IMS signalling for multiparty services based on network level multicast

Ivan Vidal, Ignacio Soto, Francisco Valera, Jaime Garcia, Arturo Azcorra
University Carlos III of Madrid
Av. Universidad, 30
E-28911, Madrid, Spain
Email: {ividal,isoto,fvalera,jgr,azcorra}@it.uc3m.es

Abstract— **The standardization process of the UMTS technology has led to the development of the IP Multimedia Subsystem (IMS). IMS provides a framework that supports the negotiation of the next generation multimedia services with QoS requirements that are envisioned for 3G networks. But even though many of these services involve the participation of multiple users in a multiparty arrangement, the delivery technology at network level is still unicast based. This approach is not optimum, in terms of transmission efficiency. In this paper, a new approach is presented proposing to use a network level multicast delivery technology for the multiparty services that are signalled through IMS. The main advantages and drawbacks related with this new approach are analyzed in the article. Finally, as a starting point in the development of the presented solution, a new SIP signalling dialogue is proposed allowing the negotiation of a generic multiparty service, and supporting at the same time the configuration of the corresponding network level multicast delivery service with QoS requirements that will be used in the user plane.**

I. INTRODUCTION

The new communication paradigm offered by 3G technologies has brought to the market of mobile communication networks a new broadband access infrastructure and enhanced terminals. These components will allow the provision of the new value-added multimedia services that are envisioned for the future in 3G networks. The proper execution of these services will require, among other things, to implement mechanisms that guarantee the QoS provision in the transport networks.

The standardization process of the UMTS technology as the 3G standard, performed by the 3GPP [1], has leaded to the development of the IP Multimedia Subsystem (IMS). With a layered design, which separates transport and signalling services, IMS provides a framework that supports the negotiation of next generation multimedia services between end users, providing at the same time integration with functionalities that are essential for a complete service architecture, such as user registration, security, QoS control and support for roaming and charging.

On the other hand, many of the multimedia services that are envisioned for UMTS involve the interaction between multiple participants in a peer to peer arrangement. Currently, these services are provided to the end users by means of unicast based delivery services. This approach, nevertheless, presents several drawbacks that will be briefly reviewed in Sect. III.

In this paper, a new approach is proposed to deliver multiparty services to the end users within UMTS network environments. The approach consists of using a network-level multicast delivery service for the multiparty services that are negotiated using IMS. This solution introduces a new framework in IMS that allows to develop multiparty services, providing at the same time a cost-effective solution to UMTS operators. In the article, a new IMS signalling dialogue involving multiple parties is presented. This dialogue supports the configuration of a network level multicast delivery service in the user plane with QoS requirements.

The rest of the article is structured as follows. The second section provides a brief overview of IMS, to give the reader a basic knowledge about the concepts that will be developed in the paper. The third section covers the issues related to the integration of a network level multicast delivery service with QoS requirements in the IMS user plane. Section IV describes the SIP signalling dialogue proposed in this work, focusing on the issues related to SIP routing, session description negotiation and resource reservation. Finally, Sect. V describes the most important conclusions achieved along the article.

II. IMS OVERVIEW

The IP Multimedia Subsystem (IMS) [2] is an IP based architecture that provides a multimedia call (or session) control service over packet networks, allowing applications to establish synchronous multimedia sessions. IMS offers this session control to the applications by means of a signalling system based on the SIP signalling protocol [3].

To negotiate the parameters associated with the media which is going to be transferred during the session, such us the media components, codecs or IP addresses and ports, the Offer/Answer model [4] of the Session Description Protocol (SDP) [5] is used. SDP provides the support to describe multimedia sessions, and the Offer/Answer model applied to this protocol allows the user equipments (UE) to reach an agreement about the session description.

This signalling system for applications provided in IMS forms a control plane in the architecture that is independent from the user plane used by those applications. Examples of applications that can be created using IMS are VoIP applications, presence services, Push to talk, etc. Figure 1 presents an overview of the IMS architecture.

