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Synchronous Voice Applied Customer Access based on IEEE 802.11

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Introduction

The modern telecommunication networks with DSL features of interactive access fulfill most of the customer demands. In Hot-Spot environments like airports or universities, the access points are implemented by using widely common wireless LAN (WLAN) equipment of IEEE 802.11 standard [1]. Initial versions of this standard were implemented for typical infrastructure conditions, presenting a lot of randomly moving users. The finite quantity of data usually is transmitted by customers. In such conditions the WLAN equipment implements the packet transmission and packet switching function. The common method of customer's access to the WLAN is contention-based random multiple-access.

The IEEE 802.11 equipment is not expensive; therefore it is used to implement access networks with small quantity of customers.

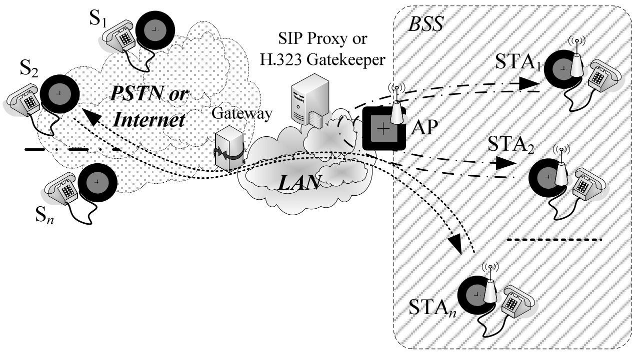

Fig. 1. Network with customer access based on IEEE 802.11

To support wireless architectures, the IEEE 802.11 MAC offers two operating modes. The first one is also defined as the contention mode and called the DCF. It defines the distributed access and typically is used for an ad-hoc network. The second operation mode is contention free and called the PCF. PCF defines a centralized access for an infrastructure network. PCF support time-bounded services as well as transmission of asynchronous data, voice or mixed.

The main research object of this task is the connection of the customer group to PSTN and Internet. Customer access is based on IEEE 802.11 network (Fig.1). The aim of research is to analyze the abilities to provide DSL-like quality level services in customer access network based on IEEE 802.11. The support of telephony services allows acceptable packet loss which can emerge because of the time limitations between CFP and CP. Analysis is made considering that the PCF is implemented and the beacon rate is synchronized with voice coding rate.

The infrastructure access network basically consists of AP and nodes (STA's). AP acts as a bridge between the wireless terminals or STA and the wired part: PSTN and Internet. General propositions, we are making further are as following:

1. The AP is connected to the wired network via a link that has a higher capacity than the system capacity of the WLAN.
2. In such system all users are fixed. To support high quality physical channel, the antennas may be used.
3. The quality of service in such system must be comparable with DSL.

In this task the analysis presented in [2] and [3] is proceeded. The main consideration is focused on features of signal transmission of few user conversations at the same time through IEEE 802.11 based customers access WLAN. The offered method is applicable to analyze technological expenditures for RT traffic in IEEE 802.11 infrastructure networks by applying ITU P.59 [4]. The modeling results based on Opnet modeler simulations are presented herein.

IEEE 802.11 abbreviations

AP – Access Point; STA – wireless station; DCF - Distributed Coordination Function; PCF - Point Coordination Function; CFP – Contention-Free Period; CP – Contention Period; ACK – Acknowledgment; SIFS – Short Inter Frame Space; PIFS – PCF Inter Frame Space; MPDU – MAC Protocol Data Unit; TBTT - Target Beacon Transmission Time; PHY – Physical Layer.

Related Work

There are many research works where wireless or radio link access from different aspects are analyzed – WLAN protocols and its features, voice and data transmissions, features of usage in the specific conditions. For WLAN protocol modernization or perfection many task solutions are proposed. A number of previous studies have evaluated the capacity in IEEE 802.11 networks for voice traffic and real time traffic in general. References [5], [6], [7] studied the use of DCF to support VoIP. The delays and its variation margins are determined. It is shown, that the capacity to accommodate voice traffic in DCF is very limited. DCF is not efficient in supporting the delay-sensitive voice traffic. The contention-based nature and exponential backoff mechanism can not guarantee that voice packet is successfully delivered within the delay bound.

