Analysis of technological requirements for IP-based modernization of aeronautical telecommunication network in Ukraine

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Abstract

The aviation infrastructure of Ukraine constantly needs to modernize telecommunication networks, especially in the context of the upcoming end of the war and the opening of airspace over Ukraine. A comprehensive analysis of technical and regulatory standards established by international organizations ICAO and EUROCAE, and systematization of their requirements in the direction of implementing a convergent network architecture ATN/IPS is a fundamental task. This work considers the need to replace outdated TDM-based networks with modern IP technologies. The main requirements for protocols at key levels of the OSI model, including IPv6, TCP/UDP, IPsec, IKEv2, SIP and RTP, as well as recommendations for quality of service. The optimal selection of network equipment parameters aimed at minimizing packet transmission delays to meet performance standards necessary for reliable operation of the ATC system is recommended. A mathematical model based on a discrete Markov system is proposed for predicting the load on the aviation telecommunications network of Ukraine using data on passenger traffic between regional airports.

Keywords

aeronautical telecommunications, ATN/IPS, voice quality, E-model

1. Introduction

The advancement of aviation infrastructure and information transmission technologies underscores the critical role of network communication in ensuring air traffic safety and operational efficiency.

Following the end of the war and the reopening of Ukrainian airspace, restoring and modernizing the country's aviation infrastructure will be a strategic priority to guarantee safe and efficient flight operations. A crucial component of this effort involves integrating advanced data transmission technologies into the Aeronautical Telecommunications Network (ATN). Transitioning to the modern ATN/IPS (Aeronautical Telecommunications Network over Internet Protocol Suite) architecture will be essential for this upgrade.

The aviation telecommunications network, based on the integrated ATN/IPS system, serves as the foundation for organizing the exchange of operational information among air traffic control services, airlines, and other airspace participants. In this context, ensuring high-quality and secure data transmission is essential. The International Civil Aviation Organization (ICAO) and the European Organization for Aviation Equipment (EUROCAE) have established a set of requirements and standards aimed at enhancing the reliability, security, and efficiency of network systems. They address a wide range of issues, from architectural solutions and data transmission protocols to mechanisms for protecting against cyber threats and other forms of external influence. Analyzing the ICAO and EUROCAE requirements

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enables not only the assessment of the current state of the ATN/IPS network but also the identification of areas for modernization and enhancement. The integration of the traditional ATN aviation network with modern packet data transmission technologies opens up new opportunities for optimizing air traffic control processes. Contemporary trends in aviation demand that existing solutions adapt to new operating conditions, which requires a thorough analysis and comparative assessment of the requirements of international organizations.

2. Analysis of research and publications

A review of recent publications reveals that the transition to ATN/IPS in Ukraine remains an underexplored topic. While some documents, such as in [1], provide a broad analysis of global IP migration requirements for aviation networks, others, like in [2], specifically focus on cybersecurity and data protection. However, detailed research on Ukraine's specific challenges and implementation strategies is lacking. This gap presents a significant opportunity for further investigation and the development of practical recommendations tailored to the Ukrainian aviation sector.

3. Statement of the task

The objective of this article is to conduct a comprehensive analysis and systematization of ICAO and EUROCAE requirements related to ensuring high-quality data transmission in converged ATN/IPS networks. Additionally, the study explores practical approaches for implementing these standards within Ukraine's aviation infrastructure, particularly in the post-war recovery phase. The findings will help identify critical challenges and propose solutions for modernizing Ukraine's aviation telecommunications network, aligning it with global best practices. Given the ongoing shift from traditional TDM-based voice communications (used in air traffic control) to Voice over IP (VoIP) and converged IP networks, this research is both timely and essential. International organizations, including ICAO, EUROCAE, SESAR, and the FAA's NextGen program, have already established frameworks for this transition. Ukraine's successful adoption of these standards will enhance aviation safety, operational efficiency, and interoperability with global air traffic management systems once airspace reopens.

