D4.1.3

Final Network Service Provisioning Concept

Version 1.1

WP 4.1 Mobile Network Architecture, Design & Implementation

Dissemination Level: Public

Lead Editor: Nuno Inácio, IT-Aveiro

19/01/2007

Status: Approved by QM

SIXTH FRAMEWORK PROGRAMME

PRIORITY IST-2002-2.3.1.18

Information Society

Grid for complex problem solving

Proposal/Contract no.: 004293

Context

Activity 4	Detailed Architecture, Design & Implementation
WP 4.1	Mobile Network Architecture, Design & Implementation
Dependencies	This deliverable depends on and has influence on (to some extent) work carried out in WP 3.1, 3.2, 4.2, 4.3, 4.4, 5.1, 5.2

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Version	**Date**	**Authors**	**Sections Affected**
0.1	05.12.06	All	Document Created – largely based on existing ID4.1.3
0.2	20.12.06	All	Updates for all chapters
0.3	28.12.06	Nuno Inácio	First draft for revision
0.4	17.01.07	All	Second draft, with corrections from external review
0.5	19.01.07	Nuno Inácio, Patrick Mandic	Transition to new template Updates to security section

Executive summary

This document presents the evolution of the Network Layer Architecture described in D4.1.1. The latter depicted the fundamentals and architecture of this particular view of Next Generation Networks (NGN), in which collaboration and integration with Grid infrastructures is considered a must. After laying the basis for this architecture and corroborating the design by means of a first implementation while taking into account the feedback of the commission on the previous work, a redesign to polish some aspects of the architecture was undertaken and summarized in this document focusing on the changes performed in the architecture and paying special attention to the most innovative issues such the interaction between Grids and Networks – Grid using Network QoS, SIP with SOAP, etc. In addition to these topics, security is also strongly prioritized and focused on the following fields:

- Security aligned to the research done in the project i.e. security issues arisen within the context of the new functionalities developed (e.g. SIP with SOAP interactivity).
- Security as a whole, in order to provide guidelines for the security of the complete platform developed. (e.g. provide authentication and Single Sign On (SSO)).
- Security for the tools used for implementation - e.g. Java stubs. (This one given the least importance in favour of the other two)

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Abbreviations

A4C	Authentication, Authorisation, Accounting, Auditing and Charging
AA	Authentication and Authorisation
AAA	Authentication, Authorisation and Accounting
Akogrimo	Access To Knowledge through the Grid in a Mobile World
AN	Access Network
AoR	Address of Record
API	Application Programming Interface
AR	Access Router
BA	Binding Acknowledgement
BE	Best Effort
BU	Binding Update
CIM	Common Information Model
CIMOM	CIM Object Manager
CM	Context Manager
COPS	Common Open Policy Service Protocol
CPU	Central Processing Unit
DiffServ	Differentiated Services
DoS	Denial of Service
DDoS	Distributed Denial of Service
DSCP	Differentiated Service Code Point
EMS	Execution Management Service
EPR	End Point Reference
FHO	Fast Handover
FBU	Fast Binding Update
FNA	Fast Neighbour Advertisement
gSDP	Grid Session Description Protocol

GUI	Graphical User Interface
GUP	Generic User Profile
HA	Home Agent
HoA	Home Address
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IntServ	Integrated Services
IP	Internet Protocol
IPSec	IP Security Protocol
IPv6	Internet Protocol version 6
LAN	Local Area Network
LDAP	Lightweight Directory Access Protocol
MIPv6	Mobile IPv6
MN	Mobile Node
MOF	Managed Object Format
MT	Mobile Terminal
MTIHO	Mobile Terminal Initiated Handover
nAR	New Access Router
NIHO	Network Initiated Handover
oAR	Old Access Router
OpVO	Operational Virtual Organization
PANA	Protocol for carrying Authentication for Network Access
PDP	Policy Decision Point
PEP	Policy Enforcement Point
PBNM	Policy Based Network Management
PBNMS	Policy Based Network Management System
PDA	Personal Digital Assistant

PGP	Pretty Good Privacy
QoS	Quality of Service
QoSB	Quality of Service Broker
QoSGGw	Quality of Service Grid Gateway
RSVP	Resource-Reservation Protocol
RTP	Real-Time Protocol
RTSP	Real Time Streaming Protocol
SDP	Service Discovery Protocol
SER	SIP Express Router
SIP	Session Initiation Protocol
SLA	Service Level Agreement
SNMP	Simple Network Management Protocol
SOAP	Simple Object Access Protocol
SSL	Secure Socket Layer
SSO	Single Sign On
TCP	Transmission Control Protocol
TLS	Transport Layer Security
UA	User Agent
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
VO	Virtual Organization
WBEM	Web Based Enterprise Management
WEP	Wired Equivalent Privacy
WLAN	Wireless Local Area Network
WP	Work Package

WPA	Wi-Fi Protected Access
WS	Web Service
XML	Extensible Markup Language

1. Introduction

As already described in previous documents, the Network layer provides means to keep the transport of information up and optimized at all times always taking into account the context and the nature of the communication, the parties involved and external factors such as saturation or breakdown.

This document provides a refined architecture of the different components and services provided by the Network and takes special interest in concepts that are based on interactions between Network and Grid such as the integration of SIP and SOAP, or the possibility of having network QoS requests from grid services in chapters 3 to 6.

It is worth mentioning the role of PBNM, described in chapter 6, which has also suffered a notable development with respect of what was related in previous documents.

In addition to this, Security also takes a central role in this document. Chapter 7 makes a survey of possible attacks and their importance and describes the most appropriate solutions according to impact, probability and interest to the project's goals.

2. Architecture

The following picture describes the general architecture inside WP4.1 (inside the blue box) and its relations to components in different WPs, the arrows among them indicating the existent interfaces. The protocol utilized for each interface to communicate is also indicated.

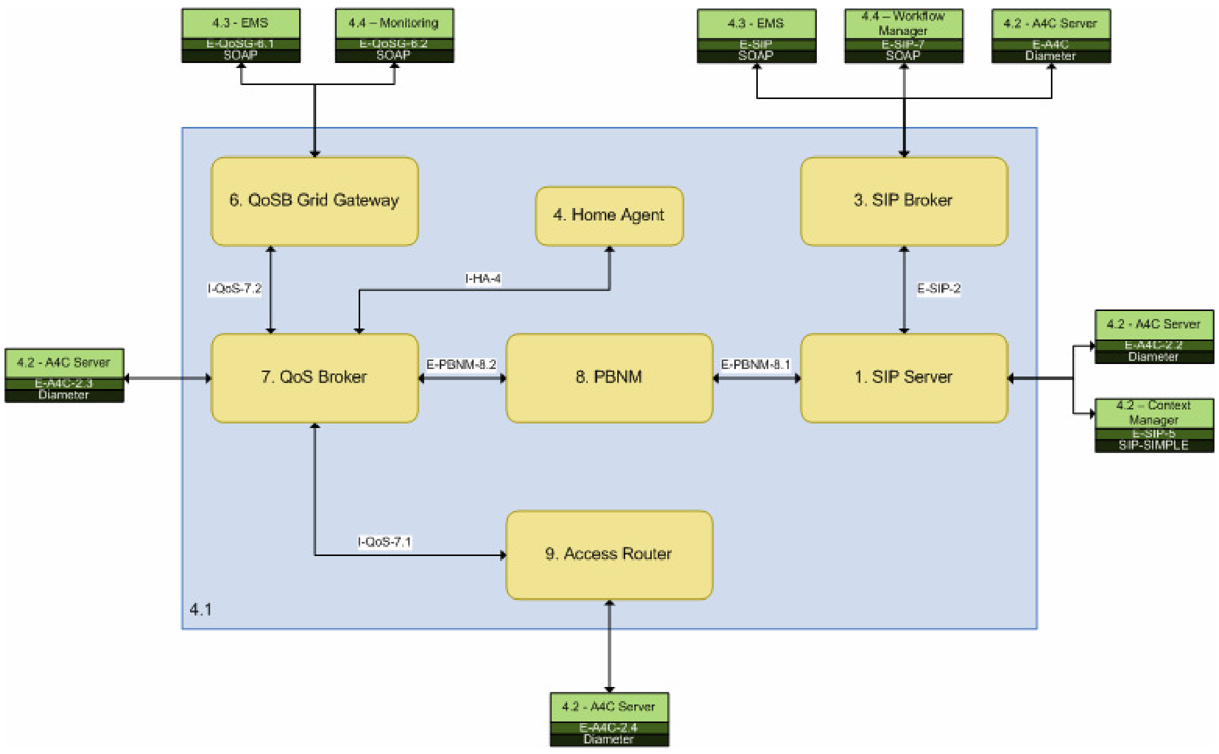


Figure 1 – General architecture

Regarding network quality of service, the main components involved are the Access Router (AR), QoS Broker (QoSB) and PBNM. The QoSB is the equivalent to a Policy Decision Point (PDP), which receives general network policies from the PBNM, and also user profile information from the A4C Server.

The AR is the equivalent to a Policy Enforcement Point (PEP). It controls all network flows that originate or are destined to mobile terminals connected to that AR. When a new flow is detected, the AR sends a message to the QoSB questioning if that flow is allowed or not. The QoSB, based upon the information it gathered from the PBNM and A4C Server, makes a decision and responds to the AR.

The SIP Server provides advanced support for multimedia sessions, session mobility, user presence and context and also support for interactions with web services. Web services may use the SIP Server to enable grid sessions with session management and mobility, session transfer or 3rd party grid session control.

Both the SIP Server and the QoS Broker have web services interfaces for providing some of their functionality to higher layers.

Security issues have also been taken into account in the architecture. They have however not been represented in the description of the architecture at module level since security needs to be handled in much broader sense. Therefore, this issue is treated separately in chapter 7.

3. Mobility

3.1. Introduction

Different types of mobility may be considered; in this chapter we present the three most important in the Akogrimo project.

3.2. User Mobility

As defined in [D411], User Mobility enables the user access to network services or resources independently of the user's terminal. A user-oriented security and authentication framework (described in WP42 deliverables) is responsible for this. For more details see [D422].

3.3. Session Mobility

Session Mobility is the ability, supported by the underlying infrastructure, to transfer an ongoing session from the device the user is currently using (the source Mobile Terminal) to another device (the target Mobile Terminal). Of course, this must be done without the session being interrupted.

Such ability could be used to continue a session in a more suitable device when it is discovered, or when the source device becomes unavailable, e.g. running out of battery. In Akogrimo's eHealth scenario, session mobility is used to transfer an already established media session to a larger display, which is more suitable for media visualisation than the small, low resolution screen of a wearable computer.

Session transfers are usually initiated by the user when needed, e.g. need to switch from a fixed to a mobile terminal (*user triggered session transfer*). However, many of the times the need of a session transfer can be anticipated by analysing user context. Taking this into account, and taking advantage of Akogrimo's infrastructure support for context, session transfers can be initiated on behalf to the user (*Grid triggered session transfer*), suggesting more suitable target MT's in the user surroundings. This functionality is demonstrated in the eHealth scenario, where a doctor is warned about the availability of a better display and the session transfer is initiated by the platform. This operation, of course, requires new entities to be defined in the platform to perform the needed behaviour, as well as to offering these services as suitable interfaces to upper layers.

In Akogrimo, session mobility support will be provided by the SIP infrastructures. The following subsections, the architecture will be presented and the involved components described, as well as their interactions with other subsystems in the platform. Finally, some MSCs are analyzed to clarify the different roles and to give an overview of the signalling involved.

3.3.1. Architecture

As D4.1.1 mentions [D411], there are two main methods to perform the session transfer using SIP:

- Mobile Node Control mode, based on SIP 3PCC (Third-Party Call Control).
- Session Handoff mode, based on SIP REFER.

After analysing the pros and cons of each method, it was decided the second one is more appropriated to the Akogrimo project. For example, it does not involve the terminal which initiated the transfer after it takes place, so a MT could transfer an ongoing session to another device if its battery is running out. For a more detailed explanation, please refer to [D411].

As mentioned in the previous section, session mobility in Akogrimo can take place in two scenarios:

- A user decides to move an ongoing session from one terminal to another (user-triggered session transfer).
- Some workflow decides, based on user context and other parameters from the workflow execution, to inform a user about the convenience of transferring an ongoing session to another device (grid-triggered session transfer). This situation typically can take place if a context-change occurs after a grid-initiated SIP call (e.g. in an eHealth environment, there is a grid-initiated SIP call between a user and a doctor. The doctor is using some device not capable to visualise a medical application, but he/she is moving and now there is a reachable beamer near him/her).

The following drawing shows the user-triggered session transfer, in the way it was implemented for the prototype in the first project phase.

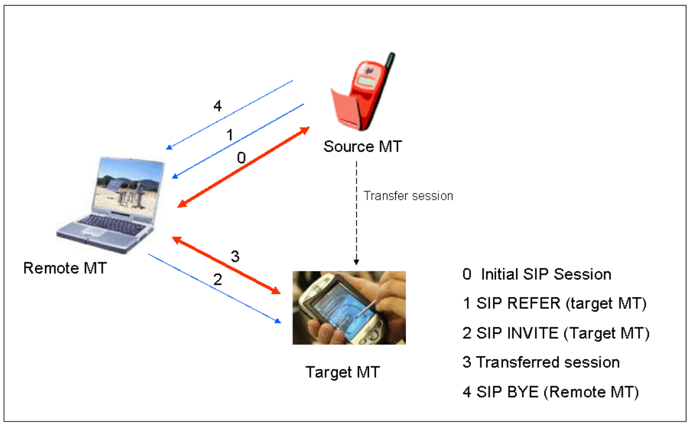


Figure 2 – User-triggered Session Transfer

In this scenario two users (Source MT and Remote MT) have established one SIP session, and one of them (Source MT) wishes to transfer his session to another device (Target MT). So, the Source MT sends a REFER SIP message to the Remote MT, including the Target MT SIP URI in this message. After the Remote MT receives this message, it can start a standard SIP session setup process with the Target MT, and the session will continue now from the Remote MT to the Target MT.

Basically the difference regarding to the scenario described at D4.1.1 (pg 67), is where the REFER message is sent. Sending it to the remote MT being involved in the communication (instead of the target MT, as described in D411) avoids the inclusion of the “Replaces” header field in the SIP request and simplifies the process. Section 3.3.3 describes the message sequence in detail.

The other scenario refers to the grid-initiated session transfer. In this context, there is a SIP session established between two users (Source MT and Remote MT), when the Workflow Manager (based on the user context information and some workflow execution parameters) decides to inform to the SIP architecture (SIP Broker) about the convenience to perform a session transfer, from the device being used (source MT, e.g a PDA) to a more suitable device (target MT, e.g. a beamer).

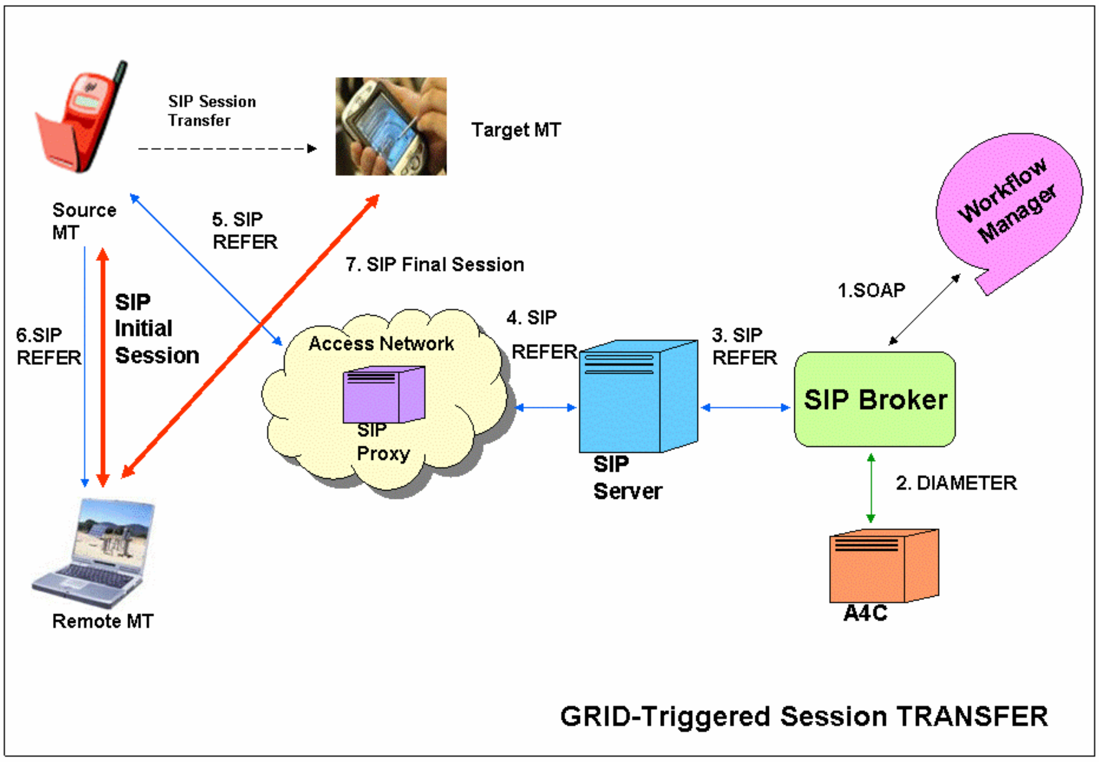


Figure 3 – Grid-triggered Session Transfer

As shows the figure above the components involved in this scenario are the following:

- **Workflow Manager**: this component is in charge of sending the SOAP request (1) to the SIP Broker, in order to indicate it that one session transfer should be started from Source MT to Target MT. The SOAP request (1) includes three parameters: the user identity of the participants of the ongoing session and the target MT's SIP AoR).
- **SIP Broker**: once it receives the SOAP request (1) from the Workflow Manager, it retrieves the SIP URIs of the participants from the A4C (i.e. concerning the source and remote MTs being used), then it builds the REFER SIP message and sends it to the SIP Server.
- **SIP Server**: it receives the REFER message from the SIP Broker and forwards it to the SIP Proxy which is supporting to the Source MT.
- **SIP Proxy**: it is placed in the Access Network, and is in charge of forwarding the REFER message to the Source MT.
- **MT**: it is the end user device. Once the Source MT receives the SIP REFER message , in order to transfer its current session to another device (more suitable), it starts the Session Transfer process described before (Figure 2)

3.3.2. Interaction with other subsystems

The SIP module offers a Java interface, which enables the communication from/to the end user application running on top. Considering the point of view of the application running on top, this interface can be divided in two main parts:

- The public interface that the SIP Module component offers to the application in order to initiate different SIP requests.

