ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

Comparative call Capacity analysis of VOIP in IEEE802.11b WLAN Environment

Jitendra Jakhar[1], Sankit R Kassa[2], Tejpal[3]

PG Student, Electronics & Communication Engineering Department, SBCET, Jaipur, Rajasthan, India[1]

Assistant Professor, Electronics & Communication Engineering Department, SBCET, Jaipur, Rajasthan, India[2]

Electronics & Communication Engineering Department, Manipal University Jaipur, Rajasthan, India[3]

Abstract: In recent years, Voice over Internet Protocol (VoIP) is growing very fast and is providing a cost effective alternative for Voice communication. VoIP over WLAN is poised to become an important Internet application. Since numbers of VoIP client over WLAN is increasing at a fast pace, the number of simultaneous VoIP streams supported by IEEE WLAN standard is an important area of study. With each VoIP stream typically requiring less than 10kbps, an IEEE802.11b WLAN operated at 11Mbps could in principle support 550 VoIP sessions, but in practice it provides very few VoIP sessions, typically 8 to 10. IEEE 802.11b standard operates in two modes i.e. Distributed Co-ordination Function (DCF) and Point Co-ordination Function (PCF). Although, both modes of operation could work for any type of traffic (Data traffic, Real-time Traffic, Video traffic), but PCF is more suitable real time traffic (Voice).

The reason for low capacity is the small packet size of VoIP packets and a large overhead at MAC layer. Many schemes, such as PCF, Modified PCF, CSSR, ICF and MICF have been proposed to reduce polling overhead to enhance call capacity. In this paper call capacity performance of PCF, Modified PCF, CSSR, ICF and MICF have been evaluated and compared using MATLAB simulation.

Index Terms: VoIP, Wireless LAN, IEEE 802.11, DCF, PCF, Capacity, Bandwidth.

1. INTRODUCTION

VOIP over WLAN is becoming a very attractive solution for wireless voice communication. It uses the existing packet switched data network for transporting the VoIP packets and provides a low cost alternative to the traditional telephonic system. Wireless LAN standard 802.11 uses two modes for wireless channel access. These are Distributed Co-ordination function (DCF) [1] and Point Co-ordination Function (PCF)[1]. DCF mode is based on random access of channel that is best suited for non-real time traffic i.e. bursty traffic and PCF mode is based on polling mechanism that is best suited for real time traffic but most of the devices do not support PCF mode. During early years of WLAN deployment DCF mode was mostly supported in WLAN devices but in recent years the importance of the PCF mode is being recognized and now the PCF mode is also being supported (for real time traffic) in new devices like Laptop, PDAs. The DCF mode is based on CSMA/CA (carrier sense multiple access with collision avoidance). The timing diagram of DCF scheme is depicted in figure 1.

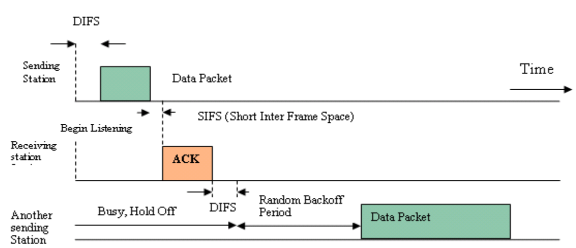


Figure 1 IEEE 802.11 DCF Scheme

In the DCF mode [1] control to the access of channel is distributed among all the stations. The DCF access method is based on the CSMA/CA principle in which host wishing to transmit senses the channel, to check if it is free. On finding the channel free the host waits for a random amount of time (to avoid two hosts starting transmission at the same time.) before transmitting.

In the PCF mode [1] of operation the access of the wireless channel is centralized by using polling based protocol controlled by the Point Coordinator. The Access points generally serve as PCs. The PCF mode provides contention free service to the wireless Stations. In PCF mode a frame is divided in two parts, contention free period (CFP) and

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

contention period (CF). The PC indicates the start of the contention free period by sending a beacon frame that contains the list of pollable stations and other polling management information. The CFP is repeated after a fixed interval. The CFP and CP together constitute a super frame whose structure is as shown in figure 2(a).

