Investigate common work of IP software phone systems and PSTN equipment

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Abstract - **This paper discusses common workbetweenPSTN andsoftware – based VoIP phone systems. The experimental part is basedon real working system including specialized interface modules and gateways.**

Keywords–**VoIP, FXO, FXS , Phone systems,SIP signalizations**

I. INTRODUCTION

The purpose of this experiment is to create a interconnection of IP-based phone systems with conventional telephone equipment in already build telecommunication network. Investigated system allows connection to various outside PSTN, ISDN and other networks. IP telephone systems like TRIXBOX and ELASTIX are used. The Network consists various conventionaland IP, software and hardware phones[1].

For realization of channelswitching commutation and packet switching commutation two methods are applied: using a hybrid interface card with four FXO ports and one FXS port and by connecting input-output device (gateway)[2].

II. EXPERIMENTAL RESULTS

2.1. Investigate and analyze the common work of conventional telephone equipment with IP software telephone system "Elastix"[3].

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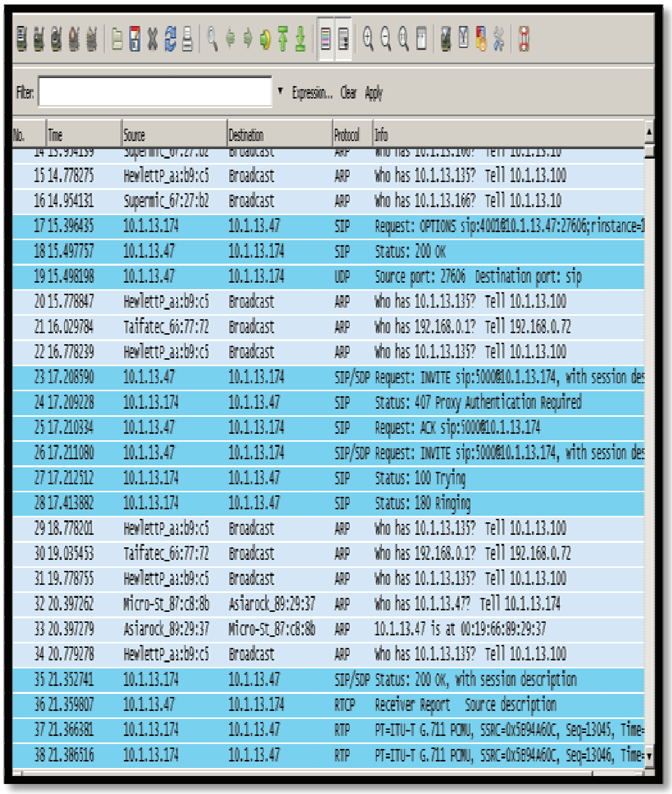
For normal use of the system a phone number is necessary to be created. This number is part of the phone system and is responsible to port on the interface module.

When initially connect two phones stream of packets contains not only conversation of the subscribers, but different SIP signalizations between them (Fig.2.1).



