Dynamic Bandwidth Control in Wireless Mesh Networks: A Quality of Experience based Approach

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***Abstract*—Wireless Mesh Networks (WMNs) are gaining an increasingly important role in next generation wireless networks. Due to their advantages over other wireless and wired networks, WMNs are undergoing rapid progress and are supposed to deliver wireless services for a large variety of applications. Especially real-time applications such as voice over IP make high demands on wireless mesh networks. A small change of the Quality-of-Service (QoS) metrics like packet loss, delay, and jitter have a significant impact on the Quality-of-Experience (QoE), a subjective measure from the user perspective of the overall value of the provided service or application.**

In this paper, we present a dynamic bandwidth control mechanism which measures the current situation in the network and adapts the bandwidth in order to ensure a high QoE level. The mechanism is implemented in a Wireless LAN mesh testbed and the results show that real-time applications are successfully protected from disturbing best effort traffic flows.

***Index Terms*—QoE, Mesh, WLAN, 802.11, Testbed**

I. Introduction

W*IRELESS mesh networks* (WMNs) provide cheap, reliable, and flexible broadband Internet access in local and metropolitan areas. Similar to wireless ad-hoc networks, no central unit is distributing traffic and the data is sent directly from neighbor node to neighbor node. If data shall reach nodes that are not directly reachable neighbors, the packets are sent on a multi hop route. All nodes provide relaying capabilities to forward traffic through the network to reach the destination. However, in contrast to wireless ad-hoc networks, WMNs are normally static devices and focus on reliability, network capacity, and are mainly used as an alternative to a wired network infrastructure.

Due to the advantages of WMNs like self-organization and self-healing, several standardization groups have been set up. The first standardization group for *Wireless Local Area Networks* (WLANs) was started in 2003 under the extension IEEE 802.11s [1]. Besides the IEEE 802.11s standard further standardization groups for WMNs like IEEE 802.15.5 [2] and IEEE 802.16j [3] underline the importance of wireless mesh networks.

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Major research aspects in WMNs are intelligent routing strategies and *Quality-of-Service* (QoS) support. In this paper, we present a distributed, measurement-based approach to support real-time traffic in WLAN-based mesh networks. The aim of the proposed mechanism is to keep track of the services currently present in the network and to ensure a stable and high QoS. The objective measurable QoS parameters are then mapped to the user-perceived *Quality-of-Experience* (QoE), expressed through the *Mean Opinion Score* (MOS) [4]. The tools for the approach are implemented in a WLAN-based mesh testbed. The results reveal that the mechanism prevents real-time flows from disturbing best effort flows by observing the QoS parameters and controlling the throughput of the best effort flows on the network layer. As an extension to our work in [5], the performance of the mechanism is measured for two different scenarios, disturbing traffic flows on the same path to the destination as well as on a crossing path in the wireless mesh network.

The remainder of the paper is organized as follows. In Section II the work related to QoS and QoE issues in wireless mesh networks is shown. This is followed by Section III, introducing wireless mesh networks and its known problems. Our approach is presented in Section IV and Section V shows the results of performance measurements. Finally, a short conclusion is given in Section VI.

II. Related Work

One step towards QoS support in IEEE 802.11 networks is defined in the IEEE 802.11e standard for service differentiation, which slightly modifies the *Carrier Sense Multiple Access/Collision Avoidance* (CSMA/CA) mechanism. However, the standard does not guarantee a good QoS level, especially in highly loaded networks. This has been tested and improved for single hop environments in [6], [7], and [8].

A MAC protocol for QoS support in WMNs is proposed by Carlson et al. [9]. It is called *Distributed end-to-end Allocation of time slots for REal-time traffic* (DARE). In this protocol, time slots are reserved in all mesh nodes along a real-time traffic's route to ensure a transmission with good QoS performance. The reservations are thus done for fix routes but repair mechanism are provided if a link fails and the route has to be changed. The DARE approach is implemented and tested in a simulation with ns-2.

Besides the simulation-based adaptation mechanisms, Guo et al. [10] implemented a mechanism called *Software-based Time Division Multiple Access* (STDMA) on top of the WLAN

MAC layer in a testbed. The approach is designed to support WLAN-based VoIP appplications and it is claimed that a significant improvement of the maximum number of G.729-quality voice conversations in a WLAN is achieved. Typical scenarios with both best effort and real-time traffic are though not in the scope. This reduction to single use cases is, besides the MAC layer changes, the second difference to the approach presented here.

There are also propositions for QoE provisioning on higher layers. He et al. [11] introduce a middleware-based QoS control in 802.11 wireless networks. The idea is to implement a traffic prioritization inside the mesh nodes based on control theory. To realize this prioritization a ”middleware design with cross-layer framework” is introduced and implemented in a Linux-based testbed. Above the middleware, the applications have the possibility to define requirements for single connections. Before a service is started, the application informs the middleware that certain QoS specifications are needed for the desired flow between two end points. The middleware's task is to choose an adequate service class on a dynamical base depending on the current performance of the service and the demanded requirements. By a control loop the current quality is measured and compared to the desired one. Depending on the current ”quality error” dynamical packet scheduling is performed.

