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Endpoint Based Solution for Resource Management and QoS Enhancement in VoWLAN

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Abstract: *This paper deals with two endpoint based solutions for resource management (call admission control and dealing with link adaptation) in VoWLAN, and its performance in mixed voice/data traffic in the network. Call admission control is based on endpoint admission control algorithm where the endpoints probe the network to determine if the call can be supported with acceptable Quality of Service (QoS). Link Adaptation results in congestion in the network which can be rectified by a codec adaptation algorithm, where the codec of the ongoing call can be changed to a better codec based on the packet transmission period. Here an endpoint based VoIP aggregation method is used to improve the voice capacity of the WLAN. The paper also solves the inherent interference problem between VoIP and TCP traffic at the buffer of the AP (Access Point) in a WLAN by using a voice priority queuing solution that effectively eliminates the problem.*

Keywords— *call admission control, IEEE 802.11b, VoIP, Link adaptation, Codec adaptation.*

I. INTRODUCTION

Voice over WLAN (VoWLAN) is a natural evolution of VoIP. VoIP allows users to make IP (Internet Protocol) based calls over the global networks. It is also a potential supplement or a potential competitor to 3G mobile systems. Wireless Local Area Networks (WLANs) have become very popular in recent years. The most popular technology is based on IEEE Standard 802.11 [7]. 802.11-based WLANs are often considered as wireless extension to Ethernet standardized in IEEE Standard 802.3. Wireless LAN is used as an alternative and a complement to wired LAN. With the increasing popularity of IP technology, real-time applications such as interactive two-way voice and multimedia over IP have become popular.

QoS of VoIP is an important concern to ensure that voice packets are not delayed, lost or dropped during the transmission over the network. VoIP quality of service is measured based on different parameters like delay, jitter, packet loss and echo. In VoIP, Echo is experienced when caller at the sender side hears a reflection of his voice after he talked on the phone or the microphone whereas the callee does not notice the echo. By adjusting these parameters the quality of service can be controlled.

In this paper the proposed method is used to support both voice and data packets at the same access point providing higher priority for voice traffic by using voice priority queuing. Call admission control and voice codec adaptation [1] methods improve the QoS of VoWLAN. Frame aggregation at the endpoints increases the voice capacity of VoWLAN.

II. Proposed Model

A. VoWLAN Architecture

VoWLAN architecture consists of IP phones, data stations, access points, router, VoIP server, voice gateway and PSTN. In this

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architecture we consider both voice stations and the data stations accessing the wireless channel to the same access point. The VoIP server used here is an SIP server. An SIP server is the main component of an IP PBX, dealing with the setup of all SIP calls in the network. A SIP server is also referred to as a SIP Proxy or a Registrar. Although the SIP server is the most important part of the SIP based phone system, it only handles call setup and call tear down. It does not actually transmit or receive any audio. This is done by the media server in RTP.

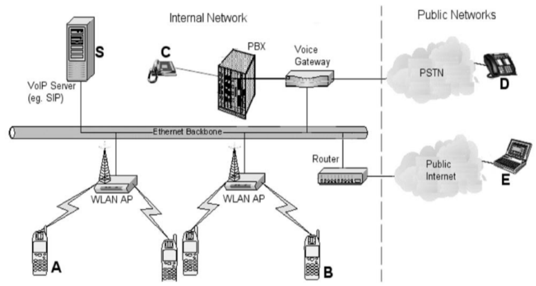

Figure 1. VoWLAN architecture diagram

If wired VoIP is already in use, then the VoIP server and gateway will be in place, and all that is needed is the addition of wireless handsets to provide wireless VoIP. The VoWLAN system can also interface with a legacy PBX through a voice gatewayIn VoIP networks, the primary purpose of voice gateways is to provide an interface between the VoIP network and the Public Switched Telephone Network (PSTN). The gateway acts as an interface between the IP network and the PSTN.