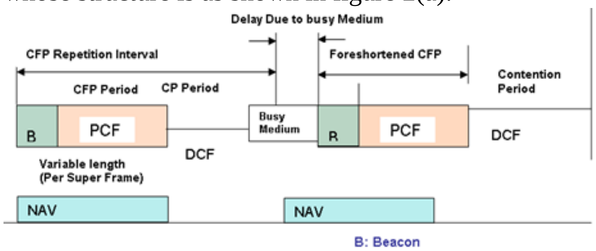


Figure 2(a) Basic PCF Mode of operation

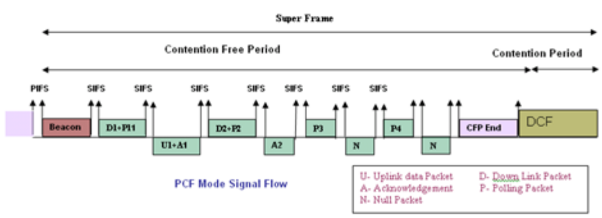


Figure 2(b) Flow of signals And data in PCF Mode

As shown in figure 2(b), after sending the Beacon the PC starts polling stations one by one in the order indicated in the beacon. In CFP, if the PC has a data packet to send to the station it sends the polling packet piggybacked on the data packet and if the PC does not have any data to send then it sends only a polling packet. The polled station responds by sending the uplink ACK packet and piggybacks any uplink data on the ACK packet. If Polled station does not have data to send in the uplink then it just sends a NULL packet in response to the poll by PC. In this scheme, some of the bandwidth will be used only for Polling and ACK which is wasted. Here in figure 2(b) stations P3 and P4 do not have any uplink and downlink data, but even then the PC polls these stations resulting in wastage of bandwidth.

These drawbacks of the basic PCF mode limit the number of simultaneous VoIP calls. There are several proposals given by various authors, details of which are available in papers like Dynamic PCF [11], Adaptive PCF [8], and Modified PCF [6], etc which improve call capacity.

2 MODIfiED PCF

As illustrated in Figure 3 during the CFP in the Modified PCF the channel transmission time comprises two transmission periods: the distributed polling protocol period (DPPP) and the real-time traffic downlink period (RTDP)[6]. The length of the DPPP and RTDP in the Modified PCF is equal to the length of the CFP in the standard PCF. Since the CFP interval provided in the standard has to be shared between the DPPP and RTDP, their length should be equal.

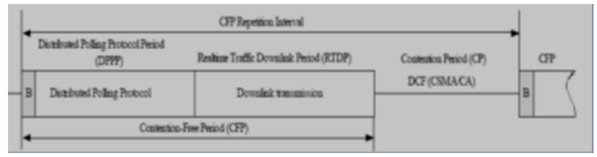


Figure 3 Modified PCF Timing Interval

However, if the DPPP does not fully use its allocated time, the remainder can be lent to the RTDP. The wireless stations send real-time traffic during uplink traffic during DPPP and, the PC sends real-time traffic during the RTDP. The RTDP traffic includes downlink traffic and real-time traffic that is relayed by AP from one station to another. In spite of sending polling frame to each station, AP creates a polling list which is sent in beacon frame to each station. So, each station gets their transmission order in beacon frame. The main task in MPCF is making polling list. To enter in polling list, the station sends an association request to the access point during the CP.

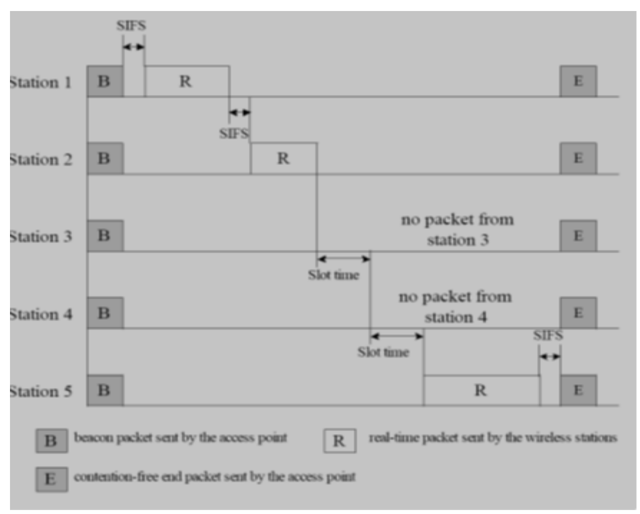


Figure 4 MPCF Access Procedure

To confirm that the station has been added to the polling list, the PC returns a polling identification and transmission order assignment. Stations that are already in the polling list updates with new transmission order after receiving a successful association. The channel access procedure of MPCF is described in figure 4. It is possible that some stations in the polling list are not able to transmit during the DPPP because the end of the DPPP period is reached before their turn. Therefore, each station in the polling list has to

