Numerical Simulation and Traffic Analysis in Carrier Voice Networks

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Abstract: - Performing traffic and voice transmission quality analysis is sometimes required in carrier voice networks due to conflicts appear between operators. One of the reasons of conflicts is, in many cases, the incomplete national regulations in the field of telecommunications. In such cases, juridical expertise should be carried out in the similar conditions as the real environment, but in almost all the situations the specific equipment is not available anymore. In order to highlight certain technical aspects and compensate for the lack of equipment, a software that simulates equipment and processes in telecommunication networks are necessary. The paper presents a simulation process of interconnection between operators, realized at the beginning of 2016, for establishing the quality parameter for a real case from 2004. The simulation is based on an interface represented by Voice Gateway, and through various settings on a dial plan, which can establish a real operation. Voice Gateway configuration is particularly complex and can go up to transforming the simulation into a gateway configuration tutorial. Finally, traffic trends are studied over a period of time, analyzing the conflicts started between the two telecommunications operators regarding traffic and quality of services.

Key-Words: - ID change, simulation, traffic, voice quality, IP networks, VoIP

1 Introduction

In Romania, as in Europe, after 2000, the number of operators developing their own telecommunication networks increased. This has helped to improve speed in accessing telecommunication services and to accomplish the appropriate level of user requirements. During this time, in the Romanian National Regulatory Authority in Telecommunications (ANCOM) defined:

- carrier operators and/or service providers as those with “significant power ... of broadband electronic communication services and public telephone services at fixed points”, who can directly provide users with Internet access services;
- dominant operators as those with “significant market power in terms of access to their mobile network in order to terminate calls”, requiring interconnection with any other carrier operator and/or telecommunication service provider.

As a result, agreements have emerged between carrier operators and/or service providers and dominant operators, stipulating the installation of bi-directional communication streams (e.g. 30 channels, E1 [1]), without establishing details on how to identify and resolve deviations from prerequisites such as exceeding the value of daily traffic (e.g. traffic exceeding 30,000 minutes/month), forced route “maturing” (e.g., provided after 6 months), etc. Services required by service provider operators from the dominant operator were [2]:

- termination of calls originated by customers of the carrier operator and /or service provider to the dominant operator’s network;
- termination of calls originated by the dominant operator’s network to the geographic numbers allocated to the carrier operator and/or service provider (for example, Romania: 021 569 xxxx where xxxx = 0,1, ... 9);
- routing calls originated by the dominant operator’s network to the carrier operator and/or service provider’s exchange for calls with carrier selection;
- termination of calls originated by the dominant operator network to non-geographic numbers allocated to the carrier operator and/or service provider (e.g., green numbers 0800 120 xxx, prepaid cards 0808 810 xxx, or value added services 0903 030 xxx, where x = 0 1, ..., 9); allocation of short xxxx numbers, callable from the dominant operator’s network, and which are routed to non-geographic numbers of the carrier operator and/or service provider, providing various services to customers depending on the set category.

- line services leased between customers and the company of the carrier operator and/or service provider through the dominant operator's network.

VoIP (Voice over IP) is a means of encoding, transmitting and reassembling sound in a network from IP (I*nternet* P*rotocol*) packets [3]. IP telephony is a means of making a phone call over an IP network, without users being aware that they do so. VoIP services strongly entered the market in 2004 [4], along with access to broadband Internet services – access to higher speed data – over 768 Kbs for transfer download rates, according to a 2008 definition of the US Federal Communications Commission. The major advantage of VoIP communications is that, as the other applications or data communication-based services develop, they can be easily integrated into VoIP services, which leads to virtually limitless possibilities for service providers. The ability of VoIPs to make calls and send faxes over an IP-based network provides quality services with an advantageous cost/benefit ratio for the user.

The case presented in our paper, traffic estimates for services established between the dominant operator and the service provider operator was 400,000 minutes/month in the first 3 months, of which 100% voice – to – voice calls, which in turn is distributed in a proportion of 70 % over the 08-18 time slot and 30% over the 18-08 time slot. Transmission hours were spread evenly from Monday to Sunday, with peak traffic from 10 am to 4 pm. The maximum number of simultaneous calls was 70, and the prospects for traffic increase are determined, depending on services, at 2 million minutes per month for the next eight months. In this case, the increasing trends of traffic over a short period of time (e.g. 20 days) resulted in the dominant operator taking additional measures to monitor traffic and moderate its growth, a fact that caused discontent in the carrier operator and/or service provider, the latter citing the fact that the traffic of their subscribers is a process that evolves in a probabilistic manner which cannot be controlled, and therefore requiring a new interconnection agreement for national and international call routing to the dominant operator's network. The carrier operator and/or service provider was already offering comprehensive services to other telecommunication companies through its infrastructure.

For the quality analysis of the services, we used numerical simulation because, at that moment, there was no possibility to make this evaluation directly on equipment. Usually, *node.js* [5] is employed to run server applications in real time and excels in the performance of using *non-blocking* I/O and asynchronous events. A real advantage is in the fact the same language can be used to construct the *backend* of the *frontend*, already constructed in JavaScript. Still in its beta phase, the language is used by many large companies (Microsoft, Yahoo!, LinkedIn, eBay, The New York Times, etc.), on account of being stable and efficient. Few programming languages have known such rapid development as *node.js*. It can be easily seen that most of the resources found on GitHub are in JavaScript or *node.js*. As such, it is a good opportunity for developing applications based on artificial intelligence, which for the time being are not numerous. The *node.js* programming language is fairly reliable for creating services e.g. API (A*pplication* P*rogramming* I*nterface*), which would allow *multithreading* – the application's capacity to provide simultaneous responses. The beneficial part of the *node.js* language is the immense amount of code, already presented on GitHub. A developer only has to combine the resources, in order to save time when creating a new application.

For testing the method, we compared the results with those obtained through experimental evaluation, a process which was carried out on similar equipment.

The following chapters will present the carrier voice network based on a real situation, the description of the simulation process, the traffic and voice transmission quality analysis and some conclusions.

2 Hardware and software used by telecommunications operators

2.1 Methods and equipment for the interconnection of telecommunication operators

The physical equipment (hardware) used by the carrier operator and/or service provider to connect with the dominant operator was an Ericsson ANS Translocal R420R2 exchange [6] with Diax A/S NMLCM v5 .43 support/OS/software, using SS7 CCS7v2.2 signalling, which allowed routing national and international calls to the dominant operator's network. Equipment used (in 2004, similar to Alcatel equipment) by the dominant operator had the following specifications: SW releases [7] SW of the platform (according to existing HW) are: Evolium SSP release R272.3.4.2/V9 (E10), Evolium RCP release R909 (EP8) and Evolium M HLR release GB910 (EP9). The exchanges of the dominant operator and the carrier operator and/or service provider fulfilled certain interconnection requirements, such as those illustrated in Fig.1 (VoIP Network ↔ dominant operator network). VoIP calls are converted to digital channel calls (64 Kbs) and oriented to the dominant operator's network.

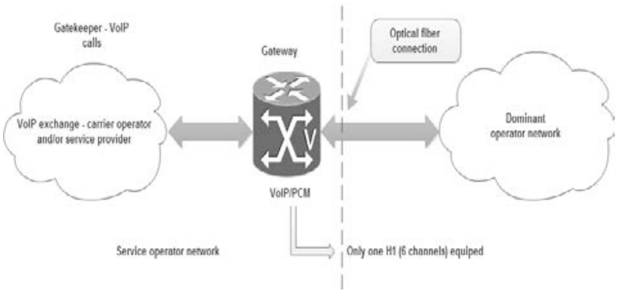


Fig.1. Explanatory scheme on interconnection between the carrier operator and/or service provider network and the dominant operator network on E1 stream, through the adapter interface VoIP-PCM (Pulse Code Modulation).