4. Key study

As stipulated in [1], the following requirements must be met:

- IPv6 serves as the primary network protocol to enable global addressing and scalability [3, 4] with IPv4 permitted only in limited scenarios;
- TCP is mandated at the transport layer for reliable communication [5];
- UDP is employed for latency-sensitive services, such as voice communications;
- Security measures are enforced through IPsec and IKEv2 protocols, with detailed security requirements analyzed in [2] ;
- Air traffic control (ATC) voice communications must utilize VoIP technology;
- BGP is designated for inter-domain routing;
- Integration with System Wide Information Management (SWIM) must be ensured;
- Support for SATCOM, terrestrial radio links, and VPNs (Virtual Private Networks) is required.

For VoIP-based voice transmission, the EUROCAE WG-67 working group has developed a series of guidance documents [6, 7, 8, 9, 10]. These documents align with the "Vienna Agreement," which outlines the components of the VoIP ATM (Air Traffic Management) system and their interconnections, as illustrated in Figure 1.

The requirements for VoIP ATM systems must be clearly defined at every level of interaction, whether through radio or telephone communications. This includes specifying physical interfaces, functional

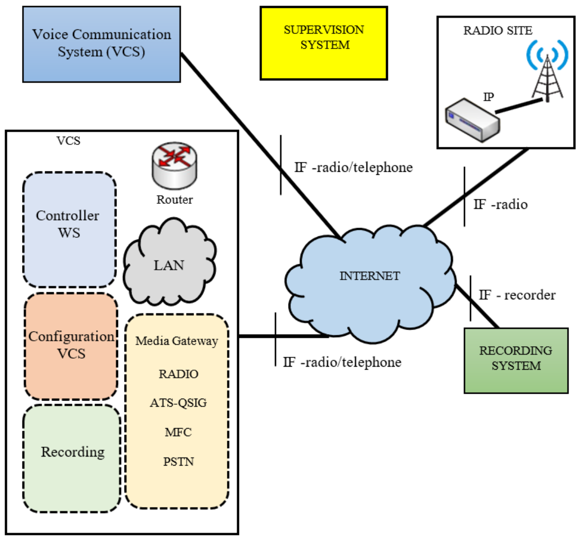

Figure 1: Vienna agreement.

capabilities, integration with existing systems, verification against established protocols such as Session Initiation Protocol (SIP) [11] and H.323 [12], as well as implementing robust monitoring, security protections, and quality of service measures.

An analysis of the relevant documentation allows for the systematization of minimum requirements that manufacturers and users of VoIP components in ATM systems must meet. These requirements are designed to guarantee performance, security, and seamless interoperability across different systems. They encompass both technical specifications and operational considerations for voice and data communications, whether between air navigation service providers (ground-ground communications) or between air traffic controllers and aircraft (air-ground communications).

A fundamental requirement for VoIP ATM systems is ensuring minimal latency in the transmission of critical aviation messages. This necessitates a structured approach to network traffic prioritization, where data streams are categorized based on their operational importance, ranging from critical and high-priority to medium- and low-priority communications. Maintaining signal integrity is equally vital to prevent distortion or loss of voice transmissions. Additionally, control signals must be delivered with high precision, adhering to stringent latency limits where the maximum permissible delay does not exceed 100 milliseconds in terrestrial network environments, as illustrated in Figure 2.

