

Issues and Flaws in Endpoint based Call Admission Control for Voice over WLAN

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ABSTRACT

This paper presents the various issues and drawbacks of the popular and existing scenario of carrying the Voice over the WLAN. The existing scenario that manages the Voice over Internet Protocol (VoIP), over an IEEE 802.11 WLAN is the Endpoint based Call Admission Control (CAC). Even though the most researches and experiments over the Endpoint based Call Admission Control shows it to be the most effective and yielding acceptable QoS, but still there are certain drawbacks and issues that arise in the real time scenario. Therefore, this paper focuses on those areas of this methodology that leads to flaws and various techniques through which issues can be reduced.

Keywords

IEEE 802.11, Admission Control (AC), Modulation and Coding Schemes (MCS), Signaling Protocol (SIP), Endpoint Admission Control (EAC)

1. INTRODUCTION

The most common and widely used methodology for carrying Voice over VoWLAN is based on the Endpoint Admission Control Paradigm. Numerous experimental works as well as the research are being done in this field to make it flawless. This is due to the usage of the VoWLAN, as various industries have adopted to use Wireless LAN so that inexpensive wireless VoIP handsets could be deployed which utilizes the wireless network to provide the services of the wireless telephony. Linksys/Vonage has proprietary solutions that use dedicated equipment based on the IEEE 802.11. [6] Also, many of the well-known organizations such as Avaya have their proprietary solutions [8] which lead to the installation of the expensive and complicated equipment. The research can only be beneficial, if it is compatible with the existing methodology, otherwise, there could be the need of designing the special devices and equipment and it has become the major research key challenge to make wireless telephony compatible with the standard infrastructure, rather than owning expensive proprietary solutions

VoIP [2][7] is defined as the technology, which is used by the IP telephony for the transportation of phone calls and refers to the communication protocols, technology and transmission techniques which are being involved in the delivery of voice communication as well as the Multimedia sessions over the IP networks such as internet. An example for it can be the internet telephony.

The major technical issues, which arise while IEEE 802.11 based WLAN is used for carrying voice over Internet Protocol (VoIP) can be divided into two major parts: Call Admission control (CAC) [11] and the Link Adaptation (LA).

It is already known that the starting of the congestion is unpremeditated or sudden. The Quality of Service (QoS) can go from the acceptable state to the unacceptable state resulting in the poor performance of the system [10]. Even with the addition of little extra traffic, or even one more call from its threshold value of call admission, the system moves from the uncongested state with high Quality of Service to the congested state with unacceptable and poor quality. Thus, it is being noted that when one more call is placed on the wireless network, then it can be supported; all calls start to suffer with unacceptable quality of Service. [3]

The several data rates are related to dissimilar Modulation and Coding Schemes (MCS), each of which has the different robustness to the channel errors.[8] The Link Adaptation can be defined as the process, which a wireless station can adopt its MCS to suit the current channel conditions with the purpose of maintaining an acceptable link quality [5][12]. For example, from the associated AP, if the mobile user roams away and due to it, its received signal strength decreases, then the wireless interface switches to a more robust modulation scheme to maintain an acceptable BER (Bit Rate Error). But the situation becomes more complex if the user is on the Active VoIP call, because, selecting the more robust modulation scheme and reduction in the transmission rates results in more channel time to carry the voice data for a given codec [13]. This increased usage of the resources adds the load to the network. Therefore, the second technical issue i.e. Link Adaptation results in the congestion in the network, leading to unacceptable QoS.

2. EXISTING SCHEME ARCHITECTURE

A typical enterprise network configuration is illustrated in Fig. 1 which shows the basic architecture of WLAN phone system. There are two cases: the first one, in which there is wired VoIP, exists and the second one is wireless, over WLAN. In the first scenario, VoIP server and gateway is present and only wireless handsets are required to provide the wireless. The VoWLAN can be interfaced with the first case with the help of legacy PBX through voice gateway. [8][4].