To distinguish our approach from [11] two things are mentioned. As the middleware approach is based on prioritization inside the mesh nodes, only problems caused by traffic passing through one of the nodes prioritizing multimedia streams can be handled. If the traffic problems occur due to collisions on the air interface caused by nodes that are not demanded to prioritize any real-time traffic among themselves, they will not recognize any problem and not control the disturber to solve the problem. There is no signaling mechanism between different nodes using the middleware software to locate a problem outside the real-time route. Depending on the focused field of application, there might be a second drawback of the approach presented in [11]: All services that need a certain QoS performance have to be announced first.

III. WLAN Mesh Networks and the MeshBed Setup

Wireless mesh networks are an interesting to provide broadband wireless Internet access. Fig. 1 shows a WMN in a hierarchical structure. Starting at the bottom, normal non-mesh capable wireless or wireline clients are attached to the mesh network by *Mesh Access Points* (MAPs). These MAPs form together with other *Mesh Points* (MPs) the mesh network itself. A MP is responsible for mesh relaying, meaning that it is capable of forming an association with its neighbors and forwarding traffic on behalf of other MPs. The top of the hierarchy in Fig. 1 constitutes a *Mesh Point Portal* (MPP). The MPP bridges traffic between different WMNs or connects the WMN to the Internet.

As todays technology and infrastructure developments have advanced, e.g. when looking at WMNs, the services used by the customers nowadays have as well. As for instance *Voice over IP* (VoIP) has become more and more popular. Networks and mechanisms are necessary to ensure a high quality. The performance of these real-time applications in WMNs has been widely studied in terms of simulation, but only a few testbeds exist. We have investigated the possibility of real-time application support in a WLAN-based mesh network testbed, called ”*MeshBed*”. The *MeshBed* has been developed and is deployed at T-Systems in Darmstadt, Germany and has been set up ”to investigate carrier-grade aspects from a network operator's point of view”. The *MeshBed* offers all main aspects of WMNs to make the testing possibilities as various as possible. Details about the *MeshBed* can be found in [12].

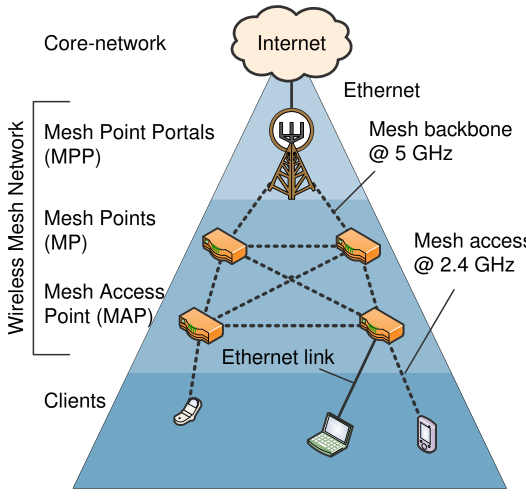


Fig. 1. *MeshBed* architecture

Fig. 2 shows the floor plan of the T-Systems building in Darmstadt, where the *MeshBed* has been set up. Currently, the *MeshBed* consists of 12 mesh points, which can all be configured to serve as MAPs. The MPs consist of embedded AMD Geode SC1100 Systems with 266 MHz CPUs and 64 MB of RAM. All mesh nodes are equipped with two Atheros Wireless Mini PCI Wi-Fi Cards as well as an Ethernet port and Debian Linux is installed together with the madwifi [13] driver. Furthermore, two MPPs are set up which are equipped with 3 GHz Intel Pentium 4 processors and 1 GB of RAM.

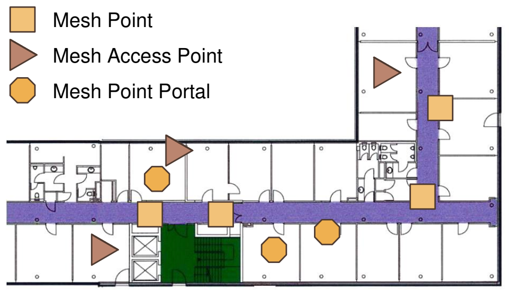


Fig. 2. *MeshBed* indoor deployment

IV. A Routing Layer based approach

A. *Idea and General Structure*

1) Idea of the Approach: The general idea of the approach is to perform the QoS support at the routing layer. MAC layer changes would be possible as well but they are not suited in this case. WLAN has already become a wide spread technology. Changing something in the MAC layer as currently standardized would not just mean an update to or recreation of all drivers for the WLAN devices but also implies possible hardware changes in those devices. This makes the deployment and usage of new MAC mechanisms very difficult.

