

New QoS Control Mechanism Based on Extension to SIP for Access to UMTS Core Network over Hybrid Access Networks

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Abstract— creating end-to-end QoS in heterogeneous wireless-wired networks beyond 3G networks in which the access to the IP core network can be accomplished via different kinds of access networks with different technologies is essential for supporting real time application. In this paper we have defined new functionalities and interfaces in addition to some extension to the existing SIP signalling to resolve some of the existing problems existing in UMTS that don't let end-to-end QoS control between different technologies and domains.

Index Terms—, Policy Based Networks, E2E QoS, SLA.

I. INTRODUCTION

The Universal Mobile Telecommunication System (UMTS) community has been increasingly looking for an architecture that can provide consistent network-independent end-to-end IP QoS that is an essential for supporting real-time application and services. Flexible QoS establishment need tight coordination between session and bearer level and it can't be considered only as bearer level task. Because even if different operators in different domains have agreement on the IP QoS requirements of a specific service, they may configure their network elements in different ways. This is why 3G has chosen a policy-based architecture to provide this co-ordination. IP Multimedia Subsystem (IMS) is being standardised by 3GPP as an overlay on the basic architecture of 3G to provide globally the services for the subscribers [1]. SIP which is an IETF standard is used in IMS as the signalling protocol in 3G networks to control multimedia sessions. CSCF (call session control function) proxy and server are introduced in IMS to establish the session between two call parties and prepare the required services according to the session characteristics. Besides, P-CSCF (Proxy CSCF) acts as the *policy decision function* and communicates with GGSN which works as the *policy enforcement point* via Go interface. The policy rules describe the amount of network resources required for different

multimedia services without going to the details of network elements configuration. Policies can handle network configuration, including Quality of Service (QoS), Service Level Agreement (SLA), Virtual Private Network (VPN), and security issues [6]. Session signalling indicates the required resources with the Session Description Protocol (SDP) inside of SIP messages. The Policy Decision Function (PDF) examines the authorisation of the user for the requested resources and then if the user passed this examination the proper policies will be transferred to the Policy Enforcement Point (PEP) in the data flow. Finally, IP QoS signalling (like RSVP) reserve the resources. In this architecture, the operators negotiate the SLA for their QoS services they have mutually contracted to provide. Hence, each operator defines its local policies based on the negotiated SLA and applies it to its network elements to implement it. However, establishing end-to-end QoS for Multimedia services in all IP mobile network beyond 3G networks toward 4G networks, which is leading more and more to heterogeneous all IP wired-wireless networks, where access to the IP-based core network can be accomplished via different kinds of wireless-wired access networks with different technologies like UTRAN, WLAN, xDSL, CATV... and data flow of a multimedia session may pass through different technologies and administrative domains, is more sophisticated and needs some development in the architecture and signalling flow of the existing configuration. From the architecture point of view, there is no a way between the access and core network or even between different domain edge proxies to exchange the policies and limitation of their network dynamically and efficiently. On the other hand, from the signalling point of view, in the current session signalling, in the SDP inside of the SIP messages the only QoS parameters that can be indicated by the user are codec and bit-rate [3] and the user can not express exactly his expectation about the QoS level of the required multimedia service; although it doesn't mean that the user receives a bad QoS but the user may wishes to have the choice in selecting the level of QoS for the same service because of the cost or end-device capabilities. For example, with the current QoS parameters in SDP, "video call" will be exactly mapped to a certain QoS class beyond of user choice but for a long international video call, the caller may desires an acceptable QoS but not a high quality to reduce his costs. In our work we have suggested some extension to SIP to exchange some additional QoS level information to satisfy the user QoS expectation for the requested multimedia service and help

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different administrative domains (or even different network technologies) negotiate SLA dynamically. In the rest of the paper we will explain the policy-based architecture of 3G networks and current session signalling flow. Then we will present our solution on the base of SIP to overcome the existing weaknesses.

II. POLICY BASED ARCHITECTURE IN 3G NETWORKS

Establishment of session for multimedia services like voice or video telephony, video streaming, messaging, multimedia gaming or virtual reality, needs co-ordination between bearer and session layer for QoS. After release 5, UMTS architecture is a layered architecture with a clean split between bearer (eg. SGSN, GGSN), session (eg. P-CSCF, S-CSCF) and service level [1]. But providing QoS is not only a bearer level issue and session layer should be involved too. This is why 3GPP has chosen the policy based architecture to provide high quality transport media with efficient resource utilization. In a policy based network, policy rules describe behavior of the network in some high-level statement without going to the detail of network element configurations. In fact, policy rules are a set of conditions and instructions; whenever a request for a service fulfills a condition, the corresponding instruction will be performed. Figure 1 has depicted the proposed policy based architecture by IETF [9,10]. Four major functional entities are defined:

Policy Repository: All the policy rules exist in this entity. Policy Repository is usually implemented inside Policy Decision Point or separately as a LDAP (light-weight directory access protocol) directory server.