Controlled access is more suitable for voice traffic delivery, because of its less overhead and guaranteed delay performance. The capacity of a system that uses the PCF for CBR and VBR voice traffic was analyzed in [8] and [9]. The VBR voice traffic was simulated using ON-OFF voice source model.

The paper [10] focuses on a throughput analysis and delay of the DCF/PCF of the IEEE 802.11 and 802.11e MAC. The results are data transmission oriented and presented for different packet lengths and different number of users. Throughput performances are also detailed for the EDCF.

Voice Source Models

Currently few different voice models are used. The simplest ones analyze single independent voice sources. Constant Bit Rate (CBR) and Variable Bit Rate (VBR) sources may be assigned to this group. Voice models which are composed by examining dialogs – the conversational speech may be assigned to other group.

The CBR voice source is quite simple. The source transmits packets at the constant rate κ. The size of the packets is determined by the bit rate of the codec R_{cod} and the packetization period τ_{pac}.

A voice source alternates between talk spurts and silent periods. Function of Voice Activity Detection (VAD) is implemented in the modern voice codec. Such codec generates VBR traffic. VBR voice traffic can be represented by an ON/OFF model. At ON state, voice packets are generated periodically at the constant rate κ, while no voice packets are generated at OFF state. The durations of the ON and OFF states typically are modeled as independently and exponentially distributed random variables, with parameter τ_{ON} and τ_{OFF}, respectively. Each voice source is modeled by a two-state discrete-time Markov chain. At instant time *t,* a voice station has voice packet to send (source is at ON state) with probability: $p_{\text{ON}}= \tau_{\text{ON}}/(\tau_{\text{ON}}+ \tau_{\text{OFF}})$, and has no voice packets (source is at OFF state) with probability $p_{\text{OFF}}=1- p_{\text{ON}}$. VBR voice source model is proper for analysis of one direction transmissions.

To imitate the conversation of two speakers applying this ON-OFF model, the assumption has to be made that the users speak independently. Realistically, the assumption would be incorrect since speakers talk alternately during the conversation. And only on rare particular occasions they either both talk or both listen.

For WLAN applications conversational model when two users (A and B) are talking dependently is more applicable.

Summarized model of the conversation between two users A and B like superposition between state of the conversation and the state of speech activity is composing and may be defined by four state Markov chain. In the conversation model possible states are: A talking B silent, A silent B talking, both talking, both silent [4]. The probabilities of these states are indexed by p_{A0}, p_{0B}, p_{AB}, p_{00} respectively.

According to ITU recommendation [4] the durations of states are identically distributed exponential uniform random variables with means 854 ms, 854 ms, 226 ms and 456 ms respectively. According to these durations calculated probabilities are: $p_{\text{A0}} = 0.357$, $p_{\text{0B}} = 0.357$, $p_{\text{AB}} = 0.095$, $p_{00}=0.191$.

The channel occupation times are evaluated incorrectly, if two-state ON-OFF speech model is used while analyzing IEEE 802.11 networks, where transmission in both directions take place in same physical channel. Two-state model fails to evaluate precisely time periods of mutual talk or mutual silence. In this article conversational model [4] is used while analyzing WLAN characteristics.

Voice coding and transmission diagram

Voice coding and transmission processing time diagram is shown in Fig. 2. At the sender, the analog voice signal is encoded by a codec which determine voice frame duration T_{F} of coding.

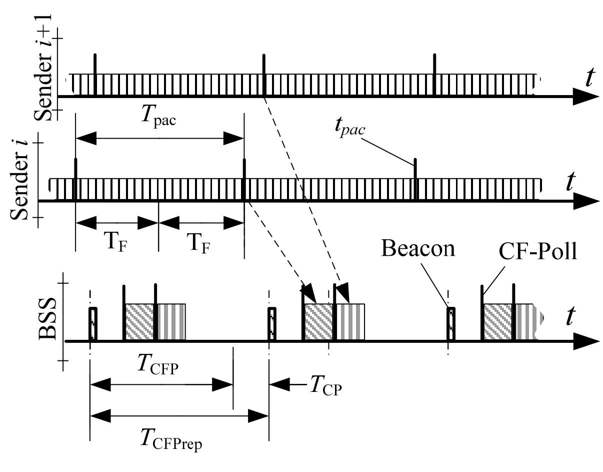

Fig. 2. Diagram of Voice coding and transmission processing

RTP/UDP/IP headers are appended to coded voice packet. This process is not shown in Fig. 2.