No.	Time	Source	Destination	Protocol	Info
15	14.778275	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
16	14.954131	Supermic_67:27:b2	Broadcast	ARP	Who has 10.1.13.166? Tell 10.1.13.10
17	15.396435	10.1.13.174	10.1.13.47	SIP	Request: OPTIONS sip:4001@10.1.13.47:27606;rinstance=1
18	15.497757	10.1.13.47	10.1.13.174	SIP	Status: 200 OK
19	15.498198	10.1.13.47	10.1.13.174	UDP	Source port: 27606 Destination port: sip
20	15.778847	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
21	16.029784	Taifatec_66:77:72	Broadcast	ARP	Who has 192.168.0.1? Tell 192.168.0.72
22	16.778239	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
23	17.208590	10.1.13.47	10.1.13.174	SIP/SDP	Request: INVITE sip:5000@10.1.13.174, with session des
24	17.209228	10.1.13.174	10.1.13.47	SIP	Status: 407 Proxy Authentication Required
25	17.210334	10.1.13.47	10.1.13.174	SIP	Request: ACK sip:5000@10.1.13.174
26	17.211080	10.1.13.47	10.1.13.174	SIP/SDP	Request: INVITE sip:5000@10.1.13.174, with session des
27	17.212512	10.1.13.174	10.1.13.47	SIP	Status: 100 Trying
28	17.413882	10.1.13.174	10.1.13.47	SIP	Status: 180 Ringing
29	18.778201	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
30	19.035453	Taifatec_66:77:72	Broadcast	ARP	Who has 192.168.0.1? Tell 192.168.0.72
31	19.778755	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
32	20.397262	Micro-St_87:c8:8b	Asiarock_89:29:37	ARP	Who has 10.1.13.47? Tell 10.1.13.174
33	20.397279	Asiarock_89:29:37	Micro-St_87:c8:8b	ARP	10.1.13.47 is at 00:19:66:89:29:37
34	20.779278	HewlettP_aa:b9:c5	Broadcast	ARP	Who has 10.1.13.135? Tell 10.1.13.100
35	21.352741	10.1.13.174	10.1.13.47	SIP/SDP	Status: 200 OK, with session description
36	21.359807	10.1.13.47	10.1.13.174	RTCP	Receiver Report Source description
37	21.366381	10.1.13.174	10.1.13.47	RTP	PT=ITU-T G.711 PCMU, SSRC=0x5B94A60C, Seq=13045, Time=
38	21.386516	10.1.13.174	10.1.13.47	RTP	PT=ITU-T G.711 PCMU, SSRC=0x5B94A60C, Seq=13046, Time=

Fig.2.1. SIP signalizations between subscribersinVoIP conversation

VoIP signal recorded in this study is presented with VoIP analyzer (Fig.2.2)[4].

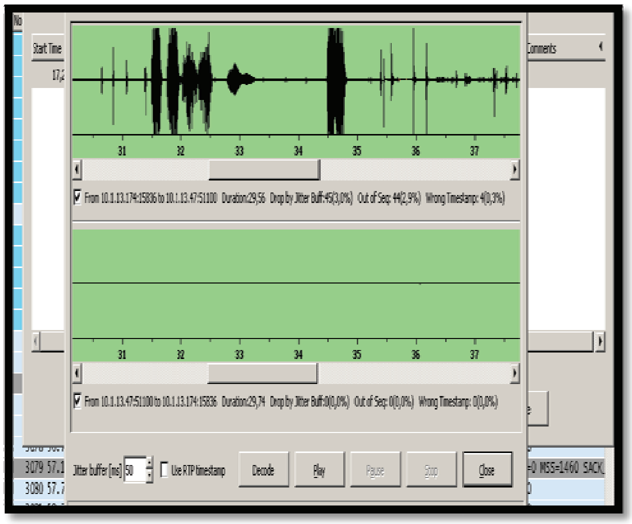


Fig.2.2. Realized VoIP call

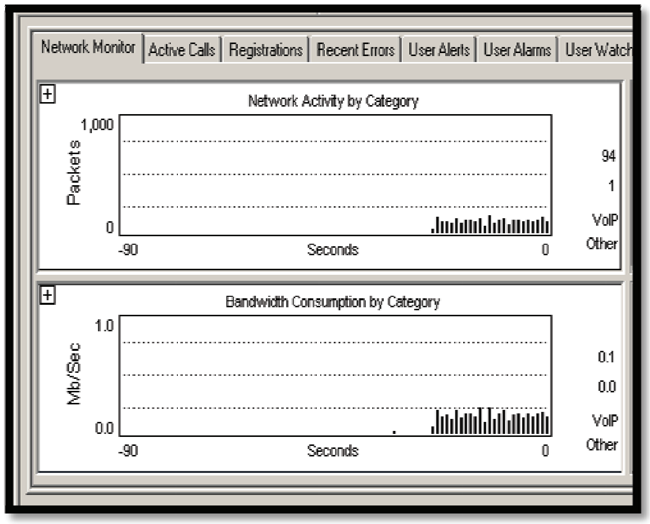


Fig.2.4. Parameters of the conversation

RTP, TCP, and SIP protocols are monitored in the study (fig.2.3).

