Possibilities of Migration to the G/G VoIP Network for Voice Communication in the Air Traffic Management

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Abstract:

Ground / Ground coordination of the participants in the Air Traffic Management, in addition to the increasing application of information technologies, continues to be inconceivable without the Voice application. According to Eurocontrol development strategies, the voice application for coordination will be include in a future plans, with the necessary transfer to the IP network.

Within Eurocontrol a majority of the participants currently uses outdated analog network to connect VCSs using ATS-MFC-R2. Eurocontrol's plans have predicted abandoning the MFC-R2 and the migration to digital ATS-QSIG network. Such migration requires a significant investment in infrastructure, and most often, the replacement of the existing VCS systems, which slows down the whole process significantly.

This paper considers the possibility of IP technology for interconnection VCSs in order to overcome the problems of transition to the ATS-QSIG network. The necessity of shifting is imminent because telecommunication operators announce the abandonment of their hitherto services as well as the necessity of migration to IP infrastructure.

This paper takes into consideration minimum technical-technological requirements for the implementation of VoIP ATM. The analysis of the capacity of links between VCSs will be carried out, which is based on the required technical and technological specifications of voice services.

Keywords: GG voice communications, VoIP, link capacity analysis

1 Introduction

Voice Communications Ground/Ground in Air Traffic Management currently uses analogue networks for voice transmission and digital TDM (Time Division Multiple Access) systems for voice routing and management of VCS (Voice Communication Switching System) [1], [2] which are non-blocking systems based on the channel time allocation method along with the application of PCM encoding techniques and voice transmission.

In recent years there has been an increasing demand for shifting to new technologies, especially after the announcement of the national telecom service provider on the abandonment of analog transport networks and migrating to IP transport networks. For the aeronautical community this announcement represents a problem, given the fact that the majority of ANSP is currently associated with analog leased lines or MFC ATS R-2 networks [3], [6]. Eurocontrol has planned to switch to ATS QSIG in 2008, but at this point that would be only a temporary solution since the migration to an IP network infrastructure is inevitable.

From the technical point of view, the transition to ATS QSIG also presents a problem for national providers of telecommunication services as they set very strict requirements for the synchronization of digital links needed to connect VCSs at national and international level. For most ANSP migration to ATS QSIG network can be difficult due to interoperability problems with other ANSP at international level, and efforts are being made to accelerate the transition to IP networks for voice transmission.

In some ATM segments the VoIP technology – the voice and data transmission through a closed IP network – can be used, which is the ultimate goal of network infrastructure development in the ATM, but for such way of connecting VCSs the standards are yet to be defined as well as guidelines both those by ICAO and Eurocontrol and the Telecom Regulator. After defining the guidelines and standards, a considerable support for VCS equipment manufacturers is expected in order to meet the strict requirements of ANSP operational services.

2 Development of ATM Voice system according to SESAR program

In order to simplify the entire process of preparing regulations, standards' development, system design, VCS's production and finally the migration, Eurocontrol has initiated a series of action plans necessary for the introduction of ATM VoIP system. The most significant projects are implemented through SESAR.

Considering the fact that the current air traffic is in a large expansion, the European Union set the problem of compatibility and interoperability as a priority and launched the SESAR project [4], [5] which should solve the problems of restructuring the air space, improvement and consolidation of services in air traffic and modernizing the ATM infrastructure. One of the biggest problems is the current fragmentation of European sky, discrepancy in operational procedures, and technical application of different technology solutions that often do not allow interoperability.

The primary task of SESAR program is to create conditions for the introduction of new modern ATM until 2020 in three phases:

- Development of Master Plan from 2005 to 2008, which will define the technological steps and priorities for modernization and implementation of a new ATM concept.
- Phase of development from 2008 to 2013, which envisages the development of new equipment, standards and regulations as well as mechanisms for the unified European sky together with the replacement of existing systems and enabling interoperability with systems outside the EU
- Phase of implementation from 1014 to 2020 which envisages large-scale production and equipment supply, implementation of the new ATM services, construction of infrastructure and installation of adequate equipment in the aircrafts as well.

In accordance with plans of Eurocontrol, SESAR and various strategies, analog networks for voice transmission would be abandoned by 2010. Since 2010 the digital network (ATS QSIG) would be used, which is based on the rented links with capacity of 64 kb/s. The start of operational application of VoIP ATM Ground/Ground voice communication is planned for 2013, while the start of operational application of VoIP ATM Air/Ground voice communication is scheduled for 2020.

