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Voice over IP

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Multimedia Services over the IP Multicast Network

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Voice over IP (VoIP) is one of the most important and complex new services that are being introduced in Internet. VoIP makes use of several different technologies like signalling, streaming of real time data, session management, etc. The development and the experimentation of video conferencing applications over IP multicast networks have contributed greatly to the maturation of some of these technologies. This article summarizes the most important topics related with IP Multicast technology and video conferecing over IP Multicast networks. After introducing IP multicast technology as a mean to support many-to-many communications, we present some of the protocols and the applications used over IP multicast service. Finally, we outline some of the problems that preclude IP multicast to be widely deployed.

Keywords: IP Multicast, MBone, MBone Tools, RTP, SAP, SDP, Multicast Routing

1 Introduction to IP Multicast

We are presently go through a revolution in the Internet world as it happened with the WWW. Terms like e-commerce, video conferencing, streaming, video on demand and many others are becoming common usage in day-to-day life. VoIP is one of the technologies contributing to this revolution. However, the present IP networks are not adequately adapted to support this kind of services. For example, there are no mechanisms that guarantee a satisfactory quality of service (QoS) for VoIP communications. To support many-to-many communication, IP multicast offers a much more efficient mechanism than the current IP unicast networks.

1.1 Unicast vs. Multicast

The typical Internet services are based on the IP unicast model, that is to say: datagrams are addressed to only one host. Such communication is called "one-to-one". In some other situations, when there are more than two parties, the use of IP unicast can be very inefficient because the same information has to be sent to several destinations, and this process could overload the senders and the network (Figure 1).

How can we avoid this problem? Traditionally, it has been solved using "reflectors" (or Multipoint Control Units – MCU – according to H.323 terminology). A reflector is the equipment responsible for sending a packet from a source to all destinations taking place in the communication. This approach has several drawbacks, the most important being the excessive bandwidth consumption.

As an alternative, IP multicast can be used to make a datagram reach all the destinations that belong to a group. The concept of group is implemented using a special range of IP addresses. When a host is interested in receiving the datagrams addressed to a group, it has to join that group by signalling it to the network. This subscription is completely dynamic. The important fact behind IP multicast is that the source only sends one packet and the network is responsible for making the necessary copies to reach all destinations. This copying is made so that only one instance of the packet is transmitted over each link.

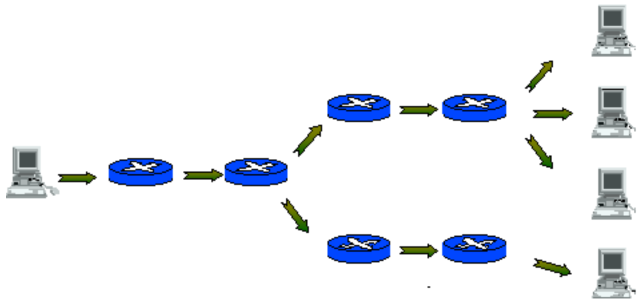

Figure 1: Distribution of one only packet to multiple receivers

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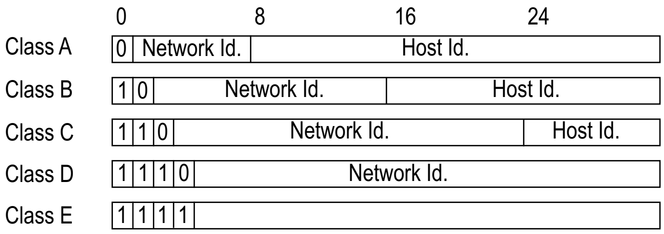

Figure 2: IP Address classification

1.2 Multicast Addressing

When the IP addressing scheme was designed, several classes of addresses were defined, as depicted in Figure 2. The class D addresses were reserved for multicast. So, IP multicast uses the range 224.0.0.0 to 239.255.255.255. These addresses are commonly called group addresses or multicast addresses. From the whole range of multicast addresses, some are reserved for specific purposes, and the rest can be used by multicast applications.

1.3 Internet Group Management Protocol (IGMP)

As we mentioned before, the way IP multicast works can be summed up as follows: receivers join the group they are interested in, and the network makes the datagrams sent to that group delivered to every receiver that joined the group. A mechanism is needed for the hosts to tell the router that they are interested in joining a certain group. This mechanism is the IGMP protocol [Deering 89], that defines the behaviour of hosts and routers when informing about joining or leaving a group.

