Comparative Analysis of VoIP Protocols Used in CATV Networks

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***Abstract* – In the present paper there has been made comparative analysis between VoIP protocols used in the CATV networks according to their operation algorithm. There have been presented the requirements for network configuration supports SIP and H.323 protocols. Furthermore, there has been made an estimation of the effectiveness of the MGCP and MEGACO protocols in case of use in the contemporary CATV networks. There has been described the applicability to these protocols within the imposed PacketCable standard.**

***Keywords* – VoIP protocols, SIP vs. H.323, MEGACO vs. MGCP, IP telephony, PacketCable.**

I. INTRODUCTION

Distribution of voice services over internet protocol achieves large implementation in modern CATV networks. Subscribers must obtain multimedia terminal adapter (MTA) in order to able use VoIP service. MTA converts the analog voice information into IP packets and vice versa. The basic problems in the VoIP implementation over the CATV/HFC networks are as follows: packets delay, accumulating phase noise (jitter), large expenses for network realization and support, necessity to support new services as well as servicing large number of subscribers.

The quality of the distributed VoIP service depends on the internet connection speed as well as on the VoIP protocols and audio codec which have been used. In general, at present there are four VoIP protocols: SIP, H.323, MGCP and MEGACO. The main purpose of the present article is the comparison of these VoIP protocols according to their operation algorithm as well as to specify the most appropriate method for implementation of the above mentioned service over CATV networks.

II. REQUIREMENTS TO NETWORK CONFIGURATION WHICH SUPPORTS SIP AND H.323 PROTOCOLS

Both protocols are very similar in their basics. SIP architecture is similar to client-server HTTP protocol [1]. Requests have been generated by client and sent to server which processes them and sent back to the client as reply. Each request and its reply is “transaction”. SIP protocol uses INVITE and ACK messages, which determine the process of opening a reliable transmission channel for the call control messages.

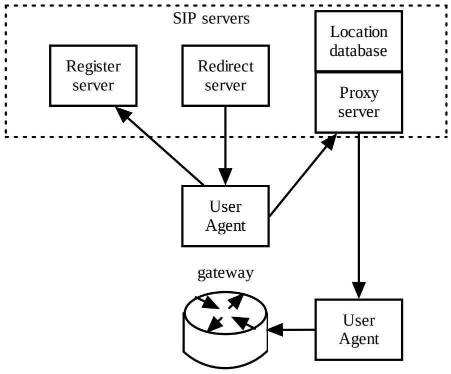

Fig. 1

The network, which supports SIP protocol consists of two components – user agent and network servers (Fig. 1). The user agent is a final system which operates by user’s side, and serves to initialize SIP requests. By server’s side it receives requests and returns acknowledgments to the client. In that system three network servers are implemented:

Register server which main purpose is to renew the current client location.

Proxy server which receives requests and retransmits them to another server that has more information about the client searched (location database). Its main characteristic is that it works either as a client or as server and transmits requests to next servers on client’s behavior.

Redirect server which function is to receive request, to locate server which has more information about the location of the client searched and to return client address. Unlike the proxy server, redirect server does not emit its own SIP requests.

SIP protocol works as follows: When a calling side accomplishes SIP call it must find an appropriate server and send a request to that server. It can be done direct through proxy server or indirect through redirect server. When client turns on SIP device (personal computer, phone or other VoIP device) he registers itself in the register server. That registration information is constantly refreshed and sent to register server and/or proxy server.

The H.323 protocol is a specific “umbrella” type i.e. it is not a detached protocol but it defines how other protocols can be used. The H.323 protocol specification defines voice, information and video traffic through IP based networks. The H.323 protocol realization requires four logical components:

“Terminal”, which is an end point for the network, ensures a real time two-way communication. All H.323 terminals must support H.245, Q.931, Registration Admission Status (RAS) and RTP protocols. The H.245 protocol is used for connection control, Q.931 protocol is ISDN signaling

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protocol, RTP is protocol for real time packets delivery and RAS is used for interaction with the Gatekeeper component.

