An Approach ofQoS by Admission Control of VoIP over WLANs

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Abstract –**Call Admission Control (CAC) is one of the key strategies to achieve satisfactory QoS for VoIP over WLANs. There are presented experiments with real VoIP network with cable and WiFi and simulations with NS3 for adaptive CAC scheme, which would provide better performance with respect to real applied solutions. The results are analyzed.**

Keywords –**QoS, VoIP, Wireless, LAN, CAC.**

I. Introduction

Voice over IP (VoIP) over Wireless LAN is getting great attention from the industry and the products, implementing different technology for VoIP over WLAN deployment, which are rapidly emerging.Nowadays, network services are proposed to customers in an environment of strong competition, so with the advent of WiMAX for longer range Wi-Fi communications, wireless phone companies (cellular carriers) are gearing up to offer hybrid phones that will use VoIP over Wi-Fi when a Wi-Fi network is available and switch to cellular when one is not. So VoIP over wireless encompasses different aspects, depending on the context. The cost of services is important when companies make decision which technology they will use and VoIP is less expensive then cellular phones. Therefore, VoIP is a real-time application, making it particularly sensitive to packet loss that can be caused in a wireless network by weak signals, range limitations, and interference from other devices that use the same frequency.To support VoIP wireless network it must be reliable because users expect more dependability from their phone systems than from their computers. They expect a dial tone every time, no dropped calls, and high voice quality.

II. Related work

Many researchesare related with the availability of the network services and how to keep them in their planned quality.IEEE 802.11e standard [6] introduces Quality of Service (QoS) supported in Wireless LANs (WLANs). The enhanced distributed channel access (EDCA) is a key component of this standard and in [2] it is proposed a model for the VoIP capacity of an EDCA WLAN, which decouples the problem of estimating stations performance and that of evaluating channel congestion, because it models the dynamics of the sending queues.

Because of the sensitivity of VoIP applications to any disruption or delay, competing with data transmissions on the same wireless network can cause degradation of voice quality. It's important to implement QoS features to ensure that VoIP packets get priority.

CAC is recognized as one of the key strategies to achieve satisfactory QoS support for VoIP over IEEE 802.11 WLANs. Most of the CAC solutions for VoIP over WLAN are centralized, and the few distributed CAC schemes proposed so far do not account for issues such as the loss of channel time due to medium contention and the coexistence of VoIP traffic with background traffic such as TCP data flows. In [4] it is described a distributed CAC scheme which aims at addressing these issues by leveraging on channel monitoring techniques, thus being readily implementable in today's consumer devices.

In ATICAC (Adaptive Transmitting Interval Call Admission Control) strategy, proposed in [7], base station adaptively changes the transmitting interval of the active stations to prevent the network from saturation by controlling the average collision probability of the network.

In this paper is shown an analysis of the maximum number of VoIP calls in real WLAN and areproposedsimulation results for an ATICAC to enhance VoIP calls in WLAN.

III. Analytical study of VoIP over WLAN

For the purposes of the approach it is accepted that the total number of active full-duplex VoIP calls within the network is *n*. In addition, to simplify the analysis it is assumed that a full duplex VoIP call consists of two half duplex links between active stations and base station. It is assumed that VoIP traffic is equally distributed among the *2n* active stations.

Let T is a period, in which $T_{suc}$is the average time of a successful transmission,$T_{col}$is the average time between collisions of packets and $T_{idle}$is the average period of time in anticipation of a denial in a given time interval. This is obtained in (1).

$$T = T_{suc} + T_{col} + T_{idle} \quad (1)$$

Unlike a wired network, the actual available bandwidth (B_{avl})in wireless network is lower than the average bandwidth (B_{avg}), due to collisions and the expectation of denial (backoff idle). Available bandwidth is calculated with formulae (2).

$$B_{avl} = \frac{T_{suc}}{T} . B_{avg} = \frac{T_{suc}}{T_{suc} + T_{col} + T_{idle}} . B_{avg} \quad (2)$$

Let P_{idle} is the probability of idle time slot of the refusal, P_{suc} is probability for one successful transmission and P_{col} is the probability of at least two transmissions during the same time slot denial. According to the IEEE 802 standard [5] and

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[1] the probability of transmission in each active session at any period T, yielding the following equations:

$$P_{idle} = (1-t)^{2n} \tag{3}$$

$$P_{suc} = 2nt(1-t)^{2n-1} \tag{4}$$

$$P_{idle} = 1 - P_{idle} - P_{suc} = $$
$$= 1-(1-t)^{2n} - 2nt(1-t)^{2n-1} \tag{5}$$

According to study [1]:

$$t = \frac{2(1-2p)}{(1-2p)(W+1)+pW(1-(2p)^m)} \tag{6}$$

Where p is conditional collision probability (when collision occurs if a packet starts transmitting over the link) and W is CW_{min}.

