Title: LINK LAYER PACKET LOSS CLASSIFICATION FOR LINK ADAPTATION IN WLAN

Abstract: A method of link layer behavior analysis to infer the cause of packet loss in a WLAN is provided. More specifically, a classification of packet loss into either congestion or wireless errors based mainly on delays and MAC layer parameters under congestion and/or link error dominated status is disclosed. A method and a system to apply a link layer packet loss classification scheme to support link adaptation for VoIP applications in WLAN are also disclosed.
before the expiration of the time limit for amending the claims and to be republished in the event of receipt of amendments

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.
LINK LAYER PACKET LOSS CLASSIFICATION FOR LINK ADAPTATION
IN WLAN

CROSS-REFERENCE TO RELATED APPLICATIONS
[0001] This application claims the benefit of U.S. Provisional Application No. 60/778,324 filed March 2, 2006, which is incorporated herein by reference.

BACKGROUND OF THE INVENTION
[0002] The present invention relates to performance classification of WLANs in terms of Packet Loss. More specifically, it relates to Link Adaptation in WLANs based on Packet Loss Classification.
[0003] Voice over IP (VOIP) and high quality video streaming become an alternative to conventional telephony and TV service in many locations by way of ongoing deployment of WLAN hotspots. The newly approved IEEE 802.11e QoS enhancement standard will be a proper solution of delay demanding for real-time multimedia applications. However, the existing Auto Rate Failback (ARF) link adaptation scheme, designed to improve wireless transmission by dynamically selecting transmission rates and frame sizes, cannot effectively solve the QoS concerns of multimedia communication due to the growing UDP traffic. A comprehensive analysis of link layer behavior points a cause of quality loss to packet loss in a WLAN. Accordingly, methods or apparatus providing improved performance when WLAN quality is deteriorating due to packet loss are required.

SUMMARY OF THE INVENTION
[0004] One aspect of the present invention presents a novel method and system that will provide a broad advantage to WLAN and to VoIP systems.
[0005] In accordance with one aspect of the present invention, a method for link adaptation in a multi-user WLAN used for VoIP is provided. The method includes determining a channel rate as a function of an error rate, a threshold backoff and defer time and a measured backoff and defer time. The error rate can be, for example, a frame error rate.
[0006] In accordance with a further aspect of the present invention, the threshold backoff and defer time is determined by the expression \( CW \max X slottime + n_{\text{fer,avg}} \times d \).
[0007] In accordance with another aspect of the present invention, the channel rate is lowered only if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

[0008] Alternatively, the channel rate can be lowered if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time. Further, the channel rate can be unchanged if the frame error rate is above the threshold and the measured backoff and defer time is greater than the threshold backoff and defer time.

[0009] In accordance with yet another aspect of the present invention, a system accomplishing the processes described herein can also be provided.

DESCRIPTION OF THE DRAWINGS
[0010] FIG. 1 is a diagram of packet loss classes.
[0011] FIG. 2 is a time diagram of two stations communicating in a WLAN.
[0012] FIG. 3 provides two performance plots of a VoIP client in a WLAN.
[0013] FIG. 4 provides two performance plots of a AP in a WLAN.
[0014] FIG. 5 provides two performance plots of multiple VoIP clients in a WLAN.
[0015] FIG. 6 provides two Link Adaptation examples.
[0016] FIG. 7 illustrates a computer system that is used to perform the steps described herein in accordance with one or more aspects of the present invention.

DESCRIPTION OF A PREFERRED EMBODIMENT
[0017] With the emergence of wireless technology, a variety of multimedia services are available today through portable devices and even more increasingly accessible in the near future. For instance, Voice over IP (VoIP) and high quality video streaming become an alternative to conventional telephony and TV service in many locations by way of ongoing deployment of WLAN hotspots and even powerful WiMAX coverage. However, present Internet is neither providing adequate Quality of Service (QoS) for users nor ready to become the universal network satisfying all our communication needs.

[0018] The quality of proposed multimedia over BP is mainly affected by throughput, delay, and data loss rate. In VoIP, less than 150 ms of one way delay and 1% of Packet Loss Rate (PLR) is considered as good quality. In contrast to the wired networks, the wireless systems such as IEEE 802.11 WLAN suffer from both delay and packet losses.