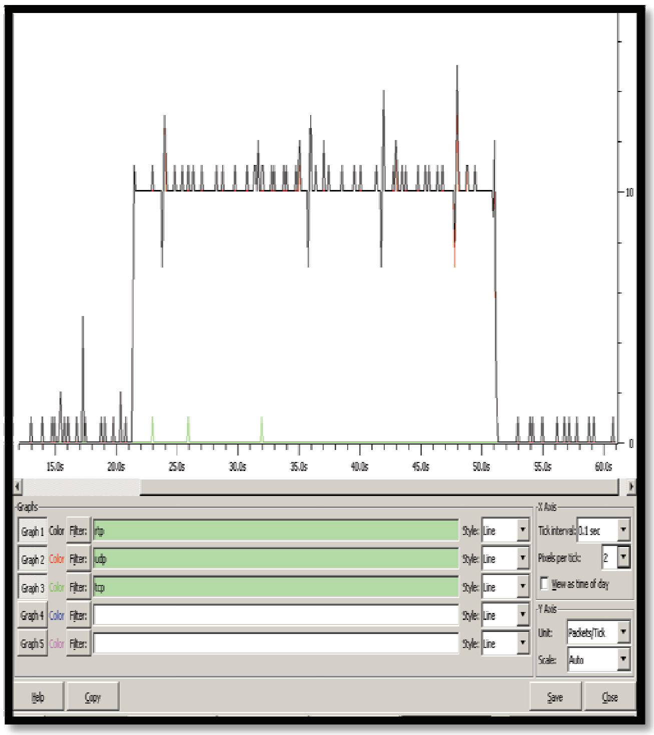


Fig.2.3. Time dependence of VoIP protocols

Main parameters of the conversation like network activity by category, bandwidth consumption by category, network activity by protocol and bandwidth consumption by protocol are shown in Fig. 2.4

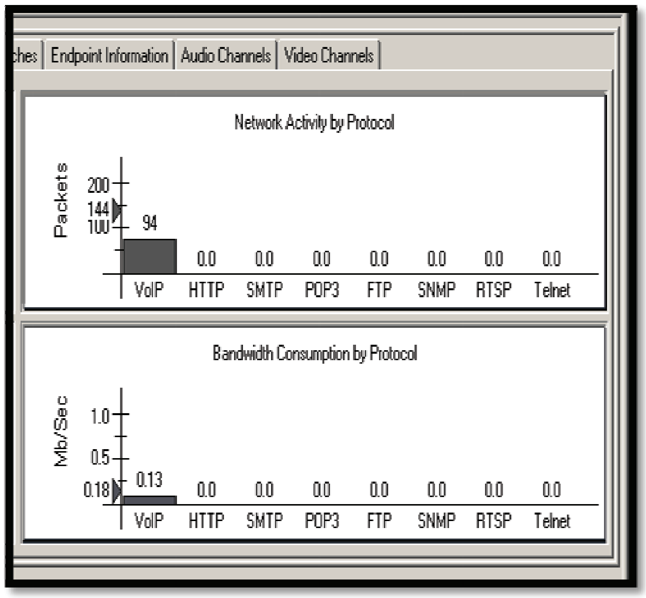


Fig.2.5Parameters of the conversation part 2

Along with that the status of the investigated system is monitored. Fig.2.5 shows profiles of the network, which tracks current employment of bandwidth and the packet employment of the network.