3 Migration of G/G communication to VoIP network

In accordance with the SESAR's objectives certain working groups have been defined which deal with solving a series of problems related to the implementation of G/G VoIP communication. To resolve the problem of implementation of VoIP network in the ATM, it is necessary to perform some of the preparatory actions:

- Conduct the analysis of the current state of G /G ATM communications
- Define the technical operational requirements for G/G ATM voice communication with the application of already existing standards as much as possible
- Define the way of dealing with security problems in the IP network, define the QoS(Quality Of Service),SLA(Service level Agreement) etc.
- Make recommendations or guidelines for the design of new VoIP ATM networks and communication systems, after the analysis of technical operational requirements.
- Develop interoperability between ATM, VoIP systems and existing systems for voice transmission.

In solving the problem of migration to VoIP ATM network many subjects are included, such as: regulators, equipment manufacturers, airlines, aircraft manufacturers, providers of telecommunication services, etc. In order to ensure all conditions for the realization of VoIP ATM network and to meet operational requirements such as connecting the radio equipment to the VCS, connecting the system for speech recording, linking VCSs in the G/G Voice ATM network, etc., it is necessary to define all the interfaces and protocols for implementation of all applications.

4 The possibility of migration to the IP network using the VCS IP Gateway

In the initial stage of migration, most ANSP will rely on the VCS IP Gateway for connecting the existing digital ATS SQIG interfaces of VCSs on the IP network. Application of VCS IP Gateway for the analog MFC R2 network is not recommended because of delay sensitivity analog signalization that occurs during the transmission of MFC signal tones through an IP network.

VCS IP Gateway will enable linking a variety of analog and digital interfaces such as: CB, LB, E&M, PSTN, MFC, ATS QSIG, and E1 to IP interface. One of the methods that can be applied is the ATS QSIG traffic tunneling through the IP network using E1 multiplexer whether installed inside or outside the VCS and connected to an IP router. (Fig. 1)

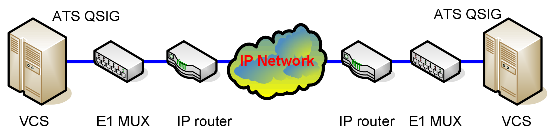

Figure 1. Connecting the IP gateway to the MUX

Voice Gateway is important for the ANSP who want to keep their existing digital VCS systems and enable the transition to IP network interfaces in order to reduce the costs of leased links connecting the VCSs (Fig. 2)

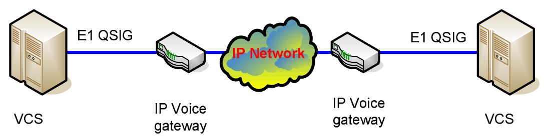

Figure 2. Connecting the IP Gateway to a VCS

The VCS IP Gateway should be applied in voice transmission from VCS's network interfaces to the IP router and over the IP network. VCS to CWP voice transmission as well as switching would continue in the very same VCS and for now it remains to be based on the TDM principle, thus allowing a full availability and the non-blocking principle of system's functioning.

5 Characteristics of voice communication in ATM

This section will present some considerations for implementing VoIP technology in G/G voice network as well as characteristics of voice communication in ATM.

Since IP was initially designed for data, mechanisms have been implemented to provide real-time, low-latency and error correction demands for voice. These mechanisms include:

- Echo cancellation
- Packet prioritization-prioritizes voice packets over other traffic
- Forward Error Correction
- Low Delay CODEC
- Bandwidth allocation and queuing
- Network delay and jitter buffering.

As voice services in G/G voice network have stricter requirements regarding call set-up time, blocking probability and voice latency than voice services in public network, it is essential to get into account those requirements for the network design as shown in [1] and [2].

Analysis of traffic for capacity dimensioning of IP based G/G voice network is based on the research carried out in paper [6] regarding the number of offered calls per hour and call duration. Using the eq. 1:

$$A_{jk} = N_{zr} \cdot \sum_{i=1}^{n} \frac{n_{zr_i} \cdot T_{s_i}}{3600} \qquad (1)$$

where:

A_{jk}-	traffic between nodes *j* and *k*
N_{zr}-	number of aircraft between nodes *j* and *k*
n_{zr_i}-	number of calls from the working position *i* per aircraft
T_{s_i}-	average call duration characteristic for working position *i*
n	number of working positions

one can obtain the traffic matrix as presented in Table 1, for the forecast number of aircraft between single nodes during busy hour.