“Gateway”, which implements “translator” function i.e. converts different transmission formats and realizes the interface between Public Switched Telephone Network (PSTN) and IP-network. It is optional because the terminals, which are in the same network, are able to communicate directly. “Gateway” is necessary when terminals from one network require communication with terminals from other network. In that case the VoIP communication uses the H.245 and Q.931 protocols.

“Gatekeeper” is the most important H.323 protocol component as it manages the system work. It performs other functions as follows: address translation, access control, call signaling, call authorization, bandwidth and conversations management.

Multipoint Control Units (MCU) are network end points that provide capabilities for multipoint (three and more subscribers) conference meeting. It uses the H.245 protocol. MCU consists of one multipoint controller for operation control and of multipoint processors which receive and process data streams as well as audio and video information.

The communication setup by means of H.323 protocol performs in five steps:

Call setup;

Initial communication and connection parameters negotiation/exchange;

Audio/video communication setup;

Call maintenance;

Call ending.

The call setup procedure means to locate a Gatekeeper which will manage the communication initiating terminal. The initial communication means registration of this end point at Gatekeeper. During the parameters negotiation/exchange process both communication sides – active and passive, negotiate their bandwidths through the Gatekeeper, payload formats recognized and transmitted, communication busy time and other restrictions. After parameters exchange through H.245 messages, the logical channels are opened and transmission started.

Comparing the MCS diagrams which illustrate call setup through H.323 (Fig. 2a) and through SIP (Fig. 2b) protocols we can see that both protocols are very similar in their basics. H.323 protocol requires decoding of the transmitted messages, while SIP protocol transmits plain text messages [2]. Description details for different messages shown on fig.2a) and 2b) are given in [3, 4].

The main disadvantages of the H.323 protocol are: its complexity as well as transmission of large number unnecessary messages in the network. These disadvantages impede H.323 protocol realization over large scope networks / networks over large geographical area /. Both protocols rely on Resource Reservation Protocol (RSVP) to provide guaranteed quality of services (QoS).

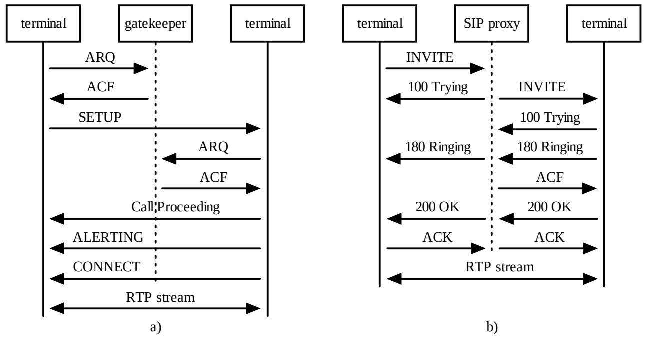

Fig. 2

The SIP protocol has been implemented wider in the modern cable television networks.

III. Estimation of the Effectiveness of MGCP and MEGACO Protocols in Case of Use over CATV Networks

Centralized architectures are built on the MGCP and H.248/MEGACO basis. Initially IETF implemented media gateway control protocol (MGCP). Thereafter it was modified by the same organization and in cooperation with ITU they developed a new standard, called H.248 according to ITU or media gateway control (MEGACO) according to IETF. MGCP and MEGACO are master/slave protocols which maintain call control elements (such as call agents, media gateway controllers (MGC) and softswitches). These elements control the media gateways (MG) operation. MGs accomplish connection among the communicating subscribers through the RTP or RTCP protocol according to the MGC managing instructions. MGs convert circuit-switched voice to packet-based traffic.

Fig. 3 shows the operation diagram of MGCP and H.248/MEGACO protocols. Connection among different MGCs is performed by protocols as SIP, H.323 and Q.BICC. They determine the route for RTP or RTCP connection among the communicating devices and send information to the MGCs managing the MG operation.