1-4244-0857-1/07/$25.00 ©2007 IEEE

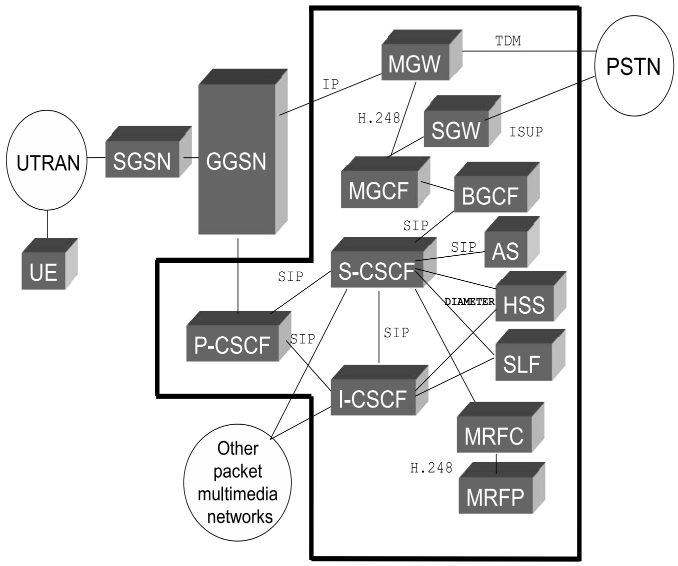

Fig. 1. IMS architecture overview

Out of the entities that compose this architecture, the key elements are the ones providing control functionalities, namely the Call Session Control Functions (CSCF), that are basically SIP servers. The Serving-Call Session Control Function (S-CSCF) provides session control, acting as a SIP registrar and a SIP server. The Proxy-Call Session Control Function (P-CSCF) is the first contact point for the UE in the IMS network and acts as an Inbound/Outbound proxy server for the UEs. The Application Servers (AS) are entities that contain service logic to provide services in IMS. On the other hand, the Home Subscriber Server (HSS), stores all the data related to subscribers and services in IMS.

Finally. although it does not really belong to the IMS infrastructure, the Serving GPRS Support Node (SGSN) links the radio access network and the packet core network. In addition, the Gateway GPRS Support Node (GGSN) provides in the packet domain of UMTS the internetworking with external packet based network, providing the UEs with IP connectivity in the user plane using PDP contexts. A PDP context is a logical connection with QoS guarantees by means of which the UE and the network may exchange IP packets.

III. IMS AND NETWORK LEVEL MULTICAST

IMS is an IP-based architecture that provides the users with means to negotiate the next generation multimedia services with QoS requirements that are envisioned for UMTS, such as Conference, Messaging or Push to Talk over Cellular (PoC). These services are based on the Internet Protocol (IP), as the network level delivery service, and involve the communication between users in a peer-to-peer arrangement.

On the other hand, the provision of some of these services imply the exchanging of information between multiple participants. The specifications developed to date by 3GPP for these multiparty services still consider an unicast based delivery service in the user plane. In this approach, the data traffic corresponding to the service is replicated in an AS and then is sent to each participant user. As it can be seen, this solution normally increases the traffic load in the network, producing a less scalable solution in terms of services and served users. It also presents a serious drawback in terms of transmission efficiency. In this respect, consider the case where several participants in a multiparty service are present in the same UMTS cell. In this scenario, many copies of the information will be sent to the cell where the users reside. This necessarily represents an inefficient use of the resources in the core network and what is even more important, in the radio access network, where the bandwidth availability is critical. From the previous considerations, it is concluded that a network level multicast based approach would bring the following advantages:

- More transmission efficiency, in the core network and in the radio access network. In the multicast approach the IP packets would be routed by the network, being replicated at network level only when necessary. The transmission from the GGSNs to the multicast receivers could be done by means of shared PDP contexts.
- Better scalability, in terms of services and served users, as a logical consequence from the previous point and due to the fact that there are no AS behaving as bottlenecks in the user plane.
- Better fault tolerance. As in the unicast based approach all the data traffic must pass through a single entity, this becomes an isolated point of failure. In the multicast based approach the data traffic is distributed by the network towards the final destinations, being the multicast routing mechanisms in charge to provide consistency to node failures in the backbone networks connecting the GGSNs.

However, using a network-level multicast based approach to deliver the data traffic in the user plane implies that the GGSNs in the UMTS core network must implement the IGMP support, to configure the multicast delivery service from each UE, and must support multicast routing by means of implementing any multicast routing algorithm, such as PIM-SM. This, on one hand may increase the processing load in the GGSNs, but on the other hand the traffic load received by the GGSNs would decrease when serving multiple UEs that receive traffic corresponding to the multicast group.

On the other hand, the solution currently proposed by the 3GPP for broadcast and multicast services in UMTS is the Multimedia Broadcast/Multicast Service (MBMS). MBMS allows a service provider to deliver multicast based services to the end users. The multicast operation mode implemented by MBMS is designed to efficiently use the available resources in the core and the radio access networks, in order to transmit the multicast data to the users subscribed to each multicast group. Further details about the broadcast and multicast services in 3GPP and about the MBMS service can be found in [6].