The gatekeeper, which provides call control for endpoints, more streamlined for incoming calls, is preceded by other equipment. Thus, VoIP calls incoming from interconnection lines of the input port, where transmission was performed by packets, were taken from an IP switch (DJAX-AS) layer 2 or 3 and, if the numbers are specific to the dominant operator, they were then routed to the receiving interface port. In the case of the service operator, the IP switch type (DJAX-AS) was layer 3 (in OSI - Open Systems Interconnection architecture) and therefore it could provide, in addition to the outgoing address, the input port address and not that of the caller. Thus, the layer on which the layer 3 switch acts was the transport layer (Fig. 2).

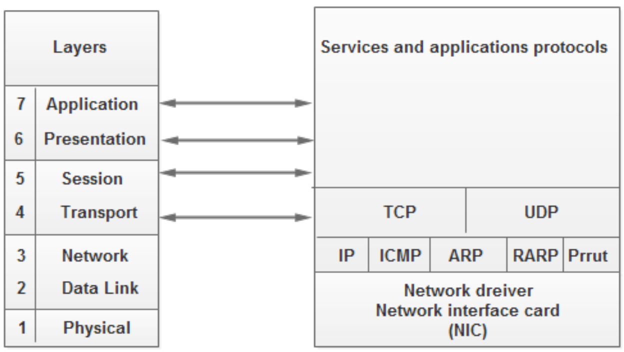


Fig.2. The OSI model applied to interconnection between the service operator's network and the dominant operator's network.

The IP package received from a call source, also called UDP (User Datagram), has the contents shown in Fig. 3, being marked by the position in the packet of the address in which that particular switch can act.

1 16	17 32
Source port	Destination port
UDP message length	Verification sequence
Date	

Fig.3. Explanatory scheme on the content of the package communicated via IP from a call source.

Thus, the switch used by the carrier operator and/or service provider can act on the identity of the call source, and thus of the caller, regardless of the network where the call originates. It is however possible, depending on the input interface equipment (gatekeeper), for the switch to be a layer 2 OSI.

In this case, the ID change for incoming calls from landline or international networka is performed in the gatekeeper. Fig. 4 shows the configuration of interconnections in the case of incoming calls in the carrier operator and/or service provider network from other national and international networks. The access code in 2004, established under the interconnection agreement of the carrier operator and/or service provider with the national landline network, was changed in 2008 by ANCOM into access code 1010, according to the National Numbering Plan.

The equipment used proved that the carrier operator and/or service provider does not preserve caller ID (e.g., 021-BPQMCDU), introducing its own authorised numbering (021-569xxxx). Or, in order to change caller ID, the operation might be assigned (allocated) to either:

a) the layer 3 switch of the VoIP equipment (the carrier operator and/or service provider exchange);

b) or directly in the channel interface - 1 - MUX converter PCM/VoIP used for interconnecting the VoIP exchange with the mobile and landline network (gatekeeper).

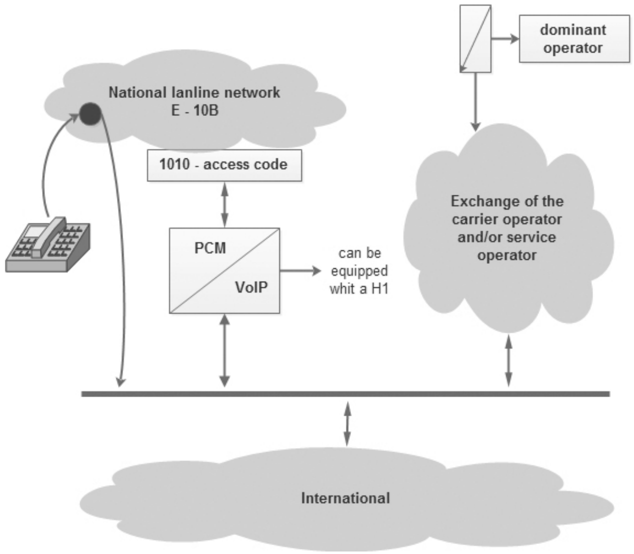


Fig.4. Explanatory scheme on the direct interconnection between the carrier operator and/or service provider network and the landline network (Alcatel) via an adapter (of the same type as for the interconnection with the dominant operator), placed PCM-VoIP, only an H1 being equipped;☎ - VoIP phone adapter; ◎ - communication route through the local loop of the landline network (only the pair of wires).

In the first case, each incoming channel from platform 1010 is assigned from the outset a specific numbering on the VoIP network of the carrier operator and/or service provider, such as 021569111x (authorised numbering, the tranche of numbers used being highlighted in bi-annual reports of the carrier operator and/or service provider to ANCOM, where the carrier operator and/or service provider declares that the number of its subscribers is 11, 12 or 15), thus sufficient to be used and allocated as ID to incoming channels on the landline or international network.

In the second case, if a simple converter is used, with no processing capacity, the ID change is performed within the layer 3 switch, the other numbering problems being the same as in the previous case. Connection with the dominant operator on the output of the VoIP - PCM converter is equipped with a 30-channel MUX, and not E1 stream. For calls originating in the dominant operator's network, calls will be connected directly via VoIP network – 4 – bypassing platform 1010 (Fig. 5).

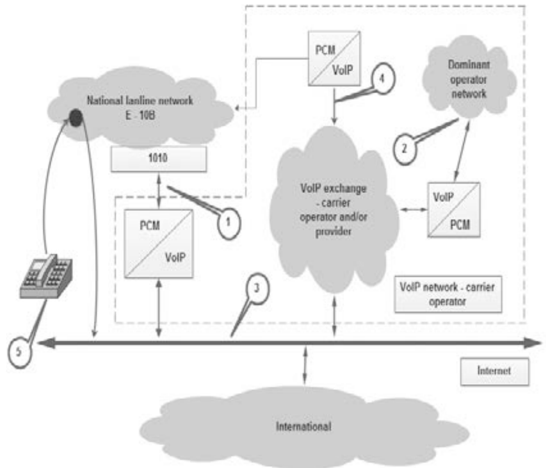


Fig.5. Explanatory scheme on the physical interconnection of the network operator and/or service provider network with the Internet data network, landline network and dominant operator's network via a local router: 1010 – no-step access platform to other networks; PCM/VoIP or VoIP/PCM – E1 PCM stream converter interface or voice over IP/E1 PCM stream

H.323 (1996) is a suite of protocols that includes both signalling protocols (signalling and controlling point-to-point and multi-point conferences – calling, data bandwidth, etc.), as well as transport protocols. The H.323 standard describes a series of functional components, which can be implemented separately in different equipments, or can be grouped into one multi-purpose equipment. The advantages of using H.323 standard are: caller ID, interoperability, detailed control over the call (to and from the gateway), integration of different network technologies, support for different media contents, H.323 gatekeeper.

The disadvantages of using H.323 standard are: gateway configuration is complicated, lack of a centralised numbering plan, call survival (if the gatekeeper connection is lost, then all calls will be interrupted).For the first 10 months, the numbering plan used by the carrier operator and/or service provider analyzed in this case was limited to 10,000 call numbers (required by the equipment used [8]), thus, as recommended by ITU-T (International Telecommunication Union), it could not be considered a public telecommunication service operator, but at most a VPN (Virtual Private Network) [9] implemented in the network of another dominant operator or a service provider between networks (ISP - Internetwork Service Provider). A virtual private network [10], VPN, is a way of simulating a private network through a public network (e.g. the Internet) and is called virtual because it depends on virtual/temporary connections, which do not have an actual physical presence, achieving transmission through data packets sent through multiple Internet transit equipments or a network built on an ad hoc basis [11]. The carrier operator and/or service provider could allow routing, through its network, national and international calls from other national telecommunication operators to subscribers of the dominant operator with which they concluded an agreement, but only if this behavior was tolerated by the dominant operator

2.1 Simulating an interconnection situation

Since it is not possible to carry out simulations on a model that has its 2004 configuration, mainly for the analysis of an E1 stream as a link between a communication service distributor (Ericsson Diax Ans) and the dominant operator network as transit network (Alcatel 1000 E10 OCB 283 HC [1]), under certain circumstances, different software can be used to simulate that described earlier in this paper. Specifically to achieve such goals, a medium that simulates the infrastructure of some operators was used, in this case the connection between the dominant operator and the carrier operator.