Policy Decision Point (PDP): This is logically a centralized entity that makes the policy decision according to the policy rules and the dynamic and static information of the network.

Policy Enforcement Point (PEP): PEPs enforce the policies in the network. They are network elements (especially edge routers) that will realize the policies for the resources by using software and hardware features (scheduling, queuing, classifying, traffic policy and shaping) in the network.

Policy Administration System: This is the point in which the operator define his policies. Policy Administrator System pushes the defined or modified policies to the Policy Repository and informs the PDPs about any modification in policies.

In policy management systems, there are two main models for interaction between PDPs and PEPs: *provisioning* and *Outsourcing* [10]. In the *provisioning* model, PDP decides which policy rules should be installed on PEP and then provision it for the resource reservation request coming to the PEP. In contrast, in *Outsourcing* model, a resource reservation request coming to PEP will trigger the process of policy request from PDP. Each model has some benefit and disadvantages. For example the Outsourcing increases the signaling load but it is more dynamic for special cases like link failure or time-dependent policies.

Now, to see how the general policy based architecture is adapted in UMTS architecture to establish end-to-end QoS for session based multimedia services, let's take a closer look on

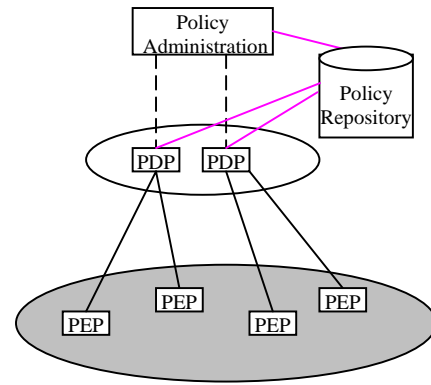


Fig 1. Policy based Architecture

the defined architecture for traffic and signaling transport. According to the popularity and variety of IP multimedia services, 3GPP has chosen IP as the data transport technology. Besides, to establish end-to-end QoS, a layered “bearer service” architecture [3] (figure 2) is proposed. In such architecture several bearer services should be established between network elements in different domains between call parties. In UMTS part, the QoS parameters between end-user and GGSN will be negotiated by using Packet Data Protocol (PDP) context [4]. But in the external network, although 3GPP has not standardized a special IP QoS method, but DiffServ is being used in GGSN to classify the IP flows [3,4]. On the other hand, with considering the fact that IP is chosen as the data transport protocol, SIP which is an IP based signaling protocol for setting up the sessions, is chosen by 3GPP (and it seems rational). Then, IMS (IP Multimedia Subsystem) was introduced in release 5 (and is being developed in releases 6 and 7) as an overlay on UMTS PS (Packet Switch) to support IP multimedia services. The data traffic is still managed by PS elements but new control functionalities are defined in IMS to control the session signaling and provide the multimedia services globally [1]: Media Gateway Function (MGCF), Media Resource Function (MRF) and Call State Control Function (CSCF). There are three kinds of CSCF: P-CSCF which acts as the SIP Proxy is the first contact point inside of IMS for user equipment (UE). The Serving CSCF (S-CSCF) resides in the home network and control the session by enforcing the service profile of the user via accessing to the home subscriber server (HSS). And the last one, Interrogating CSCF (I-CSCF) hides the network configurations for the external connections. As a first contact point for a SIP request message (which convey requested QoS specifications of the service inside in SDP) from a user, it seems rational that P-CSCF is chosen as the Policy Control Function (PCF). PCF acts same as PDP and enforces the policy rules to the PEP. GGSN as the gateway of data flow to external network acts as the PEP and translate the policy rules to the IP flow control functions (labeling diffserv flow and traffic classification, Scheduling, Traffic Policy and admission Control, Traffic Shaping). To open a gate for a resource reservation request of a data flow, the PEP component of GGSN must verify the request with PCF in the signaling path. The Go interface make