According to [1], WLAN cannot provide the acceptable QoS for VoIP in the presence of greedy data users on the same network. At the traffic buffer of Access Point, data traffic using TCP interfere with VoIP traffic, resulting in long queuing delays leading to jittery and lossy performance for the voice traffic. Hence, there comes the main reason to have a mechanism that can differentiate between the voice traffic and the other traffic on WLAN, which can be done by using the dedicated access points or either by using the traffic prioritization mechanisms, which are used by 802.11e EDCA [9].

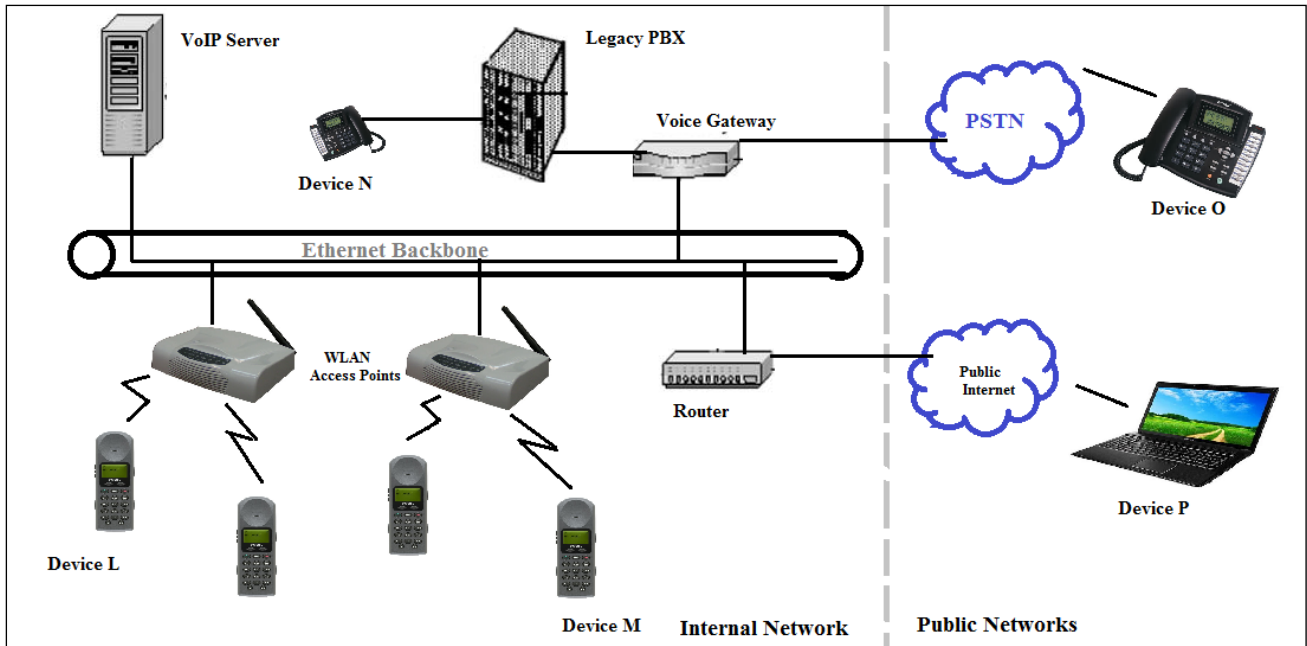


Fig 1: System architecture diagram for a WLAN phone system

3. ENDPOINT BASED CALL ADMISSION CONTROL FOR VOICE OVER WLAN

When it comes to define the Endpoint Admission Control (EAC) paradigm [3], it is a mechanism where the endpoint, such as the WLAN terminals sends the probe packets to determine the state of the network i. e. whether the network is in the situation to admit a new call or not with acceptable QoS. Because as accepting even a single call above its threshold limit could result in the poor performance, it is critical to determine that the call can be admitted or not.

One more key fact of using the Endpoint Admission Control Paradigm is that there is no need to change the existing systems and equipment. Components such as Access Points and routers can be effectively adopted without any changes in them. Only requirement is the addition of wireless VoIP handsets with their updated software.