Routing layer mechanisms to enhance QoS are a promising approach for WLAN-based mesh networks. The routing layer is easily exchangeable, as it is totally based on software. Independent of the operating system, the routing layer is logically situated on top of the network device driver and interacting with it via driver independent interfaces.

In the presented approach, maximal adaptability and flexibility is reached through a distributed solution. Every relay node is equipped with capacities to monitor, judge, and react on the current network situation.

The aim of the proposed mechanism is to keep track of the services currently present in the network. Approaching or already present problems shall be recognized as fast as possible. Solutions to those problems on different ways shall be provided to ensure a stable and high QoS level.

This aim basically needs two main tools to be realized, a *Traffic Observer* that analyzes the current network situation and a *Traffic Controller* that offers different possibilities to influence the actual situation to provide high QoS. Furthermore, an effective way to allow communication between those two components not only when present on one mesh node but also when distributed throughout the network is necessary. The following sections explain the different parts of the mechanism in more detail.

2) General Structure and Interoperability: Fig. 3 shows the general structure of the developed mechanisms. The core of the implementation is formed by the OLSR implementation of Andreas Tonnesen *OLSRd* [14]. Running this software on every node enables the mesh routers to connect to each other and to form the *MeshBed*. The *Traffic Observer* is implemented as a kernel module. It is runnable independently of *OLSRd* and can be compiled and used on any linux machine with the correct kernel version. The *Traffic Controller* is implemented as a plugin to the *OLSRd* plugin interface. It includes a signaling unit making use of the *OLSRd* broadcast messages and allows thus communication between different *Traffic Controller*s. Located on one single node, *Traffic Observer* and *Traffic Controller* are contacting each other via the Linux *netlink* sockets.

B. *Traffic Observer*

The key part of the presented approach is the component called *Traffic Observer*. Its tasks are two folded. On the one hand this module has to monitor the current situation in the

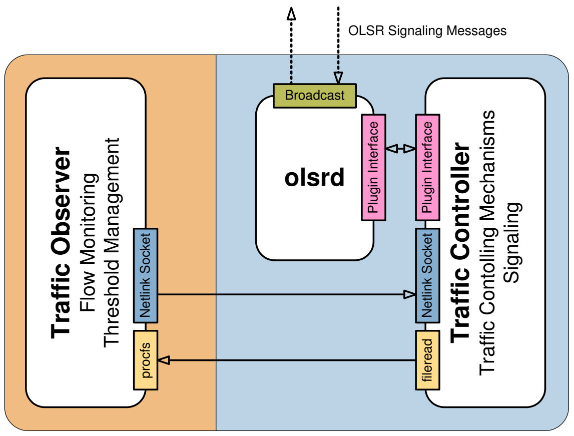


Fig. 3. General structure

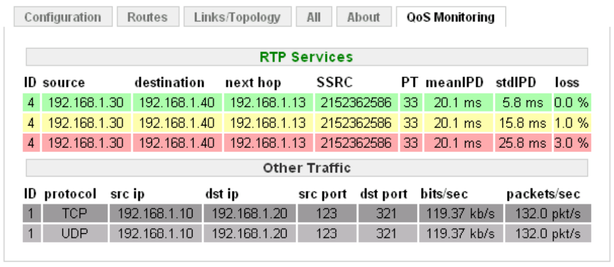
mesh network by observing the traffic flows, as well as other information that can be obtained from the network. On the other hand it has to judge whether the current network situation is acceptable or, if this is not the case, how to react on the occurring problems. To realize this, certain thresholds are needed. In the following sections each of these two tasks is presented in detail.

1) Flow Monitoring: As mentioned before, the most important task of the *Traffic Observer*, as the name says, is observing the network and the traffic inside it. Especially because *Traffic Observer* and *Traffic Controller* are normally situated in every relay node, a lot of information is obtained and analyzed. In a raw classification one might separate this information into packet or traffic related information and non-packet or -traffic related information. Even though the latter one, including things like CPU usage or memory load at the monitoring node, might also be of big interest, the main focus lies on the former.

Traffic related information are those information concerning the traffic of the network, i.e. the packets describing this traffic in the case of IP as in WLAN-based mesh networks. One of the main aims of the approach presented in this work is a distributed solution to the issue that is highly adaptable to different scenarios and network changes. This has a large impact on the possible choice of monitorable information. No information of neighbor nodes about their observations can be included in the measurements for two reasons. First, the standard packet structure of real-time services does not include any place to transport those information. Second, sending this information in separate packets with regular time intervals is impossible due to an insolvable trade off between too much signaling overhead and too imprecise information. OLSRv2 might solve this problem because it provides a more flexible signaling framework but produces more overhead due to periodic signaling.

