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Features of Voice Traffic Transmission in IEEE802.11ac Wireless Networks

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Abstract*—The aim of the study is to determine theoretically substantiated limitations of voice traffic transmission in IEEE 802.11ac wireless networks with mixed traffic prioritization. A mathematical model of IEEE 802.11 standard wireless network operation based on the virtual contention window concept was applied for the research. Voice frame parameters were selected corresponding to the G.711 codec, the results of which can be extended to voice traffic generated by other modern codecs. The prioritization mechanisms of the IEEE 802.11e standard were analyzed, implemented through reduced mandatory waiting time and shortened additional waiting interval for high-priority frames. Numerical data on the probability of transmitting AC_VO voice traffic frames in competition with AC_BE general priority traffic were obtained. It was established that the probability of transmitting a voice frame on the first attempt when selecting a "zero" time slot ranges from 0.98 (one voice station and two general priority stations) to 0.922 (eight voice stations and sixteen general priority stations). For voice stations in competition mode, the probability of successful transmission on the first attempt ranges from 0.795 to 0.527 depending on the number of stations with saturated load. It was determined that the probability of transmitting voice frames using retransmission attempts in the presence of consecutive collisions is at least 0.98 when using six retransmission attempts. Further development of the method for evaluating performance characteristics of IEEE 802.11 wireless networks using the virtual contention window mathematical model for networks with prioritized traffic was obtained. The obtained results are useful for optimizing the operation and designing IEEE 802.11ac standard local wireless networks with VoIP functionality.

***Keywords:** wireless network; voice codec; medium access control; transmission probability; prioritized traffic; IEEE 802.11ac standard; VoIP*

I. INTRODUCTION

With the development of the Internet, voice communication services based on IP protocol – commonly known as VoIP – have also advanced. Examples of such services include IP telephony and voice calls in various messaging applications. Voice traffic transmission can be carried out over either wired or wireless connections. The primary wireless technology used is Wi-Fi, based on the IEEE 802.11 standard. Originally, Wi-Fi was not designed for voice applications or real-time services, which can lead to degraded Quality of Service under certain conditions [1]. Common issues when using Wi-Fi for VoIP include [2, 3]: dropped calls, when a call ends abruptly without warning; choppy or fragmented audio caused by inconsistent packet delivery; and excessive packet delays resulting in speech delay or echo, which makes conversations unnatural and unpleasant. Over time, the IEEE 802.11 standard has evolved significantly, improving its operational performance. Currently, the most widely used specifications are IEEE 802.11ac and IEEE 802.11n, which offer high throughput. To enhance the quality of real-time services over Wi-Fi, particularly by prioritizing traffic, the IEEE 802.11e protocol was developed. This protocol is mandatory in IEEE 802.11ac networks.

II. PROBLEM STATEMENT

Numerous scientific and practical articles have addressed the use of Wi-Fi technology for voice traffic transmission, offering recommendations on optimal configuration of technical equipment and network parameters, for example, [1 - 4]. However, there is a lack of theoretical studies on the fundamental limits of voice traffic transmission in Wi-Fi networks with mixed traffic. Most publications rely on the practical experience of wireless network operators. The aim of this research is to analyze the process of voice traffic transmission in IEEE 802.11ac networks with mixed and prioritized traffic in order to identify theoretically grounded limitations that may arise in such networks.

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III. Research background

In our previous work, we examined the operation of IEEE 802.11ac networks where the mixed prioritized traffic consisted of saturated low-priority traffic BK_AC and high-priority voice traffic VO_AC [5]. A key characteristic of such networks is the significantly shorter arbitration interframe space AIFS_VO for voice traffic stations compared to the AIFS_BK for low-priority stations. These networks can support many simultaneously active voice stations, but the overall network performance tends to be limited due to considerable unproductive time losses.

The current study analyzes the processes occurring in an IEEE 802.11ac network with mixed traffic composed of best-effort traffic BE_AC and voice traffic VO_AC.

To evaluate the processes in such a data transmission network, a model based on the concept of a virtual contention window was used [6]. The virtual contention window VCW is a stochastic parameter of a Wi-Fi network under saturated load conditions. It quantitatively represents the average number of elementary time intervals (time slots) during which the backoff counter decrements the delay interval following the end of a previous frame transmission until the start of the next successful frame transmission by a given station. This mathematical model establishes a direct relationship between the system parameters of the network and its performance characteristics. It is not limited by the range of system parameter values; rather, the values are constrained by the IEEE 802.11 standard and practical experience with wireless data networks.

