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## RESEARCH OF AUDIO AND VIDEO TRAFFIC THE CHARACTERISTICS IN LOW-BANDWIDTH RADIO COMMUNICATION NETWORKS

*The construction of mass service systems, namely automated control systems, requires preliminary analysis and modeling of traffic in their communication networks. Mathematical models of various types of traffic have been developed for public networks, which allows to estimate the necessary functional characteristics of equipment for building a communication network, depending on the number of users. Low-bandwidth communication networks, which are built on the basis of ultra high frequency and very high frequency (UHF/VHF) radio stations, are distinguished by low speed, high delay and jitter of data transmission. To work in such communication networks, special data transmission protocols are adapted and developed. In this paper, a study of the characteristics of audio and video traffic in low-bandwidth communication networks is carried out, which are built on the basis of UHF/VHF radio stations, which will allow creating a software implementation of simulated traffic modeling for the further determination of the services availability at the stage of planning and designing the communication system. Two personal computers and two modern RF-7850M-HH UHF/VHF radio stations were used to study the characteristics of audio traffic. The radio stations worked in three operating modes: narrowband mode with a fixed carrier frequency FF, narrowband mode with pseudo-random adjustment of the operating frequency QL1A, and wideband mode ANW2C. The voice was transmitted in digital mode using the built-in MELP 2400 codec, and the "iperf-v2.0.5" software was used to determine the characteristics of the audio traffic on personal computers connected to these radio stations. Two personal computers, two modern RF-7850M-HH UHF/VHF radio stations, a video encoder and a video camera were used to study video traffic characteristics. The radio stations operated in ANW2C broadband mode. To evaluate the characteristics of the video traffic, the Wireshark software was used on a personal computer, with the help of which the video broadcast from the video encoder was presented. It was found that voice transmission in low-speed communication networks based on UHF/VHF radio stations occupies a bandwidth of 2 Kbit/s - 2.5 Kbit/s, and when voice and data are simultaneously transmitted in radio stations, data buffering and jitter increase. The resolution, bitrate, FPS, and necessary bandwidth of video traffic that can be transmitted via UHF/VHF radio communication channels are determined. Based on the conducted research, recommendations are provided for the transmission of video traffic through low-bandwidth communication channels.*

*Key words: low-bandwidth communication networks, traffic, audio, video, radio station, UHF, VHF, performance quality indicators, data transfer rate, jitter, channel loss, packet, video resolution.*

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## ДОСЛІДЖЕННЯ ХАРАКТЕРИСТИК АУДІО ТА ВІДЕОТРАФІКУ В НИЗЬКОШВИДКІСНИХ РАДІОМЕРЕЖАХ ЗВ'ЯЗКУ

*Побудова систем масового обслуговування, а саме автоматизованих систем управління потребують попереднього аналізу та моделювання трафіку в їх мережах зв'язку. Для мереж загального користування на сьогоднішній час розроблено математичні моделі різного типу трафіку, що дозволяє в залежності від кількості користувачів оцінити необхідні функціональні характеристики обладнання для побудови мережі зв'язку. Низькошвидкісні мережі зв'язку, які побудовані на базі ультракороткохвильових радіостанцій, відрізняються низькою швидкістю, великою затримкою та джиттером передачі даних. Для роботи в таких мережах зв'язку адаптують та розробляють спеціальні протоколи передачі даних. У даній роботі проводиться дослідження характеристик аудіо та відеотрафіку в низькошвидкісних мережах зв'язку, які побудовані на базі ультракороткохвильових радіостанцій, що дозволить створити програмну реалізацію імітаційного моделювання трафіку щодо подальшого визначення доступності сервісів на етапі планування та проектування системи зв'язку. Для дослідження характеристик аудіотрафіку було використано два персональних комп'ютери та дві сучасні ультракороткохвильових радіостанцій RF-7850M-HH. Радіостанції працювали в трьох режимах роботи: вузькосмуговий режим з фіксованою несучою частотою FF, вузькосмуговий режим з псевдовипадковою перебудовою робочої частоти QL1A та широкосмуговий режим ANW2C. Передача голосу відбувалася у цифровому режимі з використанням вбудованого кодеку MELP 2400, а для*