After inclusion of the RTP/UDP/IP headers during the packetization procedure at the transport and network layers, voice packets are transmitted over the network. Packetization procedure ends at the moment t_{pac}.

Voice packets with period T_{pac} are transmitted periodically during each CFP period. T_{pac} may be equal to T_F or multiple T_F, in case WLAN data packet carries several voice frames.

Voice frame duration T_{PAC} and T_{CFPrep} are totally independent. T_{PAC} is conditioned by voice codec (for GSM and AMR codecs T_{pac} = 20 ms), while T_{CFPrep} is adjustable WLAN parameter. To overcome voice packet loss, we suggest performing transmission synchronically, when

$$T_{\mathrm{pac}} = T_{\mathrm{CFPrep}} \tag{1}$$

or in case of carrying multiple voice frames:

$$a \cdot T_{\mathrm{F}} = T_{\mathrm{CFPrep}} \,. \tag{2}$$

In analysis and simulations presented herein, the conditions (1) and (2) is assumed to be met.

The T_{CFPrep} is controlled by AP. The AP shall define the timing for the entire BSS by transmitting beacons (Fig.2) according to the T_{CFPrep} attribute within the timing synchronization function (TSF) of AP [1]. TSF is a local AP timer which synchronizes the TSF of every other station in the BSS. TSF defines a series of *TBTT*s exactly $T_{CFPRate}$ time units apart. At each *TBTT*, the AP shall schedule a beacon as the next frame for transmission [1].

While transmitting voice packets in PCF, the queuing delay is inevitable, because the transmission to particular destined STA takes place strictly according to the polling list. This creates additional delay.

Voice Applied Customer Access Model

Customer access modeling was based on voice source synchronization with beacon interval in PCF mode, according to [2].

The modeling of voice applied customer access was implemented in two stages: the modeling of conversational speech and modeling of IEEE 802.11 hotspot and wireless PCF clients.

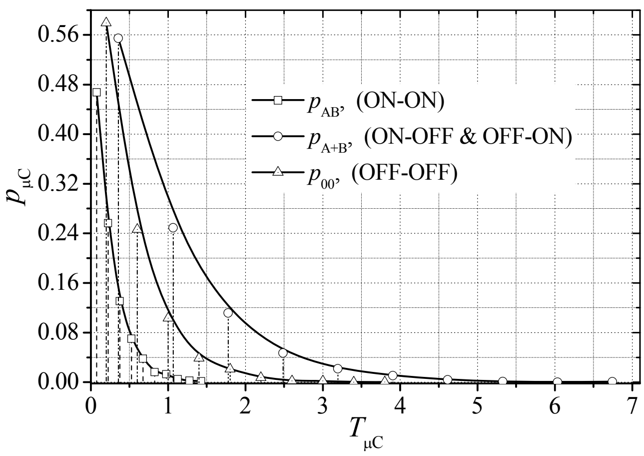

Fig. 3. The distribution of model state durations

The first stage includes the modeling of conversational speech, which follows four state ON-OFF model (as described in [4]), having transitions from double talk to mutual silence trough single talk states of one or other speakers. Model was implemented in Matlab. The outcome of first stage simulation expresses the timeline of speech activity in forward and backward directions. Unlike in two-state ON-OFF model, the two speech directions are dependent and connected in a way of conversational dialogue. Thus the algorithms implemented as Matlab scripts produce voice source in pairs – one for STA and other for corresponding user on the wired part of the network. The distributions of model state durations are shown in Fig. 3.

The generated conversational speech flows are stored as external data in form of talk-silence cycle. Although all speech pairs were created by using the same parameters, they are generated separately and are statistically independent.

The second modeling stage includes models and simulations in Opnet Modeler [11]. The network setup (Fig. 1.) was implemented using standard Ethernet station, AP and wireless IEEE 802.11 station models. User profile configuration includes voice over IP calls, which are initiated at the beginning of simulation and continues to the end.

