Experimental Results on IP-layer Enhancement to Capacity of VoIPv6 over IEEE 802.11b Wireless LAN

Youngjune Gwon, James Kempf, Raghu Dendukuri, and Ravi Jain
DoCoMo Communications Laboratories USA, Inc.
181 Metro Drive, Suite 300
San Jose, CA 95110
{gyj, kempf, dendukuri, jain}@docomolabs-usa.com

***Abstract*—We present an experimental study to improve capacity of an IEEE 802.11b wireless LAN supporting Voice over IPv6 (VoIPv6). We propose capacity enhancement schemes based on IP (or higher) layer techniques of Robust Header Compression (ROHC) and Silence Suppression (SS), which are distinguished from most of the previous literature discussing MAC layer enhancements. We adopt an existing theoretical capacity model, instantiated with IPv6, ROHC, and SS parameters, and estimate the upper bound on VoIPv6 capacity. We then empirically validate the theoretical results in an experimental IEEE 802.11b wireless IPv6 network. We find that our experimental results are consistent with the theoretical model and the proposed schemes achieve 17 – 20 %, 100 %, and 117 – 140 % capacity improvements for ROHC-only, SS-only, and combined ROHC and SS schemes.**

I. Introduction

Enabling voice communications over IEEE 802.11 wireless LAN has been a focus of rigorous research efforts. Initially, wireless LAN technologies were primarily used for data services. However, today's wireless LAN is being used for voice and other real-time multimedia, as is the case with other radio technologies. The CSMA/CA protocol used for medium access control (MAC) in wireless LAN is not optimal for real-time services. To address this problem, the IEEE 802.11 Working Group established a separate task group (TGe) and defined a set of Quality of Service (QoS) enhancements to the existing MAC, which form the main part of IEEE 802.11e.

Although completion of the IEEE 802.11e specification is under way, it is not clear at this point how many of its effective mechanisms will become incorporated into commercial products. Previous attempts [16, 17] to include a polling-based (TDM-like) MAC mechanism have failed to achieve widespread deployment due to the additional complexity and cost of the MAC implementation. While there are other mechanisms in IEEE 802.11e [13] that enhance real-time QoS of the basic CSMA/CA contention resolution protocol, these mechanisms are less effective in general. Consequently, it is worthwhile to investigate IP (or higher) layer mechanisms that improve voice performance in wireless LAN without requiring any modification at the MAC layer.

In this paper, we focus on Voice over *IPv6* (VoIPv6) over IEEE 802.11b wireless LAN (Vo6WLAN). In particular, we propose the use of two higher layer mechanisms, Robust Header Compression (ROHC) [6] and Silence Suppression (SS), to enhance the *capacity* of VoIP calls. We have evaluated the capacity of a plain Vo6WLAN and the proposed enhancements using both theoretical and experimental methods. We have carried out empirical measurements using commercially available IEEE 802.11b systems. Our experimental results indicate a best-case capacity improvement of 140 % with combined use of ROHC and SS compared to plain Vo6WLAN, which validates our *a priori* estimate computed from the theoretical model.

The rest of the paper is structured as follows. In the following section, we survey related work. In Section 3, we propose the use of ROHC and SS and explain how they can be used to enhance the capacity of Vo6WLAN. Section 4 presents a theoretical capacity analysis. Section 5 describes our experimental methods in detail. Section 6 presents experimental results and comparison with the theoretical results. Section 7 concludes the paper.

II. Related Work

Cole and Rosenbluth [1] proposed a method to assess Voice over IP (VoIP) performance based on reducing the ITU-T's E-Model [11] to transport level measurable quantities. The proposed method is extremely useful for quantifying the impairment factors specified by the E-Model from empirically measurable delay and packet loss.

Prior work in VoIP over wireless LAN is mainly based on analytical modeling and simulation. In Garg and Kappes [2], limitations of the IEEE 802.11a/b distributed coordination function (DCF) for supporting VoIP were investigated. They showed that the maximum allowed simultaneous VoIP sessions (calls) in IEEE 802.11b are limited to 6 (assuming IPv4 and a G.711 voice codec) using an analytical model; the model approximates the upper bound on the number of VoIP sessions based on the IEEE 802.11 MAC-specific properties. In Hole and Tobagi [3], a similar but more complete analytical model that also approximates the upper bound on capacity was derived. They validated the model using simulations. The literature presenting experimental results using the actual IEEE 802.11 devices is relatively sparse in comparison. Anjum et al. [4] presented an experimental VoIPv4 performance study in IEEE 802.11b, which indicated a capacity of 7 calls; they proposed a MAC modification for capacity improvement named 'Back-off Control and Priority Queuing (BC-PQ).' Recent experimental measurements of VoIP capacity for IEEE 802.11b in the industry estimate a maximum of 6 – 7 calls per channel in the presence of 3 Mbps data traffic [18], even with proprietary QoS techniques.