IV. Characteristics of processes in a network with mixed traffic

The system parameters that define the features of prioritized traffic transmission in an IEEE 802.11 network are presented in Table 1 [7].

Table 1 – IEEE 802.11ac 20 MHz wireless channel characteristics

Default EDCA parameters for each access category			
Access Category	***CWmin***	***CWmax***	***AIFSN***
Background (AC_BK)	15	1023	7
Best Effort (AC_BE)	15	1023	3
Video (AC_VI)	7	15	2
Voice (AC_VO)	3	7	2
Legacy DCF	15	1023	2

Table 1 includes the following parameters: CW (Contention Window) – a value that defines the set of numbers [0, 1, 2, …, CW], from which a station randomly selects a number to load the backoff counter and introduce an additional access delay to the channel. AIFSN – a number (N) used to determine the duration of the arbitration interframe space (AIFS – Arbitrated Interframe Space):

$$AIFS[AC\_XX] = SIFS + AIFSN \cdot ST, \qquad (1)$$

where SIFS is the Short Interframe Space (16 µs); ST – system time slot (9 µs).

The IEEE 802.11e specification provides two prioritization mechanisms: 1) High-priority frames are assigned a shorter mandatory arbitration interframe space (AIFS); 2) These frames are also assigned a smaller contention window value to enable faster recovery from collisions.

The channel access cycle during the successful transmission of any data frame is shown in Figure 1.

In Fig. 1, the following notations are used: LP – Long Preamble of the frame, intended for recognition by stations operating under earlier IEEE 802.11 specifications; RTS – Request to Send frame indicating the beginning of channel access; CTS – Clear to Send frame indicating readiness to receive a frame; VHTP – Very High Throughput Preamble of the IEEE 802.11ac

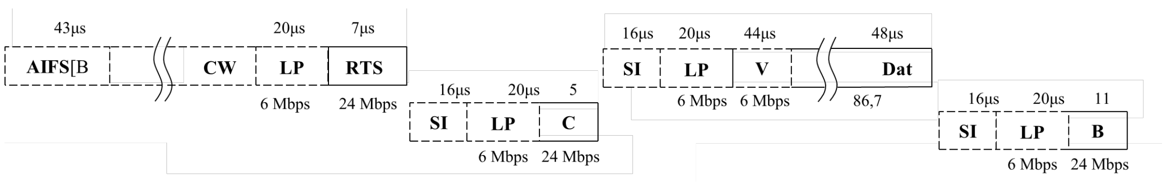

Fig. 1 Distribution of time intervals during access to the IEEE 802.11ac wireless channel [8]

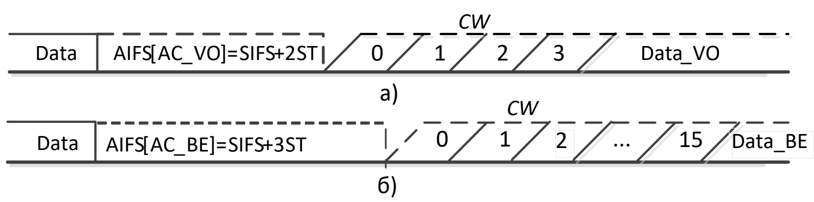

Figure 2 – Time interval distribution during channel access for AC_VO and AC_BE frames

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standard; Data – Block of useful data; BA (ACK) – Block Acknowledgment frame confirming successful data reception.

The time interval distribution diagram for medium access during the transmission of an RTS frame by voice traffic stations AC_VO, according to Table 1, is shown in Fig. 2a, while for best-effort stations AC_BE, it is shown in Fig. 2b.

The duration of the mandatory arbitration interframe space AIFS after the completion of the previous data frame transmission is one time slot shorter for voice traffic stations than for best-effort stations. The initial set of values for randomly determining the additional backoff interval (CW) is [0, 1, 2, 3] for voice traffic stations, and [0, 1, 2, ..., 14, 15] for best-effort stations.

The following features of voice traffic frame transmission can be highlighted:

- In each access cycle after the previous frame transmission by any station, there is one time slot during which only voice traffic stations (AC_VO) compete for channel access;
- Voice traffic stations do not create saturated traffic; instead, they generate one voice data frame periodically (typically with a sampling interval of 20 ms) [8] and attempt to transmit the frame from a random point in time;
- – In the event of a collision, a voice traffic station increases its contention window once and begins loading the backoff counter with a value from the set [0, 1, 2, 3, 4, 5, 6, 7], which is half the size of the initial set for best-effort AC_BE frames [7]. That means a voice traffic station will attempt retransmissions much more frequently compared to stations transmitting best-effort frames.

