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Link Capacity Dimensioning Model of ATS Ground Voice Network

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1. Introduction

Apart from a number of new technologies which are currently implemented in air traffic control for the exchange of messages the ground/ground (G/G) voice communication is still very significant and is currently irreplaceable. The problem regarding introduction of new technologies that would replace voice communication lies in insufficient development and linking of very expensive ATM (*Air Traffic Management*) systems that have been already in implementation for a number of years. The best indicator showing this is the implementation of MFC (*Multi Frequency Coding*) standard which is still being implemented in the majority of Eurocontrol member countries.

The introduction of advanced automatic message exchange system in ATM will result in the reduction of voice communication in coordination. It will not be possible, however, to perform the implementation of the new systems that will enable exchange of data essential for the coordination, integrally for all the ATM users, so that voice communication will continue to be implemented either as a basic service or as a *backup* service.

In order to realize the planning and dimensioning of the telecommunication network for G/G voice communication which is to be the basic task of this paper, it is necessary to have all the data on the relevant network parameters that may affect the very operation process in air traffic control system. The most important parameter used for this purpose is the telecommunication traffic which is generated among the network nodes and the intensity of which has to be adequately forecast. It will obviously depend on the number of flights between two network nodes during the peak hour and the type of flight which will affect the duration of communication between two working positions in G/G Voice network.

The problem that always occurs is due to the telecommunication network which is designed on the basis of forecast values of several different parameters. The forecast errors are always present in the network design process. They occur either through overestimation or underestimation of the future traffic requirements in the network. In order to correct these errors, eliminate them, or at least alleviate them, the routing design procedure and introduction of dynamic routing in the network are used.

The advances in the technology of modern telecommunication systems have brought to significant interest in the development of schemes that can dynamically manage the calls in the network. The purpose of developing such dynamic routing schemes is the harmonization of the routing pattern in accordance with the variations of the supplied traffic that are not deterministic, in order to better use the network capacity and to enable additional flexibility and robustness that will be able to react to the errors and overloads.

Dynamic routing is nothing new, and the dynamic routing phenomenon is considered apart from the circuit switched networks also in the packet switched networks. There is a number of categories of dynamic routing in the telecommunication networks, both operative ones and many others, recommended through various criteria, such as network planning efficiency, price and complexity of implementation, performances, etc. Up to now a large number of dynamic routing methods have been proposed, and some of them have been partially implemented in the networks of some countries.

One may observe two basic approaches that have attracted significant attention. In the USA, AT&T has implemented the scheme called Dynamic Non-Hierarchical Routing (DNHR) (Ash et al., 1981), which used traffic forecasts for different periods during the day in order to pre-determine the routing patterns. In Canada, Bell-Northern Research presented a scheme called Dynamically Controlled Routing (DCR) based on the controllers receiving information on the current condition for links in the network in regular time intervals (Cameron & Hurtubise, 1986).

Apart from the basic approaches, also the approach which features the advantages for certain network formations is used and it refers to the scheme that is implemented in the British Telecom main network (Stacey & Songhurst, 1987), (Gibbens et al., 1988) and (Key & Whitehead, 1988). This scheme does not use the central controllers, but the information related to the planning of routing pattern is exchanged among the nodes. It has been primarily designed for use in the fully connected networks or nearly full connected networks and employs random search techniques in order to find the beneficial routing pattern.

Since the problems of considering the dynamic routing increase with the number of possible network structures and they vary with the set limitations, the attention will be focused here on special network structure which is not fully connected, and it is imposed by the arrangement of sectors and organization of airspace. This network structure will be used to study the dependence between the efficiency, robustness, simplicity and planning.

This chapter has been organized in the following manner. The second section analyzes the existing and expected communication needs of the G/G Voice network users in order to design the supply of services, forecast the traffic and plan the network capacities. It discusses the specific characteristics of communication needs and requirements in ATM. The technical and technological specification of the telecommunication services has been expressed in a set of specified parameters necessary for further analysis.

The third section studies the specified parameters and the set constraints within which their measures need to be realized.

The chapter presents the adapted version of the dynamic routing method for voice traffic, and therefore the ICAO recommendations for call routing i.e. traffic in the G/G Voice network are considered in the fourth section of the chapter. These have been used as the criterion to define the alternative routes in the analyzed network.

The fifth section describes the used dynamic routing scheme, using the defined conditions to derive a routing table for the analyzed network structure. The relations for determining the route usage probability for all the node pairs in the network have been derived. On their basis and based on the expected traffic among the nodes obtained by measuring of the call duration in aircraft coordination the link capacities between individual network nodes have been determined.

Since in recent years there has been an increasing demand for shifting to new technologies like as migrating to IP (Internet Protocol) transport networks (Markežić et al., 2009), the sixth section analyse bandwidth requirements for the voice transmission over an IP based network between individual nodes.

2. Specification of users' communication needs and requirements

2.1 Current and expected communication needs

Today's European Air Traffic Service Ground Voice Network (AGVN) is composed of many nodes, and each of them contains Voice Communication Systems (VCS). They represent the node entities that contain the functions for automatic switching management in order to provide the telecommunication services. The majority of networks of the Eurocontrol member countries use analogue signalization, whereas the countries with larger traffic have changed to digital signalization. Connecting of such different networks represents a problem that is currently solved by gateway within VCS (Eurocontrol (a), 2005), (Eurocontrol (b), 2005).

It often happens that the network configurations are also distinguished on international links which leads to difficulties in the interworking between VCSs of different Aeronautical Service Providers (ANSP). Sometimes these problems of interoperability can be solved by special border agreements, but this type of solution is not suitable for the realization of the concept of seamless Trans-European network for voice transmission between the ATM services.

