VoIP service performance optimization in pre-IEEE 802.11s Wireless Mesh Networks

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***Abstract*—802.11-based Wireless Mesh Networks are seen as a means for providing last mile connections to Next Generation Networks. Due to the low deployment cost and the mature technology used, they are scalable, easy to implement and robust. With an increasing coverage of wireless networks, VoIP becomes a cheaper alternative for traditional and cellular telephony. In this paper, we carry out a feasibility study of VoIP in a dual radio mesh environment. Heading towards 802.11s, we present the design of a mesh testbed and methodology for performing the measurements. Additionally, we address the problem that small voice packets introduce a high overhead leading to a low voice capacity of 802.11 based mesh networks. In order to alleviate this problem and increase the voice capacity, a novel packet aggregation mechanism is presented and evaluated using the ns-2 simulator.**

I. Introduction

Wireless Mesh Networks (WMN) are gaining attention as a cost-efficient way for providing broadband wireless Internet access. The IEEE 802.11s task group is aimed to form a transparent 802.11 broadcast domain with the same functionality as its wired counterpart. Hence, it is supposed to support the protocols located at higher layers as well as to perform frame forwarding and path selection at OSI Link Layer. Recently, there has been a lot of research done in WMNs. Most of these activities are based on simulations, which provide an appropriate means for optimization and detailed analysis. However, to gain a basic understanding about the behavior of WMNs, measurements and experiences derived from testbeds in realistic scenarios are essential. There are already several testbeds developed, like Roofnet [1], UCSB Meshnet [2] or MCG-Mesh [3]. In this paper, we present a testbed designed and deployed by T-Systems in Darmstadt, Germany. As a part of the Triple Play bundle, Voice over IP (VoIP) was chosen for performance tests.

Multi-hop WMNs have several benefits. In comparison to infrastructure networks with single wireless links, multi-hop WMNs can extend the coverage of a network and improve the connectivity. The number of fixed Internet access points can be reduced leading to a cheaper network access as several users share Internet connectivity by multi-hopping towards the access routers. Multi-hop WMNs avoid a wide deployment of cables and can be rapidly deployed in a cost-efficient way. In case of dense multi-hop networks, the use of multi-radio multi-channel mesh nodes increases network capacity, and therefore several paths might become available increasing the network's robustness.

The provisioning of VoIP in multi-hop WMNs is an important service for the future wireless Internet. However, VoIP service poses new challenges when deployed over a multi-hop WMN. Packet losses and an increased delay due to interference in a multiple hop network can significantly degrade the end-to-end VoIP call quality. High traffic leads to high medium contention which increases packet loss rates compared to single hop deployments. The existence of potential hidden nodes further intensifies this problem. Moreover, the transmission of small (voice) packets imposes a high MAC layer overhead, which leads to a low capacity for VoIP over IEEE 802.11-based WMNs.

Several studies of VoIP in mesh networks concentrate on the analysis of the impact of multiple hop on VoIP performance [4], [5]. The impact when using multi-radio multi-channel techniques is not well exploited, however. The availability of several radio interfaces provides scalability to the system, while the availability of several channels across the mesh networks provides frequency diversity [6]. Hyacinth [7], a multi-channel multi-radio architecture, presents a distributed channel assignment algorithm that adapts to traffic loads, and shows through testbed experiments that the aggregate throughput of multiple FTP sessions may be increased by a factor of 5. The impact of switching channel cost over multi-radio for UDP and TCP traffic is investigated in [8], however such impact on multimedia traffic (e.g. VoIP) is not taken into account. Research on improving VoIP scalability in WMNs by employing multi-radio multi-channel is presented in [9]. By using 2 radio interfaces and 3 independent channels in 802.11b, interference reduction and path diversity are achieved, and consequently a greater number of calls can be supported.

The enhancement of the VoIP capacity in WMNs by aggregating packets is studied in [9], [10], [11], [12], [13] and [14]. While trying to reduce the IEEE 802.11 MAC overhead, different techniques were applied, such as end-to-end, hop-by-hop, and hybrid aggregation schemes. As an example, the proposed accretion (hybrid) aggregation algorithm in [10] proved to increase the number of supported calls with the given quality measured over single-radio single-channel mesh networks. In such a scheme, the aggregation is done at the ingress node for all flows routed to a common destination. The medium access queuing delay of intermediate nodes is used for a further aggregation without imposing an extra delay to the packets. The mechanism proposed in [14] adapts the size of aggregated packets to the quality of wireless channel as smaller packets lead to less packet loss for low quality links. In addition, header compression schemes such as robust header compression (ROHC) are presented in [10] and [15] as a complementary technique to aggregation, while increasing VoIP scalability over mesh networks.