Most work on capacity improvement tries to overcome shortcomings of the standard IEEE 802.11 MAC by proposing modified MAC protocols [4, 14, 15, 16, 17]. These approaches specifically aim to replace the DCF protocol, which is based on CSMA/CA, and causes unacceptably large delays for accessing the wireless channel. The replacements use either priority-based queuing mechanisms [4, 14, 15] that can differentiate packet types, or polling-based schemes [16, 17], such as the Point Coordination Function (PCF) protocol, that better handle streaming traffic. Although PCF is a part of the IEEE 802.11 standard, it is unlikely to be introduced in actual products. Other MAC layer modification proposals off the standard track are even more unrealistic since voice is not the only application over wireless LAN.

Our work differs from others in numerous aspects. We propose the use of compression schemes at IP layer (ROHC) and above (Silence Suppression) to enhance the capacity, which provide no additional complexity and cost concerns for wireless LAN products, unlike existing proposals mandating MAC modifications. We present, to our knowledge, the first experimental study of Vo6WLAN, improve its capacity, construct a theoretical capacity model for *IPv6*, and validate it using the results obtained from common off-the-shelf IEEE 802.11b products. To the best of our knowledge, the previous literature provides none of the above.

III. ENHANCING CAPACITY OF VOIPV6 OVER WIRELESS LAN (VO6WLAN)

A. Robust Header Compression (ROHC)

Robust Header Compression (ROHC) is specified in RFC 3095 [6], which describes highly robust and efficient header compression schemes for RTP/UDP/IP, UDP/IP, and ESP/IP. ROHC is designed to provide a flexible header compression framework capable of supporting multiple protocol stacks by defining packet formats, compression states, modes of operation, error recovery and correction mechanisms, and encoding methods. Figure 1 depicts the case for Vo6WLAN, which compresses RTP/UDP/IP headers carrying voice data over a wireless link.

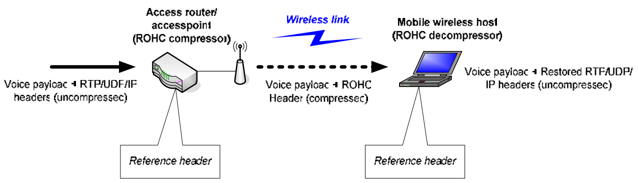

Figure 1. ROHC

ROHC achieves its compression gain by establishing state information at both ends of the link (i.e., compressor and decompressor) through which the compressed data stream transmits. ROHC efficiency depends on the shared property of all packets in a stream having the same field values such as IP addresses, port numbers, and RTP parameters (e.g., payload type). Especially, the ROHC of RTP/UDP/IP establishes functions from the RTP sequence number (SN) to other fields, and then reliably communicates the SN. The complete list and classification of RTP/UDP/IP header fields used by ROHC are detailed in the appendix of RFC 3095 [6].

ROHC is essentially an interaction between two state machines deployed at the compressor and decompressor having three states each. The three states are Initialization and Refresh (IR), First Order (FO), and Second Order (SO). ROHC typically starts in the lowest compression state (i.e., IR) and transits gradually to higher states (i.e., FO and SO). Transitions need not be synchronized between the compressor and decompressor. During FO and SO states, uncompressed RTP/UDP/IPv6 headers of 60 bytes can be fully compressed to a 1 – 2 byte ROHC header while the IR state results with 4 byte-header.

B. Silence Suppression (SS)

More than half of a typical voice conversation is silent. Thus, using a voice codec that generates a constant bit rate regardless of speech activity is bandwidth wasteful. Typically, Speech Activity Detector (SAD) is implemented in voice applications to distinguish talkspurt from silence. In case of talkspurt, the codec generates speech frames in a normal bit rate. When silence is detected, the codec operates in silence suppression either generating minimal bits to encode comfort noise or not generating a bit at all to reduce data transmission. VoIP applications use constant bit-rate (CBR) codecs without SS (e.g., G.711) for broadband channels, but there are other codecs that implement SS, such as Adaptive Multi-Rate (AMR), which is used widely for cellular channels.

