in processing power must be accounted for. Our simulation results show a significant increase in user perceived service quality for delivery over WiMax based Mesh Networks, with an emphasis on video content, but also for audio communication and file transfer. Moreover, the scheduler's approach of analyzing all services combined and of contemplating multiple packet drops from any individual service leads to a more balanced distribution of quality, as packets can be dropped from one service to improve another service.

As future work, we aim to evaluate the performance of the optimized scheduler with other MOS models, especially for video content, as this specific research is still ongoing. Also, an evaluation of this scheduler in a wireless mesh testbed would yield important results regarding viability of the algorithms in network setups where hardware performance is normally inferior to that of simulation setups.

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