Emulator GNS3 [12] and the application Cisco Unified Communication Manager (CUCM) [13] were used for simulation, the latter being an integrated communication system, based on voice, video and data IP, also allowing the exchange of information between operators.

CUCM standardized components, by levels, are:

- infrastructure containing the routers, switches and Voice Gateway; the infrastructure level enables exchange of data, voice and video between network devices and applications.
- call control which is responsible for handling calls, controlling device and managing dialing plans.
- an application that is an independent level of the call control, which allows applications to run anywhere on the network, such as Voice Mail, Cisco Unified Meeting Place, Cisco Emergency Responder, as well as of the integration of SOAP, Q.SIG, H.323, MGCP and SIP protocols [8]. The CUCM node in this simulation will be located in the carrier operator's website. GNS3 emulator was used to configure the Cisco 3725 router and voice gateway, and to simulate calls we used Cisco IP Communications application [14].

In the proposed simulation scenario, both locations responsible for each operator simulate the connection to dial the number corresponding to operators, each location having a specific set numbering plan (Fig. 6).

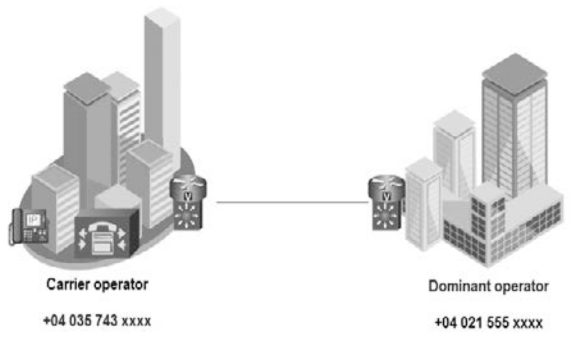


Fig.6. Structure of interconnections between operators. Sources: Authors Compilation [12].

The hardware resources we used to achieve the simulation were a Personal Computer with a 3.6G i7 processor and 16 GRAM. CUCM was installed on a VMWare virtual machine, with a storage space of 160G and 8 GRAM.

The IP addresses used – assigned to the devices – and the dial table/plan are given in Tab.1. CUCM installation can sometimes be complicated, while CISCO provides an ISO file (file system used with a CD-ROM mass-media) in this respect, and not a bootable application. There are quite varied solutions to transfer these ISO images to a bootable DVD.

Table 1. IP addresses and numbering plan.

Location	Devices	IP address	Numbering plan
Dominant operator	Cisco 3725 Router Phone 1	Fa 0/0 192.0.20.1	+021 555 xxxx 6000
Carrier operator	Cisco 3725 Router Phone 1 IP Phone 2 IP Voice Gateway CUCM	Fa 0/0 192.168.0.100 Fa 0/0 192.168.0.102 Fa 0/0 192.168.0.103 Fa 0/0 10.10.10.10 Fa 0/0 192.168.0.250	+0357 43 xxxx 4001 4002

For the virtual machine that will host CUCM, parameters were set as follows: Linux Red Hat Enterprise operating system, dedicated 2-core processor, 8 GB memory, 160 GB HDD. After setting these parameters, we proceeded to install CUCM. To simulate connections, CICO 3725 virtual routers were used, emulated by GNS3 application (Fig. 7 and Fig. 8).

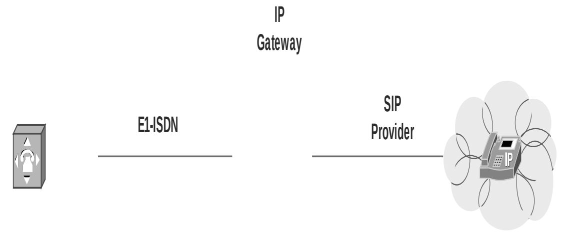


Fig.7. Explanatory scheme on the interconnection of service providers in the GNS3 emulator. Sources: Authors Compilation [12].

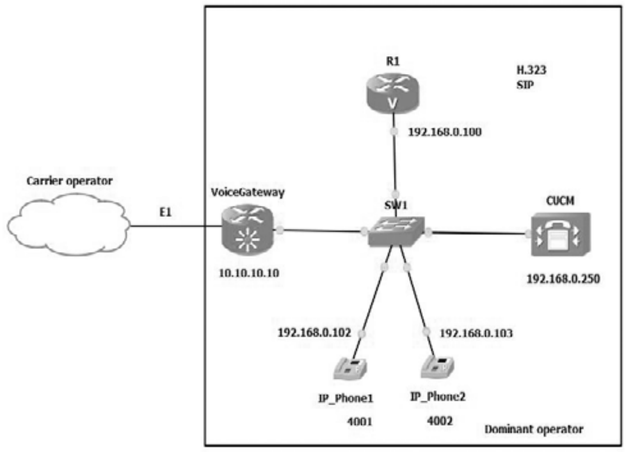


Fig.8. Explanatory scheme on the implementation of connections in GNS3. Sources: Authors Compilation [12].

The idea is to simulate an environment as similar as possible to that described in section 2, based on the H.323 protocol and SIP (Session Initiation Protocol) on the VoIP part. SIP, a signalling protocol, found in the application layer of the OSI stack, is used to create, modify, and terminate sessions between two or more participants in the conversation. The H.323 protocol will be used between the routers and SIP. Topology is created in GNS3, and CUCM is the connection between the node and the Vmnet8 interface of the virtual machine. Routers will be configured to provide connections based on H.323 protocol and SIP to CUCM. Ethernet port configuration and protocol H.323 activation for Voice Gateway are described in the next secvence:

```
Router>en
Router#conf t
Router(config)#hostname VoiceGateway
VoiceGateway(config)#int eth 1/0
VoiceGateway(config-if)#ip      add      10.10.10.10
255.255.255.0
VoiceGateway(config-if)#no shut
VoiceGateway(config-if)#
*Mar 1 00:08:23.823: %LINK-3-UPDOWN:
```

```
*Mar 1 00:08:24.823:
%LINEPROTO-5-UPDOWN:  Line  protocol  on  Interface
Ethernet1/0,
changed state to upno shtdo
VoiceGateway(config)#ip   route   0.0.0.0   0.0.0.0
10.10.10.1
VoiceGateway(config)#end
VoiceGateway#conf t
VoiceGateway(config)#voice call send-alert
VoiceGateway(config)#voice rtp send-recv
VoiceGateway(config)#voice service voip
VoiceGateway(conf-voi-serv)#h323
VoiceGateway(conf-serv-h323)#end
VoiceGateway#
H.323 service is up
This gateway is not registered to any gatekeeper
Alias list (CLI configured) is empty
Alias list (last RCF) is empty
VoiceGateway#conf t
VoiceGateway(config)#voice service voip
VoiceGateway(conf-voi-serv)#allow-connection h323 to
sip
VoiceGateway(conf-voi-serv)#allow-connections sip to
h323
VoiceGateway(conf-voi-serv)#voice translation-rule 1
VoiceGateway(cfg-translation-rule)#rule 1 /^9(.*$\)/
/\1/
VoiceGateway(cfg-translation-rule)#voice
translation-profile OUT
VoiceGateway(cfg-translation-profile)#translate
called 1
VoiceGateway(cfg-translation-profile)#exit
VoiceGateway(config)#end
VoiceGateway#conf t
VoiceGateway(config-if)#int fa 0/0
VoiceGateway(config-if)#ip     add     192.168.0.100
255.255.255.0

VoiceGateway(config-if)#h323-gateway    voip    bind
srcaddr 192.168.0.101
VoiceGateway(config-if)#exit
VoiceGateway(config)#dial-peer voice 4021555 voip
VoiceGateway(config-dial-peer)#destination-pattern
4021555
VoiceGateway(config-dial-peer)#session        target
ipv4:10.10.10.10
VoiceGateway(config-dial-peer)#codec g711ulaw
VoiceGateway(config-dial-peer)#exit
VoiceGateway(config)#write
VoiceGateway#
```