Taking the above into account, the further research was divided into two stages:

- Investigation of the probability of successful transmission of voice frames that begin transmission during “zero” slots;
- Investigation of the probability of successful transmission of voice frames that begin transmission in competition with frames having AC_BE priority.

First, we analyze the transmission characteristics of voice frames by stations that loaded the value “0” into the backoff counter, i.e., such stations start transmitting their frames immediately after the mandatory AIFS[AC_VO] delay interval ends.

When multiple voice traffic stations operate simultaneously in the network, on average, one-fourth of these stations will gain access to the channel immediately after the arbitration interval ends, and collisions may occur among them.

For further analysis of the operation of a network with mixed traffic, we will use the calculation results presented in [10] and [5].

Reference [10] provides values for: the probability of collisions – pc, the contention window size – VCW, the number of collisions during the virtual contention window – Nc, the duration of the virtual contention window – TVCW, and the channel throughput in cases where all stations transmit identical data blocks (512 and 1500 bytes) – S512, for networks with varying numbers of stations (from 2 to 16) using the same priority level AC_BE under saturated load conditions (Table 2).

The data presented in Table 2 were obtained for an IEEE 802.11ac network operating on a 20 MHz channel in MCS8 mode (forward error correction rate 3/4). Also useful are the transmission durations of data frames with 512-byte or 1500-byte payloads: ΔT_{512} = 140.4 µs; ΔT_{1500} = 230.4 µs.

In IP networks, voice traffic is transmitted using both specialized and universal voice codecs, which may differ significantly in their parameters [9]. Generally, all network applications are required to support G.711 codec packets. This codec, originally developed for digital telephony (ISDN), produces a compressed stream at 64 kbps. For the analysis of voice traffic transmission features, we will use the parameters of the G.711 codec, since other codecs typically have better charac-

TABLE 2 – CHARACTERISTICS OF THE IEEE 802.11AC WIRELESS CHANNEL (20 MHZ)

N	p_c	*VCW*	*Nc*	T_{VCW}^{512}, µs	S_{512}, Mbps	T_{VCW}^{1500}, µs	S_{1500}, Mbps
2	0,0625	8,036	0,067	605,255	13,535	785,254	30,563
4	0,1760	9,519	0,427	1169,764	14,006	1529,722	31,377
6	0,2758	11,772	1,142	1763,176	13,932	2302,24	31,260
8	0,3635	14,905	2,282	2390,852	13,674	3104,312	30,853
10	0,4406	18,779	3,925	3056,364	13,304	3930,445	30,308
12	0,5083	23,067	6,149	3761,613	12,841	4768,249	29,679
14	0,5679	27,367	9,023	4507,34	12,296	5600,975	28,989
16	0,6202	31,307	12,602	5293,491	11,676	6410,825	28,245

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teristics, and the results obtained can be generalized to them. The G.711 codec samples voice signals at intervals of 20 or 30 ms. In the following analysis, we consider the 20 ms sampling mode. The sampling rate is 8 kHz, with each sample consisting of 160 bytes. The transmission time of a single voice information frame in an IEEE 802.11ac network operating under
the specified conditions is 68 µs [5].

To determine the probability of successful voice frame transmission during zero slots, the following must be clarified:

- How many zero time slots will occur within a 20 ms interval;
- What is the total number of time slots counted during the 20 ms period;

What is the collision probability for voice frames transmitted in a zero slot.

Since the transmission interval for voice frames is 20 ms, we need to determine how many “zero slots” occur during this time. Let us consider two scenarios where best-effort AC_BE stations transmit frames of 512 or 1500 bytes under saturated load conditions.

“Zero slots” occur immediately after the AIFS_VO interval counted following an acknowledgment frame (ACK or BA) confirming the successful transmission of any data frame by any station, or after the arbitration interval counted following a collision.

Based on this observation, we can propose the following expression to calculate the number of “zero slots” (i.e., arbitration intervals) N_{int} within the voice frame transmission interval:

$$N_{int} = \frac{T_1}{T_{VCW}} \cdot (N_s + N_c), \quad (2)$$

where T_1 – duration of the observation interval (20 ms); T_{VCW} – duration of the virtual contention window; N_s, N_c – number of successfully transmitted frames and number of collisions that occurred during the virtual contention window, respectively.

