ELECTRONICS AND ELECTRICAL ENGINEERING

ISSN 1392 – 1215 2009. No. 7(95)

ELEKTRONIKA IR ELEKTROTECHNIKA

T180 TELECOMMUNICATIONS ENGINEERING

TELEKOMUNIKACIJŲ INŽINERIJA

Research of Real Time Traffic Transmission in 802.11 WLANs

D. Pauliukas, V. Vosylius

Department of Telecommunications, Kaunas University of Technology,
Studentų str. 50-453, LT-51368 Kaunas, Lithuania, e-mail: donatas.pauliukas@stud.ktu.lt, valdas.vosylius@stud.ktu.lt

Introduction

Transmitting real time traffic, such as VoIP, is challenging task for IEEE 802.11 wireless networks, because initially this wireless customer access technology was designed for best-effort traffic, such as web browsing, emails, etc. This paper concentrates on transmitting real time traffic over the 802.11b/g and presents wireless network's capabilities to handle different types of voice codecs.

Coordination Functions

Access to the wireless medium is controlled by coordination functions. Ethernet-like CSMA/CA (Carrier Sense Multiple Access With Collision Avoidance) access is provided by the distributed coordination function (DCF). The DCF is the basis of the standard CSMA/CA access mechanism and like Ethernet, it first checks to see that the radio link is clear before transmitting. To avoid collisions, stations use a random backoff after each frame, with the first transmitter seizing the channel. This mechanism ensures that all transmitting stations have equal rights to access the medium and therefore it does not ensure the quality of such services as VoIP [2].

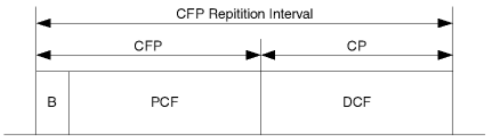

Fig. 1. PCF and DCF coexistence [2]

Point coordination function (PCF) provides contention-free services. To gain priority over standard contention-based services, the PCF allows stations to transmit frames after a shorter interval. The PCF is not widely implemented, because so called point coordinators have to reside in access points (AP). That is why the PCF is restricted to infrastructure networks [1].

Contention-free Period (CFP), controlled by central authority, is used to set the frequency of the PCF. Even when 802.11 network provides contention-free services, some contention-based (CP – Contention Period) access to the wireless medium is also allowed. The beacon (B) sent by the transmitting station shows the beginning of the CFP.

Subject and purpose of the research

In this paper the IEEE 802.11b/g networks are under investigation, where *k* number of stations are connected to one access point and using VoIP service. Analytical model is proposed, which lets to calculate how many stations can simultaneously use the VoIP service in such kind of networks. Also shown, how changing the CFP values does effect the number of stations in one WLAN environment, described above. Results are shown to some 802.11b/g network data rates. Also here is presented the efficiency of the wireless media and VoIP data rate dependencies.

802.11b media parameters and dependencies

One of the most important parameter, which affects the VoIP conversation quality of service and shows it‘s capabilities to be transmitted over the wireless media, is the size of the voice packet. It consists of digitalized voice and headers of various protocols:

$$l_{G.711\,MAC} = l_{G.711} + l_{MAC} + l_{LLC} + l_{IP} + l_{UDP} + l_{RTP}\,, \quad (1)$$

where $l_{G.711\ MAC}$ – G.711 packet size of the codec with headers [B], l_{RTP} = 12 [B] – RTP header size, l_{UDP} = 8 [B] – UDP header size, l_{IP} = 20 [B] – IP header size, l_{LLC} = 8 [B] – LLC header size, l_{MAC} = 34 [B] – MAC header size.