It is important for Voice Gateway to ensure an equal amount of calls simultaneously, both between the VoIP provider to the Gateway, and between the Voice Gateway and the dominant operator. E1 interface provides a capacity of 24/32 calls simultaneously. If necessary to increase call capacity, multiple E1 streams can be installed. It can be considered that within the carrier network to Voice Gateway, the numbering of the caller and the call recipient may be stored according to SIP. It is possible that, by processing ID and numbering, the interconnection interface between operators can change the identity of the call source. SIP is widely used today as a signalling protocol for VoIP. The protocol is based on text messages, similar to HTML, which allows SIP messages to be read directly by human users. Use of a PBX (Private Branch Exchange) by the dominant operator, which does not support SIP Trunking, will require a Voice Gateway to connect to the SIP provider. The gateway will be placed between the PBX and the SIP provider, using an E1 ISDN (Integrated Services Digital Network) stream via the IP protocol [15]. These settings are not complete and are only the initial setup of voice characteristics. H323 calls are activated to SIP and SIP to H323 for Voice Gateway and R1, respectively. Subsequently, the connection for CUCM will be configured. Cisco Unified Communications Manager supports several types of Cisco gateways. To configure Voice Gateway, they are used for communication terminals when IP telephony infrastructure communicates with PSTN (Public Switched Telephone Network) or other devices such as PBXs, analogue phones, fax machines, etc. The interfaces specify how the gateway communicates with the PSTN or other devices. Different types of interfaces can be used, but overall CUCM supports two major gateway types based on H323 protocol or MGCP (Media Gateway Control Protocol). This simulation used a gateway based on the H.323 protocol and configured for IP 10.10.10.10 (Fig. 9). An important component of a voice communication system, which uses CUCM, is establishing a dial plan. The dial plan describes how calls are processed and routed. This plan can be described based on certain main functions [16], of which the following are listed:

- *Endpoint addressing*, function used for IP phones, fax machines, analogue phones, conference systems and applications;
- *Calling privileges* which can be used to configure different types of access depending on the users' level of access;
- *Path selection* through which the same destination can be reached by different routes, and if a path to the gateway is not possible, then the traffic can be routed through another operator;
- *Digit manipulation*, when calls are initiated from the carrier operator network to the dominant operator, and numbers can be converted to suit the specific numbering plan. In this case prefixes can be used to access the dominant operator's network. Thus the number of the source can be modified to remove the prefix (Fig. 10);
- *Call coverage* with which output groups can be created based on digit analysis. Thus calls can be routed to the gateway or to another communication trunk.

- *Patterns* by which routing plans can be created relying on a predefined model (Fig.11). Also, the absence of a patterns route, can determine display an unknown number (Fig. 10).

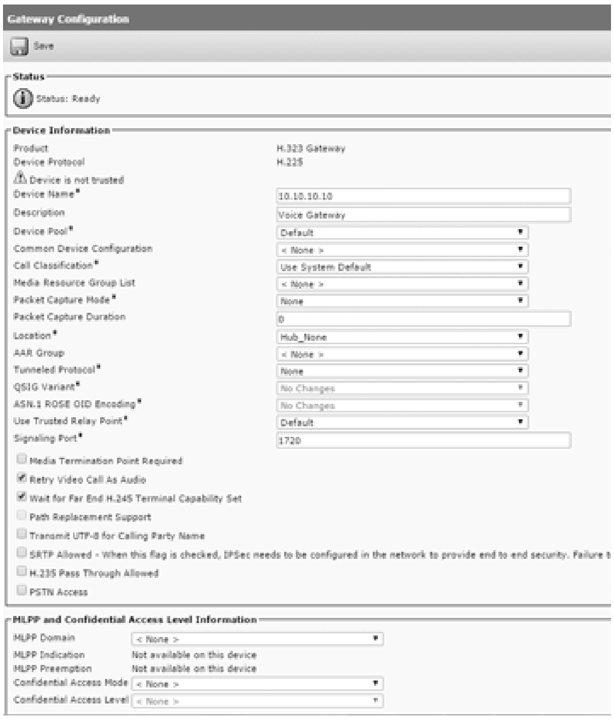


Fig.9. Creating Voice Gateway based on H323 protocol in CUCM. Sources: Display Capture [12].

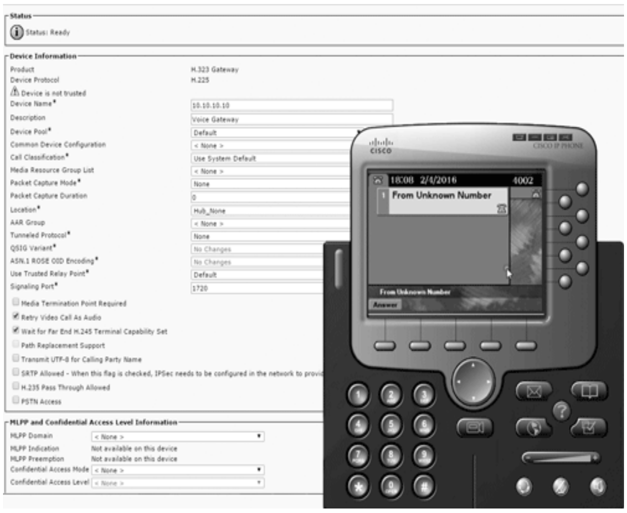


Fig.10. Explanatory scheme on showing an unknown number on the phone display in absence of a routing plan. Sources: Display Capture [12].

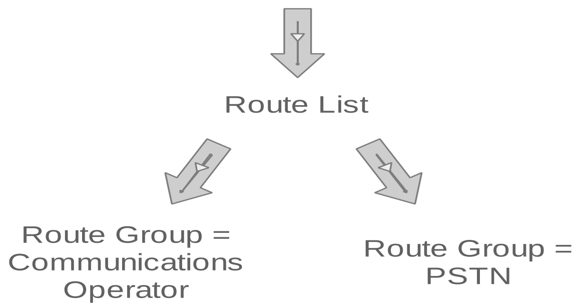
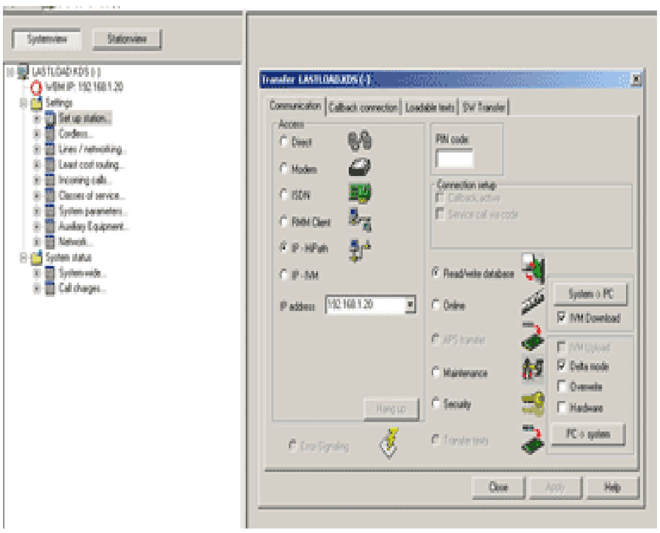


Fig.11. Pattern-based routing plan. Sources: Authors Compilation [12].