Additionally few editions were made to the original Opnet 10.0A STA IEEE 802.11 MAC layer:

1. In CFP operation mode the STA could be polled more than once in the same CFP period. After AP have finished to poll all STAs listed in its polling list, but still some CFP time remains and there is some data to send, the AP starts the polling over again resulting some STAs to be polled more than once. This kind of operation had to be eliminated in order to maintain the precision of voice packet delays and channel capacity. Modified model ends contention free period prematurely rather than starts polling once more.
2. All STAs could send data in CP even if it is PCF enabled STA. Original model was modified to prevent DCF transmissions in PCF enabled stations in order no to lose any voice packets, which would not be included in statistics while calculating CFP durations. Despite of modifications, STA registration related traffic is allowed during DCF to maintain regular registration routine.
3. All upper layer traffic in MAC layer were handled equally – stored in a single buffer. Separation was done only in the means of coordination function type – separate buffers for PCF and DCF traffic, but no differentiation within PCF buffer. This makes the voice packets to be collected and stored in buffer until sent together with poll (in case of AP) or until poll is received and data can be sent with an acknowledgement to the poll (if this is a STA). In case of overload in the PHY, when more STAs are registered than can be served by AP, the packets accumulate in the buffer, thus effecting increased delays or even buffer overflows if the overload happens to be harsh. Such principle is not suitable for delay sensitive applications. It is acceptable rather to drop packet than to store it for unlimited amount of time (usually it is limited only by size of buffer). Packet drop would affect voice quality, but only for the particular flow, while storing and delaying packet may negatively affect other voice flows introducing unacceptable delay and jitter. Furthermore, if

the delayed packet is not usable, but still transmitted, the bandwidth used for transmission is wasted. The model was modified in such way, that it would drop the data packet if it failed to be transmitted due to overload, i.e. it was not transmitted because previous STAs used up all available CFP time. Generally speaking, all data frames older than CFP repetition time are dropped.

Opnet simulations were executed using prepared conversational speech models to imitate talking users. G.711 was used as voice codec with voice activity detector (VAD) activated in order to produce VBR traffic: voice flow bandwidth – 64 kbps; packet rate – 50 pps. Voice payload frame size is forced to be equal to CFP repetition time: $T_{\text{pac}} = T_{\text{CFPrep}} = 20$ ms. The voice payload of 20 ms produces 160 byte voice frames. This may be considered as multiple voice frame packing in WLAN packet (2) with multiplier $a = 2$, since in this case two codec sample intervals of 10 ms form the 160 byte payload. The IP/UDP/RTP header of 40 bytes size is added to voice frame before passing to WLAN layer.

Since OpNET is discrete time simulator, it does not require any additional effort to keep the voice sample length and CFP repetition time exactly the same. Thus, no additional packetization time fluctuations over CFP repetition time can possibly occur. This means that we always precisely keep the equality (1) during the simulations.

Performance Results

The first set of simulations was intended to show the wireless channel occupation time while different number of STAs operate in the PCF controlled network. The simulations were performed for IEEE 802.11b physical data rates 5.5 and 11 Mbps, while all other parameters including CFP repetition and voice packetization times are constant.

In order to simplify the modeling, the simulations were combined with some calculations. The simulations were performed for STA numbers up to 10, next the distributions of channel occupation time were obtained from simulation results. The distributions for STA counts over 10 were calculated using convolution of already obtained distributions.

It was done for the following reasons. Firstly the simulation of great number of nodes is a time and processing power consuming task, therefore it is more efficient to perform longer simulations to obtain higher precision, than to run short simulations of high quantity of nodes. Secondly, it is not possible only by simulating to show visually the distributions which extend over CFP occupation time. Since when actual required CFP occupation time exceeds maximum allowed CFP duration, this indicates the network overload, therefore probability distributions are distorted due to packet loss. This kind of solution does not bring any bias to the results whatsoever.

Let the $p_l(t_{\text{oc}})$ and $p_j(t_{\text{oc}})$ be the cannel occupation time t_{oc} distribution functions, when STA count is l and j respectively.