We consider a network in which each active best-effort station AC_BE operates under saturated load conditions, and the number of such active stations varies from 2 to 16. It is assumed that the number of simultaneously active voice traffic stations can range from 2 to *M*.

To calculate the value of N_{int}, we use the previously provided data from Table 2. The results of the calculations using equation (2), the number of arbitration intervals formed in an IEEE 802.11ac network with a 20 MHz channel bandwidth under saturated load conditions with data frames carrying 512 or 1500 bytes

Table 3 – Approximate number of arbitration intervals N_{int}^{512} and N_{int}^{1500} available in a network with saturated traffic (AC_BE) during a 20 ms interval.

N	N_s	N_c	T_{VCW}^{512}	T_{VCW}^{1500}	N_{int}^{512}	N_{int}^{1500}
2	2	0,066667	605,2545	785,2544	68	52
4	3,999979	0,427256	1169,764	1529,722	75	57
6	5,999272	1,142386	1763,176	2302,24	81	62
8	7,993291	2,282442	2390,852	3104,312	85	66
10	9,968072	3,925061	3056,364	3930,445	90	70
12	11,8948	6,148615	3761,613	4768,249	95	75
14	13,7333	9,023238	4507,34	5600,975	100	81
16	15,43527	12,60204	5293,491	6410,825	105	87

Note. N_{int}^{512} and N_{int}^{1500} refer to the number of arbitration intervals occurring in networks under saturated load conditions with data frame payloads of 512 and 1500 bytes, respectively

of payload, along with the initial calculation parameters, are presented in Table 3.

Assume there are *M* voice traffic stations (AC_VO) in total. On average, m = *M*/4 of these stations will attempt to transmit during “zero slots.” The probability that one of the m AC_VO stations starts transmitting in a given (predefined) zero slot is $1/N_{\mathrm{int}}$. The probability that a given station does not start transmission in a specific zero time slot is $(1\text{-}1/N_{\mathrm{int}})$. If there are (m-1) other AC_VO stations, the probability that none of them start transmitting in the same zero slot as the first station is: $(1\text{-}1/N_{\mathrm{int}})^{(m\text{-}1)}$.

Based on the reasoning presented above, we can formulate an estimate for the probability of successful transmission without any collision P_{s0} for those network stations that load the value 0 into their backoff counter. This probability can be expressed as:

$$P_{0S} = \left(1 - \frac{1}{N_{int}}\right)^{m-1}. \quad (3)$$

The numerical values obtained using equation (3) are presented in Table 4.

The graph of the probability of successful voice frame transmission during zero slots on the first attempt, in networks with different numbers of saturated AC_BE stations, is shown in Fig. 3.

We now determine the probability of successful transmission of voice frames for stations that, at the beginning of their first channel access attempt, loaded a value from the set [1, 2, 3] into the backoff counter. These stations will compete with saturated AC_BE stations and may also compete with each other.

The presence of stations transmitting voice frames may slightly increase the duration of the virtual contention window (VCW). However, this does not reduce the total number of “zero” slots during the 20 ms interval, as the duration of a voice frame is shorter than that of an AC_BE frame carrying 512 or 1500 bytes of payload. Therefore, the proposed approach provides estimates that are close to real values.