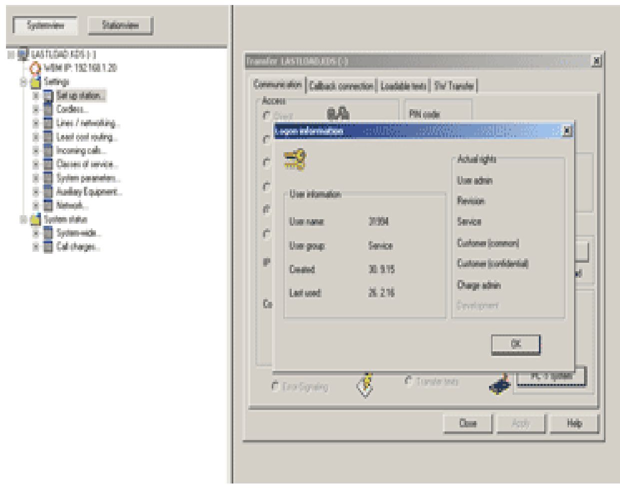
2.3 Testing the interconnection situation

Since it was not possible to run tests on a mockup with exactly the above-mentioned configuration, similar equipment was used to test those previously described in the paper. In order to attain our goals, we used an environment simulating the infrastructure of some operators (Fig. 12 b,c), namely the connection between the dominant and the carrier operator. The tests were performed for 3 relevant situations [2]:

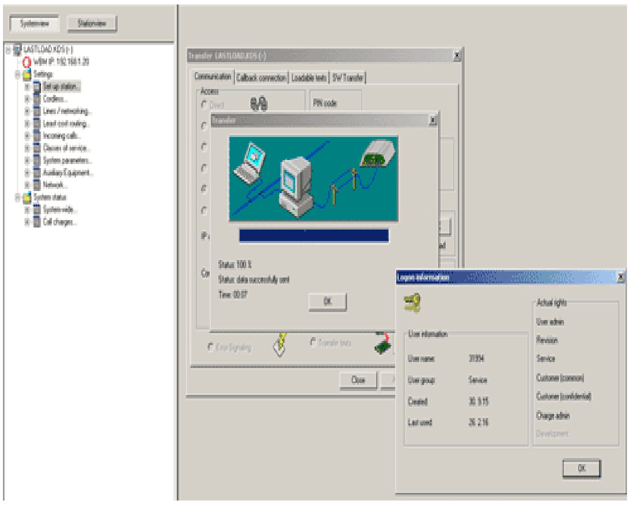
Case 1: - one unaltered and one altered port. The call is initiated on a phone with the number 0132 130 002 –port 1 (unaltered port), and the recipient mobile phone shows the unaltered number 96 850 112, which is also registered in the logs (ABOIN) (Fig.13).



a)



b)



c)

Fig.12. Explanatory scheme on the hardware (a) and software equipment (b-c) used for the tests.

```
ABOIN:    CEN=1/91-09-16/03  H  24  MN  52/INTERROG.
CARACTERISTIQUES D'ABONNES
@ND=96850112:
   TRAITEMENT TGLAIN ACC
       ND=96850112            NE  =085-00-112
            TAX  =  00000005  +  00000000  +  00000000  +
00000000
               TY= NBS2+ NBA30+PRAC3
               CAT= NAT2+CLAB4
                MAR=ASD
                 DS= 96 850212     <96850215
           +96850991      <96850995
    TRAITEMENT TGLAIN EXC
@
@URAIL:
 CEN=1/91-09-16/03  H  36  MN  25/LISTAGE  EQUIPEMENTS
U.R.A.
@AFCN=85-0<20:
    TRAITEMENT TPSSER ACC
AFUR=085 TYPUR=URA     NBEQPT=0064
    NE= 085-00-000  S2F
    NE= 085-00-001  ND=      96850001 S2F
    NE= 085-00-002  ND=      96850002 S2F
    NE= 085-00-003  ND=      96850003 S2F
    NE= 085-00-004  ND=      96850004 S2F
```