Eurocontrol has recognized the need to prepare and define the documents that would provide guidelines for ANSPs for the configuration of their VCSs that will be included in the Trans-European ATS Ground Voice Network. The first goal of these recommendations is to ensure the advice in the configuration of a large number of parameters of Voice Communication Systems (VCS). Another goal is to define in which way such a network should be planned and implemented, in relation to the services and functions, with the aim of smooth achievement of network operations. The introduction of unique standards and network technologies sets the basics for the service quality improvement and the possibility of introducing new functions in VCSs.

2.2 Technical and technological specification of services

In order to design the supply of services, forecast the traffic and plan the network capacities it is necessary to study the existing and expected communication needs of the G/G Voice network users. Well defined users requirements need to be related with the technical and technological specifications of the telecommunication services and networks, which means that a number of parameters need to be specified out of which the following are crucial for further analysis:

- *bandwidth;*
- *call setup time;*
- *delay;*
- *blocking.*

2.3 Specific characteristics communication needs and requirements in ATM

In order to enable coordination among sectors, the ATM has to provide message exchange either by means of data exchange or by voice. In individual airspace communication links are necessary among those participants who are involved in coordination in ATM. In case of increased density of air traffic new sectors may be opened within the already existing sector or in case of a reduction in air traffic density several sectors may merge into one. This change dynamics in number of sectors also affects the G/G and A/G (air/ground) communication needs.

Regarding the organization of airspace both in the Upper Space (ACC- Area Control Centre) and in the Lower space (TMA - Terminal Manoeuvring Area, APP - Approach Control, TWR - Tower Control) there is need for several users to access the communication resources at any moment. The ATM needs for coordination, dynamics of sectors opening and organization of airspace may render the design of such systems a very complex task. The complexity is additionally increased if the category of priority is introduced into the G/G communication systems.

3. Specification of requests for G/G Voice communication affecting AGVN planning

3.1 Access method in the G/G Voice network

Since the call setup time represents an important factor in air traffic safety, the access method will be described here briefly. The required time values of call setup affect the number of nodes through which the call in the network may be set up. According to (Eurocontrol (c), 2005) the following access methods are distinguished:

- *Instantaneous Access (IA)*: This access type is most frequently used for coordination between APP and TWR services when no action by the called user to set up the connection is necessary. The call has to be set up within 1s or less in 99% of the time, (ICAO, 2002). The interval starts from initiating the call from the A side until voice link is established. According to EUROCONTROL recommendations the IA call has to be set up within 100ms.
- *Direct Access (DA)*: This call is usually used between sectors, both in case of a routed link or in case of a point to point link whose characteristics will be presented in Chapter 5 of this paper. The call has to be set up within 2s in 99% of the time, (ICAO, 2002). The interval starts from initializing the call by A side to the moment of obtaining indication of incoming call at the B side.
- *Indirect Access (IDA)*: This method is most often used for coordination with other ATM users that have not been defined in the previously mentioned access types or in accessing the public and closed private networks. The call in this case has to be set up within 15s or less in 99% of the time, (ICAO, 2002). The interval starts from initiating the call by A side to the moment of obtaining an indication of the incoming call at B side.

3.2 Bandwidth requirements for voice transmission in G/G Voice network

The analogue G/G networks use the 300 – 3400Hz bandwidth which is necessary for smooth operation of MFC signalization methods. This bandwidth has to be secured in order to provide high-quality transmission of voice and signalization throughout the network.

The implementation of digital communication systems has enabled better usage of bandwidth so that with the implementation of ATS-QSIG methods and standards for compression per single channel of 64kbps capacity, three voice channels and one signalization channel can be transmitted. The implementation of the methods of voice compression and coding affects also the increase in delay compared to the analogue transmission due to additional voice processing.

3.3 Voice delay in G/G Voice network

Voice delay understands the time necessary to perform voice transmission from end to end between a speaker and a listener. The delay occurs already in A/D (analogue/digital) conversion and depends on the applied method of voice compression. Thus, e.g. for PCM A law (G.711) compression it amounts to 0.75ms whereas for ADPCM (G.726) compression it amounts to 1ms.

Voice delay in transmission through the telecommunication network is defined according to ITU-T G.114 recommendation, which defines 150ms as acceptable end-to-end delay. Detailed analysis of delay components in G/G network is presented in (Markežić et al. 2007).

3.4 Call Blocking

Voice communication systems (VCS) are designed as non-blocking systems, which means that the communication resources have to be availability at any moment at any working position. This property also has to be transferred to the transmission network where the implementation of signalization methods, standards and recommendations for network design ensures minimal call blocking through the network and increase in the system availability. According to recommendations in (ICAO, 2002) for the dimensioning transmission links of G/G Voice network the GoS (Grade of Service) value is 0.001. GoS is defined as the probability that a call is lost during the peak hour due to the lack of transmission links (capacities). Based on this criterion the capacities of the communication links between VCSs will be dimensioned which will be presented in Section 5.

4. Traffic and technological characteristics of G/G voice communication

The previously presented requests for the specified parameters affect the call routing strategy through AGVN; therefore, this section will present the recommendations defined according to (Eurocontrol, 2006), which are used to form the call routing table (Subsection 5.2).

4.1 Recommendations and network routing strategy in G/G Voice network

The basic routing strategy is done according to the following steps (Figure 1):

1. VCS should always try to route a call to the *Direct Point-to-Point Route* or *Direct Network Route.*

2. In case the route is out of service or congested, VCS should try to route the call to another *Direct Point-to-Point Route* or *Direct Network Route* of another network operator (presented by dash line in Figure 1), if such a one is configured.