Nevertheless, the solution presented in MBMS is mainly focused on the figure of a service provider as multicast source, not allowing the end users to negotiate the features of the service over IMS nor considering the possibility of an user becoming a traffic source in the multicast service. These features are essential for the full integration of a network level multicast delivery service in IMS that supports the exchange of information in the user plane. To make this integration possible the following important issues must be considered:

- In the control plane, a new SIP signalling dialogue to negotiate generic multiparty services between the end users must be provided. This dialogue must result in an adequate resource reservation in the user plane for the proper execution of the service, as well as in the configuration of the network level multicast delivery service that will be used in the user plane.
- In the data plane, the multicast routing of the IP packets must be performed in the UMTS core network. In addition, the bearer services in the user plane for the data transfer must provide an efficient use of the available resources in the core network and in the radio access network (e.g. by using a common radio channel over the radio interface).

Next section covers the first objective presented for the control plane, presenting a SIP signalling dialogue that allows to negotiate multiparty services over IMS, supporting at the same time the configuration of a network level multicast delivery service in the user plane with the QoS requirements that were negotiated by the participants in the session.

IV. IMS SESSION ESTABLISHMENT FOR MULTIPARTY SERVICES

As in a typical one-to-one IMS session, the first thing the originator UE must do in order to establish the multimedia session is to create a SIP dialogue with the UEs belonging to the end users that will be involved in the communication. This SIP dialogue will allow the participant parties to negotiate the characteristics of the media components that will be exchanged during the session, to reserve the adequate resources in the user plane to provide the QoS required for each media component and to support the configuration of a network-level multicast delivery service for each media component in the multimedia session.

Figure 2 shows a general overview of the session establishment procedure which is proposed in this article to setup a multimedia multiparty session, from the point of view of the originator UE.

Figure 3 shows the session establishment procedure from the point of view of each destination UE.

For the exchange of the SIP signalling messages, it is assumed that all of the participants UEs count with a dedicated signalling PDP context that was previously established. The session establishment procedures are covered in detail in the following subsections.

A. Routing of SIP signalling

This subsection details the routing procedures that are applied to properly route the SIP signalling messages that are