The rest of the paper is organized as follows. In Section II, we describe the design of the testbed and show some experimental results. In Section III, we propose a new packet aggregation scheme for VoIP traffic and in Section IV we demonstrate the efficiency of the scheme through simulations. Finally we discuss future work in Section V and draw a conclusion on VoIP traffic in WMNs in Section VI.

II. MeshBed architecture and test results

This Section handles the MeshBed which is a next generation WLAN based Wireless Mesh Network, developed and deployed at T-Systems in Darmstadt, Germany. In its current state the MeshBed consists of 10 Mesh Router Nodes (MRNs) and 2 Mesh Gateways (MGWs) that are all deployed indoors. As hardware platform an embedded AMD Geode SC1100 Systems with 266 MHz CPUs and 64 MB of RAM is used. For nodes that require more processing power, e.g. MGWs, bare bone desktop PCs with 3 GHz Intel Pentium 4 processors and 1 GB of RAM are used. All mesh nodes are equipped with Atheros Wireless Mini PCI WLAN cards as well as Ethernet ports and use operating systems based on Linux together with "madwifi" [16], an open-source WLAN driver.

This mesh environment is designed in accordance with a strategy towards 802.11s, which assumes the usage of advanced MAC technique, namely link layer routing [17]. Currently, packets are still routed at the network layer. The testbed emulates dual-radio feature, we call it pre-IEEE 802.11s.

The MeshBed architecture is depicted in Figure 1 and consists of an access and a backbone network. The backbone network operates at the 5 GHz frequency band. This backbone network is used for the communication between mesh nodes and for forwarding user traffic towards the Internet. The Optimized Link State Routing protocol (OLSR) [18], based on the implementation of Andreas Tonnesen [19], is deployed as the routing protocol in the backbone. The Ethernet port on the MGWs acts as the gateway to the Internet.

The access network is used to connect users to the WMN in order to provide Internet access. This access can be either wired via the Ethernet interface of each MRN or wireless. The second WLAN card is configured to act as an access point operating in the 2.4 GHz frequency band. Due to the separation of backbone and client traffic, the user terminals do not need any mesh specific functionality. From the users' point of view, there is no difference whether mesh or standard wired backbone network is used. More details about the MeshBed can be found in [20].

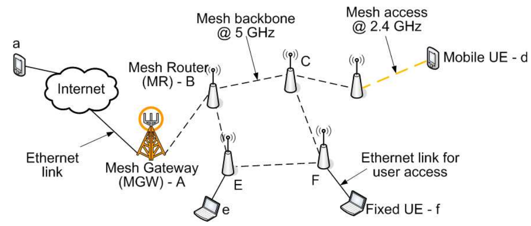

Fig. 1. MeshBed architecture

In the following we present results from VoIP performance measurements with the aim to detect and to analyze possible problem sources arising from transporting VoIP traffic over WMNs. In a first experiment the performance of an undisturbed VoIP call over multiple wireless hops is investigated. The experiment resulted in an acceptable voice quality. On the five hops path, e.g., the resulting end-to-end delay did not exceed $t = 5\,ms$ while the corresponding jitter stayed below $1\,ms$.

A second experiment was designed to investigate the sensitivity of VoIP traffic to an increasing load in the WMN. Due to the nature of mesh networks, different sources for quality degradation are possible. Among others, these include packet collisions on the air interface as well as overloaded queues in the MeshBed nodes. Key parameters like mean inter packet delay, packet loss and bandwidth are measured at each hop to state precisely at which point of the network the quality decrease originates and in which way it becomes visible.

In a second experiment, the impact of background traffic is investigated. In the scenario presented in Figure 1 a VoIP call from node A to D over the three hop path A-B-C-D is disturbed by traffic on the one hop route E-F. Note that hop E-F is in the same collision domain as B-C and thus the traffic flows compete for the access. The bandwidth of the interfering background traffic is linearly increased. The results of the experiment are shown as traces in Figure 2. The upper two graphs depict packet loss rate and jitter expressed by the standard deviation of the inter packet delay at node D. The lowermost graph shows how the bandwidth of the disturbing flow from E to F increases during the experiment. The curves provide a practical quantification of the theoretically expected problems on the air interface. Obviously in this scenario the packet loss ratio is influenced more by disturbing traffic than the jitter. An increasing number of packets is thrown away, when the sending attempts fail. The packets that are delivered, still arrive without any bigger change of inter packet delay. During the first 500 seconds, the cross over traffic has no significant impact on the VoIP connection until it exceeds a certain threshold.