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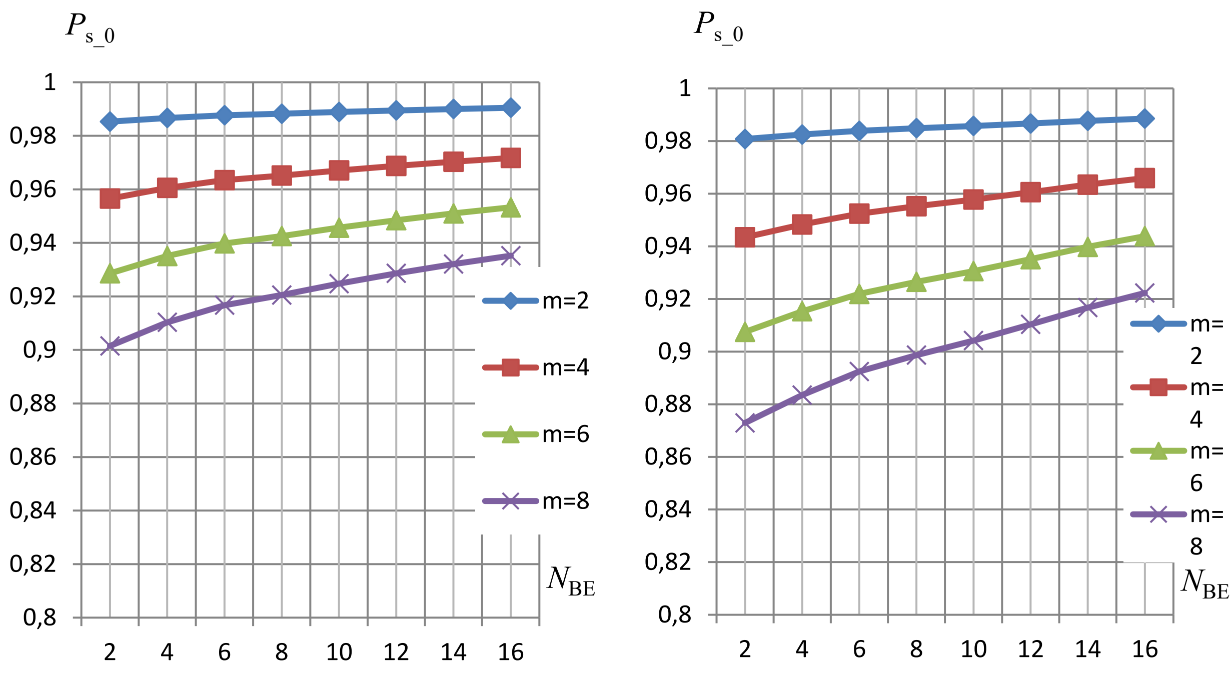
TABLE 4 – PROBABILITY VALUES FOR SUCCESSFUL AC_VO FRAME TRANSMISSION IN A ZERO SLOT

N_{BE}	N_{int}^{1500}	P_{s0}, *m* = 2	P_{s0}, *m* = 4	P_{s0}, *m* = 6	P_{s0}, *m* = 8	N_{int}^{512}	P_{s0}, *m* = 2	P_{s0}, *m* = 4	P_{s0}, *m* = 6	P_{s0}, *m* = 8
2	52	0,980	0,943	0,907	0,872	68	0,985	0,956	0,928	0,901
4	57	0,982	0,948	0,915	0,883	75	0,986	0,960	0,935	0,910
6	62	0,983	0,952	0,921	0,892	81	0,987	0,963	0,939	0,916
8	66	0,984	0,955	0,926	0,898	85	0,988	0,965	0,942	0,920
10	70	0,985	0,957	0,930	0,904	90	0,988	0,967	0,945	0,924
12	75	0,986	0,960	0,935	0,910	95	0,989	0,968	0,948	0,928
14	81	0,987	0,963	0,939	0,916	100	0,990	0,970	0,950	0,932
16	87	0,988	0,965	0,943	0,922	105	0,990	0,971	0,953	0,935



a) b)

Fig. 3 Probability of AC_VO frame transmission in a mixed network with several (m) voice traffic stations and several (N_{BE}) saturated stations: a – packet payload size of 512 bytes; b – packet payload size of 1500 bytes

Since AC_VO stations do not generate saturated traffic and voice traffic transmissions start at random times, the likelihood of competition among voice stations is low when their number is small.

Let us now examine the characteristics of voice frame transmission under conditions of competition with saturated AC_BE traffic. One important feature of voice communication in IEEE 802.11 networks is that both uplink (from client stations to the access point) and downlink (from the access point to client stations) channels are used. A key difference is that client stations may attempt to transmit simultaneously, thereby increasing the risk of collisions. The access point, on the other hand, transmits all voice frames sequentially, reducing the likelihood of collisions. All voice frames sent by the access point are placed in a queue and transmitted one after another.

To estimate the probability of successful voice frame transmission in a mixed network with a single active AC_VO station, we calculate the relative share of free time slots among the total number of time slots observed during the 20 ms interval.