When a host wants to join a group, it sends an IGMP report message to the 224.0.0.1 address (which should be included with every multicast-enabled host). When the router receives this message, it takes into account that there is at least one host interested in receiving that group on that interface.

Periodically, the router sends IGMP query messages to the IP multicast address of the group to ask for renewals. If there are still receivers, one of them must answer with an IGMP report addressed to the group it wants to renew. The IGMPv2 is now commonly used, however IGMPv3 implementations start coming up.

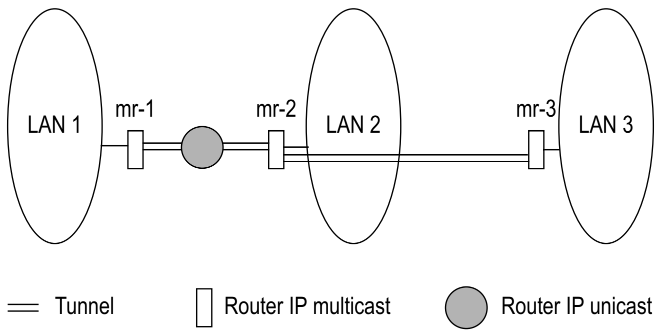

Figure 3: A multicast tunnel schema

1.4 Mrouters and tunnels

Most of the first Internet routers were manufactured without taking into account IP multicast traffic[1]. In order to experiment with IP multicast over Internet, it was necessary to define a way to interconnect IP multicast-enabled networks through networks without IP multicast support. This activities brought about what we know today as the Multicast Backbone, or simply MBONE [Macedonia/Brutzman 94].

IP multicast-enabled routers are sometimes called *mrouters*. They must satisfy two basic requirements:

- Implement the IGMP protocol.
- Use some IP multicast routing algorithm.

In order to connect a network to MBONE, one of their *mrouters* has to be connected to the rest of the IP multicast clouds. If our Internet Service Provider (ISP) offers the native IP multicast service, a simple configuration will suffice to do the task. Otherwise, we will need to configure a "tunnel" to another router connected to the MBone. This tunnel will be used to encapsulate every IP multicast datagram that has to be transmitted between these two routers into IP unicast datagrams addressed to the other end of the tunnel. So, an IP multicast-disabled network can be traversed (Figure 3).

1.5 Multicast routing

IP multicast routing is in charge of making the datagrams to flow from sources to every destination joined to a group. A good routing algorithm should guarantee that

- an IP multicast datagram addressed to a multicast group G, reaches all the hosts that have joined G,
- there are no loops, i.e., a datagram reaches its destination only once and, if possible, using the shortest path.

The IP multicast routing protocols are usually classified according to the way they work: There are protocols that work in *dense mode* like *Distance Vector Multicast Routing Protocol* (DVMRP [Waitzman et al. 98]). Some others work in *sparse mode* like *Protocol Independent Multicast Sparse Mode* (PIM-SM [Estrin et al. 98]). There are algorithms that do not fit exactly in one of these two models, an example is *Multicast Open Shortest Path First* (MOSPF [Moy 94]). The common practice is to use *PIM-sparse-dense mode* as intradomain multicast routing protocol because it is very efficient and needs no special routing messages interchanged between neighbours. Instead, it uses the unicast routing table in the router to do the calculations and decide on the better paths.

Multicast routing algorithms are usually much more complicated and difficult to configure than the unicast ones.

2 Services over IP Multicast

2.1 Multimedia Protocols

IP multicast as a network service offers no service directly to the user. But it offers an excellent framework for many-to-many multimedia data internetworking. Many protocols have come up in the IP multicast and multimedia content distribution

1. Almost every router manufactured today implements at least IGMP and one or more IP multicast routing protocols.

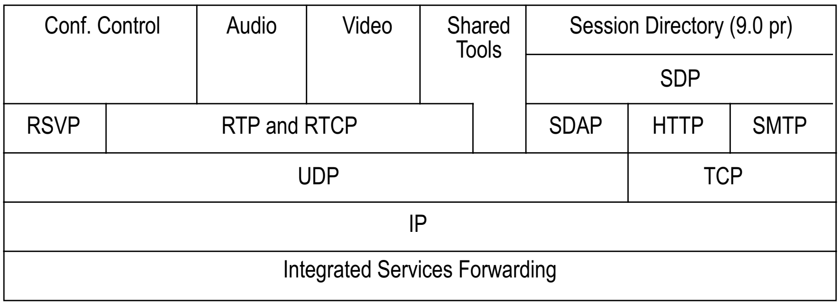

Figure 4: Multimedia protocols

environment (Figure 4). This set of protocols range from generic protocols like those used for audio and video streaming (i.e. RTP), or to manage multimedia sessions (i.e. SAP, SDP, SIP); to others much more specific used only for certain applications. It is important to note that some of these protocols, although designed, developed and tested over MBONE, have been widely adopted in other frameworks like VoIP.