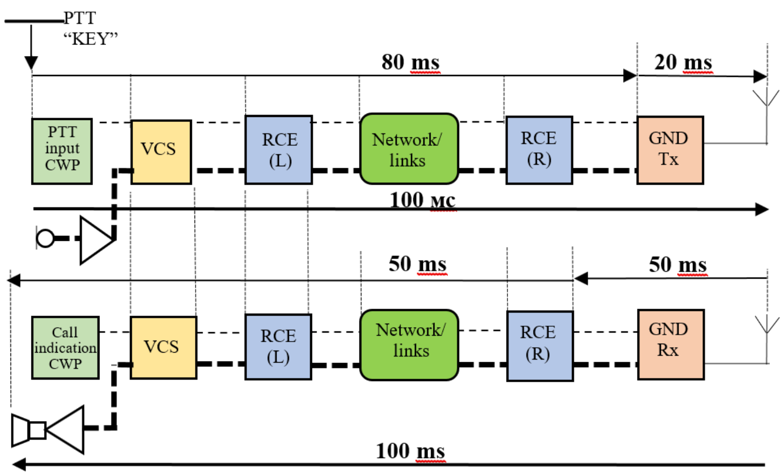

Figure 2: Requirements for control signal delay.

The transmission delay of voice packets for terrestrial components should not exceed 130 ms (Figure 3).

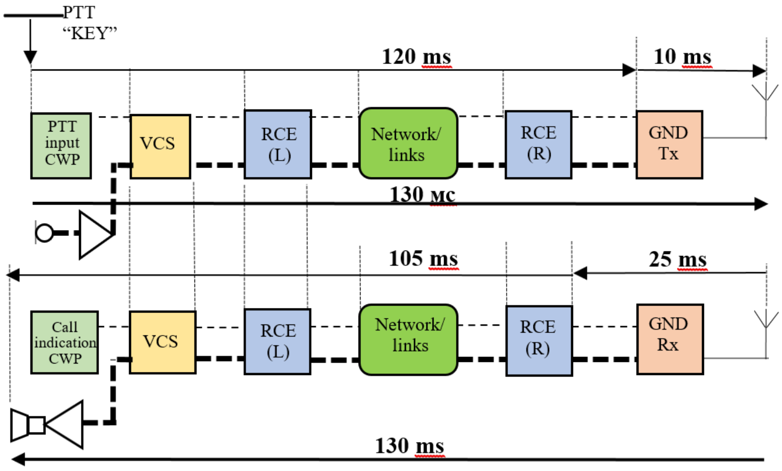

Figure 3: Requirements for voice signal delay.

Additionally, jitter should not exceed 30ms to maintain high voice quality, and packet loss should be less than 1 % to ensure acceptable voice clarity. The key indicator, MOS (Mean Opinion Score), should be no less than 4.0 for high voice quality.

These performance benchmarks are crucial for supporting real-time decision-making in air traffic control operations. The system must guarantee reliable data transmission through mechanisms that prevent packet loss and compensate for network jitter, while consistently meeting the rigorous demands of modern air traffic management in terms of both security and operational reliability. The implementation of these standards ensures that VoIP-based communications in ATM systems maintain the highest levels of efficiency, security, and uninterrupted service required for critical aviation operations.

4.1. Security and protection

The Secure Real-Time Transport Protocol (SRTP) should be used to encrypt voice traffic. Authentication and access control procedures are provided by the Transport Layer Security (TLS) and IPSec protocols. It is also necessary to provide protection against Denial of Service (DoS) attacks and implement mechanisms for detecting traffic anomalies to ensure reliable system operation in the event of a connection loss or emergencies.

4.2. Interoperability requirements between IP components of the VoIP ATM system

The ATN/IPS network must support interaction with air traffic control systems of the European Union countries and comply with SESAR (Single European Sky ATM Research) standards. The key point is the support of IPv6 protocols to ensure long-term scalability of the network, detailed in [3].

4.3. Protocols and standards

The SIP should be used to establish, manage, and terminate calls. Provide support for Real-Time Transport Protocol (RTP) [13] for voice traffic transmission and compatibility with ITU-T G.711 [14], G.729 [15], and other audio codecs. Simple Network Management Protocol (SNMP) is used to monitor the status of components.

4.4. Equipment compatibility

The ability to connect ATS consoles, gateways, and signaling servers should be provided. Integration with existing ATM voice communication systems and support for routing and load balancing protocols (OSPF, BGP, MPLS).