3. In case all the defined *Direct Routes* are not available, VCS should then try to route the call via *Detour Route*. If multi-*Detour Routes* have been configured in the preferred routing tables, then VCS should have an order of selecting *Detour Routes* with respect to the call establishment time.

4. In case of congestion of the *Direct Point-to-Point Route* or *Direct Network Route*, VCS should attempt to route the call to another *Direct Point-to-Point Route* or *Direct Network Route* (of a different network operator), if such a one is configured.

5. In case all the planned routes are congested, VCS should determine whether there is a call that has priority. In that case the procedures need to be followed to realize the priority call.
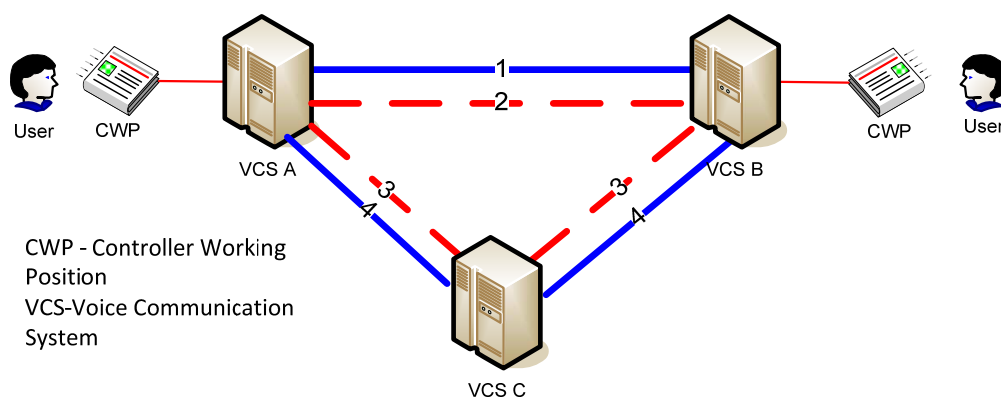

Fig. 1. Call routing strategy with two network operators

4.2 Topology and design of G/G Voice network

Figure 2 presents an example of the network topology for which transmission capacities analysis in this paper is to be carried out. The network consists of five nodes (VCS) that can be of national or international character.

Each node can contain several working positions that can be ACC sectors, TWA or APP working positions. The network design has to allow communication among all sectors at any moment regardless of the air traffic density in a sector. The network also has to be flexible in order to be able to respond to the requests for dynamic changes in the number of sectors and the size of sectors. G/G network for voice transmission has to have the possibility: call routing, priority call, call diversion as well as call waiting.

As can be seen in Figure 2 all VCSs do not have to be in direct connection with everyone, and the call setup time as well as voice delay for the analyzed network with special purpose, impose the need of defining the set of routes (direct and alternative ones) that will satisfy the required criteria. The following chapters tend to present the characteristics of single routes as defined in (Eurocontrol (c), 2005).
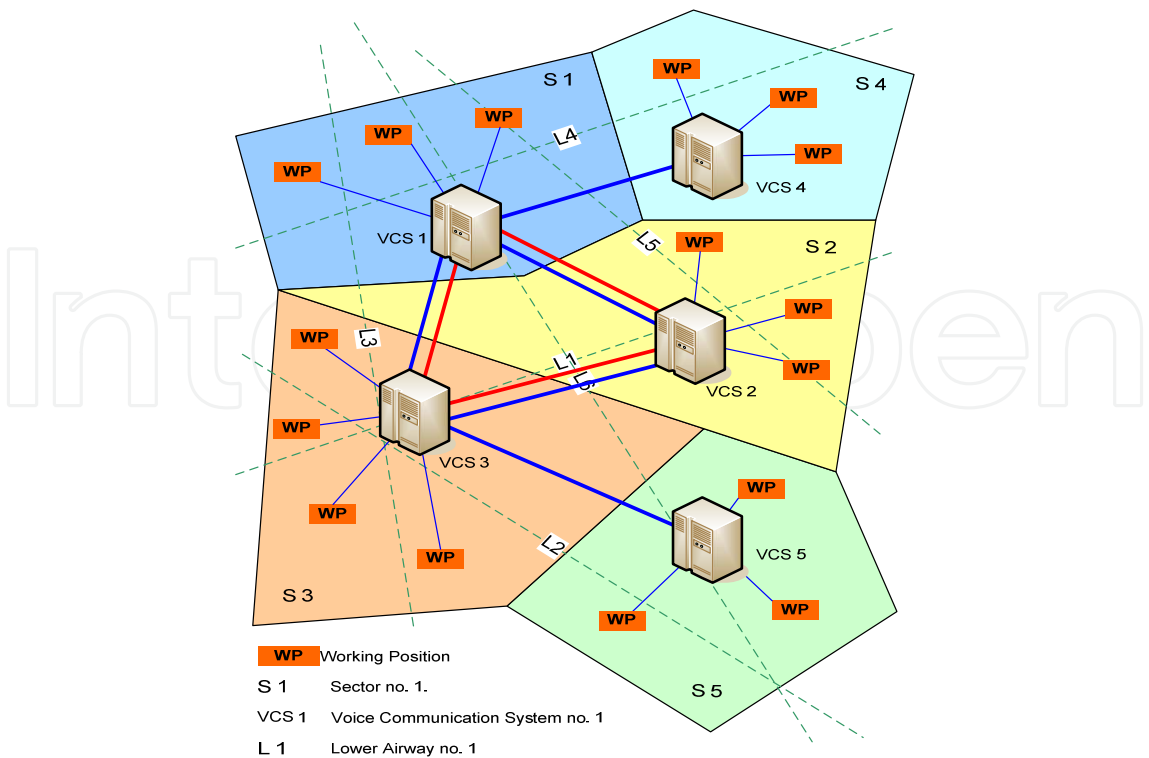
Fig. 2. Sectors and VCS nodes

4.2.1 Direct Point-to-Point Route

The most direct path from the originating to terminating VCS is called *Direct Point-to-Point Route* (Figure 3). In order to respect the time limitations for the IA calls determined in (ICAO, 2002) it is recommended that it be a physical circuit or inter-VCS link that does not pass through the transit/gateway VCS and is not switched by network.

