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Provisioning Call Quality and Capacity for Femtocells over Wireless Mesh Backhaul

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Abstract—The primary contribution of this paper is the design of a novel architecture and mechanisms to enable voice services to be deployed over femtocells backhauled using a wireless mesh network. The architecture combines three mechanisms designed to improve Voice Over IP (VoIP) call quality and capacity in a deployment comprised of meshed femtocells backhauled over a WiFi-based Wireless Mesh Network (WMN), or *femto-over-mesh*. The three mechanisms are: *(i)* a Call Admission Control (CAC) mechanism employed to protect the network against congestion; *(ii)* the frame aggregation feature of the 802.11e protocol which allows multiple smaller frames to be aggregated into a single larger frame; and *(iii)* a novel delay-piggy-backing mechanism with two key benefits: prioritizing delayed packets over less delayed packets, and enabling the measurement of voice call quality at intermediate network nodes rather than just at the path end-points. The results show that the combination of the three mechanisms improves the system capacity for high quality voice calls while preventing the network from accepting calls which would result in call quality degradation across all calls, and while maximizing the call capacity available with a given set of network resources.

I. INTRODUCTION

Wireless access technologies, such as WiFi, have increased in popularity due to the widespread use of smart-phones. The amount of data traffic has surpassed voice traffic driving both the Mobile Network Operators (MNOs) and wireless access standardization bodies to adapt accordingly. Widespread Long Term Evolution (LTE) deployments are being undertaken in an effort to cope with these increased traffic demands, while in parallel, operators are looking to complementary technologies, such as femtocells and WiFi, in an effort to reduce congestion on the cellular radio access networks.

Reducing the radius of cell towers is a possible solution to increase the performance of cellular networks and increase frequency reuse [1]. Femtocells are small access-point-sized devices with much smaller coverage areas when compared to typical cellular base stations. Unlike traditional base stations, femtocells are backhauled over normal Internet Protocol (IP) connected networks such as residential Internet connections.

The focus of this work is to increase the cellular capacity in a defined region where femtocells are deployed. This is particularly beneficial for transient deployments; such scenarios include, but are not limited to, social events where attendees need access to voice and data services at venues which are used for a relatively short period of time. In such scenarios, the networking infrastructure needs to be flexible and quickly deployable.

A backhaul solution where each femtocell is provided with wired infrastructure to forward its traffic to the Evolved Packet Core (EPC) is not feasible as wired deployments tend to pose serious logistical problems. This work proposes a new deployment scenario which utilises a WMN backhaul infrastructure for femtocells, in order to overcome the deployment difficulties imposed by wired solutions. WMNs have become increasingly popular due to their capabilities: relatively large coverage areas with minimal cabling requirements, quick deployment, fair price, and ease of maintenance.

This paper considers the usage of a WiFi-based WMN infrastructure, where the mesh nodes are hybrid stations featuring multiple WiFi mesh interfaces and one LTE femtocell embedded or co-located with the mesh node. Figure 1 depicts the scenario this work is focused on. It can be seen that clients roam with their User Equipment (UE) in an area serviced by a grid of femtocells backhauled over a WMN network into the femtocells' network operator's core network.

The WMN access-medium's capacity saturates when traffic demand is high, resulting in high latency and packet loss causing the performance of services to degrade. A special case of such services are voice calls. In LTE, both data and voice traffic payloads are encapsulated in IP packets, hence the voice calls in an LTE network are VoIP calls.

WMNs can be highly unpredictable in terms of Quality of Service (QoS) offered to VoIP calls, hence the VoIP QoS on WMNs degrades rapidly when only one call more than the system's capacity is added. In such situations detecting the capacity barrier is critical to ensure ongoing calls are guaranteed with satisfactory QoS levels. This work compares the individual and combined effects of three mechanisms designed to improve the call quality and capacity in a femto-over-mesh deployment scenario: *(i)* a CAC mechanism which uses samples of the calls' quality to determine when to restrict the addition of new calls into the network; *(ii)* the frame aggregation feature of the 802.11e wireless standard, allowing the aggregation of multiple small frames into a single frame, thus reducing the delays induced by transport protocol's

Figure 1: Topology of WMN-backhauled femtocells

overhead; and *(iii)* a proposed delay-piggy-backing mechanism which attaches to every VoIP packet, while inside of the WMN, the cumulative delay experienced as it propagates through the network — this enables the scheduler to choose from a Push-In-First-Out (PIFO) queue more delayed packets first, and the WMN gateway to compute call quality samples based on which the CAC mechanism takes actions regarding new call requests.