In a third experiment the performance improvement of dual radio over single radio WMN deployment in noisy environment has been evaluated. To introduce interference an external traffic was generated on the same channel in which

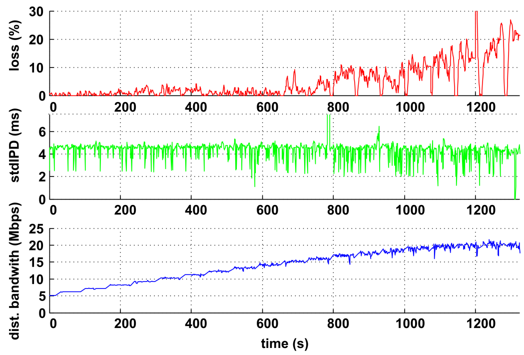

Fig. 2. Influences of Cross Over Disturbers

the backbone tier is operating. In a single radio scenario, access tier, backbone tier and interfering traffic were utilizing the same channel. In a dual radio scenario non-overlapping channel was set for an access tier. To obtain the results presented in Table I, a set of 3 test runs with varying packet size was performed over 3 hop path in single/dual radio environment. Each test run lasted for 10 minutes.

	Single Radio			Dual Radio		
Packet Size [B]	60	500	1470	60	500	1470
Packet Loss [%]	3	5	9	3	4	6
Delay [ms]	12.6	14.3	24.2	12.7	12.9	16.5
Jitter [ms]	10.5	14.7	25.9	8.7	6.3	9.0

TABLE I
DUAL RADIO VS SINGLE RADIO IN NOISY ENVIRONMENT

According to the results obtained, dual radio brings significant gain in delay, jitter and Packet Loss Ratio (PLR). In the case of the packets of a small size (single VoIP packets) dual radio does not bring valuable improvement in comparison to single radio. With increasing size of the packet (aggreagation of VoIP packets from different flows), an effect of dual radio becomes stronger. Therefore, introduction of dual radio along with packet aggregation techniques will bring improved performance of VoIP in WMNs.

III. AGGREGATION SCHEME FOR VOIP PACKETS

VoIP overhead reduction is critical to increase capacity and meet the customer demands. Packet aggregation aims to combine several small-sized packets into a single one. Figure 3 illustrates the proposal for an aggregation scheme, that significantly reduces MAC and PHY layer overhead. The upper part of the figure depicts timing in normal DCF basic access. DATA is assumed to be composed of RTP, UDP and IP headers along with compressed voice samples. The lower part of the figure presents the idea of packet aggregation where aggregated packet contains three original IP packets. Transmitting bigger packets reduce MAC contention and medium occupation, and also increase the throughput. The drawback of this solution is the introduction of an additional delay caused by holding several packets for aggregation before forwarding and by the aggregation process itself.

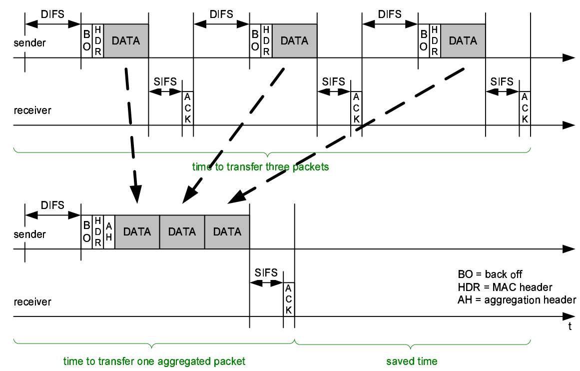

Fig. 3. Principle of Packet Aggregation

Now our hop-by-hop aggregation algorithm is presented. It aggregates the packets at IP layer and is able to adapt to different network conditions and traffic characteristics. The algorithm should not increase the delay unless it provides a good aggregation ratio. These properties can be controlled by three configurable parameters: $SIZE_{min}$ and $SIZE_{max}$ specify the maximum and the minimum size of an aggregation packet, and MAX_{delay} denotes the maximum forced delay.