All information the *Traffic Observer* can analyze about the currently active services is obtained by the observation of the packets passing by in the own node. Three different types of information can be obtained for a certain packet stream. First of all there is the explicit time independent information readable out of the packets content, as for instance source or destination address or protocol type. Next, there is the implicit time dependent information which is obtainable at the moment of the packet monitoring, e.g. the packet absolute arrival time or relative arrival time after the last packet of the same service. Finally, there is statistical information that is based on a series of packets rather than on a single one. This information provides a long term analysis of the monitored services, for instance packet loss over the last n packets or the standard deviation of the packet inter arrival time. The measurement of the widely used one way delay metric is evidently not possible in this approach as information of more than one time stamp at other nodes in the network would be necessary. Though obtaining this information is impossible as explained before.

Fig. 4 shows a screen shot of the graphical information page displaying the information provided by the *Traffic Observer*. In the following section all displayed values are shortly described and assigned to the above classification. Furthermore, the equations to calculate the statistical information and to compute the MOS are given.



RTP Services

ID	source	destination	next hop	SSRC	PT	meanIPD	stdIPD	loss
4	192.168.1.30	192.168.1.40	192.168.1.13	2152362586	33	20.1 ms	5.8 ms	0.0 %
4	192.168.1.30	192.168.1.40	192.168.1.13	2152362586	33	20.1 ms	15.8 ms	1.0 %
4	192.168.1.30	192.168.1.40	192.168.1.13	2152362586	33	20.1 ms	25.8 ms	3.0 %

Other Traffic

ID	protocol	src ip	dst ip	src port	dst port	bits/sec	packets/sec
1	TCP	192.168.1.10	192.168.1.20	123	321	119.37 kb/s	132.0 pkt/s
1	UDP	192.168.1.10	192.168.1.20	123	321	119.37 kb/s	132.0 pkt/s

Fig. 4. A screenshot from the browsers monitoring page

The information collected for Premium and RTP services are as follows: source, destination, and next hop IP address of the packet can be obtained as explicit information, either out of the packet header, or in case of the next hop address out of the routing table by knowledge of the destination address. The payload type of the RTP service and its unique SSRC number are also explicitly readable from the packet header. The combination of SSRC and next hop address is used to assign a unique ID to each service. Packets with the same SSRC and next hop obtain the same ID and are collected together.

The values $mean_{IPD}$, std_{IPD}, and $loss$ are statistical information. To explain their calculation, the following definitions are given: For every packet p_i the following implicit and explicit information can be obtained:

ϕ_i: unique identification number of p_i,

t_i: absolute arrival time of p_i,

$\Delta t_i = \frac{t_i - t_{i-1}}{\phi_i - \phi_{i-1}}$: relative arrival time of p_i, and

l_i: total length of p_i in Bytes.

Furthermore, sets are held containing the obtained values for the last window size w packets $P = \{p_{last-w+1}, \ldots, p_{last}\}$ sorted by time of packet arrival:

$\Phi = \{\phi_{last-w+1}, \ldots, \phi_{last}\}$,

$T = \{t_{last-w+1}, \ldots, t_{last}\}$,

$\Delta T = \{\Delta t_{last-w+1}, \ldots, \Delta t_{last}\}$, and

$L = \{l_{last-w+1}, \ldots, l_{last}\}$.

Using these definitions, the statistical information can be obtained as follows:
The mean inter packet delay $mean_{IPD}$ is defined as

$$mean_{IPD} = mean[\Delta T] = \frac{\sum_{x \in \Delta T} x}{w}. \tag{1}$$

The standard deviation of the inter packet delay std_{IPD} is defined as

$$std_{IPD} = \sqrt{\frac{w}{w-1} \cdot \left(\frac{\sum_{x \in \Delta T} x^2}{w} - \left(\frac{\sum_{x \in \Delta T} x}{w} \right)^2 \right)} \tag{2}$$

and the packet loss $loss$ is defined as

$$loss = 1 - \frac{|\Phi|}{max[\Phi] - min[\Phi] + 1} = 1 - \frac{w}{max[\Phi] - min[\Phi] + 1} \tag{3}$$

The information collected for other traffic, i.e. non real-time traffic are as follows: The protocol type, source and destination addresses, and ports are explicit information of the packet header. The combination of source and destination addresses and ports are used to assign a packet to the correct monitored service. Bits/sec and pkts/sec are statistical information calculated as follows using the above definitions:
The bandwidth in bits/sec bps is defined as

$$bps = \frac{\sum_{l \in L} l}{max[T] - min[T]} \tag{4}$$

The packet rate in pkts/sec is defined as

$$pktps = \frac{|L|}{max[T] - min[T]} = \frac{w}{max[T] - min[T]} \tag{5}$$

2) Threshold Management: The preceding section has offered a look inside the *Traffic Observer*'s monitoring facilities. It displayed which different types of information and parameters are measurable and how they are obtained. All information provided by the *Traffic Observer* is always available up to the most recent packet on demand via the Linux proc filesystem *procfs*.