4.5. Quality of Service (QoS)

Voice traffic packets should have a higher priority compared to other types of traffic to avoid delays during network congestion. The system should be sufficiently robust to prevent voice data loss even in the event of individual component failure. It is also essential to provide support for Climax mode, which allows you to choose the best available signal source to obtain a high-quality signal. It is advisable to use DiffServ and MPLS-TE traffic prioritization mechanisms to maintain the proper level of performance. It is necessary to guarantee a minimum bandwidth of 64 kbit/s for voice traffic per call. In addition, effective queue management and buffering are essential, which allow avoiding packet loss and ensuring the stability of voice message transmission.

4.6. Reliability and fault tolerance

Ensuring the reliable functioning of air traffic control systems involves continuous and stable data exchange between ground air traffic control centers (ATC) and aircraft [16, 17]. For this, it is necessary to implement a communication channel redundancy, as well as mechanisms for automatic switching in the event of failure (failover, redundancy), which minimizes the risk of communication loss. The system must guarantee stable operation even under high-load conditions, which requires an effective organization of resources and adaptive solutions [18, 19]. Constant monitoring of network parameters allows you to detect deviations in operation and promptly eliminate problems. The implementation of automatic recovery mechanisms ensures a swift return of the system to regular operation in the event of failures.

4.7. Testing methodology and compliance criteria

It is necessary to use laboratory tests to check delays, packet losses, and system performance, as well as simulate real operating conditions and network loads. Provide for testing for protection against attacks and communication security. There must be compatibility with ITU-T, ETSI standards, and EUROCAE recommendations. Fulfillment of requirements for QoS, security, and compatibility of components to ensure reliable operation in a real ATM environment.

5. Calculation of elementary parameters that can affect voice data quality in VoIP ATM networks

5.1. General methodology

It is necessary to maintain voice transmission quality that meets the ITU-T criteria [20] for high-quality communication. The voice quality of radio communication is determined using the voice quality assessment methodology of the nominal rating MOS. The MOS scale was formulated as a result of subjective research. In subjective testing, subjects were asked to classify the perceived voice quality according to the voice signal categories: excellent (5), good (4), satisfactory (3), poor (2), and very poor (1). To minimize the effects of voice quality degradation, the MOS value of the radio call over the terrestrial segment "to" and "from" the radio equipment MUST be equal to or greater than 4. The most common approach to quantifying speech quality (and therefore its intelligibility) in VoIP networks is to use the E-model (Figure 4) [21]. This model calculates the transmission rating factor, so-called "R value", which takes into account both the effects of delay and the impact of packet loss (along with other factors).

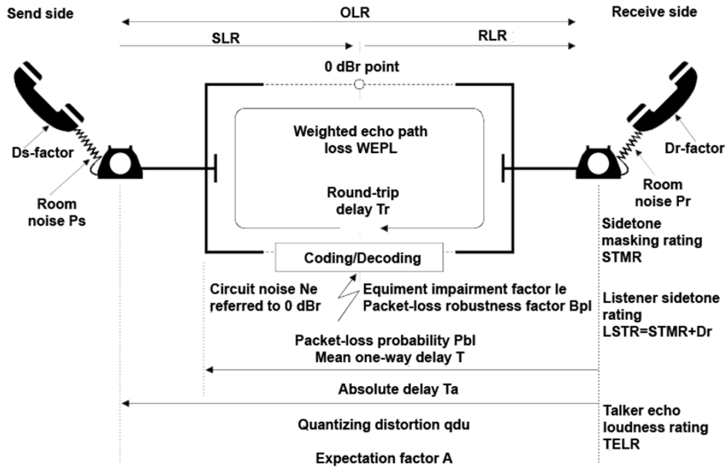

Figure 4: Reference image of the E-model.

One of the basic representations of this model has the form of a formula

$$R = R_0 - I_s - I_d - I_e + A,$$

where

- R_0 is a baseline value that takes into account noise effects and the overall signal-to-noise situation.
- I_s is the simultaneous quality degradation factor.
- I_d is the delay impairment factor.
- I_e is the equipment impairment factor.
- A is the advantage factor (e.g., mobility).

