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).

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	19 15.939159	Super Inc. or . er. uz	Bruducasu	RP	
	15 14.778275	HewlettP_at:b9:c5	Broadcast	AKP 	who has 10.1.13.135? Tell 10.1.13.100
	16 14.954131	Supermic_67:27:b2		ARP	who has 10.1.13.166? Tell 10.1.13.10
	17 15,396435	10.1.13.174		SIP	Request: OPTIONS sip:4001010.1.13.47:27606;rinstance=1
	18 15.497757			SIP	Status: 200 OK
_	19 15.498198	10.1.13.47		UDP	Source port: 27606 Destination port: sip
	20 15.778847	HevlettP_arcb9:c5	Di Concente	жp	who has 10.1.13.135? Tell 10.1.13.100
	21 16.029784	Taifatec_6:77:72		жp	who has 192.168.0.1? Tell 192.168.0.72
	22 16.778239	HevlettP_a:b9:c5		ЖР	who has 10.1.13.135? Tell 10.1.13.100
	23 17.208590	10.1.13.47			Request: INVITE sip:5000010.1.13.174, 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		SIP	Request: ACX sip:5000010.1.13.174
	26 17.211080	10.1.13.47	10.1.13.174	SIP/SOP	Request: INVITE sip:5000010.1.13.174, with session des
	27 17.212512	10.1.13.174	10.1.13.47	SIP	Status: 100 Trying
	28 17.413682	10.1.13.174	10.1.13.47	SIP	Status: 180 Ringing
	29 18.778201	HewlettP_acb9:c5	Broadcast	ARP	who has 10.1.13.135? Tell 10.1.13.100
	30 19.035453	Taifatec_65:77:72	Broadcast	ARP	who has 192.168.0.1? Tell 192.168.0.72
	31 19.778755	HewlettP_a: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	Asianock_89:29:37	Micro-St_87:c8:8b	жp	10.1.13.47 is at (0:19:66:89:29:37
	34 20.779278	HewlettP_aa:b9:c5	Broadcast	AKP	who has 10.1.13.135? Tell 10.1.13.100
	35 21.352741	10.1.13.174	10.1.13.47	SIP/SOP	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=0x5694A60C, Seq=13045, Time
_	38 21.386516	10.1.13.174	10.1.13.47	RTP	PT=ITU-T G.711 RONU, SSRC=0x5B94A60C, Seq=13045, Time=

Fig.2.1. SIP signalizations between subscribersinVoIP
conversation

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

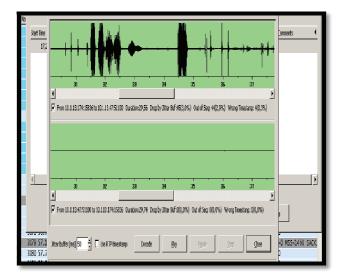


Fig.2.2. Realized VoIP call

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

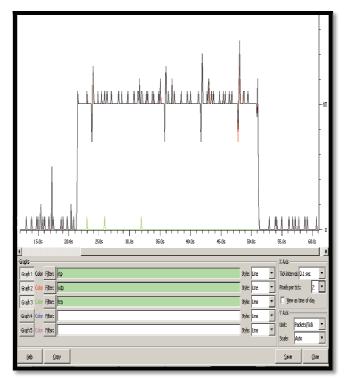


Fig.2.3. Time dependence of VoIP protocols

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

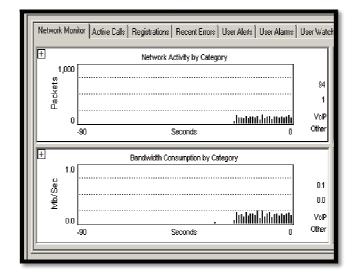


Fig.2.4. Parameters of the conversation

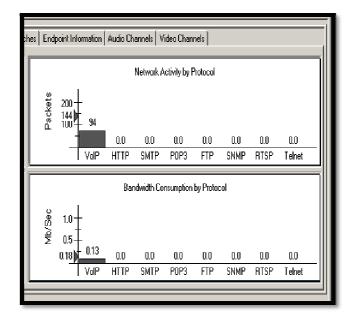


Fig.2.5Parameters of the conversation part 2

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.