- A Java interface that the applications have to implement in order to receive notifications from the SIP UA. These notifications can be a response to a process initiated by the application, an incoming request from another SIP entity, or intermediate notifications on the status of a transaction in progress.

So as part of the middleware offered to the MT SIP-aware applications, the SIP module offers a set of functions to initiate, accept, reject, and so on, a SIP session transfer, even if not initiated by another MT. In fact, the SIP Broker WS interface includes a method to notify a user on the availability of a more suitable device for an ongoing session. This method can be invoked from a grid entity (typically the Workflow Manager) to induce a session transfer.

3.3.3. Signalling

SIP signalling for session transfer in Akogrimo is based on the concepts and scenarios presented in [SHACH03]. The proposed mechanisms have been adapted to Akogrimo platform specific requirements. In the draft, two possibilities to perform session transfer are presented: Mobile Node Control Mode and Session Handoff (SH) Mode. The main difference between them is that the Mobile Node Control Mode allows the Source MT to retrieve a previously transferred session. Session Handoff Mode is not as flexible as Mobile Node Control Mode, but it simplifies signalling and implementation. SH Mode is the approach selected in Akogrimo, because its capabilities satisfy session transfer requirements for Akogrimo scenarios, so there is no real advantage in making signalling and implementation more complex.

In the Session Handoff Mode, the session is completely transferred to the Target MT. The Source MT is no longer related with the session being held. As stated before, this is how sessions are transferred in Akogrimo.

Figure 4 shows the signalling involved in a user-triggered session transfer. In the MSC, the Source MT has established a session with the Remote MT. Then, the user decides to transfer the session to a Target MT for any reason. Session transfer is started with a REFER message, which tells the Remote MT to transfer the session to the Target MT. The REFER message also creates a subscription relationship between the Remote MT and the Source MT. This subscription is used by the Remote MT to inform the Source MT about the progress of the transference by means of NOTIFY messages. The Remote MT then establishes a new session with the Target MT according to a standard SIP session setup. Once the new session is established, the Remote MT informs the Source MT, which sends a BYE message to the Remote MT to tear down the previous session.

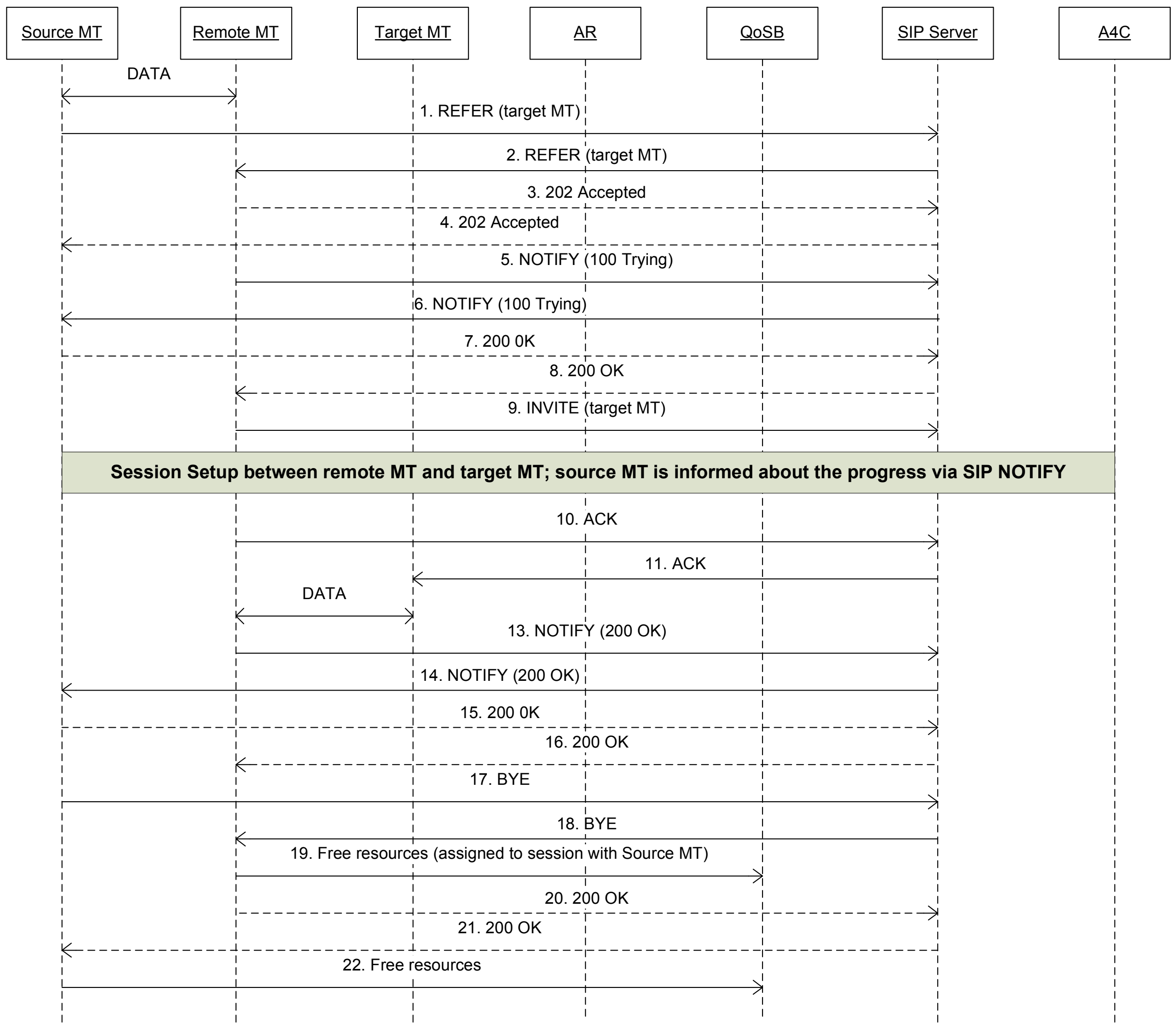


Figure 4 – SIP Session Mobility (session handoff mode). User triggered

Note that the signalling presented in the last figure is a bit different from the one proposed in [SHACH03], where the Source MT sends the REFER message to the Target MT instead of the Remote MT. The Target MT then contacts the Remote MT and establishes a session with it. As the Remote MT does not maintain an ongoing session with the Target MT, some mechanism to distinguish the session transfer from an independent new session is needed. This is achieved via the *Replaces* header, defined in [RFC3891]. This header is included by the Source MT in the first REFER and copied by the Target MT in the INVITE it sends to the Remote MT. Its presence in the INVITE message indicates a session transfer, and the header contains the needed data for the Remote MT to know which session is actually being replaced.

The signalling proposed in Figure 4 for Akogrimo simplifies the messages exchanging and UA behaviour, its drawback being that the Target MT does not know it is engaged in a session transfer, as it receives a standard INVITE from the Remote MT. Whether this is important or not depends on each application. In the Akogrimo's eHealth scenario, session transfer is used for a multimedia videoconference application, and when the session is transferred the application running on the remote MT just continues sending audio and video coming from the microphone and the webcam, but to another device. Target MT does not need to be aware of a session being transferred. This issue should be taken into account for future Akogrimo applications using SIP. If these applications need remote party awareness of session transfer, more complex signalling should be implemented.

Figure 5 shows the signalling needed to perform a Grid-triggered session transfer. In this scenario, the SIP Broker makes use of the REFER message to ask the Source MT to transfer the session. This should not be strange, as the REFER method can be used to make a remote UA send any other method, not only INVITE. In this case, it is use to cause another REFER, and this is why this mechanism is known as *Nested REFER*.

Apart from this, the signalling involved is the same as in the previous scenario. The REFER sent by the SIP Broker also creates a subscription in the Source MT, which will inform the SIP Broker about the progress of the transfer via NOTIFY messages.

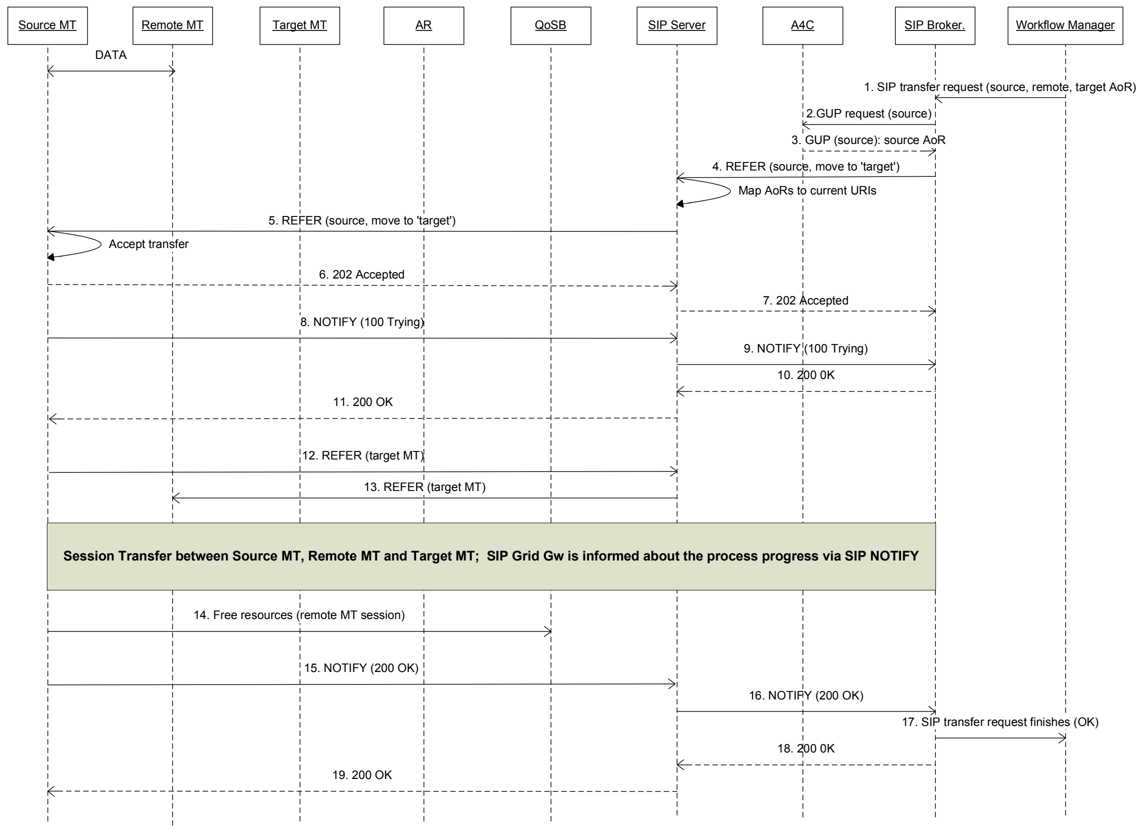

Source MT
Remote MT
Target MT
AR
QoSB
SIP Server
A4C
SIP Broker.
Workflow Manager
DATA
1. SIP transfer request (source, remote, target AoR)
2.GUP request (source)
3. GUP (source): source AoR
4. REFER (source, move to 'target')
Map AoRs to current URIs
5. REFER (source, move to 'target')
Accept transfer
6. 202 Accepted
7. 202 Accepted
8. NOTIFY (100 Trying)
9. NOTIFY (100 Trying)
10. 200 OK
11. 200 OK
12. REFER (target MT)
13. REFER (target MT)
Session Transfer between Source MT, Remote MT and Target MT; SIP Grid Gw is informed about the process progress via SIP NOTIFY
14. Free resources (remote MT session)
15. NOTIFY (200 OK)
16. NOTIFY (200 OK)
17. SIP transfer request finishes (OK)
18. 200 OK
19. 200 OK

Figure 5 – SIP Session Mobility (session handoff mode). Grid triggered

3.4. Terminal Mobility

Terminal mobility refers to the ability of a terminal to remain connected to the network even if it changes access point or access technologies.

SIP alone is not enough to guarantee seamless terminal mobility, for if a non-Mobile IPv6 capable terminal changes access point, it loses connection to its previous network and gains a new IP address in the new network. In practice, that means that all its ongoing communications before the change, are stopped and have to be restarted.

3.4.1. Architecture

Terminal mobility in Akogrimo is achieved with the usage of Mobile IPv6 [MIPv6]. The Mobile IPv6 protocol enables mobility at layer 3 in a transparent manner, i.e. higher layers don't need to know, and in fact are not aware if the terminal they are communicating with is using mobile IPv6 or not.

For implementing Mobile IPv6 in a network, one new entity has to be added to the network:

- Home agent (HA): A node that resides at the MT's home network, which is in charge of forwarding traffic directed to the MT while it is away from its home network so that it looks as though the MT is virtually at his home network. To this end, the HA must be aware of the MT's current binding.

Additionally, mobile terminals have to be modified in order to have support for Mobile IPv6. The Mobile IPv6-enabled terminal may described as

- Mobile terminal (MT): A node with the ability to change its point of attachment to the network keeping connections alive.

Any entity communicating with the MT is called the correspondent node.

- Correspondent node (CN): Any peer node which a MT is communicating with. A CN does not necessarily have to implement mobility i.e., a CN may be either mobile or stationary, with or without mobile IPv6 support.

When the MT is away from his home network the HA will act on behalf of it forwarding all the packets addressed to the MT at its Home Address (HoA) to its current Care of Address (CoA) by means of a tunnel. The CN is not aware that the MT is roaming unless it too supports Mobile IPv6. If this is the case, the CN will realize that its traffic is being tunnelled by the HA and may perform a route optimisation by directly communicating with the MT by means of the MT's CoA instead of using the tunnel provided by the HA. This should be the standard mechanism because it notably increases the performance. In order for the MT to communicate its position to the HA and all the CNs it is communicating with, a procedure is launched in which a packet called Binding Update (BU) is sent with the address currently used and a packet called Binding Acknowledgement (BA) is received by the MT as a confirmation.

3.4.2. Interaction with other subsystems

Mobile IPv6 is transparent for higher layers. However, in some situations the QoS Broker must know whether or not a terminal is outside its home network. For that purpose, Akogrimo's Home Agent includes a module which will communicate with the QoS Broker.

That module's function is solely to translate any HoA into CoA, and vice-versa. The results of that translation will allow the QoS Broker to better perform its decisions.

3.4.3. Signalling

3.4.3.1. Fast Handover

The goal of Mobile IPv6 is to achieve seamless mobility of devices. Due to its movement, the MT may have to change its point of attachment to the network by using a new access router that is closer to its current position. This change of point of attachment to the network is called handover and involves acquiring a new CoA and communicating the HA and CNs its new location before it can be fully operative again and the communication can be re-established. Voice applications, for example, are very sensitive to delays and need the handover to be fast enough in order to for it to be seamless.

Fast Handover [FHO] reduces handover latency by anticipating the handover by means of the link layer. It allows the MT to start sending packets on a new network link as soon as it is detected, and also allows the MT to receive packets as soon as its attachment to the new link is detected. This way, while maintaining the previous connection intact and sending packets through it, the MT also sends packets through the new network link (effectively duplicating the packets). This process, also known as bicasting, avoids the loss of packets during the handover process and consequent degrading of quality in applications that are making use of the network.

3.4.3.1.1. Mobile Terminal Initiated Handover

Figure 6 depicts a scenario in which a Mobile Terminal is moving from the Old Access Router (oAR) to the New Access Router (nAR). Numbered messages show how the FHO is executed. The explanation of the messages is given below.

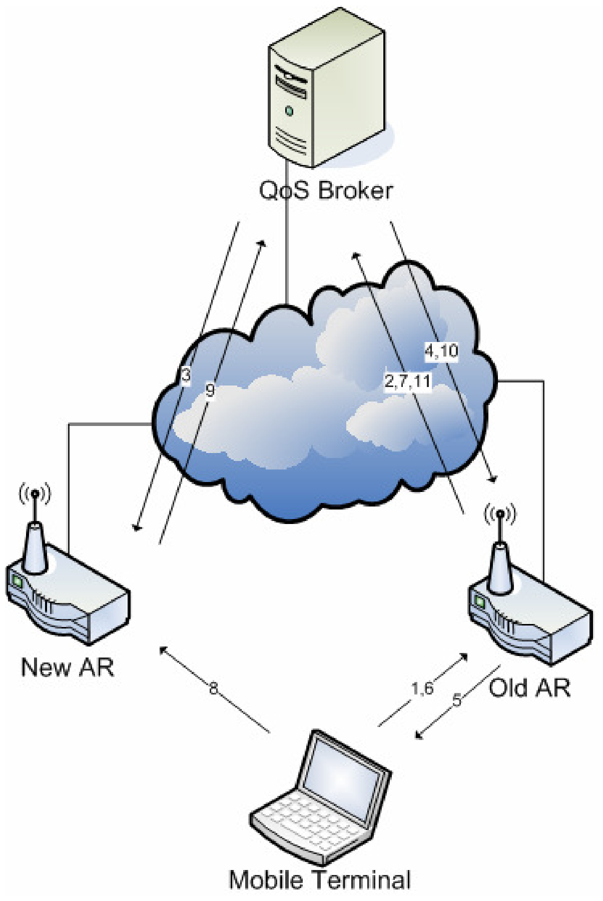


Figure 6 – Fast Handover

The Mobile Terminal detects a new access network and decides that it should move there. The reason may be related to better signal in the new network or user preferences. The MT begins the FHO process by sending a *HandoverRequest* message to the current AR (1). Upon receiving this message the AR sends a request for approval of the handover to the QoS Broker (2). The QoSB, after verifying that the requested access network has available resources to match the user's requirements, sends a *HandoverDecision* message to both the oAR and nAR (3,4).

The oAR informs the MT that it can move to the new access network by sending it a *HandoverResponse* message (5). When the MT receives that message, it performs some internal checks and sends a *FastBindingUpdate* to the oAR (6) which is subsequently reported to the QoSB (7). The MT then configures its layer 2 connection to the new link and sends a *FastNeighbourAdvertisement* message in order to populate the nARs neighbour cache (8).

Having received the FNA, the nAR notifies the QoSB that the handover was successful (9) and the QoSB informs the oAR of the handover's success (10). Finally, the oAR informs the QoSB that information about this MT has been deleted and that MT's control has been handed to the nAR (11).

3.4.3.1.2. Network Initiated Handover

In this scenario, it is the network which decides that the MT must perform a handover to a new access network. The QoSB sends the oAR a *HandoverDecision* unsolicited message specifying that the MT must roam to the new access network. The oAR sends a *HandoverResponse* to the MT, which upon its reception must perform the necessary steps to make the handover to the nAR.

In case the new access network is no longer available to the MT, it can request aborting the procedure by sending a FBU with a negative acknowledge.