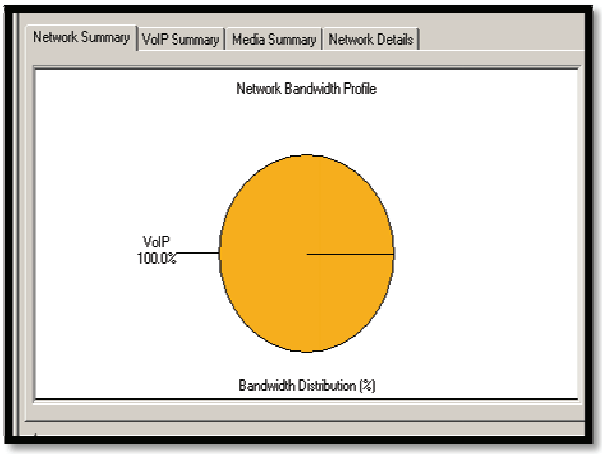


Fig.2.6 Parameters of the network

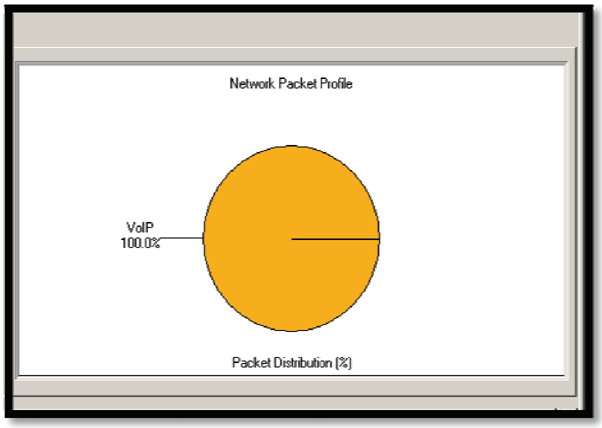


Fig.2.7Parameters of the network part 2

Information about ongoing conversations in the study is shown in fig.2.6:

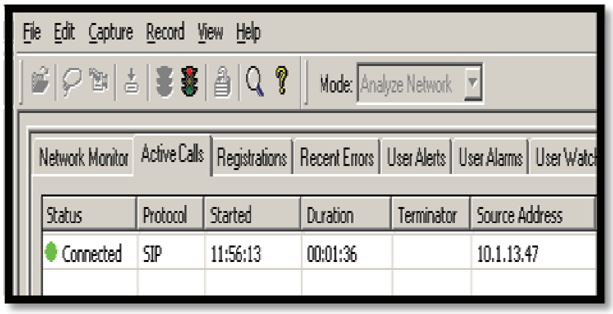


Fig.2.8Information for current conversations

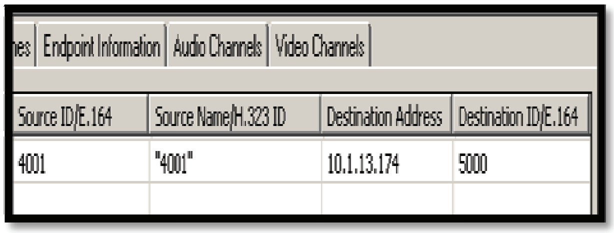


Fig.2.9 Information for current conversations

2.2. Investigate and analyze the common work of interface commutation modules and PSTN network

In this study connections are created between PSTN network, interface hybrid card and the input-output module (Sangoma B600 and Micronet SP5014)[5].

In the first case, the Sangoma card is part of Elastix phone system, therefore it is necessary some important settings to be configured in the phone system for the proper functioning of the module[6].

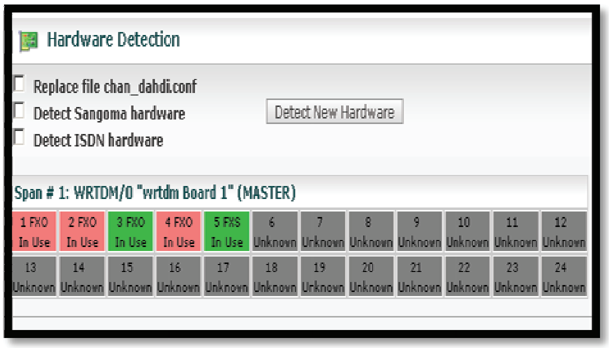


Fig.2.10. FXOport configuration

The second way of connection to the PSTN network in this experiment is using the gateway.