Depending upon the state of the network, in EAC, routers mark or drop the packets so that it is known that a call can be admitted in network or not. The Marking is usually based on the setting special Explicit Congestion Notification bits (ECN Bits). The congestion information detected by the routers in the duration when the probing is done, is being conveyed to the endpoints and upon receiving the probes, the destination endpoint echoes the probe packet back to the source endpoint. Then, at last, the information carried out is being used to perform the Call Admission Control (CAC) decisions [14].

Referring to Fig. 2, the endpoint based call admission is based on the ICMP echo messages which are being used to measure the Round Trip Time (RTT), packet loss and Jitter. The probe packets are being used which mimic the traffic into factors: packet interval and packet size. The core motive to use the ICMP echo message is this that all type of VoIP servers can respond to them, without any reforms. In EAC, the echo request mimics the uplink call leg and the echo reply mimics the downlink call leg. The call is then admitted if the measured Round trip Time (RTT), Jitter values are less than that of the

threshold limits. When call is being admitted, it enters into the data phase where the VoIP packets are being transmitted over the WLAN. [4]

Total three situations occur while using the endpoint based call admission control. These are:

1. A call originates and also terminates on the WLAN phone systems.
2. A call originates but does not terminate on WLAN
3. A call terminating only on the WLAN

The first scenario takes place when the device L initiates the call. Then the Endpoint based client on the handset sends the probe packet to the VoIP server to verify that the originating cell is capable of supporting the call. If it is, the SIP INVITE message is sent to the server to initiate the call setup. Now, after receiving the SIP invite message from the server, device M also sends the probe packets to the VoIP server to verify that the terminating cell is also capable of supporting the call. In this case, if the call can be admitted, SIP (Signaling Protocol) completes the call setup procedure and the transmission of the voice packets occurs with acceptable Quality of Service [15].

If in case, the originating cell is not able to support the call, and then device L starts playing the busy tone. Then no INVITE message is being sent to M. If, in case, the terminating cell is unable to support the call, it sends the 'BUSY HERE' message to device L via Server. Then device L plays the busy tone for its user.

The second scenario, deals with the outgoing calls which originated on the WLAN. Referring to the fig.1, this scenario comes into the picture when there is call between L and N, L and O, and, L and P. In this, the originator sends the probe packets to the server and waits till the call is being admitted as shown in fig.3. Here one key fact to be noticed is this that back to back probing is being done continuously which

becomes the major drawback of this mechanism as described further in this paper.