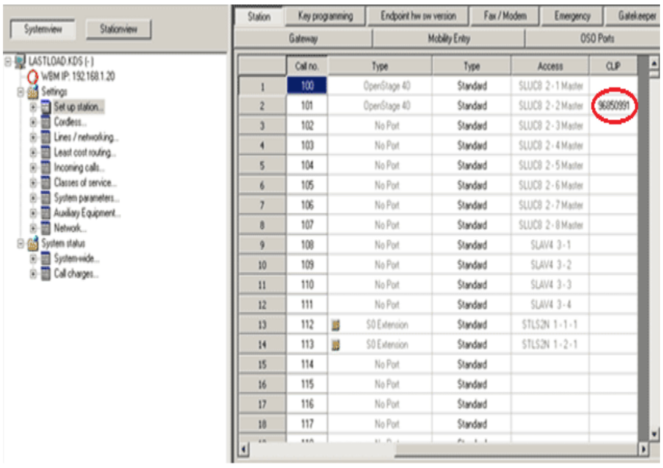
```
    NE= 085-00-005  ND=     96850005 S2F
    NE= 085-00-006  ND=     96850006 S2F
    NE= 085-00-007  ND=     96850007 S2F
    NE= 085-00-008  ND=     96850008 S2F
    NE= 085-00-009  ND=     96850009 S2F
    NE= 085-00-010  ND=     96850010 S2F
    NE= 085-00-011  ND=     96850011 S2F
    NE= 085-00-012  ND=     96850012 S2F
    NE= 085-00-013  ND=     96850013 S2F
    NE= 085-00-014  ND=     96850014 S2F
    NE= 085-00-015  ND=     96850015 S2F
    NE= 085-00-016  ND=     96850016 O2F
    NE= 085-00-017  ND=     96850017 O2F
    NE= 085-00-018  ND=     96850018 O2F
    NE= 085-00-019  ND=     96850019 O2F
    NE= 085-00-020  ND=     96850020 O2F
    NE= 085-00-021  ND=     96850021 O2F
    NE= 085-00-022  ND=     96850022 O2F
    NE= 085-00-023  ND=     96850023 O2F
    NE= 085-00-024  ND=     96850024 O2F
    NE= 085-00-025  ND=     96850025 O2F
    NE= 085-00-026  ND=     96850026 O2F
    NE= 085-00-027  ND=     96850027 O2F
    NE= 085-00-028  ND=     96850028 O2F
    NE= 085-00-029  ND=     96850029 O2F
    NE= 085-00-030  ND=     96850030 O2F
    NE= 085-00-031  ND=     96850031 O2F
    NE= 085-00-033  ND=     96850033 AQE
    NE= 085-00-034  ND=     96850034 AQE
    NE= 085-00-035  ND=     96850035 AQE
    NE= 085-00-036  ND=     96850036 AQE
    NE= 085-00-037  ND=     96850037 AQE
    NE= 085-00-038  ND=     96850038 AQE
    NE= 085-00-039  ND=     96850039 AQE
    NE= 085-00-040  <  085-00-047 NQA
    NE= 085-00-048  ND=     96850048 S2F
    NE= 085-00-049  ND=     96850049 S2F
    NE= 085-00-050  ND=    296853050 S2F
    NE= 085-00-051  ND=     96850051 S2F
    NE= 085-00-052  ND=     96850052 S2F
    NE= 085-00-053  ND=     96850053 S2F
    NE= 085-00-054  ND=     96850054 S2F
    NE= 085-00-055  ND=     96850055 S2F
    NE= 085-00-056  ND=     96850056 S2F
    NE= 085-00-057  ND=     96850057 S2F
    NE= 085-00-058  ND=     96850058 S2F
    NE= 085-00-059  ND=     96850059 S2F
    NE= 085-00-060  ND=     96850060 S2F
    NE= 085-00-061  ND=     96850061 S2F
    NE= 085-00-062  ND=     96850062 S2F
    NE= 085-00-063  ND=     96850063 S2F
    NE= 085-00-064  ND=     96850064 AQE
    NE= 085-00-065  ND=     96850065 AQE
    NE= 085-00-066  ND=     96850066 AQE
    NE= 085-00-067  ND=     96850067 AQE
    NE= 085-00-068  ND=     96850068 AQE
    NE= 085-00-069  ND=     96850069 AQE
    NE= 085-00-070  ND=     96850070 AQE
    NE= 085-00-071  ND=     96850071 AQE
    NE= 085-00-072  <  085-00-079 NQA
    NE= 085-00-080  ND=     96850080 ACP
    NE= 085-00-081  <  085-00-095 NQA
    NE= 085-00-096  <  085-00-111 NEQ
    NE= 085-00-112  ND=     96850112 ACP
    NE= 085-00-113  <  085-00-127 NQA
    NE= 085-01-000  <  085-39-127 NEQ
    NE= 085-40-000  <  085-41-127 NEQ
    TRAITEMENT TPSSER EXC
@
@ABOIN:      CEN=1/91-09-16/03  H  37  MN  30/INTERROG.
CARACTERISTIQUES D'ABONNES
@
@FSMIN;
```

```
 CEN=1/91-09-16/03  H  38  MN  13/INTERROGATION  SUR
    TRAITEMENT TFSMIN ACC
    NFSM=   LAN6   TYR=RN
    PS= 00201 LOI= 31      D= N   CORR= BASE
COC RANC LSD    SRV ILS      TSV-VTSV TSM-VTSM  COM-
LRX-IT  ETCS
    00     00   0004   S7  LAN61      001-00    000-00
1-001-00  ACTI+NBLO
    01     01   0005   S7  LAN62      002-00    001-00
1-009-00  INIT+NBLO
    NFSM=   LAN3   TYR=RN
    PS= 13000 LOI= 02      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0014   S7  LAN30      001-02    000-01
1-001-01  INIT+NBLO
    NFSM=   LAN4   TYR=RN
    PS= 14000 LOI= 31      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0011   S7  LAN41      001-03    000-03
1-001-03  INIT+NBLO
    01     01   0012   S7  LAN42      002-03    001-03
1-009-03  INIT+NBLO
    NFSM=   LAN1   TYR=RN
    PS= 00195 LOI= 31      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0013   S7  LAN11      001-04    000-04
1-001-04  INIT+NBLO
    NFSM=   LAN5   TYR=RN
    PS= 11100 LOI= 31      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0015   S7  LAN5COC0  001-05    000-06
1-001-06  INIT+NBLO
    01     01   0016   S7  LAN5COC1  002-02    001-01
1-009-01  INIT+NBLO
    NFSM=   LAN2   TYR=RN
    PS= 00196 LOI= 02      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0017   S7  LAN2       001-06    000-05
1-001-05  INIT+NBLO
    NFSM=   FHLR   TYR=RN
    PS= 01930 LOI= 31      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    01     00   0020   S7  HLR        001-07    000-07
1-001-07  ACTI+NBLO
    NFSM=   LAN8   TYR=RN
    PS= 11200 LOI= 02      D= N   CORR= BASE
    COC  RANC  LSD    SRV  ILS       TSV-VTSV  TSM-VTSM
COM-LRX-IT  ETCS
    00     00   0023   S7  LAN81      001-08    000-08
1-001-08  INIT+NBLO
    01     01   0024   S7  LAN82      002-04    001-04
1-009-04  INIT+NBLO
    NFSM=   LAN9   TYR=RN
    PS= 00200 LOI= 02      D= N   CORR= BASE
  COC RANC LSD   SRV ILS      TSV-VTSV TSM-VTSM  COM-
LRX-IT  ETCS
    00     00   0025   S7  LAN900     001-09    000-10
1-001-10  INIT+NBLO
    TRAITEMENT TFSMIN EXC
@
@ABOMO:
    CEN=1/91-09-16/03   H   39   MN   38/MODIFICATION
DISCRIMINATIONS LIGNE
@ND=96850112,CAT=IAM+CAM2:
    TRAITEMENT TGLAMO ACC
       ND=96850112           NE  =085-00-112
```

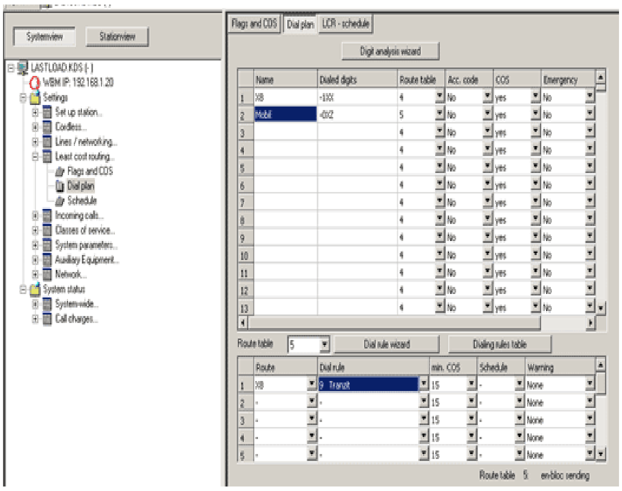
```
        TAX  =  00000005  +  00000000  +  00000000  +
00000000
            TY= NBS2+ NBA30+PRAC3
            CAT= IAM+ CAM2+ NAT2+CLAB4
    TRAITEMENT TGLAMO EXC
@
*   ^R0809/054/'      '/LIBERATION DU TERMINAL
@
*   ^R0809/054/'      '/LIBERATION DU TERMINAL
@
*   ^R0809/054/'      '/LIBERATION DU TERMINAL
@
*   ^R0809/054/'      '/LIBERATION DU TERMINAL
@
@ABOIN:
              CEN=1/91-09-16/04 H 16 MN 57/INTERROG.
CARACTERISTIQUES D'ABONNES
@ND=96850112:
    TRAITEMENT TGLAIN ACC
       ND=96850112           NE  =085-00-112
         TAX  =  00000006  +  00000000  +  00000000  +
00000000
            TY= NBS2+ NBA30+PRAC3
            CAT= IAM+ CAM2+ NAT2+CLAB4
            MAR=ASD
         NDS= 96850212     <96850215
             +96850991     <96850995
    TRAITEMENT TGLAIN EXC
@
```

Fig.13. Explanatory scheme on the recipient phone screen and the number registration in logs (ABOIN).

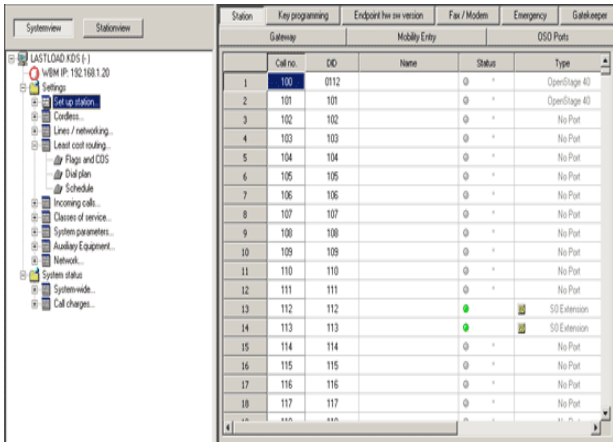
The call is initiated on a phone with the number 0132 140 005 – port 2 (altered port), and the recipient mobile phone shows the altered number (the number set in the switch application), i.e. number 96 850 991 (Fig.14).



a)



b)



c)

Fig.14. Explanatory scheme on initiating the call in case 1 – one unaltered and one altered port.