B. Frame Aggregation

Frame aggregation is also an endpoint based concept to increase the voice capacity of the WLAN. At the endpoints the VoIP frames generated are aggregated (specific number of frames) into a single VoIP payload by a multiplexing unit. This VoIP payload is then transmitted to the receiver. At the receiver there is a demultiplexing unit which retrieves all the VoIP frames from the received payload. This is then send to the playout buffer.

C. Endpoint Admission Control(EAC)

The simplest admission control approach that can be implemented at the endpoints is based on detecting packet losses due to congestion in the network. The EAC procedure consists of a type of "local path" probing of the route to the VoIP server, which focuses on detecting congestion in the wireless access network, as opposed to "end-to-end" probing.

In EAC all admission control decisions are taken by the endpoint devices like the VoIP phones, laptops etc. The VoWLAN EAC probing procedure is based upon the use of ICMP echo messages (ping), which can be used to measure Round-Trip Time (RTT), jitter, and packet loss The probe is designed to mimic the traffic of an actual voice call in terms of packet size and packet interval. The echo requests mimic the uplink call leg and the echo responses mimic the downlink call leg.

Algorithm

1. Wait for w_{AC} probes to be received (to allow moving window to fill)
2. If all w_{AC} probes have RTTs > threshold, T_{AC}
    a. terminate probing and reject call
3. For each new probe received
    a. If N_{AC} probes have been received go to step 4
    b. Else repeat step 2, firstly updating the moving window with the new RTT value
4. If all w_{AC} probes have RTTs > T_{AC}, then reject call, else admit the call Any lost probes are counted as an RTT delay > T_{AC}

At the handset, a moving window of length w_{AC} which contains the individual RTT delays of the probes echoed back by the VoIP server is used to perform the AC decisions. To allow the window to fill, w_{AC} probe responses must first be received. If all w_{AC} RTT delays exceed a predetermined threshold, T_{AC}, then probing is terminated and the call is rejected. Otherwise, the window is updated upon the arrival of the next probe response, and the admission criterion is again tested. This last step is repeated until the call is either rejected or until NAC probes have been transmitted, at which point the procedure completes and the call is admitted.

D. Link Adaptation

The problem of effective resource management is further exacerbated due to the fact that 802.11-based WLANs provide multirate support. IEEE 802.11b supports data rates of 11, 5.5, 2, and 1 Mb/s. The various data rates are based on different modulation and coding schemes (MCSs), each of which has a different robustness to channel errors.

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Link Adaptation is the process whereby a wireless station adapts its MCS to suit the current channel conditions with the aim of maintaining an acceptable link quality. LA may be invoked, if a mobile user roams away from its associated AP such that the received signal strength deteriorates and the wireless interface switches to a more robust modulation scheme to maintain an acceptable Bit Error Rate (BER). For a user on an active VoIP call, the more robust modulation scheme and resulting reduction in transmission rate means that more channel time is required to carry the voice data for a given codec. This increased resource usage effectively adds a load to the network and may result in the WLAN capacity being exceeded. Thus, LA like the addition of new calls may result in congestion on the WLAN and hence unacceptable quality for all existing call connections.

In the proposed solution, a call which undergoes LA and which results in the onset of congestion in the WLAN is permitted to continue with acceptable QoS through the use of an intelligent voice codec adaptation scheme also implemented at the handset.

E. Voice Codec Adaptation

Voice codec adaptation is the process by which the codec of the ongoing call is specifically adapted to maintain approximately the same level of channel occupancy time as before the LA took place. Channel occupancy time is defined here as the fraction of channel time per unit time required by a full duplex VoIP call connection for a given codec setting and transmission rate. This metric will allow different codecs with possibly differing packetization intervals to be compared based on their “per second” channel occupancy requirement. This is calculated by separating the time required to transmit a VoIP packet into its constituent components, determining the time required to transmit each component for a given codec configuration and transmission rate, and then summing the individual components. This result is then normalized relative to unit time to give the channel occupancy time for that codec setting at the specified transmission rate.