The steps of the NIHO are the same as the MTIHO, except the former does not include steps 1 and 2.

3.4.4. Security

The use of Mobile IPv6 should not make the network less secure. In order to do that, MIPv6 has already some security mechanisms. Others mechanisms are also used to further improve security.

Binding Update protection is necessary for preventing an attacker from disrupting or hijacking the communication between two nodes.

The use of IPSec extension headers allows the protection of messages exchanged between the MT and the HA. It also protects the authenticity and integrity of Mobile Prefix Solicitations and Advertisements.

Protection of messages between MT and the correspondent node is achieved through a procedure called Return Routability.

Security is developed further in chapter 7 of this deliverable.

4. Quality of Service

4.1. Introduction

Quality of service can be defined in a very basic sense as the consistent and predictable delivery of data. The Akogrimo network must be able to provide such diverse services as interactive audio (e.g. a SIP audio session) or large data transfers (e.g. some Grid service which requires large portions of data to be transferred among different machines for computation). These diverse services have completely different requirements on a network level. The former requires a relatively small amount of bandwidth (e.g. 64Kbps) but very strict latency requirements, in order to allow a normal conversation without interruptions. The latter requires large amounts of bandwidth (e.g. 512Kbps or more) but is tolerant when it comes to latency, since a delay does not affect the computation of the data. Quality of service is used in this case to accommodate the different network flows according to their requirements and at the same time optimize the use of the network resources. QoS also has an impact on the reduction of network infrastructure costs.

This section will present the functioning of the QoS system and the interactions among the major components involved in QoS.

4.2. QoS Architecture

4.2.1. Overview

Figure 7 shows the 4.1 work package components which are responsible for managing QoS in the Akogrimo network. All other components, including possible QoS clients which can make QoS requests but are not involved in managing it, are dimmed.

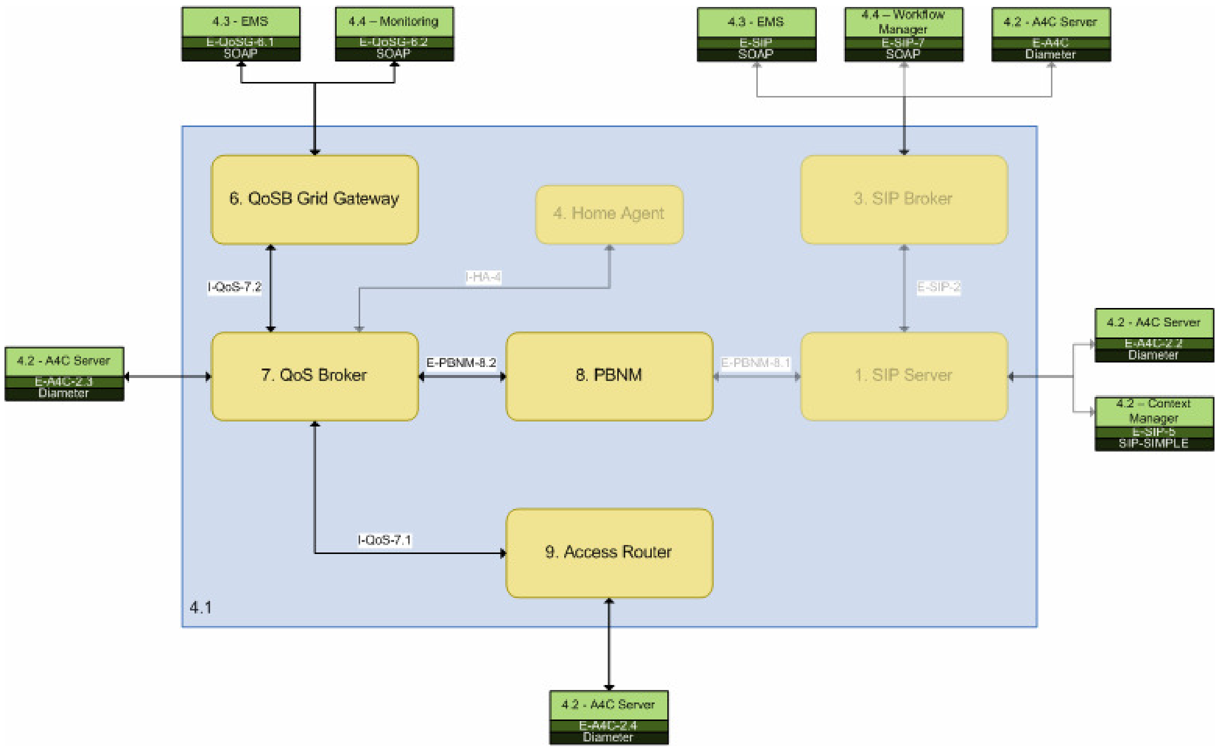


Figure 7 – QoS subsystem components

4.2.2. Components involved in QoS subsystem

The PBNM system is responsible for distributing policies to the QoS Broker. It is discussed in chapter 5.1.

The QoS Broker manages all network resources that belong to the network(s) it is responsible for. It knows at every moment the usage level of the network resources and the users; with that knowledge combined with the policies it receives from the PBNM, it is able to make informed decisions whenever a request from a user arrives. Additionally, the QoS Broker can receive requests from the Execution Management Service (EMS) through the QoS Grid Gateway. The QoS Broker is described in chapter 4.3.1.

The Access Router, besides providing access to users of either wireless or wired technology, will enforce the QoS rules it gets from the QoS Broker. Whenever a user starts a network flow with a determined QoS level, the AR checks if the user is allowed that QoS level or otherwise and correspondingly allows or blocks the network flow. The Access Router is described in chapter 4.3.2.

4.2.3. QoS Bundles

To make an effective use of network resources and to assure that a user can utilize the services he is entitled to without disruptions, the Mobile Network Layer will implement end-to-end Quality of Service, for all types of mobility. The QoS system allows fine-grained QoS reservations at the access networks, but aggregates different flows with the same QoS requirements in the core network.

For implementation and scalability reasons, the network supports well defined QoS bundles, strongly influenced by the existing models for mobile technologies (UMTS). Each of the three defined bundles is designed for a specific usage profile, audio, video and data. A QoS Bundle is comprised of several well defined services, which the user may choose from when using the Akogrimo network. Table 1 presents these bundles.

Bundle 1	Bundle 2	Bundle 3
Mix audio + data	High data + video	Mostly voice
10 – Interactive	20 – Interactive	10 – Interactive
100 – Data	1000 – Data	1 – Priority
1 – Priority	200 – Priority	1 – Signalling
1 – Signalling	1 – Signalling	250 – Best Effort
250 – Best Effort		

Table 1 – QoS bundles and usage. Values are in kilobytes per second

Signalling: This is traffic needed so as to maintain and support the network infrastructure; therefore it is the highest priority. It is time-critical and, in fact, essential to network operations as a whole. Typically its bandwidth requirements are very low.

Interactive real-time: This is time-critical traffic that will be used mainly for video conferencing or audio communications. Interactive multimedia applications are very sensitive in regards to delays. The delays allowing for optimal functioning of interactive applications are less than 100ms. Latency, jitter and out-of-order packets also affect adversely voice communications.

Priority: This type of traffic is not time-critical, but it is important, such as multimedia streaming, or some grid application data exchange. It is higher priority than Data Transfer, but has lower bandwidth available typically.

Data Transfer: Data transfer is somewhere in between Priority and Best Effort (BE). This type of traffic is not time-critical but may be loss-sensitive. Furthermore, out of order packets are typically not a concern for applications that use Data Transfer.

Best Effort: As the name implies this service offers best effort. If the network conditions are good, this should be fine for most applications. If the network is heavily loaded, BE will be the most affected. This is basically what Internet provides.

4.3. QoS subsystems and interactions

4.3.1. QoS Broker

The QoS Broker is the network component which effectively manages all network resources. It receives global network policies from the PBNM system. Requests for QoS from access routers have to be approved by the QoS Broker which then sends appropriate configurations to the ARs. It also exchanges information with other QoS Brokers.

Mobility is also dependent on the QoS Broker, since a user may not change Access Network (AN) if, for example, the new AN has all its resources already occupied. When a user moves from an access network to another then both brokers are involved in the process, assuming that the different access networks are supported by distinct QoS Brokers.

Figure 8 shows the QoS Broker functional architecture:

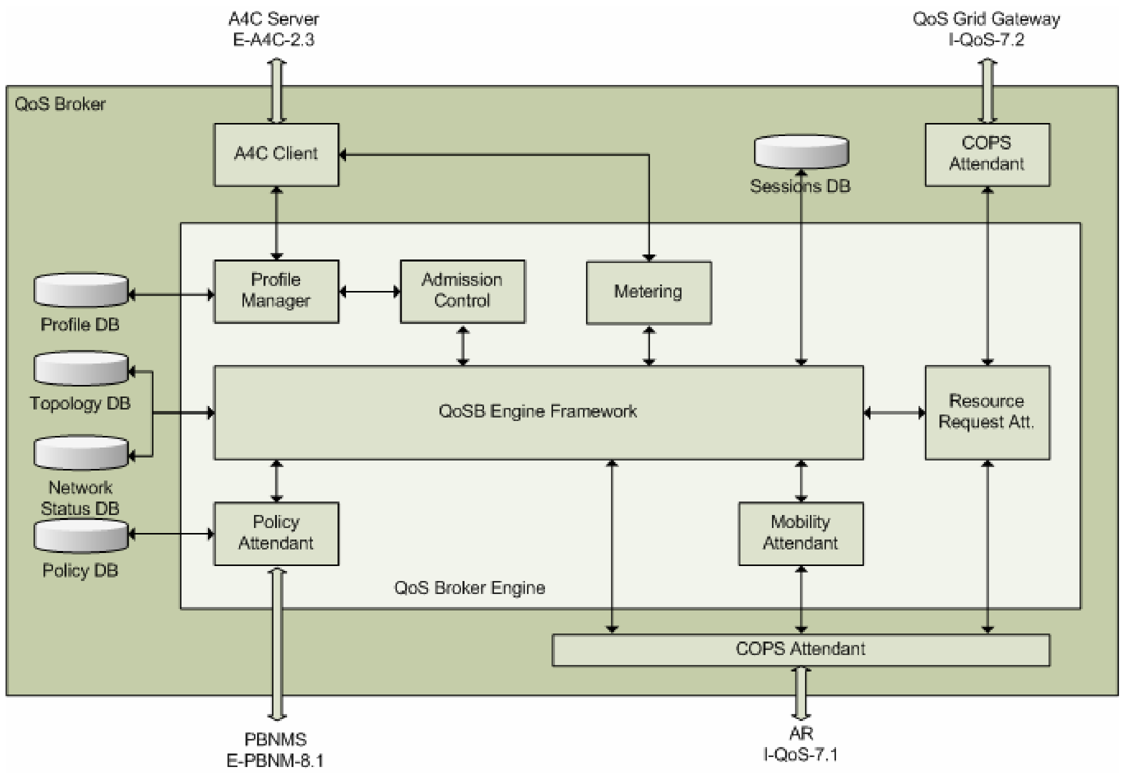


Figure 8 – Akogrimo QoS Broker

The QoS Broker has a Policy Attendant which communicates with the PBNMS and stores the policies it receives in a policy database. It is also possible for the QoS Broker to request policies by itself.

The COPS Attendant module allows it to communicate with the AR using the COPS protocol [COPS]. It is through this interface that the QoS Broker will receive QoS requests and other information from the Access Routers. It will also be used to send appropriate configurations to the ARs.

The A4C Client module allows the QoS Broker to retrieve user profiles from the A4C Server. Those profiles are then cached in the Profile database for efficiency reasons. This eliminates the need to do a request to the A4C Server each time the QoS Broker needs information from the user's profile. The A4C Client module is also used to communicate metering data, received from the Access Routers, to the A4C Server after having been processed by the QoS Broker. For a sample user profile, see Annex C.

A COPS Attendant module is responsible for receiving QoS requests from the QoS Web Service. This module also sends QoS status notifications, for SLA monitoring purposes. These messages are transmitted to the QoS WS which then communicates with the SLA Monitor.

The Topology, Network Status and Sessions databases are used for QoS Broker internal operation:

- Topology DB – holds information about routers and respective interfaces. It allows the QoSB to find an actual route for a data flow.
- Network Status – allows the evaluation of whether there are available network resources for establishing a new session.
- Sessions DB – holds information about ongoing sessions.

4.3.1.1. Differences to previous version

There are two main differences between the present architecture and the previous version which was presented in deliverable D4.1.1.

- **Accounting:** in the previous version accounting was only present in the Access Router. Upon the implementation of the actual software it was realised that it would be beneficial to move the accounting module to the QoS Broker. Having the AR do all the accounting work would imply added communications between the AR and the QoS Broker for retrieving user information which is necessary to correlate the network flows and the users responsible for them. It would also mean that it would be necessary to add logic to do that operation. The present architecture is simpler, albeit introducing an extra step in the accounting communications.
- **QoS Grid Gateway:** while in the previous version a web service interface was presented as a solution for communicating with the EMS, that interface has evolved into a separate component. This component has a web service based interface to the EMS. That interface is in fact not limited to the EMS – it can also be used by any other service that wishes to do so. For communicating with the QoS Broker it uses a COPS interfaces, much like the Access Router.

4.3.2. Access Router

4.3.2.1. Overview

The Access Router acts as an interface between wired/wireless terminals and the core network. A single AR may support different access technologies. The rest of the network will be able to communicate with different mobile terminals in the same way, regardless of the technology they are using for network access.

The Access Router also provides network management components (such as QoS Broker, for example) with a uniform way of managing a heterogeneous network environment such as the one which will form the Akogrimo network.

The AR resides on an Access Network at the edge of the Akogrimo network, and thus, any attempt to unlawfully use network resources without proper permission will be halted at the AR.

A user is required to authenticate himself to the provider before he can use any resource of a provider's network. User authentication requests are intercepted by the AR and then forwarded to the A4C Server.

A user is immediately authorised once he has successfully authenticated. That authorisation allows him to use basic network resources such as moving between different ANs.

Figure 9 shows a functional view of the AR.

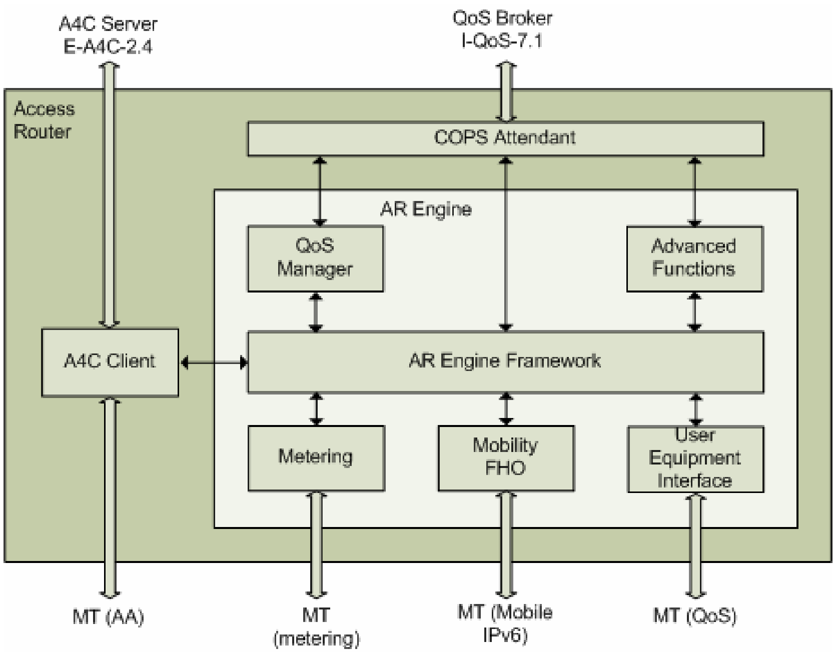


Figure 9 – Akogrimo Access Router

The Access Router is comprised of the AR engine which provides most of its functionalities. It also incorporates an A4C Client module which communicates, using the Diameter protocol, with the A4C Server for authenticating users. The AR engine is, in its turn, formed by the following modules:

- **COPS Attendant:** The COPS Attendant is the main interface of the AR. It is through this interface that the AR communicates with the QoS Broker. QoS requests from the mobile terminal will be translated into COPS messages and then delivered to the QoS Broker. QoS requests may come in the form of explicit RSVP messages or DSCP packet marking; whatever the case, and assuming that the AR is properly configured to handle those options, the AR will process them and deliver the appropriate COPS message to the QoS Broker.
- **QoS Manager:** The QoS Manager module is responsible for managing per-user flows established in the AN, as well as the per-hop behaviour of the AR.
- **Metering:** The metering module is able to do per-flow metering. Its results are communicated to the QoS Broker, which then does some processing for relating flows with users and finally delivers the results to the A4C Server.
- **Mobility+FHO:** The Mobility and FHO module will deal with users' mobility requests.
- **Advanced Functions:** Advanced Functions is a group of modules that implement certain advanced mechanisms which may be used by the AR:
    - QoS Trans-signalling: The translation of external QoS signalling into internal DiffServ signalling, appropriate for the CN, is done by the QoS Manager module. This translation is then sent to the QoS Broker, where it is validated and communicated to the rest of the

network. Having the AR do the translations and the QoSB the validations and communications allows the distribution of processing effort among ARs and QoSB's.

- Advanced Signalling Processing: This module is capable of processing signalling messages from layers above layer 3, such as SIP. It can inspect packets and retrieve QoS related information, such as session bandwidth from SDP packets. With this information, the AR may setup a temporary resource allocation which would then be used in case the session setup is successful, thus diminishing the time necessary compared with normal setup.
- Legacy QoS: This module provides support to legacy applications that are unaware of the Akogrimo network QoS framework.

- **User Equipment Interface:** The User Equipment Interface is in fact a group of modules that deal with communications to and from the mobile terminal. These communications can be done through a signalling mechanism, be it explicit or implicit. Alternatively, a mechanism such as connection tracking can be used. The modules that compose the UEI are:
  - DSCP: The DSCP module is capable of recognizing and controlling network flows marked with a specific DSCP code. It is also capable of re-marking the packets, should the need arise.
  - RSVP: This module extracts information from RSVP requests and passes that information to other modules.
  - SIP: If required, SIP requests are intercepted by the ARs Advanced Functions module and they are processed, else they are simply forwarded to the SIP Proxy. Processing of SIP messages is done to allow the extraction of relevant QoS information or even change the messages themselves.
  - Connection Tracking: This module is capable of tracking TCP sessions. Other transport protocols may also be tracked, allowing customised services on a per-flow/session basis.