IV. THEORETICAL CAPACITY ANALYSIS

Garg and Krappes [2] present a brief mathematical analysis estimating the capacity of a voice-only IEEE 802.11b network. Hole and Tobagi [3] adopted the base mathematical analysis by Garg and Krappes, expanded it to practical upper bound capacity estimation, and validated their estimates using ns-2 IPv4 network simulation. Hole and Tobagi upper bound capacity N (i.e., number of full-duplex VoIP sessions/calls) for IEEE 802.11 wireless LAN is:

$$N = \left\lfloor \frac{1}{R\left[2\left(T_{VOICE} + SIFS + T_{ACK} + DIFS\right) + \frac{T_{SLOT} \cdot CW_{MIN}}{2}\right]} \right\rfloor, \qquad (1)$$

where parameters in (1) are explained in Table I. The denominator of (1) contains sum of two components: 1) time to transmit voice packets from both caller and callee; 2) time to access wireless medium in CSMA/CA. Inverse of the sum scaled by number of packets generated per second (codec-dependent) yields the capacity, whose upper bound is derived by taking the floor value. Further details are found in [3].

Table I. Capacity estimation parameters

Parameters	Description	Value
R	Number of packets generated by voice encoder per second	For G.711 with 10 msec frame duration and no SS, R = 100.
T_{VOICE}	Transmission time of a voice data frame comprising: 1) T_{PLC}: packet loss concealment (PLC) preamble and header processing time; 2) T_{MAC}: MAC header processing time; 3) T_{HEADER}: RTP/UDP/IP or ROHC header processing time.	For IEEE 802.11b having data rate of 11 Mbps, T_{VOICE} is sum of T_{PLC} = 192 usec, T_{MAC} = 20.4 usec, and T_{HEADER} * Voice octets * 8 / 11 usec. T_{HEADER} = total header size (bits) / 11 (Mbps).
SIFS	Short Interframe Space.	For IEEE 802.11b, SIFS = 10 usec

T_{ACK}	Transmission time of acknowledgment comprising: 1) T_{PLC}: packet loss concealment (PLC) preamble and header processing time; 2) T_{ACK_FRAME}: actual transmission time of acknowledgment.	T_{ACK} is sum of T_{PLC} and T_{ACK_FRAME}. For IEEE 802.11b having the peak data rate of 11 Mbps, T_{PLC} = 192 usec, T_{ACK_FRAME} = 10.2 usec.
DIFS	DCF Interframe Space	For IEEE 802.11b, DIFS = 50 usec
T_{SLOT}	Slot duration	For IEEE 802.11b, T_{SLOT} = 20 usec
CW_{MIN}	Number of minimum random slots picked during backoff ~ U(0, CW_{MIN})	For IEEE 802.11b, CW_{MIN} = 31

We consider the following Vo6WLAN schemes: 1) plain (no enhancement); 2) ROHC of voice packets over wireless LAN link; 3) silence suppression (SS) of voice packets; 4) combined use of ROHC and SS. The last three schemes represent proposed capacity enhancements for Vo6WLAN. We estimate Hole and Tobagi upper bound capacity for each scheme based on the following assumptions:

- ITU-T G.711 [5] voice codec generating 80-byte payload for each frame of 10 msec is used.
- R = 100 for Schemes 1 and 2.
- R = 50 for Schemes 3 and 4.
- T_{HEADER} for Schemes 1 and 3 (no ROHC) is: [RTP (12 bytes) + UDP (8 bytes) + IPv6 (40 bytes) headers] / 11 Mbps = 43.64 usec and T_{HEADER} = 43.64 * voice octets (80 bytes) * 8 / 11 = 2538.8 usec.
- T_{HEADER} for Schemes 2 and 4 (ROHC), 2-byte ROHC header (on average) replaces 40-byte RTP/UDP/IPv6 headers: ROHC header (2 byte) / 11 Mbps = 1.455 usec and T_{HEADER} = 1.455 * 80 * 8 / 11 = 84.63 usec.