визначення характеристик аудіотрафіку використовувалось програмне забезпечення «jperf-v2.0.5» на персональних комп'ютерах, що підключені до цих радіостанцій. Для досліджень характеристик відеотрафіку було використано два персональних комп'ютери, дві сучасні ультракоротковхвильових радіостанцій RF-7850M-HH, відеоенкодер та відеокамера. Радіостанції працювали в широкосмуговому режимі ANW2C. Для оцінки характеристик відеотрафіку використовувалось програмне забезпечення «Wireshark» на персональному комп'ютері, за допомогою якого була представлено відеотрансляцію від відеоенкодера. Було отримано, що передача голосу в низькошвидкісних мережах зв'язку на базі УКХ радіостанцій займає полосу пропускання 2 Кбіт/с - 2,5 Кбіт/с, а також при одночасній передачі голосу та даних в радіостанціях збільшується буферизація даних та джиттер. Визначено роздільну здатність, бітрейт, FPS, необхідну полосу пропускання відеотрафіку, що може передаватися через УКХ радіоканали зв'язку. На основі проведених досліджень надано рекомендації щодо передачі відеотрафіку через низькошвидкісні канали зв'язку.

Ключові слова: низькошвидкісні мережі зв'язку, трафік, аудіо, відео, радіостанція, УКХ, показники якості функціонування, швидкість передачі даних, джиттер, втрати в каналі, пакет, роздільна здатність відео.

## Introduction

Modeling the load on the communication system during the construction of automated control systems allows you to obtain the necessary functional characteristics of the equipment even at the stage of designing such systems. For public networks, the characteristics and types of traffic circulating in these communication networks have already been defined. For example, for the LTE network, several QoS (Quality of Service) service classes [1], service traffic parameters (service duration, required bandwidth) are described, and the law of interval distribution between service requests for each type of service is defined [1,2]. Unfortunately, for the construction of automated control systems in low-speed communication networks based on ultra high frequency and very high frequency (UHF/VHF) radio stations, service classes and traffic characteristics are not described. UHF/VHF radio networks are characterized by low speed, high delay and jitter of data transmission [3], so special transmission protocols are developed or existing ones are adapted to work in such communication networks.

In works [4, 5], automated control systems in low-bandwidth communication networks were simulated, the STANAG 4677 and AdatP-36 standards traffic indicators were obtained, and the distribution density of time intervals between the moments of packets arrival for the maintenance of these standards was determined. In [6], a study of file transfer protocols in the UHF/VHF radio network was conducted, traffic characteristics were obtained, and recommendations were given for the use of these protocols. Through modern low-bandwidth radio networks based on UHF/VHF radio stations, it is possible to transmit audio and video traffic.

Traffic parameterization in radio communication networks is performed according to QoS service classes [1, 2]: conversational, interactive, streaming and background. Audio and video traffic services are included in the conversational class of services. Each type of service has its own law of distribution of intervals between requests for service, for example, for audio traffic - it is exponential, and for video conferences - Pareto distribution [1, 2]. According to ITU-R.1768 [1], it is determined that audio traffic is transmitted at a speed of 64 Kbit/s and a service duration of 90 - 450 s. But in low-bandwidth communication networks built on the basis of UHF/VHF radio stations, the maximum bandwidth usually does not exceed 64 Kbit/s, and the value of jitter is much higher than in public networks. For the transmission of voice messages in such communication networks, special codecs are used, which have their own speed and influence on the jitter of data transmission in UHF/VHF radio networks.

The purpose of the work is to determine the bandwidth required for the transmission of audio traffic, to investigate the effect of voice transmission on jitter in low-bandwidth communication networks, to determine the optimal characteristics of the video stream and its bandwidth in communication channels built on the basis of UHF/VHF radio stations, with regard to the further implementation of simulation modeling traffic and determining the availability of services at the stage of planning and designing the communication system.