Monitoring of the services alone is though not enough to do QoS/QoE monitoring and enhancement. There is also the need for a mechanism that judges the monitored information and reacts in the case of a possible quality decrease. To realize this task, a threshold management in the *Traffic Observer* is necessary. Following a common way of illustration, traffic light charts with colors green, yellow, and red depicting good, average, and bad quality are used.

Key parameters have to be compared to adequate thresholds to assign them with the correct color, i.e. quality level. The key parameters chosen in this work to judge QoS and a possible QoS degradation are the previously introduced std_{IPD} and $loss$.

In this work, the thresholds to do the QoS judgment on this parameters are configured service dependent. Each RTP payload type can be configured with four own values describing the $std_{IPD_{green-yellow}}$, $std_{IPD_{yellow-red}}$, $loss_{green-yellow}$, and $loss_{yellow-red}$ thresholds. One might imagine that thresholds could become less demanding in case of a larger number of services in the network or more claiming in an empty network. The thresholds defined in this work are though intentionally not adapting to different network situations. They are set to fixed values for every type of service.

As said before, the monitored values of the *Traffic Observer* are always available on demand via the *procfs*. More precisely, the explicit and implicit information for the w last packets are saved internally. At the moment of access to the *procfs*, the statistical information is calculated. The judged key parameters std_{IPD} and $loss$ belong to the statistical information as well. Nevertheless, they have to be compared to the thresholds regularly and not just on demand. std_{IPD} and $loss$ are thus calculated when $\lfloor \frac{w}{10} \rfloor$ new packets have arrived. For instance in case of $w = 100$ with the arrival of every 10th packet the std_{IPD} and $loss$ values are updated. Afterwards, the values are compared to the thresholds shown in Table I. If the thresholds are exceeded, an alert is broadcast via the linux *netlink* socket. To avoid an alert flooding during the process of the reaction period, alerts are sent with an interval of 1 second.

TABLE I
QoE THRESHOLDS

Quality Level	MOS	$loss$	threshold $loss$	std_{IPD}	threshold std_{IPD}
good	3.8-5	$<$ 0.3 %		$<$ 1.7 ms	
average	3-3.8	0.3-1.7 %	0.1 %	1.7-7.2 ms	1.5 ms
bad	1-3	$>$1.7 %	1.5 %	$>$7.2 ms	7.0 ms

As we have seen a really small std_{IPD} during the measurements for this paper, the parameter was neglected in Section V. Instead, we just used the QoS parameter $loss$ to calculate the *Mean Opinion Score* (MOS). According to Hossfeld et al. [15], [16] there is a clear exponential relationship between the packet loss ratio and the MOS for the ITU-T G.711 voice codec [17]. As we are using this codec for the measurements, the MOS can be calculated using the following equation from Hossfeld et al.:

$$MOS = 2.861 \cdot e^{-29.816 \cdot loss} + 1.134. \tag{6}$$

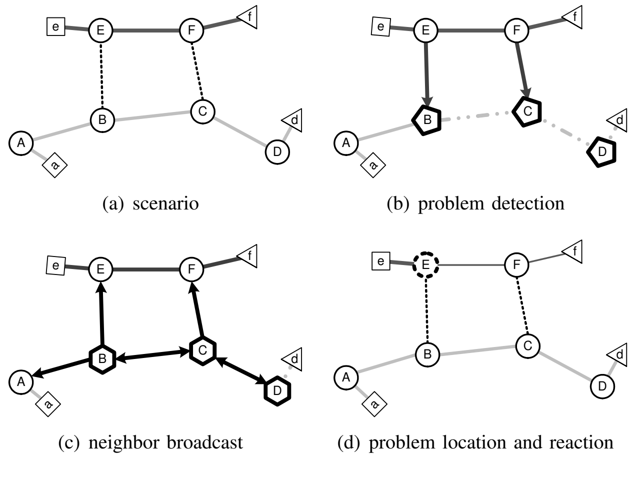
C. *Traffic Controller*

The second important unit of the mechanism is the so called *Traffic Controller*. So far, the possibilities of the *Traffic Observer* to detect a problem and its ways to give alerts have been presented. The remaining logical steps of the mechanism to solve quality problems are signaling the quality problems to other nodes in the *MeshBed* and to react on the disturbing influence to increase the quality. These tasks are realized by the *Traffic Controller* and are presented in this section.

1) Traffic controlling mechanisms: Quality degradation can occur for several reasons like packet loss, jitter, and long end-to-end delays. A common approach to decrease the packet loss and the jitter is packet prioritization using the type of service bit in the IP header. However, due to problems on the air interface caused by subsequent nodes when relaying traffic over multiple hops, a prioritization alone does not work in WMNs.

Considering the possibilities of automated and manual WLAN channel choice, it can be estimated that there are no external influences to the WMN on the air interface. All colliding packets are originating from one of the own mesh routers in the *MeshBed*. Under these circumstances a reaction to these collisions can be done by a reduction of the disturbing traffic's packet amount. By reducing the allowed bandwidth for non real-time traffic to a lower but still acceptable level, the frequency of possible disturbing packets is automatically decreased as well.