4.3.2.2. Differences to previous version

There is one significant difference between the present architecture and the previous version presented in deliverable D4.1.1.

- **Accounting:** as stated in the previous chapter, the AR is no longer responsible for sending accounting information to the A4C Server. Its responsibilities in that matter are limited to gathering metering data and providing it to the QoS Broker, which will handle them appropriately.

4.3.3. Mobile Terminal

The mobile terminal runs a Linux kernel patched with Mobile IPv6 and Netfilter. The MIPv6 patch obviously enables Mobile IP functionality. The Netfilter patch is necessary for implicit signalling to work. It permits

- the marking of packets with a specific DSCP
- the matching of packets with a specific DSCP

4.3.4. Policies/PBNM interactions

The network components must be configured according to the network operator's requirements. Since configuring large networks can be tedious and error-prone, Akogrimo will use Policy Based Network Management system, which will allow network administrators to use a simple interface for configuring global network policies that will be propagated to the rest of the network.

The PBNM system is responsible for providing network policies and configurations to the QoS Brokers, which will then configure the Access Routers. For more information on PBNM, see chapter 5.1.

4.3.5. User profiles/A4C interactions

The user profile is stored in the A4C Server. When a user connects to the network, the QoS Broker will request the user's profile to the A4C Server, and will cache it during the user's session in order to avoid repeating requests for the user profile.

The user profile holds information about the services which the user has at his/her disposal. With that information, the QoSB is able to perform the decisions on whether or not the user is allowed access to a specific service.

A sample skeleton of a user profile is available in Annex C.

4.4. Signalling

This section will present the signalling mechanisms which QoSB, AR and MT use to communicate with one another.

4.4.1. QoS Broker <-> Access Router

Signalling between the QoS Broker and the Access Router is done with the COPS protocol which provides all necessary functionality as well as the possibility of being extended to accommodate new functionalities.

The Common Open Policy Service protocol (COPS) is a client/server policy exchange protocol. The clients (Policy Enforcement Point or PEP) send requests to the server (Policy Decision Point or PDP). The PDP then makes a decision and sends it to the PEP. The COPS protocol is extensible; it can recognize new self-identifying objects and supports client specific information without changes to the protocol. It uses TCP for transport, provides message level security for authentication, replay protection and message integrity. It can also use IPSec [IPSEC] or TLS [TLS] for authorisation and securing communications between the PDP and PEP.

4.4.2. Access Router <-> Mobile Terminal

The signalling between the Mobile Terminal and the Access Router may be handled in different ways. The functionality provided by the multiple different ways is, however, equivalent. The next sections present the different signalling schemes which can be adopted.

4.4.2.1. Implicit signalling

4.4.2.1.1. Overview

Implicit signalling is done by having the Mobile Terminal mark its packets according to the QoS class required, or by having the AR automatically mark the packets of a flow for which a given policy exists.

Implicit signalling is closely tied to the notion of well-known services. Akogrimo well known services that require a specific QoS may be used without requiring any signalling from the part of the MT.

Implicit signalling is also used for support of legacy applications, which are unaware of the Akogrimo QoS-enabled network.

In any case, two distinct possibilities exist for implicit signalling usage:

- The MT marks the packets that applications send. In this case the MT is pre-configured for handling application specific packets and marking them accordingly.
- The AR recognizes a particular flow for which appropriate policies exist which require that flow to be treated with a particular QoS level. In this case the AR re-marks the packets from the default value to the desired value, thus enabling QoS on that flow.

4.4.2.1.2. Packet Marking

Packet marking means setting the Traffic Class field of an IPv6 packet to a specific value. The Traffic Class field is composed by eight bits; the six most significant bits are used for the value of the DSCP (Differentiated Services Code Point), while the remaining two bits are used for ECN (Explicit Congestion Notification). In Akogrimo the ECN bits are also used as part of the DSCP.

The Access Router analyses each and every packet that is sent by the Mobile Terminal. Knowing the DSCP of a packet, the AR then matches it with its internal rules, which have been previously setup by the QoS Broker. If a packet does not match to a rule, it is dropped. Else the packet is sent to its destination through the core network and using the appropriate QoS.

4.4.2.1.3. Message sequence

The following picture shows a message sequence depicting two mobile terminals communicating with each other using network QoS.

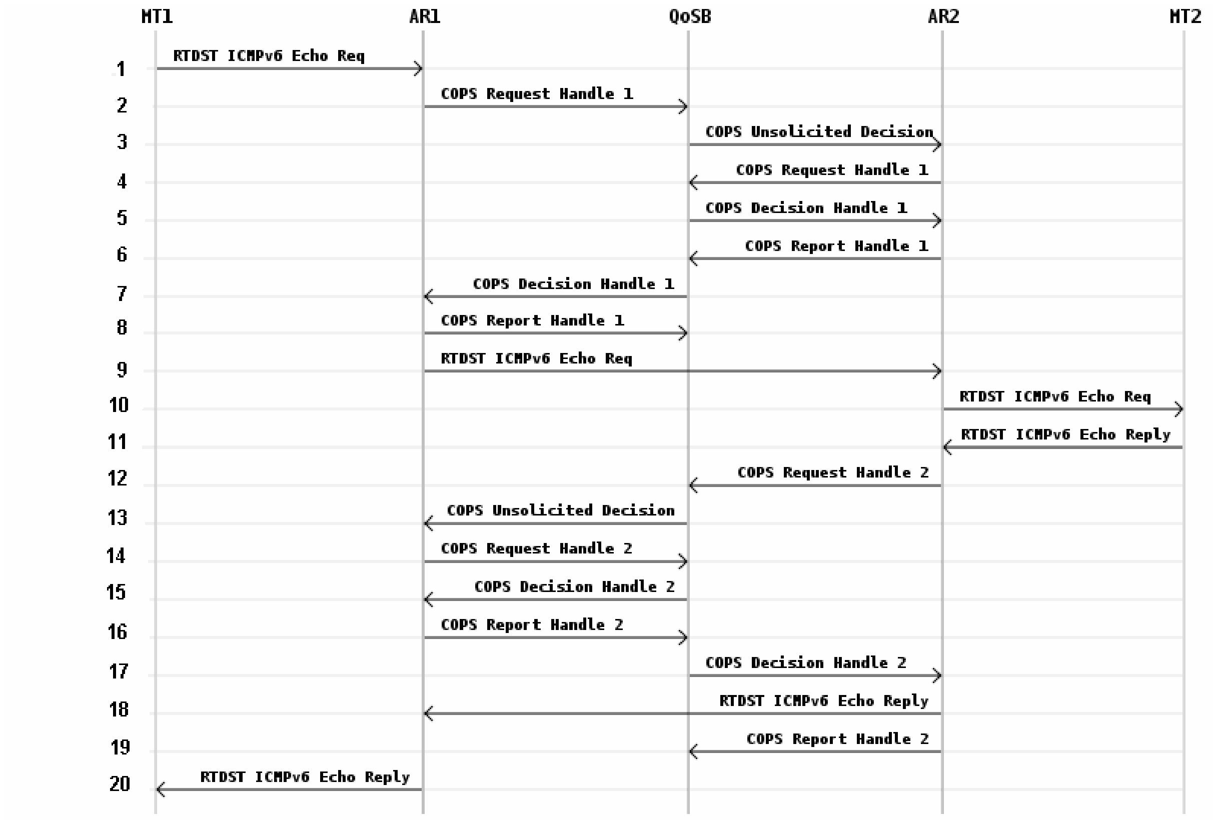


Figure 10 – Implicit QoS request

MT1 initiates the communication with a ping packet which was marked in the mobile terminal (1). AR1 analyses the packet and recognises by its DSCP that it requires a determined QoS level. Since it does not know if the user is allowed that QoS or not, it contacts the QoS Broker (2). The QoS Broker finds out that MT1 is indeed allowed to use that QoS for that flow, as is MT2.

Before responding to the AR1, however, it will inform AR2 that MT1 and MT2 will be having a QoS session and that it must reserve resources for that (3). AR2 then receives the configuration and sends a report stating that all went OK (4, 5, 6). Only then does AR1 receive its configuration for the flow (7, 8). The ping packet then traverses the core network and is delivered by AR2 to MT2 (9, 10). The ping reply will trigger the equivalent message sequence, but in the inverse direction.

It is worth pointing out that all this signalling occurs only for the first packet of a given flow. Other packets will be processed according to the configuration that was received before and require no signalling overhead.

4.4.2.2. RSVP signalling

It is possible to use RSVP [RSVP] for QoS requests thanks to the QoS trans-signalling capabilities of the AR. Figure 11 shows an RSVP request message sequence.

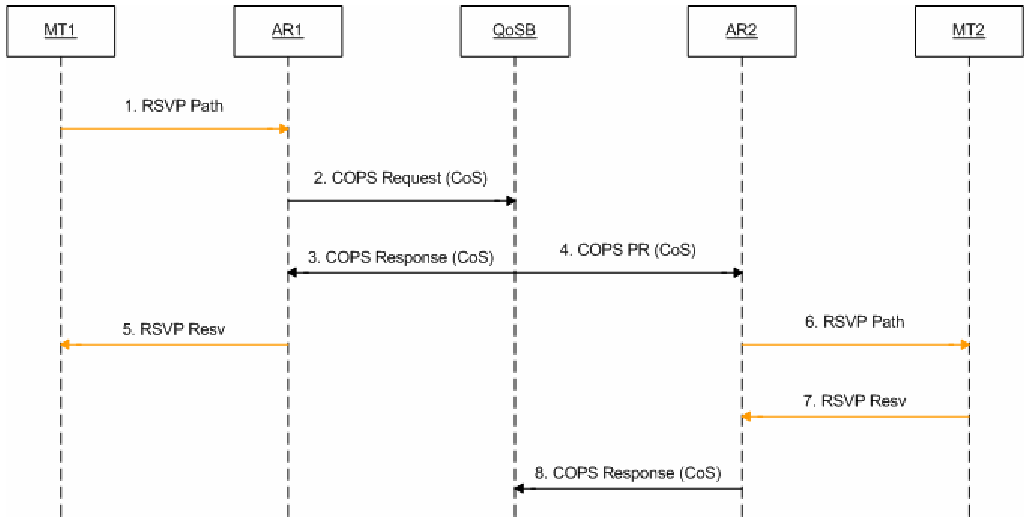


Figure 11 – explicit QoS request using RSVP

In this scenario MT1 issues an RSVP Path request that is detected by AR1 (1). AR1 extracts relevant information and requests the QoS Broker an appropriate configuration and mapping of that RSVP reservation into DiffServ Class of Service (CoS) (2). The QoS Broker then responds to AR1 (3) with appropriate configurations and filters. AR1 then sends a RSVP Resv message to MT1. At the same time, the QoS Broker sends a COPS provision message to AR2 (4) with information that will be used to build a new RSVP Path message and also for configuring AR2. The RSVP message built by AR2 is then sent to MT2 (6) which will respond accordingly (7) with a RSVP Resv message. Finally AR2 responds to the QoSB provisioning message (8).

4.5. QoS and EMS interoperation

The interoperation between QoS Broker (QoSB) and Execution Management Services (EMS) is provided by the QoSB Grid Gateway. This component exposes some of the QoSB's functionality as Web Services, enabling the EMS to interact with the QoSB.

4.5.1. Architecture

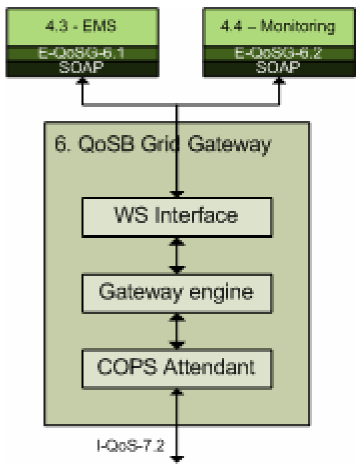


Figure 12 – QoSB Grid Gateway

Figure 12 presents a functional view of the QoSB Grid Gateway. It is composed of three software modules.

- The WS Interface is responsible for communication with other Web Services, namely the EMS and the Monitoring service. It is a web service developed with the Globus Toolkit.
- The Gateway Engine is where most of the logic resides and is developed with the C++ programming language. It communicates with the WS Interface via standard internet sockets, using a simple purpose-specific protocol that allows the Gateway Engine to receive QoS requests and to send the responses to the WS Interface.
- The COPS Attendant is responsible for communicating with the QoS Broker. It is a C++ program that makes reuse of some of the software classes already developed for the AR and the QoS Broker. Communication between COPS Attendant and the Gateway engine is done through normal function calls.

4.5.2. EMS Interface

The QoSB Grid Gateway exposes the methods described below. Note that the interface is not limited to the EMS - its use in the methods description is just for simplicity.

Method	Arguments	Purpose
check_qos_availability	user qos_bundle qos_service	This method is used by the EMS to check if the network is able to accommodate the new flow required by the EMS. Note that no reservation is performed by this method.
set_qos	user SLAid qos_bundle qos_service	This method is used by the EMS to perform the actual QoS reservation.

Table 2 – QoS-EMS interface methods

The *set_qos* method allows the EMS to select not only the *qos_service* (real-time, data, etc.) but also to request the QoS Broker to permit the user access to a better QoS Bundle. This particular function is not yet implemented as some issues such as who is to be charged for that are still not clear.

4.5.3. Monitoring Interface

The QoSB Grid Gateway also has an interface to the Monitoring Service. The QoSB Grid Gateway will be the information producer and the monitoring service is the information consumer.

An appropriate notification will be sent if an SLA is not being complied with by the network infrastructure. Since the QoSB will always check the status of the networks, this situation only arises when the network is at or near full load and a high priority network flow, such as a request from a doctor for patient data, arrives at the QoSB.

4.6. Security

Security at the network level involves a wide array of problems, most of which are not Akogrimo specific. As such our focus has been in matters more relevant to Akogrimo.

One of the major concerns is user's safety and data confidentiality. For this reason, IPSec will be used between MT and AR to provide data integrity and confidentiality. It is not possible to use end-to-end tunnel mode, since that would prevent the AR from inspecting packets sent by the MT.

Network entities may use IPSec for communication among themselves. This will prevent an intruder who gains access to the core network from posing as a given machine.

More details can be found in chapter 7.

5. The Akogrimo SIP framework

5.1. Introduction

The Session Initiation Protocol (SIP) is an application-layer protocol that provides mechanisms to establish, modify and terminate sessions. In conjunction with other IETF protocols it can provide advanced support for multimedia sessions, although it is not restricted to multimedia. These protocols are SDP, Session Description Protocol for describing multimedia sessions; RTP, Real-time Transport Protocol for transporting real time traffic and RTSP, Real Time Streaming Protocol for controlling delivery of streaming data. It can provide also user and session mobility. This section is intended to describe how SIP infrastructures cooperate with the rest of the network infrastructures to provide such kind of services.
The Akogrimo SIP-related components are the Mobile Terminal (MT) SIP Module, the SIP Server and the SIP Broker (the old SIP Grid Gateway). A detailed description of them can be found in D4.1.1 [D411]; however, since the architecture of each one has been evolving, an analysis of this evolution will be performed, both at architecture and implementation levels.

5.2. SIP Architecture

5.2.1. Mobile Terminal: SIP Module

The internal architecture of the MT SIP Module (which was described in D411 [D411]) has been slightly modified to avoid dependencies with the type of session being handled. Figure 13 shows the old version of the component (focussed only on the session control part; subcomponents related to the SIP presence handling has not been depicted). The modified elements and interfaces have been highlighted.

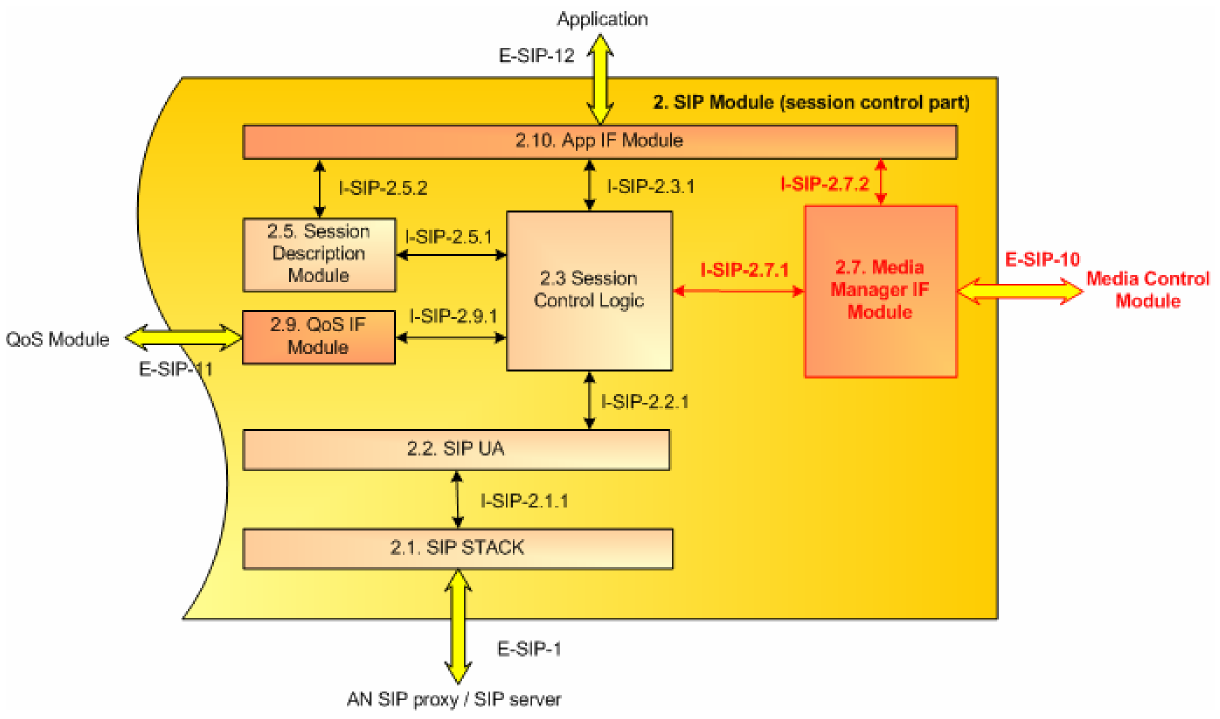


Figure 13 – Old MT SIP module architecture

Session data handling is an attribute very close to the meaning of "session" within the context of each type of application. For example, for a classic multimedia application a "session" means the exchange of RTP traffic, so "session data handling" will consist on the control of the ports being used to exchange this kind of information. But for other applications (like the gSDP application described in section 5.3.1) it can have an absolutely different meaning.