Fig. 3. Direct Point-to-Point Route

Such route cannot consist of more than two nodes and there is no call routing. This is the most frequent communication between the positions within one node or between two adjacent nodes or even direct communication between two VCSs without routing through the network. DA access is used for this type of communication.

4.2.2 Direct Network Route

Direct Network Route can be defined as fixed and pre-established path through the network, between the originating and terminating VCS. This path can comprise of successive physical circuit or inter-VCS link passing through transit or *gateway* VCSs as presented in Figure 4. Owing to shorter call setup times achieved by the usage of digital signalization methods,

ICAO recommendations for ATS Ground-Ground Voice Switching and Signalling allow that maximally three inter-VCS links are used on the *Direct Network Route* among the ATS units (i.e. two transit VCSs) for calls with direct access, under the condition that the network is fully digital and that the criterion for call realization with direct access of 2s can be satisfied.
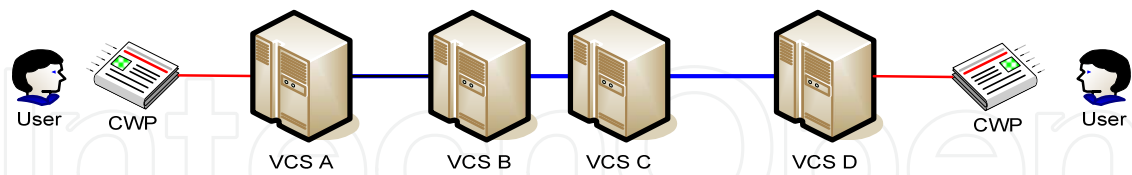

Fig. 4. *Direct Network Route* with maximum number of VCSs

4.2.3 Detour Route
The characteristics of *Detour Routes* are distinguished regarding of whether the network is an analogue or a digital one. Regarding analogue network, the *Detour Route* is an indirect physical path between the originating and terminating VCS through transit VCSs. VCS selects this path when the defined direct routes (*Point to Point or Network Route*) between two points and are not available (do to congestion or failure). The maximum number of inter-VCS links is two for DA calls in analogue networks.

Owing to the shorter call setup time realized by using the digital signalization method, a larger number of inter-VCS links is allowed. Thus, for calls with direct access up to three links on *Detour Route* between ATS units is allowed (Eurocontrol (c), 2005). An example of *Detour Route* with maximally allowed number of links is presented in Figure 5.
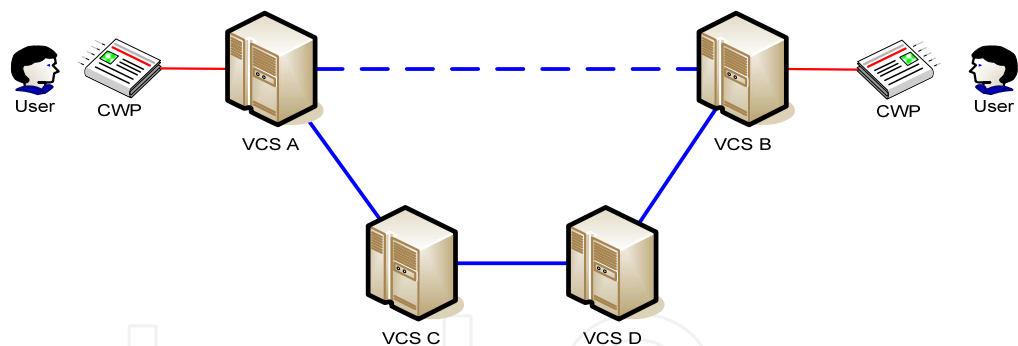

Fig. 5. *Detour Route* in digital network

In case of a stricter criterion that call setup has to be maximum 1s, planned for calls with instantaneous access, it is recommended that these calls (IA calls) are not routed via *Detour Routes*. If there is still need for this, the *Detour Route* in digital network should not contain more than two inter-VCS links.

4.3 Line diversification strategy for G/G Voice network
It is recommended that even for minimal traffic at least two leased lines are available on the inter-VCS link and leased by two network operators. ANSP should check that the network operators have carried out the line separation (i.e. that the leased lines occupy different physical paths in the network) so that a single point of failure would not cause complete

disruption in customer service provision. The recommendation is to configure the VCSs so that they separate traffic into two routes of different operators.

It is extremely important to make it possible for the routing tables for calls on detour routes to be correctly applied for every VCS within AGVN in order to avoid long delays. Badly configured routing table can lead to a closed loop in network routing and for this reason the network can become congested. It causes network degradation due to activation of all resources and a drop in service level experienced by users. With correct call routing and routing table definition it is important in fact to limit the number of transit VCSs through which the call can pass within the network.

Consequently, one may conclude that the maximal number of nodes that a call can pass is four.