2) Steps of Controlling: Fig. 5 shows the steps of a *Traffic Controller* reaction in an example scenario inside the WMN environment displayed in Fig. 1. A constant bitrate real-time connection between *a* and *d* via *A-B-C-D* is disturbed by crossover high bandwidth traffic from *e* to *f* via *E-F*, see Fig. 5(a).



(a) scenario (b) problem detection
(c) neighbor broadcast (d) problem location and reaction

Fig. 5. Steps of controlling

The packets relayed from *E* to *F* and from *F* to *f* collide on the air interface with the packets relayed from *B* and *C* which results in a quality decrease of the real-time service, as illustrated in Fig. 5(b). The *Traffic Observer*s at *B*, *C*, and

D detect the quality problem and send an alert to their *Traffic Controller*s. At first the nodes try to find possible disturbances in their own queues. To avoid quality decrease caused by overloaded queues, all non real-time applications in the own node are checked first, if a certain bandwidth threshold is exceeded. If this is the case, the bandwidth of the non real-time applications is reduced to a predefined threshold. A dynamical stepwise adaptation of the bandwidth for non real-time traffic is an interesting topic to be researched and tested by simulation studies in future work. In the next step as neighbor nodes might cause crossover problems, like *E* and *F* in this scenario, signaling messages are sent to all one-hop neighbors via the *OLSRd* Hello Message system. This is shown in Fig. 5(c). All nodes receiving such a broadcast message of a disturbed node are as one-hop neighbors of the disturbed node possibly responsible for the disturbance. Therefore, they check and control the bandwidth of possible disturbing traffic the same way as the disturbed node did before. In the displayed scenario, *E* will activate the bandwidth control. *F* then recognizes that the bandwidth is already reduced and no further reaction is necessary. Fig. 5(d) shows the situation after the reaction of the mechanism. *E* is performing bandwidth control that leads to a slower but still working high bandwidth traffic from *e* to *f*. The performance of the real-time flows increases again and the QoS/QoE demands can be met.

V. Performance Measurements

To analyze the performance of the presented approach and to see if the user perceived quality can be kept on a constant and high level, two WMN scenarios are set up at T-Systems, see Fig. 2.

A. In-Band Traffic Disturbance

In the first scenario, shown in Fig. 6, the disturbing best effort flow has to use the same wireless link between mesh point *A* and mesh point *B*. We call this scenario the in-band scenario. The cause of a quality degradation should thereby directly be recognized by mesh point *B*.

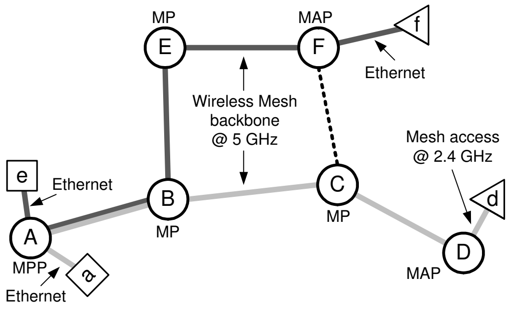


Fig. 6. In-band disturbing traffic

The real-time connection between *a* and *d* is realized by a VoIP connection, similar to the ITU-T G.711 voice codec, with an inter arrival time of 20 ms and a packet size of 200 Bytes. The bandwidth of the disturbing best effort connection between *e* and *f* is stepwise increased from 1 to 6 Mbps. Whenever the *Traffic Controller* detects a QoS degradation of the VoIP connection, the bandwidth of the disturbing best effort flow is decreased to 1 Mbps.

Fig. 7 and Fig. 8 successively present the results of measurements with deactivated and activated controlling mechanism. The x-axis shows the time of the measurement in seconds, the y-axes show the estimated MOS and the $loss$ in percent of the real-time traffic measured at *D* as well as the bandwidth in Mbps of the disturbing service measured at *F*.

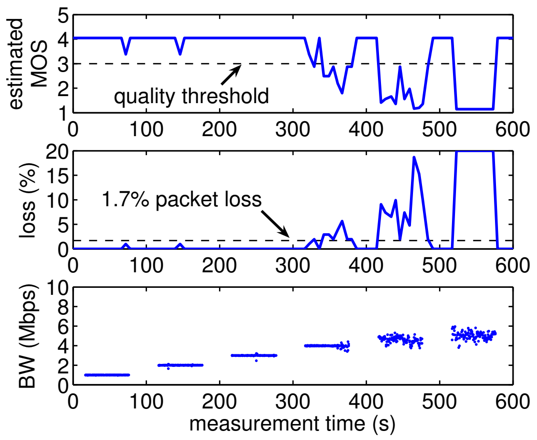


Fig. 7. In-band scenario without *Traffic Controller*

The std_{IPD} has also been measured at *D*. However, the measurements have shown that even for the highest disturbing bandwidth, this parameter still stays at an acceptable level below 5 ms. Therefore, it is not displayed in the measurement results. However, the packet loss has a large influence on the estimated MOS. Whenever the dashed line at 1.7% packet loss is crossed, the MOS drops below 3, resulting in a bad voice quality. This is already the case when the bandwidth of the disturbing best effort is increased to 4 Mbps. A further