Table II presents the theoretical upper bound capacity of each Vo6WLAN scheme. The use of ROHC (Scheme 2) over the wireless link increases the capacity of plain Vo6WLAN (Scheme 1) by 1 more VoIP session (i.e., 17 % improvement). The capacity improvement with silence suppression (Scheme 3) is greater with 7 more VoIP sessions than Scheme 1 (or 117 % improvement). Yet, simultaneous use of ROHC and SS (Scheme 4) further improves the capacity, resulting 8 more VoIP sessions than Scheme 1 or 133 % capacity improvement.

Table II. Theoretical upper bound capacity of Vo6WLAN

	Vo6WLAN schemes			
	1 Plain	2 ROHC only	3 SS only	4 ROHC + SS
Upper bound capacity, N	6	7	13	14
Capacity improvement	-	17 %	117 %	133 %

V. DESCRIPTION OF EXPERIMENTAL METHODS

To evaluate the Vo6WLAN schemes presented in Section IV, we consider eight different experimental scenarios as summarized in Table III. A single VoIP session is defined by simultaneous full-duplex voice streams conveying voice conversation between a pair of caller and callee. For each scenario, we start from running 5 VoIP sessions. We then add a session and repeat until the total number of VoIP sessions reaches to 14. Each VoIP session lasts exactly 15 minutes. Scenarios I, III, V, and VII are voice-only. For scenarios with background web traffic (i.e., Scenarios II, IV, VI, and VIII), the number of web clients remains the same as VoIP clients.

Table III. Experimental Vo6WLAN scenarios

Background web traffic	Vo6WLAN schemes			
	1 Plain	2 ROHC only	3 SS only	4 ROHC + SS
No	I	III	V	VII
Yes	II	IV	VI	VIII

A. *Vo6WLAN Testbed*

Figure 2 depicts our experimental testbed. Wireless access is provided by the IEEE 802.11b wireless LAN access point (AP), which is co-located with an IPv6 access router. Wired backhaul features three IPv6 routers connected in 100BaseT Ethernet.

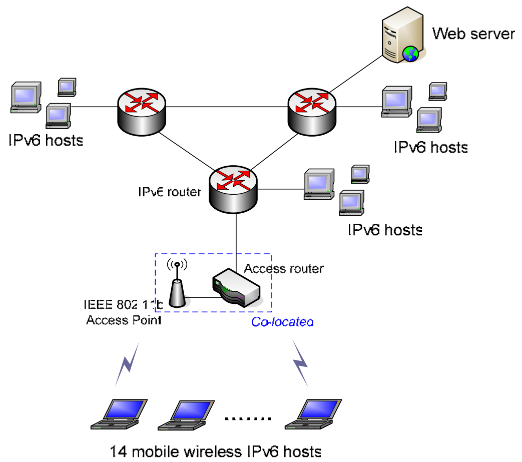

Figure 2. Vo6WLAN testbed

We deploy 14 wireless IPv6 hosts in a couple of feet apart. The wireless hosts are located within 6 – 10 foot range from the AP to ensure that the wireless link quality is good (60 % or higher link quality metric) for most of time. There are also wired IPv6 hosts and a web server connected to a wired Ethernet. Detailed specification of the testbed components are as follows:

- Co-located AP/AR and wireless IPv6 hosts are laptop computers equipped with LinkSys™ IEEE 802.11b WPC 11 client adaptors.
- The co-located AP/AR runs Linux 2.4.20 kernel with HostAP [7] driver.
- The mobile wireless hosts run Linux 2.6.8.1 kernel with HostAP [7] drivers.
- IPv6 routers and stationary hosts are workstations running Linux 2.4.20 kernel.
- All IPv6 hosts run Linux IPv6 implementation [8].
- ROHC is set up between co-located AP/AR and wireless IPv6 hosts.
- Our SS implementation achieves maximum compression gain by generating no voice packets during silence.

B. Speech Generation

We adopt a four-state speech model from ITU-T Recommendation P.59 [9] to generate VoIP sessions. The model details a method of generating artificial conversational speech that is accurate for a real telephone conversation. Summary of the method is as follows:

- Four states are two single-talk (ST) states (caller-talking/callee-silent and caller-silent/callee-talking), mutual silence (MS), and double talk (DT) states.
- Duration of each state varies according to the following equations: i) $T_{ST} = -0.854 \ln(1 - x_1)$ for ST; ii) $T_{DT} = -0.226 \ln(1 - x_2)$ for DT; iii) $T_{MS} = -0.456 \ln(1 - x_3)$; where x_1, x_2, and x_3 are random variables with uniform distribution, $0 < x_1, x_2, x_3 < 1$.
- State transition is based on Figure 3. The values of P_1, P_2, and P_3 are 0.4, 0.5, and 0.5, respectively.