## Audio traffic research

To conduct a study of audio traffic indicators in low-speed communication networks built based on UHF/VHF radio stations, the scheme shown in Figure 1 is being built. Modern RF-7850M-HH radios manufactured by Harris are used as UHF/VHF radio stations. The radio stations are located at a distance of 100 m from each other, personal computers (PCs) are connected to the radio stations using a special Ethernet cable with RJ-45 (12067-5220-01) from the set of radio station accessories. Antennas of the pin type are connected to the radio stations. The power of the radio stations is set to 1 W, the measurement is carried out in an open space at a temperature of 24-26°C, air humidity of 60% and a pressure of 761 mm Hg.



Fig. 1. Communication network diagram for audio traffic research

To evaluate the characteristics of audio traffic, we will use three operating modes of radio stations: FF - narrowband mode with a fixed carrier frequency, ANW2C - wideband mode, QL1A - narrowband mode with pseudorandom adjustment of the operating frequency. Voice in all operating modes is coded and compressed using the MELP 2400 codec [7]. MELP 2400 is a low-speed digital audio codec with a bit rate of 2400 bit/s, adopted by the NATO standard STANAG-4591 [7]. To transmit voice messages, it is enough to press the side button and start speaking. The use of a digital codec allows simultaneous transmission of voice and data over low-bandwidth communication channels built based on modern UHF/VHF radio stations. Radio stations are configured with the ability to transmit voice messages in parallel with data transmission.

The "iperf-v2.0.5" software is installed on personal computers. Before starting the study of traffic characteristics, it is necessary to determine the parameters of the radio channel between personal computers. According to work [3], we will measure the bandwidth and jitter of the low-bandwidth radio network in three operating modes: FF, ANW2C and QL1A (Table 1) for three minutes.

Tabl 1.

Measurement results without voice transmission

Operating mode	Bandwidth	Jitter
FF	64,8 Kbit/s	258,31 ms
ANW2C	357 Kbit/s	46,07 ms
QL1A	15,8 Kbit/s	955,81 ms

Figure 2 shows a fragment of the "iperf-v2.0.5" program when measuring bandwidth and jitter of a low-bandwidth radio network in FF mode.