5. Traffic capacity analysis of G/G voice network

5.1 Presentation of dynamic alternative routing scheme

The routes in routing tables can contain a group of direct routes and a group of alternative routes that are defined in compliance with the requirements given in Chapters 3 and 4, taking into consideration the network instability that can be caused by dynamic call routing. This understands the avoidance of a closed loop and tromboning, (Eurocontrol (c), 2005). It should be noted that the majority of papers that refer to the dynamic routing problems in the network are based on fully connected network formations and limit the set of alternative routes only to those with two links, which is not the case in this paper.

Next, in forming the routing table the nodes capabilities are respected (i.e. their intelligence). This refers to the possibility that a node can recognize which is the originating node for the call that has entered it, and that it has information on the state of all the links that come out of it (whether they are available or not). Consequently, the routing is done in a way that is known in literature as the call-by-call.

A method is analyzed, according to which, the table of alternative routes is formed with the order which has been determined according to the pre-adopted criterion, and call-by-call routing is carried out in the following way. The selection of an alternative route from the routing table is done according to the order of route in the table, i.e. sequentially. This means that always first the alternative route is selected (of course after the *Direct Network Route*), which is on the first place in the table. If the call cannot be routed along this alternative route, it is directed to the next route in the table, etc. Which means that the call will use one of the alternative routes not completely randomly, but rather conditioned by the occupancy of the previous routes defined by the routing table.

Since a single VCS has information only about links availability to the first next node, it may happen that some of the links further on the stipulated route are not available, and therefore the attempt of setting up the call is returned to the node that offered the alternative routes for a certain observed call. Thus, the analytical procedure of determining the probability that the call will use one of the supplied routes corresponds to the sub-method of alternative routing known as "originating node management with possibility to move management options to other nodes", (Sinković, 1994). In case all the defined routes are occupied after one checking, the call will be rejected, unlike the similar method described in (Kostić-Ljubisavljević et al., 2000) where the attempt will be made to set up the call on a set of pre-defined routes until a given time has passed for the call setup. The authors call this method

sequential routing (i.e. dynamic automatic alternative routing) since the selection of alternative routes follows the order determined by some in-advance adopted criterion (route length, delay, capacity).

The criterion for the definition of the set of routes and their order in selection is exclusively the route length that it is derived from the conditions presented through previous sections.

5.2 Defining the routing tables

The routing strategy can be completely described by the routing table and call management rule. For the presented network (Figure 2) the routing is described by Table 1 (Mrvelj et al., 2009). In order to describe the routing a "typical routing table" can't be used because when a call reaches a certain node, it's further routing depends on the originating node. Therefore, the routing rule will be defined by a three-dimensional field (i,j,k), where i denotes the node in which the call is currently positioned, j is the originating node, and k is the terminating node of the respective call.

The n-tuple in a certain table cell has the following meaning. If you look at the n-tuple in the table cell (1,4,3), (1st row, 4th sub-column of the 3rd column) which is (3,2), it means that the call that is in node 1 whose terminating node is 3, and which originated from node 4, will be routed in two ways according to the order of priority into node 3, and if the link towards it is occupied then to node 2.

		Node k																								
		1					2					3					4					5				
		Node j					Node j					Node j					Node j					Node j				
		1	2	3	4	5	1	2	3	4	5	1	2	3	4	5	1	2	3	4	5	1	2	3	4	5
Node i	1	x	x	x	x	x	2.3	x	2	2.3	2	3.2	3	x	3.2	x	4	4	4	x	4	3.2	3	x	3	x
	2	x	1.3	1	x	1	x	x	x	x	x	3	3.1	x	3	x	x	1.3	1	x	x	3	3.1	x	x	x
	3	x	1	1.2	x	1.2	2	x	2.1	2	2.1	x	x	x	x	x	x	1	1.2	x	1	5	5	5	5	x
	4	x	x	x	1	x	x	x	x	1	x	x	x	x	1	x	x	X	x	x	x	x	x	x	1	x
	5	x	x	x	x	3	x	x	x	x	3	x	x	x	x	3	x	X	x	x	3	x	x	x	x	x

Table 1. Routing table

It will depend on the following condition which route the call will use. If link 1-3 is free, it means that the call on this route has reached its destination. If link towards 3 was blocked, and link 1-2 available, new $i = 2$ (j and k remain unchanged), and then the table cell (2,4,3) is considered. This means that the call that has reached node 2 whose origin is 4, and terminating node is 3, will be made on this route if link 2-3 is available (n-tuple in table cell is 3). If the call is not set up on the last in the series of pre-defined routes, it will be rejected.

5.3 Traffic analysis in G/G Voice network

5.3.1 Analysis of ATM users communication time

For the dimensioning of the telecommunication network transmission capacity it is necessary, apart from the requirements previously presented, that are required from a specific telecommunication network to know also the traffic volume between individual location areas, i.e. switch nodes. In order to determine the traffic volume between the nodes measurements were carried out for the purpose of paper (Mrvelj et al., 2009), measuring the duration of calls for various working positions. The measurement results are presented in Table 2.

In the observed peak hour there were 16 aircraft in the coordination of which node 1 and 2 participate. The number of calls and their duration per working positions for that aircraft number are presented in the table (3rd and 4th column). Based on the measured values of the link occupancy times the average values of the call duration per aircraft were obtained (column 5). Current communication which is used to obtain the measured values is performed on the point-to-point principle between working positions of the same category, which facilitated obtaining of realistic picture on the link occupancy for a certain working position.

Apart from measuring traffic on the links i.e. link occupancy duration, the call duration analysis per single working position was carried out also by measuring the time of certain working procedures that refer to communication between the working positions. For the purpose of the analysis of the technological processes the UML diagrams were used, and the sequence diagram is given in Figure 6.