While applying linear system model, the channel occupation time is not limited. The distributions of channel occupation time are calculated using rules of adding casual variables for the STA count $n=l+j$. Then the convolution can be written as:

$$p_n(t_{\text{oc}\Sigma}) = \sum_{t_{\text{oc}}} p_l(t_{\text{oc}}) p_j(t_{\text{oc}\Sigma} - t_{\text{oc}}) . \qquad (3)$$

To keep the maximum possible accuracy, the distributions were produced using precise t_{oc} values, obtained from simulations. Since those values differ in constant steps, the discrete function convolution has been applied directly, without resampling frequency counts consequently avoiding any bias. It is easy to notice that the step size is equal to difference between data and control packet (empty packet) sizes.

Distributions of CFP occupation time rate are shown in Fig. 4 and Fig. 5. The graphs show distributions of different STA number under different physical data rate.

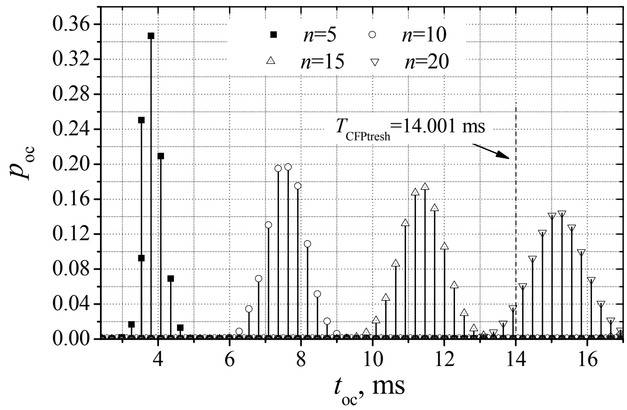

Fig. 4. Distributions of CFP occupation time rate for 5.5 Mbps PHY rate of IEEE 802.11b

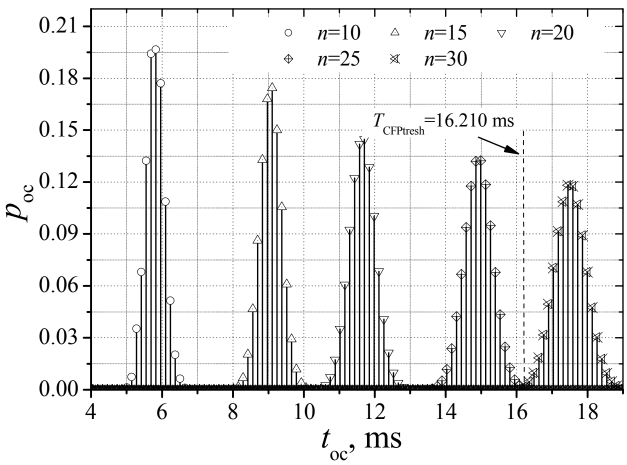

Fig. 5. Distributions of CFP occupation time rate for 11 Mbps PHY rate of IEEE 802.11b

Capacity

IEEE 802.11 standard [1] defines minimal allowed CP duration T_{CP}:

$$T_{\text{CPmin}} = T_{\text{PPDUmax}} + 2\tau_{\text{SIFS}} + 2\tau_{\text{TimeSlot}} + 8\tau_{\text{ACK}} . \qquad (4)$$

At least that much transmission time has to be dedicated for CP in T_{CFPrep} cycle.

When the CFP repetition time duration T_{CFPrep} is chosen (2), the allowed CFP duration has to consider the deduction of minimal CP time and can be marked as threshold $T_{CFPtresh}$:

$$T_{CFPtresh} = T_{CFPrep} - T_{CPmin}\ . \quad (5)$$

According to [1], and [3] when with each frame of CFP micro-cycles the equal User Date payload (all in ON-ON state [3]) is transmitted, the maximal duration of CFP (T_{CFPmax}) may be expressed:

$$T_{CFPmax} = nT_{AB} + \tau_{Beacon} + \tau_{CF\text{-}End}\ , \quad (6)$$

$$T_{AB} = 2(\tau_{PPDU0X} + \tau_{SIFS})\ , \quad (7)$$

where n – count of micro-cycles in the CFP (the n is equal to count of pollable STAs in WLAN); τ_{Beacon} – beacon frame duration; $\tau_{CF\text{-}End}$ – CF-End subtype of Control type frame duration [3].