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

circularly shift its transmission order after it has received the contention-free end packet. These proposals seek to overcome the call capacity deficiencies of the PCF mode of operation thereby providing capability to the WLAN network to accommodate more number of simultaneous VoIP calls. One of the proposed techniques introduces a new modified multiple access mechanism to improve the capacity termed as Isochronous Coordination Function.

3 ICF

Isochronous Coordination function was introduced to handle real time traffic and especially the voice traffic. The voice traffic is correlated to some extent and therefore voice data of some lost packets can be reconstructed from the other received voice packets. For providing acceptable quality of the service the lost packet number should be no greater than 1% of the total number of packets sent by a particular station. This characteristic of voice traffic is exploited by the ICF technique which trades off delay with packet loss. To ensure QoS constraints, it is taken care that no consecutive packet loss occurs. The AP initiates an ICF cycle during the optional CFP or during CP by sending the ICF poll frame. ICF polling frame consist of a status vector (SV). Each VoIP node is allocated a polling bit in SV at the time of association. After receiving ICF polling frame each station checks its bit in status vector and based on the status (0/1) of that bit and numbers of zero's in SV [9] before their bit, stations calculate the time slot allotted to them for transmission. Due to limited numbers of time slots in an ICF cycle, all station may not be polled, so an efficient polling list management is implemented by using cyclic polling queue [9]. Stations which do not get time slot in ICF cycle for transmission, discard their packet. The cyclic polling queue management ensures that there will not be too many consecutive packet loss.

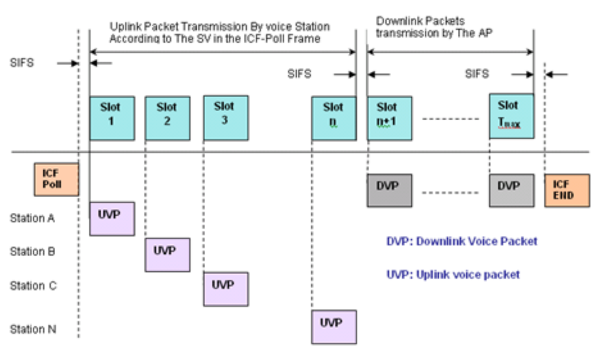


Figure 5 Isochronous Coordination Function

4 MODIFIED ICF

To improve the performance of ICF scheme a Modified ICF (MICF) scheme for channel access has been proposed. Here the M-M scheme in downlink stream is used. This scheme exploits the fact that there is an opportunity with the access point to multiplex the download packets from the Different downlink STAs and multicast them using a multicast IP address. This will reduce the overhead from that of multiple VoIP packets to that of a single packet (thereby resulting in better bandwidth utilization). Also, this scheme saves the time period corresponding to SIFS intervals between the two adjacent time slots (for data to be sent in the downlink direction). The Modified ICF scheme is as shown in figure 6 saves large amount of MAC and PHY layer overheads by using a single packet rather than multiple packets with their individual overheads. Also, the MICF scheme reduces the RTP, UDP, IP (combined header size of 40 bytes) header to a compressed mini Header of 2 bytes. In the Modified scheme, at the start of the ICF cycle, the uplink stations send the packets according to the entries in the SV. When all uplink transmission is complete, the access point senses the channel free for SIFS time interval and then takes the control of channel to transmit the downlink voice traffic.