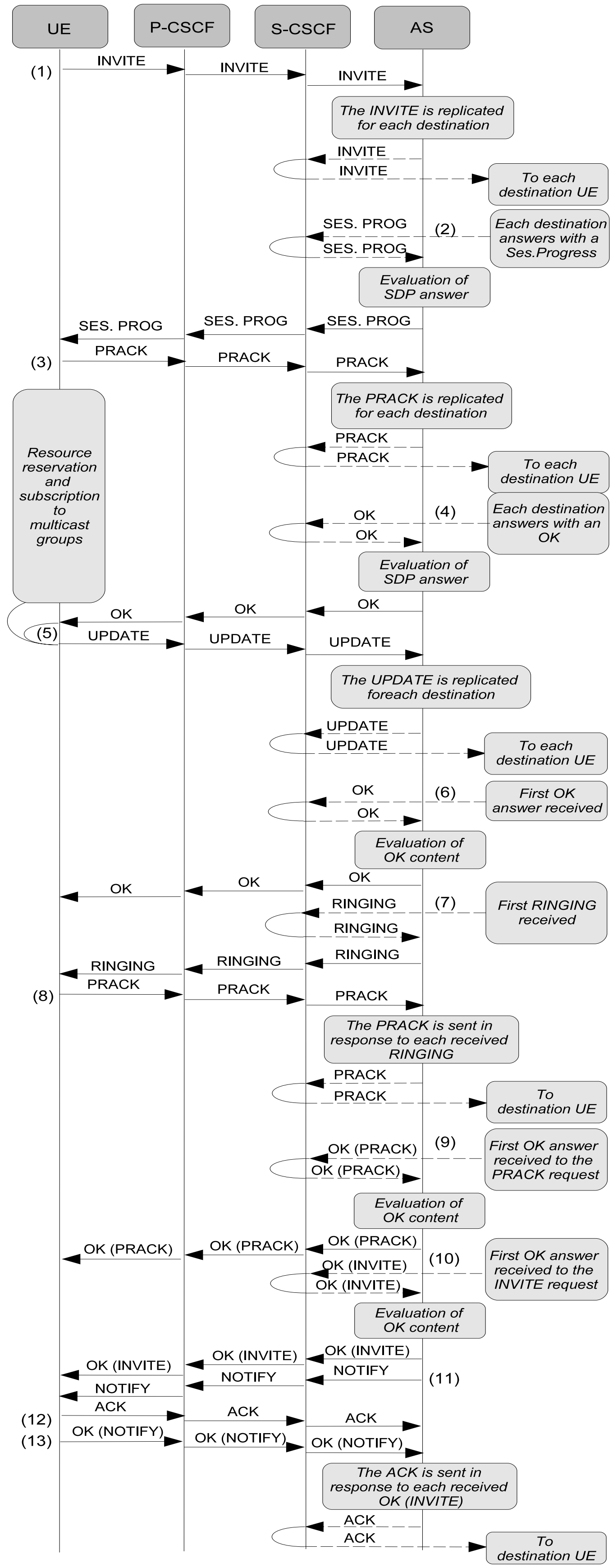

Fig. 2. IMS session establishment, originating side

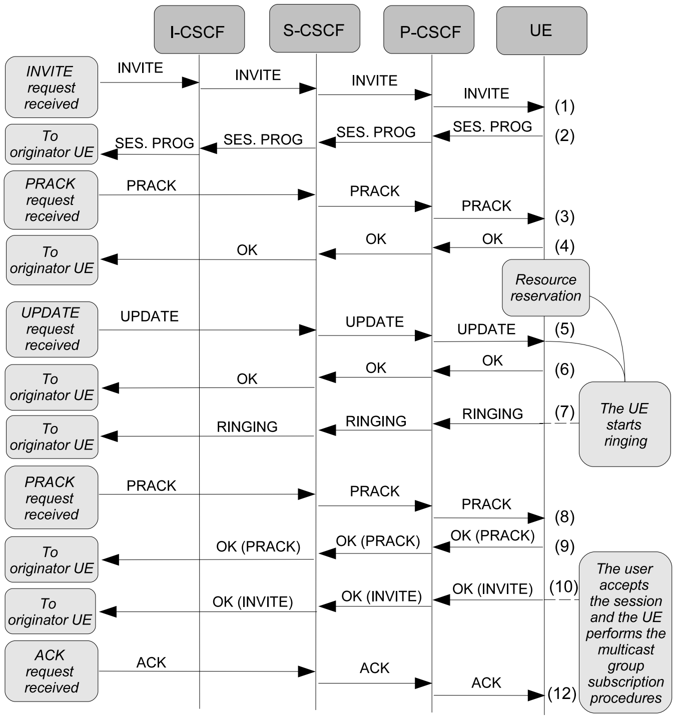

Fig. 3. IMS session establishment, destination side

exchanged between the originator UE and the destination UEs within the SIP dialogue.

1) Routing of the INVITE request: to create the SIP dialogue, the originator UE must send a SIP INVITE request to the destination UEs belonging to the users the originator wants to communicate with (point 1 in figures 2 and 3) . Nevertheless, at this point the only information the originator UE counts with to properly route the INVITE request to each destination UE are the SIP URIs of the destination users and the route to its own S-CSCF. With this information, the originator UE constructs the initial INVITE request including the following elements:

- A request URI in the start line of the INVITE request, containing the list of SIP URIs corresponding to the end users that the originator wants to involve in the multimedia session.
- A Via header with the address of the originator UE. This header will allow the originator to receive all the responses to the INVITE request.
- A set of Route headers, specifying the route from the UE to the S-CSCF, passing through the P-CSCF which acts as the outbound proxy for the originator UE.
- A Contact header, with the address where the originator UE wants to receive future requests within the SIP dialogue.

The INVITE request is then sent to the next hop specified in the Route headers, which is the P-CSCF of the originator UE. After receiving the request, the P-CSCF removes the first entry from the Route header (the one containing its own information), it includes its address in a Via header (so as to receive the responses to the INVITE request) and it creates a new Record-Route header. This header will contain the address of the P-CSCF, so as to receive future requests sent in the dialogue from any of the participant parties. Finally, the P-CSCF routes the INVITE request to the next hop specified by the Route headers, which in this case is the S-CSCF.

The S-CSCF checks the information contained in the INVITE request against the initial filter criteria corresponding to originator user. Based on this checking process, the S-CSCF certifies that the INVITE message must be processed by a multiparty Application Server (AS). This AS will be provided by the operator to manage the establishment of the multiparty sessions initiated by a set of originator users.

The filter criteria which is necessary to verify whether an INVITE request should be sent or not to a multiparty AS could be based on the content of the Require header within the INVITE request. A new option tag will be defined for this SIP header, indicating the necessity to process the INVITE request according to the necessary extensions to proceed with the multiparty session establishment.

Therefore, the S-CSCF must send the INVITE request to the multiparty AS. However, to guarantee that after the processing performed by the AS it receives the request back, it includes two Route headers with the address of the AS and its own address, so as to route the request to the AS as the next hop and to get the request back once the AS concludes with the processing. In addition, it includes a Via header and a Record-Route header with its own address.

After receiving the INVITE request, the multiparty AS:

- Will remove the first Route header, which contains its own information.
- Will assign a multicast IP address to each media component specified in each m-line within the SDP payload of the SIP message. This way, each media component will be assigned to a multicast group, so different UE with different capacities will be able to subscribe to the media components that are able or are willing to accept in the session. Each assigned multicast address will be reserved for the media component while the session remains active.
- Will include a new Via and Record-Route header with its own address. By including its address in the Via and Record-Route headers, the AS assures that it will receive all the SIP signalling messages exchanged between the parties involved in the session. This configuration, made by the AS, will be essential so as to be able to establish, modify and release the multimedia multiparty session.
- Finally, the AS will replicate the INVITE request for each destination SIP URI specified in the request URI within the start line. The request URI within each replica is set to the corresponding destination SIP URI. Finally, each replica is routed based on the Route header back to the S-CSCF.