The voice transmissions in analyzed network model are lossless until the following condition is met:

$$T_{CFPmax} \le T_{CFPtresh}\ . \quad (8)$$

From inequality (8) and formulas (5) – (7) follows the expression of capacity for transmission system without losses:

$$n_0 \le \frac{T_{CFPrep} - T_{CPmin} - \tau_{Beacon} - \tau_{CF\text{-}End}}{T_{AB}}\ . \quad (9)$$

Consequently, if the T_{CFPrep} = 20 ms, the maximum allowed CFP duration or CFP threshold $T_{CFPtresh}$ for IEEE 802.11b 5.5 Mbps, and 11 Mbps is 14.401 ms and 16.210 ms respectively.

Using these parameters we get n_0 12 and 21 channels for 5.5 Mbps and 11 Mbps PHY throughput respectively.

The system capacity n_c can be revised and increased by declaring the allowed packet loss probability. This task is solved as follows. The simulation results are processed to obtain channel occupation duration distributions, which are complemented by calculated (3) distributions, when $n > n_0$. The results are presented in Fig. 6 and Fig. 7.

The capacity evaluation from the provided graphs may be observed flexibly in the means of allowed packet loss probability. If distribution of channel occupation duration does not fit into timeframe up to $T_{CFPtresh}$, it leaves some distribution bars behind. This means that under certain load (number of STAs) the last voice streams will experience packet loss, which probability is equal to sum of unfitted values.

From Fig. 6 it can be seen, that up to 16 STA's distributions fits perfectly within $T_{CFPtresh}$ limits, thus no packet loss is ever introduced in the particular scenario. It can also be seen that 17 STA distribution leaves some values behind $T_{CFPtresh}$, but the sum of it is very small – the calculations show, that the loss probability for 17 STA stays below 0.8 %, which is certainly acceptable as will not have noticeable effect on voice quality. In case of 18 STAs, the $T_{CFPtresh}$ cuts the bigger part of distribution, thus the packet loss is expected to be greater. The calculation shows almost 4.5 % packet loss probability. So channel number is extended from 12 to 17 or 18 depending what packet loss probability we are willing to accept.

As for 11 Mbps PHY rate (Fig. 7), the distributions up to 25 STA show no packet loss, distribution of 26 STA shows 0.4 % packet loss probability, and 5.4 % in case of 27 STA. Thus in case of 11 Mbps PHY, we gain 5 channels extending from 21 to 26.

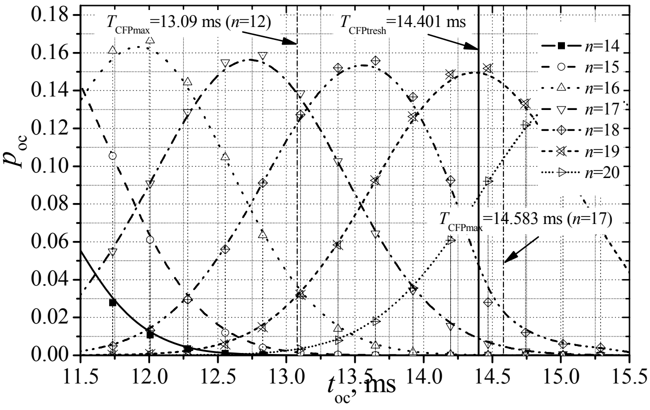

Fig. 6. Distributions of CFP occupation time rate in range of maximal CFP duration for 5.5 Mbps PHY rate

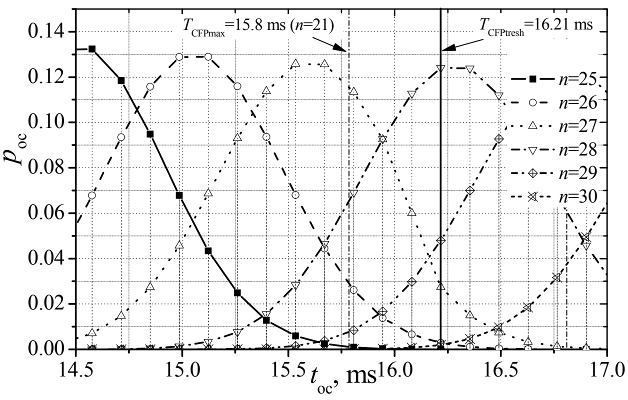

Fig. 7. Distributions of CFP occupation time rates in range of maximal CFP duration for 11 Mbps PHY rate

Thus on the particular situation in both PHY rate cases, the channel capacity is increased by 5 channels keeping reasonable packet loss probability (less than 1%). The greater capacity allows serving more users or using greater oversubscription. The acceptable packet loss ratio can be chosen to allow more simultaneous voice streams instead of allowing less voice channels with no packet loss, which is not efficient way of predicting the capacity, since the probabilities of full (or near to it) channel usage is really small.

