Interface Modules of the Voice Protocols Using Software

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ABSTRACT: *We in this work, have described the VoIP systems that use the PSTN and software. The interfaces in the voice protocols and the modules contained in them are tested. Besides, we have used the voice gateways to describe the process. The experimental trials and the results are presented.*

Keywords: VoIP, FXO, FXS, Phone systems, SIP signalizations

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1. 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 conventional and IP, software and hardware phones [1].

For realization of channel switching 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].

2. Experimental Results

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

When initially connect two phones stream of packets contains not only conversation of the subscribers, but different SIP signalizations between them (Figure 1).

VoIP signal recorded in this study is presented with VoIP analyzer (Figure 2)[4].

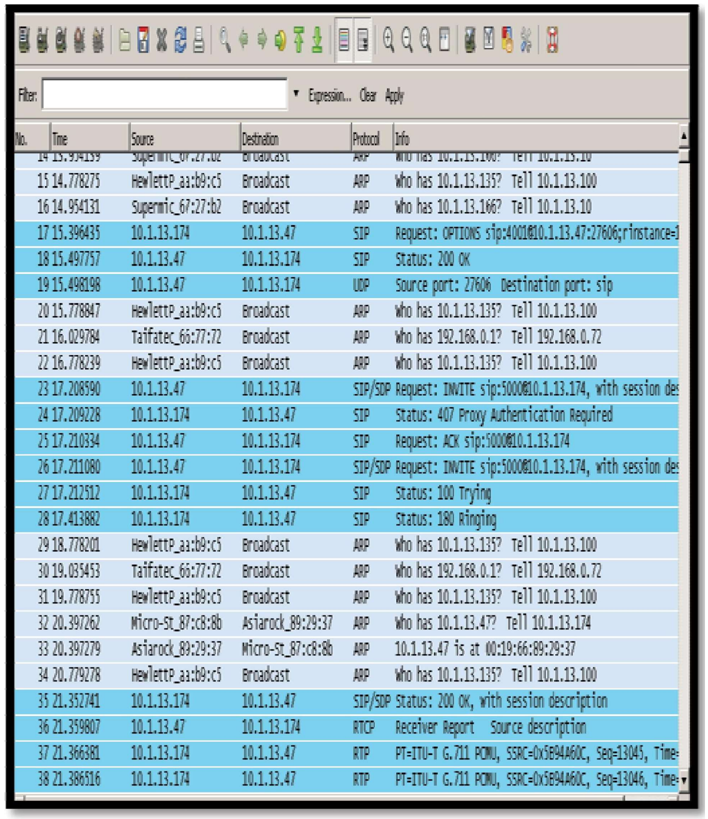

Figure 1. SIP signalizations between subscribersin VoIP conversation

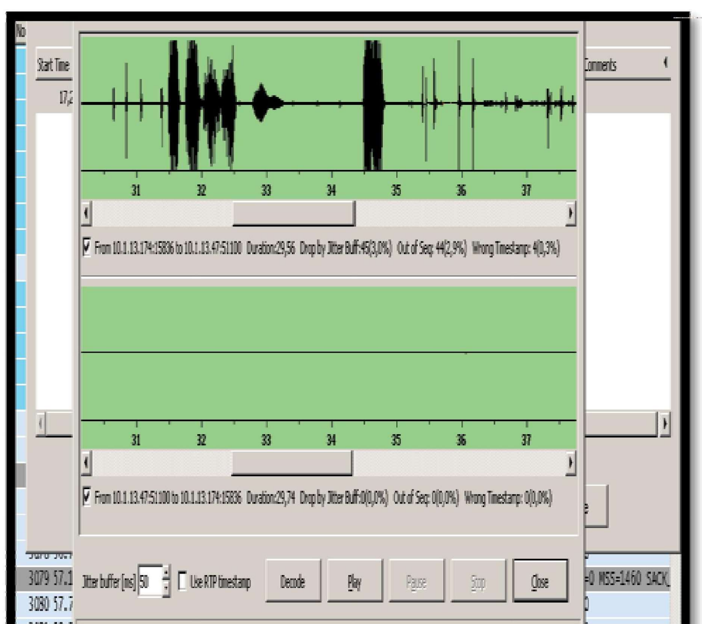

Figure 2. Realized VoIP call

RTP, TCP, and SIP protocols are monitored in the study (figure 3).