Baseline Value R_0 is a baseline value that takes into account noise effects and the overall signal-to-noise situation.

$$R_0 = 15 - 1.5(\text{SLR} + N_o),$$

where:

- SLR — Transmission loudness indicator
- N_o — Total power of different noise sources

Simultaneous Quality Degradation Factor I_s is the simultaneous quality degradation factor, determined as:

$$I_s = I_{olr} + I_{st} + I_q,$$

where:

- I_{olr} — Coefficient representing the decrease in quality due to low Overall Loudness Rating (OLR)
- I_{st} — Coefficient for degradation caused by suboptimal sidetone
- I_q — Coefficient for degradation due to quantization noise

Delay Coefficient I_d is the delay impairment coefficient, calculated as the sum of three components:

$$I_d = I_{dte} + I_{dle} + I_{dd},$$

where:

- I_{dte} — Estimates the quality degradation due to the echo heard by the speaker
- I_{dle} — Estimates the quality degradation due to the echo heard by the listener
- I_{dd} — Represents the degradation caused by excessive absolute delay T_a

In the context of the critical requirements for signal propagation delays within the ground segment of the Aeronautical Telecommunication Network (ATN), a detailed analysis of the contributing factors to voice data delay is essential.

The total transmission delay D_{total} consists of several components:

$$D_{\text{total}} = D_{\text{net}} + D_{\text{enc}} + D_{\text{sec}} + D_{\text{buf}},$$

where:

- D_{net} — Network delay (includes routing time and inter-node transmission)
- D_{enc} — Encoding delay (time required to convert an analog signal to digital)
- D_{sec} — Delay due to encryption and decryption
- D_{buf} — Buffering delay (used to smooth out jitter, but may cause synchronization issues)

The buffering delay is further defined as:

$$D_{\text{buf}} = k \cdot J,$$

where:

- J — Jitter (variation in packet arrival time)
- k — Buffering coefficient (depends on jitter compensation algorithm)

Typically, if jitter $J \geq 30$ ms, buffering delay D_{buf} starts to noticeably degrade speech quality.

5.2. Example: Delay calculation and codec selection

In ATN, the maximum delay of voice packet transmission should not exceed 130 ms in two critical sections: "before the radio transmitter" (ground network delay) and "after the radio receiver" (ground network delay after receiving a packet from the air segment).

To minimize the impact on this parameter, we analyze the permissible audio codecs [14, 15, 22] based on the recommendations in [7]. A summary of their key characteristics is provided in Table 1.

Table 1
Comparative Characteristics of Audio Codecs

Codec	Coding Method	Bitrate (kbps)	MOS (Quality)	Delay (ms)
G.711 (A-law)	PCM	64	≈ 4.1	$\sim$0.125–1
G.728 (no PLC)	LD-CELP	16	≈ 3.8–4.0	$\sim$3–5
G.728 (LD-ELP)	LD-CELP	16	≈ 3.8–4.0	$\sim$3–5
G.729	CS-ACELP	8	≈ 3.92	$\sim$10–15

We select **G.711 (A-law)** as it provides the highest Mean Opinion Score (MOS) and the lowest delay among listed codecs. The total network delay D_{net} includes:

- Packet transit time through routers,
- Processing time at switching devices,
- Transmission time across terrestrial links.

According to recent studies [23], taking into account full-scale intrusion and route adjustments, typical delays for the regional IP network of Ukraine are in the range of 15–20 ms, while delays for the European domain are typically 40–50 ms.