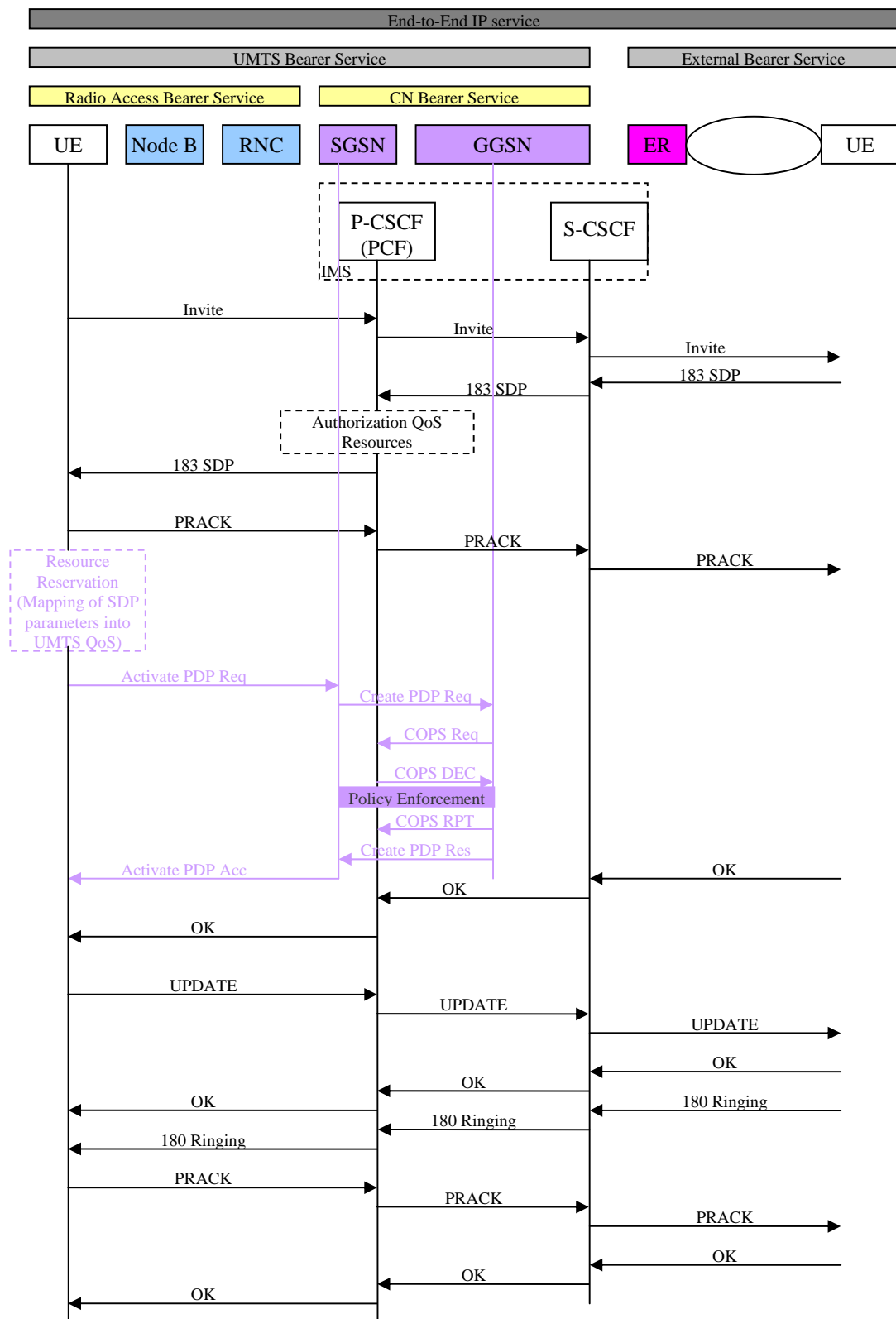


Fig. 2: Session Signaling Flow

this co-ordination feasible. 3GPP has agreed on COPS-PR protocol as the communication protocol on the Go Interface [5].

III. SESSION SIGNALLING FLOW AND QOS CONTROL MECHANISM IN UMTS

3GPP has defined the signaling flow (figure 2) of resource reservation for session based Multimedia application using SIP. In fact, establishing a UMTS multimedia session has three phases [2] :

1) Session Initiation: negotiation of the media components between calling and called party. The caller sends an INVITE message containing its supported multimedia capabilities for the session. The called party responds with a 183 (SDP) and indicates its acceptable multimedia parameters. The PCF element of P-CSCF on the way of this signaling flow, evaluate the service QoS characteristics inside of the SDP and add an authorization token to the message if it doesn't conflict with the user service profile. Then, the caller will acknowledge the receipt of 183 (SDP) with a PRACK.