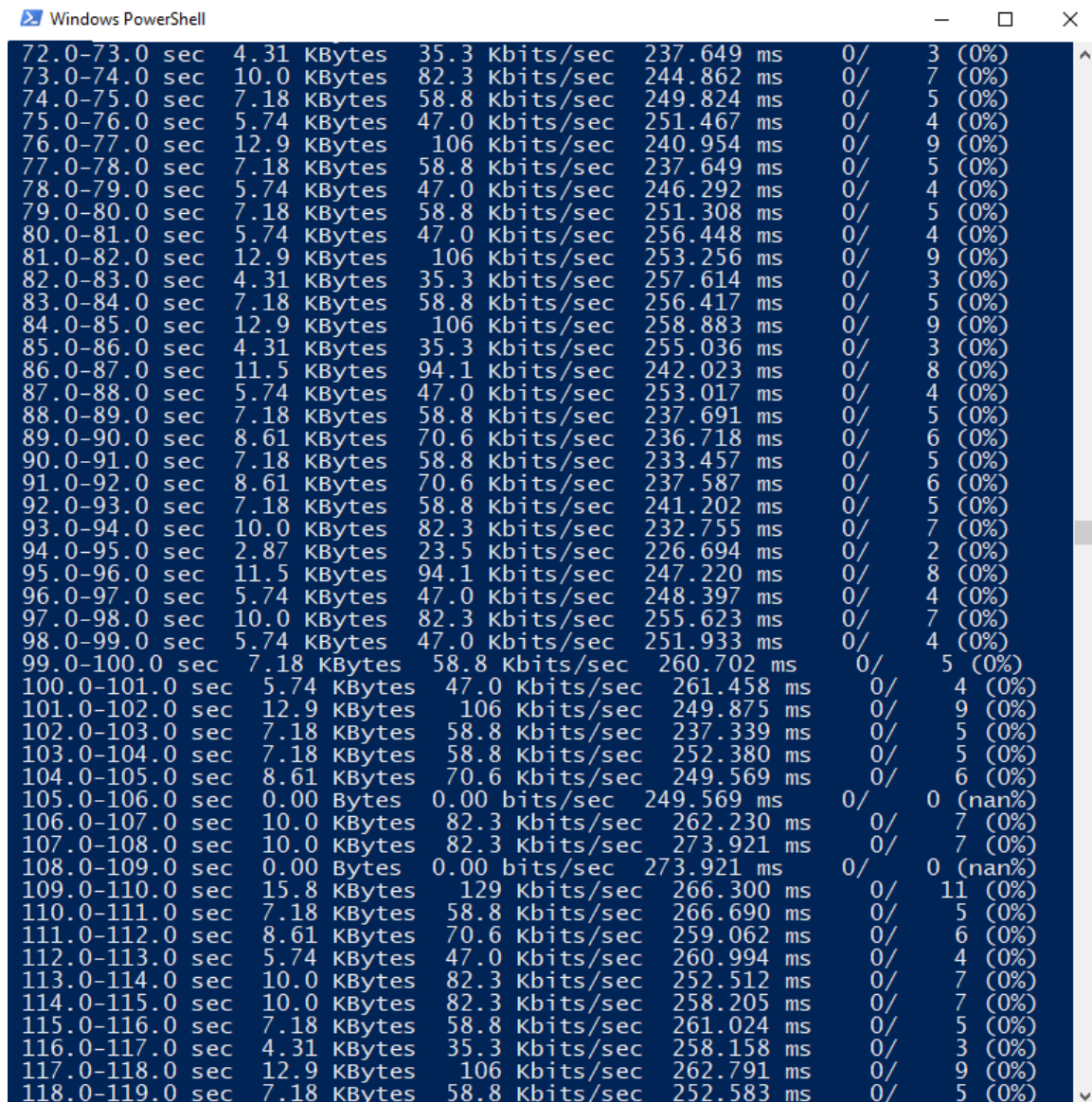


Fig. 2. Measurement of bandwidth and jitter in FF mode

To evaluate the influence of voice transmission on the UHF/VHF parameters of the communication network, we will conduct the following experiment. According to work [3], we will measure bandwidth and jitter, but during the measurement we will transmit a voice message from one VHF radio station to another. The results of the study are shown in Table 2.

Table 2.

Measurement results with voice transmission

Operating mode	Bandwidth	Jitter
FF	62,3 Kbit/s	310,11 ms
ANW2C	355 Kbit/s	49,28 ms
QL1A	13,4 Kbit/s	1005 ms

Figure 3 shows a fragment of the "iperf-v2.0.5" program when measuring the bandwidth and jitter of a low-bandwidth radio network in FF mode with voice transmission.