To ensure robustness and include a margin for error, we assume a total network delay of:

$$D_{\text{net}} = 70 \text{ ms}.$$

Encryption delay. The use of VPN or IPSec introduces an additional delay of approximately 10–15 ms. For the purposes of this analysis, we select:

$$D_{\text{sec}} = 12 \text{ ms}.$$

Buffering delay. Buffering is used to compensate for jitter but contributes to overall transmission delay. For VoIP applications, a recommended buffering delay is:

$$D_{\text{buf}} = 20 \text{ ms}.$$

Total delay and compliance check. Assuming an encoding delay of:

$$D_{\text{enc}} = 1 \text{ ms}.$$

the total estimated one-way delay is:

$$D_{\text{total}} = D_{\text{net}} + D_{\text{enc}} + D_{\text{sec}} + D_{\text{buf}} = 70 + 1 + 12 + 20 = 103 \text{ ms}.$$

The obtained total delay of 103 ms is less than the 130 ms threshold defined by EUROCAE requirements. Therefore, the selected parameters meet the performance criteria for ATN (Aeronautical Telecommunication Network) communications.

5.3. Equipment quality influence coefficient

It primarily reflects the impact of packet loss due to reduced codec bitrate. This value is based on results from subjective quality assessments (Mean Opinion Score, MOS) and empirical observations in network environments.

Historically, the Ie values (equipment impairment factors) were determined using tabulated data that related codec performance to packet loss. Today, this parameter is often described analytically as a function that considers the baseline impedance of the codec and the effective packet loss rate, as follows:

$$Ie = Ie_0 + (95 - Ie_0) \cdot P_{\text{eff}},$$

where:

- Ie_0 is the baseline equipment impairment factor for the codec under ideal (lossless) conditions.
- P_{eff} is the effective percentage of packet loss, expressed as a value between 0 and 1.

The resulting R value can be converted into a MOS (Mean Opinion Score) scale or directly interpreted to assess speech intelligibility.

This model enables the assessment of voice communication quality, taking into account the primary network parameters, such as latency and packet loss, which directly impact speech intelligibility.

5.4. Relationship between information transmission and e-model parameters

The relationship between the amount of transmitted information and the parameters of the E-model can be described using Information Theory. The formula for the information transmission rate is determined through entropy [24]:

$$I_{\text{inf}} = \nu \cdot H(\lambda),$$

where:

- ν is the phoneme pronunciation rate (in phonemes per second),
- $H(\lambda)$ is the entropy of speech (in bits per phoneme).

To estimate the amount of information contained in a single phoneme, Shannon entropy is applied:

$$H(\lambda) = \log_2(m),$$

where m is the number of distinct phonemes in the language.

For English, which contains approximately 44 phonemes:

$$H(\lambda) = \log_2(44) \approx 5.46 \text{ bits/phoneme}.$$

However, due to the natural correlation between phonemes (resulting from syntactic and semantic constraints), the actual entropy is lower than the theoretical maximum. For English, which is the official language of ICAO communications, this effective entropy is typically estimated in the range of:

$$H_{\text{eff}}(\lambda) \approx 3.5\text{–}4.0 \text{ bits/phoneme}.$$

5.5. Impact of information loss on the R-factor

The amount of information lost due to packet loss, noise, and signal distortion contributes to a reduction in the overall R-factor. This information loss can be expressed as:

$$I_{\text{lost}} = I_{\text{inf}} \cdot \text{PLR},$$

where:

- I_{inf} is the total information rate (bits per second),
- PLR is the packet loss rate (a value from 0 to 1).

The impairment factor due to codec losses, Ie, can be modeled using the following expression:

$$Ie = k_1 \cdot \text{PLR} + k_2 \cdot \left(1 - \frac{I_{\text{out}}}{I_{\text{inf}}}\right),$$

where:

- k_1, k_2 are codec-dependent coefficients reflecting sensitivity to packet loss and compression artifacts, respectively,
- I_{out} is the amount of information preserved and successfully transmitted after codec processing.

Thus, a smaller output information value I_{out} leads to a higher impairment factor Ie, which in turn reduces both the R-factor and the MOS (Mean Opinion Score). This highlights the importance of codec efficiency and network reliability in preserving speech quality.