Since UDP data does not utilize the transport layer congestion control, congestion and the resulting packet loss can cause substantial degradation to the quality of multimedia applications. In addition, link adaptation could mistakenly adjust the rate as if the loss was due to the poor channel condition and further create more congestion. In G. Bianchi, "Performance analysis of the IEEE 802.11 distributed coordination function," _IEEE Journal on Selected Areas in Communications_, Vol. 18, Issue 3, Mar. 2000, the author shows that the 802.11 system throughput drops significantly once congestion occurs.

To overcome this problem, a necessary add-on packet loss classification (PLC) scheme to support link adaptation under 802.11e Enhanced Distributed Channel Access (EDCA) MAC as described in S. Mangold, S. Choi, G. R. Hertz, O. Klein, B. Walke, "Analysis of IEEE 802.11e for QoS Support in Wireless LANs," _IEEE Wireless Communications_, Vol. 10, pp. 2-12, Dec. 2003, is presented in accordance with one aspect of the present invention. The main objective of the provided PLC is to support ARF in making the following critical decisions, i.e.,

1) only to lower the transmission rate for error prone wireless channel condition, but
2) remain the same rate when congestion is the major cause of frame loss.
[0021] With QoS in mind, packet delays will be analyzed in many aspects for the highest priority VoIP traffic and it will be concluded that the Backoff + Defer Time (BDT) of one complete frame transmission attempt is the best differentiating factor among various contributing factors of wireless delays. Based on known backoff parameters, Contention Window (CW) size, minimum CW (CW min), and maximum CW (CW max), thresholds of congestion levels can be set to determine the channel status and a QoS support link adaptation architecture can be incorporated. Practical VoIP scenarios with simple link adaptation are simulated using NS-2 as available on the Internet at [http://nsnam.isi.edu/nsnam/index.php/Main Page](http://nsnam.isi.edu/nsnam/index.php/Main Page). This will demonstrate the effectiveness of the solution here provided in accordance with a further aspect of the present invention.

[0022] RELATED WORKS

[0023] Link Adaptation

[0024] In order to select proper transmission rates according to the channel condition for a WLAN, several strategies have been proposed for effective link adaptation. Similar to the ARF algorithm as proposed in A. Kamerman and L. Monteban, "WaveLAN-H: A High-performance wireless LAN for the unlicensed band," *Bell Lab Technical Journal*, pp. 118-133, Summer, 1997, a scheme that switches between different transmission rates depending on thresholds in failure and success of transmissions was proposed in P. Chevillat, J. Jelitto, A. Noll Barreto, H.L. Truong, "A Dynamic Link Adaptation Algorithm for IEEE 802.11a Wireless LANs," *IEEE ICC*, 2003. If the success and failure counters exceed certain predefined thresholds, the transmission rate is switched to the next higher or lower data rate, respectively. The optimal thresholds for ARF are analyzed for Voice over Wireless traffic in T. Braskich, N. Smavatkul, S. Emeott, T. Wilson, "Link adaptation evaluation for WLAN using a voice quality metric," *IEEE GLOBECOM*, 2004. These optimal values are determined by using a voice quality metric as described in T. Braskich, N. Smavatkul, S. Emeott, "Optimization of a link adaptation algorithm for voice over wireless LAN applications," *IEEE Wireless Communications and Networking Conference*, 2005 and previously proposed, in response to scenarios of a different number of Voice over Wireless users. Other than ARF methods, D. Qiao, S. Choi, K. G. Shin, "Goodput Analysis and Link Adaptation for IEEE 802.11a Wireless LANs," *IEEE Transactions*
on Mobile Computing, Dec. 2002 presented an algorithm that uses a lookup table to
determine the best PHY rate and fragment size for the next data transmission using the
most up-to-date system status. Another variation of link adaptation can be found in Y.
Sun, A. Chindapol and J. Rosea, "SYSTEM AND METHOD FOR LINK
ADAPTATION FOR WLAN VOICE TRANSMISSION," US patent application

[0025] Through existing link adaptation, failed transmissions from unstable wireless
channel conditions can be reduced, but congestion may increase and hurt the overall
throughput when unsuccessful transmissions are due to collision, resulting from traffic
congestion. Consequently, in congestion dominated circumstances, the data link rate
should maintain high. If the wireless link quality is getting worse, a lower channel rate,
smaller frame size, and higher error protection code ratio should be adopted. Solutions
of two network statuses obviously go opposite way in terms of desired data rates, so
link layer PLC is believed to be helpful toward an effective link adaptation in UDP
congested channel.