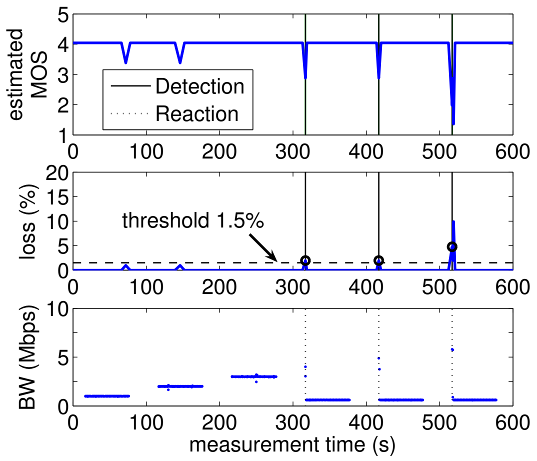


Fig. 8. In-band scenario with *Traffic Controller*

bandwidth increase leads to a packet loss of up to 20 percent and a MOS of 1.

However, if the *Traffic Controller* is activated, the MOS is kept on a high level as shown in Fig. 8. The vertical lines in the curves show the time of the problem detection and the time of the controller reaction. The first exceeding values alerted at the time of the detection of a new problem are marked with a circle in the $loss$ graph. The *Traffic Observer* threshold between average and bad $loss$ values is set to 1.5 % and displayed in the graph by a dashed horizontal line.

The functionality of our mechanism is most obvious after 420 s of measurements. At this point, the bandwidth of the disturbing traffic flow is increased to 5 Mbps which results in 6 percent $loss$ of the RTP packets. The problem is then detected by the *Traffic Observer* and the bandwidth for the best effort flow is reduced to 1 Mbps. Afterwards, the $loss$ decreases and the mean opinion score increases to 4 again.

B. Out-Band Traffic Disturbance

The second measurement scenario is shown in Fig. 9. This time, the RTP service from a to d is disturbed by subsequent crossover high bandwidth connections from e to f via E-F. This scenario is called out-band scenario. A reaction to a bad QoE is performed like shown in Fig. 5. However, this time the bandwidth of the best effort flow is reduced to 5 Mbps instead of 1 Mbps because the quality decrease does not originate from overloaded queues but from interferences on the air interface which have less influence on the voice traffic flow.

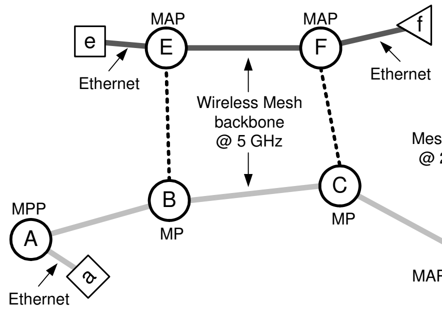


Fig. 9. Out-band disturbing traffic

In contrast to the first measurement scenario, the bandwidth of the disturbing best effort traffic can now be increased from 5 to 25 Mbps in steps of 5 Mbps, due to the above mentioned reason. In the first scenario, both flows share the same queues at the WLAN MAC layer at A and B. In the second scenario, the quality of the VoIP flow is just degraded by the interference on the wireless link.

Fig. 10 and Fig. 11 show the measurement results with deactivated and activated controlling mechanism. Similar to the previous results, the std_{IPD} is negligible and not plotted. Again, the $loss$ value is a lot more sensible to collisions on the air interface. Fig. 10 shows that the threshold is already exceeded for a disturber bandwidth of 10 Mbps. For disturber bandwidths of 20 Mbps and more, the quality remains always