The measure of QoS in VoIP network with G729a is based on [3]

$$R = 94.2 - 0.024d - 11 - 40\log(1+10e) -$$
$$-0.11(d-177.3)H(d-177.3) \tag{8}$$

$$d = d_{codec} + d_{jitterbuffer} + d_{network} \tag{9}$$

$$e = e_{network} + (1 - e_{network})e_{jitter} \tag{10}$$

$$H(x) = \begin{cases} 1 \; if \; x > 0 \\ 0 \; if \; x \le 0 \end{cases} \tag{11}$$

where$R$-score for QoS, d- delay, e- total loss, $H(x)$ – Heaviside function.

If R is more than 70, QoS is acceptable. In next experiments this value is used to assess QoS of real network and QoS of simulation network.

IV. EXPERIMENTS

A. Real Experimental Networks

For the purposes of the study are made experiments with real VoIP network over cable and Wi-Fi network. The first experiment is with HybSys3000(WirelessIP3000), which is on the market from 2006. One main disadvantage is that this system works with IEEE 802.11b standard.

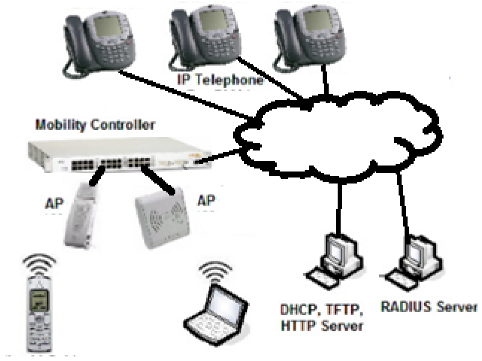

Fig.1.Experimental Network Diagram for Hitachi Cable WirelessIP-3000 SIP Telephone

The second experiment is with Trixbox (Asterisk PBX) basic appliance, because it is open platform for business telephony. Tests are provided in small software company "VBS", which have head office in Sofia and branch office in Varna and costumers all over the country. Diagram of VoIP network is presented on Fig. 1.

The compliance testing focused on verifying QoS for voice traffic while low priority background traffic was competing for bandwidth. The WirelessIP-3000 was verified to roam successfully between access points on the same network (Layer 2 roaming) and between access points on a different network (Layer 3roaming) while maintaining voice calls. Security schema WPA2-AES 802.1X-PEAP with Certificates and codec G.729AB was tested.

B. Results from Real Experimental Networks

When the SSID of the wireless LAN client is set to 'ANY connection', any wireless LAN access point can be connected to. However, access points that reject LAN clients set to 'ANY connection' cannot be connected to. The jitter size that can be tolerated in fulfilling the required quality of conversation differs according to the jitter buffer of the receiving device. The role of the jitter buffer is to store the arriving VoIP packets in the buffer and adjust the latency inthe arrival times of packets prior to sending to end user.If the jitter buffer is made bigger, the jitter certainly becomes less, but if the size is made too big, intolerable delays in conversation is forced onto the end users. The results showed that calls were maintained for durations over one minute without degradation to voice quality. The telephony features verified to operate correctly included attended/unattendedtransfer, conference call participation, conference call add/drop, multiple call appearances, caller ID operation, call forwarding unconditional, call forwarding on busy, call forwarding clear.

The maximum number of VoIP calls (N) depends on the bit rate of codec and the available bandwidth. The payload rate of voice data for each stationis very low (for G.729a is 8kbps), the required bandwidth to transmit these data payload is very large, in comparison with the period T_{suc}. If T_p is the time to send payload information, it is calculated that:

$$T_p = \frac{Size_{\ of_{frame}} . Frames \ per_{packet}}{R_{data}} \tag{12}$$

Where R_{data} is the bit rate for data packets (2Mbps).

Based on experiments with real equipment $B_{avl}$of a network in the saturate status is 0.9 times of the maximum B_{avl}a network can provide. Then calculation of the number of calls is:

$$N = \frac{\frac{B_{avl}}{0.9}}{2.B_{req}} \tag{13}$$

According to (13) the more VoIP calls are allowed in the network -the lower the required bandwidth (B_{req}) is, whilst larger number of frames per packet means larger delay. If the number of frames per packet is too large, it will not satisfy the quality of VoIP. In Table 1 is presented maximum number of calls. Third column presents calculated results, based on formulae (13). Next columns present measured maximum number of success calls in real VoIP network.