[0026] Packet Loss Classification

[0027] In wired/wireless hybrid networks, where wireless is serving as the last-mile
connection, a fair amount of research has been conducted in application layer to
classify congestion loss of wired network and wireless loss to improve the performance
of existing congestion and error control algorithms. As shown in FIG. 1, PLC schemes
assume that packet loss can take place in three network statuses: congested, bad
wireless signal, and mixed. Spike-train as presented in Y. Tobe, Y. Tamura, A.
Molano, S. Ghost, H. Tokuda, "Achieving Moderate Fairness for UDP Rows by Path-
Status Classification," IEEEELCN, Tampa, FL, pp. 252-261, Nov. 2000, and ZigZag as
presented in S. Cen, P. C. Cosman and G. M. Voelker, "End-to-End Differentiation of
Congestion and Wireless Losses," Proceedings of ACM Multimedia Computing and
Networking, San Jose, CA, Jan. 2002 investigate the relative one way trip time (ROTT)
statistics between these classes of packet loss and distinguish them by thresholds. In H-
F. Hsiao, A. Chindapol, J. Ritcey, Y.-C. Chen, J.-N. Hwang, "A New Multimedia
Packet Loss Classification Algorithm for Congestion Control over Wired/Wireless
Channels," IEEE ICASSP, Mar. 2005, the authors take advantage of the fact that an
increasing trend of ROTT can be observed during congestion due to building up packet
queues in wired routers. They also improve the thresholding scheme around the gray zone of two statistic classes which cannot be well differentiated by simple thresholding. [0028] The wireless network can suffer from problems of both congestion and unstable wireless link quality as mentioned before. It was found by the inventors that ROTT and its increasing trend may not be a good indicator to classify packet loss. A link layer approach is thus presented in accordance with a further aspect of the present invention. [0029] To clarify the terms used herein, data units of (and above) the transport layer will be named as packets; lower layer units as frames, but keep the known name of PLC without inventing the term of frame loss classification. Because of the use of the Automatic Repeat-reQuest (ARQ) for all unicast packets in 802.11 MAC, a frame loss is not necessarily corresponding to a packet loss. [0030] NETWORK SETUP [0031] To measure link layer behavior and PLC performance, a network environment using NS-2 simulator was carefully set-up. [0032] IEEE 802.11b PHY Layer [0033] Four data rate options are provided, 1Mbps, 2Mbps, 5.5Mbps, and 11Mbps. The Bit Error Rate (BER) vs. Signal to Noise Ratio (SNR) can be found in C. Heegard, J. T. Coffey, S. Gummadi, P. A. Murphy, R. Provencio, E. J. Rossin, S. Schrum, M. B. Shoemake, "High-performance wireless ethernet," IEEE COMMUNMAG, Vol. 39, no. 11, pp. 64-73. Nov. 2001. Frame errors are inserted according to the transmission rate, signal strength, and frame size. Standard specific parameters such as slot_time, Short InterFrame Space (SIFP), preamble length, PLCP header length, and PLCP data rate are set accordingly. [0034] IEEE 802.11e MAC Layer [0035] The IEEE 802.11e standard introduces the Hybrid Coordination Function (HCF) for QoS support as described in S. Mangold, S. Choi, G. R. Hiertz, O. Klein, B. Walke, "Analysis of IEEE 802.11e for QoS Support in Wireless LANs," IEEE Wireless Communications, Vol. 10, pp. 2-12, Dec. 2003. It defines two medium access mechanisms: contention-based EDCA and Controlled Channel Access (HCCA). The present invention focuses on EDCA where prioritization of different traffic is used. [0036] To achieve QoS, 802.11e uses multiple queues for the prioritized and separately handled Access Categories (ACs) labeled according to their target applications:
AC_VO (voice), AC_VI (video), AC_BE (best effort), and AC_BK (background). Each AC queue works as an independent legacy 802.11 Distributed Coordination Function (DCF) station with its own contention parameters such as Arbitrary InterFrame Space (AIFS) number, CW min, and CW max, where

\[ AIFS = SIFS + AIFS\_number \times slot\_time \]  \hspace{1cm} (1)

Before starting the transmission, the terminal picks the backoff time by selecting a random number between 0 and CW, which is initially set to CWmin. The transmission starts when the medium is idle for at least ABFS and the backoff counter reaches 0. The counter stops when the medium is busy and resumes when the medium is idle for AIFS time.