2.2 Real Time Protocol (RTP)

Internet follows a *best-effort* delivery model. That means that, when a datagram is sent, the network does guarantee neither the delivery nor that the different datagrams sent arrive in correct sequence or in time. The network only guarantees that it will do as well as possible in delivering the datagram. In some cases, specially with real-time data like VoIP traffic, this model does not work properly.

Besides, when sending continuous media (like digitised voice or video) over a packet network, it is essential to provide the means for the receiver to be able to reconstruct the original information. The packets generated at the origin – typically a flow of packets equally spaced in time – will arrive to the receiver possibly out of sequence, with losses or changes in inter-packet times (jitter). A mechanism is needed for the receiver to be able to reproduce the audio or video being sent in such environment.

Real-Time Transport Protocol (RTP, [Schulzrinne et al. 96]) is the protocol defined by the IETF for real-time data transport over the Internet. The protocol basically labels each datagram to be sent, attaching them information like the time when they were generated or the type of codec being used, so that when they reach the receiver, it can reproduce the original flow adequately.

It is important to note that RTP provides no mechanism for on-time delivery or any other QoS guarantees. In order to offer QoS support for real-time communications, RTP have to be used together with other protocols or solutions like Integrated Services or Differenciated Services QoS models. However, RTP provides specific means to control the quality of the distributed data, the Real Time Control protocol (RTCP), that allows the senders and receivers to know, for example, the packet loss rate in a session. In addition, RTCP provides identification mechanisms for RTP communications.

RTP has become the *de facto* standard to send continuous media over packet (mainly IP) networks. It is used by most of VoIP proposals like H.323, MEGACO or SIP.

2.3 SAP and SDP

In a multicast environment where all participants can send and receive data between them, the concepts of client and server makes no more sense. Hence there is a need for a mechanism to allow for the user to locate a conference he is interested in. This mechanism must be based in the concept of session, which is an aggregation of related contents. For example, in a video conference, a session would be defined as the multicast group used for audio transmission, the multicast group used for audio transmission, the codecs being used both for audio and video, and so on.

For session management, two protocols have come up in MBONE: SDP and SAP. SAP (*Session Announcement Protocol* [Handley et al. 00]) defines the use of specific multicast groups to distribute session information. The session creator is responsible for periodically reannouncing it so that people joining the special "announcement group" after the creation can still know about the session. The information used to describe the session is specified by SDP (*Session Description Protocol* [Handley/Jacobson 98]).

As RTP, SDP protocol has been reused in other environments, for example, it has been incorporated into MEGACO VoIP proposal.

2.4 SIP

The SDP/SAP model requires the user to look for the session he wants to attend. However, in some scenarios like IP telephony, a way to invite other parties to participate in a session is needed. The *Session Initiation Protocol* (SIP, [Handley et al. 99]) came up for covering this need.

SIP defines the signalling mechanisms that are necessary to establish a session and to negotiate the parameters to be used in it, such as codecs, media, location, etc. As other protocols mentioned, SIP has surpassed the MBONE environment were it was originally created and now it has become one of the main proposals for VoIP. In fact, SIP has been recently selected by 3GPP to be used as the VoIP protocol for 3G mobile networks based on "All-IP" proposal.

3 MBone Tools

Several applications have been developed to test the advantages of the IP multicast model at the initial stages of the MBone. These tools, commonly known as the *MBone Tools* allow us to participate in different kinds of video conferences and meetings using IP multicast as the network technology. The typical MBone tools are (Figure 5):

- SDR. This tool is equivalent to a TV guide. It shows all planned and ongoing MBONE sessions. Recent versions also allow us to use a "quick call" service based on SIP.