For each INVITE request, the S-CSCF removes its own entry from the Route header and evaluates the rest of initial filter criteria. Finally, it routes the request to the IMS domain corresponding to the destination user. To do so, it resolves the host part of the request URI within the INVITE request in the Domain Name Service (DNS), obtaining a set of addresses of I-CSCFs in the destination IMS domain. The S-CSCF selects one of these I-CSCF and sends it the INVITE request.

The I-CSCF, upon receiving the INVITE request, recovers from an HSS the address of the S-CSCF assigned to the destination user and appends a Route header to the request with this address. It also includes a Via header with its own address to the request. Note that, with this configuration, the I-CSCF will not receive any further requests sent within the dialogue, as it does not append any Record-Route header to the INVITE request. Finally, the request is routed to the S-CSCF of the destination user.

The S-CSCF removes its entry from the Route headers, appends a new Route header with the address of the P-CSCF corresponding to the destination user and includes a new Via and Record-Route header with its own address. It also changes the request URI in the INVITE request by the registered contact address of the destination user. Finally, the request is routed to the next hop specified in the first Route header, which is the P-CSCF of the destination user.

The P-CSCF receives the request and removes its own entry from the Route headers. It also includes its own address in a new Via and Record-Route header and finally sends the request to the destination UE indicated by the request URI.

Once the destination UE receives the INVITE request, it saves the Record-Route headers and the Contact header. These headers jointly conform the path that will be used by this UE to send further requests in the dialogue to the originator UE. Note that, as the multiparty AS address is contained in the Record-Route headers, all the requests sent from any destination UE will pass through this functional entity.

2) Routing of the Session in Progress response: after receiving the INVITE request, each destination UE answers back with a SIP Session in Progress message (point 2 in figures 2 and 3). This SIP response includes a Contact header, with the address of the destination UE. It also includes the Record-Route and Via headers that were received within the INVITE request.

Each Session in Progress response is routed back to the originator UE following the path specified in its own Via headers. Each proxy SIP that receives the response, removes its own entry from the Via headers and sends the response to the next hop in the Via list. This way, each response will eventually reach the multiparty AS. As it will be explained in Sect. IV-B, the AS will wait to receive all the Session in Progress responses to the INVITE request. With all these messages, the AS will generate a single Session in Progress message that will finally be sent to the originator UE. To guarantee the consistency of the routing procedures during the dialogue, the AS will include the following routing related information in the response:

- First of all, it will generate a new SIP URI that uniquely identifies the multiparty session. This SIP URI will be included in the final Session in Progress response, within the Contact header. The AS will save the Contact headers included in the received Session in Progress messages for further use, as it will be explained next.
- On the other hand, the Record-Route headers in the final response will only include information about the path between the originator UE and the multiparty AS. Note that these are the Record-Route headers that are common in all the Session in Progress messages received from the destination UEs. Again, the AS will store, for each destination UE, the rest of the Record-Route headers that were received in its corresponding Session in Progress response for further use.

Upon receiving the Session in Progress message, the originator UE stores the SIP URI of the multiparty session, as it was received in the Contact header, and the Record-Route headers. This information will be used to route subsequent requests sent by this UE within the dialogue.

3) Routing of subsequent requests and responses: whenever any UE in the session needs to send a new request within the dialogue (e.g. PRACK, UPDATE or BYE), it will copy the infomation contained in the stored Record-Route headers in new Route headers, and the stored Contact header in the request URI of the new SIP request. With this routing related information, the request will be properly routed towards its destination.

If the SIP request is sent by the originator user, the request will eventually reach the multiparty AS. Upon receiving the request, the AS will recover the routing related information for the session from the SIP URI contained within the request URI field. Then, it will replicate the message for each destination user associated with the session. To route each replica to the proper destination UE, the AS will set the request URI in the replica to the value of the Contact header received in the Session in Progress response from the destination UE. In addition, it will include the Record-Route headers that were also stored for the destination UE in new Route headers. Jointly, the Route headers and the request URI, provide all the information that is needed to properly route each replica to the corresponding destination UE.

Finally, any SIP proxy in the path towards the UE specified in the request URI appends its own address to a Via header. As usual, all the responses sent back from that UE will follow the path specified by the Via headers.