In WLANs the CFP interval starts when access point sends a beacon. It takes time, presented in formula bellow:

$$\tau_B = \frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}} + 8\frac{l_B}{c}\,, \quad (2)$$

where τ_B – beacon transmission time [s], $l_{PLCPpreamble}$ – PLCP preamble size [b], $l_{PLCPheader}$ – PLCP header size [b], $c_{PLCPpreamble}$ – PLCP preamble transmission rate [bps], $c_{PLCPheader}$ – PLCP header transmission rate [bps], l_B – beacon size [B], c – wireless media data rate [bps].

The end of CFP interval is marked by the CF-End frame

$$\tau_{CF-End} = \frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}} + 8\frac{l_{CF-End}}{c}, \qquad (3)$$

where $\tau_{CF\text{-}End}$ – CF-End frame transmission time [s], $l_{CF\text{-}End}$ – CF-End frame size [B].

In one period, between these two frames the connection for data transmission (refer 4 and 5 equations) is established between a station and AP. The difference is in the CF-Poll frame transmission – it is performed only by AP. In these calculations is assumed that this frame is sent over the media every time when AP is receiving the data.

$$\tau_{AP} = 3\left(\frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}}\right) + 8\frac{l_{G.711MAC} + l_{CF-ACK} + l_{CF-Poll}}{c}, \qquad (4)$$

where τ_{AP} – AP data transmission time [s], $l_{CF\text{-}ACK}$ – CF-ACK frame size [B], $l_{CF\text{-}Poll}$ – CF-Poll frame size [B];

$$\tau_{PC} = 2\left(\frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}}\right) + 8\frac{l_{G.711MAC} + l_{CF-ACK}}{c}, \qquad (5)$$

where τ_{PC} – station data transmission time [s].

To count how many data is transmitted over one CFP interval (τ_{CFPint}), it is needed to set the optional length, which is limited by the maximum and the minimum values. The shortest value of CFP consists of the minimum possible CFP (τ_{CFPmin}) and CP (τ_{CPmin}) periods sum:

$$\tau_{CFP\min} = 4\left(\frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}}\right) + 8\frac{2l_{MPDU\max} + l_B + l_{CF-End}}{c} + 3\frac{\tau_{SIFS}}{1000000}, \qquad (6)$$

where τ_{SIFS} – SIFS duration [μs], $l_{MPDUmax}$ – biggest possible MPDU size [B];

$$\tau_{CP\min} = 4\left(\frac{l_{PLCPpreamble}}{c_{PLCPpreamble}} + \frac{l_{PLCPheader}}{c_{PLCPheader}}\right) + 8\frac{l_{MPDU\max} + l_{RTS} + l_{CTS} + l_{ACK}}{c} + \frac{3\tau_{SIFS} + \tau_{PIFS} + \tau_{DIFS} + \tau_{cont\min}}{1000000}, \qquad (7)$$

where l_{RTS} – RTS frame size [B], l_{CTS} – CTS frame size [B], l_{ACK} – ACK frame size [B], τ_{SIFS} – SIFS duration [μs], τ_{PIFS} – PIFS duration [μs], τ_{DIFS} – DIFS duration [μs], $\tau_{cont.min}$ – shortest possible contention duration [μs].

It is not possible to avoid the usage of the DCF, so in all calculations it will be designated the minimum value of the CP period duration.

Using calculations above, some characteristics can be found. The amount of frame pairs transmission between the station and the AP is found using the equation:

$$n = \frac{\tau_{CFP} - \tau_B - \tau_{CF-End}}{\frac{\tau_{SIFS}}{500000} + \tau_{AP} + \tau_{PC}}, \qquad (8)$$

where n – possible amount of frame pairs, τ_{CFP} – CFP period duration [s].

When n is calculated, the amount of time in the entire CFP interval could be found. It is used just for voice data (in this case for the G.711 codec):

$$\tau_{voice} = 16\frac{l_{G.711} \cdot n}{c}, \qquad (9)$$

where τ_{voice} – time when the voice information is transferred [s].