The CAC mechanism resides in the Local Femto Gateway (LFG) [2] which is an entity placed in the neighbourhood of the WMN's gateway. Simulation results show that a CAC mechanism is necessary in such scenarios due to the fact that uncontrolled additions of voice calls into the network will negatively impact all existing calls. The overall call quality and network VoIP capacity are the two performance indicators analysed in this work.

The remainder of this paper is structured as follows. In Section II, the relevant and related work is discussed. This is followed by Section III which provides an overview of the Mean Opinion Score (MOS) and E-Model concepts, a description of the LTE femtocell architecture, and the solutions proposed in this work. Section IV details the simulation settings followed by the presentation of the simulation results in Section V. This paper is then concluded in Section VI.

II. RELATED WORK

CAC is not a recent concept as it applies to any system where reaching capacity is a concern. However, Wei et al. [3] addressed the issues related to employing CAC in a WMN serving VoIP as the main service for users. The possibility of aggregating packets to improve the overall system's capacity is also presented in [3].

The concept of calculating an Intermediate Mean Opinion Score (iMOS) was introduced in our previous work [4]. In that work, the authors assumed the intermediate node is time-synchronized with one of the end-points, hence the packet delay could be measured in order to obtain the iMOS.

In [5] a possible implementation of PIFO queues is presented together with a performance analysis. As PIFO queues

require more computational power during the *push-in* phase than normal First-In-First-Out (FIFO) queues, their performance evaluation showed that current hardware is able to cope with the increased processing power imposed by PIFO queues.

In [2] a solution to offload the mobile core network is presented together with a proposed architecture addressing an enterprise femtocell deployment. The work presents the concept of the LFG entity where a proxy Mobility Management Entity (MME) and a proxy Serving Gateway (S-GW) are used to offload the core network by locally managing functions related to mobility and data access.

This work combines all the solutions mentioned above and further extends their scope by coupling them with a novel method of prioritizing VoIP packets and obtaining the iMOS.

III. ARCHITECTURE DESCRIPTION

A. LTE Architecture and LFG concept

The LTE security requirements specify that all traffic exchange between a femtocell and the EPC needs to be transported over IPsec tunnels. The tunnel terminates at the entry point of the EPC, which in Figure 1 is depicted as the Home *e* Node B Gateway (HeNBGW).

Inside the EPC, the MME terminates the control-plane signalling from femtocells and UEs. The S-GW handles the data plane from the UEs. The PDN Gateway (P-GW) is the anchor point of the UEs to the Internet and manages the IP addresses domain.

One of the roles of the MME is to manage the bearers and connections with the UEs, which further include the acceptance of new calls and potentially dropping calls when such action is required. Our proposed CAC mechanism needs to have access to these functions in order to maintain the QoS level for the VoIP calls inside the WMN.

However, these functions are not accessible to the WMN operator. Zdarsky et al. [2] have shown in their proposed architecture that these functions could be assigned to a LFG entity. In their proposal, the LFG intercepts control messages using its Proxy-MME function. This enables the LFG to take

actions regarding new or existing calls, based on the iMOS reported by the WMN.

B. iMOS-based CAC

The International Telecommunication Union-Telecommunication Standardisation Sector (ITU-T) report P.800 [6] introduces a general scoring system to assess the quality of a speech transmitted via telephone lines, on a range from 1 to 5. Human subjects grade the speech quality using this scale, the result being the Mean Opinion Score (MOS).

This work uses an estimative model which does not involve human subjects in rating speech quality. The most popular model for the estimative speech assessment is the E-Model (ITU-T G.107 [7]) which is based on transmission parameters and is widely accepted as an accurate tool for transmission network planning.

The E-Model was initially not designed for real-time measurements, but its adaptation to VoIP calls [8] allows the calculation of MOS scores on-the-fly. The E-Model was designed to measure the MOS on an end-to-end basis where the packet delay can be estimated as the half of the Round Trip Time (RTT) obtained form Real-Time Control Transport Protocol (RTCP) packets. When trying to measure the MOS at an intermediate point in the path, even the rough RTT delay estimation is not possible. Time synchronization protocols such as Network Time Protocol (NTP) can synchronize the intermediate point with the end-points, hence the intermediate nodes would be able to precisely estimate the network delay. However, the synchronization solution is not viable, as usually the end-points do not support synchronization from unknown network entities.