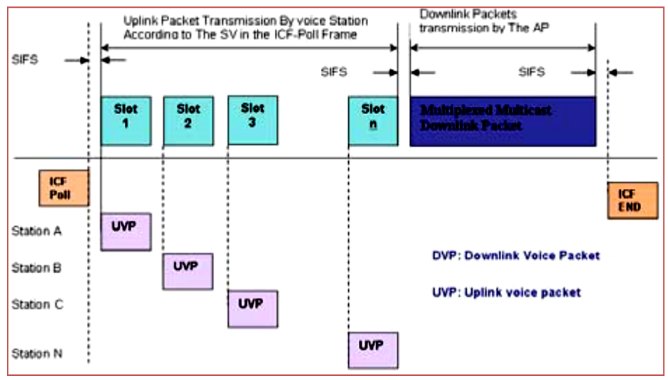


Figure 6 Modified ICF

5 CAPACITY ANALYSIS

IEEE 802.11 Capacity Analysis: A constant bit rate (CBR) [6] VoIP client generates one VoIP packet every packetization interval. Therefore, the numbers of packets that can be sent during one packetization interval is the maximum numbers of calls that can be supported. The capacity of VoIP can be calculated by the following equation:

$$N_{max} = T_p / (2T_t) \quad \ldots\ldots \ldots\ldots\ldots\ldots \quad (1)$$

Where Nmax is the maximum numbers of calls, Tp is the packetization interval, and Tt is the time for sending one packet of voice. The reason for Tt being multiplied by 2 is that the voice communication is full duplex. Nmax can be higher if we account for the fact that normally we do not have voice data in both directions simultaneously. Tp depends upon the codec used in the VoIP client. Table I lists these values for typical codecs.

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

TABLE I

Codec	Packet Duration (ms)	Packet Size (Bytes)
G.711	20	200
G.729	20	60
G.723.1	30	64
GSM	20	73

5.1 VoIP Capacity of PCF

To avoid the delay, VoIP station needs to be polled every packetization interval, which means CFP cannot be more than the packetization interval. Therefore the Nmax is maximum numbers of station that can be polled in CFP, which can be calculated by following equation:

Nmax = (TCFP-TB-TCE)/ (2Tp)

= (TCFP-TB-TCE)/ (2(Tv+TSIFS)) …………..... (2)

where TCFP, TB, TCE, Tv and TSIFS are the durations of contention free period, beacon frame, contention free period end frame, transmission time for voice packet and SIFS period respectively. Ordinarily, in voice communication uplink and downlink stations do not transmit voice packet simultaneously. Therefore, polling the STA for uplink data in the frame in which downlink data for that STA is transmitted is not an efficient method of polling as it will result in unnecessary polls. So, CFP is further divided into uplink transmission period and downlink transmission period. In uplink period, CF-Pollable STAs are polled according to the polling list management scheme implemented in AP.

TABLE II

IEEE 802.11b Parameters	Values
Data Rates for Data Packets	11Mbps
Data Rates for Control packets and PHY overheads	1Mbps
MAC layer Overheads	34 Bytes
PHY layer overheads	24 Bytes
Transmission Time for TB	744µs
Tx Time for Null Frame	464µs
Tx Time for Polling Frame	464µs
Tx Time for CF End Frame	352µs
Tx Time for payload (T_v)	609.45µs
CFP Contention Free Period	15ms-19ms
SIFS Interval	10µs
Slot Time Interval	20µs
Probability of ON Period	0.43
Probability of OFF Period	0.57

The parameter values listed in Table II, are for the G.711 codec, with voice traffic modeled as markov bi-state [7].

If assigned uplink transmission period is not fully utilized by the stations in the uplink polling list, the remaining duration is utilized to send downlink voice data to STAs which do not appear in the downlink polling list. Downlink traffic is transmitted using FIFO mode. Therefore Nmax can be calculated using following equation:

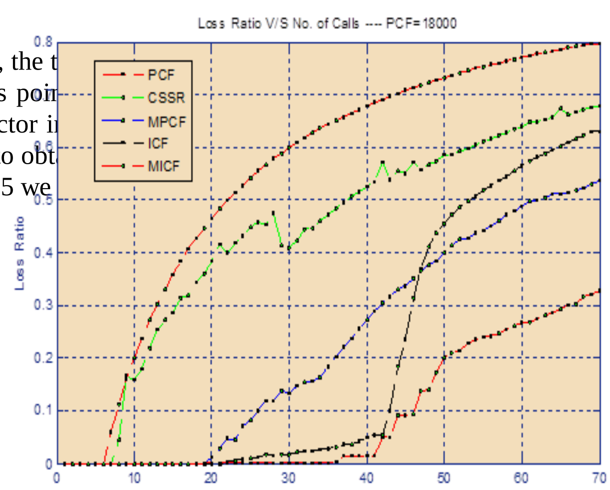
Nmax=(0.5(TCFP-TB-TCE))/(Tv+TP+2TSIFS) ……….(3)

Here, Tp is transmission time for polling frame.