Because of the difficulty to detect a collision or fading loss from the channel, each transmitted frame needs an acknowledgement (ACK) immediately followed to be considered as a successful transmission. After an unsuccessful transmission with the number of attempt no more than retransmission limit for a frame, an exponential backoff process is performed with double-sized \( CW_{new} \) up to \( CW_{max} \).

\[ CW_{new} = (CW_{old} + 1) \times 2 - 1 \]  \hspace{1cm} (2)


The present implementation herein of delay analysis and PLC mechanism below is based on a TKN 802.11e module as in Sven Wietholter, Christian Hoene, Adam Wolisz, "Perceptual Quality of Internet Telephony over IEEE 802.11e Supporting Enhanced DCF and Contention Free Bursting", *Telecommunication Networks*
Group, Technische Universität Berlin, September 2004 with contention parameters as in
the following Table.

<table>
<thead>
<tr>
<th>Access Categories</th>
<th>AC_VO</th>
<th>AC_VI</th>
<th>AC_BE</th>
<th>AC_BK</th>
</tr>
</thead>
<tbody>
<tr>
<td>AIFS number</td>
<td>2</td>
<td>2</td>
<td>3</td>
<td>7</td>
</tr>
<tr>
<td>CW min</td>
<td>7</td>
<td>15</td>
<td>31</td>
<td>31</td>
</tr>
<tr>
<td>CW max</td>
<td>15</td>
<td>31</td>
<td>1023</td>
<td>1023</td>
</tr>
</tbody>
</table>

[0040] Application Traffic

[0041] QoS demanding VoBP traffic is applied to the WLAN in infrastructure mode
with an Access Point (AP) to route all packets. The VoIP setup simulates a G.711
PCM codec at 64Kbps voice data rate and 20ms per packet. The data rate and packet
size including IP/UDP/RTP headers are 80Kbps and 200 bytes respectively. Each
client with a two-way VoIP link to AP needs 160Kbps of bandwidth and can talk with a
client in or out of the WLAN. TCP connections are used as background traffic in some experiments.

[0042] LINK LAYER DELAY ANALYSIS

[0043] Due to the importance of packet loss information in wireless link adaptation, it is
necessary to understand the fundamental differences between link layer and
conventional end-to-end PLC.

[0044] Similarly, there are three channel conditions resulting in packet loss:
congestion, wireless, and mixed. In hybrid networks, the bottleneck link within an end-
to-end path results in high end-to-end delay and packets dropped at routers' queues.
Therefore, congestion loss carries much higher delay than last-mile wireless loss.
Statistics associated to end-to-end delay (e.g. ROTT) has been proven to be a good
indicator of congestion as in Q. Liu and J. N. Hwang, "A Scalable Video Transmission
System using Bandwidth Inference in Congestion Control," IEEE Int'l Conf On

[0045] Though some packet loss may occur at link layer interface queues in heavily
congested cases, common packet loss is due to collision and poor channel conditions.
In 802.11 MAC, however, those two types of losses, resulting either from congestion or
wireless error, are treated identically by ARQ and CW increasing to assure successful
rate. Regardless of implementation issues, the ROTT of a frame between two stations
from queuing to receiving, which includes all ARQ attempts, cannot tell the difference
of congestion from wireless loss because queuing delay is dominated by the same factor of ARQ.

[0046] In accordance with another aspect of the present invention it is herein presented and simulated that BDT (backoff plus defer time) of one transmission attempt is the best differentiator of congested and bad PHY link status. FIG. 2 shows examples of un-deferred and deferred backoffs. In un-deferred backoff, BDT can be at most backoff max = CWmax x slot_time.

[0047] When defer happens in a backoff process, BDT includes transmission times of stations with smaller remaining backoff numbers and is much higher than backoff max. Since defers appear more frequently if it is heavily competing for the channel with other stations, BDT can be a promising indicator of congestion level (to be defined in a following Section) as seen from the measuring station. The following simulations illustrate the present analysis.

[0048] **Queue-to-ACK vs. Backoff + Defer Time**

[0049] Firstly, BDT with ROTT of frames under congested and bad wireless channel are compared. Since there is no timestamp information provided in MAC header, one has to take advantage of ACKs to derive ROTT. In other words, queue-to-ACK time begins at the packet being placed at the appropriate interface priority queue to the ACK of a successful transmission is received at the transmitting station. This queue-to-ACK time involves queuing, backoff, defer, transmission, and retransmission delays.