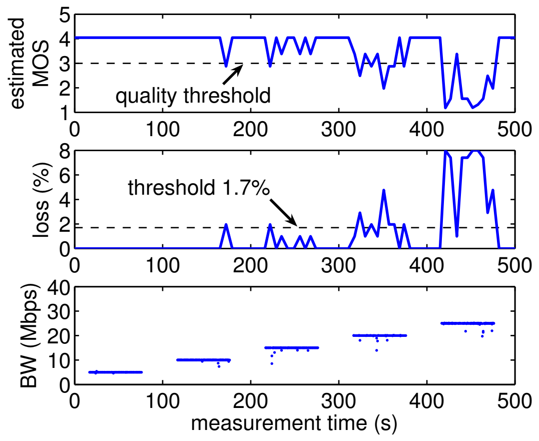


Fig. 10. Influences of crossover disturbers

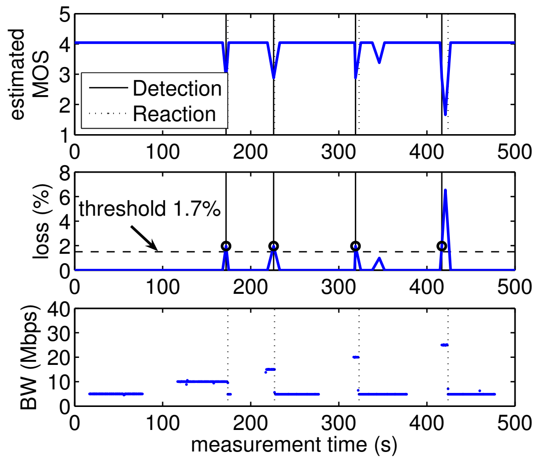


Fig. 11. Improvements by the *Traffic Controller* in the out-band scenario

below the threshold. For the highest tested bandwidth of 25 Mbps, the service quality at D is totally unacceptable as the $loss$ value increases drastically.

Fig. 11 shows the same case as Fig. 10 but with activated mechanism at all nodes. Obviously, as a first perception, the phases with high $loss$, invoking low MOS, are a lot shorter than without the influences of the mechanism. The bandwidth graph shows the reduction of the disturbers bandwidth to the configured value of 5 Mbps. This obviously leads to a direct return to acceptable quality values in the $loss$ and MOS curves.

To quantify the performance of the mechanism, the key parameters, reaction time and signaling message load, have been analyzed. Depending on the number of neighbors a mesh router in the depicted scenario receives on average between 400 Byte, about 3 to 4 packets, and 2000 Byte, 15 to 20 packets, of *OLSRd* messages per second. As said before, the *Traffic Observer* does not send alerts more frequently than with an interval of 1 second to avoid an alert flooding. An alert is furthermore broadcast by an *OLSRd* message of a size fitting in one single *OLSRd* packet. This one additional packet per second does not show any increase of the average *OLSRd* signaling bandwidth. Even the highest measured *OLSRd* signaling bandwidth of 2 kbps is ignorable even in a highly loaded network. The signaling load issue is thus no problem of the presented mechanism.

The second important metric to quantify the mechanism's performance is the reacting time. As upcoming quality loss is recognized latest within the first w disturbed packets, i.e. in the default case with $w = 100$ and constant bitrate $20\,ms$ in the first two 2 seconds, the delay between the occurrence of a quality decrease and the recognition can be disregarded. Then again an activation of the *Traffic Controller* e.g. reducing the disturbers bandwidth is supposed to solve the problem in maximally w packets as well, what can be confirmed by a look at Fig. 11. The time between the activation of the *Traffic Controller* and the return of an acceptable quality level is thus also negligible. The scope lies on the delay between the detection and the reaction. Fig. 11 shows that this delay depends on the bandwidth of the disturber. For the bandwidths of 5, 10, 15 Mbps the delay remains between 1 and 3 seconds. Such a delay results only in a short QoS loss which is still acceptable for a user.

For the test cases with higher bandwidths of 20 Mbps and 25 Mbps, which are though not expected to occur in real mesh networks, the delays increase significantly up to 7 seconds. An analysis of the single controlling steps has shown that the high delays in the measurement setup are mainly caused inside the *Traffic Controller* while activating the traffic reduction. The delays are due to high CPU use of the used mesh points. In this case the prerequisite of Section IV that the nodes can be chosen fast enough to not be the bottleneck of a transmission is not met anymore with the used equipment. The effects are though expected to disappear when more powerful machines or a hardware based realization are used. Investigating such a realization might be a promising topic for future work.

VI. Conclusion

In this paper, we presented an approach to meet a high Quality of Experience level for real-time applications in wireless mesh networks. The approach is based on two main entities, a *Traffic Observer* and a *Traffic Controller*. The *Traffic Observer* concurrently measures the network situation. Whenever a problem is detected in the wireless mesh network, for example a high rate best effort flow blocks a real-time application, the *Traffic Controller* forces this low priority flow to reduce the bandwidth.

In contrast to other publications in this area, the developed approach was not just tested in a simulation environment, but implemented in a real WLAN-based mesh network. Two different scenarios have been investigated, one with disturbing best effort traffic on the same path and one scenario with disturbing traffic within the coverage area of the mesh points. The results have shown that without the *Traffic Controller*, the subjective quality, expressed in the mean opinion score, decreases drastically when only a small disturbing bandwidth is set up. However, when the *Traffic Controller* is activated, the MOS only drops for one to three seconds below 4.

Comparing the two scenarios, it can be said that the real-time application is by far more influenced by a best effort flow on the same path than on a crossing path. This is due to queuing effects on the MAC layer. The next step is to reduce the best effort bandwidth not to a fixed value, but to automatically adapt it to the maximum possible bandwidth without disturbing the real-time traffic flows.

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