The time when the speech information is transmitted over the network can be called as "useful time". The amount of the useful time could be marked as efficiency coefficient (10th equation). Also in the same way it is possible to calculate the real data (voice information) transmission rate (11th equation) and later – the amount of conversations using particular codec (12th equation).

$$\rho = \frac{\tau_{voice}}{\tau_{CFP\,\mathrm{int}}}, \qquad (10)$$

where ρ – data rate transmission coefficient of efficiency, τ_{CFPint} – duration of the CFP frame [s].

$$c_{useful} = \rho \cdot c, \qquad (11)$$

where c_{useful} – useful information data rate [bps].

$$k = \frac{c_{useful}}{2c_{G.711}}, \qquad (12)$$

where k – the amount of conversations.

Results of the 802.11b medium research

Following the calculations, clear trends of the values are seen. The bigger voice codec packets are the less conversations could be held in one AP environment. In other hand, the more information could be filled in one packet, the bigger bandwidth could be achieved of the useful voice traffic and it goes without saying that in such way the better efficiency of the wireless media is reached.

Comparing our results with DCF mode calculations [3], network efficiency is about 50% (when the τ_{CFPint} = 0,1 s) better when the network bandwidth is 11 Mbps and the transferred useful amount of data is about 160 B. The difference is clear – contention-free mode increases 802.11 network efficiency for voice data.

That is more, when the τ_{CFPint} reaches 0,1-0,2 s, the curves are not "growing" so fast. We can conclude that these τ_{CFPint} values are optimal for the largest amount of calls in WLANs with PCF configured.

Best results of the ρ (fig. 2) are reached when the size of the packets is largest and the network bandwidth is smallest. It is because the PLCP and interframe intervals do not influence the transmission of the data very much and the data is transmitted slowly too. When throughput of voice data information increases but the PLCP and interframe intervals do not change, these values do affect voice data throughput sufficiently and it is noticeable.

Transmission of the useful data varies depending on codec. Highest (fig. 3) values are reached using G.711 codec within the 11 Mbps network bandwidth. Then the c_{useful} does not even reach 3 Mbps. Results are explained that the MAC frame is not filled enough with useful data.

The largest number of simultaneous calls (fig. 4) is reached using G.723.1 codec with network bandwidth of 11 Mbps. This codec generates larger voice packets than

G.729, but it's packetization interval is 50% longer and that's why network could deal with 43 simultaneous calls.

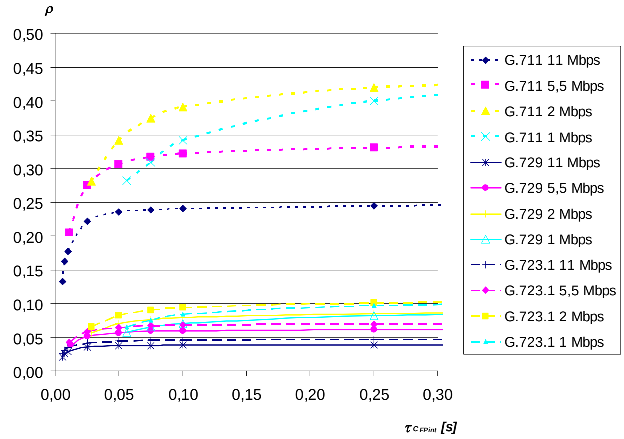

Fig. 2. Wireless media efficiency when different types of codecs and 802.11b network bandwidth are used

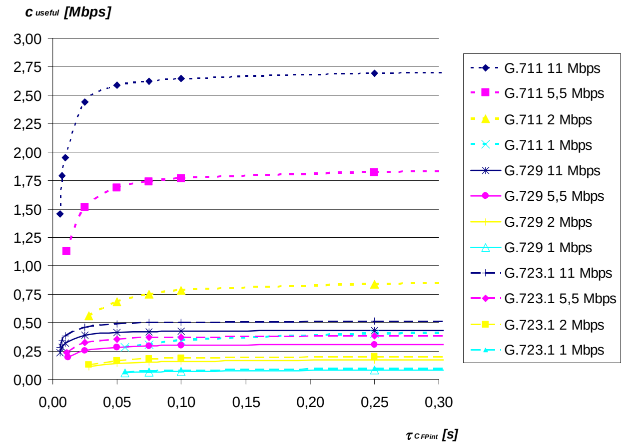

Fig. 3. Useful voice data rate dependencies on different codecs and 802.11b network bandwidth