In the old architecture, there was a specific component that was responsible for interfacing the Media Control module (which was responsible for the RTP traffic handling). Considering that media control is something particular for multimedia applications, this subcomponent introduced a strong dependency with some specific type of application (in this case, a multimedia application), so one of the potentials of the SIP protocol (its independency from the type of session or application) was lost.

The proposed architecture is depicted in Figure 14. In the new version, the Media Control IF module has been removed; additionally, a clear differentiation between session control (i.e. signalling) and session data handling (i.e. data exchange) has been introduced. The control plane comprises all subcomponents related only to signalling, which are independent from the type of application being handled. The core part of this subset of components is the Session Control Logic. This component interacts with the data plane core part trough a dedicated internal interface (I-SIP-2.7.1) which is used, for example, to inform that a problem with the data exchange has occurred, or to stop this data transmission if the user ordered it.

In order to consider the application-dependant nature of the session handling, in the new architecture each application will be responsible for the definition of its specific session data handlers, which have to be registered in the MT SIP Module. All registered session handlers are managed by a single component named Session Data Manager, which substitutes the old Media Manager IF module and offers a generic interface to the signalling core part.

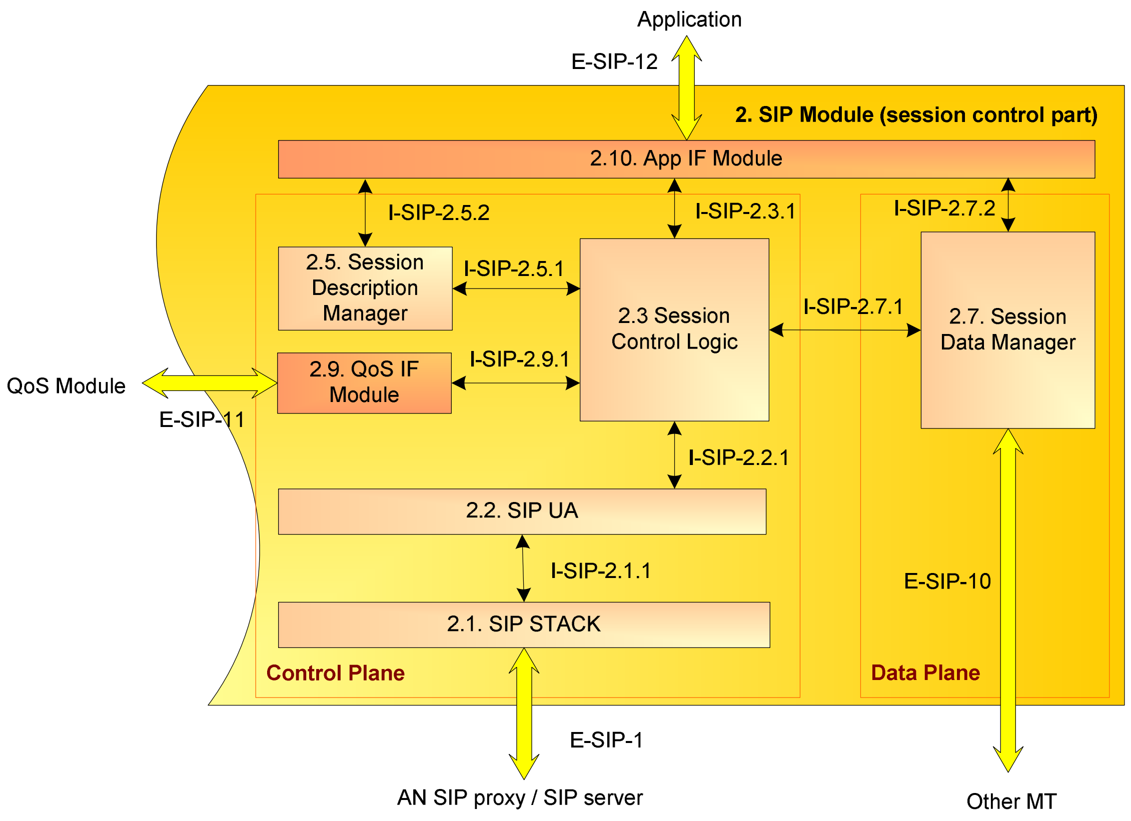


Figure 14 – Revised MT SIP module architecture

This change in the architecture introduces also a modification on the nature of the E-SIP-10 external interface, while the old E-SIP-13 (RTP interface between the Media Manager of the MTs) becomes unnecessary. This is described in the table below.

External interface	Old architecture	New architecture
E-SIP-10	MT SIP module – Media Manager (Java interface between MT components)	MT SIP module – MT SIP module (session-specific interface between different MTs– e.g. RTP for multimedia sessions)
E-SIP-13	MT Media Manager – MT Media Manager (RTP for multimedia sessions)	Not necessary

Table 3 – Changes in the external interfaces of the MT SIP module

5.2.2. Network components

5.2.2.1. *SIP Server*

No significant changes from the architecture described in D4.1.1 [D411].

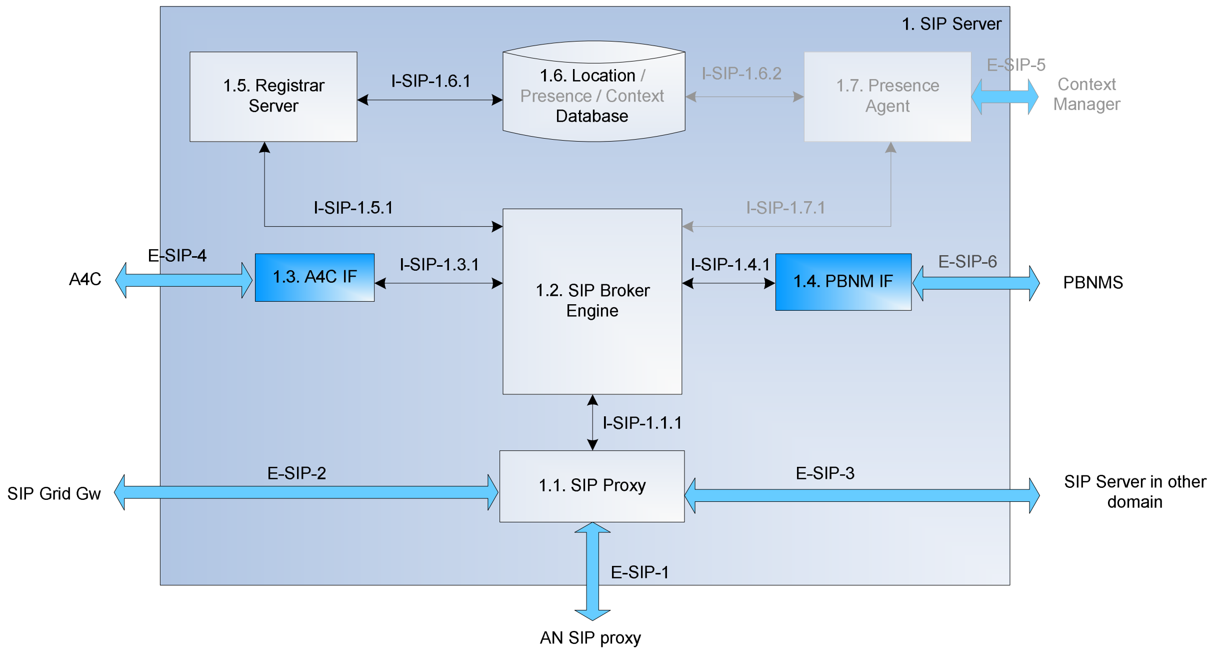


Figure 15 – SIP Server architecture

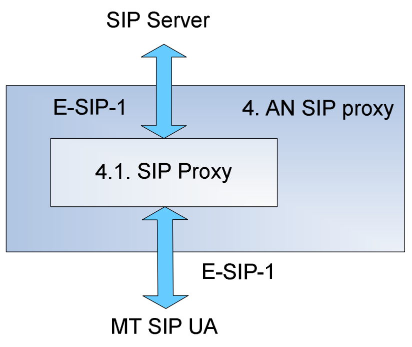


Figure 16 – AN SIP proxy architecture

5.2.2.2. SIP Broker

The SIP Broker can be understood as an application server (following the IMS terminology) which offers SIP-based services (e.g. grid initiated SIP calls and session transfers, gSDP session establishment and termination…) to the upper layers, exposing them as web services.

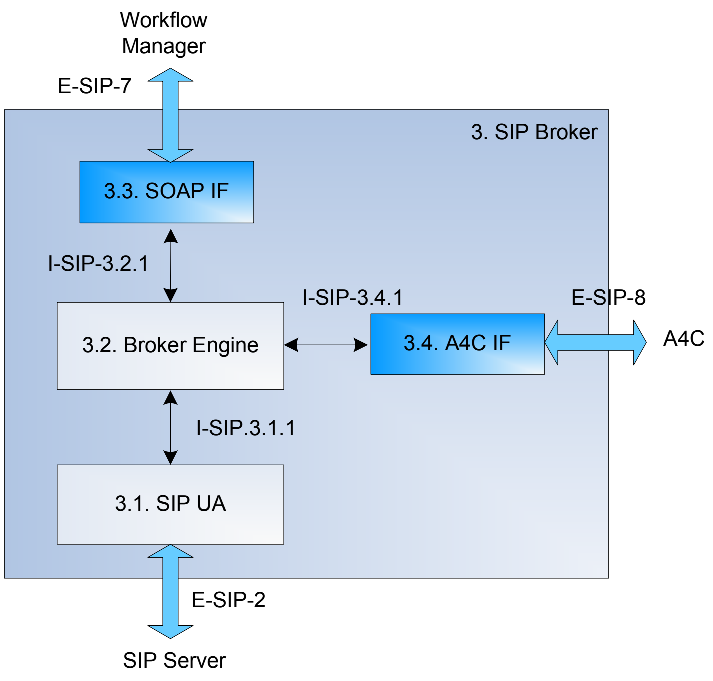


Figure 17 – SIP Broker architecture

5.3. SIP-SOAP synergies

As it has been defined in the updated architecture documents, it is necessary to address the integration of the GRID environment in a ubiquitous and mobile context where the different actors (users and services) can move to a different terminal (with a different network address), or appear/disappear suddenly.

The agreed approach is to integrate SOAP-based Web Services with SIP signalling protocols in a framework for addressing this mobility and ubiquity feature in non-multimedia sessions like Grid sessions, problem also known as the ***"SIP with SOAP interaction"*** where the main protocols (SIP for ubiquity and SOAP for Web Services) have to interact.

In the proposed approach, GRID applications will be able to:

- Establish a GRID session with a mobile/ubiquitous service without a prior knowledge of its current location.
- Use the SIP mechanisms to manage sessions, like transfer a session, or control a Grid session by a 3rd party.
- Using the presence and context management mechanisms established by the SIP SIMPLE infrastructure.

From a service provisioning point of view, it is needed to provide an interface to support the functionality needed by Grid applications, which includes the following:

- At start up phase, mobile/ubiquitous services will be associated to a certain SIP URI, which has been registered in the SIP registrar with the SIP REGISTER method. This SIP URI can be simply the URI of the user that started the MT where the services are running, or an URI representing the services. Within the context of the SIP-SOAP synergies, from now on we will refer to it as "service URI".

- The applications, instead of using EPRs (End Point References, as stated in WS-Addressing) for identifying resources, should use in the establishment phase SIP URIs for identifying mobile/ubiquitous services, so they will send a SIP INVITE to the service URI.
- In the SIP answer, by means of a specific "Grid Session Description Protocol" (from now on, gSDP), the Grid application will get the needed Grid session information to start the Grid invocations (i.e. the current service EPR).

Additionally, Grid applications could include the adequate logic needed to:

- Use further session management features provided by SIP like session save/load/restore, useful also for session transfer. There are already some proposals for this feature (Web Services Continuity), like the approach presented in [WS CONT].
- React to context changes which are notified via SIP.

This service provisioning can be done in two ways:

- Provide a SIP API to the Grid applications. Grid applications will handle SIP interactions by themselves, managing registrations and invitations. This is the cleanest approach in terms of architecture, but implies that Grid applications have to be modified in order to use the interfaces offered by the SIP UA or the Session Control Logic for SIP protocol handling.
- Provide some intermediate modules, which will be invoked by the Grid applications and manage the needed SIP interactions on behalf of the Grid applications, even establishing a "Grid session" on behalf of them. These modules will offer their capabilities through standard Web Service interfaces. This approach will ease the implementation of the Grid applications, since they will need to use just SOAP messages even for SIP-related methods. Thanks to these modules, SIP information such as the current location as well as notifications about the status of mobile services running at registered MTs on the SIP Server, are propagated to the Grid layer and vice-versa. In this way, the Grid layer can keep track of the mobile users and readjust itself to changes on the fly.

Finally the second approach (the usage of intermediate modules) was adopted in order to minimise the changes of the grid applications; however, the first option has not been discarded for a next stage of the project.

Next section describes how the chosen solution (named "Services SIPification" works within the Akogrimo context.

5.3.1. Services SIPification framework

As mentioned before, the adopted solution involves some intermediaries in which the grid applications delegates in order to handle all the underlying SIP interactions. Depending on the role and the nature of the involved Grid services, two types of intermediaries are envisaged:

- For the mobile/ubiquitous grid services running at MTs, considering that these MTs already have a SIP module, the simplest solution is to have a specialised application handling the gSDP sessions. This MT gSDP application should be active in the MTs where the service to be invoked is running and it will offer some API to the services in order to modify the services information. In order to interact with the MT SIP module, it must follow the general rules for building Akogrimo SIP applications described in section 5.4.1.
- For grid services not running at the MTs (tipically the EMS service), the current SIP Broker module WS interface will be extended, adding a WSRF-compliant WS developed on top of GT4 that acts as a gateway service in front of the SIP infrastructure. Using these new

facilities, a grid service will be able to retrieve the current EPR of another service running on a MT, be aware of the service availability, etc.

The way in which all these elements interact is described in the following MSCs.

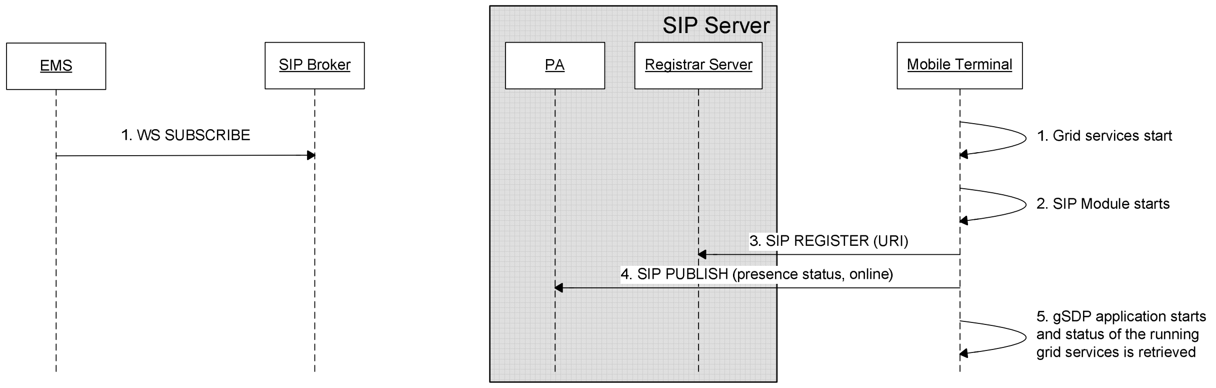


Figure 18 – Intermediaries and interfaces setup

During the MT startup, the grid services, the SIP module and the gSDP application which will act in behalf of the grid services are started following the depicted sequence in the right side of the picture. The SIP module startup involves a SIP registration and a SIP presence publication, in order to make visible the MT for other grid services that could be interested in the grid applications running on it. After the gSDP application starts, the MT will be ready to provide information about the grid services running on it. The service URI to be used will be the SIP URI being used during the SIP registration.

The interface of the intermediate module running on the SIP Broker has also to be initialised, as shown in the diagram (left side). After the initialisation, the consumer service (e.g. the EMS in the sequence) will be ready to receive WSRF notifications.

Next sequence shows how the EPR of a mobile/ubiquitous service can be retrieved using this framework, in order to invoke it.

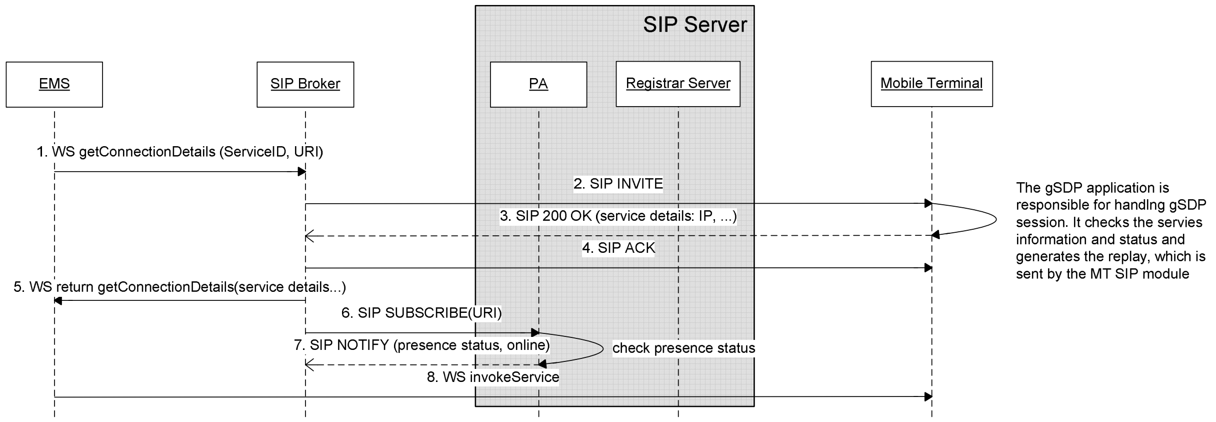

SIP Server
EMS
SIP Broker
PA
Registrar Server
Mobile Terminal
1. WS getConnectionDetails (ServiceID, URI)
2. SIP INVITE
3. SIP 200 OK (service details: IP, ...)
4. SIP ACK
5. WS return getConnectionDetails(service details...)
6. SIP SUBSCRIBE(URI)
7. SIP NOTIFY (presence status, online)
check presence status
8. WS invokeService
The gSDP application is responsible for handlng gSDP session. It checks the servies information and status and generates the replay, which is sent by the MT SIP module

Figure 19 – SIPification session establishment and service invocation

When the EMS needs to know the EPR of a service running on a MT, it invokes the operation `getConnectionDetails` offered by the SIP Broker, indicating which service it is interested in and the SIP URI being used by the MT in which the service is expected to be running (1). The EMS can also request details about all the services running in the specified terminal. The SIP Broker starts a SIP gSDP session setup with the required MT (2, 3, 4), whose status basically provides the information the EMS needs to invoke the services. Finally, the method `getConnectionDetails` returns with the required information (5). Annex D.1 describes the agreed format used by the SIP gSDP session, which is also used as return value to the EMS. At this point, the EMS knows the EPR to be used in order to invoke the service or services it is interested in (8).