B. Session description negotiation

Before the establishment of the multimedia session, the different parties must reach an agreement about the description of the media components that they will exchange during the session. To do so, the Session Description Protocol (SDP) and the Offer/Answer model of SDP are used. Below, a proposal is detailed to extend the Offer/Answer model of SDP to cope with the multiparty scenario. This proposal is based on the support provided by the mechanisms for sending provisional SIP responses reliably. These mechanisms are specified in [7] and its support is mandatory for any UE accessing to IMS.

The proposal is as follows:

- The originator UE includes an SDP offer in the INVITE request, which is sent to each destination UE. This offer mainly contains the set of media components that the originator user wants to exchange during the session, the set of supported codecs for each media component and the bandwidth requirements and addressing information for each media component at the originator side.
- Afterwards, each destination UE answers back with a first SDP answer within a Session in Progress message. This answer may discard some of the media components proposed in the SDP offer. For each accepted media component, the destination UE indicates in the SDP answer the codecs it supports out of the codecs that were present in the offer.
- Eventually, the Session in Progress responses go back to the multiparty AS. The AS waits until all the responses are received (up to a maximum configured timeout), and stores the corresponding SDP answers for further use. The received SDP answers will be used to compute a combined SDP answer, which will be sent to the originator UE. This SDP answer is created as follows:
  - For each media component that is present in the SDP offer, if no SDP answer accepts the media component then it is discarded in the combined answer, and it is indicated by setting the corresponding media port to zero. In other case, the media component is accepted in the combined SDP answer.
  - For each accepted media component, the AS derives the codecs that will be present in the combined SDP answer. Only those codecs that were accepted by all the destination UEs that accept the media component will be included in the answer. If there are no codecs in common for one media component, then the media component is discarded in the combined SDP answer.
- After receiving the combined SDP answer, the originator UE selects one single codec for each accepted media component and sends a second SDP offer in a PRACK request (point 3 in figures 2 and 3). This PRACK request will be routed to the multiparty AS.
- The multiparty AS receives the PRACK message with the second SDP offer. As it has been explained before, this PRACK message is replicated for each destination UE. The SDP offer which is included in any specific replica for a destination UE is constructed as follows:
  - Any media component that was discarded in the previous SDP answer received from the UE will be discarded.
  - Any media component that was accepted in the previous SDP answer received from the UE will only be accepted if it has been accepted in the second SDP offer received from the originator UE. In any other case, the media component will be discarded.

  This way, the destination UE always receives an SDP offer which is consistent with its previous SDP answer and the result of the negotiation
- Every destination UE will receive the PRACK request, containing the second SDP offer. Now, each UE accepts this second offer and sends a confirmation by means of a second SDP answer which is sent in a SIP OK message (point 4 in figures 2 and 3).
- Eventually, the SIP OK response reaches the multiparty AS. The AS waits until it receives enough OK responses so as every media component that was present in the SDP offer received from the originator UE is accepted by at least one destination UE. When this happens, it generates a combined SDP answer with all the accepted media components and sends this SDP answer within a SIP OK message to the originator UE.

  In certain situations, the multiparty AS may timeout without having received the confirmation for some media components. In that case, it is assumed that the communication path with the subset of destination UEs that accepted the media components does not exist any more, and the combined SDP answer which is sent to the originator UE discards these media components by setting the corresponding media ports to zero. In addition, the AS is prevented to send any further SIP requests to this subset of UEs.

One possible improvement for this schema, in case that the codecs in common for an specific media component is null after evaluating the first combined SDP answer, could be to select the subset of codecs that maximizes the number of destination UEs that might participate in the session. Another more complex possibility would be to introduce transcoding facilities in the user plane.

C. Resource reservation

After the session description negotiation, all the participant UEs agree on the media components that will be exchanged during the multimedia session, as well as on the codec that will be used for each media component. Now, each UE will have to establish the PDP contexts that will be necessary to transport the media components between the UE and its corresponding GGSN. This procedure states for resource reservation, and it will be initiated by the originator UE after sending the PRACK request and by each destination UE after sending the OK response corresponding to the PRACK request.

Each UE will establish up to two PDP contexts for each accepted media component:

- One PDP context for the transmission of the media component in the upstream traffic direction (i.e. from the UE to the GGSN).
- One PDP context for the transmission of the media component in the downstream traffic direction (i.e. from the GGSN to the UE). This PDP context will be shared between all the destination UEs that receive the media component and that are served by the same SGSN and GGSN. So, whenever the GGSN receives a multicast packet for a multicast group that has associated a PDP context, the GGSN will transmit the multicast packet by

the PDP context. The multicast packet will reach the SGSN that, in turn, will replicate the packet for each cell where multicast receivers are present. Finally, all the multicast receivers located in that cells will receive a copy of the packet. This way, the transmission of the multicast packets is done in an efficient way, between GGSNs by means of network level multicast routing, and from the GGSN to the receiver UEs by means of the shared PDP contexts.