In this work, we enable the WMN gateway to obtain an accurate measurement of the packet delay from the delay-value attached to each VoIP frame coming from the WMN. This accurate delay measurement enables the WMN gateway to measure the call quality. Since the WMN gateway is an intermediate point in the calls' path, we name the MOS obtained in the WMN gateway as iMOS.

The WMN gateway keeps track of all ongoing VoIP calls and their corresponding iMOS scores. The iMOS scores used by the mechanisms presented in this work are obtained only from the up-link traffic. However, the mechanism can be employed also on the down-link, but would require each WMN node to be iMOS-aware and to have the means to inform the WMN gateway about possible issues. Both of these assumptions unnecessarily increase the load on the WMN nodes and on the network traffic itself.

The iMOS values obtained from the up-link voice packets are the basis of the CAC mechanism. CAC can be invoked automatically or manually, allowing a network operator to prevent extra traffic from being injected in the network when certain conditions occur. Usually, the conditions revolve around maintaining a pre-established QoS level.

The CAC mechanism is controlled by the LFG based on call quality reports received from the WMN gateway, i.e. the WMN node in Figure 1 equipped with a wired interface.

This work assumes the usage of a LFG placed in the architecture as indicated by Figure 1. The LFG is able to intercept and interpret signalling messages between the femtocells and the EPC. In this way the CAC mechanism can take actions regarding new and existing calls.

As VoIP applications generate tens of packets per second, it would be infeasible to take a CAC decision based on every VoIP packet traversing the WMN gateway. However, the WMN gateway calculates the iMOS for each traversing VoIP packet and places the value into an accumulator. Periodically the CAC mechanism, residing on the LFG, will request the WMN gateway to provide one aggregated value which represents the overall iMOS of all ongoing calls during the last period. If that value is under a certain pre-established threshold, then CAC is activated and new call requests are rejected.

We define the network's capacity as being identified when the addition of the most recent call causes the average iMOS value across all calls to drop below some threshold. In addition to activating the CAC mechanism when the overall iMOS value falls under the threshold, the CAC mechanism in our scenario, drops the call with the worst iMOS. A few calls are dropped in this manner until the network reaches a steady state after it was destabilized by the extra call accepted above the capacity.

As the CAC mechanism periodically checks the aggregated iMOS value, it is important to determine a proper rate at which this checking is done. According to [8], human subjects are able to determine a quality drop within 5 seconds. Taking that into account, we used 1 second for the CAC loop interval, allowing the mechanism to drop a few calls, hence restoring the actual call quality before the user can detect it.

C. Delay-Piggy-Backing based Priority Scheduler

This work uses a packet scheduler which prioritizes more delayed VoIP packets over less delayed VoIP packets. In conjunction with the frame aggregation, the proposed scheduler levels the delay distribution over all VoIP calls so that the quality variation between calls decreases thereby increasing fairness.

In order to achieve this delay-based prioritization, the queueing delay, expressed in time units, is attached to each WiFi frame. In case of multiple aggregated frames, the delay value is attached to each individual frame. The value is additive, thus next hops will add to it the queueing delay suffered while waiting in their queues. When packets are placed in the queue, the delay value is used to find the proper location of insertion, assuming the WMN node uses PIFO queues.

In a PIFO queue, packets are enqueued in the *push-in* phase at a specific location based on a comparison criteria. In our work, the criteria is the piggy-backed delay value. Specifically, the current packet's cumulative delay is compared against that of each packet's in the queue, starting with the head of the queue. The newly arrived packet will take the position in the queue where its cumulative delay is for the first time bigger than that of the packet's used for comparison.

In Figure 2, a 35 seconds sample of a voice call shows the absolute difference between the actual and the estimated delay
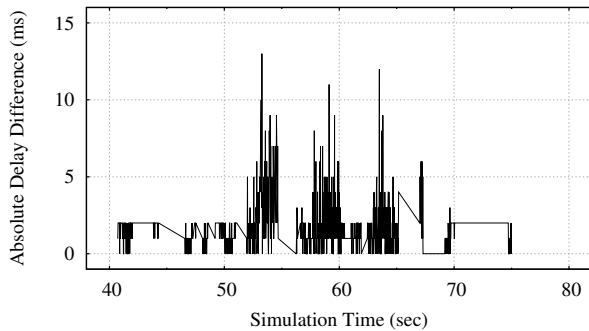
Figure 2: Validation of delay measurement precision.

values. The actual values were obtained from the simulator, by computing the difference between the arrival time and the sending time (only possible in a simulation environment). The estimated delay is extracted from the value attached to every packet. There is a high degree of correlation between both values with a variation typically lower than 5 milliseconds which is an insignificant amount when related to its influence on the MOS. This demonstrates the ability of the piggy-backed-delay solution to provide accurate estimations of network packet delay without requiring any time synchronisation, and furthermore it enables the WMN gateway to calculate accurate iMOS values for ongoing calls.