At each hop, incoming packets are marked with a timestamp and put into the queue. The aggregation algorithm creates an aggregation packet as soon as the MAC layer becomes idle. Potentially, all packets with common next hop may be aggregated as long as they do not exceed $SIZE_{max}$. If only the cumulative size of packets becomes greater than $SIZE_{min}$, the packets are aggregated and passed to the MAC layer. If $SIZE_{min}$ is not reached, only packets older than MAX_{delay} are aggregated. If none is older, nothing is sent. If exactly one packet is older, it is sent as it is without additional aggregation header. There have be at least two packets older than MAX_{delay} in order for an aggregation to takes place.

Aggregation adds an additional 20 bytes IP header to each aggregated packet. The protocol field in the new IP header is set to IP_META (or the corresponding numerical value) so that the node can recognize aggregation packets. Destination address is set to the next hop. Deaggregation is done by inspecting the first aggregated IP packet - P0, calculating the offset of the next IP header and so forth. Every intermediate node is assumed to be capable of aggregate and deaggregate packets.

The parameter $SIZE_{min}$ should ensure that the ratio of overhead and payload (the aggregated packets) remains small. If $SIZE_{min}$ is too large, it may result in many packets being sent without aggregation. On the other hand, it needs to accommodate at least two packets. $SIZE_{max}$ must be smaller than the MTU minus 20 bytes. MAX_{delay} denotes the maximum forced delay that a single packet may experience for aggregation. In case of low network traffic, this parameter causes some artificial delay and increases the aggregation ratio. As VoIP is time-critical, the value of MAX_{delay} should be kept low.

IV. Performance evaluation and discussion

In order to evaluate the proposed aggregation algorithm, this section presents simulation results obtained with ns-2.26 [21]. The network topology, depicted in Figure 4, comprises both wired and wireless nodes.

Node 0 represents a server connected with a Fast-Ethernet link to the router Node 1, which itself has a wired Ethernet connection to Node 2. Node 2 is a Mesh Gateway connecting the wired and wireless network. Node 3 and 4 are Mesh Relay Nodes, which only forward traffic inside the mesh network. Node 5 and 6 are Mesh Relay Nodes where clients are connected.

IEEE 802.11a DCF without RTS/CTS mechanism is used by wireless nodes. As stated in [22], it may not always be beneficial to be used together with packet aggregation in VoIP. The basic rate is set to 6 Mbps, the data rate to 24 Mbps.

Communication between neighbors can be affected by transmissions in a different way depending basically on the distance between nodes and the packet length. The effect of bit error rate (BER) and signal-to-noise ratio (SNR) over packet loss are considered according to [23]. Each simulation run lasts 180 seconds. We used AODV-UU [24] as the basic routing protocol to provide connectivity for mesh relay nodes. AODV-UU is used in half-tunneling mode, which adds an encapsulation header for all packets forwarded towards the Internet. Our packet aggregation mechanism can be deployed with any routing protocol more suitable to a multi-channel, multi-radio mesh scenario.

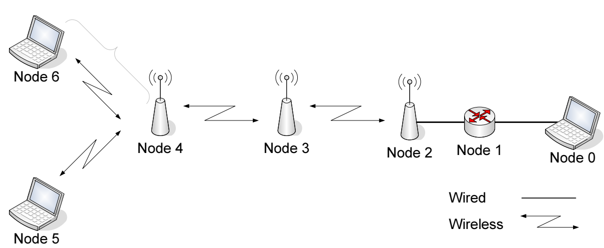

Fig. 4. Simulation topology

All MRNs can aggregate and deaggregate the traffic. $SIZE_{min}$ is set to 300 bytes, $SIZE_{max}$ to 2302 bytes and MAX_{delay} to 10 ms. All MRNs are stationary. The distance between the nodes is 45 m, and the VoIP traffic is exchanged between Nodes 6 and 0, as well as between Nodes 5 and 0, and in the respective reverse direction. The traffic is generated and analyzed with the ns-2 VoIP-extension presented in [25]. ITU G.729a with Voice Activity Detection is used as speech codec. Accordingly, 50 packets per second of 60 bytes including RTP/UDP/IP-headers are sent during talk spurts. At the receiver an adaptive playout buffer copes which jitter. Based on packet loss ratio and end-to-end delay after the playout buffer the Mean-Opinion-Score (MOS) is calculated. The MOS is a quality measure promoted by the ITU-T [26]. A MOS of 5 can be interpreted as "Excellent Quality" and 1 as "Bad Quality". A MOS of 3.5 is considered to be "Satisfactory Quality".