[0050] The comparison is based on similar VoIP quality at the application layer measured in PLR and high frame error rate (FER) for possible rate fallback. For congested only cases, SNR is set at SNR =10dB at 11Mbps channel rate for theoretically no wireless loss according to pre-measurements (see the earlier Section on IEEE 802.11b PHY Layer), and use 12 VoIP client stations to fill the channel with 41% FER and 2.8% of overall PLR observed throughout the simulation. Under the same channel rate, SNR = 6dB is set resulting in 48% theoretical FER and 1.64% PLR after ARQ with 5 VoEP clients for a bad wireless channel. Each station runs 10 seconds of VoIP traffic with the join-time being 0.1 seconds apart. The retransmission limit is 7 for both scenarios.

[0051] FIG. 3 plots a moving average with 1 second window width at a VoIP client for 301 Queue-to-ACK time and 302 BDT of congested and bad wireless condition in a 10-
second run. It shows (one-second) moving average of in graphs 301 queue-to-ACK time plots and in 302 BDT plots, respectively, recorded at a client for both cases. Due to the consecutive join-time of stations, more representative and stable behavior can be seen after the 3 second. In 301, both cases have similar values and overlap frequently which imply that congestion and wireless failures create similar amount of $ARQ + queuing$ delay, a large part of queue-to-ACK time. Also, the number of queuing VoEP packets in a client stays low (<2) because of the low incoming packet rate (20ms/packet) and an alike situation is expected in case of the existence of competing flows of lower priorities. But the moving average BDT plots in 302 are clearly separated. The defer time caused by competing traffic from other stations is the dominating factor in the congestion environment.

**[0052] The Access Point vs. Client Stations**

One also expects to see different delay behaviors between AP and clients for the following reasons: 1) the queue builds up at a speed faster than clients as many times as the number of clients. 2) The AP transmits half of the total traffic, so it sees much less channel competition from other nodes than a client.

**[0054]** Fig. 4 shows two plots of moving average with 1 second window width at AP for 401: Queue-to-ACK time and 402: BDT of congested and bad wireless condition in a 10-second run. Similar to FIG. 3 in FIG. 4 401 queue-to-ACK and in FIG. 4 402 BDT plots are measured at the AP. Queue-to-ACK delay goes up dramatically under congestion reflecting its fast queue inflow while staying low consistently in the other case. For BDT, one first notices that the average value is always below 1 ms which is even lower than the no-congestion values of the clients, and does not have significant difference between conditions. One concludes that BDT is not a good indicator for VoIP traffic type of congestion, and queuing delay can be the better one. From link adaptation perspective, however, client-dependent decisions are needed, which is better to be made in distributed fashion at clients.

**[0055] Mixed Classes**

Nevertheless, WLAN is subjected to having both wireless and congestion losses. In this section, various high (low) wireless loss conditions will be setup to represent when one should (should not) perform rate fallback and increase number of
VoIP stations to see the feasibility of using BDT to differentiate the dominant packet loss type.

[0057] BDT is monitored at any one of the VoIP client stations. FIG. 5 shows two experiments conducted with different SNRs corresponding to two theoretical FERs (excluding collision loss) of (a) 48% and (b) 4.8% respectively on the monitored station, while all other stations are with 4.8%. The number of VoIP nodes increases from 1, 5, 8, 10, to 12. One can observe a big jump of BDT between 10 and 12 nodes, where 12 is the number to indicate congestion in former sections, and a 45% of FER is observed in the congested station. Therefore in a link adaptation algorithm with appropriate PLC support, the transmitter in a mixed situation can tell the dominating losing source for correct adaptation. For instance, both suffering more than 40% FER, enough to mislead ARF, the station with low BDT (e.g., 48%, J stations) should go to lower rate since the losing source is wireless error, while the one with high BDT (e.g., 4.8%, 12 stations) will stay at the same rate since the losing source is congestion.