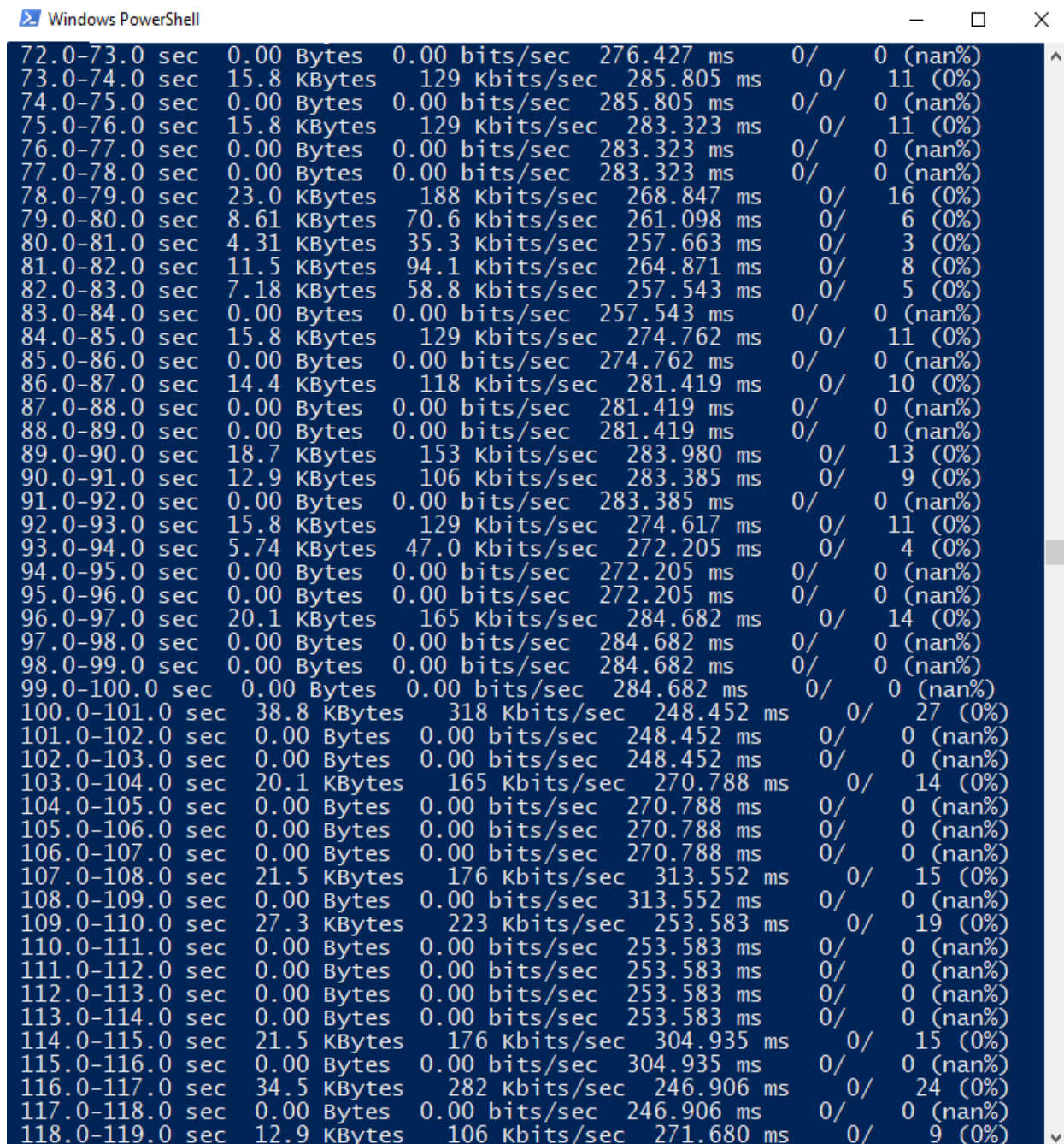


Fig. 3. Measurement of bandwidth and jitter when transmitting voice in FF mode

If you compare the operation of the "iperf-v2.0.5" program without voice transmission and with voice transmission in FF mode, you can see that transmitting voice in radio stations, more buffering of data occurs than when transmitting only data (Fig. 2 and Fig. 3). This buffering affects the jitter in the communication channel, which can be seen from the results of the "iperf-v2.0.5" program (Table 1 and Table 2). Data buffering also occurred in the ANW2C and QL1A operating modes, which led to an increase in jitter with these operating modes (Table 2).

It is not difficult to see that the QoS parameters of a low-bandwidth communication network is degraded when transmitting voice. For FF mode, bandwidth decreased by 2.5 Kbit/s and jitter increased by 52.2 ms, for ANW2C mode, the bandwidth difference is 2 Kbit/s, jitter is 3.21 ms, and for QL1A mode, the bandwidth difference is 2.4 Kbit/s, jitter 49.91 ms, respectively. The difference in QoS indicators of a low-bandwidth communication network without voice transmission and with voice transmission is shown in Table 3.