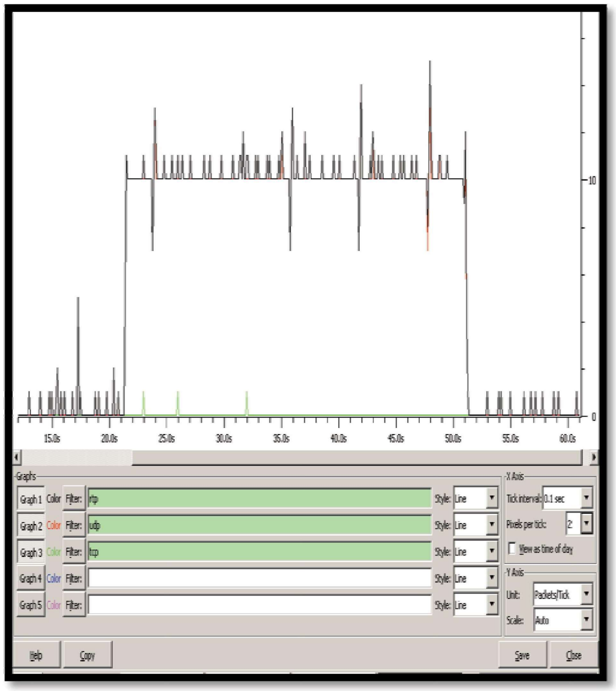

Figure 3. Time dependence of VoIP protocols

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 Figure 4

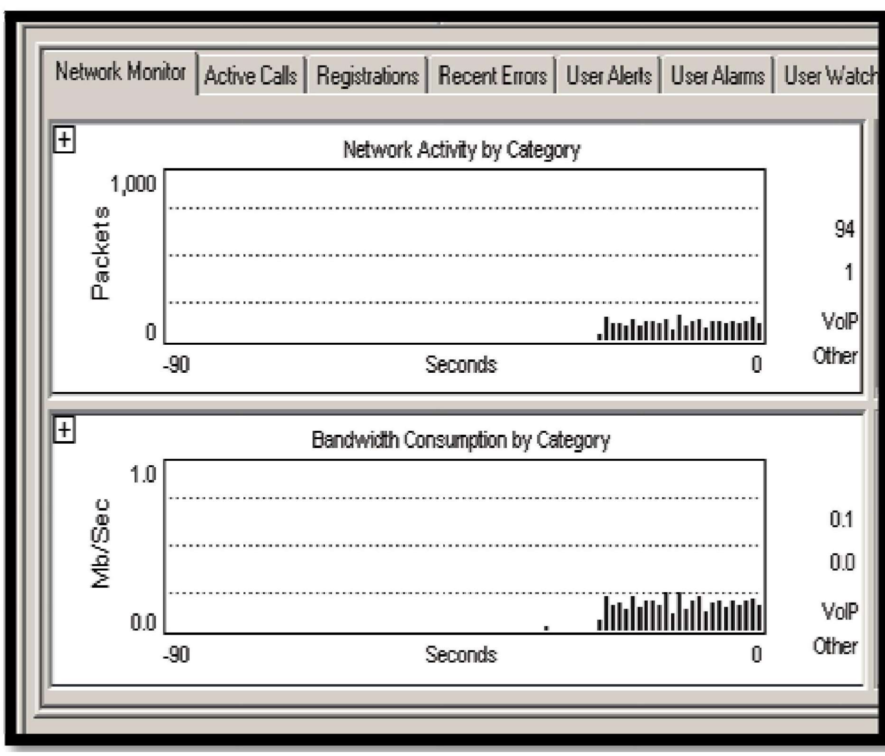

Figure 4. Parameters of the conversation

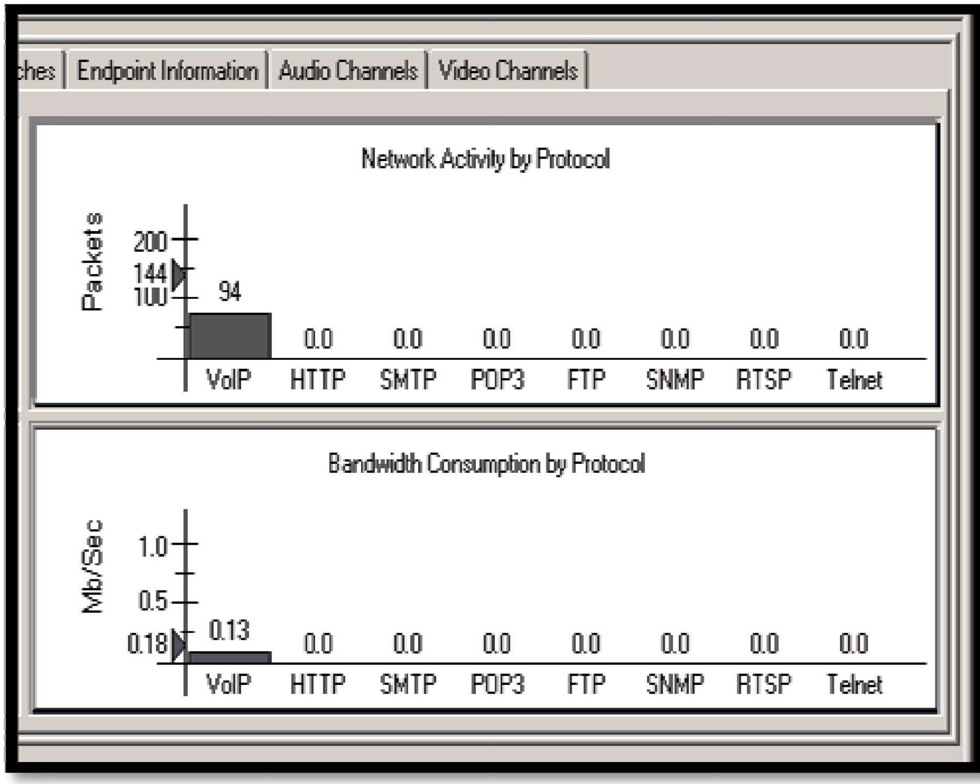

Figure 5. Parameters of the conversation part 2

Along with that the status of the investigated system is monitored. Figure 5 shows profiles of the network, which tracks current employment of bandwidth and the packet employment of the network.