Additionally, the SIP Broker performs a subscription to the SIP presence information of the service SIP URI (6, 7); in this way, it will be aware on the availability status of the whole terminal and it will be able, for example, to inform the EMS if the user disconnects from one location and then connects from a different one. Using this feature, the EMS can take an appropriate decision, like getting the services EPRs at the new services location. This is the behaviour described in the figure below.

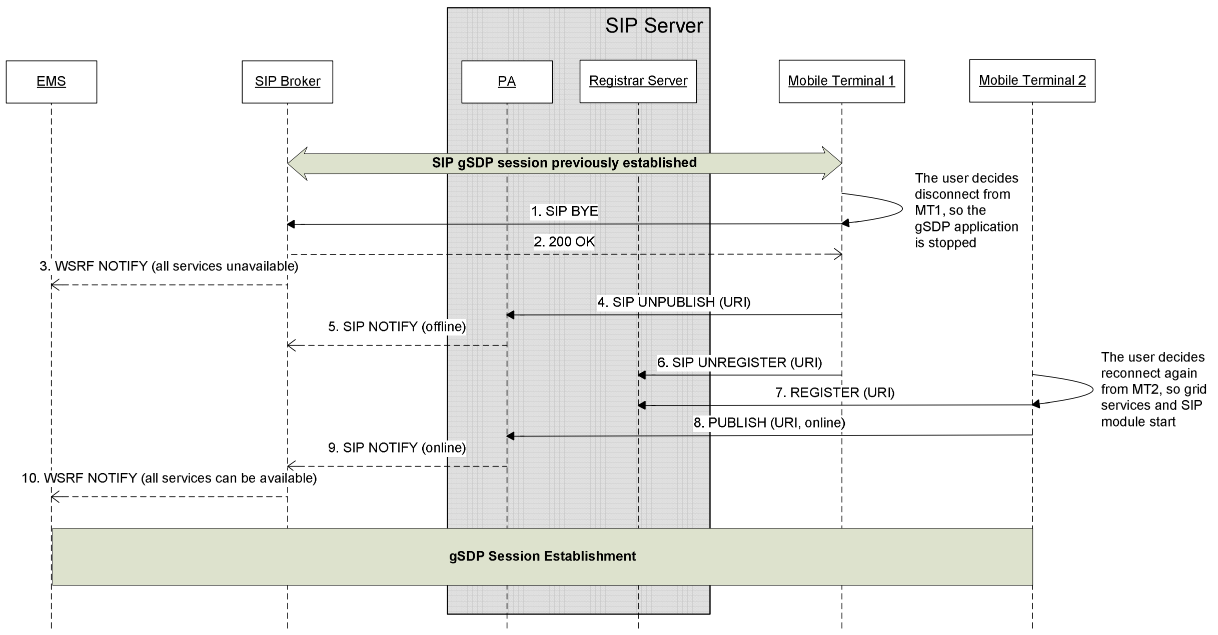

SIP Server
EMS
SIP Broker
PA
Registrar Server
Mobile Terminal 1
Mobile Terminal 2
SIP gSDP session previously established
The user decides disconnect from MT1, so the gSDP application is stopped
1. SIP BYE
2. 200 OK
3. WSRF NOTIFY (all services unavailable)
4. SIP UNPUBLISH (URI)
5. SIP NOTIFY (offline)
6. SIP UNREGISTER (URI)
The user decides reconnect again from MT2, so grid services and SIP module start
7. REGISTER (URI)
8. PUBLISH (URI, online)
9. SIP NOTIFY (online)
10. WSRF NOTIFY (all services can be available)
gSDP Session Establishment

Figure 20 – User disconnection and reconnection from another location

Steps from 1 to 7 represent a controlled disconnection from certain location. From the point of view of the services SIPification, most relevant steps are:

- The SIP Broker informs the EMS about the gSDP session termination using a WSRF notification (3), indicating that all services running at the MT become unavailable. The format of the content of this notification is described in Annex D.2.
- The Presence Agent informs about changes on the presence status while the presence subscription done during the gSDP session setup phase remains active (5).

Then the service URI is used to connect again from another location (steps from 7 on). Thanks to the active presence subscription, the SIP Broker detects that the presence status of the associated service URI becomes online again (9) and it can inform on this to the EMS (10), which typically will request again for the service details at the new location.

Even if an accidental disconnection takes place, the EMS will be aware on this situation, since the SIP Broker will receive a SIP presence notification indicating that the presence status publication of the service URI has expired. This event will be interpreted as an incidental disconnection and the EMS will be informed about the complete services unavailability at the corresponding URI.

Formally, the presence information is not needed, since it is used only as an indirect mechanism to detect a further reconnection (i.e. a SIP re-registration). The SIP Broker would be aware on the registration status of the service URI using the method proposed in RFC3680 [RFC3680], also based on the SIP event framework but for the registration event itself. Additionally, incidental disconnections could be prevented by monitoring the status of the established gSDP session, as described in RFC4028 [RFC4028]. Both mechanisms are being evaluated.

The SIP session established will be used as the main mechanism to inform the EMS about changes on the MT grid services availability or EPRs. So, if the information related to some service changes, the MT gSDP application will send a reINVITE containing the new gSDP session properties. The EMS will be informed on this through a WSRF notification, whose content indicates a change on the services using the format described in Annex D.2. This sequence is depicted in Figure 21.

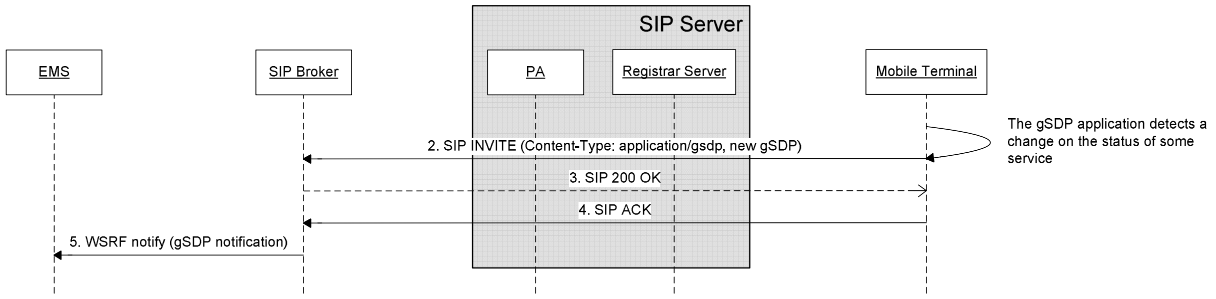

SIP Server
EMS
SIP Broker
PA
Registrar Server
Mobile Terminal
The gSDP application detects a change on the status of some service
2. SIP INVITE (Content-Type: application/gsdp, new gSDP)
3. SIP 200 OK
4. SIP ACK
5. WSRF notify (gSDP notification)

Figure 21 – Change on services information

When the information related to the services running on certain MT is no longer needed, the EMS can terminate an existing SIP gSDP session. In this case, the SIP presence subscription is also terminated. See next figure for more details.

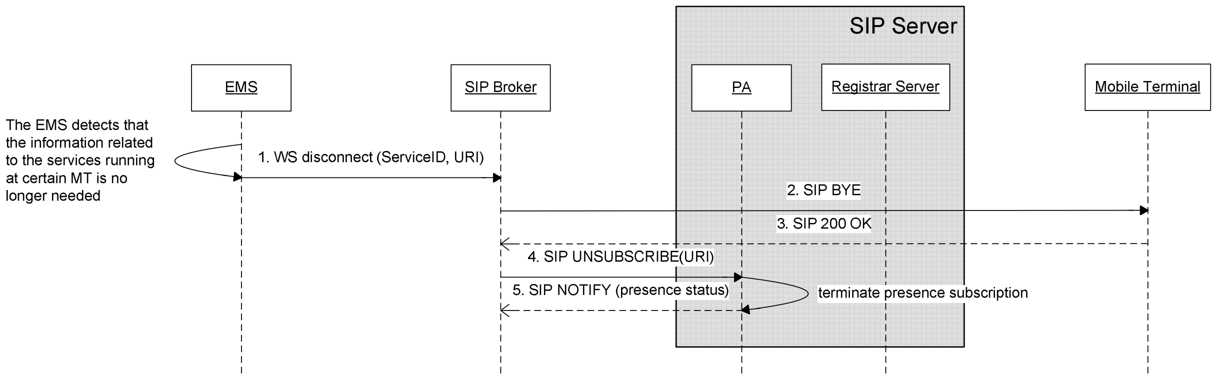

SIP Server
EMS
SIP Broker
PA
Registrar Server
Mobile Terminal
The EMS detects that the information related to the services running at certain MT is no longer needed
1. WS disconnect (ServiceID, URI)
2. SIP BYE
3. SIP 200 OK
4. SIP UNSUBSCRIBE(URI)
5. SIP NOTIFY (presence status)
terminate presence subscription

Figure 22 – gSDP session termination (from EMS)

5.4. Implementation

Main technologies being used for the SIP architecture development are Web Services, Java and C. In the MT all components have been developed in Java, while in the SIP Server C was used (SIP Server was developed using and extending SER).

The SIP Broker functionalities are exposed by means of the SOAP IF subcomponent (see Figure 17) like a Web Service, so, the Workflow Manager or the EMS simply has to invoke (by means of SOAP requests) some method exposed by the Web Service and which are described in the Web Service WSDL file. Additionally, it includes a client application that acts over a property of the WSRF producer, which induces a WSRF notification to the consumer services subscribed to that property. This is the way in which the Broker Engine can send WSRF notifications to the EMS when required.

Other SIP Broker subcomponents (Broker Engine and SIP UA) were implemented with Java.

5.4.1. MT SIP Module

The MT SIP module has been developed in Java, and it has been designed to support multiple and simultaneous SIP applications and sessions running over it. Applications interact with the

SIP Module process using Java RMI technology, but the way in which the SIP Module has been designed hides these details to the application developers.

Each application may be responsible for the handling of certain types of session, so they have to define both the content type of the sessions they are responsible for, and the number of permissible simultaneous sessions they can support: for example, the multimedia application manages “application/sdp” sessions while the gSDP application is responsible for the handling of “application/gsdp” sessions. In both cases, due to the nature of the sessions themselves, only one session is allowed.

When the SIP Module starts, it checks if the login application was started in order to make use of the current login information, which includes the SIP Address of Record, or main SIP URI (SIP AoR) to be used during the SIP registration. If yes, the SIP module is successfully registered; if not, a GUI is shown to request for the login data.

Once the SIP Module has been successfully started, SIP applications can be initiated. The final version of the SIP module could be configured to automatically start some applications after a successful registration. In the other hand, since SIP applications cannot work if the SIP module is not running, all applications running on top will be closed when stopped.

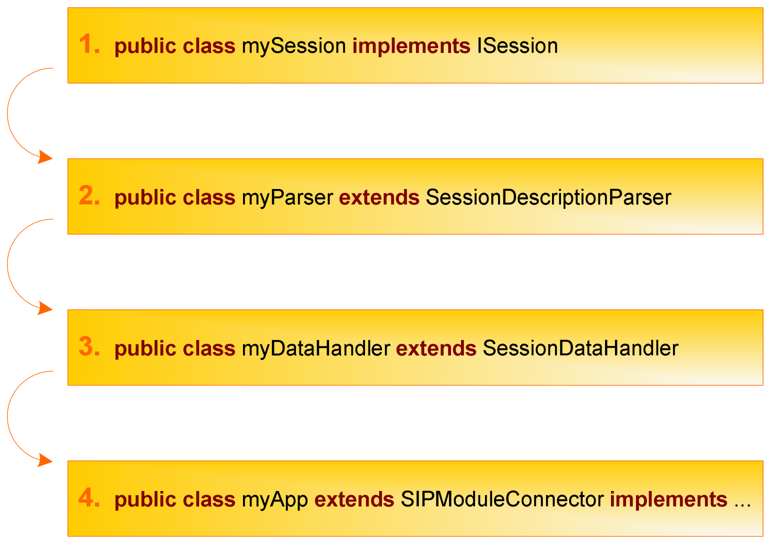
The corresponding debian package install several .jar files that can be used to build an Akogrimo-compliant SIP application following certain rules, as will be described below.

The mapping between the functional components and their implementation is described in the table below.

Component	**.jar file**	**Package**
2.1 Sip Stack	JainSipApi1.1.jar nist-sip-1.2.jar	These libraries contain the JAIN SIP specification and RI, and a JAIN-compliant SIP Stack developed by NIST. They are used by the SIP UA.
2.2 SIP User Agent	SIPUA2.0.jar	org.akogrimo.sip.ua
2.3 Session Control Logic	MTSIPSessionControlLogic.jar	org.akogrimo.sip.sipmodule.sessioncontrollogic
2.5 Session Description Manager	MTSIPSessionControlLogic.jar	org.akogrimo.sip.sipmodule.sessiondescription
2.7 Session Data Manager	MTSIPSessionControlLogic.jar	org.akogrimo.sip.sipmodule.sessiondata
2.9 QoS IF module	MTSIPSessionControlLogic.jar	org.akogrimo.sip.sipmodule. qos
2.10 App IF module	MTSIPSessionControlLogic.jar	org.akogrimo.sip.sipmodule

Table 4 – MT SIP module implementation: libraries and packages

These libraries offer a middleware that can be used by the programmers to build a SIP-aware Akogrimo-compliant application, under certain preconditions (see Figure 23):



```
1. public class mySession implements ISession
2. public class myParser extends SessionDescriptionParser
3. public class myDataHandler extends SessionDataHandler
4. public class myApp extends SIPModuleConnector implements ...
```

Figure 23 – Akogrimo MT SIP application's preconditions

1. Each application must define its own session object that implements the generic session interface, named ISession. This empty interface will be used in all interactions with the SIP Module; the application will be responsible to perform an appropriate casting from the generic type to the specific one, and vice versa.
2. Each application must define how to parse its specific session description data from and to SIP message bodies. This is done by extending from the generic parser base class, named SessionDescriptionParser.
3. As mentioned before, each application must define its own session data handler, by extending from the SessionDataHandler base class.
4. The MT SIP application must extend from the class SIPModuleConnector. This class just provides an implementation of the key utilities that all Akogrimo SIP applications need in order to make use of the SIP Module. But a session-oriented application may want receive notifications from the session control logic (e.g. a new incoming session has just arrived), or event notifications from the SIP Presence User Agent. In that case, it would additionally have to implement one of the interfaces that define all possible incoming events from the SIP UA, so the application developers will be able to decide what to do when the event comes – if the application is not interested in the event, the method will be simply empty. For session handling, the MT API offers an utility class (named SIPSessionApp) that already extends from SIPModuleConnector and implements the required interface (for session handling, ISessionControlLogicListener), so developers can extend directly from it.

Next table summarises these classes and the SIP Session Control Logic library package which contains them.

Class/Interface	**Package (located at)**
ISession	org.akogrimo.sip.sipmodule.sessiondescription
SessionDescriptionParser	org.akogrimo.sip.sipmodule.sessiondescription
SessionDataHandler	org.akogrimo.sip.sipmodule.sessiondata
SIPModuleConnector	org.akogrimo.sip.sipmodule

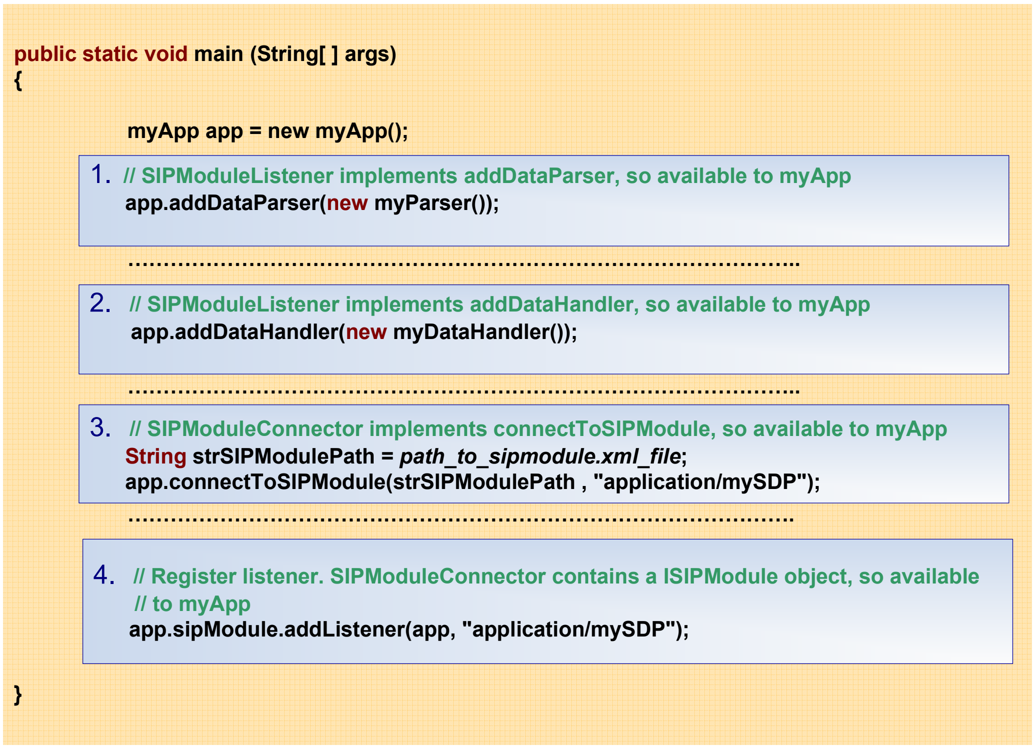
Class/Interface	Package (located at)
ISessionControlLogicListener	org.akogrimo.sip.module.event [1]
SIPSessionApp [2]	org.akogrimo.sip.sipmodule

Table 5 – MT SIP module implementation: location of the main classes

After that, the main function of the application must do the following:

1. Create and initialise the Session Description Parser subcomponent, registering its own session parser.
2. Create and initialise the Session Data Manager object, registering its own session data handler.
3. Connect to the SIP Module.
4. Register the corresponding listeners.

