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

Group applications like audioconferences 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.

Three types of multicast exist on the state of the art: IP multicast, application layer multicast (ALM) and overlay multicast (OM). Each one of them has its advantages and disadvantages. However, it is necessary to develop architectures that permit the interoperability between any of them.

Another important aspect is the use of conference servers with multicast domains jointly, by connecting the audioconference members that are located in different points of private and public networks. There are not currently implementations of that, as a consecuence of the complexity for the deployment of IP multicast, which is the most common implemented type. Some difficulties in routing and drawbacks in the recognition of group members inside local are networks are present.

By other hand, signalling protocol SIP is being implemented in voice over IP architectures extensively, and does not support multicast signalling based on application layer. It uses IP multicast addresses in some functions like registration process but members (SIP servers on this case) have to belong to the same group.

It is possible to extend SIP protocol, so that multicast signalling can be supported natively, like the way IGMP (Internet Group Management Protocol) supports H.323 on IP multicast approaches.

The proposed architecture permits a conference server interact with multicast domains of Overlay and IP Multicast. It is the result of the improvement of conventional audioconferences with 3 complementary modules: a SIP extender, a multicast agent inside the conference server called MGA (Multicast Gateway Agent) and a multicast manager.

The first one lets unicast traffic pass normally between a SIP UA (user agent) and the conference server bidirectionally. If the multicast manager gives any instructions, SIP messages are converted into multicast SIP extended ones and viceversa. The traffic of voice is also routed towards the conference server, or towards MGA for the correct dispatching.

MGA is in charge of routing multicast traffic, after interpreting SIP multicast extended messages sent from the SIP extender and sends back responses according to a multicast applicative table. It can also forward voice traffic to IP multicast groups by using a set of external reflectors. The conference server could also include this function itself.

The multicast manager sends instructions to the SIP extender just to let it know exactly which user agents should be included into multicast groups according to operator orders.

These three components permit the reutilization of SIP unicast messages as multicast ones to solve the drawbacks of multicast on LAN, join multicast domains to conference servers and the maintainance of the information.

Preliminary qualitative results through case studies indicate that the multicast architecture proposed based on SIP can work perfectly with every conference server. Future research will give quantitative results to show the improvements on quality of service and security with the architecture proposed.