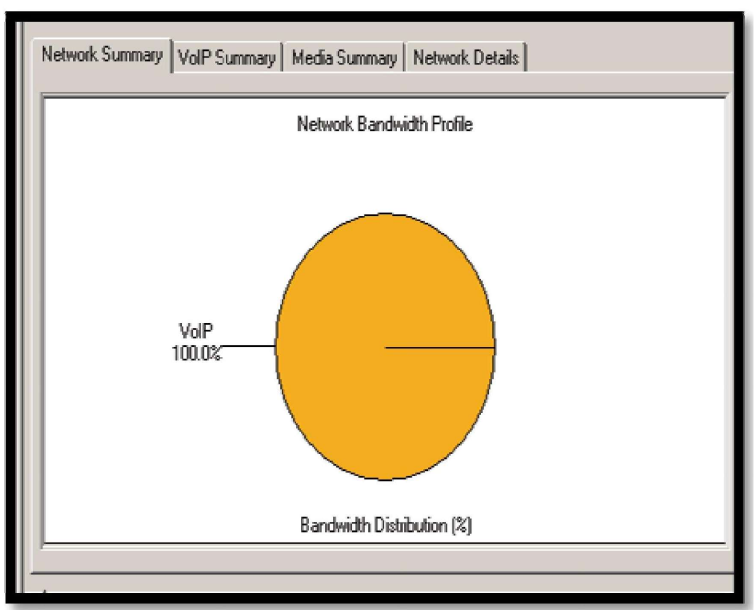

Figure 6. Parameters of the network

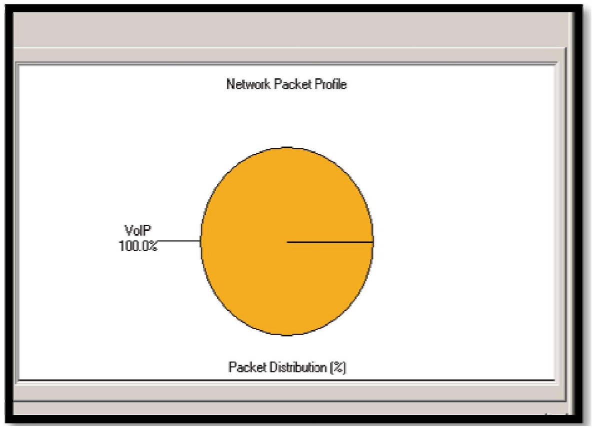

Figure 7. Parameters of the network part 2

Information about ongoing conversations in the study is shown in figure 6:

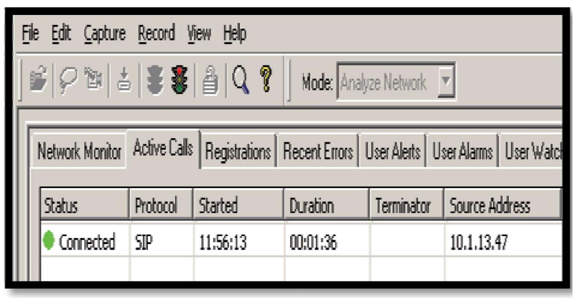

Figure 8. Information for current conversations

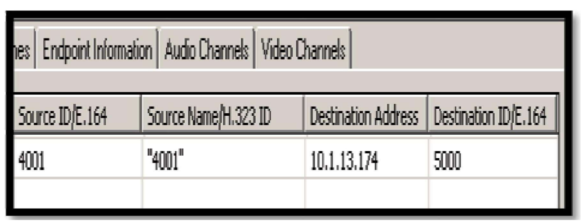

Figure 9. Information for current conversations

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].

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 (figure 8 and figure 9):

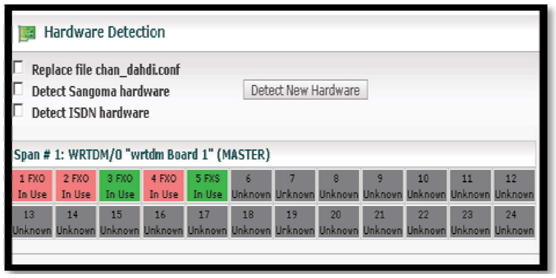

Figure 10. FXOport configuration

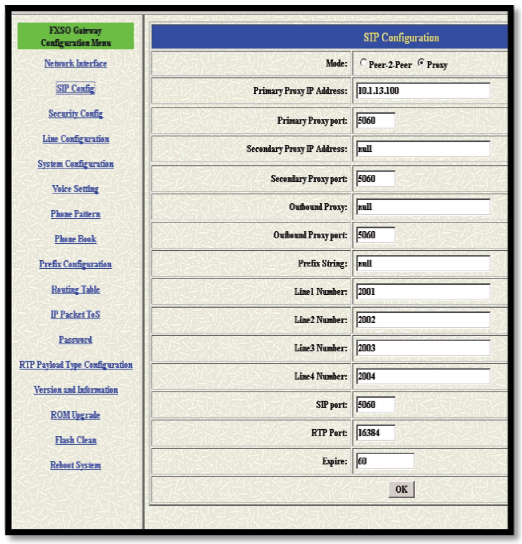

Figure 11. Sip configuration

Figure 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.

3. 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

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