Table 3.

**The difference in bandwidth and jitter without voice and with voice transmitting**

Operating mode	Bandwidth	Jitter
FF	-2,5 Kbit/s	+52,2 ms
ANW2C	-2 Kbit/s	+3,21 ms
QL1A	-2,4 Kbit/s	+49,91 ms

Thus, voice transmission in low-bandwidth communication networks based on UHF/VHF radio stations occupies a bandwidth of 2 Kbit/s - 2.5 Kbit/s, which approximately corresponds to the bitrate of the MELP 2400 audio codec. Based on the obtained traffic characteristics, it is possible to create a model for simulating audio traffic in low-bandwidth communication networks to further determine the availability of services at the stage of planning and designing the communication system.

#### Video traffic research

It is advisable to evaluate video traffic parameters only in broadband ANW2C mode, since in other modes of operation there is not enough bandwidth, high delay and jitter of data transmission [3]. To conduct a study of video traffic indicators in low-bandwidth communication networks based on UHF/VHF radio stations, the scheme shown in Figure 2 is being built. Modern RF-7850M-HH radios manufactured by Harris are used as UHF/VHF radios. The radio stations are located at a distance of 100 m from each other, PCs are connected to the radio stations using a special Ethernet cable with RJ-45 (12067-5220-01) from the set of radio station accessories. Antennas of the pin type are connected to the radio stations. The power of the radio stations is set to 1 W, the measurement is carried out in an open space at a temperature of 24-26°C, air humidity of 60% and a pressure of 761 mm Hg.

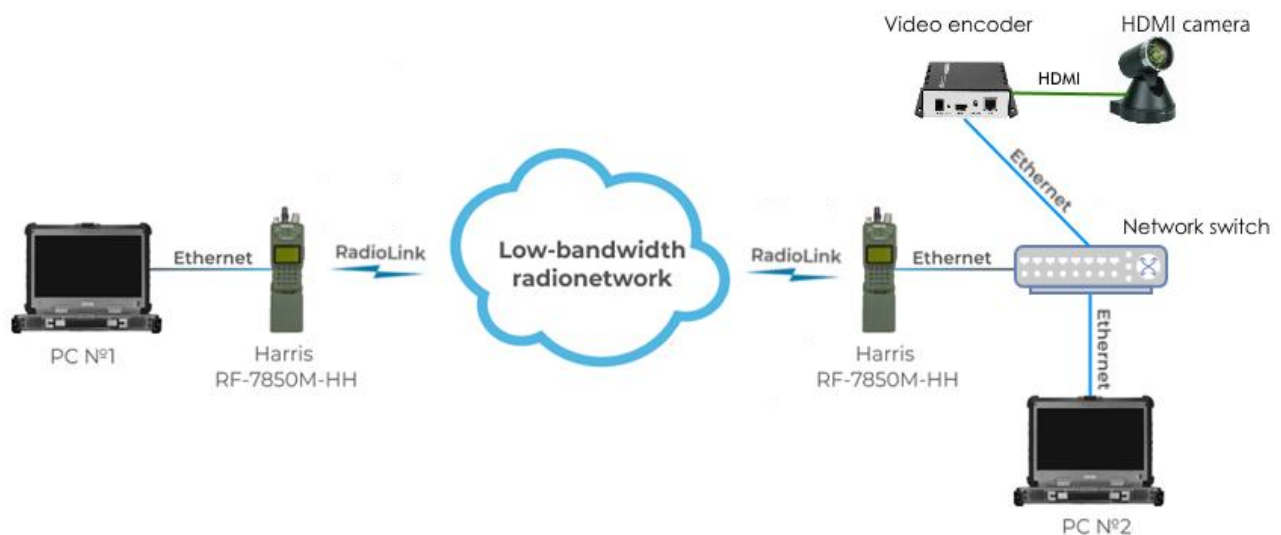


Fig. 4. Communication network diagram for video traffic research

As a source of video traffic, we will use a video encoder and a video camera connected to it. The video encoder has the ability to configure the following parameters of the video stream (Fig. 5):

- number of frames per second (FPS);
- bitrate (Bitrate);
- video resolution (Encoder size);
- H.264 level (H.264 Level);

- bitrate control mode (Bitrate control);
- network video transmission settings.