Case 2: - both ports are altered. The call is initiated on a phone with the number 0132 130 002 – port 1 (altered port), and the recipient mobile phone shows the altered number 96 850 991 (the number set in the switch application) which is registered, and thus featured in the logs (ABOIN). The call is initiated on a phone with the number 0132 140 005 – on port 2 (altered port), and the recipient mobile phone shows the altered number (the number set in the switch application), i.e. the number 96 850 991 (Fig.15), which was also noticed in the logs (ABOIN).

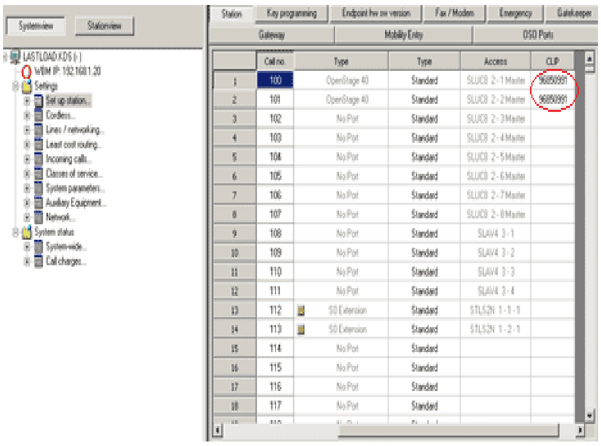


Fig.15. Explanatory scheme on initiating the call in case 1 – one unaltered and one altered port.

Case 3: - both ports altered (Fig. 16). The call is initiated on a phone with the number 0132 130 002 – port 1 (altered), and the recipient mobile phone shows the altered number 96 850 991 (the number set in the switch application) which was also noticed in the logs (ABOIN) The call is initiated on a phone with the number 0132 140 005 – port 2 (altered), and the recipient mobile phone shows the altered number (the number set in the switch application), i.e. the number 96 850 995, which was also seen in the logs (ABOIN).

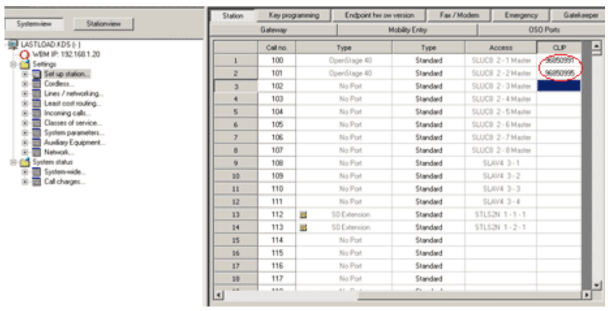


Fig.16. Explanatory scheme on initiating the call in the case when both ports are altered.

3 Traffic and voice transmission, quality analysis

3.1. The carrier operator and/or service provider has routed into the dominant operator's network international traffic and/or national traffic originated in other networks than its own

Following the ANCOM decision on the standardization of national and international rates (2007), the carrier operator and/or service provider has reduced traffic (starting in the 36th month, Fig. 17), but only for a short time period [2].

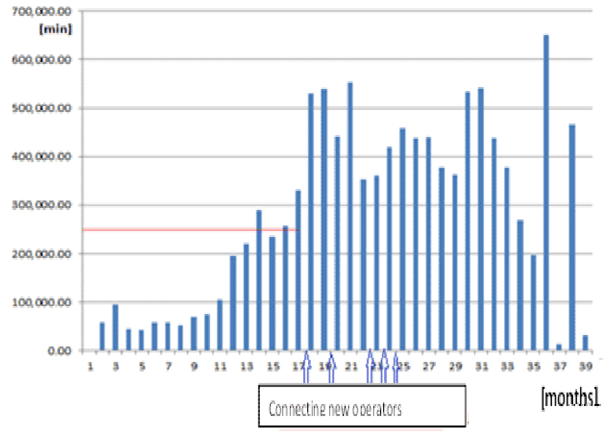


Fig.17. Traffic trends during the connection of the carrier operator and/or service provider with the dominant operator and the moment of clients' connection to the carrier operator and/or service provider network and traffic increase in the dominant operator's network.

a) One argument for the major increase in traffic compared to the amount established in the agreement (30,000 minutes/month) is given by the small number of subscribers declared by the carrier operator and/or service provider in relation to the dominant operator (about 100 times). Thus, during traffic monitoring by the dominant operator the given traffic values (Fig. 18) are real, i.e. in accordance with the number of subscribers declared by the carrier operator and/or service provider.

b) During the period under review, according to the existing agreement between the carrier operator and/or service provider and another dominant operator on the market, prepaid cards and smartcalls are also used by subscribers of those networks, the service operator offering conditions for routing traffic for subscribers of the other dominant operator and international traffic to subscribers of the dominant operator.

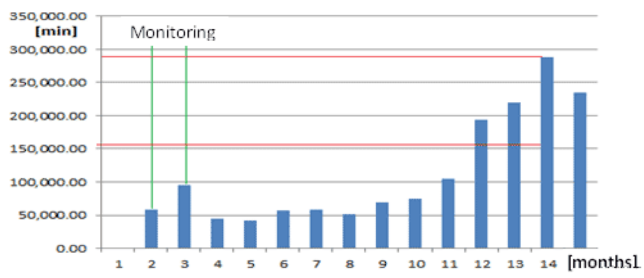


Fig.18. Traffic trends during monitoring by the dominant operator.

The carrier operator and/or service provider would use card count-type cards by which the subscriber, dialling a phone number, after entering the account number, could access any other number in the telecommunication networks connected to the card issuer (card issuer identifier code), these short numbers being called carrier selection code. Such a card, during its validity, could be used by any subscriber. In other words, routing calls to the dominant operator was also performed by use of prepaid cards by subscribers of other networks than that of the carrier operator and/or service provider, but with direct access to the network, with the consequence of rates applied between the two operators (carrier operator and/or service provider – dominant operator), for calls made. These rates were lower than those applied for direct calls between operators of those subscribers' networks. The dial numbers of the carrier operator and/or service provider, from which different subscribers of the dominant operator network were called, are the same over a long period of time, which shows that they were used for routing calls from other telecommunication networks to the dominant operator's network. By using the smartcall service, a subscriber can dial the desired number after previously entering a "short" phone number. The caller's ID may be replaced with another, pertaining to the carrier operator and/or service provider network, so that when reaching destination, the caller may be considered to be the carrier operator and/or service provider's own subscriber.

c) Use of traffic routing through VoIP protocol (allows changing caller identity upon entering another network), by the carrier operator and/or service provider, illustrates the fact that the carrier operator and/or service provider will be able to route national and international traffic to the dominant operator's network. Thus, after converting voice signals and preparing the packets (Fig. 19), upon entry into the operator's network operator and/or service provider, the true identity of the caller is replaced with an ID (caller number) pertaining to the transit network (of the carrier operator and/or service provider), and upon destination/reception the call appears as coming from the transit network and not the real one. Although ITU-T recommendations require just one signalling change for establishing communications in a national area, it can be noted that two changes are made in the signalling system, namely, into and out of the carrier operator and/or service provider network.