The value of traffic presented in Table 1 concerning the network shown in Fig. 3. As can be seen in Fig. 3, the network for which transmission capacities analysis is to be carried out in this paper consists of five nodes (VCS). Each node can contain several working positions that can be ACC (*Area Control Centre*) sectors, TWA (*Tower Control*) or APP (*Approach Control*) working positions. This presents an example of the network topology where all VCSs do not have to be in direct connection with everyone. Those nodes (VCSs) arrangement is imposed by airport and sector arrangement, although number and size of sectors can vary according to air traffic density in a sector.

Table 1. Traffic matrix

		Traffic towards node ' k' [Erl]				
		VCS1	VCS2	VCS3	VCS4	**VCS5**
Traffic from node'j' [Erl]	VCS1	A_{11}=0	A_{12}=0.1	A_{13}=0.2	A_{14}=0.05	**A_{15}=0.06**
	VCS 2	A_{21}=0.1	A_{22}=0	A_{23}=0.15	A_{24}=0.15	**A_{25}=0.05**
	VCS 3	A_{31}=0.14	A_{32}=0.34	A_{33}=0	A_{34}=0.14	**A_{35}=0.1**
	VCS 4	A_{41}=0.2	A_{42}=0.1	A_{43}=0.1	A_{44}=0	**A_{45}=0.08**
	VCS 5	**A_{51}=0.15**	**A_{52}=0.08**	**A_{53}=0.08**	**A_{54}=0.06**	**A_{55}=0**

Source: [3]

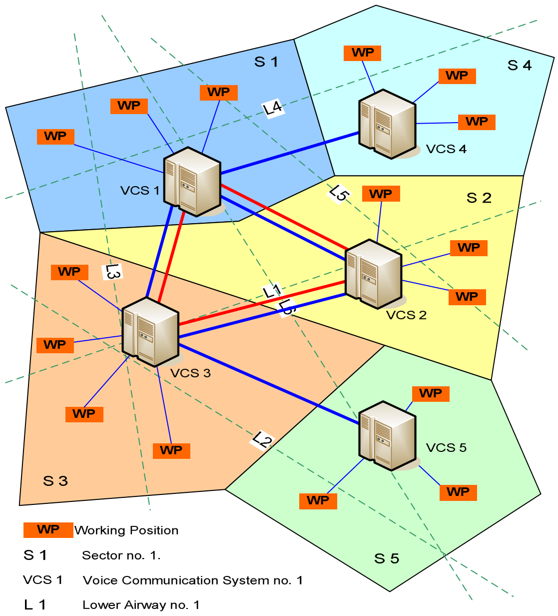

Figure 3. Arrangements sectors and VCS nodes in the analysed network

6 Bandwidth requirements for voice transmission over IP based network

6.1 Impact factor for bandwidth calculation

There are many factors involved when calculating the bandwidth required through a network. This section of paper aims to explain these factors, and to offer a simple means of making such calculations.

Detailed consideration of each coding method is beyond the scope of this paper, but it should be understood that the various coding methods vary in the levels of complexity, delay characteristics and quality. The CODEC which is used for bandwidth calculation in this paper is G.711 (PCM) and G.728 [7], (CODEC is used in ATS QSIG).

There are many ways to reduce the bandwidth requirements, and these can be particularly important in the specific network like Air Traffic Service Ground Voice Network (AGVN). These include silence suppression, RTP header compression and RTP multiplexing.

In common with many communications systems, the protocols involved in Voice over IP (VoIP) follow a layered hierarchy which can be compared with the theoretical model developed by the International Standards Organization (*OSI seven layer model*).

This paper does not discuss header compression schemes and does not include them in calculation of bandwidth requirements. Furthermore, this paper only considers IPv4 and does not discuss layer 2 protocols which increase overall bandwidth requirements, depending on type of protocol.

The selection of payload duration is a compromise between bandwidth requirements and quality. Smaller payloads demand higher bandwidth per channel band, because the header length remains at forty octets. However, if payloads are increased, the overall delay of the system will increase, and the system will be more susceptible to the loss of individual packets by the network.

It is known that there are no strict recommendations concerning packet duration. Although codecs vary in their quality and delay characteristics and there is not yet an agreed standard for ATM VoIP. The Table 2 shows bandwidth requirement depending on packet duration for G.711 (PCM coding) and G.728 (LD-CELP) which is used for bandwidth calculation in this paper.

There is no absolute answer to this question, but for the purpose of this paper, it will be assumed that voice samples representing 30ms and 20 ms are sent in each packet, respectively.