So, each media component will be assigned the desired QoS independently from the rest of the media components. In addition, several UEs served by the same SGSN and GGSN will be able to exchange an specific media component in an efficient way, each of them transmitting the media by means of an independent PDP context and all of them receiving the media within a shared PDP context. On the other hand, by having separate PDP contexts for the upstream and downstream directions, the procedures related to the resource management are simplified, such as is not necessary to modify the downstream PDP context whenever a new user joins or leaves the session.

Nevertheless, the establishment of PDP contexts may fail, for example when there are not enough resources available in the radio interface. On the other hand, each destination UE should not alert its user about the incoming session unless the session can be really established at least with the originator UE. For this reason, the establishment of PDP contexts should have succeed both in the destination UE and in the originator UE before the destination UE starts ringing. To guarantee this, the preconditions mechanism specified in [8] will be used, in the same way as it is used in a typical one-to-one IMS session. The support of the preconditions mechanism is mandatory for any UE that access to IMS. This way, the preconditions mechanism is used, assuring that:

- The originator UE will send an UPDATE request (point 5 in figures 2 and 3) to each destination UE when all the necessary PDP contexts are successfully established at the originator side. This UPDATE request will contain an third SDP offer, indicating that the local resource reservation has been successfully completed. The UPDATE request will be received by the multiparty AS, that will send one replica of this message to each destination UE with a consistent SDP offer.
- Each destination UE will not send the RINGING request until it receives the UPDATE message from the originator UE and has successfully established all the necessary local PDP contexts.

The UPDATE request is answered from each destination UE with a SIP OK response (point 6 in figures 2 and 3), which contains an third SDP answer with information about the status of the local resource reservation in the UE. After receiving the first OK response the multiparty AS modifies the SDP answer if necessary, so as to be consistent with the SDP payload expected by the originator UE, and routes the possibly modified SIP OK response to the UE. Subsequents SIP OK responses to the PRACK request are simply filtered out by the multiparty AS.

Finally, when any of the destination UEs finishes with the resource reservation procedures and once it has received the UPDATE message, it alerts its user, i.e it starts ringing, and it sends a RINGING response back to the originator UE (point 7 in figures 2 and 3). This RINGING response will eventually reach the multiparty AS. After receiving the first RINGING response, the multiparty AS routes it towards the originator user. From the point of view of the originator UE, this RINGING response means that at least one of the destination UEs has finished with the resource reservation procedures and it is alerting its corresponding end user.

The originator UE confirms the reception of the RINGING response with a PRACK request (point 8 in figures 2 and 3), which is routed towards the multiparty AS. This PRACK message will be sent from the AS in response to any received RINGING message. After receiving the PRACK request, the receiver UE answers back with a SIP OK response (point 9 in figures 2 and 3) that finally reaches the multiparty AS.

When the first SIP OK response, corresponding to any PRACK request is received by the AS, it routes the response towards the originator UE. Any subsequent SIP OK message, confirming the reception of the PRACK request, is filtered out in the multiparty AS.

To conclude with this subsection, there are some refinements to this basic scheme that must also be taken into account:

- When the SIP OK response, containing the second combined SDP answer, arrives to the originator user, if any media component that was accepted in the corresponding SDP offer sent by the originator UE appears as discarded in the answer, then the originator UE will have to modify the resource reservation, by deactivating the PDP contexts corresponding to the discarded media components.
- On the other hand, once the resource reservation has succeeded at the originator side, the originator UE can subscribe itself to all the multicast groups corresponding to the media components that it is going to receive. This is done by means of the IGMP protocol, which is executed between the UE and the GGSN.

D. *Concluding the session establishment*

Finally, when the first destination user accepts the session (by pressing the Accept button in its mobile terminal), the following actions are performed by the UE:

- First, the UE subscribes itself to all the multicast groups corresponding to the media components that it is going to receive. This way, the UE guarantees that it will start receiving traffic for that media components from then on.
- Next, it sends a SIP OK response to the originator UE (point 10 in figures 2 and 3), indicating that the end user has accepted the session establishment.

The SIP OK response will eventually be received by the multiparty AS. Upon receiving the SIP message, the AS routes the SIP OK message to the originator UE. The UE confirms the reception of this message by means of a SIP ACK request (point 12 in figures 2 and 3). The main problem now is that this OK response does not contain any information about the media components that the sender of the message is waiting to receive. So, at this point, the originator UE knows that the session is established but it does not know anything about the media components it can start transmitting for. To solve this problem, the SIP specific event notification framework detailed in [9] will be used.