Working position	Aircraft number	Call number	Results obtained by measuring link occupancy			Results obtained by analysis using UML formalism		
			Duration of a single call [second]	Duration of call per aircraft [second]	Total duration of all calls [second]	Duration of a single call [second]	Duration of call per aircraft [second]	Total duration of all calls [second]
1	2	3	4	5	6	7	8	9
1		9	20	11.2	180	20	11.2	180
2	16	6	16	6	95	20	6	95
3		14	37	32.3	518	50	43.7	700
4		4	24	6	96	23	5.7	62
Total/ *average*	16	33	*24.5*	55.5	889	*28.2*	64.8	1037

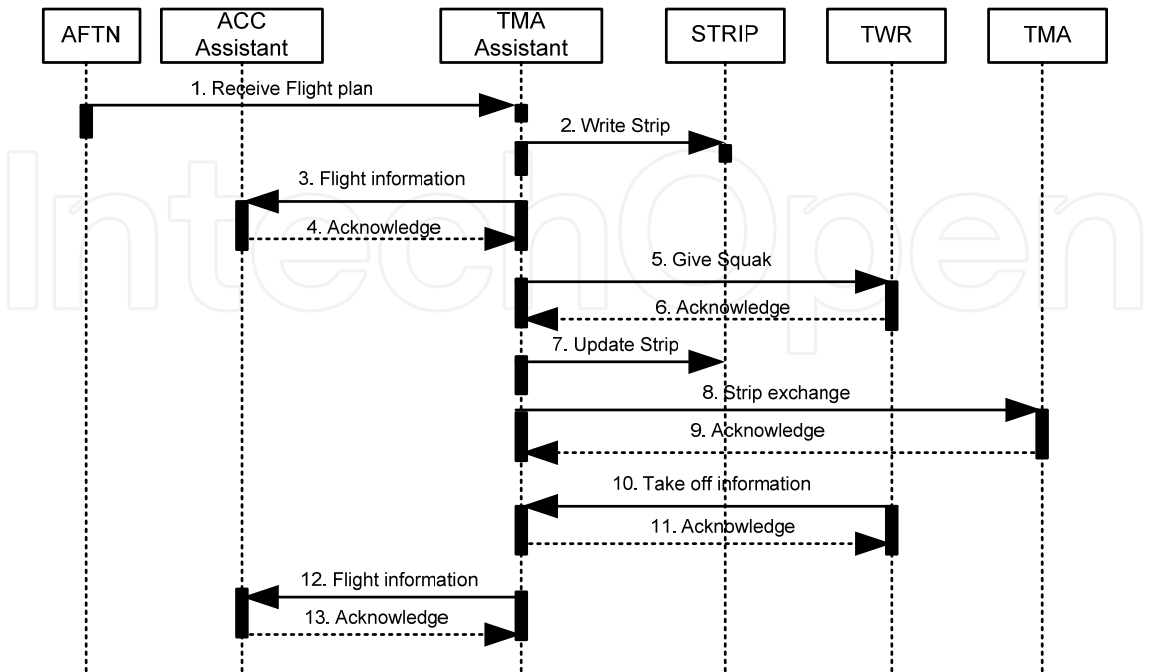Table 2. Call duration in aircraft coordination between two nodes (VCS)

Fig. 5. UML sequence diagram

It may be observed that there are certain differences in the call duration between the data obtained in these two ways, and the reason is that the increase in air traffic often results in the reduction of the coordination time. Thus, e.g. for working position 3 the call duration per single aircraft is longer than according to the measured link occupancy. Regarding working position 4, it is of shorter call duration per aircraft in the analysis using UML formalism than the call duration obtained on the basis of link occupancy. The reason may also be the shorter call duration due to increased traffic.

5.3.2 Traffic matrix

Based on the obtained data on the duration of individual types of calls and data on the number of aircraft in the unit of time the expected values of telecommunication traffic between individual nodes can be determined. It should be noted also, that there is no high uncertainty regarding the volume of the telecommunication traffic such as present in public telecommunication networks. The reason is that the number of aircraft handling is limited by the capacities of single airports.

For the purpose of the analysis, a period of one hour was taken, as usual in the analysis of telephone telecommunication network, and the total traffic between two nodes can be expressed by the following formula

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

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.

Applying formula 1 and data from Table 2, one can obtain the traffic matrix as presented in Table 3, for the forecast number of aircraft between single nodes.

		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	$A_{51}=0.15$	$A_{52}=0.08$	$A_{53}=0.08$	$A_{54}=0.06$	$A_{55}=0$

Table 3. Traffic matrix

5.3.3 Probability of route usage

After having described for the considered network presented in Figure 2, the traffic routing using the routing table (Table 1), and after having determined the expected traffic between the nodes (Table 3), for further analysis it is necessary to determine the probability of the usage of individual route. Before this it is necessary to develop the expanded routing trees

and based on them using the expression which represents the recursive formula for determining the probability of using a certain route (expression 2) according to (Sinković, 1994) determine the probability of using the defined routes for every origin-destination pair. Examples of expanded routing trees are presented in Figure 6, where Li shows nodes where a call could be blocked.
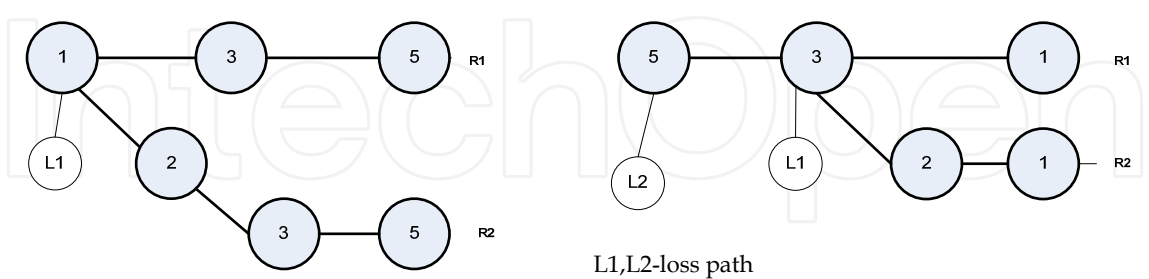