[0058] Threshold Determination

[0059] From the previous analysis, the gap of BDT moving average is apparent between congested/non-congested channels. The threshold $TH_{BDT}$ can thus be determined as:

$$TH_{BDT} = CW \times X_{slot_{time}} + n_{jefer.avg}X_d$$

(3)

where $n_{\phi \_avg}$ is the average number of defers seen in a frame transmission attempt, and $d$ is the average frame transmission duration excluding backoff time, which is already counted in the first term. Intuitively, equation (3) is maximum backoff plus expected defer time. According to the 802.11b standard, $d$ at 11Mbps including all PHY/MAC headers can be calculated as:

$$d = AIFS + d_{Data} + SIFS + d_{ACK}$$

(4)

$$= 50 + 358 + 10 + 304 = 722 \, (\mu s)$$

One can verify this number by estimating the throughput:

$$Throughput = \frac{Data\_packet\_size}{Transmission\_time}$$

$$= \frac{200\, bytes}{AIFS + backoff_{avg} + d_{Data} + SIFS + d_{ACK}}$$

(5)

$$= 1.865 \, Mbps$$
The number is right between the bandwidth needed for 10 (1.6Mbps) and 12 (1.92Mbps) VoIP sessions. In this preliminary algorithm, one empirically selects $a_{deg}.\text{avg} = 2.5$ resulting $\text{Th}_{BDT iiM} = 1.744\text{ms}$. The following Table includes thresholds and characteristics for rate options. Similar BDT gaps are observed in all rates.

<table>
<thead>
<tr>
<th>Rate (Mbps)</th>
<th>11</th>
<th>5.5</th>
<th>2</th>
<th>1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Throughput (Mbps)</td>
<td>1.865</td>
<td>1.558</td>
<td>0.995</td>
<td>0.635</td>
</tr>
<tr>
<td># of congested nodes</td>
<td>12</td>
<td>10</td>
<td>7</td>
<td>5</td>
</tr>
<tr>
<td>$h_{\text{defer.avg}}$</td>
<td>2.5</td>
<td>2</td>
<td>2</td>
<td>1.5</td>
</tr>
<tr>
<td>$\text{Th}_{BDT}$ (ms)</td>
<td>1.744</td>
<td>2.074</td>
<td>3.236</td>
<td>3.870</td>
</tr>
</tbody>
</table>

A LINK ADAPTATION EXAMPLE

To perform a simple link adaptation with PLC support, a scenario is created with stable SNRs and focus on the effect of PLC. The experiment is setup as follows: some VoIP sessions start with 1Mbps channel rate with SNR = 7dB, one of the client suffers interference and SNR drop to 2dB at the 5th second, some new clients join at the 10th second, and the channel becomes better at the 15th second. Performance in terms of PLR and transmission rates with and without PLC is compared.

Assuming ARF select lowers transmission rate in case of 1) low SNR measured, 2) consecutive frame loss and try to go back to a higher rate periodically. FIG. 6 shows to graphs, 601 and 602 depicting in 601 a Link Adaptation Example with PLC and 602 a Link Adaptation Example without PLX. Two solid lines in FIG. 6 601 and 602 show steady rates only ignoring unsuccessful periodical attempts. There are 5 VoIP stations from beginning and one is being monitored. At the 5th second, the client starts to fallback to 2Mbps, reducing theoretical FER from 100% to 3.2%. Due to the ARQ mechanism with a 7 retransmission limit, only few packet losses are observed during the fallback. At the 10th second, 2 more VoIP stations join the WLAN, so the required overall bandwidth is higher than 2Mbps channel can afford (1.12 Mbps : 0.995 Mbps). Several collisions happen, causing a 35.4% FER, and average BDT is detected to be higher than the preset threshold. The station without PLC performs another backoff to 1Mbps due to high FER, while the one with PLC detects congestions and stay at 2Mbps (still suffering 35.4% FER) avoiding high packet loss and can go back to high rate quicker when the signal becomes stronger. The station is hard to go back to 2Mbps for high congestion FER at that rate but still has a chance if congestion relieved a little bit.
Note congestion generates high FER easily because of the small CW size for the highest priority queue.

[0064] The methods for determining a Link Adaptation action such as a Channel Rate change that are part of the present invention can be executed by a system as shown in FIG. 7. The system is provided with data 701 representing relevant data for determining individual and network performance of a WLAN and specifically for determining congestion and wireless performance. An instruction set or program 702 executing the methods of the present invention is provided and combined with the data in a processor 703, which can process the instructions of 702 applied to the data 701 and output a control signal, for instance for adjusting a Channel Rate on a control device 704, which may be a network controller. The processor can be dedicated hardware. However, the processor can also be a CPU or any other computing device that can execute the instructions of 102. Accordingly the system as shown in FIG. 7 provides a system for Link Adaptation.