2) Resource Reservation: After the two end-users have agreed at session level on the media characteristics to be used for the session, the resources for the media flows can be reserved at bearer level according to the policy of the network domain. On the UMTS access network, resource reservation means PDP context activation. Then, the QoS signaling translation should be accomplished in different elements of the data path as depicted in Figure 3. But for the cases that the access network is another IP based network, resource reservation in local access network can be based on the IP resource reservation protocol like RSVP or DiffServ.

3) Session Completion: First the calling party notifies the called party that resources are reserved (with UPDATE message). The called party is alerted and the calling party is informed. When the called party hooks off, the session is established and required bearers are dedicated to call.

The end-to-end QoS control mechanism defined by UMTS is limited to a single domain and doesn't work well for multi-domain data path or inter-technology inter-operation. In [6] the limitations of the existing system are well explained as follow:

1) There is no E2E resource based admission control: The PCF will authorize all resource reservation if a session at application level can be established. The GGSN can perform local resource based admission control and won't care of service network or external network!

2) PCF is limited to SIP signaled services: PCF is supposed to be in P-CSCF which is a SIP Proxy. So it can support only SIP based multimedia application. Although in release six this problem will be resolved by defining the proper interface between application servers and PCF.

3) PCF scope limited to GGSN: PCF only serves GGSN as the policy control function and doesn't control other network elements such as inter-domain edge routers.

In the next section the architectures that permit end-to-end QoS for heterogeneous networks with resolving the existing architecture will be discussed.

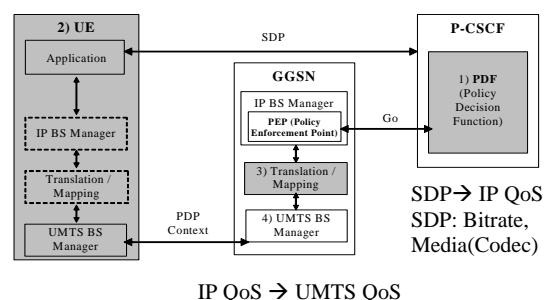


Fig. 3 : QoS Parameter Translation

IV. ARCHITECTURE OF HETEROGENEOUS IP MOBILE NETWORKS FOR END-TO-END QOS

As discussed in previous section, the defined end-to-end QoS architecture by 3GPP has some limitations that can't support E2E QoS for multi-domain data path and in addition, the existing architecture is not flexible enough to support access of different networks with different technologies to the core network.

The existing limitations can be divided in two categories: 1- architectural problems. 2- Weakness of signalling protocols. In this section the possible architectures for different situation which can resolve these limitations will be discussed and our solutions for enriching signalling protocols will be presented in next section.

When we are accessing to IMS via another access network we need some more co-ordination between session and bearer layers; because, the QoS signaling and protocol, in addition to availability of resources in access and UMTS-CN can be completely different. For example, the IP QoS protocol in access network can be Intserv and in UMTS-CN can be Diffserv. In addition it is very likely that the four QoS classes defined in the UMTS [3] don't have exact equivalents in other access networks. In [6] an architecture like what is depicted in figure 4 is proposed so that PCF can control the edge router of other access networks too. This is a good solution for the cases that: a) the operators of all access networks are the same or b) there is a big trust between two operators and the access network operator has agreed that the policies be pushed by the core network operator. To cope with this problem two other architectures are proposed: in figures 5 the Local PDF (LPDF) will exchange the policies with the PCF in the IMS (PCF) and control the edge router of the access network. In the one proposed in figure 6, Local Policy Repositories of each accesses network will exchanges their policies with a shared S-PDF and the S-PDF will control the edge routers of all access networks. Each architecture has its benefits and drawbacks and the use of them depend on the policies and capabilities of the access network operators. In the first architecture, for example for the SIP based applications, the L-PDF should support SIP and acts as a SIP proxy and this push more cost but is more dynamic for policy enforcement according to the local policies. This method is more suitable for the access networks which had had this kind of proxy for their local services regardless of their connection to core network of UMTS; then

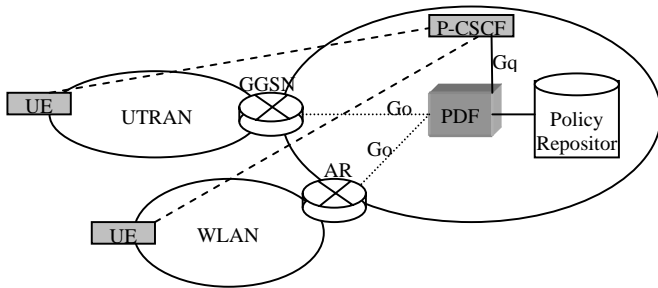


Fig 4 : Modified Architecture for Multi Domain E2E QoS :
All the Edge/Access routers are controlled by the policies defined in UMTS core network

by upgrading the existing proxies, a flexible and dynamic policy control for end-to-end QoS control will be possible.