QoS support can be activated by enabling the 802.11e [9] protocol. This protocol defines four Access Categories (ACs) as follows: AC_BK for *Background* traffic, AC_BE for *Best Effort* traffic, AC_VI for *Video* traffic, and AC_VO for *Voice* traffic. In this work, we enabled the delay-piggy-backing prioritization only in the AC_VO queue.

IV. SIMULATION SETTINGS

The parameters of the simulations carried out in this work are presented in Table I. We simulated a WMN grid of 16 nodes using NS-3.10 [10]. The WMN nodes are equipped with two 802.11a interfaces for backhauling femtocell traffic. For simplicity we placed the UE applications directly on the WMN nodes.

The queues used in our simulation have a maximum capacity of 50 packets, as this is the queue size used by the most widespread wireless drivers, i.e. *MadWiFi* [11] and *ath5k* [12].

The impact of the packet delay on the MOS score is significant when its value is higher than 150 ms [7], and values higher than 400 ms render a conversation as non-interactive. Therefore the AC_VO queue in our simulations employs an early-dropping mechanism by removing packets when their queueing delay on a node becomes larger than 250 ms.

The routes are statically assigned in order to being able to draw statistically significant conclusions from the results. However, it is worth noting that Optimized Link State Routing Protocol (OLSR) [13] was used for initial route discovery.

We injected 100 calls into the network which is higher than the expected capacity. Calls are injected sequentially and the inter-call arrival rate is exponentially distributed with a mean of 1 second. Figure 3 depicts the call duration and start times
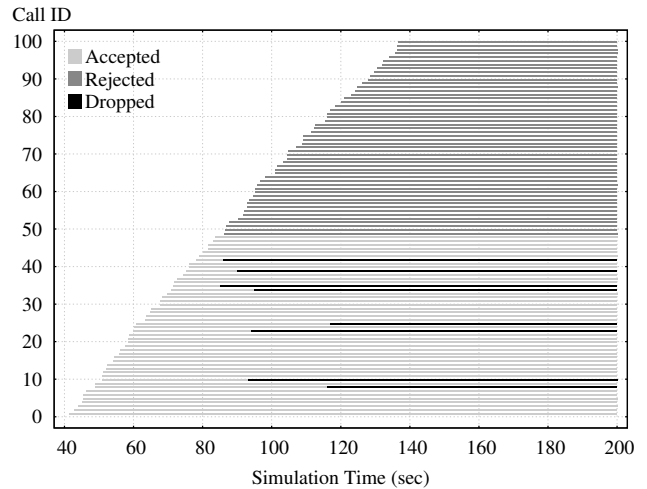

Figure 3: Call duration and status; dropped calls are the calls which were initially accepted but the CAC decided later to drop, as their corresponding iMOS was low.

Parameter	Value
Simulator	NS-3.10 [10]
Topology	Grid 4x4
Distance between nodes	100 m
Number of interfaces	2
WiFi Mode	802.11a
WiFi Data Rate	6 Mbps
Network Access Method	CSMA-CA
Propagation Model	LogDistancePropagationLossModel
Error Rate Model	YansErrorRateModel
Remote Station Manager	ConstantRateWifiManager
WiFi interfaces queue size	50 packets per AC
Early-drop threshold	250 milliseconds
Routing Algorithm	Fixed routes, pre-discovered by OLSR
No. of injected calls	100
Call Duration	Exp. dist. between 160 and 40 seconds
Call Direction	Full-duplex
Voice codec	AMR_12.20 & AMR_SID modes only
Speech model	ITU-T/P.59 [14]
iMOS threshold	R=70 (MOS=3.6)
CAC loop interval	1 second
Number of simulation epochs	10

TABLE I: Simulation Setup

of all calls injected in the simulation. For consistency of results only the MOS scores obtained during the time span of the shortest accepted call are considered.