5.2 VoIP Capacity of ICF

If we compare the time required for sending the voice traffic and polling frame, it becomes apparent that polling each STA individually constitutes a very large overhead. This procedure becomes even more inefficient when some of the stations do not have voice packet to send (here a polling frame is sent and a NULL frame is sent as response, either of these packets does not carry any useful traffic). Calculation shows that only one additional STA can be polled when three STAs do not have voice traffic to transmit.

In ICF mode, the t
by the access poi
the status vector i
information to obt
Using figure 5 we



ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

Nmax = (TCFP-TB-TCE)/(Tv+TSIFS) ……………..(4)

6. SIMULATION RESULTS

This section presents some simulation results to compare the ICF with the existing basic PCF, and MPCF schemes. Using the information provided in tables and equations in the previous sections, the call capacity (number of simultaneous voice calls) has been calculated. Figure 7 & 8 shows a comparison between basic PCF, CSSR, MPCF, ICF and MICF. In this simulation, the CFP is taken as 15 ms and frame repetition interval as 20 ms. Figure 7 & 8 represents loss ratio as a function of number of simultaneous voice calls. As is evident from the plot, the MICF scheme has the lowest loss ratio for a given number of simultaneous calls. A more important measure of the efficacy of scheme is the number of simultaneous calls while maintaining the QoS requirement. Generally, a loss ratio of less than 0.01 results in acceptable QoS. For this loss ratio MICF can support more number of simultaneous calls as compared to MPCF and ICF. Table III lists the call capacity of the various schemes for different values of CFP interval.

TABLE III

CFP	N_{max} (Simulated)				
	PCF	CSSR	MPCF	ICF	MICF
15 ms	6	8	19	41	44
18 ms	7	8	26	43	47

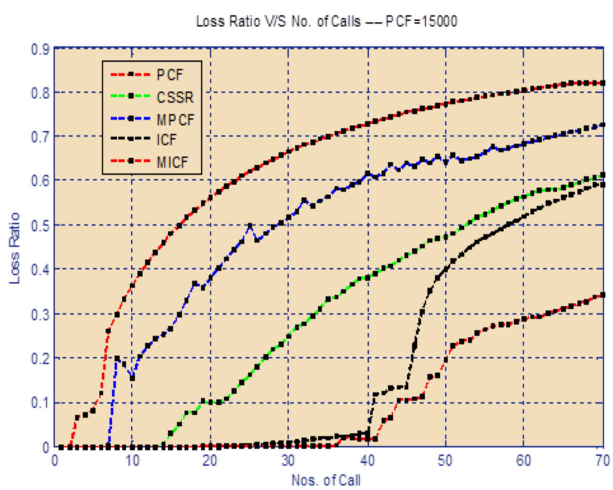


Figure 7 Comparison of PCF, CSSR, MPCF, ICF and MICF at CFP=15ms

Figure 8. Comparison of PCF, CSSR, MPCF, ICF and MICF at CFP=18ms

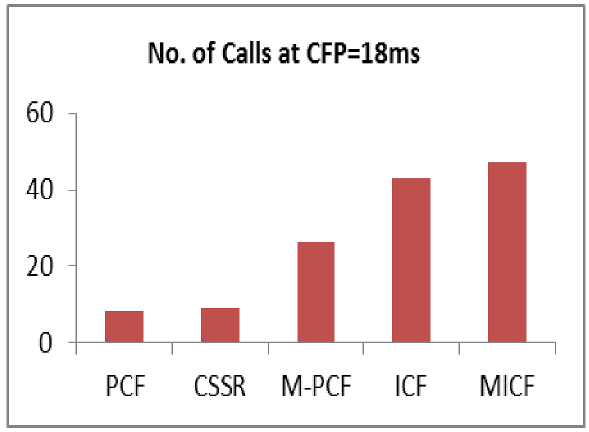


Figure 9 No. of calls at cfp =18ms

7. CONCLUSIONS

In this paper we have measured the performance of the PCF, Modified PCF,CSSR, ICF and MICF by analyzing the number of supportable stations in voice traffic, and comparing it. The simulation results shows that the MICF can support a much higher number of wireless stations than that of the standard PCF, Modified PCF and ICF. This has been widely used in VoIP. This indicates that the channel utilization of the MICF is significantly better than the standard PCF, Modified PCF and ICF.