On the other side, in the architecture of figure 6, there is no need for supporting the session signaling in the access networks and then the cost will be decreased. But first, the policy exchange can't be as dynamic as the previous architecture and second, the S-PDF may be the bottle-neck of the system.

V. THE PROPOSED ARCHITECTURE FOR ENRICHING SIGNALING FLOW FOR BETTER E2E QoS CONTROL IN HETEROGENEOUS DATA PATH

To reach a proper E2E QoS control over the heterogeneous networks for the multimedia application, a tight co-ordination between bearer and signaling level is necessary. The policy based architecture proposed by 3GPP has some limitation from the view point of signaling and policy rules for this co-ordination.

From the side of policy rules, it should be considered that the PCF as specified in IMS does not evaluate the policy rules and only authorize the services and negotiate the resources locally in application layer.

From the side of signaling, we can note that QoS parameters that can be extracted from current SDP in the body of SIP messages are too poor to allow the user to express its expectation about the QoS level he wishes to receive for each media component. The only QoS parameters that can be extracted from the SDP are codec and bit-rate. So with this level of information there is no way except a one to one mapping between SDP QoS parameter and UMTS QoS classes as defined in [4] (table 1). Hence, it is impossible for user to have different level of QoS for a certain media. (e.g. Video with low quality). We have defined some extension to SIP to solve some existing problem and facilitate the co-ordination between bearer and application level for resource reservation and allow the UE to express exactly its required QoS level.

Some extensions to SDP are defined in two categories in [8]. The Traffic Information (TI) and the Sensitivity Information (SI) are added to the information of an SDP message. TI characterizes the traffic type of the bearer associated with codec (bandwidth, packet size). But SI defines the parameters like end-to-end delay, delay jitter and maximum packet loss that defines the level of quality that a user wish to have. In [7] an extension to SIP named Q-SIP is introduced where QoS

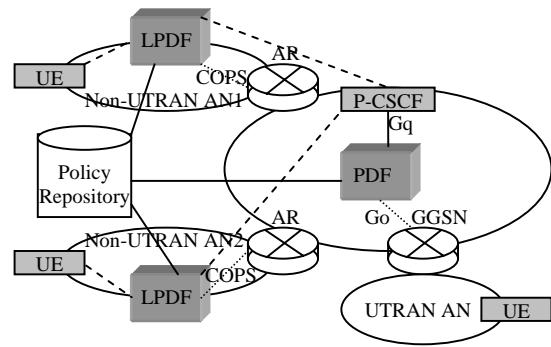


Fig 5 : Modified Architecture for Multi Domain E2E QoS :
The access networks own their Local PDF to control the AR

information will be carried by a SIP INVITE message in a

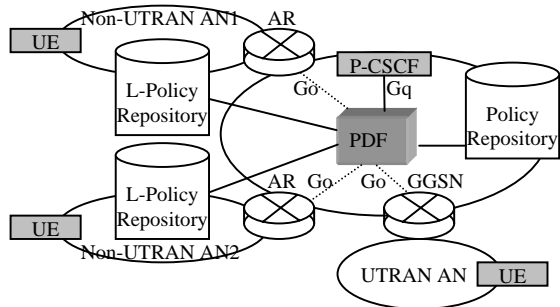


Fig 6 : Modified Architecture for Multi Domain E2E QoS :
The access networks don't have their Local PDF to control the AR. But defines their policies themselves

manner that keep backward compatibility to the standard SIP elements. The proposed Q-SIP proxies detect these QoS messages and use them for resource reservation.

The architecture defined in [7] makes some possibilities to exchange dynamic SLA between end-user and service network but not for inter-domain and inter-technology architecture.

We have considered some extension to SIP so that, a user can adds its context in the Registration time and TI and SI to the Invite message on the time of starting a session. Besides, in our proposed methodology, the border proxies have the capabilities to negotiate dynamically their policies.