The structure of the main function is depicted in Figure 24. Note that the base class SIPModuleConnector already provides the implementation of the required methods, so they are available to the MT SIP application subclasses.



```
public static void main (String[ ] args)
{
        myApp app = new myApp();
    1.  // SIPModuleListener implements addDataParser, so available to myApp
        app.addDataParser(new myParser());
        ......................................................................
    2.  // SIPModuleListener implements addDataHandler, so available to myApp
        app.addDataHandler(new myDataHandler());
        ......................................................................
    3.  // SIPModuleConnector implements connectToSIPModule, so available to myApp
        String strSIPModulePath = path_to_sipmodule.xml_file;
        app.connectToSIPModule(strSIPModulePath , "application/mySDP");
        ......................................................................
    4.  // Register listener. SIPModuleConnector contains a ISIPModule object, so available
        // to myApp
        app.sipModule.addListener(app, "application/mySDP");
}
```

Figure 24 – Akogrimo MT SIP application. Main function structure

Just to mention that the connectToSIPModule method is responsible for creating the RMI connection between the application process and the SIP Module, but these details remain hidden

[1] All available listeners are defined in this package

[2] This class extends from SIPModuleConnector and implements ISessionControlLogicListener, so it can be used as the base class for all session-oriented SIP applications.

to the application developers. This makes Akogrimo SIP applications clearer and simpler, by avoiding the developers to know the concrete details of the RMI connections.

5.4.1.1. MT gSDP application

As mentioned earlier, a specialised SIP session-oriented application has been developed to act in behalf of the grid services running at the MTs. The first version is just a proof-of-concept prototype oriented to demonstrate the feasibility of the Akogrimo SIP-SOAP synergies approach, so not all the required functionality has been included; however, missing features will be included in further versions. Next table shows a comparison between the main characteristics of the first prototype and the envisaged refined version:

Current prototype	**Final version**
Started by the SSO login application after the SIP Module	Started by the SIP Module itself when successfully logged in
Interface with the grid services simulated; the services information is read from a preloaded xml file Console application to simulate services input	Real interface with the grid services, which will be able to use the gSDP application. Additional GUI for admin. and testing purposes.

Table 6 – gSDP application 1st prototype vs. final version

The mapping between the installed functional components and their implementation is described in the table below.

Component	**.jar file**	**Package**
gSDP application	gSDPApp.jar	org.akogrimo.sip.gsdpapp

Table 7 – MT gSDP application implementation: libraries and packages

The corresponding debian package installs a shell script (gsdpApp) to handle the gSDP application from the console and an xml file (services.xml) with the simulated services information. The format of this xml file is identical to the gSDP session specification defined in Annex D.1.

5.4.2. SIP Server

The whole SIP Server has been build using SIP Express Router (SER) implementation as the basis [SER]. SER is a powerful, easily configurable, C-written and free SIP server implementation. SER can be easily configured to acts as a proxy, registrar or redirect server. Most relevant characteristics of SER are:

- Speed: SER is able to process thousands of calls per second even on low-cost platforms.
- Extensibility: using SER it is possible to link new C code to redefine, or extend, current logic. New code, which can be developed independently on SER core, is linked to it in run time. In this way, new functionality is included by adding SER modules (*.so libraries) keeping the SER core really fast and compact. This has been the adopted solution to add the interface modules with the A4C and the PBNMS.
- Flexibility: routing decisions and SIP method processing can be modified by means of textual scripts that administrators may write. By modifying the main script (named ser.cfg) it is possible to control and modify lot of parameters (e.g. expiration time of registration requests)

and introduce new logic (loading new modules, and invoking exported functions when necessary).

- Portability: written in ANSI C, and deeply tested on PC/Linux, Sun/Solaris platforms.
- Interoperability: SER is based in open SIP standards. SER core implements basically RFC 3261.
- Small size: core footprint is 300KB, add-on modules take up to 630KB.

As mentioned before, SER has been built to be configured using a configuration file, named ser.cfg. It is a C-Shell script which defines how incoming SIP requests are processed. During the server startup, this file is translated into an internal binary representation, because the direct interpretation of this file would make SER very slow. Considering the functional description provided in D4.1.1 [D411], the main configuration file and the main part of the SER core implements both the SIP Proxy and the SIP Broker Engine modules. The appropriate manipulation of this file enables the delivery of incoming SIP requests to the specific module that implements a more concrete functionality, like the Registrar or the A4C interface. Annex B includes the SER configuration file (ser.cfg) used for the demo.

Extensions to implement the required interfaces are described in next subsections.

5.4.2.1. Akogrimo A4C IF SER module

The Akogrimo A4C IF sub-module has been implemented as a SER extension. The resulting dynamic C library is named a4c.so.

The basic functionality of this module is to check and validate the A4C Token included in a SIP request, acting as a filter for the corresponding SIP transaction. In order to validate the token, the C library makes use of the A4C C++ client library. If the validation finishes with success, such transaction is allowed. D4.1.1 [D411] provides detailed information on how this mechanism works.

Next table summarises the new functionality that this library exports, which have to be invoked from the ser.cfg configuration file in order to validate the corresponding transaction. Annex B shows a usage example for the validation of a SIP REGISTER.

Table 8 – a4c.so exported functions

Function	Description	Parameters
aa_request	This function checks and validates the A4C token included in a SIP transaction	None when invoked from ser.cfg file [3]

The library offers some configurable parameters. Most significant are:

- The detail level of the logs (for debugging purposes a traces mechanism – independent from the SER traces framework – has been included. This allows the library debugging without "interferences" from the rest of the SER code).
- The name of the non-standard SIP header field in which the A4C token will be included.

[3] The SER framework passes implicitly the incoming SIP message structure to the function.

- The executable command for validation (a wrapper for the A4C C++ client library to be invoked from the a4c.so code).

5.4.2.2. Akogrimo PBNMS IF SER module

As the PBNM section describes (section 5.1), policies related to SIP proxies or the SIP Server may also exist, but since the envisaged policies involves others systems besides the SIP components (i.e. the QoSBroker and the A4C system) they have been primarily defined for these systems, so the PBNMS IF sub-module has not been implemented yet; however, it would be also implemented as a SER extension if required and documented in a further update of this document.

5.4.3. SIP Broker

The SIP Broker is a standard Web Service deployed on an application server which exposes SIP-based functionality. To be precise, in the prototype version two functionalities were offered:

- Grid-initiated SIP Call, which enables the possibility to arrange a grid call between two SIP users.
- Grid-initiated SIP transfer, which enables the possibility to inform a user he/she can move an ongoing session to a most suitable device.

Other functionalities (e. g. a WS interface to expose the SIP SD facilities) are being considered to be included in next versions. A detailed description of the offered interface is described in section 3.3.3.

The internal structure of the SIP Broker comprises four differentiated sub modules, following the functional description provided in Figure 17:

- The SOAP interface, which accepts incoming WS requests from the "grid world": to be precise, from the Workflow Manager.
- The A4C IF, an embedded overlay of the A4C Java client used to request the A4C Server for the generic user profile (GUP) of the users to be contacted. The GUP contains (apart from other information) the SIP AoR of the user, which is crucial for all SIP interactions.
- The Broker Engine. This Java component is the core of the SIP Broker. Its main task is to convert the WS request parameters (typically the Akogrimo IDs of the user to be put in contact, or the device URI the user is allowed to move to) into a suitable format that the SIP UA can understand. This is made through the facilities provided by the A4C IF submodule. For efficiency purposes, and considering that only one SIP UA and one A4C client are needed, it has been implemented as a separate Java process that can be accessed through Java RMI.
- The SIP UA, which handles the SIP process itself. It has been implemented as a Java library based on JAIN SIP that other components can use.

The mapping between the functional components and their implementation is described in the table below.

Component	**.jar file**	**Package**
3.1 SIP UA	JainSipApi1.1.jar nist-sip-1.2.jar	These libraries contain the JAIN SIP specification and RI, and a JAIN-compliant SIP Stack developed by NIST. They are used by the SIP UA.
	SipUA2.0.jar	org.akogrimo.sip.ua

Component	.jar file	Package
3.2 Broker Engine	SIPBrokerEngine.jar	org.akogrimo.sip.broker.brokerengine
3.3 SOAP IF	sgg.war (deployed on Tomcat5.0.X)	akogrimo.sipBroker
3.4 A4C IF	liba4cclient.jar	This library (UniZh) contains several utility packages that are used by the Broker Engine.

Table 9 – SIP Broker implementation: libraries and packages

Most of the parameters of the SIP Broker can be configured through an xml file (named sipbroker.xml) which is installed together with the mentioned libraries. Parameters that can be configured using this file are:

- SIP Broker Engine RMI port and policy file.
- SIP UA related parameters (IP address, port, transport…).
- SIP proxy server parameters (IP address and port).
- A4C related parameters (A4C client library location, A4C Server IP address…).

The SOAP IF component receives the request of arranging a SIP Call or transferring a session from one device to other. The SOAP-IF then translates the SOAP request to a SIP request. The invocation of the corresponding SIP Broker Engine is made via RMI technology. The SOAP IF implements a client whose URL contact is stored on a properties file that belongs to the SOAP IF component.

This component has been developed using J2EE with a Web Service interface, so, in principle, any application server J2EE-compatible can host this web service. In our case, Tomcat 5.0.9 has been used and the service follows the specifications of a standard Web Service.

6. PBNM

6.1. Introduction

Policy Based Network Management (PBNM) goal is to help bridge part of the conceptual gap between a human policy maker and a network element that is configured to enforce the policy. When a human business executive defines network policy, it is usually done using informal business terms and language. For example, a human may come with a policy statement that reads: "traffic generated by our human-resources application should have higher priority for making it through to its destination compared to traffic generated by people browsing the WEB during their lunch breaks" (abstract policy). While this statement clearly defines QoS policy for the network, the network itself cannot enforce it. Translation to "network terms and language" (concrete configuration) is required.

Clearly this wide gap implies several translation levels, from the abstract to the concrete. Two levels will be analysed here: CIM ([CIM]) closer to human natural language and used in many Akogrimo systems -WebServices will use WSDL ([WSDL])- and COPS ([RFC2748]) (and its extension [RFC3084]) closer to configuration commands and used in QoS System.

6.2. Architecture

Akogrimo is currently deciding and designing the PBNM framework. Two models, CIM [CIM] and WebServices WS-Policy [WSPOLICY], seem to be the winning candidates for PBNM. They are focussed on two different aspects, CIM for network management and WS-Policy for services control. To give coherence to these two models, we propose integration in the highest (closest to human language) policy level.

Figure 25 depicts de PBM model that can be followed. It is divided in PDPs and PEPs (Policy Decision/Enforcement Points). PDPs take decisions and control and configure the PEPs who enforce the policies. PDPs and PEPs may communicate using e.g. COPS or DIAMETER protocols. The actual protocol depends on the system considered: COPS for QoS (see section 4.4), DIAMETER for A4C. PDPs can gather the policies from a PBNM Server (PBNMS). PDPs should communicate with the PBNMS all using the same protocol e.g. CIM (XML based) over HTTP. PBNMS will store the policies (e.g. in a LDAP repository) and offer a Graphical User Interface (GUI) for the manager to introduce the policies. Policy conflict is detected in the PBNMS. PBNMS can be integrated with the WebServices WS-Policy Policy Manager thus forming the coherent Akogrimo PBM. Both CIM and WS-Policy are based on XML. Details of this integration are to be provided, but on a first approach, two components that can be shared by PBNMS Server and PM are the policy repository and the GUI. Here we concentrate on the CIM approach and on QoS issues.

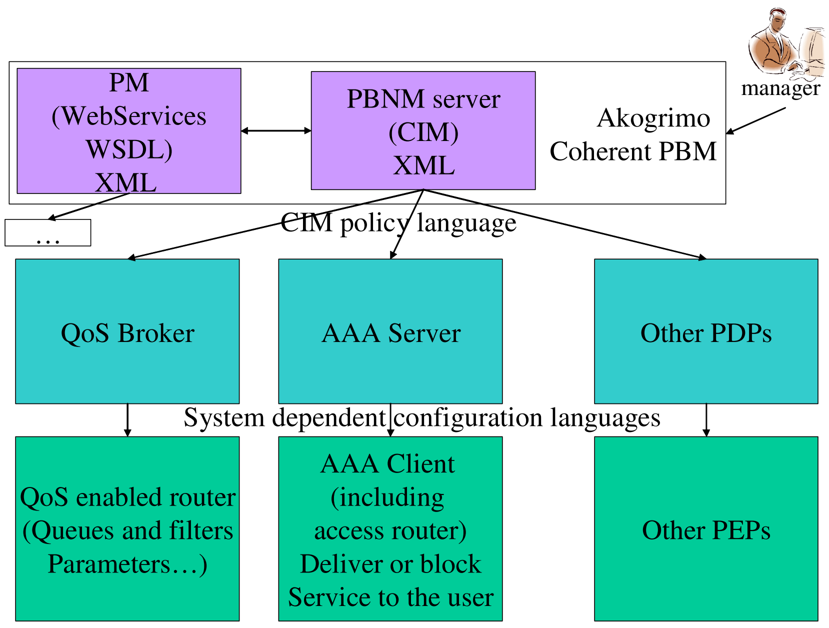

manager
PM (WebServices WSDL) XML
PBNM server (CIM) XML
Akogrimo Coherent PBM
…
CIM policy language
QoS Broker
AAA Server
Other PDPs
System dependent configuration languages
QoS enabled router (Queues and filters Parameters…)
AAA Client (including access router) Deliver or block Service to the user
Other PEPs

Figure 25 – PBM Architecture. Relation with other components

6.3. Sample scenarios

PBNM covers a wide range of scenarios and situations. For instance we have "policy conflict" cases. This is better explained by an example. We have two policies: policy 1 "users under 18 years old, have a 50% discount in MMS between 20:00 and 8:00" and policy 2 "gold profile users have a 60% discount in MMS between 21:00 and 9:00". If a user is gold and under 18, which policy should be applied?? Mechanisms handling these conflicts must be designed, for instance assigning priorities to policies, or policy "set-union" rules.

Other scenarios, more related to QoS, may be as following. Let's consider the following policy:

"On any interface to which these rules apply, guarantee at least 30% of the interface bandwidth to transport QoS class conversational, and at least 40% of the interface bandwidth to transport QoS class transactional."

Policies in the system level can be grouped in *PolicySet CIM* class. This class can also contain information about the roles of the entities to which these policies apply. In our example the desired role is an interface, because we want to allocate for QoS Classes an amount of bandwidth available through the interface. The resources assigned to specific roles are described in *PolicyRoleCollection* CIM class.

In the following scenario another aspect of PBNM system occurs. Let's consider the following policy:

"If the EF traffic load between A and B exceeds 70% of the allocated bandwidth, allocate more bandwidth to the EF traffic from other classes." (1)

This is an example of dynamically applied (reconfiguration) policy. In order to initiate this policy, the QoS system (namely the QoSBroker) needs to know from where and how to get more bandwidth. It means that in PBNM system other policies (describing any conditions) must have been defined earlier and sent to the QoSBroker. They can be configuration or reconfiguration policies.

Firstly, QoSBroker must have information from the measurement system (this system can be the same as the QoS System) about the actual situation in network. This implies that a policy should be supported by the measurement system. This policy could be:

"Provide measurements of the EF traffic between A and B every 10 minutes and send the trigger to QoS Broker if it exceeds 70% of the allocated bandwidth". (2)

Additionally, another policy defining how to get more bandwidth for EF traffic load is required. It can be defined in the following way:

"If the AF traffic load is under 40% you can take up to 30% of its unused bandwidth." (3)

Abovementioned policy is a typical network view level policy. It seems simple but in fact it engages a lot of aspects of network management. In order to enforce the policy it must be translated into lower levels of abstraction. It also interacts with other policies on the same (network) level.

Policies related to SIP proxies may also exist. One may think in a policy that reads:

If the user profile is not "high priority" and the network load is above 50% then, in any audio-video call, suppress the video codecs and force to make an audio-only call. (1)

This policy involves others systems besides the SIP proxy, namely the QoSBroker and the A4C system. As in the example above, policies should be defined for these systems. For instance, for the QoSBroker one may define a policy such as:

If the network load is above 50% then on any reservation request prevent the SIP Proxy about such a status and send an alarm to the SIP Proxy.(2)

6.4. Documents, standards, protocols & interfaces

The communication between PDPs and PEPs is well defined in standards. In our PBM model this communication can be different between the several systems, the coherence is achieved in the PBNM Server level. For QoS the PDP-PEP communication is based on COPS-PR [RFC3084] and the data representation on PIBs [RFC3571].

At the abstract end of the policies (closer to human language level) is CIM. CIM [CIM] is defined by DMTF. CIM facilitates a formal representation of network business rules, thus providing the first concretization level: formally representing humanly expressed policy. CIM is based on UML (Microsoft Visio files are available) and XML representation exists. CIM policies are based on Policy conditions (if …) and actions (then …). CIM covers a wide range of policies. Figure 26 represents an example of QoS policies definition. Due to the innovative nature of Akogrimo some new policies will need to be defined.

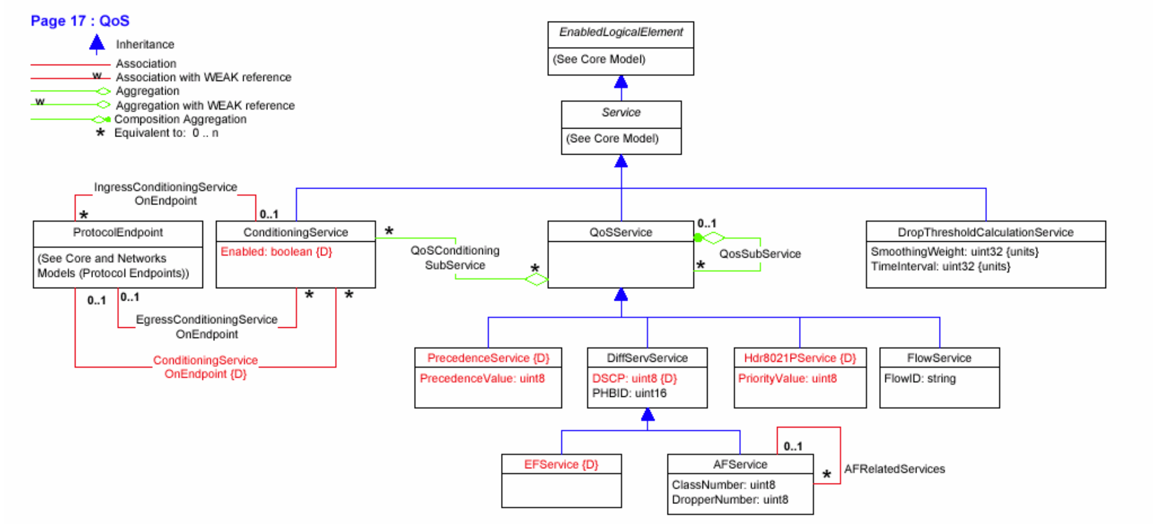


Figure 26 – A declaration of a QoS policy in CIM

As we said CIM is based on UML but XML representation exists. Figure 27 shows the QoS Service Class present in Figure 26 but defined in XML. The communication between the PBNM Server and the PDPs is based on this XML representation which is transported over HTTP.

```
<?xml version="1.0" ?>
- <CIM CIMVERSION="2.2" DTDVERSION="2.1">
- <DECLARATION>
- <DECLGROUP>
- <VALUE.OBJECT>
- <CLASS SUPERCLASS="CIM_Service" NAME="CIM_QoSService">
- <QUALIFIER TOSUBCLASS="false" TRANSLATABLE="true" NAME="Version" TYPE="string">
  <VALUE>2.7.0</VALUE>
  </QUALIFIER>
- <QUALIFIER TRANSLATABLE="true" NAME="Description" TYPE="string">
     <VALUE>This is a concrete class that represents ....</VALUE>
  </QUALIFIER>
  </CLASS>
  </VALUE.OBJECT>
  </DECLGROUP>
  </DECLARATION>
  </CIM>
```

Figure 27 – Representation of a CIM class in XML

We recall that for WebServices and service management [WSDL] will be the high (closer to human natural language) level language.