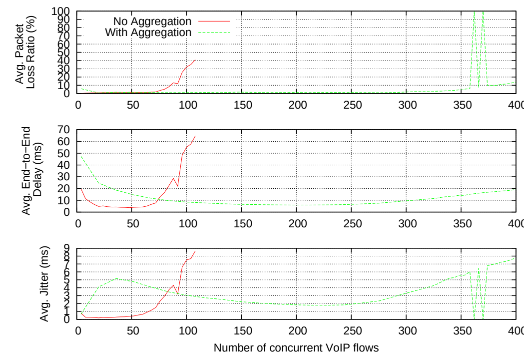

Fig. 5. PLR, delay and jitter of VoIP flows

Subsequently simulation results are presented through the evaluation of the capacity and performance in terms of "supported VoIP flows" for "no aggregation" and the proposed aggregation algorithm. First we look into packet loss ratio, delay and jitter of all flows. Then we investigate the MOS and regard a flow as "supported" when its average MOS over the simulation period is greater than 3.5. We use this information to calculate a performance metric "VoIP capacity", which we define as the maximum number of concurrent flows, such that at least 95% of the injected flows are supported.

Figure 5 depicts average values of packet loss ratio, delay and jitter versus the number of injected flows without and with the proposed aggregation. Results are averaged across all flows. The diagram shows that below a certain threshold the number of injected flows can be increased while keeping the QoS parameters within acceptable boundaries. Above this threshold, packet loss, delay and jitter increase unacceptably. Without aggregation this threshold is about 80 flows. If aggregation is used, it is about 354 flows. However, in low traffic scenarios, aggregation may lead to higher delay and jitter. Here, some packets wait until MAX_{delay} is reached in order to get aggregated, and jitter increases. In case the MAX_{delay} is very low, aggregation would act similarly as if no aggregation were used.

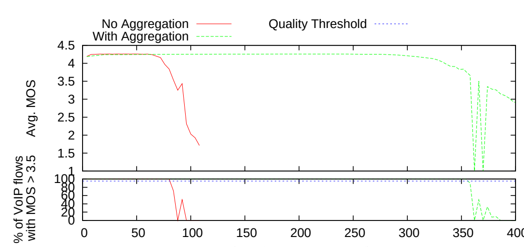

Fig. 6. Average MOS and number of supported flows

The upper part of Figure 6 shows that the average MOS over all flows is constantly at about 4.3, whereas "no aggregation" is a bit better in low traffic scenarios due to its lower end-to-end delay and jitter. When the traffic load is increased, the

MOS begins to drop at about 80 flows with no aggregation and 354 flows with aggregation. This reduction of the MOS is mainly a result of the increased packet loss. The packet loss is caused by collisions and the incapability of the MAC layer to serve more traffic. Since the packet loss ratio and delay might vary among flows, it is also necessary to investigate the quality of individual flows. Therefore the lower part of Figure 6 displays the percentage of supported flows with respect to the injected flows. With ”no aggregation” the number of injected flows can be increased up to 80, while still all flows are supported. However, if only 4 more flows are added, only 60 flows have an average MOS greater than 3.5. Thus the VoIP capacity is 80 flows. With 88 flows no flow has an MOS greater than 3.5. With the use of aggregation the capacity can be raised to 354 flows. Thus the capacity is increased by a factor of 4.4 here.

V. Conclusions

The paper addresses the deployment VoIP service in pre-IEEE 802.11s WMN and means for its performance optimization. VoIP, being a part of Triple play service bundle, was chosen as a reference service for extensive measurements. The general finding of the experiments is, that VoIP can be supported with good quality in mesh environment. However, under high load, quality drops and additional mechanisms are needed to overcome these problems. Moreover, it was demonstrated how the VoIP traffic may benefit from the small packet aggregation. A novel hop-by-hop packet aggregation mechanism was proposed. It significantly improves the performance of VoIP traffic in WMNs and reduces MAC layer busy time.

The factors causing VoIP quality drop in highly loaded networks are to be identified and discussed in detail at further steps. Finding the right packet size for aggregation is a complex task and is left for future work. Therefore, the algorithm will be revised to make it more adaptive to channel conditions and multi-hop contention. Future improvements will also include comparison of packet aggregation performance over a greater variety of scenarios and implementation of more realistic mixed traffic scenarios, including TCP.

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