The calls are full duplex and use the AMR codec. For simplicity we implemented only the 12.20 kbps and Silence Indicator (SID) modes. We implemented the speech model defined in [14] to mimic realistic conversations. During an active period of the speech model, i.e. when someone speaks, AMR_12.20 packets are sent, and during silent periods AMR_SID packets are sent.

A threshold value of 3.6 for the iMOS is used by the CAC mechanism to detect if it is necessary to activate CAC, as on the MOS scale it represents the border between *Some users dissatisfied* and *Many users dissatisfied*.
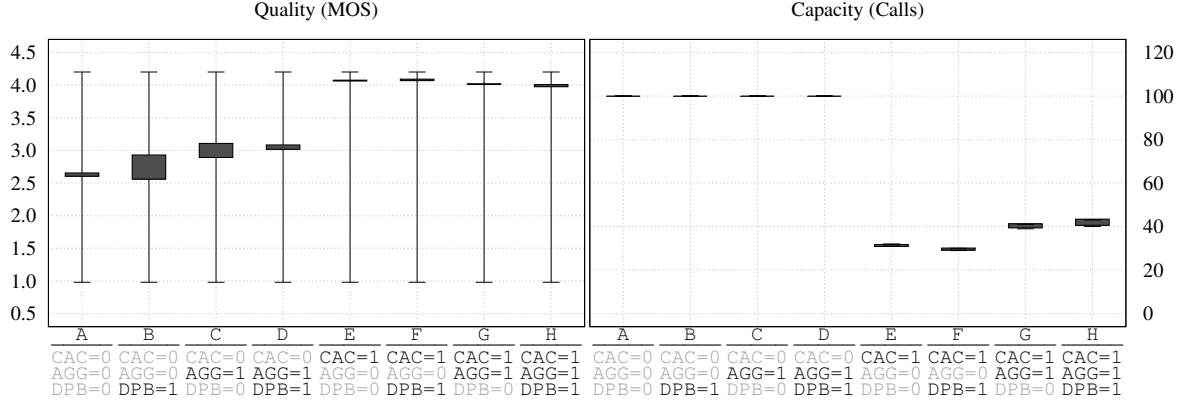
Figure 4: Comparison between all combinations resulting from using the CAC, frame aggregation, and delay-piggy-backing mechanisms in enabled or disabled mode.

V. SIMULATION RESULTS

A comparison between different scenarios is presented in Figure 4. Each scenario represents a specific combination of features. There are eight possible combinations with the following cases: CAC enabled or disabled, frame aggregation (AGG) enabled or disabled, and delay-piggy-backing mechanism (DPB) enabled or disabled.

The two diagrams in Figure 4 compare the cases related to the overall call quality observed by the users (left plot) and the overall system's call capacity (right plot). The middle of the filled box represents the average value, the margins of the box is the standard deviation around the mean, and the whiskers are the max and min values.

When CAC is disabled (cases A to D), the VoIP capacity diagram shows that all injected calls are accepted. However, the overall call quality suffered a high amount of degradation which is beyond acceptable levels. The frame aggregation feature increases the overall call quality by a small amount (cases C and D), however not enough to satisfy any user.

When CAC is enabled (cases E to H), the number of accepted calls is lower than the number of injected calls, but the overall quality is maintained at high levels on the MOS scale. When employing frame aggregation (cases G and H), the VoIP capacity increased by about 10 calls compared to the case when frame aggregation is not used (cases E and F), whereas the decrement on the quality scale is minimal. It can be seen that the delay-piggy-backing mechanism (case H) allows the network to accept a few more calls compared to the case when it was not employed (case G) with imperceivable reduction in call quality.

The result clearly shows that there is an obvious need for a CAC mechanism to detect and act in situations where the number of call requests is higher than the network's capacity. Regardless of the number of injected calls, the three mechanisms enabled in the WMN assure a maximum VoIP capacity of 43 calls, with our simulation parameters. In order to maintain the quality, some of the calls had to be dropped as shown in Figure 3.

The CAC mechanism is triggered by a timer to check the overall call quality and take a decision. The frequency of this periodical check allows the network operator to increase the overall quality or the VoIP capacity. A more frequent check allows the mechanism to detect quality degradations sooner and reject further requests, thus assuring for a high overall call quality at the expense of VoIP capacity. Figure 5 shows that checking periods longer than 5 seconds are not suited as the call quality drops to undesired levels.