TABLE I
MAXIMUM NUMBER OF VOIP CALLS FOR G. 729A

Frames per packet	Required bandwidth (kbps)	Analytical Number of calls	Number of calls in HybSys3000	Number of calls in Trixbox
1	144.4	5.9251	5.8300	5.8000
2	76.2	10.4945	10.5000	10.4000
3	35.28	20.9946	20.9900	20.9000
4	27.48	26.1102	26.1100	26.1000
5	23.15	30.4697	30.4700	30.4000

To be possible to calculate *R*, the delay *d* and the total loss *e*must be measured. On Fig. 2 and Fig. 3 are presented these parameters for both networks. Because of the fact that experiments are based on the same network and same codec, the difference in the delay depends only onthe jitter buffer of the receiving device. Based on these results the score for QoS (*R*) is calculated by formulae (8) and it is average 72.596 for HybSys3000 and 72.553 for Trixbox. This means, that QoS is acceptable, but if network delay is less than 130ms, average loss will be less than 2.5%.

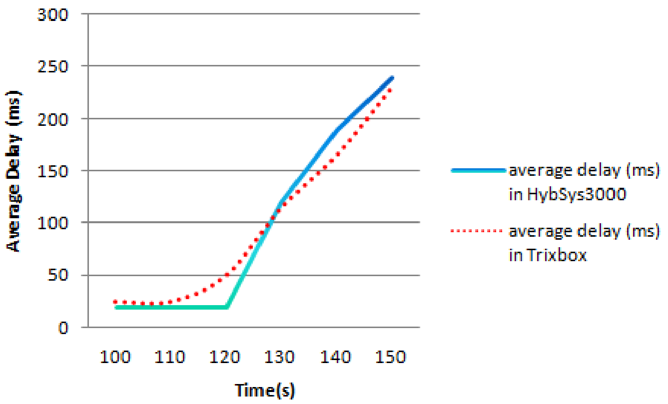

Fig. 2.Average Delay in msfor Hitachi Cable WirelessIP-3000 SIP Telephone and TrixBox

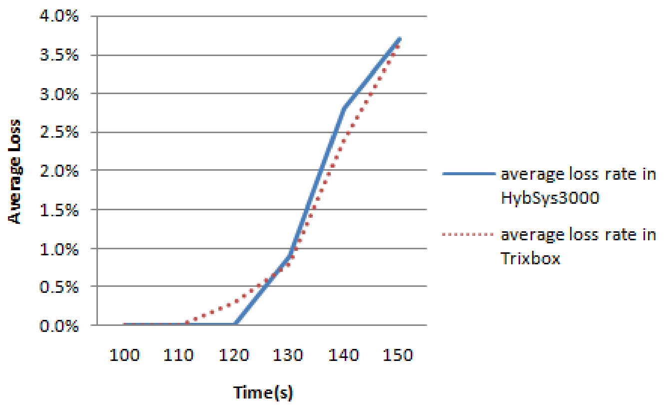

Fig. 3.Average Loss in Percentsfor Hitachi Cable WirelessIP-3000 SIP Telephone and TrixBox

A. Experimental Results from Simulation of Adaptive Interval

Forthepurposeoftheexperimentit is used ATICAC algorithm, which is presented in [7]. This algorithm is implemented in VoIP module in Network Simulator 3 to be possible to compare with experimental results from real VoIP networks. In the simulation network, traffic generator creates sequence of calls every 2 seconds with transmitting interval 20ms. Data traffic in the same network is constant with the rate of 50kbps.

On Fig. 4 is presented the maximum number of calls, which depends on different *CWmin* with the same data rate of *2M*. It is clear to see that the maximum number of success call supported increases with the increase of transmitting interval, but the change of *CWmin*takes little effect on the maximum number of calls in VoIP network. This is dueto the fact, that when *CWmin*grows up, it provides more backoff timer selections.

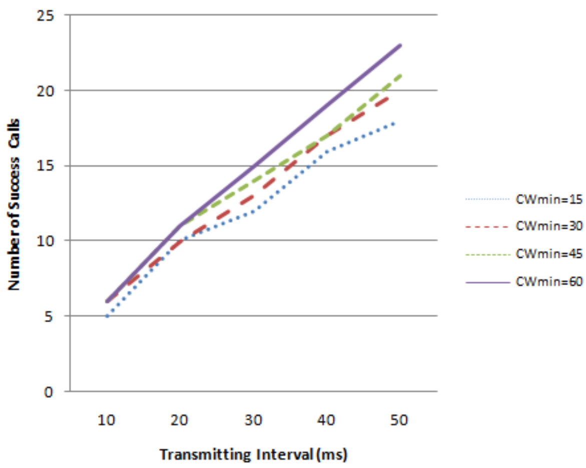

Fig. 4.Maximum number of success calls with data rate of 2M

Next step is presenting statistics for network QoS when number of calls is increased – with ATICAC and with normal CAC.To be possible to calculate *R*, the delay *d* and the total loss rate *e*must be measured. On Fig. 5 and Fig. 6 are presented these parameters for simulated networks with normal CAC and with ATICAC. When network is with ATICAC, it reduces voice traffic blocking probability.