A new event package will be designed according to [9] that will allow the AS to notify the originator UE the IP addresses of the multicast groups the originator UE can start transmitting media for. This information will be sent in a SIP NOTIFY request (point 11 in figure 2) from the multiparty AS to the originator UE immediately after the SIP OK response is sent. Upon receiving the NOTIFY request, the originator UE answers back with a SIP OK response (point 13 in figure 2) to the multiparty AS, and can start transmitting media for the multicast groups indicated in the SIP NOTIFY. This media will finally reach the UE of the destination user that accepted the session establishment.

E. Notifying changes within the session

Whenever another end user accepts the session establishment, the procedure is repeated until the SIP OK response, sent from the destination UE, reaches the multiparty AS. As at this point the session is already established, the AS:

- Examines the list of media components that were accepted by the destination UE in the session description negotiation. For any accepted media component that is going to be received by the destination UE and that has not been yet notified to the originator UE, the multiparty AS includes the address of its corresponding multicast group in a NOTIFY request and routes it towards the originator UE.
- Next, the AS sends a SIP ACK request to the destination UE (note that the SIP OK message is not routed to the originator UE).

The event package could be extended, in order to notify the active UEs the changes that result from the users joining and leaving the session. For instance, whenever a new SIP OK message is received in the AS, indicating that one user is joining the multimedia session, the AS could send a SIP NOTIFY message to all the participants that have joined the session up to that moment. In this NOTIFY, the AS could include any information about the new member, such as its identity and the media components it will exchange with the rest of participants, or even any information about the current session status, such as the number of active participants, the media components that are being exchanged and the number of participants that are exchanging each media component.

V. Conclusion

IMS provides a framework that allows the negotiation of the value added multimedia services that are envisioned for UMTS. These services, that are peer to peer in nature, are delivered in the user plane by means of the IP protocol in an unicast based approach. But this unicast approach is not optimum, in terms of transmission efficiency and scalability, for services that involve the participation of multiple users in a multiparty arrangement.

With the previous considerations, this paper has proposed to use a multicast approach to provide multiparty services over IMS. The main advantages and drawbacks related with the implementation of this solution on the UMTS infrastructure have been analyzed, and the key elements to develop the proposal have been highlighted. As an starting point, a SIP signalling dialogue that allows to negotiate a generic multiparty service over IMS has been presented, supporting at the same time the configuration of a network level multicast delivery service with QoS provision capabilities.

However, more scenarios will have to be considered before having a complete multicast solution, such as the session release cases and issues regarding the SIP signalling delays. On the other hand, the details concerning the shared PDP contexts to receive the multicast traffic must be provided, looking for synergies with the MBMS specifications. Finally, in this article the UMTS access network has been considered. Nevertheless, future work will extend the proposed solution to other access network technologies, with the purpose to provide the multicast delivery service in IMS based Next Generation Networks scenarios.

Acknowledgment

This article has been partially granted by the European Commission through the MUSE project (IST-026442), and the Spanish MEC through the CAPITAL project (TEC2004-05622-C04-03/TCM).

References

[1] The 3rd Generation Partnership Project (3GPP), "http://www.3gpp.org."
[2] G. T. . version 7.5.0:, "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); IP Multimedia Subsystem (IMS); Stage 2."
[3] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, and E. Schooler, "SIP: Session Initiation Protocol," RFC 3261 (Proposed Standard), June 2002, updated by RFCs 3265, 3853, 4320. [Online]. Available: http://www.ietf.org/rfc/rfc3261.txt
[4] J. Rosenberg and H. Schulzrinne, "An Offer/Answer Model with Session Description Protocol (SDP)," RFC 3264 (Proposed Standard), June 2002. [Online]. Available: http://www.ietf.org/rfc/rfc3264.txt
[5] M. Handley and V. Jacobson, "SDP: Session Description Protocol," RFC 2327 (Proposed Standard), Apr. 1998, updated by RFC 3266. [Online]. Available: http://www.ietf.org/rfc/rfc2327.txt
[6] 3rd Generation Partnership Project, "3GPP TS 22.146: Technical Specification Group Services and System Aspects; Multimedia Broadcast/Multicast Service; Stage 1 (Release 8)."
[7] J. Rosenberg and H. Schulzrinne, "Reliability of Provisional Responses in Session Initiation Protocol (SIP)," RFC 3262 (Proposed Standard), June 2002. [Online]. Available: http://www.ietf.org/rfc/rfc3262.txt
[8] G. Camarillo, W. Marshall, and J. Rosenberg, "Integration of Resource Management and Session Initiation Protocol (SIP)," RFC 3312 (Proposed Standard), Oct. 2002, updated by RFC 4032. [Online]. Available: http://www.ietf.org/rfc/rfc3312.txt
[9] A. B. Roach, "Session Initiation Protocol (SIP)-Specific Event Notification," RFC 3265 (Proposed Standard), June 2002. [Online]. Available: http://www.ietf.org/rfc/rfc3265.txt