REFERENCES

[1] IEEE 802.11, 1999 Edition, IEEE Standards for Information Technology – Telecommunication and Information Exchange between Systems – Local and Metropolitan Area Network – Specific Requirement – Part 11: "Wireless LAN Medium Access Control (MAC) and Physical Layer (PHY) Specification".

[2] Praveen Gopalkrishnan, David Famolari, Toshikazu Kodama, "Improving Wlan voice capacity through dynamic priority Access" ,pg.3245-3249, in Globecom 2004

[3] A. Kopsel and A. Wolisz, "Voice Transmission in an IEEE 802.11 WLAN Based Access Network," Proc. ACM WOWMOM'01, July 2001, pp. 23–32.S. Drabowitch, A. Papiernik, H. Griffiths, J. Encinas and B. L. Smith, "Modern Antennas", 1st Edition, Chapman & Hall, 1998.

[4] D. P. Hole and F. A. Tobagi. Capacity of an IEEE 802.11b wireless LAN supporting VoIP. In IEEE ICC, pages 196–201, 2004.Huseyin Arslan, Zhi Ning Chen, Maria-Gabriella Di Benedetto, "Ultra Wideband Wireless Communication", John Wiley & Sons. Inc, 2006

[5] H.-H. Liu and J.-L. C. Wu, "A Scheme for Supporting Voice over IEEE 802.11 Wireless Local Area Network," Proc. Natl. Sci. Council ROC(A), vol. 25, no. 4, 2001, pp. 259–68.M. Ojaroudi, Sh. Yazdanifard, N. Ojaroudi, and R. A. Sadeghzadeh,Band-Notched Small Square-Ring Antenna With a Pair of T-Shaped Strips Protruded Inside the Square Ring for UWB Applications", IEEE ANTENNAS AND WIRELESS PROPAGATION LETTERS, VOL. 10, 2011.

[6] L. Zhao and C. Fan, "M-PCF modified IEEE 802.11 PCF protocol implementing QoS", Electronic Letters, Nov. 2002. pp.1611-1613.

[7] P. Brady, "A Model for Generating On-Off Speech Patterns in Two-Way Conversation," Bell Syst. Tech. Journal, vol. 48, no. 7, pp. 2245-2272, Sept. 1969.

ISSN (Online) 2321 – 2004
ISSN (Print) 2321 – 5526

INTERNATIONAL JOURNAL OF INNOVATIVE RESEARCH IN ELECTRICAL, ELECTRONICS, INSTRUMENTATION AND CONTROL ENGINEERING
Vol. 1, Issue 9, December 2013

[8] Y.-J. Kim and Y.-J. Suh, "Adaptive Polling MAC Schemes for IEEE 802.11 Wireless LANs Supporting Voice-over-IP (VoIP) Services," Wireless Commun. Mob. Comp., vol. 4, 2004, pp. 903–16.
[9] Ray Y.W.Lam, Victor C.M.Leung,"Polling Based Protocols for Packet Voice Transport Over IEEE 802.11 Wireless local Area Networks", pg 22-29, IEEE Wireless communication ,February 2006
[10] Wei Wang, S.Chang Liew and Victor O.K.Li, "Solutions to Performance Problems in VoIP Over a 802.11 Wireless LAN", IEEE Transaction on vehicular technology, volume 54, pp. 366 – 384, Jan 2005.
[11] Takehiro Kawata, Sangho Shin, Andrea G. Forte, "Using Dynamic PCF to improve the Capacity for VoIP Traffic in IEEE 802.11 Networks" IEEE Communications Society / WCNC 2005, pp. 1589 -1595
[12] Jonathan Davidson, James Peters, Voice over IP Fundamentals, Cisco press 2001 edition.
[13] Oliver Hersent, Jean-Pierne petit, David Gurle, Beyond VoIP Protocols, John Wiley and sons, 2005