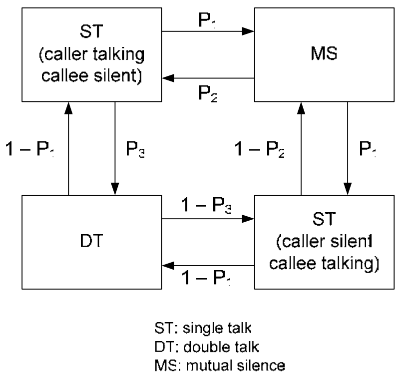

Figure 3. State transition of the speech model

C. Background Traffic Generation

Wireless LAN is presumably a data network for both real-time and non-real time traffic. VoIP performance with the presence of other background data traffic is an important aspect of our investigation. For this reason, we consider a web traffic model based on Choi and Limb [10] and generate HTTP/TCP/IPv6 data traffic in addition to VoIP sessions. Table IV presents the summary.

Table IV. Web traffic generation parameters

Parameters		Mean	Standard deviation	Distribution
Request size (bytes)		360.4	106.5	Lognormal
Object size	Main	10710	25032	Lognormal
	In-line	7758	126168	Lognormal
Number of in-line embedded objects		5.55	11.4	Gamma
In-line inter-arrival time (sec)		0.86	2.15	Gamma
Reading (OFF) time (sec)		39.5	92.6	Weibull

VI. Results and Discussion

In this section, we present experimental performance evaluation for Vo6WLAN. Using the ITU-T E-Model [11], we obtain experimental capacity for each Vo6WLAN scheme. We present a comparative analysis between the theoretical and experimental results.

A. Voice Packet Loss and Delay Performance

Figure 4 depicts downlink voice packet loss over increasing number of VoIP sessions for all experimental scenarios described in Table III. We observe that the plain Vo6WLAN (Scheme 1) results the worst performance reaching nearly 50 % packet loss with 8 VoIP sessions. Scheme 2 (ROHC) outperforms Scheme 1 by 11 – 25 % less packet loss while Scheme 3 (SS) results with further improvement at about 30 – 50 % less packet loss than Scheme 1. Scheme 4 (ROHC + SS) results the best performance with another 5 – 10 % less packet loss than Scheme 3.

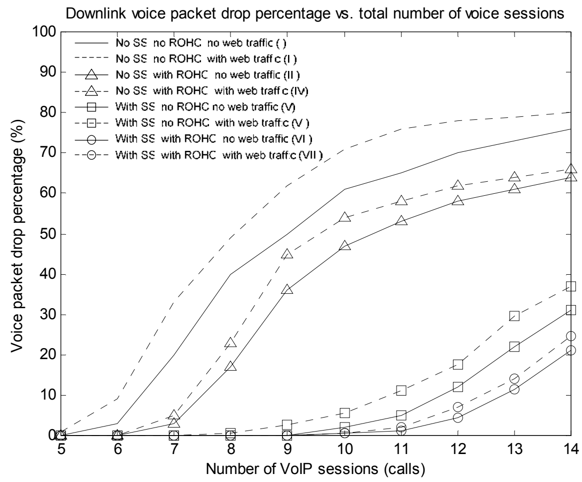

Figure 4. Downlink voice packet loss vs. number of VoIP sessions

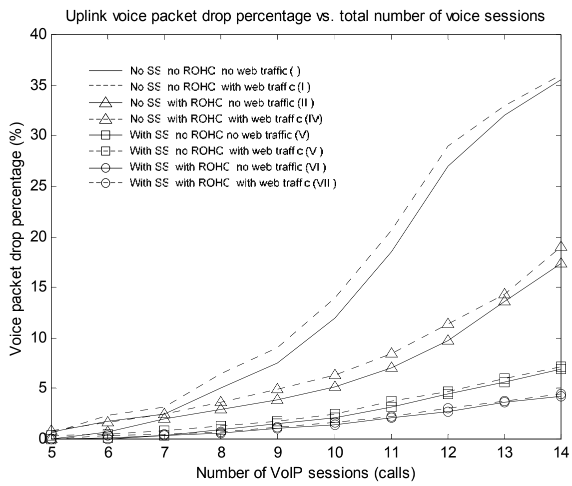

Figure 5. Uplink voice packet loss vs. number of VoIP sessions

The effect of web traffic seems consistent throughout all schemes adding extra packet loss ranging from 1 – 10 %. However, we observe more bursty loss patterns when the web traffic is present. This occurs partly due to flash web requests creating up to the average peak rate of 30 kbps per client. Given 64 kbps constant bit-rate of G.711, the web traffic (TCP) contributes up to 47 % of the entire wireless link causing the overall medium access significantly delayed and packet loss in bursts.