- VIC. This tool is used for video transmission with a great variety of codecs available. It can be used on almost every platform and is compatible with several standards for capturing video. So, it allows a simple personal computer to send video without needing to buy an expensive video capturing hardware.
- VAT and RAT are used for audio conferencing. They are also available for many platforms and support several codecs like GSM, PCM, DVI, and so on.
- WB. This tool is a distributed shared whiteboard that can be used by all the participants and offer the same functionality as the usual blackboard in a classroom.
- NTE stands for *Network Text Editor* and offers the functionality of a distributed word processor. It supports tokens for asking permission to write and is quite comprehensive.

Recent applications using IP multicast are more complex but offer new important functionalities. The goal is to integrate all these tools into one specific tool possibly in the Web. In fact, some big projects like MASH are working on Web integration. Some of these "new generation multicast tools" are:

- DLB is an improved version of an electronic whiteboard that supports two modes (on-line and off-line). It thus allows for editing slides off-line and then present them on-line.
- MiNT is a very complete application developed by the German GMD that supports the SIP protocol, includes an RSVP agent for bandwidth reservation and even an integrated GUI for audio and video transmission.
- MASH is a very big project initiated at Berkeley and its main goal is to integrate the typical MBone Tools into a common GUI. In addition, tools for playing and recording sessions are offered. They are also deploying "mashlets" that aim to integrate the GUI into the WWW.
- RELATE (REmote LAnguage TEaching). This tool was developed at the University College London and is very interesting for teaching on-line. Although it was initially thought for language teaching it can also be used to teach some other subjects. This tool integrates into the same GUI the audio, video, whiteboard and text editor applications.