L1,L2-loss path

Fig. 6. Expanded routing trees for origin destination pair 1-5 and 5-3

The recursive formula is

$$P(R_i \text{ used})=\left(\prod_{C_k \in R_i} x_k\right)\cdot\left(1-\sum_{j=1}^{i-1}P\{R_{j(i)} \text{ used}\}\right)$$ (2)

where:

R_i- analyzed route from the defined set of routes

x_k- probability that link k in route is available (the link availability understands that at least one voice channel is free between two nodes)

$R_{j(i)}$- set of links as result of the difference of two routes and not a route in itself

C_k- link k which is element of the observed route i.

Since the number of recursive calculations depends on index i, and on the number of node pairs (originating node and terminating node of call) here the expressions will be developed only for node pair 1-5. Based on the routing table it may be read that the routes according to search order for this pair of nodes are as follows: R_1(1-3-5), R_2(1-2-3-5). The probabilities of using a route are:

$$P(R_1 \text{ used})=x_{13}\cdot x_{35}$$ (3)

$$P(R_2 \text{ used})=x_{12}\cdot x_{23}\cdot x_{35}\cdot(1-x_{13}).$$ (4)

As seen in formulas 3 and 4 the probabilities of using a route will depend on the probability of the link availability on the route. In calculating the probability of using a route for every call origin – destination pair the probabilities of link availability between nodes have been used in the amount of 0.999 as recommended by ICAO.

VCSs have the possibility of assigning priorities to certain calls, and they are not included in the calculation of the usage probability of a certain route since this would additionally complicate the calculation (increase in the number of conditions in the formula). Therefore,

the probability of link availability can be reduced since the calls with higher priority can interrupt the call with lower priority. The Quality of Service expressed by the probability of path availability between two nodes will be realized anyway, since all calls do not have to be guaranteed the same probability of path availability.

Table 4 presents the probabilities of route usage for different values of link availability probability x_k for those pairs of nodes to which node 1 is the originating one and those to which node 2 is the origin, (Mrvelj et al., 2009).

Origin–destination pair	route	Probability of using route for $x_k = 0,999$	Probability of using route for $x_k = 0,99$	Origin–destination pair	Probability of using route for $x_k = 0,999$	Probability of using route for $x_k = 0,99$
1-2	direct	0,999	0,99	2-1	0,999	0,99
	alternative	0,000998001	0,009801		0,000998	0,009801
1-3	direct	0,999	0,990000	2-3	0,999	0,99
	alternative	0,000998001	0,009801		0,000998	0,009801
1-4	direct	0,999	0,990000	2-4	0,998001	0,9801
	alternative	0	0		0,000997	0,009703
1-5	direct	0,998001	0,980100	2-5	0,998001	0,9801
	alternative	0,000997003	0,009703		0,000997	0,009703

Table 4. Probabilities of route usage

The probabilities of connection realization for a pairs of nodes (origin - destination) are obtained by summing up the probabilities of usage of all routes between pair of nodes. That is, the measure for the assessment of the quality of network communication properties entitled *node-to-node Grade of service* (NNGoS) can be presented by the following expression:

$$NNGoS = 1 - \sum_{i=1}^{\substack{\text{number} \\ \text{of rutes}}} P\{R_i \text{ used}\}$$

(5)

5.4 Dimensioning of link capacities between VCSs

The basic goal of this chapter is to determine the link capacities between individual nodes whose arrangement depends on the airport location. Using all the previously introduced restrictions on the route length, on avoiding of closed loops and tromboning, and knowing the traffic requirements between individual nodes, using expression 2 the values of the expected traffic on a link have been obtained. The obtained values are presented in Table 5, 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.

For determining the capacities between the nodes the Erlang B-formula is used as well as all its assumptions defined in (Akimaru & Kawashima, 1993). The obtained values are presented in Table 5 for different Grades of Service (GoS) expressed by the blocking probability (Mrvelj et al., 2009).

Link between two adjacent nodes	Expected traffic in *[Erl]* on the link with $x_k = 0{,}999$	Number of channels for expected traffic and permitted blocking probability $p_b = 0{,}001$	Expected traffic in *[Erl]* on the link with $x_k = 0{,}99$	Number of channels for expected traffic and permitted blocking probability $p_b = 0{,}01$
1-2	0.330997	4	0.339661	3
1-3	0.929339	6	0.923257	5
1-4	0.75896	5	0.749569	4
2-3	0.670267	5	0.672438	4
3-5	0.70901	5	0.700065	4
Total		25		20

Table 5. Expected traffic on link and necessary number of channels with defined quality

It may be observed from the table that there is significant saving in reducing the link availability even on such a small network. By setting priorities for a certain group of calls satisfactory quality can be achieved that will guarantee air traffic safety. In (Eurocontrol (b), 2005) it has been suggested that the number of links between VCSs be determined based on adding redundancy to links between VCSs depending on the number of routes defined between the pair of nodes (call origin - destination). However, this redundancy is equal also for the expected traffic on a link of 0.33*[Erl]* and for the traffic of 0.99*[Erl]*, since it is only indicated that the number of links has to exceed the sum of the expected traffic between two VCSs.