5.6. Analysis of the impact of bandwidth and noise

The formula for the amount of information at the output of the codec [24]:

$$I_{\text{out}} = 2kF_{\text{max}}\Delta tN \log_2 \left(1 + \frac{\sigma_s^2}{\sigma_n^2}\right).$$

It shows:

- If $\frac{\sigma_s^2}{\sigma_n^2}$ is low (high noise level), then I_{out} drops $\rightarrow R$ decreases.
- If k (quantization bit depth) is low, then information loss increases $\rightarrow$ MOS decreases.

This is consistent with the E-model, where I_s accounts for noise, and I_e accounts for compression losses.

The main conclusion: The more information lost in the codec, the higher I_e and the lower MOS.

5.7. The impact of packet loss on MOS

Packet loss without recovery mechanisms can significantly affect voice intelligibility. The ratio of loss to quality can be expressed in terms of a special variable α, which depends on the codec:

$$MOS_{\text{loss}} = MOS_{\text{max}} - \alpha \cdot P_{\text{loss}},$$

where:

- MOS_{max} – the maximum possible MOS level for the selected codec (e.g., 4.1 for G.711 or 3.9 for G.729),
- P_{loss} – packet loss rate in percent,
- α – sensitivity coefficient to packet loss (for G.711: 0.1, for G.729: 0.3).

For example, if $P_{\text{loss}} = 5\%$ and the G.711 codec is used:

$$MOS_{\text{loss}} = 4.5 - 0.1 \times 5 = 4.0.$$

That is, the voice quality is reduced due to the packet loss.

5.8. Impact of equipment availability factor

The equipment availability factor takes into account its reliability:

$$MOS_{\text{final}} = MOS_{\text{loss}} \times R_{\text{eq}},$$

where R_{eq} is the equipment availability coefficient (from 0 to 1, where 1 is full availability).

If, for example, $MOS_{\text{loss}} = 4.0$, and equipment reliability $R_{\text{eq}} = 0.98$, then:

$$MOS_{\text{final}} = 4.0 \times 0.98 = 3.92.$$

The lower the R_{eq}, the more speech intelligibility decreases.

The final MOS formula, taking into account all parameters, looks like this:

$$MOS = (MOS_{\text{max}} - \alpha \cdot P_{\text{loss}}) \times R_{\text{eq}}.$$

6. Load forecast for the Ukrainian ATN

The basis for forecasting the load on the ATN/IPS network in Ukraine can be the mathematical model of passenger traffic distribution on domestic airlines in Ukraine, which is proposed and considered in detail in [25]. Since aviation telecommunications (voice and digital data exchange) serves passenger traffic, the forecast of the load on it directly depends on passenger movements.

In the current forecasting model, taking into account possible military-political situations, temporary ceasefire with the presence of temporarily occupied territories, airports in the following cities are considered: Dnipro (D), Kyiv (Kyiv + Boryspil) (K), Lviv (L), Odesa (O), Kharkiv (Kh) and potentially Zhytomyr (Zh) and Uzhgorod (U).

The model represents the transport system as a discrete Markov system, where city airports are considered as possible states of the system, and flights between them are described as transitions between these states. We assume that the transition from one state to another (a flight between some pairs of airports) occurs with the corresponding time dependence of the local transition probability $P(t)_{ij}$, in the form of a system with a discrete number of states $N = 7$, transitions between which are possible at discrete moments (for example, 1, 2, ..., 365). The indices i and j represent the departure airport and the arrival airport, respectively.

Model assumptions:

1. If a flight is made between certain airports, this means the transition of the system from state i to state j.
2. The probability of such a transition at a discrete time is denoted as $P(t)_{ij}$,
3. We conventionally assume that this indicator depends only on the very fact of the occurrence of a transition-jump.