In this section we have considered the architecture defined in figure 5 to explain the detail of signaling in the following:

In the Registration process, the UE adds its context (equipment capabilities, battery life time, ...) into the Register message. This is very important because the user can connect to the network with different equipment in different access networks; so, the S-CSCF in the home network will be informed about the capabilities of the UE and decide about the sessions destined to this user without informing him. To give an example, consider a user who has the video call in his service profile but in a special situation he has registered to the network via an access network and equipment who can't support video. Then, when the home S-CSCF receives a video call destined to this user, it can inform the caller to change its

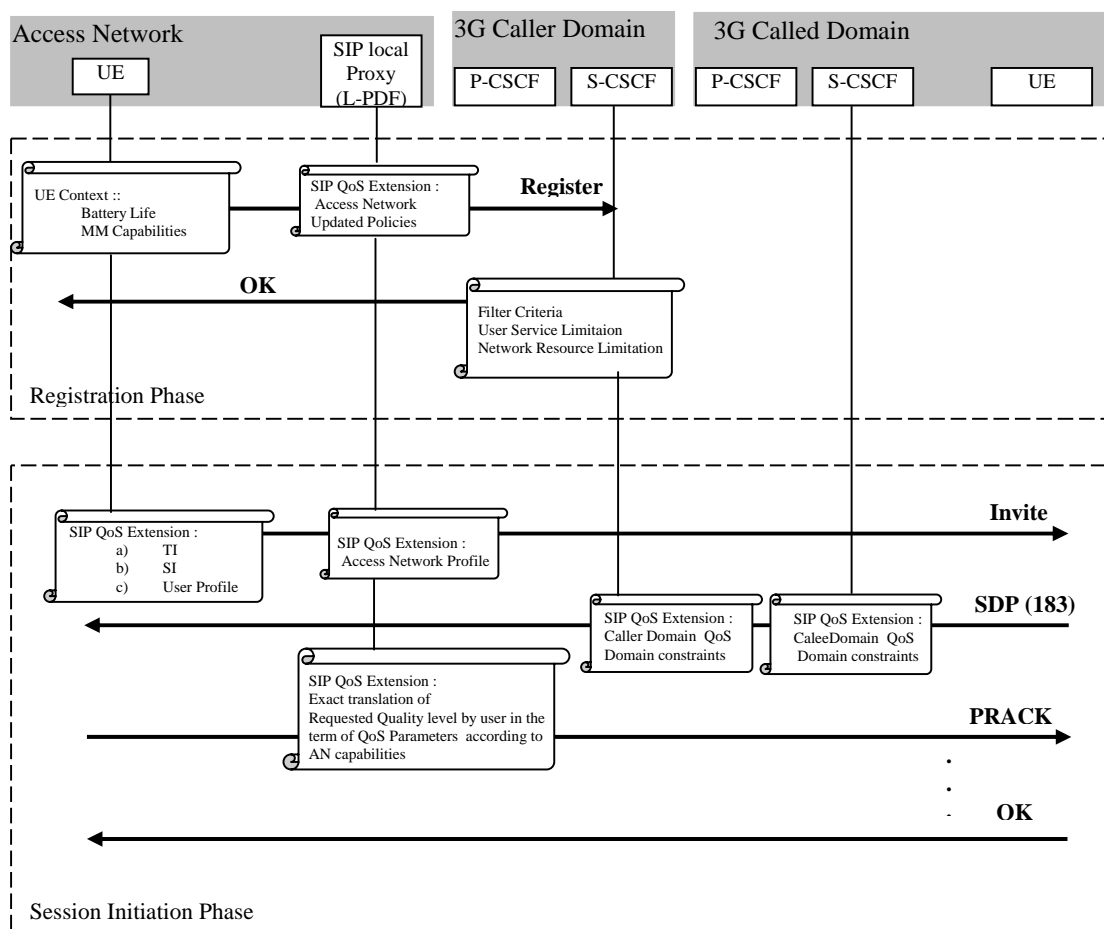


Fig. 7 : Extended Signaling Flow

session characteristics before informing the called user. This can reduce the signaling load of the network especially when the called user is in a visiting network. Before the Register request reaches the S-CSCF in the home network all the border proxies can benefit this Register request and add some information about their modified policies to negotiate their policies without pushing extra load to the network for exchanging new updated policies. It means that although the static SLAs between operators will be exchanged in a provisioning method but time dependent policies can be exchanged in SIP signaling dynamically without increasing the signaling load. The main useful SLAs that can be exchanged with this signaling method are the resource assignment policies for different services according to the time and load of the network. These information is very important according to the fact that although different networks compromise on policy rules but the way that they configure their network elements can be different (according to the technologies and the algorithms used for this purpose). To reach a better co-ordination informing other domains about the amount of resources they allocate to different services and the general capabilities of their data path elements by using SIP messages can be really useful. On the other side, S-CSCF after receiving the registration request, by accessing to the HSS invoke the user profile and filter criteria and inform the

PDFs in the visiting core and access network about the special policies for the user (if exists). With this method PDF can really examine the policy rules for a session and doesn't act only as a service authorization entity. Figure 7 shows the Registration signaling flow with these extensions.