Since polling list is not reallocated during communication process (during voice call in our case), it is reasonable, that only the latest registered STAs will suffer from packet discards when CFP time is used up.

Conclusions

The analysis accomplished herein shows that it's possible to create customer access with quality, comparable to DSL, when the IEEE 802.11 protocol equipment with PCF is used. This can be achieved by synchronization of beacon interval and voice coding cycles.

The offered method will allow increasing capacity of the IEEE 802.11 PHY channel while keeping reasonable packet loss probability. The results of IEEE 802.11b customer access simulations for 5.5 and 11 Mbps PHY rates show that implementation of offered method improve channel capacity, thus allowing to serve more users or apply oversubscription. Note that defined capacity, reflecting the marginal abilities of wireless customer access network, is greater than capacity evaluations in DCF mode presented in other publications.

The synchronization of CFP repetition interval and period of voice codec in real IEEE 802.11 equipment has been and still remains problematic topic. When these periods are not synchronized, the voice coding window is sliding time-wise during transmission. Since the sliding effect can create many noteworthy scenarios, this problem will be analyzed in-depth in the future works.

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The analysis of infrastructure IEEE 802.11 protocol based wireless access network with point coordination (PCF) for synchronized voice and reliable quality data transmission is presented. It is shown, that it is possible to create wireless customer access with wired network quality by synchronizing voice source coders with beacon periods. The analysis of this system has been validated by simulations in Opnet. The modeling additionally evaluates wireless channel capacity in terms of provided voice channels while keeping reasonable packet loss. The defined capacity is greater than capacity evaluations presented in other publications. Ill. 7, bibl. 11 (in English; summaries in English, Russian and Lithuanian).

А. Каяцкас, Л. Павиланскас, А. Виндашюс. Передача голоса по синхронным линиям доступа на основе стандарта IEEE 802.11 // Электроника и электротехника. – Каунас: Технология, 2007. – № 8(80). – С. 23–28.

Рассматривается задача применения беспроводной сети с определенной инфраструктурой на основе протокола IEEE 802.11 для передачи голоса и данных. Считается, что времена передачи распределяются централизовано (версия PCF), а цифровые источники голоса по радиомаякам синхронизированы с оборудованием сети. Показано, что такой режим обеспечивает качество передачи, сопоставимое с качеством цифровых абонентских линий. Анализ произведен путем моделирования с применением пакета OpNet. Определенная предельная емкость сети оказалась выше тех значений, которые приведены в опубликованных работах. Ил. 7, библ. 11 (на английском языке; рефераты на английском, русском и литовском яз.).

A. Kajackas, L. Pavilanskas, A. Vindašius. Balso perdavimas sinchronizuotomis IEEE 802.11 standarto vartotojų prieigomis // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2007. – Nr. 8(80). – P. 23–28.

Nagrinėjamos IEEE 802.11 protokolo apibrėžtos infrastruktūros bevielio tinklo su centralizuotai skirstoma perdavimo trukme (PCF versija) galimybės bei talpa pritaikant sinchronizuotiems balso šaltiniams bei garantuotos kokybės duomenims perduoti. Straipsnyje parodyta, kad, sinchronizuojant balso šaltinius pagal prieigos mazgo švyturio paketus, vartotojų prieigos tinkluose galima pasiekti aukštą perdavimo kokybę, prilygstančią laidinių skaitmeninių linijų kokybei. Analizė iliustruota ir pagrįsta modeliavimo rezultatais, gautais OpNet aplinkoje. Nustatyta bevielio tinklo ribines galimybes atspindinti talpa. Nagrinėjamu atveju ji yra didesnė už kituose skelbtuose šaltiniuose nurodytas talpas. Il. 7, bibl. 11 (anglų kalba; santraukos lietuvių, anglų ir rusų k.).