The indexing of system states is as follows:

$$\mathrm{D} = 1, \quad \mathrm{K} = 2, \quad \mathrm{L} = 3, \quad \mathrm{O} = 4, \quad \mathrm{Kh} = 5, \quad \mathrm{Zh} = 6, \quad \mathrm{U} = 7.$$

Since the system can be in one of N states, from which it may transition into one of the remaining $N-1$ states, then for each time point t, it is necessary to specify $N^2 - N$ transition probabilities $P(t)_{ij}$. The transition system can be represented as an $N \times N$ matrix:

$$P(t) = \begin{bmatrix} 0 & P(t)_{1,2} & \cdots & P(t)_{1,N} \\ P(t)_{2,1} & 0 & \cdots & P(t)_{2,N} \\ \vdots & \vdots & \ddots & \vdots \\ P(t)_{N,1} & P(t)_{N,2} & \cdots & 0 \end{bmatrix}.$$

This transition matrix can be represented as a discrete-state signal graph (Fig. 5), where system transitions from state i to state j are interpreted as signal transmissions from one node to another, with the transmission coefficient $P(t)_{ij}$.

The probability of a passenger transitioning from one state to another is determined using a function depending on certain parameters:

$$P(t)_{ij} = f(A_i, K_i, L_{ij}, C_{ij}, PB_{ij}, \lambda(t)),$$

where each parameter reflects the influence of various socio-economic and transport factors:

- A_i — City population: the relative share of the population of each city in the overall system is taken into account.
- K_i — Wage coefficient: defined as the ratio of the average salary in the city to the total average wage for all cities.
- L_{ij} — Distance between cities: the normalized value of the distance between cities is used.
- C_{ij} — Destination attractiveness: assessed by the level of business activity, tourist attractiveness, the presence of industrial zones, etc.

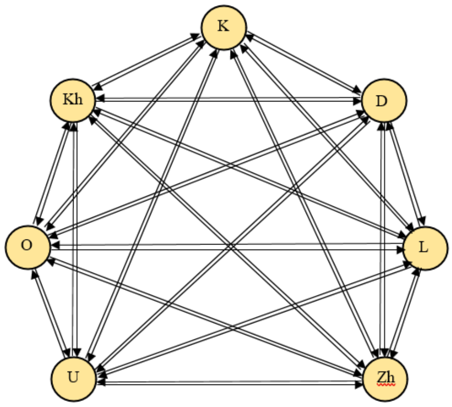

Figure 5: Graph of discrete states of the system.

- PB_{ij} — Probability of choosing transport: takes into account travel time, ticket price, availability of alternative modes of transport.
- $\lambda(t)$ — The intensity of receipt of applications reflects variable seasonality and other time factors that affect transport flows.

The main parameters for assessing the load on air communication are:

- Data traffic volume: directly proportional to the number of flights between the nodes of the graph.
- Voice traffic intensity: depends on passenger traffic and travel time.
- Seasonal changes: peak loads on the communication network coincide with peak traffic periods.
- Capacity forecasting: the same transition probability matrix is used, but with special weighting factors for communication.

The proposed model allows for effective forecasting of passenger traffic distribution and load on the aviation telecommunications network. It takes into account complex factors that affect route and vehicle choices. Additionally, it enables the estimation of peak loads to optimize aviation infrastructure and communication at both a specific airport and the network as a whole.

7. Conclusions

The analysis of ICAO and EUROCAE guidance documents systematizes the fundamental requirements for VoIP systems in the aviation control field. Their compliance ensures high quality of voice transmission, system reliability, and communication security. These requirements are mandatory for implementing modern solutions in ATM systems and contribute to increasing the efficiency of air navigation services.

The implementation of ICAO and EUROCAE requirements in the ATN/IPS network is a strategically important step in modernizing Ukraine's aviation infrastructure following the end of the war. The introduction of modern data transmission technologies will allow:

- To ensure high-quality communication between dispatching services, airports, and aircraft.
- To increase the level of cybersecurity of aviation infrastructure.
- To integrate the Ukrainian ATN with the European air traffic control system.

Thus, the gradual transition to ATN/IPS will enable Ukraine to restore its aviation sector, meet international standards, and ensure the safe operation of airspace after its reopening.

Declaration on Generative AI

The authors have not employed any Generative AI tools.

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