After registration, when the user wants to start a session, the UE sends the INVITE message and as an extension insert SI and TI information (which contains the exact QoS level expectation of user).

The L-PDF in access network, will examine this invitation and if none of the requested media satisfied the exchanged policy rules between operators the request would be declined but if not, it will forward it. In addition in special cases, L-PDF may find out that according to the updated local policies the amount of resources negotiated for the services indicated in the receiving Invite message is changed, therefore it will add the new QoS conditions to the message. The UE in called party (or the destined application server in service domain) receives Invite and choose one of proposed media capabilities of caller and send back an SDP (183) message. The S-CSCF in called party (or service domain) will add its constraints about resources and forward it. It is important to note that, according to the fact that S-CSCF can have access to the MRCF (Media Resource Control Function) in the called (service) domain which acts as a Bandwidth Broker. And by adding these

information to the SDP(183) message, the PCF in caller domain can be informed about the resource constraints in the destination domain. When this SDP reaches PCF inside of the P-CSCF in caller network, now according to the updated information about local and called domain (service) resource limitations and other negotiated policies in PCF the authorization token will be issued for the user.

By using these extensions, first of all, the authorization token won't be issued only based on resource negotiation in application layer anymore and the resource availability in local and service domain will be considered too. This is very helpful, because when the resource reservation begins the probability of success resource reservation increase and therefore the signaling load will be decreased. Because in this strategy, if the resources are not available in network elements, it will be detected in signaling stage and authorization token won't be issued anymore. Second, this method let the user to express its exact expectation about the QoS level.

Table 1: Mapping of SDP media to UMTS QoS Classes defined by 3GPP

Media inside of SDP	UMTS QoS Class
Audio	Conversational or Streaming
Video	Conversational or Streaming
Application	Conversational
Control	Interactive Priority 1
Data	Interactive Priority 3
Others	Background

VI. TEST-BED

Our platform consists of six PC Pentium IV 2.4Ghz in Linux Redhat 9, one 100Mbps Eth AP, one 1Gbps Cisco Router and six Labtops with WiFi Card as the user equipment. In our test-bed we have implemented the architectures depicted in figures 4-6. The PCs are used to implement SIP server and Policy repositories. The open source software used in our platform are: SER v09 developed by IPTEL [11] for SIP server, RADIUS server v0.9.1 and RADIUS client v0.3.2. We have used COPS 1.4 developed by VOVIDA [12] as the interface between Policy decision point (which is implemented in SER SIP server by our team) and AP and core router which act as the policy enforcement points (PEP). The modifications on SIP signaling to add the defined extensions to the SIP are done by our team. These modifications are down in two main categories: SIP signaling and functional elements.

For SIP signaling, firstly we needed to add PRACK and UPDATE requests defined by 3GPP as extension to SIP and secondly, adding our new SLA and QoS extensions depicted in fig. 7 to the SIP signaling body.

On the other hand we have developed the functionality of User Agent (UA) and SER as the SIP server to detect the extensions defined in the SIP messages and triggers the new functions defined in them for these extensions.

VII. TEST SCENARIOS AND RESULTS

Two main aims have been followed in the proposed architecture and our implementation:

First, we desire to give the user the freedom in choosing their QoS level more freely; instead of a strict mapping between a special kind of application and QoS class (like what is depicted in table 1). It means that for example for voice the user has freedom to ask for High Quality, Acceptable or Poor service and then these QoS levels will be mapped to one QoS class as depicted in Table 2. But in addition to this, for VoIP, we implemented a simple middleware to adapt the application codec to the requested QoS level. The idea is that when we choose for example Poor level which is mapped to Interactive Priority 2 class, because the packet loss rate will be more, it is better to choose a codec with smaller packet-length; so in the case of packet loss a smaller amount of information will be lost and the quality will be less affected. Hence for voice we support three codec: G.711 (64 Kb/s), GSM-EFR (12.2 Kb/s) and G.729 (8 Kb/s). The results are shown in Fig. 8 for the high load and light load traffic and quality level of Poor. The bar chart is obtained with a survey over 20 persons rating on a five-point scale (MOS) from 1 (bad) to 5 (excellent) for the quality of the voice. As it is clear, in the light load of the network the performance of two methods are similar but for the High load (high packet loss) because the quality of voice with G.711 elapse dramatically, the service adaptation method will have a recognizable better quality by switching to the G.729 codec.