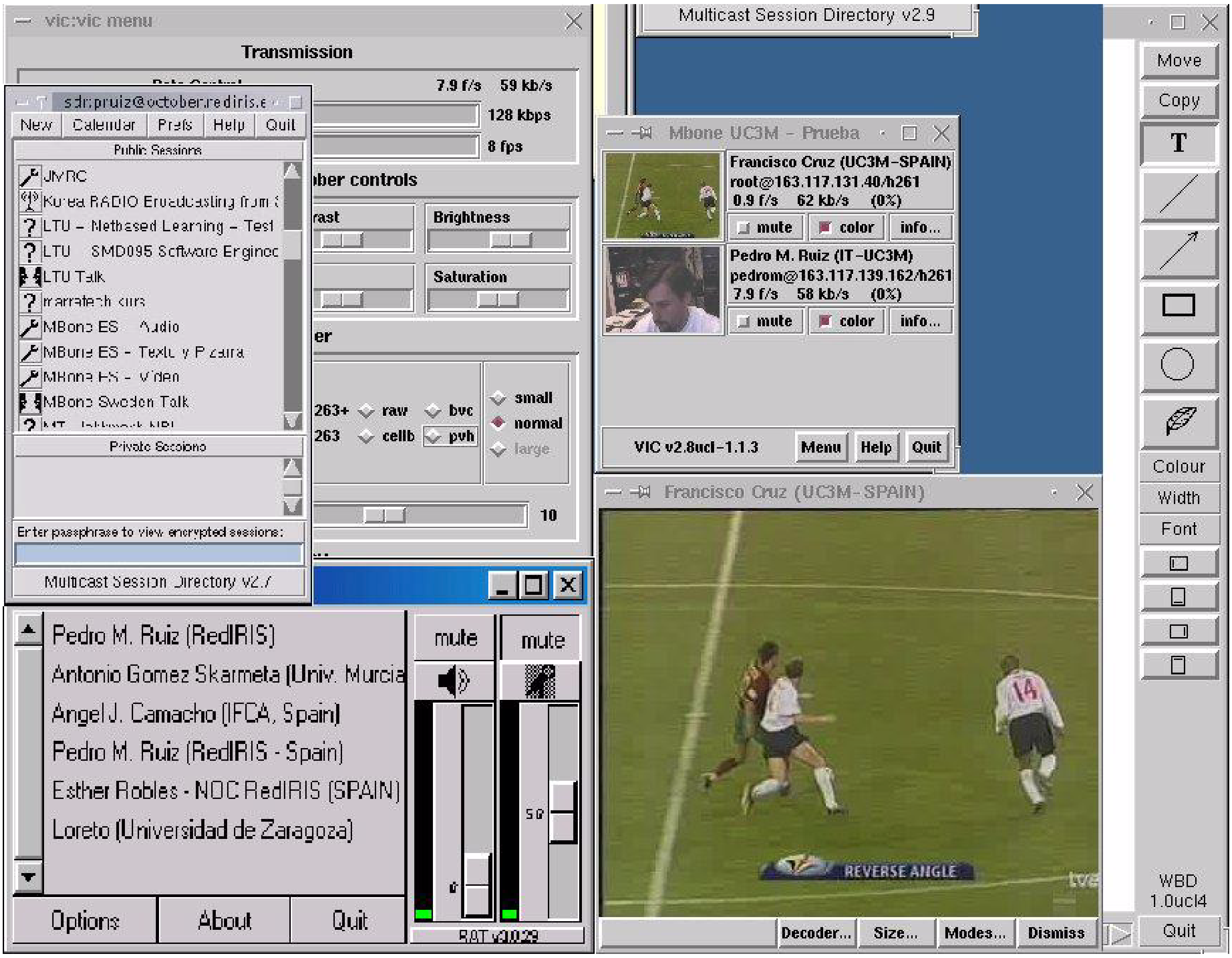

Figure 5: Some of the MBone tools

4 Advanced services over IP Multicast

As mentioned in this article, video conferencing over IP Multicast initiatives led to an important number of applications and protocols. Many are used in other environments, mainly in VoIP architectures (i.e. RTP, SIP, SDP, codecs, etc).

However, IP multicast and its video conferencing tools have limitations that hindered the deployment of MBONE tools.

4.1 Multimedia services integration

Multimedia services over IP multicast present the following limitations:

- No integration with solutions based on some other technologies. It is desirable to have a solution to integrate IP multicast with H.320, H.323 and other VoIP solutions in general. SIP will be very useful as a glue element between all these technologies.
- No integration between the MBone tools. Currently, one different tool is used for every service (i.e. VIC for video, VAT for audio and so on). Although this model simplifies the design and development of the tools, it becomes an issue due to the lack of synchronization and user interface integration between tools. Nowadays, several proposals are coming up within the IETF to solve this problem. There are two basic models: mbus (Multicast Bus) and SCCP (*Simple Conference Control Protocol*).

4.2 Security and access control in multicast environments

In the same way that the video conferencing tools over IP Multicast present several limitations that are slowing its deployment, the limitations of the IP Multicast model are making ISPs to think twice before offering the IP Multicast service to their customers. The main problems of that model can be summed up as:

- Denial of Service (DoS) attacks.
- Policy of use, because there is not a defined method to control access to the network.
- Authentication.
- Address allocation.

To avoid these problems, several working groups at the IETF are defining new protocols or even updates to the current IP multicast model. Some such initiatives are:

- BGMP (*Border Gateway Multicast Protocol*). It is thought to be the successor of the currently used MBGP (Multiprotocol BGP). This protocol is very scalable and, if combined with other protocols for address allocation like MASC, is able to solve the address allocation problem.
- MSEC (*Multicast SECurity*). It is a new IETF working group that is responsible for studying and solving security concerns in IP multicast.
- SSM (*Source Specific Multicast*). This is a new multicast model based on the concept of *channel*: a pair of a source and a multicast group. The key concept in SSM is that routing decisions are taken based on channels instead of multicast groups.
- GLOP. This mechanism divides statically and according to the AS (autonomous system) number, the 233.0.0.0/24 range so that there won't be collisions between ASs when selecting an IP multicast group. Figure 6 shows how to know what groups belong to what Autonomous System. For example, in the case of RedIRIS, the AS number is 766, so the range 233.2.254.0/24 is available for being used within the RedIRIS AS without worrying about possible collisions.

0	8	16	24
233	16 bits AS		local bits

Figure 6: Structure of GLOP addressing

5 Conclusions

In this article, we have covered the most important topics related with IP Multicast technology and video conferecing over IP Multicast networks. We have showed how important protocols developed inside MBONE initiative have been later reused in the most outstanding VoIP proposals like H.323, SIP or MEGACO. Although multimedia over packet networks is a wide subject and has been vastly investigated and experimented, we can consider MBONE as an important testbed where basic technologies nowadays used in VoIP have been matured.

Although multicast video conferencing is not popular at this moment – most of present VoIP scenarios resemble the ones found in conventional telephone networks, and so, they are unicast –, IP multicast opens possibilities for new applications, improving the use of network resources. But, as mentioned before, much more development and research is needed to improve multimedia conferencing over IP Multicast.

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