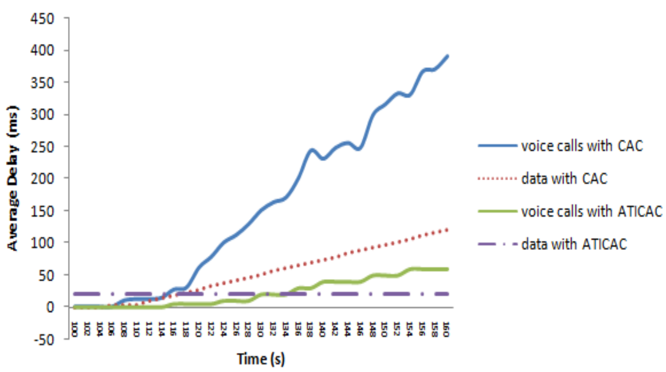

Fig. 5.Average Delay in the Simulated Network

Experiments are based on the same network and same codec. Based on these results the score for QoS (*R*) is calculated by formulae (8) and it is average 71.974 for network with normal CAC (In real experiments it is 72.596 or 72.553) and 74.413 for network with ATICAC. This means, that QoS is acceptable for both networks, but if network delay is less than 120ms, average loss will be less than 5% for network with normal CAC and for network with ATICAC loss rate is close to 0% when network delay is up to 160ms.

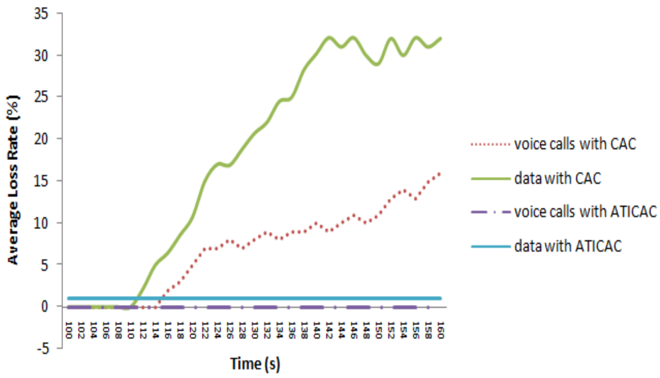

Fig. 6.Average Loss Rate in the Simulated Network

This affects on QoS and parameter *R* for network with ATICAC is a little bit bigger than parameter *R* for network with normal CAC and QoS is better than network with normal CAC. This means that network with ATICAC will support more VoIP calls than network with normal CAC.

V. CONCLUSION AND FUTURE WORK

VoIP wireless network must be reliable because users expect a dial tone every time, no dropped calls, and high voice quality.In this paper is presented study how VoIP over WLAN can support more VoIP calls with same quality.

There have beenexperimental studiesof two market decisions.It is presented analytical model and criteria for network QoS. Based on this analytical model and data from real VoIP over WLAN it is made simulation and the results are presented. Because of the high collision probability,QoS of VoIP calls over WLAN is decreased when too many calls join the network in the same time. The results for QoS show that with ATICAC loss rate in network is less than other decisions, it improves VoIP over WLAN by carefully exploring the tradeoff of voice delay and the number of calls through adaptive transmitting interval.

In this study is presented a calculation of the complex parameter for QoS for each experimental or simulated network and it is better for network with ATICAC, but it is close to the existing decisions on the market.

There are a number of avenues for future work, e.g., similar studies in data rate of 10M, to verify the conclusions, which are presented in this study about data rate of 2M; examine cases in which traffic in the system is of low arrival rate and short dispatch time, in order to monitor the effect on the likelihood of ATICAC immobilization of data traffic; through simulation will track average loss rate, average delay, maximum number of success calls for a different amount of system capacity, earmarked for data traffic. In addition, the results will be compared with the simulation of the strategy in which only data traffic is restricted to the same capacity. The aim is to find the maximum voice traffic arrival rate, which the system can support with certain voice traffic blocking probability.

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