A multicast architecture for audioconferences made with SIP extensions and conference servers

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I. INTRODUCTION

Group applications like the audioconference require a minimun level of quality of service for the adequate transmision of voice packets and the improvements of signalling processes for joining, maintaining and leaving multicast groups. The protocol SIP (Session Initiation Protocol- RFC3261) is being implemented in voice over IP architectures extensively, and does not support multicast signalling properly. It is supported by IP multicast for Registration process with static IP addresses.The proposed architecture permits a conference server to interact with multicast domains of Overlay and IP Multicast. It is conformed by: a SIP extender , an agent called MGA (Multicast Gateway Agent) and a multicast manager. The main objective consists on including within SIP multicast extensions the basic functions of join, maintain and leave group members which are present at IGMP (Internet Group Management Protocol v3-RFC3376).

II.OBJECTIVES

a) Design of a network architecture for multimedia services with SIP multicast

b) Analize the performance of the architecture according to quality of service and security criteria

III. BACKGROUND

There are 3 Multicast approaches on the state of the art:

a) **IP multicast:** multicast routers (mrouters) exchange group membership information and use multicast routing protocols to establish trees to deliver data. On the LAN network the Dynamic Group Member Protocols are used by L2 switches like IGMP snooping, CGMP(Cisco Group Management Protocol and RGMP (Router-Port Group Management Protocol).

b) **Application Layer Multicast (ALM):** Group membership, multicast tree (or some other delivery structure) construction, and data forwarding are solely controlled by participating end hosts.

c) **Overlay Multicast:** proxies or service nodes that form the "overlay backbone" cooperatively construct an infrastructure and establish multicast trees (or other structures). Outside the backbone overlay, the proxies can deliver data packets to end hosts by using POM (Pure Overlay Multicast) or TTOM (two-tier Overlay Multicast) or unicast.

Figure 1 shows the three types graphically.

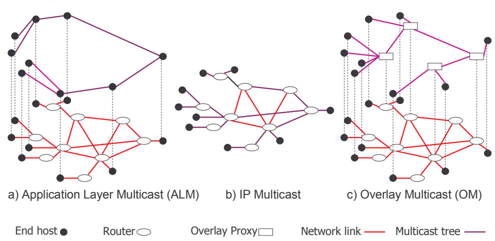

Figure 1. Multicast Approaches

Parallely, the protocol SIP (RFC3261) supports IP multicast for registrations. It is used on: single-hop-discovery-like service for registrations. (**Figure 2**) . SIP protocol uses SDP (Session Description Protocol – RFC4566) as a tool for describing each multimedia session. SDP sintaxis defines unicast or IP multicast sessions inside the payload of SIP messages. However, the IP multicast sessions can not be interactive (**Figure 3**).

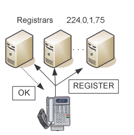

Figure 2. Registration with IP multicast

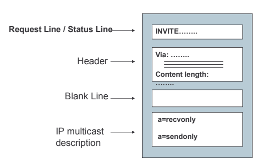

Figure 3. SDP describes IP multicast (unidirectional)

IV. TESTBED – ARCHITECTURE PROPOSED

Figure 4 depicts the architecture designed in this research.

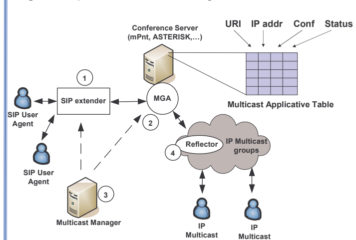

Figure 4. Architecture proposed as TestBed

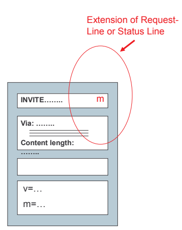

Figure 5. SIP message extended

(1) The SIP extender module permits → Forward SIP unicast messages transparently; Convert SIP unicast messages into multicast ones and viceversa

(2) The MGA (Multicast Gateway Agent) module permits → Interpret SIP multicast messages and modify the Applicative Multicast Table; Forward SIP multicast traffic onto IP multicast domain and viceversa

(3) The Multicast Manager module permits → Create, erase, join and separate multicast groups

(4) The Reflector permits → Send and receive media from IP multicast groups

V. CASE STUDIES

IP telephone joins an audioconference in high congestion

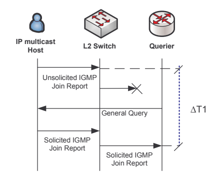

Figure 6a. Case study 1. Joining with IGMP Snooping

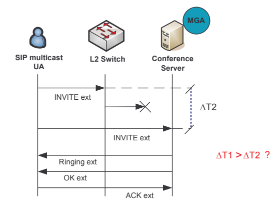

Figure 6b. Case study 2. Joining with SIP multicast

IP telephone maintains into audioconference in high congestion

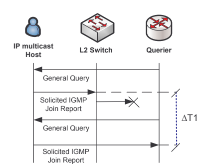

Figure 7a. Case study 3. Maintenance with IGMP Snooping

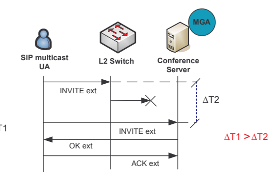

Figure 7b. Case study 4. Maintenance with SIP multicast

IP telephone leaves audioconference in high congestion

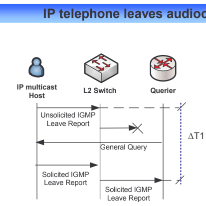

Figure 8a. Case study 5. Leaving with IGMP Snooping

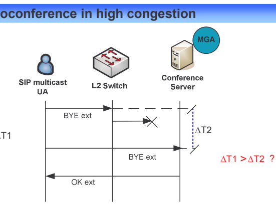

Figure 8b. Case study 6. Leaving with SIP multicast

VI. PERSPECTIVES AND FUTURE WORK

Important analysis have been done with case studies. The development of each module is being done based on JAVA and / or OPEN SOURCE. The idea is to "minimize the time of response" on multicast signalling in high congestion scenarios by using an extended SIP protocol.

It is also previewed an evaluation on quality of service and security of the architecture proposed.

Such architecture permits to work with both SIP multicast and IP multicast by having a role of Interworking

Further study is necessary to be done in environments with NAT and ALG (Application Layer Gateway).

VII. REFERENCES

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[2] Moreno, C ., Vincent, P, " Design of a SIP Open Source Architecture for Multicast Environments" *BBE2006 (BroadbandEurope2006)*, Geneva, Dec. 2006

[3] Moreno, C., Becker, M ., Vincent, P. "An improved multicast network architecture for multimedia services with SIP extensions" ICLAN2007 (*International Conference on Latest Advances in Networks*), Paris, Dec. 2007

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