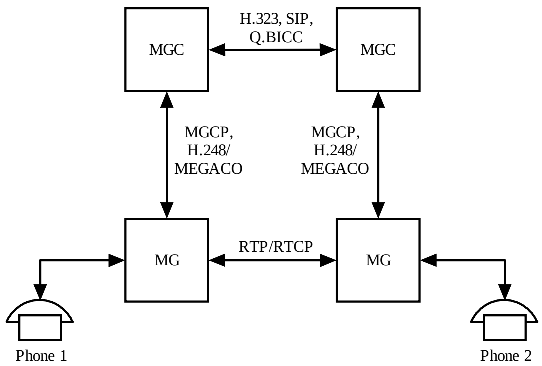

Fig. 3

As a MEGACO predecessor, the MGCP protocol is similar in many aspects. One of the differences between the two protocols is that MEGACO protocol uses many integer arrays while MGCP is plain text protocol [5]. MEGACO supports transport over TCP, UDP, SCTP and ATM, which makes it more flexible in comparison with MGCP, which supports only transport over UDP. MEGACO protocol disposes of better mechanism for guaranteed QoS by comparison with MGCP [1]. MGCP is more simplified and inexpensive for implementation over the CATV/HFC than H.248/MEGACO. That is the reason MGCP has been adopted by the CableLabs Company which develops PacketCable specifications, in spit of the supposition that it may be replaced by the H.248/MEGACO in near future [6].

IV. VoIP Realisation over CATV Networks According to the PacketCable Specification

PacketCable specification enables capability to cable operators to provide subscribers with low cost and good quality.VoIP.

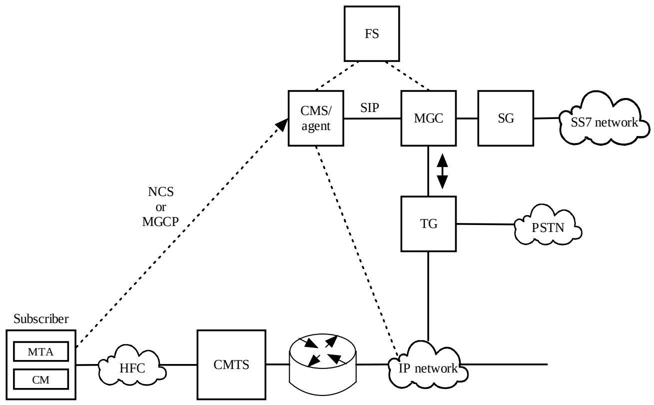

Fig. 4

PacketCable offers new real-time multimedia service specification which is DOCSIS standard compatible. One of the most important components of PacketCable specification is the MGCP protocol which provides mechanism for call set up and management. This specification also supports SIP protocol. Fig. 4 presents a summary diagram of VoIP service supply to CATV/HFC networks subscribers in compliance with PacketCable specification [7, 8].

This architecture consists of the following components:

Cable Modem (CM) - its purpose is to translate cable access network data signals into data suitable for subscriber data equipment use (computer or wireless router).

Multimedia Terminal Adapter (MTA), that converts the analog signal from the phone to IP packets and vice versa. The MTA can be a standalone unit or integrated in the Cable Modem.

Call Management Server (CMS) / Agent that manages the call control of the MTAs. It would receive events such as onhook and off-hook messages from the MTAs, and send commands such as ringing to the MTA.

Trunk Gateway (TG) – it provides translation of telephone voice signals into IP packets and capabilities to public switched telephone network (PSTN) subscribers to use VoIP.

Signaling Gateway (SG) - intended to provide connection with the SS7 system of the PSTN.

Media Gateway Controller (MGC) – it implies call control management to/from the PSTN. It controls the Trunk Gateway and implies connection to SS7 network through the Signaling Gateway.

Feature Server (FS) – intended to provide enhanced services.

Network based Call Signaling (NCS) or MGCP protocols are commonly used for establishing connection between MTA and CMS.

V. Conclusion

From the VoIP service protocols over CATV networks comparative analysis which has been performed at the present article the following conclusions are available:

SIP protocol is more appropriate by comparison with H.323 in case of realization over the CATV/HFC networks because of its simplified work algorithm and opportunity to support large scope area.

MGCP protocol is more simplified and inexpensive for implementation over the CATV/HFC networks than MEGAGO.

MEGACO protocol is more flexible considering its transport layer and disposes of better mechanism to provide QoS by comparison with MGCP protocol.

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