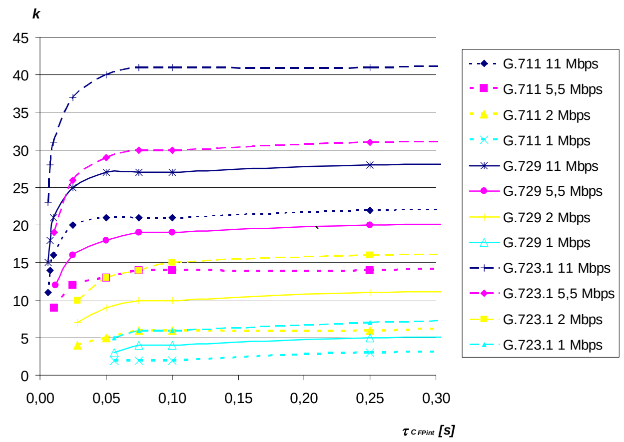

Fig. 4. Possible amount of VoIP conversations using different types of voice codecs and 802.11b network bandwidth

When DCF is used [5] and comparing various types of codecs, results of *k* do not differ much, because voice transmissions are influenced by contention mechanism in the wireless media. Using PCF with G.711 codec, values are improved about 69% using 11Mbps bandwidth and about 17% with 2 Mbps. The same trends are noticed with the G.729 codec - 110% with 5,5 Mbps and 33% - with 2 Mbps. In conclusion, to use PCF is reasonable with bigger bandwidth, otherwise it will not give a significant profit.

In the reference [4] a very abstract WLAN system with priorities is investigated. As in our case, author investigates unencrypted voice data capacity in WLAN. The results are similar to ours (the difference is not bigger more than 5 calls with the same codec and network settings). This difference is of the lack of management information in author's calculations.

Results of the 802.11g medium research

This part of the paper analyses 802.11g network using the same model as in previous chapters. 802.11b and 802.11g WLANs have the same MAC layer, but there are some minor changes in PHY part (e.g. PLCP preamble and header transmission, additional tail and pad bits) which were applied to our model [6, 7].

The results are similar to those in 802.11b part and are not analyzed so deep here. It is worth to say that all the graphs stop rapid growth at τ_{CFPint} of 0,025-0,050 s. It is about 4 times faster as it was for 802.11b WLAN.

The coefficient of efficiency (fig. 5) is similar too, but a few percent lower and that allows voice traffic to reach the throughput rate as high as 12 Mbps (fig. 6) in this case.

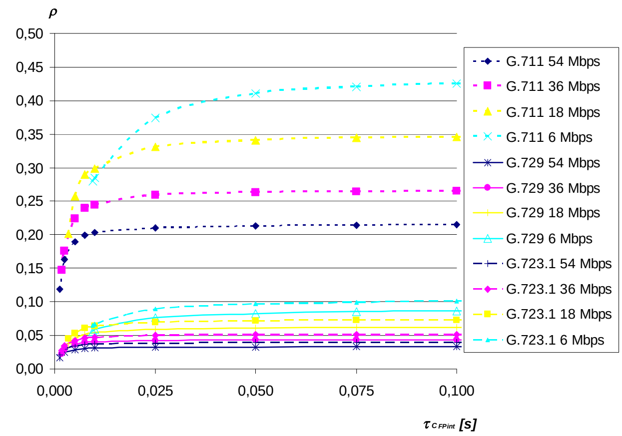

Fig. 5. Wireless media efficiency when different types of codecs and 802.11g network bandwidth are used