6. Bandwidth requirements for voice transmission over IP based network

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. Analysis capacity of transport link for IP (Internet Protocol) based G/G voice network is based on the research carried in previous section regarding the number of offered calls per hour and call duration.

6.1 Impact factor for bandwidth calculation

There are many factors involved when calculating the bandwidth required through a network. This section of chapter aims to explain these factors, and to offer a simple means of making such calculations. The designer of any network solution that includes voice will need to decide upon which coding algorithm to use. Detailed consideration of each coding method is beyond the scope of this section, but it should be understood that the various coding methods vary in the levels of complexity, delay characteristics and quality. The CODECs which are used for bandwidth calculation in this section are G.728 (ITU-T, 1992), whereas the same CODEC is used in ATS QSIG, and G.711 (ITU-T, 2000).

There are many ways to reduce the bandwidth requirements, and these can be particularly important in the specific network like AGVN. These include silence suppression, RTP (Real-time Transport Protocol) 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 Organisation (*OSI seven layer model*). Standard method of transporting voice samples through an IP based network required the addition of

three headers; one for each layer. These headers are IP, UDP (User Datagram Protocol) and RTP. An IPv4 header is 20 octets; a UDP header is 8 octets and an RTP header is 12 octets. The total length of this header information is 40 octets (bytes), or 320 bits, and these headers are sent each time a packet containing voice samples is transmitted. The additional bandwidth occupied by this header information is determined by the number if packets which are sent each second. The effect of each layer's contribution the communication process is an additional header preceding the information being transmitted.

This section does not discuss header compression schemes and include them in calculation of bandwidth requirements. Furthermore, this section 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 not recommendations concerning packet duration. Although codecs vary in their quality and delay characteristics and there is not yet an agreed standard, there are only the most common codecs used for voice transmission over IP. Similarly, there is no recommendation on the packet duration to use in the different environments, but it is considered that 20ms is a good choice for normal Internet conversation with acceptable bandwidth. For office environments where there is almost no bandwidth restriction, G.711 at 20ms packet duration is recommended. In RFC 1889, the Internet Engineering Task Force includes an example where the duration is 20ms, but they do not suggest this as a recommended value. The Table 6 shows bandwidth requirement depending on packet duration for G. 711 (PCM) and G.728 (LD-CELP) which is used for bandwidth calculation in this in this section.

Codec	Packet duration	Bandwidth [kbps]
G.728 (LD-CELP) 16kbps compression	30 milliseconds (48 samples)	27
G.711 (PCM) 64kbps uncompressed	20 milliseconds (32 samples)	80

Table 6. Bandwidth requirements for G.711 and G.728 at different packet duration

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

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

Respecting all the assumptions and restrictions introduced in previous sections regarding the route length, avoiding of closed loops and tromboning and knowing the values of the expected traffic on a link, the capacities between the nodes have been obtained.

The results of bandwidth calculation for the transmission of voice over an IP based network have been presented in table 7 (Markežić et al., 2009) for the same number of voice channel which is planned for circuit switch network considered in section 5 and shown in Figure 2.

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 16k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
2-3	0,670267	5 x 16k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
3-5	0,70901	5 x 16k	2 x 64 **[kbps]**	5 (421 **[kbps]**)	5 (140,3**[kbps]**)
Total		*25x16k (400kbps)*	*640* **kbps**	*2105* **kbps**	*561,2* **kbps**

Table 7. Bandwidth calculations for different type of links

The values in columns 5 and 6 are obtained respecting all previously introduced in section 6.1 and for two types of codecs: G.711 (column 5) and G.728 (column 6). Furthermore, obtained values are presented in Table 7 (column 5 and 6) have been achieved without the impact factors regarding layer 2 protocols.

The data in Table 7 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.

7. Conclusion

Modern communication networks have to be capable of responding to random fluctuations of requests and errors in different ways. One of them is traffic routing i.e. resource allocation. The designing of such networks (intelligent ones) and their management represent a challenge in mathematical, engineering and economic manner. This chapter describes the scheme of dynamic routing and the derived and presented model which is useful for dimensioning of initial link capacities as well as in the analysis of network stability. Emphasis is on the telephone network for G/G communication in ATM, for which the user's requirements have been described together with the technical requirements that are necessary to support them.

For the design of AGVN the usual methods of determining the telecommunication traffic are used. It should be emphasised, however, that there is a difference in relation to public telephone networks in that the calls in ATM are shorter and the recommended GoS value is lower (0.001). The chapter presents the necessary capacities for GoS value that is used in public networks and for the recommended GoS value for AGVN. The results show substantial savings in the number of channels. Since VCSs can distinguish the type of call and allocate priorities, for the dimensioning of the transmission link capacities a higher GoS value can be used, realizing at the same time a satisfactory Quality of Services for certain calls.

The improvement of dimensioning models of the transmission capacities requires a detailed analysis of traffic flow characteristics in AGVN, as well as inclusion of priorities for a certain group of calls in the model that represents the goal of further research.

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Air Traffic Control

Edited by Max Mulder

Improving air traffic control and air traffic management is currently one of the top priorities of the global research and development agenda. Massive, multi-billion euro programs like SESAR (Single European Sky ATM Research) in Europe and NextGen (Next Generation Air Transportation System) in the United States are on their way to create an air transportation system that meets the demands of the future. Air traffic control is a multi-disciplinary field that attracts the attention of many researchers, ranging from pure mathematicians to human factors specialists, and even in the legal and financial domains the optimization and control of air transport is extensively studied. This book, by no means intended to be a basic, formal introduction to the field, for which other textbooks are available, includes nine chapters that demonstrate the multi-disciplinary character of the air traffic control domain.

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