Figure 5 depicts uplink voice packet loss over increasing number of VoIP sessions. In general, the voice packet loss is less than the downlink case by 20 – 40 %. Presence of the web traffic does not seem as significant as the downlink case. This is anticipated since the web traffic is mainly downlink.

As for voice packet delay in our testbed, the longest path between a caller and a callee has 4 IP hops. Mean end-to-end delay for voice packets was 10 msec while the maximum empirical value could reach up to 50 msec. We observed that most delay contribution came from traversing the wireless link and almost negligible delay was added from the wired portion.

B. *Voice Quality Evaluation Using E-Model*

The E-Model [11] is an analytic model to assess voice quality used for network planning purposes. A basic result of the E-Model is the calculation of the R-factor, which is a simple measure of voice quality ranging from a best case of 100 to a worst case of 0. The R-factor uniquely determines the more familiar metric used in telephone speech, namely Mean Opinion Score (MOS), which is the arithmetic average of opinion when “excellent” quality is given a score of 5, “good” a 4, “fair” a 3, “poor” a 2, and “bad” a 1. The R-factor is related to the MOS through the following set of expressions:

- For $R < 0$: MOS = 1
- For $R > 100$: MOS = 4.5
- For $0 < R < 100$: $MOS = 1 + 0.035 R + 7 \times 10^{-6} R (R - 60) (100 - R)$ (2)

For reference, (2) is plotted in Figure 6. Typically, the values of the R-factor are interpreted as shown in Table V.

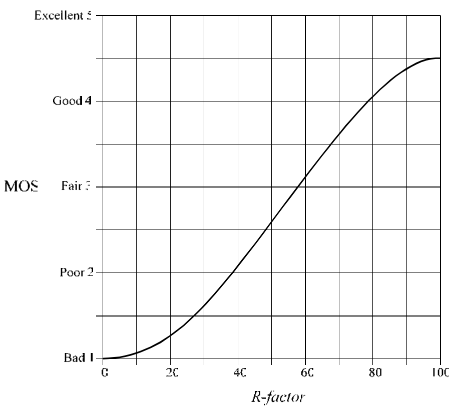

Figure 6. MOS vs. R-factor

Table V. R-factor, voice quality rating, and MOS

R-factor	Voice quality rating	MOS
90 < R < 100	Best	4.34 - 4.50
80 < R < 90	High	4.03 - 4.34
70 < R < 80	Medium	3.60 - 4.03
60 < R < 70	Low	3.10 - 3.60
50 < R < 60	Poor	2.58 - 3.10

The R-factor is expressed as the following equation:

$$R = 93.2 - (I_d + I_e). \quad (3)$$

I_d and I_e are delay and equipment impairment factors associated with mouth-to-ear path (end-to-end) delay and packet loss, respectively. The value of I_d is codec dependent while I_e depends on loss patterns such as random or bursty. Determining I_d and I_e requires a lengthy procedure involving a number of quantities. Cole and Rosenbluth [1] suggest a widely-used method for VoIP, which accurately estimates I_d and I_e values using measured latency and packet loss.

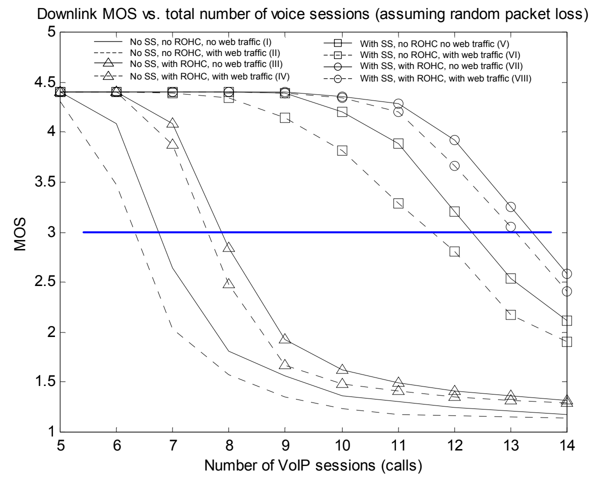

Figure 7. Downlink MOS vs. number of VoIP sessions (random packet loss)