The device is connected to one of the telephone systems with IP address: 10.1.13.100. Phone numbers are chosen for the ports (two FXO and two FXS ports) and the relevant settings are applied (fig.2.8 and fig. 2.9):

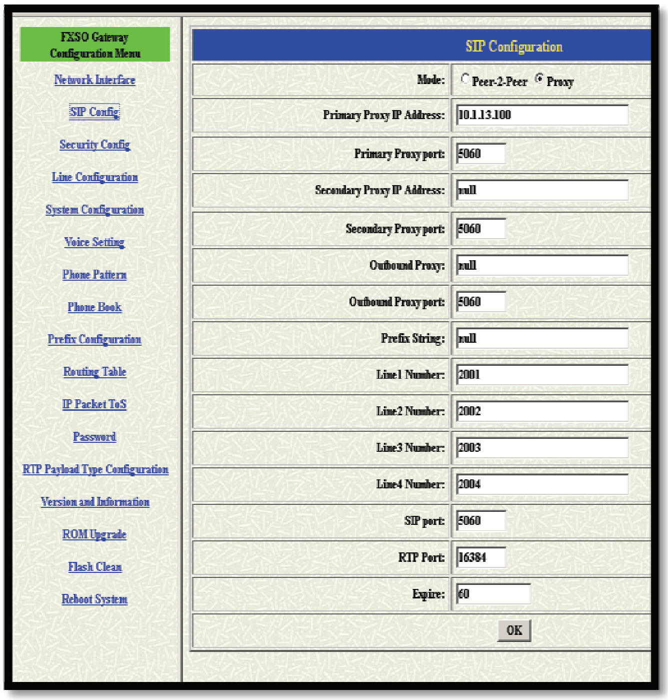


Fig.2.11.Sip configuration

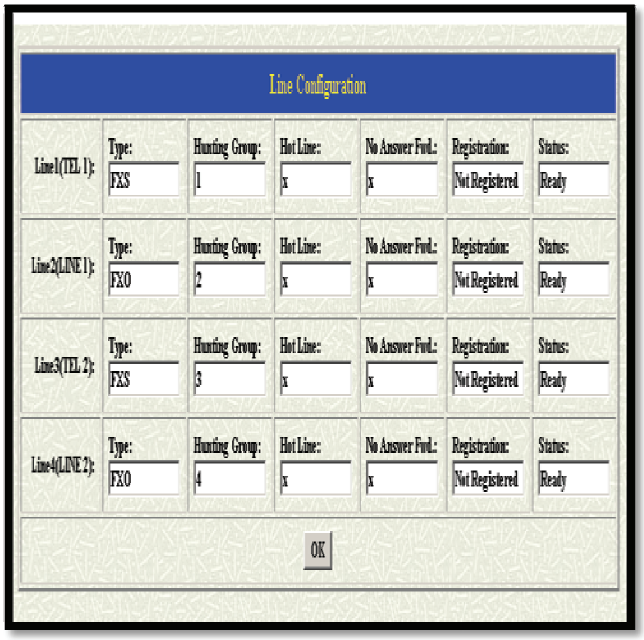


Fig.2.12. Ports settings

External connections are provided through one of the FXO entrances of the device andthereby a connection is created between PSTN and IP telephone systems.

III.CONCLUSION

The conducted experiments show successful collaboration of IP-based systems, PSTN networks and the connected to them hardware and software communication devices. The results confirm the effectiveness of the established communication connections and ensure the quality of conversations. The realized system requires further study in obtaining parameters to ensure quality teletrafficparameters and QoS.

REFERENCES

[1]DavidKelly, CullenJennings, LaanDang,Practical VoIP Using Vocal, 2010

[2]TedUelingford,VoIP Hacks - Tips& Tools for Internet Telephony, 2005

[3] Kerry Garrison, Trixbox CE 2.6: Implementing, managing, and maintaining an Asterisk-based telephony system, Packt Publishing, 2009

[4]BruceStuart, Speaking about VoIP, 2011

[5]Тед Уелингфорд,Преход към VoIP, 2006

[6]BenSharifен Шариф, Elastix Without Tears, 2010