Another influencing parameter in the behaviour of the combination of mechanisms is the distance between the WMN nodes, or inter-node distance. Figure 6 shows that increasing the inter-node distance influences differently the overall call quality when comparing the situations where CAC is enabled (case A to D) or disabled (cases E to H). When disabled, the overall quality is unacceptable for any user. It is worth noting that starting with 60 meters between the nodes, the wireless interference level drops allowing the call quality to increase, but still it remains under satisfactory levels. Beyond 120 meters signal quality of the wireless radios cannot sustain proper communication thus influencing the overall call quality to decrease.

When the CAC mechanism is enabled (cases E to H), an increase in inter-node distance results in a decrease in the overall call quality. However, the falling slope is small hence the quality is maintained to satisfactory levels for the entire investigated inter-node distance interval. The capacity is higher when frame aggregation is enabled (cases G and H), particularly when the inter-node distance is between 80 to 110 meters.

VI. CONCLUSION

This paper proposed a) a novel architecture for femtocell deployment utilising Wireless Mesh Networks (WMNs) as backhaul and b) mechanisms to assure high Quality of Service (QoS) for Voice Over IP (VoIP) calls by employing an Intermediate Mean Opinion Score (iMOS)-based Call Admission Control (CAC). The work presented a novel method of obtain-
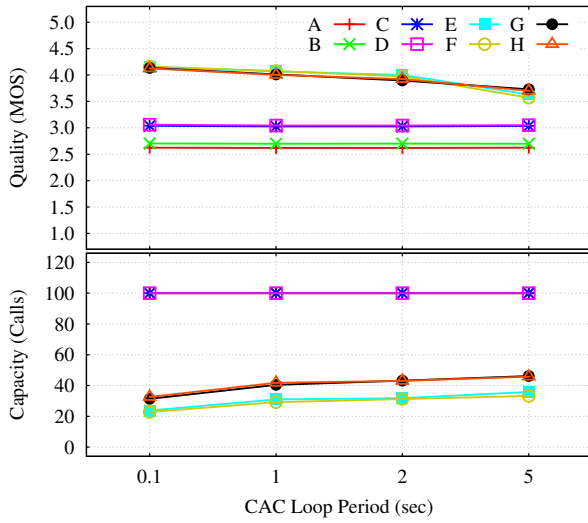
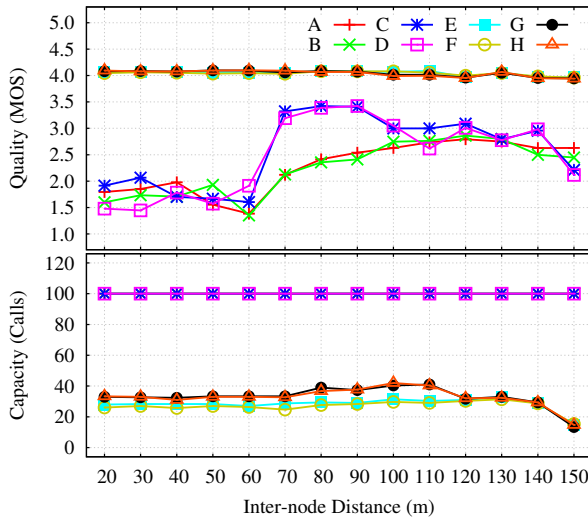Figure 5: Influence of CAC Loop Period on voice quality and capacity.

Figure 6: Influence of inter-node distance on voice quality and capacity.

ing accurate packet delay measurements needed to compute the call quality at intermediate points within the architecture.

The results showed that such CAC mechanism is needed in deployment scenarios involving a WMN as backhaul infrastructure. Furthermore, the usage of the proposed piggybacking-delay mechanism combined with the well known frame aggregation mechanism, showed improved capacity in terms of number of accepted calls in the network.

Two important parameters were varied over a range of values in order to test the flexibility of the proposed mechanisms. Namely, the frequency at which the CAC mechanism checks the network status and the WMN inter-node distance. The results showed firstly that higher polling frequency preserves the call quality at the expense of capacity, however a polling

interval larger than 5 seconds results in reduced and unguaranteed QoS. Secondly, when the inter-node distance was varied, the mechanisms are able to maintain high levels of VoIP call quality with increased VoIP call capacity around 30%, rising to 40% for distances between 80 to 110 meters.

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