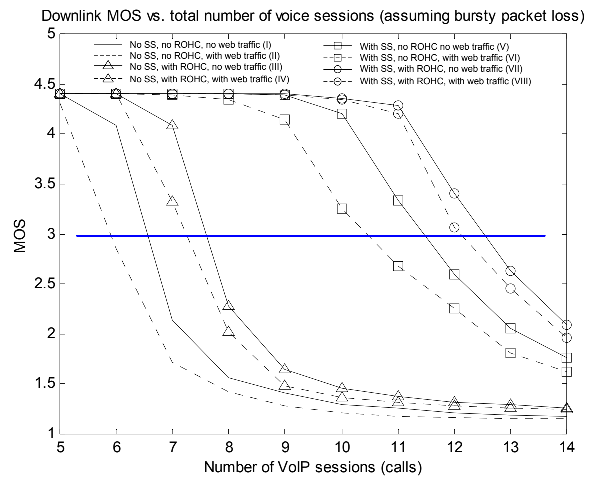

Figure 8. Downlink MOS vs. number of VoIP sessions (bursty packet loss)

C. *Capacity Analysis*

We apply the E-Model and Cole and Rosenbluth method to our empirical measurements and compute MOS for our experimental scenarios. Figures 7 and 8 depict downlink MOS over increasing number of voice sessions on random and bursty packet losses, respectively. Similarly, Figures 9 and 10 depict uplink MOS for random and bursty packet losses. As anticipated, bursty packet loss results with inferior voice quality. The higher I_e values for bursty packet losses lead to lower R-factor and MOS.

Due to the asymmetrical packet loss statistics, the downlink becomes the bottleneck to determine the capacity. Table VI presents the experimental Vo6WLAN capacity for each scheme determined based on the following criteria: 1) MOS > 3.0; 2) I_e is estimated assuming random packet loss for Scenarios I, III, V, and VII (voice-only); 3) I_e is estimated assuming bursty packet loss for Scenarios II, IV, VI, and VIII (both voice and web traffic).

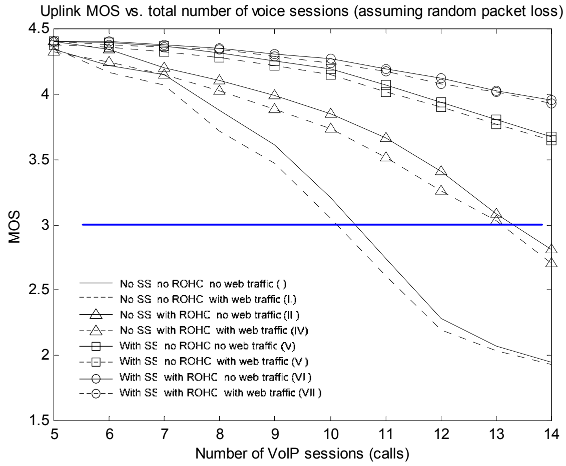

Figure 9. Uplink MOS vs. number of VoIP sessions (random packet loss)

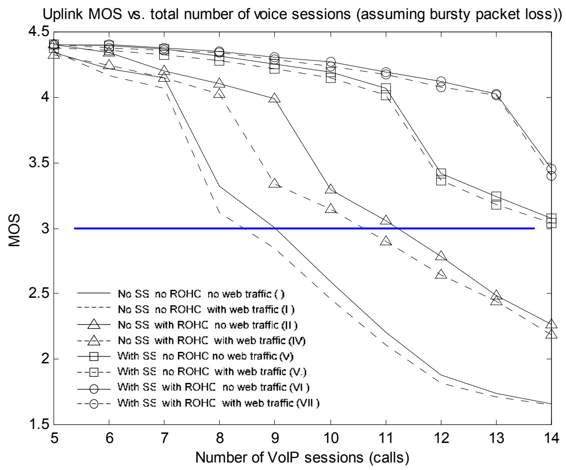

Figure 10. Uplink MOS vs. number of VoIP sessions (bursty packet loss)

The experimental capacity results presented in Table VI seem in accordance with the theoretical upper bound capacity calculated in Section IV (values in parentheses are theoretical figures). There is only 1 session difference between the experimental and theoretical results for the schemes with Silent Suppression (SS). Finally, Table VII presents capacity improvement of Schemes 2, 3, and 4 over Scheme 1 (values in parentheses are theoretical figures).