Second, we aimed to test the situation that user is asking for a service or service level but it conflicts the network available resources. In this condition we desire to test the capability of our proposed architecture to consider resource availability and authorize service requests not only in application layer but also with considering available resources; and stop such a conflicting session initiation request in early phase of its emission. Because in conventional architectures such a request that conflicts the policies not only won't be stopped in the access network in the early beginning of its creation but also it won't be stopped in the signaling negotiation phase; the process will be continued until resource reservation will be started and the bandwidth broker find that there is no possibility to reserve the requested resources for this user. To implement this scenario, the user will send an invite and he will indicate the desired TI parameters somehow that conflicts with local policies of the access network.

Table 2: Different QoS level and their mapping to QoS classes

Real Time Media and QoS level	UMTS QoS Class
Audio, Video High Quality	Conversational
Video Acceptable	Streaming
Audio Acceptable	Interactive Priority 1
Audio, Video Poor	Interactive Priority 2

Table 3: Results of resource admission control in signaling phase

	Traffic Load in Access Network	Asked QoS Level	Conflict Detection Location	Conflict Detection Time	Reaction Location to Conflict	Delay before Reaction
Arch1 (Fig. 4)	High	High Quality	Core Network	After Resource Reservation Process	CN	383ms
Arch2 (Fig. 4)	High	High Quality	Access Network	Before Resource Reservation	AN	48ms

5)			k	Process		
Arch3 (Fig. 6)	High	High Quality	Core Network	Before Resource Reservation Process	CN	72ms

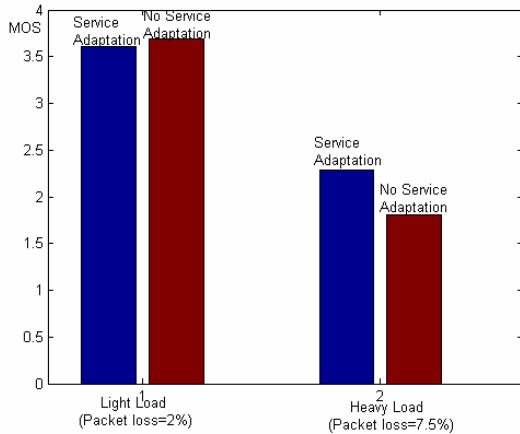


Fig. 8: Evaluation of Service adaptation performance for the voice QoS

The results of this test are indicated in table 3. The Architecture 2 (Fig. 5) has the best performance because the SIP proxies there can detect the conflicts of the TI parameters with local policies and stop the request without propagate it to core network. This improvement is obtained in the cost of implementing a local SIP server who is developed and detect the defined extensions. Architecture 3 (Fig. 6) detects these conflicts in core network; because in access network there is only a simple SIP forwarding proxy. On the other hand, in Architecture 1 (Fig. 4) these conflicts won't be detected in signaling level but when the in resource reservation process after receiving of SDP response from other party (Fig. 2). So with these two methods first, propagation of a request, which conflicts the network resource policies, to the rest of network will be avoided and the overall signaling load in the network will be reduced. Second the delay before detecting the conflict can be reduced until 80%.

VIII. CONCLUSION

In this paper we proposed new architecture with modified functional elements existing in IMS of UMTS to resolve the existing limitations in the architecture and control end-to-end QoS over the data path between different technologies and domains in heterogeneous wired-wireless networks beyond 3G. Some extensions into SIP signaling flows were suggested. With these extensions, a better dynamic policy exchange between different domains will be possible. In addition the PDF won't authorize service requests only in application layer. Instead, it will consider resource availability in both of local domain and service (destination) domain without pushing extra signaling exchange by using SIP messaging in early invitation step without propagating the request to rest of network. With this method the delay of conflict with resource limitation will be reduced until 80%.

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For the MS thesis during the years 1998-99 he worked on improving the performance of Handover algorithms in cellular networks. In 1999 after his MS he joined the DayCo where he worked as a senior design engineer and team header, designing high speed network processors. In 2000 he joined Payam Mashreq Co. where he worked as the project manager in R&D in the field of high speed router designing. In 2003 before starting his PhD he participated in founding SarvNet Co. in Isfahan-IRAN where he was the CEO. His area of interest is Digital Communication Systems (Specially Wireless Systems), Architecture of next generation Telecommunication Systems, Network Management, Network Topology and Measurement, Switching, Security in the Network, Multimedia Systems and Optical Networks.

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