According to the concept of the virtual contention window, in a network under saturated load, the number of free time slots during the duration of this window T_{VCW} is equal to the numerical value of the virtual con-

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TABLE 5 – NUMBER OF FREE TIME SLOTS WITHIN A 20 MS INTERVAL FOR VOICE FRAME TRANSMISSION IN A NETWORK UNDER SATURATED LOAD

N	2		4	6	8	10	12	14	16
VCW	8,036		9,519	11,772	14,905	18,779	23,067	27,367	31,307
T_{VCW}^{512}, μs	605,2		1169,7	1763,1	2390,8	3056,3	3761,6	4507,3	5293,4
Q_1	265,5		162,7	133,5	124,6	122,8	122,6	121,4	118,2
T_{VCW}^{1500}, μs	785,2		1529,7	2302,2	3104,3	3930,4	4768,2	5600,9	6410,8
Q_2	204,6		124,4	102,2	96,0	95,5	96,7	96,7	97,6
$N_{s\ 20\ 512}$	66,08		68,38	68,05	66,86	65,22	63,24	60,93	58,31
$N_{s\ 20\ 1500}$	50,94		52,29	52,11	51,49	50,72	49,89	49,03	48,15

TABLE 6 – PROBABILITIES OF VOICE FRAME TRANSMISSION UNDER COMPETITION WITH SATURATED AC_BE STATIONS

N_{BE}	2	4	6	8	10	12	14	16
$Q_{0\ 512}$	333,83	238,44	214,54	210,64	213,79	218,57	222,41	224,21
$p_{1\ 512}$	0,795	0,682	0,622	0,591	0,574	0,561	0,545	0,527
$Q_{0\ 1500}$	257,32	182,33	164,31	162,23	166,25	172,43	178,98	185,13
$p_{1\ 1500}$	0,795	0,682	0,622	0,591	0,574	0,561	0,545	0,527
$p_{c\ VO}$	0,205	0,318	0,378	0,409	0,426	0,439	0,455	0,473
$P_{S\ VO}$	0,9999	0,9989	0,9971	0,9954	0,9941	0,9929	0,9912	0,9889

tention window VCW. Based on this, the number of free time slots Q during an arbitrary time interval T_1 can be expressed as:

$$Q = \frac{T_1}{T_{VCW}} \cdot VCW. \quad (4)$$

The calculated average number of free time slots that occur during a 20 ms interval for two saturated networks, one with data frames carrying 512 bytes of payload and the other with 1500 bytes, is presented in Table 5 as Q1 and Q2, respectively.

Table 5 also includes the total number of saturated traffic frames $N_{s\_20\_512}$ and $N_{s\_20\_1500}$ successfully transmitted during the 20 ms interval.

The total number of time slots Q0 available within 20 ms is calculated as the sum of: – the slots during which successful frame transmissions began N_s, – the slots during which collisions occurred N_c, – and the free slots VCW:

$$Q_0 = \frac{T_1}{T_{VCW}} \cdot (N_s + N_c + VCW). \quad (5)$$

The probability of successfully transmitting a data frame, p_{1_512} and p_{1_1500}, for AC_VO stations during their first attempt, under competition with saturated AC_BE stations transmitting frames of 512 or 1500 bytes, is calculated as the ratio of the number of free time slots to the total number of time slots using the following expression:

$$p_{1\_512} = \frac{Q_1}{Q_0} \text{ та } p_{1\_1500} = \frac{Q_2}{Q_0}. \quad (6)$$

The numerical values of the probabilities p_{1_512} and p_{1_1500} are presented in Table 6, and their graphical representations are shown in Figure 5. As follows from the obtained data, the probability of a voice frame being successfully transmitted by an AC_VO station on the first attempt, under competition with saturated AC_BE stations, is the same for networks in which AC_BE stations transmit frames with 512 bytes (p_{1_512}) or 1500 bytes (p_{1_1500}) of payload. This is due to the principles of time interval formation and the rules of contention-based medium access in IEEE 802.11ac networks.

Thus, in a mixed network, the probability of successful first-attempt voice frame transmission under competition with saturated AC_BE stations ranges from 0.795 (2 AC_BE stations and 1 AC_VO station) to 0.527 (16 AC_BE stations and 1 AC_VO station).

If a collision occurs during the first attempt, the voice traffic station will double its contention window (CW) once and continue with retransmission attempts for the same voice frame.

Transmitting one additional (voice) frame within a 20 ms interval in a saturated network operating in a quasi-stationary mode will not disrupt this mode: the total number of successfully transmitted frames during this time is between 48 and 68 (Table 5).

Based on the above, we assume that during retransmissions of a single voice frame in a mixed network, the probability of successful transmission remains unchanged and equal to that of the first attempt.