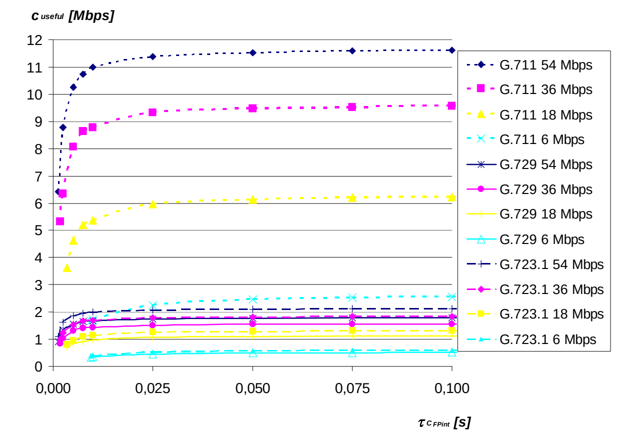

Fig. 6. Useful voice data rate dependencies on different codecs and 802.11g network bandwidth

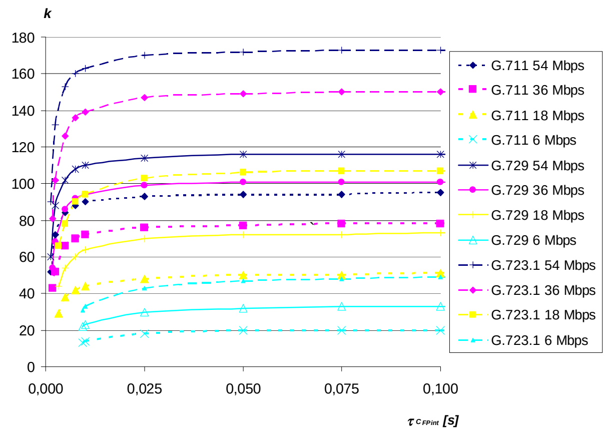

Fig. 7. Possible amount of VoIP conversations using different types of voice codecs and 802.11g network bandwidth

With 54 Mbps network throughput 802.11g WLAN allows to make 95 simultaneous calls with G.711 codec, 117 – with G.729 and 174 – with G.721.3 (fig. 7).

Conclusions

1. The biggest amount of conversations is achieved using G.723.1 with the network bandwidth of 11 Mbps for 802.11b and 54 Mbps for 802.11g standards.

2. τ_{CFPint} values are optimal in the interval of 0,1-0,2 s to use maximum VoIP calls in the 802.11b networks with PCF mechanism configured. Respectively the values of 0,025-0,050 s are optimal for 802.11g WLANs.

3. Using PCF is reasonable with bigger network bandwidth, otherwise it does not give an advantage comparing to DCF.

References

1. **Gast M.** 802.11 Wireless Networks: the Definitive Guide. – USA CA Sebastopol: O‘Reilly, 2002. – 464 p.
2. **Prasad R., Prasad N.** 802.11 WLANs and IP Networking: Security, QoS, and Mobility. – USA MA Norwood: Artech House, Inc., 2005. – 318 p.
3. **Rindzevičius R., Grimaila V., Pilkauskas V.** Traffic analysis of the next generation wireless access technologies // IPSI - 2006 AMALFI. – Amalfi, Italy. – March 23–26, 2006 / IPSI Belgrade, Serbia. – Belgrade: IPSI Belgrade Academic Mind, 2006. – P. 1–7.
4. **Gast M.** How Many Voice Callers Fit on the Head of an Access Point? Accessed at: www.oreilly.com/pub/ /a/etel/2005/12/13/ [2009-02-08].
5. **Gast M.** Maximum number of VoIP telephone calls per 802.11b access point. Accessed at: www.oreillynet.com/ /etel/blog/2005/06/ [2009-02-08].
6. **Kajackas A., Pavilanskas L.** Analysis of the Technological Expenditures of Common WLAN Models // Electronics and Electrical Engineering. – Kaunas: Technologija, 2006. – No. 8(72). – P. 19–24.
7. **Villaseñor-González L., Portillo-Jiménez C., Sánchez-García J.** A Performance Study of the IEEE 802.11g PHY and MAC Layers over Heterogeneous and Homogeneous WLANs // Ingeniería Investigación y Tecnología. – 2007. – Vol. VIII, No. 1. – P. 45–57.