Table VI. Experimental (theoretical) capacity of Vo6WLAN schemes

Background web traffic	Vo6WLAN schemes			
	1 Plain	2 ROHC only	3 SS only	4 ROHC + SS
No	6 (6)	7 (7)	12 (13)	13 (14)
Yes	5	6	10	12

Table VII. Experimental (theoretical) capacity improvement of proposed schemes

Background web traffic	Vo6WLAN schemes		
	2 ROHC only	3 SS only	4 ROHC + SS
No	17 % (17 %)	100 % (117%)	117 % (133 %)
Yes	20 %	100 %	140 %

VII. CONCLUSIONS AND FUTURE WORK

We presented an experimental study of VoIPv6 over IEEE 802.11b wireless LAN. In particular, we focused on capacity improvement and proposed three IP-layer enhancement schemes, using Robust Header Compression (ROHC), Silence Suppression (SS), and combined ROHC and SS, which are distinguished from existing MAC layer-based enhancement schemes. We adopted Hole and Tobagi's theoretical model, instantiated with IPv6, ROHC, and SS parameters, and estimated the theoretical upper bound on capacity for each scheme. We then experimentally validated the theoretical results using a real IEEE 802.11b wireless IPv6 network. For further work, our investigation on the following issues is in progress: 1) use of low-bit rate codecs and impact on capacity; 2) use of higher layer QoS mechanisms (e.g., DiffServ); 3) theoretical model for VoIP capacity with presence of data traffic; 4) mobility of voice users.

REFERENCES

[1] Cole, R., G.., and Rosenbluth, J., H., "Voice over IP Performance Monitoring," Proc. of ACM SIGCOMM, August 2001
[2] Garg, S., and Kappes, M., "Can I Add a VoIP Call?," Proc. of IEEE ICC, May 2003
[3] Hole, D., P., and Tobagi, F., A., "Capacity of an IEEE 802.11b Wireless LAN Supporting VoIP," Proc. of IEEE ICC, June 2004
[4] Anjum, F., et al., "Voice Performance in WLAN Networks – An Experimental Study," Proc. of IEEE GLOBECOM, December 2003
[5] ITU-T Recommendation G.711, "Pulse Code Modulation (PCM) of Voice Frequencies," 1993
[6] Bormann, C., et al., "Robust Header Compression (ROHC): Framework and Four Profiles – RTP, UDP, ESP, and Uncompressed," RFC 3095, July 2001
[7] HostAP Driver for Intersil Prism2/2.5/3 and WPA Supplicant, http://hostap.epitest.fi/
[8] Universal Playground for IPv6 (USAGI) – Linux IPv6 Development Project, http://www.linux-ipv6.org/
[9] ITU-T Recommendation P.59, "Artificial Conversational Speech," March 1993
[10] Choi, H., and Limb, J., "A Behavioral Model of Web Traffic," Proc. of International Conference on Network Protocols (ICNP'99), October 1999
[11] ITU-T Recommendation G.107, "The E-Model, a Computational Model for Use in Transmission Planning," July 2002
[12] ITU-T Recommendation G.113, "Transmission Impairments Due to Speech Processing," February 2001
[13] IEEE 802.11 Working Group, http://grouper.ieee.org/groups/802/11/
[14] Aad, I., and Castelluccia, C., "Differentiation Mechanisms for IEEE 802.11," Proc. of IEEE INFOCOM, April 2001
[15] Suzuki, T., and Tasaka, S., "Performance Evaluation of Priority-based Multimedia Transmission with PCF in an IEEE 802.11 Standard Wireless LAN," Proc. of IEEE PIMRC, October 2001
[16] Yeh, J., Y., and Chen, C., "Support of Multimedia Services with IEEE 802.11 MAC Protocol," Proc. of IEEE ICC, April 2002
[17] Veeraraghavan, M., et al., "Support of Voice Services in IEEE 802.11 Wireless LANs," Proc. of IEEE INFOCOM, October 2001
[18] Newman, D., "Aruba is the top dog if you want to add voice traffic to an enterprise wireless LAN," NetworkWorldFusion Online Article on Voice over WLAN, http://www.nwfusion.com, January 2005