**Main stream**

FPS:  [5-60]

Bitrate(kbit):  [32-32000]

Encoded size:  ▼

H.264 Level:  ▼

Bitrate control:  ▼

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RTSP URL:   ▼

Multicast IP:   ▼

Multicast port:  [1-65535]

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RTMP PUBLISH URL:   ▼

rtmp://ip/xxx/xxx or rtmp://user:pass@ip/xxx/xxx

Fig. 5. Setting the parameters of the video stream

To transmit a video stream, the encoder is configured to transmit data to a multicast group. This setting allows you to receive video on both PCs. PC №2 receives video over the local network, and PC №1 - through the radio network. The "VLC mediaplayer v.3.0.7.1" program is used to display the video stream on the PC. The software "Wireshark v.3.4.3" is used to analyze the video traffic on both PCs, which allows you to compare the traffic before and after transmission over the radio channel.

We will transmit the video every 2 minutes, changing the parameters: number of frames per second, bitrate and resolution. Let's consider the parameter of the number of frames per second equal to five and the bitrate - 32 Kbit/s. The results of the study are shown in Table 4.

Table 4.

**The results of the video traffic study at FPS - 5, and the bitrate - 32 Kbit/s**

№	Video resolution	PC	Number accepted packages	Loss, %	Average, Kbit/s
1	320x240	2	483	-	43
2	320x240	1	468	3,1	42
3	640x360	2	473	-	42
4	640x360	1	462	2,3	41
5	850x480	2	490	-	44
6	850x480	1	474	3,3	42
7	1280x720	2	520	-	47
8	1280x720	1	499	3,9	44
9	1920x1080	2	790	-	71
10	1920x1080	1	701	12,7	63

The video in all experiments on both PCs was blurry, pixels appeared on the image when the camera or the user moved, and the loss of video data was visible when transmitting video with a resolution of 1920x1080, which led to the freezing of the image.

Let's set the bitrate parameter to 64 Kbit/s and repeat the traffic study. The results of the study are shown in Table 5.

Table 5.

**The results of the video traffic study at FPS - 5, and the bitrate – 64 Kbit/s**

№	Video resolution	PC	Number accepted packages	Loss, %	Average, Kbit/s
1	320x240	2	880	-	79
2	320x240	1	870	1,1	79
3	640x360	2	883	-	79
4	640x360	1	869	1,6	78
5	850x480	2	886	-	79
6	850x480	1	865	2,4	78
7	1280x720	2	910	-	81
8	1280x720	1	865	4,9	78
9	1920x1080	2	1172	-	104
10	1920x1080	1	1070	8,7	96

Video with a resolution of 320x240, 640x360 and a bit rate of 64 Kbit/s did not differ in image quality from a bit rate of 32 Kbit/s. Video with a resolution of 850x480, 1280x720 and a bit rate of 64 Kbit/s was transmitted much better compared to a bit rate of 32 Kbit/c, the facial expressions of the user who was conducting a conversation and video transmission were clearly visible, the blurring of the image was insignificant. The loss of video data was visible when transmitting video with a resolution of 1920x1080, which led to the freezing of the image

Let`s set the bitrate parameter to 96 Kbit/s and repeat the traffic study. The results of the study are shown in Table 6.

Table 6.

**The results of the video traffic study at FPS - 5, and the bitrate – 96 Kbit/s**

№	Video resolution	PC	Number accepted packages	Loss, %	Average, Kbit/s
1	320x240	2	1269	-	114
2	320x240	1	1250	1,5	114
3	640x360	2	1285	-	115
4	640x360	1	1275	0,8	115
5	850x480	2	1320	-	119
6	850x480	1	1268	4	114
7	1280x720	2	1341	-	121
8	1280x720	1	1266	5,6	114
9	1920x1080	2	1468	-	132
10	1920x1080	1	1337	8,9	120

Video at 320x240, 640x360 and 96 Kbit/s bitrate was less blurry compared to 64 Kbit/s or 32 Kbit/s bitrates, but still appeared pixelated when the camera or user moved. When transmitting video with a resolution of 850x480, 1280x720 and 1920x1080, a loss of video data was visible, which led to the image freezing every few seconds.