Table 2. Bandwidth requirement for G.711 and G.728 at different packet duration

Codec	Packet duration	Bandwidth [kbps]
G.728 (LD-CELP) 16kbps compression	10 milliseconds (16 samples)	48
G.728 (LD-CELP) 16kbps compression	20 milliseconds (32 samples)	32
G.728 (LD-CELP) 16kbps compression	30 milliseconds (48 samples)	27
G.728 (LD-CELP) 16kbps compression	40 milliseconds (64 samples)	24
G.728 (LD-CELP) 16kbps compression	80 milliseconds (128 samples)	20
G.711 (PCM) 64kbps uncompressed	10 milliseconds (16 samples)	96
G.711 (PCM) 64kbps uncompressed	20 milliseconds (32 samples)	80
G.711 (PCM) 64kbps uncompressed	30 milliseconds (48 samples)	75
G.711 (PCM) 64kbps uncompressed	40 milliseconds (64 samples)	72
G.711 (PCM) 64kbps uncompressed	80 milliseconds (128 samples)	68

Table 3. Bandwidth calculations for different type of links

Link between two adjacent nodes	Expected traffic in *[Erl]* on the link with x_k= 0,999	Number of ATS QSIG channels for expected traffic and permitted blocking probability p_b = 0,001	Link capacity requirements for the transmission voice over ATS QSIG based network	Bandwidth requirements for the transmission of voice over an IP based network (CODEC G.711: RTP/UDP/IP, RTCP, packet duration 20ms)	Bandwidth requirements for the transmission of voice over an IP based network (CODEC G.728: RTP/UDP/IP, RTCP, packet duration 30ms)
1	2	3	4	5	6
1-2	0,330997	4 x 16k	2 x 64 **[kbps]**	4 (336,8 **[kbps]**)	4 (112,2 **[kbps]**)
1-3	0,929339	6 x 16k	2 x 64 **[kbps]**	6 (505,2 **[kbps]**)	6 (168,4**[kbps]**)
1-4	0,75896	5 x 64k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
2-3	0,670267	5 x 64k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
3-5	0,70901	5 x 64k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
Total		***25x16k (400*kbps*)***	***640* kbps**	***2105* kbps**	***561,2* kbps**

6.2 Comparative analysis of the bandwidth requirements for the transmission of voice

The basic goal of this paper is to determine bandwidth requirements for the transmission of voice over an IP based network between individual nodes whose arrangement depends on the VCS location (Fig. 1).

For determining link capacity between nodes in [6], alternative routing scheme was assumed and fully described.

In addition, it was assumed that the routing is done in a way that is known in literature as the call-by-call. This assumption is also retained in this paper.

Respecting all the introduced restrictions regarding the route length, avoiding of closed loops and tromboning, and knowing the traffic requirements between individual nodes, by applying the expression for determining probability of using the defined routes for every origin-destination pair, the values of the expected traffic on a link have been obtained.

The obtained values are presented in Table 3, and they have been achieved by summing up traffic that is expected to be on that link of direct routes and of all the alternative routes in which this link is included [6].

For determining the capacities between the nodes the Erlang B-formula is used as well as all its assumptions defined in [8]. The obtained values are presented in Table 3 for Grades of Service (GoS) expressed by the blocking probability as ICAO recommend ($p_b \leq 0{,}001$).

The results of bandwidth calculation for the transmission of voice over an IP based network have also been presented in table 2 for the same number of voice channel which is planned for circuit switch network considered in [6] and shown in Fig. 1. The values in columns 5 and 6 is obtained respecting all previously introduced in section 6.1 and for two type of codecs: G.711 (column 5) and G.728 (column 6)

Furthermore, obtained values are presented in Table 3 (column 3 and 4) have been achieved without the impact factors regarding layer 2 protocols.

7 Technical characteristics of the VoIP implementation

The data in Table 3 show that in all cases a part of the bandwidth remains unused with respect to calculated capacities. Implementation of ATS QSIG link requires the G.703 physical interfaces that allow data transmission speed of 64 kbps. Such a physical link allows a maximum of three voice transmission channels and one common signaling channel. Better use of 64k links can be achieved by introducing the definition of additional routes in the VCS, which also increases the network availability. During the realization of an ATS QSIG network, we recommend using links rented from two service providers [3], [5] which consequently has impact on increasing the reliability and availability throughout the network.

When implementing a VoIP ATM network Fig. 4, it is also recommended to use two service providers and at least two Gateway interfaces towards the VCS. According to data presented in Table 3 for the implementation of VoIP network, only a part of leased capacity would be used, which depends on the technical possibilities of the IP network realization by the service provider. Unused capacity of IP links can also be utilized to define additional routes.