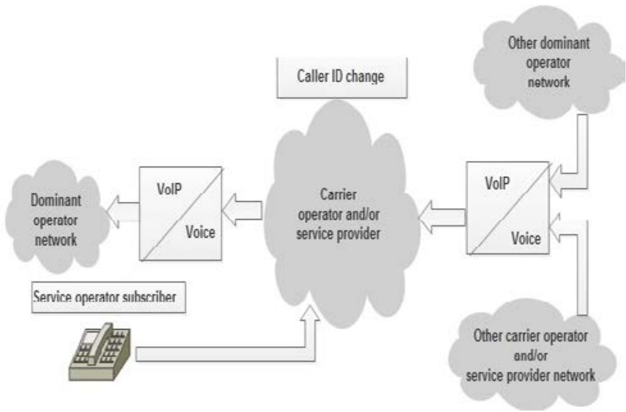


Fig.19. Explanatory scheme on communication mode with ID change (another dial number) via a backhaul network.

d) During the period under review, the carrier operator and/or service provider concluded a large number of interconnection agreements with various companies providing telecommunication services, concentrating and routing their traffic to subscribers of the dominant operator network (Fig. 20).

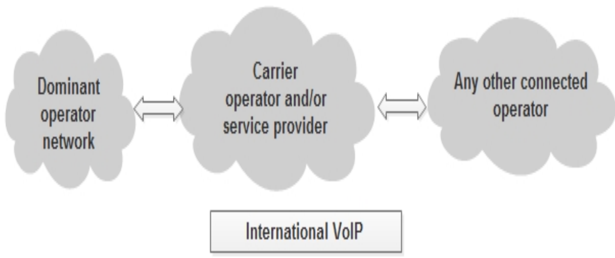


Fig.20. Diagram of connections for routing national and international traffic.

Routing other operators' traffic to the dominant operator network is also apparent from Fig. 17, where traffic increases are highlighted every time a new network connects to the carrier operator and/or service provider network [8].

e) In addition to the amount of incoming traffic (Fig. 21a), the dominant operator also monitored the average duration of incoming calls (Fig. 21b) tolerating the behaviour of the carrier operator and/or service provider.

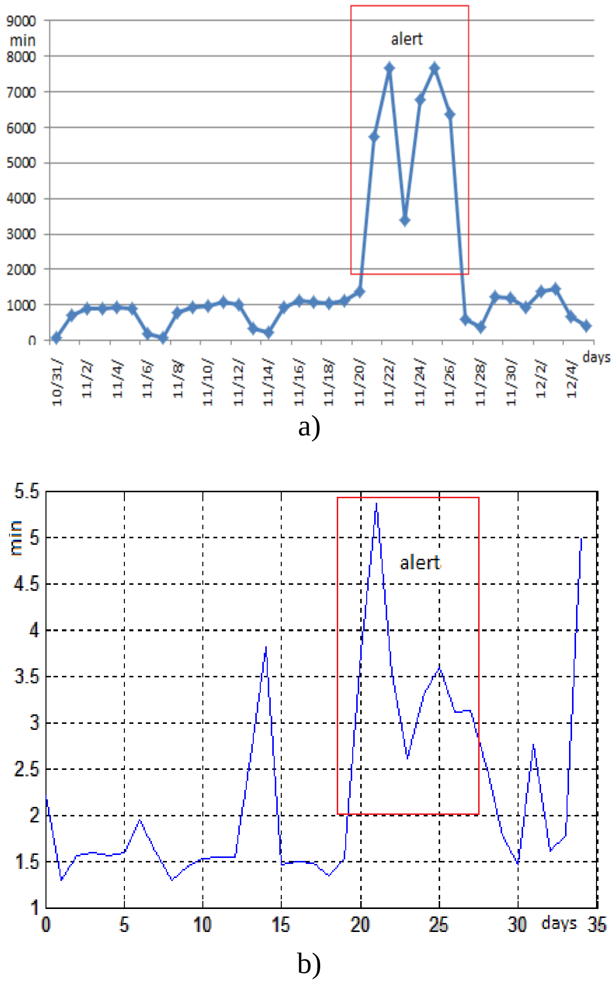


Fig.21. Explanatory scheme on incoming traffic (a) and average duration of calls (b).

Thus, there is a sudden increase in the average duration of calls from 1.5 minutes in the first 20 days to more than 4 minutes in the next 5 days, which indicates that the carrier operator and/or service provider began to route additional traffic (possibly international traffic) from the one originated in the home network to the dominant operator network.

3.2. The quality of VoIP communications, the relationship between the carrier operator and/or service provider and the dominant operator

a) Quality parameters established for national and international communications, valid for both the carrier operator and/or service provider and the dominant operator, are [17]:

- the link upload domain, necessary to enable the carrier operator and/or service provider to use VoIP [18], with values ranging between 0.2 - 0.4 Erlang [19]; exceeding the upper threshold involves increasing waiting “queue” and thus loss of packets;

- the route loading indicator, ranging between 10-90% Erlang B, determines the probability for a call to be blocked; very high values will highlight the poor quality of equipment and software used by the carrier operator and/or service provider; reaching the upper limit cannot be accepted even in the case of route transport capacity falling below 20%;

- the percentage of erroneous seconds < 1.4% (14 missed calls in 1,000); reaching the upper limit is not acceptable for national/international traffic;

- Answer Seizure Ratio (ASR)> 50%, being dependent on the subscriber’s presence in the network.

The above values, listed in the studied VoIP communications, do not meet the quality requirements desired by subscribers and do not fall into the general characteristics of voice communications, namely, ID change, interruptions and lengthy shutdowns.

b) The above-mentioned parameters as quality indicators are insufficient to ensure that the traffic routed through and originated on the carrier operator and/or service provider network maintains its quality levels upon arrival to the dominant network. VoIP-based service quality was contested by many international operators. Quality problems have only reached acceptable levels in 2008, by implementing Internet Protocol version 8 (IPv8), which allowed controlling packets carrying voice information and reducing the number of lost packets. Full implementation of this version, by all users, was due to be completed in 2013, which never happened.

c) As the transmission of voice packets can be affected by lack of synchronization occurring between the telecommunication network and telecommunication terminal adapters, resulting in loss of packets, VoIP communications are regarded as having a “best effort” quality level. However, if the bandwidth is much larger than that of an E1 stream (>2Mbps), then communications are not affected by packet loss, and transmission performance is “premium rate”.

d) The existence of two changes in the signalling system drastically affects the quality of communications. In this respect, reference is made to the poor quality of communications before 1989, in Romania, when analogue communication systems were used, requesting two changes in the signalling system (Rotary [20] and Pentaconta technologies).

4 Conclusion

As conclusion, we demonstrated that the increasing trends of traffic over a short period of time (e.g. 20 days) resulted in the dominant operator taking additional measures to monitor traffic and moderate its growth. This fact caused discontent in the carrier operator and/or service provider, the latter mentioning that the traffic of their subscribers is a process that evolves in a probabilistic manner which cannot be controlled, and therefore requiring a new interconnection agreement for national and international call routing to the dominant operator's network.

Performing simulations is required for the numerical analysis of the conflict situation arising between telecommunication operators. Thus, it has been observed that there is a possibility that numbers in a VoIP network be converted into another numbering plan based on pre-established settings. If the call source number does not correspond to a specific predefined pattern, the dominant operator may convert that number into one of its own network. Using a prefix to access the dominant operator network can result in conversion of the call source number to match the numbers in that network by removing the prefix and changing it. For the case presented in our paper, the quality assesment of VoIP communications, in the connection between the dominant operator and the carrier operator and/or service provider, demonstarted that the services did not meet the quality requirements desired by subscribers and did not fall into the general characteristics of voice communications, namely, ID change, interruptions and lengthy shutdowns.The quality standard for this type of communication was achieved only after the implementation of Internet Protocol version 6, started in 2009 and expected to be completed in 2013, although currently in Romania there are communication networks using old versions of the Internet Protocol, namely IPv4.

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