Algorithm

1. Wait for w_{LA} phase jitter delay to be received (to allow moving window to fill)
2. If all w_{LA} probes have jitter values > threshold,
    a. T_{LA} terminate probing and reject call
3. For each newly arriving packets
    a. If N_{LA} jitter values have been received go to step 4
    b. Else repeat step 2, firstly updating the moving window with the new jitter value
4. If all w_{LA} packets have jitter values > T_{LA}, then reject call, else undergo codec adaptation.

If congestion is detected after an LA event, then codec adaptation is triggered where the new codec setting chosen is based on table lookup. A list of possible codec configurations that may be used during the VoIP call can be established at call setup time using the signaling protocol (e.g., SIP/SDP INVITE message). Then, either call participant can switch dynamically to any of the other codec configurations from the agreed list during the call. When one of the call participants adapts its codec configuration the other participant detects this from the RTP headers of the incoming voice stream and consequently adapts its decoding and encoding profile to match the new codec settings of the received voice stream. Thus, when one endpoint adapts its codec configuration the other endpoint automatically adapts its coding profile in immediate response, without the need for any explicit signalling.

III Simulation Results

Simulation was carried out using NS-2 as the simulation tool (fig. 1).. VoWLAN architecture is setup with mobile nodes, access points, routers and gateways (fig. 1).

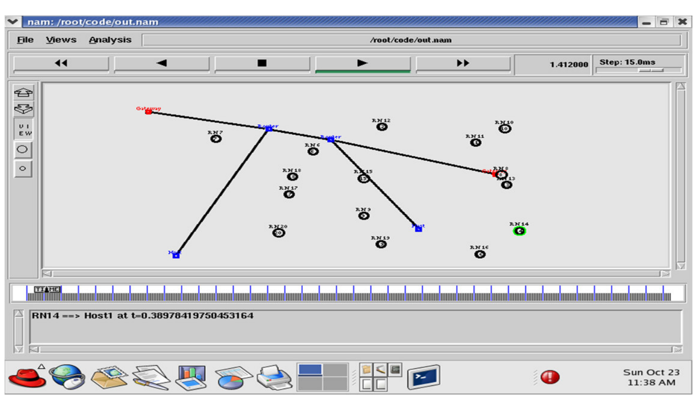

Figure 2. VoWLAN simulation setup

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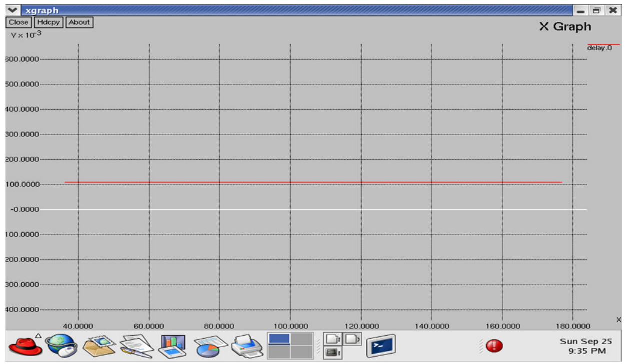

Figure 3. Delay graph

VoIP connections were established between the mobile nodes. A frame aggregation process is carried at the endpoint and the total end to end delay is calculated (fig. 3, x-axis denotes total duration of simulation, y-axis denotes end to end delay) to be 100ms which is good for wireless voice transmission.

IV Conclusion

The approach of differentiating voice and data traffic at the access points into separate queues and providing voice prioritization will make sure that the network resources are used accurately. And the endpoint based voice packet aggregation increases the voice capacity of Voice over WLAN. More over Endpoint based call admission control, and voice codec adaptation which overcomes the congestion caused by the link adaptation, increases the quality of service (QoS) of the ongoing calls.

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