Considering the service providers' announcements and Eurocontrol abandoning the development of standards for ATS QSIG at the E1 level, the majority of ANSP in the transition period plan to install their own primary E1 multiplexer in order to ensure the operational work until the final transition to VoIP on the existing VCS systems, independently of the cancellation of the service by the telecom service provider.

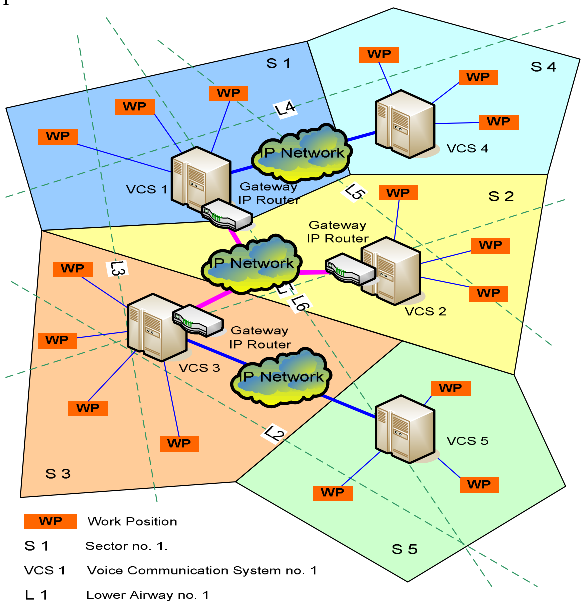

Figure 4. Arrangements sectors and VCS nodes in the IP network

Connection via E1 and primary multiplexers ensures better capacity usage of leased links because the primary multiplexers can be used for other applications and ANSP services, such as transmission of radar data, transfer of small-capacity Ethernet Traffic, transfer of flight plans, transfer of supervision and control of systems, video surveillance, voice transmission to the radio sites and between sites by analog interfaces, [3], [6], [9] etc. By analyzing all the available solutions this way of connecting VCSs significantly reduces the total cost of renting.

Eurocontrol and SESAR in their strategies [4] , [5] foresee the beginning of the operational application of ATM VoIP in G/G voice communication during 2013, for A/G voice communication the operational implementation is expected during 2020. Given the predicted dynamics, it is still necessary for a part of ANSP services to remain on the TDM network of primary multiplexers until 2020.

For the migration to VoIP, it is necessary to establish a series of standards and guidelines for introducing the VoIP in ATM. Upon completing the testing of the system, including the operational work test, all the ICAO documentation will be published. Currently, EUROCAE [10] is preparing the issuing of the following documents:

- ED136: Voice over Internet Protocol (VoIP) Air Traffic Management (ATM)System Operational and Technical Requirements
- ED137: Interoperability Standards for VoIP ATM Components
- ED138: Network Requirements and Performances for VoIP ATM Systems (Network Specification and Network Design Guideline)
- ED139: Qualification tests for VoIP ATM Components and System

Publication of these documents will enable the faster preparation of all participants in the construction of network infrastructure and will solve interoperability problems at different application levels.

8 Conclusion

This paper considers currently available technical solutions of the service provider, necessary capacity of the ATM Voice network, links capacities for the application of different technical solutions for voice transmission in the ATM. According to current pace of activity, it is to be expected that the migration to the VoIP network is going to be completed in the following decade, meeting the operational requirements, QoS and SLA.

Finally, the migration to VoIP can significantly reduce the cost of leased capacity, comparing the current analogue network for voice transmission. The process of migration itself will be supported by issuing new standards and guidelines for the design and implementation of VoIP networks.

Telecom service providers' announcements about cancellation some analog services and the necessity of transition to the IP network infrastructure raises the problem for ANSP because they are not able to do the migration to IP infrastructure in a short period of time, since the whole process is still in its preparatory phase.

In order to ensure the smooth service without compromising safety of air traffic until the beginning of the operational application of VoIP, it is necessary to maintain functioning of all systems with the existing interfaces and standards, regardless of changes in the telecom service provider infrastructure, and allow the system preparation for the introduction of VoIP ATM network. One of the possibilities for the ANSP is a temporary migration of service to their own primary multiplexers and a gradual migration to VoIP infrastructure. This kind of migration also significantly reduces the cost of lased lines from the telecom service provider and creates prerequisites for the deployment of VoIP service with the retention of full functionality of G/G and A/G communication system.

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