[0065] In summary, the performance degradation of link adaptation under real-time UDP traffics was considered. Comprehensive analysis of link layer delays has been presented to form a PLC scheme. Thresholds are empirically determined for a simple link adaptation test. Link Adaptation based on a PLC scheme can improve performance of WLAN.

[0066] While there have been shown, described and pointed out fundamental novel features of the invention as applied to preferred embodiments thereof, it will be understood that various omissions and substitutions and changes in the form and details of the device illustrated and in its operation may be made by those skilled in the art without departing from the spirit of the invention. It is the intention, therefore, to be limited only as indicated by the scope of the claims appended hereto.
CLAIMS

1. A method for Link Adaptation in a multi-user WLAN used for VoIP, comprising
determining a channel rate as a function of an error rate, a threshold backoff and defer
time and a measured backoff and defer time.

2. The method of claim 1, wherein the error rate is a frame error rate.

3. The method of claim 1, wherein the threshold backoff and defer time is determined
by the expression \( CW_{\text{max}} \times \text{slot}\_time + n_{\text{frame,avg}} \times d \).

4. The method of claim 2, wherein the threshold backoff and defer time is determined
by the expression \( CW_{\text{max}} \times \text{slot}\_time + n_{\text{frame,avg}} \times d \).

5. The method of claim 2, wherein the channel rate is lowered only if the frame error
rate is above a threshold and the measured backoff and defer time is less than the
threshold backoff and defer time.

6. The method of claim 3, wherein the channel rate is lowered if the frame error rate is
above a threshold and the measured backoff and defer time is less than the threshold
backoff and defer time.

7. The method of claim 6, wherein the channel rate is unchanged if the frame error
rate is above the threshold and the measured backoff and defer time is greater than the
threshold backoff and defer time.

8. The method of claim 4, wherein the channel rate is lowered only if the frame error
rate is above a threshold and the measured backoff and defer time is less than the
threshold backoff and defer time.
9. The method of claim 5, wherein the channel rate is lowered if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

10. The method of claim 9, wherein the channel rate is unchanged if the frame error rate is above the threshold and the measured backoff and defer time is greater than the threshold backoff and defer time.

11. A system, comprising:
   a multi-user WLAN; and
   means for determining a channel rate as a function of an error rate, a threshold backoff and defer time and a measured backoff and defer time.

12. The system of claim 11, wherein the error rate is a frame error rate.

13. The system of claim 11, wherein the threshold backoff and defer time is determined by the expression \( CW_{max} \times slot_time + ndefer,avg \times d \).

14. The system of claim 12, wherein the threshold backoff and defer time is determined by the expression \( CW_{max} \times slot_time + n_{err,avg} \times d \).

15. The system of claim 12, wherein the channel rate is lowered only if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

16. The system of claim 13, wherein the channel rate is lowered if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

17. The system of claim 16, wherein the channel rate is unchanged if the frame error rate is above the threshold and the measured backoff and defer time is greater than the threshold backoff and defer time.
18. The system of claim 14, wherein the channel rate is lowered only if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

19. The system of claim 15, wherein the channel rate is lowered if the frame error rate is above a threshold and the measured backoff and defer time is less than the threshold backoff and defer time.

20. The system of claim 19, wherein the channel rate is unchanged if the frame error rate is above the threshold and the measured backoff and defer time is greater than the threshold backoff and defer time.
FIG. 1
FIG. 2
FIG. 3
FIG. 4
FIG. 6
FIG. 7
INTERNATIONAL SEARCH REPORT

International application No
PCT/US2007/005430

A. CLASSIFICATION OF SUBJECT MATTER
INV. H04L1/00

According to International Patent Classification (IPC) or to both national classification and IPC.

B. FIELDS SEARCHED
Minimum documentation searched (classification system followed by classification symbols)
H04L

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)
EPO-Internal

C. DOCUMENTS CONSIDERED TO BE RELEVANT

<table>
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<th>Citation of document, with indication, where appropriate of the relevant passages</th>
<th>Relevant to claim No</th>
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<td>WO 03/003657 A (KONINKL PHILIPS ELECTRONICS NV [NL]) 9 January 2003 (2003-01-09) abstract</td>
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D. Further documents are listed in the continuation of Box C

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Date of the actual completion of the International search

Date of mailing of the International search report
28 June 2007
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Authorized officer
Bernardini, Andrea
### INTERNATIONAL SEARCH REPORT

**Information on patent family members**

#### PCT/US2007/005430

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