Let`s set the bitrate parameter to 128 Kbit/s and repeat the traffic study. The results of the study are shown in Table 7.

Table 7.

**The results of the video traffic study at FPS - 5, and the bitrate – 128 Kbit/s**

№	Video resolution	PC	Number accepted packages	Loss, %	Average, Kbit/s
1	320x240	2	1650	-	148
2	320x240	1	1635	0,9	147
3	640x360	2	1708	-	154
4	640x360	1	1679	1,7	150
5	850x480	2	1720	-	155
6	850x480	1	1552	9,8	149
7	1280x720	2	1730	-	156
8	1280x720	1	1506	13	135
9	1920x1080	2	1750	-	157
10	1920x1080	1	1389	20,6	125

Video with a resolution of 320x240, 640x360 and a bitrate of 128 Kbit/s was blurry and the pixels in the image also appeared when the camera or the user moved. When transmitting video with a resolution of 850x480, 1280x720 and 1920x1080, the loss of video data was visible, which led to the freezing of the image every second.

In subsequent experiments, increasing the FPS or bitrate only resulted in greater loss of video data and constant freezing of the image or its complete absence.

Attempting to transmit two video streams simultaneously from two video encoders resulted in loss of video data and constant freezing of the image or its complete absence, even with an image resolution of 320x240, a bit rate of 32 Kbit/s and an FPS value of 5.

### Recommendations

Based on the video traffic research, the following recommendations can be made:

1. Narrowband FF and QL1A modes are impractical to use for video traffic transmission due to insufficient bandwidth, high delay, jitter, and the probability of data transmission loss, so it is necessary to use only wideband operating mode - ANW2C.
2. It is impractical to transmit video with a resolution higher than 1280x720, as video with a higher resolution leads to significant data loss - from 8.7% to 20.6%.
3. If it is necessary to receive a clear image when transmitting video traffic over UHF/VHF radio networks, it is necessary to use video with a resolution of 850x480, 1280x720, a bit rate from 64 Kbit/c to 96 Kbit/c and an FPS value equal to 5. Otherwise, it is advisable to use a video with a resolution 320x480, 640x360 to minimize loading of the permissible bandwidth.

### Conclusions

1. A study of the influence of voice message transmission on jitter in low-bandwidth communication networks was carried out and indicators of the bandwidth occupied by voice message transmission in three operation modes of the UHF/VHF radio station were obtained: FF, ANW2C and QL1A. It was determined that voice transmission in UHF/VHF radio networks occupies a bandwidth of 2 Kbit/s - 2.5 Kbit/s. In the narrowband operation modes of UHF/VHF radio stations, there is a significant increase in data transmission jitter approximately by 50 ms.

2. It was determined that the maximum resolution of the video should not exceed 1280x720 when transmitting video traffic through the UHF/VHF radio communication channel, otherwise the video with a higher resolution leads to significant data loss - from 8.7% to 20.6%.

3. The value of the loss during the transmission of video traffic is from 0.8% to 20.6%, and the necessary bandwidth for the transmission of video traffic through low-bandwidth communication networks is from 43 Kbit/s to 157 Kbit/s at a bit rate of 32 Kbit/s up to 128 Kbit/s and FPS values - 5.

4. Recommendations for the transmission of video traffic through low-bandwidth communication channels, which are based on UHF/VHF radio stations, are provided.

5. Based on the received audio traffic characteristics, it is possible to create a software implementation of simulated traffic modeling in low-bandwidth communication networks for the further determination of service availability at the stage of planning and designing the communication system.

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