Received 2009 02 14

D. Pauliukas, V. Vosylius. Research of Real Time Traffic Transmission in 802.11 WLANs // Electronics and Electrical Engineering. – Kaunas: Technologija, 2009.– No. 7(95). – P. 111–114.

Analytical model and calculations of voice transmission over the wireless IEEE 802.11b/g networks which are using PCF coordination functions are proposed. Using this model, some wireless networks characteristics are unrevealed, like their efficiency to provide voice over IEEE 802.11b/g networks when different voice codecs are used. What is more, wireless network capacity for calls is calculated for different types of voice codecs. These results are compared with the same network parameters but DCF coordination functions implemented. It is found that VoIP calls capacity could be two times bigger with higher network bandwidth compared with the results of lower bandwidth from the references' calculations. It is because of the usage of PCF coordination function in 802.11b/g WLANs. Ill. 7, bibl. 7 (in English; summaries in English, Russian and Lithuanian).

Д. Паулюкас, В. Восилюс. Исследование сервисов реального времени в 802.11 сети // Электроника и электротехника. – Каунас: Технология, 2009. – № 7(95). – С. 111–114.

Предлагается аналитическая модель и расчеты параметров передачи голоса беспроводными IEEE 802.11b/g сетями, которые используют PCF функцию. Используя эту модель можно вычислить некоторые характеристики беспроводных сетей, такие как эффективность обеспечения передачи голоса IEEE 802.11b/g сетями, когда используются различные голосовые кодеки. А также рассчитывается объем беспроводных сетей для звонков, используя различные типы голосовых кодеков. Эти результаты сравниваются с такими же параметрами в тех случаях, когда используется DCF функция координации. Установлено, что наши расчеты, особенно с более высокой пропускной способностью сети дают в 2 раза больше объема для VoIP звонков, а с более низкой пропускной способностью сети результаты похожи на расчеты, приведенные в других статьях. Так получается из-за использования PCF функции координации в 802.11b/g WLAN. Ил. 7, библ. 7 (на английском языке; рефераты на английском, русском и литовском яз.).

D. Pauliukas, V. Vosylius. Realaus laiko duomenų perdavimo tyrimas 802.11 bevieliuose tinkluose // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2009.– Nr. 7(95). – P. 111–114.

Analitiniu metodu nagrinėjamas balso perdavimas IEEE 802.11b bei IEEE 802.11g tinklais naudojantis PCF koordinacijos funkcija. Tyrimo tikslas buvo išsiaiškinti, kaip efektyviai galima panaudoti WLAN išteklius balsui perduoti paketiniais tinklais. Sukurtu analitiniu modeliu apskaičiuoti tinklų naudingumo koeficientai, tinklų talpa galimiems pokalbiams, kai balsui perduoti naudojami skirtingi kodavimo algoritmai. Nustatyta, jog bevieliuose tinkluose su PCF funkcija balso paslauga gali naudotis daugiau vartotojų nei tuose tinkluose, kuriuose veikia tik DCF funkcija. Naudojantis PCF funkcija, pokalbių kai kuriais atvejais galima sutalpinti dvigubai daugiau nei su DCF, esant didesnei tinklo spartai, o esant mažesnei spartai, šis skirtumas nėra toks akivaizdus. Il. 7, bibl. 7 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).