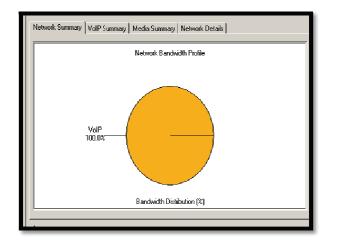


Fig.2.6 Parameters of the network

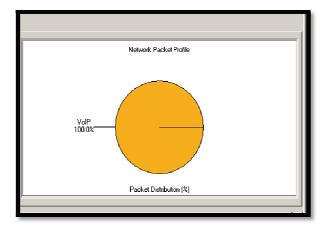


Fig.2.7Parameters of the network part 2

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

Ē	le <u>E</u> dit <u>C</u> apture	<u>R</u> ecord <u>V</u>	<u>í</u> iew <u>H</u> elp			
	2 2 % 4		🏻 🛛 🗶	Mode: Analy	ze Network	-
	н. ты х	Active Colle	do na f	ուն	i vitu	a fu sau
	Network Monitor	ADIVECTIS	Fiegistrations	Hecent Errors L	User Alerts Us	ser Alarms User Watch
	Status	Protocol	Started	Duration	Terminator	Source Address
	Connected	SIP	11:56:13	00:01:36		10.1.13.47

Fig.2.8Information for current conversations

nes Endpoint Informati	on 🛛 Audio Channels 🗍 Video C	hannels	
Source ID/E.164	Source Name/H.323 ID	Destination Address	Destination ID/E.164
4001	"4001"	10.1.13.174	5000

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].

🗆 Repl	ace file (chan_dal	hdi.conf								
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1 FX0	2 FXO	3 FXO	4 FXO	5 FXS	6	7 Urknown 19		9 Unknown 21			

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):

		$N_{2}(E) = F(N_{1} = 2) (1) (2EE)$
FXSO Gateway Configuration Menu		SIP Configuration
Network Interface	Mode:	C Peer-2-Peer @ Proxy
SIP Config	Primary Proxy IP Address:	01.13.100
Security Config	Primary Proxy port:	5060
Line Configuration	Secondary Proxy IP Address:	nil 🛛
System Configuration	Secondary Proxy port:	5060
<u>Voice Setting</u> Phone Pattern	Outbound Prexy:	
Phone Book	Outhound Proxy port:	5060
Prefix Configuration	Prefix String:	nu
Routing Table	Linel Number:	2001
P Packet ToS	Line2 Number:	2002
Password	Line3 Number:	2003
RTP Payload Type Configuration	Line4 Number:	2004
Version and Information	SIP port:	5060
<u>ROM Upg rale</u> Flash Clean	RTP Pert:	16384
Reboot System	Expire:	60
经过了这个社会	720-540-16-5720-572	OK
公式在这方面		到小学会大的知道
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Fig.2.11.Sip configuration

Linel(TEL I);	Type:	Hunting Group:	Hot Line:	No Answer Find.:	Registration:	Status:
	FXS	1	x	x	Not Registered	Ready
Line2(LINE I):	Type:	Hunting Group:	Hot Line:	No Answer Fint.:	Registration:	Status:
	FX0	2	x	X	Not Registered	Ready
	Type:	Hunting Group:	Hot Line:	No Answer Fud.:	Registration:	Status:
ne3(TEL 2):	FXS	3	X	x	Not Registered	Ready
	Type:	Hunting Group:	Hot Line:	No Answer Find.:	Registration:	Status:
e4(LINE 2):	FX0	4	X	x	Nt Registered	Ready

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.

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