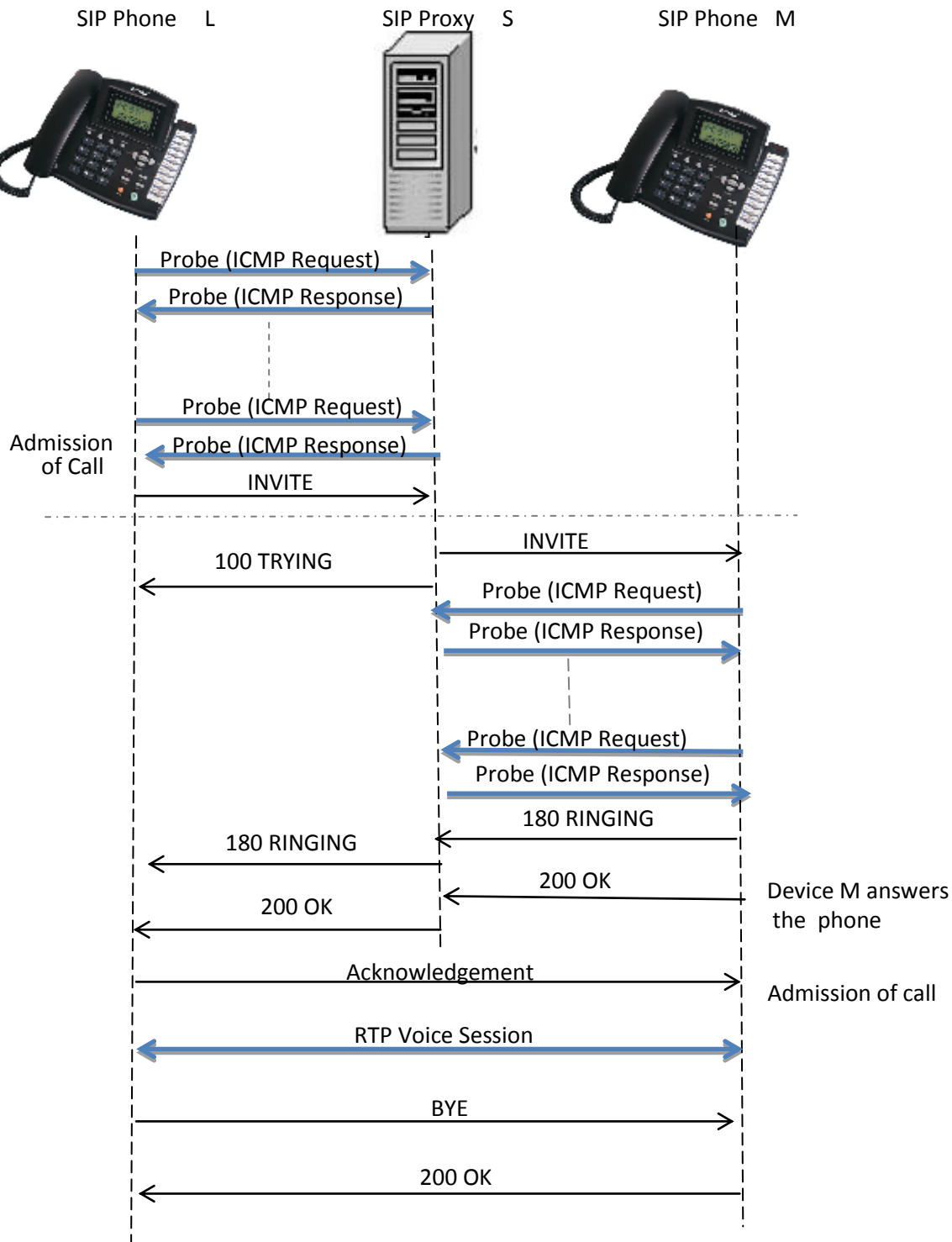


Fig 2: Sequence of Call Admission through Probing

The third and the final scenario deals when the call exists between N and L, O and L, and, P and L. (refer fig. 1.). Here, when the callee receives the 'SIP INVITE' message, the client probes the server to verify if the terminating cell is capable of supporting the call. The process can be seen in fig. 2 below the dashed line.

3.1 Algorithm of Endpoint Based Call Admission Control

It is appropriate to estimate the downlink delay through the Round trip time of the probe packets. As probes are sent by the handsets, they have the fixed size and are sent at the regular interval. So, to track the Round Trip Time of the

probes that are echoed back by the VoIP server, a window of length tw_{AC} is being set in the VoIP enabled handsets.

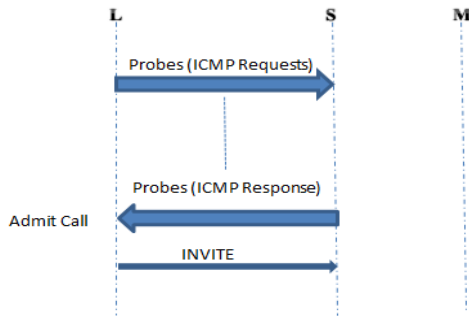


Fig 3: Call setup sequence when call terminates only on the WLAN

According to [4], if N_L probes are first transmitted into the network. Then, if the next N_{AC} probes transmitted have an average RTT delay greater than tv_{AC} , the call is being rejected else this step is performed n times. Thus, a call is either rejected after $N_L + (c \times N_{AC})$ probes where $1 \leq c \leq n$, or admitted after $N_L + (n \times N_{AC})$ probes.