The probability of collision for a voice frame p_{c_VO} in a saturated network is given by the expression:

$$p_{c\_VO} = 1 - p_{1\_512}. \quad (7)$$

If a voice frame encounters a sequence of (R) consecutive collisions, then the probability of successful transmission can be determined using the law of total probability as follows:

$$P_{S_{VO}} = p_{1_{512}} + p_{c_{VO}} \cdot p_{1_{512}} + p_{c_{VO}}^2 \cdot p_{1_{512}} + p_{c_{VO}}^{i-1} \cdot p_{1_{512}} + \dots + p_{c_{VO}}^{R-1} \cdot p_{1_{512}} =$$

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$$= p_{1\_512} \cdot [1 + p_{c\_VO} + p_{c\_VO}^2 + \ldots + p_{c\_VO}^{i-1} + \ldots + p_{c\_VO}^{R-1}] = 1 - p_{c\_VO}^R. \quad (8)$$

Equation (8) reflects the fact that progressing to each new transmission attempt is conditioned by a collision in the previous attempt, with a probability of p_{c_VO}, while the probability of a successful transmission during the current attempt is p_{1_512}.

The final form of equation (8) is derived as the sum of terms in an arithmetic progression, taking into account equation (7).

The calculated values of the probability of successfully transmitting a voice frame (P_{S_VO}) under competition with AC_BE stations, using up to six retransmission attempts, are presented in Table 6. For convenience, Table 6 also includes the collision probability p_{c_VO} for a voice traffic station that gains access to the channel.

The graph illustrating the probability of transmitting a voice frame in a saturated mixed network on the first attempt (p_{1_512}), as well as the probability of successful transmission after up to seven consecutive attempts (P_{S_VO}), is shown in Fig. 4.

We estimate whether a voice traffic station can transmit its voice frame within the voice sampling interval (20 ms) in a network with a large number of saturated stations. To do this, we determine the maximum number of time slots that may be required by a voice station to successfully transmit a voice frame in a mixed network over all seven attempts.

Based on the previously obtained probability of successful voice frame transmission P_{S_VO}, shown in Fig. 4, we assume that seven consecutive attempts are sufficient to ensure delivery of a voice frame. To estimate the maximum number of backoff time slots needed for these seven attempts, we use the data from Table 1.
– During the first access attempt, the contention window for AC_VO stations has a maximum value of CW_{max} = 3. For each of the following attempts (after a collision), CW_{max} = 7. Thus, the maximum number of waiting time slots across all seven attempts is: 4 + 6 × 8 = 52. The approximate number of available time slots during a 20 ms interval is provided in Table 5 – Q_1, Q_2.

The maximum number of time slots that may be required by a voice traffic station to carry out seven consecutive transmission attempts, due to a series of collisions (worst-case scenario), is 52, while the average (most probable) number is 26. According to the calculation results in Table 5, in networks with 2 to 16 saturated AC_BE stations, the number of free time slots available within 20 ms significantly exceeds the number of slots required by a voice station to recover from collisions. If necessary, a voice station can perform more than two full cycles of seven attempts and still successfully transmit a frame. Based on the results shown in Table 6 and Figure 5, it can be concluded that in mixed networks with prioritized traffic, individual voice frames are very likely to be transmitted successfully due to the reduced contention window.

During the 20 ms sampling interval, voice frame transmission will reduce the number of best-effort (AC_BE) frames transmitted within the same period. Based on the above analysis, it can be predicted that a network with prioritized traffic can support a relatively large number of simultaneously active voice stations (e.g., 10–20 stations) without requiring additional system tuning.

However, in saturated networks with frame aggregation, the number of free time slots within a 20 ms interval – during which voice frames could be transmitted –significantly decreases, since the transmission cycle of an aggregated frame is much longer than that of a regular data frame. It should also be taken into account that IEEE 802.11ac technology is designed primarily for indoor use, and the coverage area of a single access point is limited. In such an area, the simultaneous presence of a large number of active voice sessions is unlikely or even infeasible. The transmission duration of maximum-size aggregated frames in an IEEE 802.11ac network with a 20 MHz channel is 5424 µs, which is 12.85 times longer than the transmission cycle of a single non-aggregated frame with 1500 bytes of payload (421.9 µs). Thus, the number of free time slots within a 20 ms interval in the examined mode, according to the data in Table 5, ranges from 10 to 20. This number of free time slots can support no more than 1–2 stable voice sessions.