According to the existing scenario [4] [8], if a probe, is not received within the tv_{AC} , is considered to be lost and the threshold value is incremented by one. This is the major drawback of this scenario as this algorithm could not specify that is the probe being actually lost or it suffered from long Round Trip Time and the case treated as the ‘delay only’ admission threshold. This existing methodology does not provide us the reason that what happened to the probe packet and if it is lost, the reason behind that is not known.

Under the certain experimental conditions, it was being noted that the system enters into three states: (Refer Fig: 4)

1. Steady State: having lesser delay, jitter and no loss.
2. Transient State: have increasing delays, high average jitter and no loss
3. Steady State: have unacceptable high and constant delay, low jitter and the periodic loss.

These phases were distinguished when in experiment, the shortest transient period was measured as 1Mb/s lasting with 36 packets and the longest transient period was noted as 2Mb/s lasting with 120 packets.

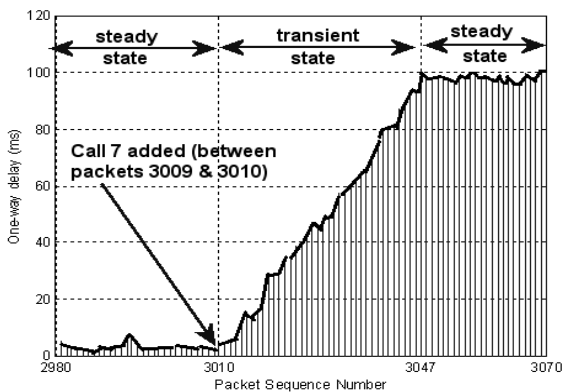


Fig 4: Various phases along with one way loss statistics and delay in downlink direction when the system capacity is being exceeded.

Referring to fig.5, in which Round Trip Time delays of probing the network has reached the quota because the probing traffic pushed the WLAN capacity and the call was rejected after the 19th probe packet.

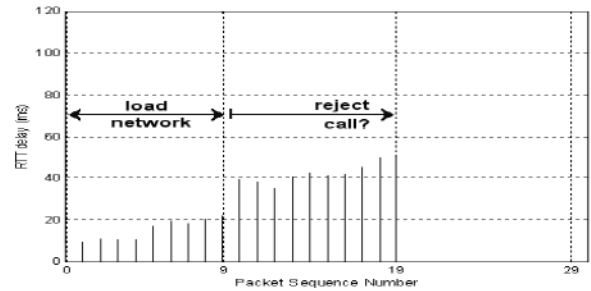


Fig 5: RTT delays seen by probes of a call which is not admitted.

4. ISSUES AND FLAWS:

Apart from the certain advantages of the existing system such as detecting congestion in the WLAN used by each party to the call, using probe packets, there are certain issues and flaws in the existing scenario.

While initiating the call, or call admission, the VoIP enabled handsets sends the probe packets continuously till the call is accepted for the admission into the network, which becomes the major basis of causing the congestion into the network by flooding probe packets back to back.

Most of the wireless devices such as the VoIP enabled handsets work on the battery. Hence, sending the back to back probe packets may drain their battery in the circumstances when the client needs to wait for the call admission for the lengthier duration

These limitations could be fixed if special concern is been taken of the lost probes in efficient way and by reducing the need of back to back probing by using the retransmission timer. By using the retransmission timer, a time window of certain length could be set by which the time frame could be set by determining the average values. Then, instead of back to back probing, the client will send the probe packet and if the timer expires, it will send the other probe packet. If in case, the packet is being lost, the timer will expire and then the client will send the new probe. Therefore, by using the retransmission timer, the better results can be achieved in the real time scenario by taking into consideration the threshold value (measured by RTT), Retransmission timer interval, and the time count along with other parameters.

5. CONCLUSION

The existing Endpoint based Call Admission control can be improved by working on its limitations and flaws. It can be improved by using the retransmission timer and so that the problem of the congestion and the difficulties faced by the Call Admission in VoWLAN and the problems created by back to back probing could be reduced. By enhancing it by this way, certain issues termed above can be fixed.

6. ACKNOWLEDGMENTS

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