Difficulties with using voice connections in networks with mixed traffic may also arise due to software configurations made by network equipment manufacturers. For example, if a certain number of collisions occur, the access point may limit the number of active users and deny others access to the channel without terminating their association. In such a case, the client remains connected to the network but is unable to initiate a connection. Another potential configuration

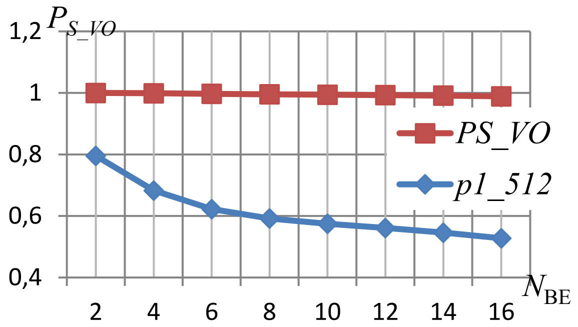

Fig. 4 – Probability of successful voice frame transmission by a single AC_VO station in a mixed network with multiple saturated AC_BE stations

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is one where the access point switches to a lower Modulation and Coding Scheme (MCS) index in response to collisions. This reduces the network's throughput and degrades the performance of the mixed traffic environment [5].

Conclusions

As a result of the study on the features of voice traffic transmission for organizing IP telephony services in office environments using IEEE 802.11ac networks, it was found that, with the use of traffic prioritization, high-quality service can be ensured for a large number of users even under conditions of intensive mixed traffic transmission.

The results were obtained based on the characteristics of the G.711 codec, which generates a relatively large block of digital data at the output. However, these results can also be extended to networks using other voice codecs that produce data blocks of equal or smaller size.

With an increasing number of stations in the network transmitting lower-priority traffic (AC_BE), the probability of collisions increases. As a result, the likelihood of a voice frame (AC_VO) being transmitted on the first attempt when the backoff counter is initialized with a "0" also increases.

When designing a mixed network with support for voice connections, it is advisable to avoid frame aggregation for low-priority traffic, or to implement adaptive network management in which frame aggregation is disabled in the presence of active voice connections.

To organize a network with a dedicated channel for voice connections, it is recommended to use a channel with a 20 MHz radio frequency bandwidth.

The above conclusions apply to networks where IP telephony services are provided using traffic prioritization in accordance with the IEEE 802.11e specification, with voice traffic assigned to the AC_VO priority, and all other data transmitted in the same network assigned to the AC_BE.

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Особливості організації голосових з’єднань в безпроводових мережах IEEE 802.11AC з пріоритизацією трафіку

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Анотація - Метою дослідження є визначення теоретично обґрунтованих обмежень передавання голосового трафіку в безпроводових мережах IEEE 802.11ac з пріоритизацією змішаного трафіку. Для дослідження застосовано математичну модель функціонування безпроводової мережі стандарту IEEE 802.11, що ґрунтується на концепції віртуального конкурентного вікна. Параметри голосових кадрів обрано відповідними кодеку G.711, результати для якого можна розповсюдити на голосовий трафік інших сучасних кодеків. Проаналізовано механізми пріоритизації стандарту IEEE 802.11e, що реалізуються через зменшення обов'язкового часу очікування та скорочення додаткового інтервалу очікування для кадрів з високим пріоритетом. Отримано числові дані щодо ймовірності передавання кадрів голосового трафіку AC_VO в умовах конкуренції з трафіком загального пріоритету AC_BE. Встановлено, що ймовірність передати голосовий кадр з першої спроби у разі вибору «нульового» часового слоту коливається від 0,98 (одна голосова станція і дві станції загального пріоритету) до 0,922 (вісім голосових станцій і шістнадцять станцій загального пріоритету). Для голосових станцій у режимі конкуренції ймовірність успішної передачі з першої спроби становить від 0,795 до 0,527 залежно від кількості станцій з насиченим навантаженням. Визначено, що ймовірність передавання голосових кадрів із застосуванням повторних спроб за наявності послідовних колізій становить не менше 0,98 за умови використання шести повторних спроб. Отримано подальший розвиток методу оцінювання експлуатаційних характеристик безпроводових мереж IEEE 802.11 із застосуванням математичної моделі віртуального конкурентного вікна для мереж з пріоритизованим трафіком. Одержані результати корисні для оптимізації функціонування та проєктування локальних безпроводових мереж стандарту IEEE 802.11ac з функцією VoIP.

***Ключові слова:** безпроводова мережа; голосовий кодек; доступ до середовища передавання; ймовірність передавання; пріоритизований трафік; стандарт IEEE 802.11ac; VoIP*