6.4.1. Example: mapping policies to CIM

We should see in this section, how one of the policies presented in the scenarios in section 6.3 can be mapped into CIM classes. Let's recall the policy:

"On any interface to which these rules apply, guarantee at least 30% of the interface bandwidth to transport QoS class conversational, and at least 40% of the interface bandwidth to transport QoS class transactional."

The proposed mapping of this policy for different level of abstraction is presented in Table 10

Levels		Sample Objective	Sample objects	Relevant model (see 6.4)
Domain	System	On any interface to which these rules apply, guarantee at least 30% of the interface bandwidth to transport QoS class Conversational, and at least 40% of the interface bandwidth to transport QoS class Transactional.	Conversational Transport Class	CIM (PCIM, PCIMe, QPIM)
	Network	If (DSCP value is 01) THEN guarantee 30% of available BW If (DSCP value is 02) THEN guarantee 40% of available BW	Conversational DiffServ Service	CIM (PCIM, PCIMe, QPIM)
Device	Device	A possible implementation of these rules within a PEP (router in this case) would be to use a Weighted-Round-Robin scheduler with 3 queues. The first queue would be used for DSCP01 traffic, the second queue for DSCP02 traffic and the third queue for the rest of the traffic. The weights of the Weighted-Round-Robin scheduler would be 30% for the first queue, 40% for the second queue and 30% for the last queue.	Routers, Interfaces, Queues	CIM (PCIM, PCIMe, QDDIM)
	Instance	Specific commands configuring the queues of the devices.	Objects representing PIBs	PIB, COPS

Table 10 – Illustration of a process of mapping a policy from system level to instance level

6.5. Implementation

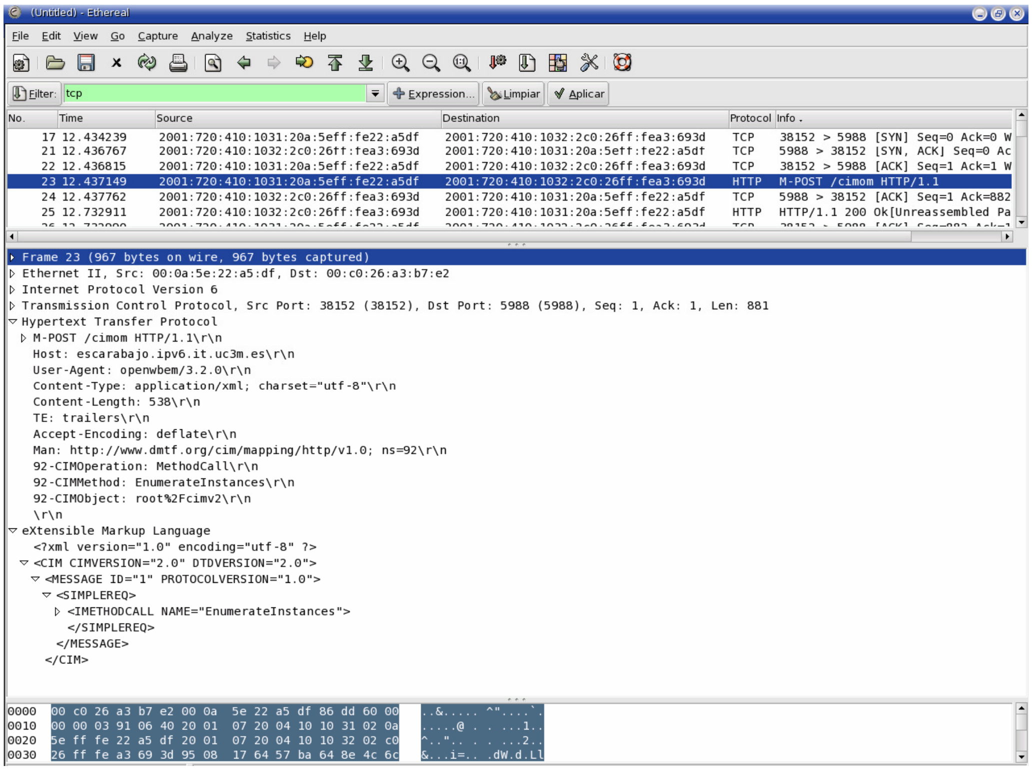
The first implementation will follow closely the work done in the IST-Daidalos project. For PBNMS-PDPs communications we will use CIM and the OpenWBEM [OPENWBEM] tool. It is free source code and not based on XML but on Managed Object Format (MOF) , but this should cause no problems. Indeed MOF is also an accepted representation of CIM (Figure 28).

```
// Copyright (c) 2005 DMTF.  All rights reserved.
// ==================================================================
// CIM_QoSService
// ==================================================================
  [Version ( "2.7.0" ), Description (
      "This is a concrete class that represents….")]
class CIM_QoSService : CIM_Service {
};
```

Figure 28 – Representation of a CIM class in MOF

Note that other open source tools for managing PBM CIM framework are available, such as OpenPegasus [OPENPEGASUS]. OpenPegasus and OpenWBEM offer very similar characteristics, indeed both could be employed.

The interface between the PBNMS and the management entities (PDPs) can be implemented as a CIM OpenWBEM provider running in the PBNMS. Currently this provider main functionality is to handle Akogrimo policy extensions and, based on the type of policy considered, "forward" it to the appropriate PDP(s). HTTP and CIM-XML (over IPv6) are used for this communication, see for instance Figure 29, where the PBNMS address is 2001:…:a5df and the PDP's (a QoS Broker) one is 2001:…:693d.



Filter: tcp

```
No.     Time       Source                               Destination                          Protocol Info
     17 12.434239  2001:720:410:1031:20a:5eff:fe22:a5df  2001:720:410:1032:2c0:26ff:fea3:693d  TCP      38152 > 5988 [SYN] Seq=0 Ack=0 W
     21 12.436767  2001:720:410:1032:2c0:26ff:fea3:693d  2001:720:410:1031:20a:5eff:fe22:a5df  TCP      5988 > 38152 [SYN, ACK] Seq=0 Ac
     22 12.436815  2001:720:410:1031:20a:5eff:fe22:a5df  2001:720:410:1032:2c0:26ff:fea3:693d  TCP      38152 > 5988 [ACK] Seq=1 Ack=1 W
     23 12.437149  2001:720:410:1031:20a:5eff:fe22:a5df  2001:720:410:1032:2c0:26ff:fea3:693d  HTTP     M-POST /cimom HTTP/1.1
     24 12.437762  2001:720:410:1032:2c0:26ff:fea3:693d  2001:720:410:1031:20a:5eff:fe22:a5df  TCP      5988 > 38152 [ACK] Seq=1 Ack=882
     25 12.732911  2001:720:410:1032:2c0:26ff:fea3:693d  2001:720:410:1031:20a:5eff:fe22:a5df  HTTP     HTTP/1.1 200 Ok[Unreassembled Pa
```

```
▷ Frame 23 (967 bytes on wire, 967 bytes captured)
▷ Ethernet II, Src: 00:0a:5e:22:a5:df, Dst: 00:c0:26:a3:b7:e2
▷ Internet Protocol Version 6
▷ Transmission Control Protocol, Src Port: 38152 (38152), Dst Port: 5988 (5988), Seq: 1, Ack: 1, Len: 881
▽ Hypertext Transfer Protocol
  ▷ M-POST /cimom HTTP/1.1\r\n
    Host: escarabajo.ipv6.it.uc3m.es\r\n
    User-Agent: openwbem/3.2.0\r\n
    Content-Type: application/xml; charset="utf-8"\r\n
    Content-Length: 538\r\n
    TE: trailers\r\n
    Accept-Encoding: deflate\r\n
    Man: http://www.dmtf.org/cim/mapping/http/v1.0; ns=92\r\n
    92-CIMOperation: MethodCall\r\n
    92-CIMMethod: EnumerateInstances\r\n
    92-CIMObject: root%2Fcimv2\r\n
    \r\n
▽ eXtensible Markup Language
    <?xml version="1.0" encoding="utf-8" ?>
  ▽ <CIM CIMVERSION="2.0" DTDVERSION="2.0">
    ▽ <MESSAGE ID="1" PROTOCOLVERSION="1.0">
      ▽ <SIMPLEREQ>
        ▷ <IMETHODCALL NAME="EnumerateInstances">
          </SIMPLEREQ>
        </MESSAGE>
      </CIM>
```

```
0000  00 c0 26 a3 b7 e2 00 0a  5e 22 a5 df 86 dd 60 00   ..&..... ^"....`.
0010  00 00 03 91 06 40 20 01  07 20 04 10 10 31 02 0a   .....@ . . ...1..
0020  5e ff fe 22 a5 df 20 01  07 20 04 10 10 32 02 c0   ^.."... . . ...2..
0030  26 ff fe a3 69 3d 95 08  17 64 57 ba 64 8e 4c 6c   &...i=.. .dW.d.Ll
```

Figure 29 – Communication between the PBNM Server and a PDP

OpenWBEM provider implementation is based on a CIMOM (CIM Object Manager) loadable library, a UNIX .so file, completely controlled by the CIMOM. OpenWBEM proposes a remote provider concept: remote providers use a light-weight CIMOM implemented in the managed entity (the PDP) that, making use of a loadable library, implements the provider interfaces to the managed entity. In the managed entity (the PDP) the provider is in charge of translating the CIM policy to PDP understandable policies. Figure 30 illustrates the remote provider concept.

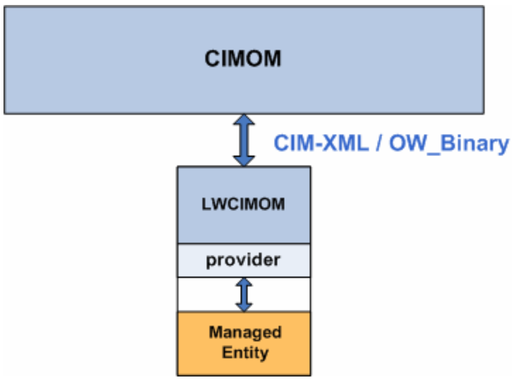
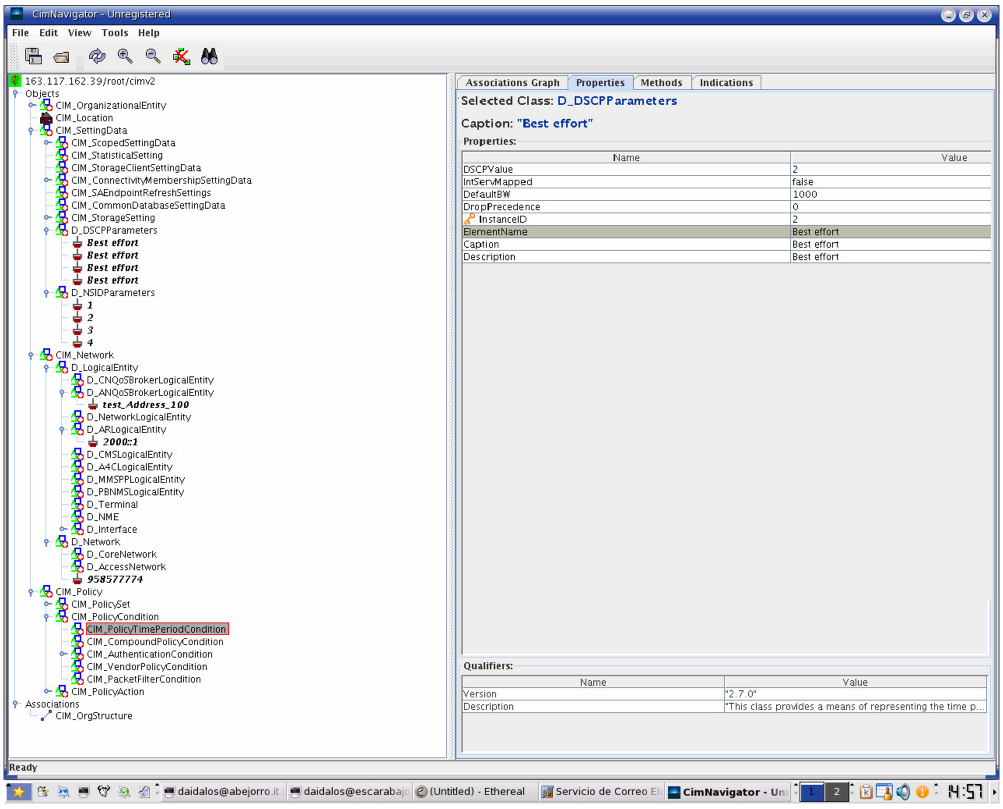


Figure 30 – CIM Remote Providers

The communication between both CIMOMs is performed in CIM-XML over HTTP. The light-weight CIMOM interfaces the managed entity. This is exploited in the implementation of some units. OpenWBEM CIMOM is able to store policies on a LDAP repository and offers a GUI (Figure 31) for controlling them. We can check there the Akogrimo Policy extensions (termed D_***) that will be handled by the implemented provider in the PBNM Server and sent to the appropriate provider(s) in the PDP(s).



```
163.117.162.39/root/cimv2
Objects
    CIM_OrganizationalEntity
    CIM_Location
    CIM_SettingData
        CIM_ScopedSettingData
        CIM_StatisticalSetting
        CIM_StorageClientSettingData
        CIM_ConnectivityMembershipSettingData
        CIM_SAEndpointRefreshSettings
        CIM_CommonDatabaseSettingData
        CIM_StorageSetting
        D_DSCPParameters
            Best effort
            Best effort
            Best effort
            Best effort
        D_NSIDParameters
            1
            2
            3
            4
    CIM_Network
        D_LogicalEntity
            D_CNQoSBrokerLogicalEntity
            D_ANQoSBrokerLogicalEntity
                test_Address_100
            D_NetworkLogicalEntity
            D_ARLogicalEntity
                2000::1
            D_CMSLogicalEntity
            D_A4CLogicalEntity
            D_MMSPPLogicalEntity
            D_PBNMSLogicalEntity
            D_Terminal
            D_NME
            D_Interface
        D_Network
            D_CoreNetwork
            D_AccessNetwork
            958577774
    CIM_Policy
        CIM_PolicySet
        CIM_PolicyCondition
            CIM_PolicyTimePeriodCondition
            CIM_CompoundPolicyCondition
            CIM_AuthenticationCondition
            CIM_VendorPolicyCondition
            CIM_PacketFilterCondition
        CIM_PolicyAction
Associations
    CIM_OrgStructure
```

Selected Class: D_DSCPParameters

Caption: "Best effort"

Properties:

Name	Value
DSCPValue	2
IntServMapped	false
DefaultBW	1000
DropPrecedence	0
InstanceID	2
ElementName	Best effort
Caption	Best effort
Description	Best effort

Qualifiers:

Name	Value
Version	"2.7.0"
Description	"This class provides a means of representing the time p...

Figure 31 – GUI for the CIMOM (PBNM Server)

In the QoS Broker, its provider will "translate" the policies into a mysql database understandable by the QoS Broker. The interface between the provider in the QoS Broker and the PBNMS' Provider is bidirectional: the QoS brokers will request the PBNMS for a configuration at startup time, and along the time it will send alarms/notifications to the PBNMS; the PBNM Server (PBNMS) will send a policy definition whenever it is (re)defined by the human operator or by automatic application triggers.

The QoS broker configuration information may include the network topology description, the QoS description of the services offered by the operator, and the initial resource distribution that must be defined in its domain. The services available are described by the PBNMS in terms of the DSCP code, the required resources both for the downlink and the uplink flows. It is very important for the operator to have a central network entity defining, in an integrated way, the service QoS parameters. A central service parameter definition avoids network configuration inconsistencies as well as it makes the configuration job much easier. Furthermore, when a service redefinition occurs the PBNMS performs an almost instantaneous QoS entities configuration. Figure 32 illustrates the QoSBroker policy database.

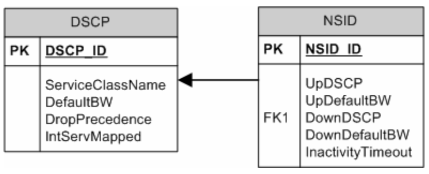


Figure 32 – QoSBroker Policy database

The QoS network configuration is performed accordingly to the network usage. The broker can also use historical network usage information namely at start-up. The initial resource distribution is sent by the PBNMS. Several CIM classes will need to be developed in order to extend CIM model in order to support the QoS Broker model.

7. Security

7.1. Introduction

Security tackles a wide range of different fields and one of its characteristics is that is something we can't get enough of. No matter how many security mechanisms are provided and how strict they are there is always a something left out. The reason being that security is a trade off with usability. Usually the more secure a system is, the less usable it is. The objective is always to provide a fairly secure system optimizing usability. There is a need to define how far one would like to go in terms of security and draw a line delimiting an acceptable level of security that is considered more than necessary for the purposes aimed. For instance, a user could be requested to be authenticated in order to make a transaction of a considerable amount of money, but just be requested to show a token in case of a small amount purchase, making it less secure but faster.

7.2. Attack model

In order to be able to deliver a consistent security architecture, first the targets and severity of attacks that can be performed need to be identified. Table 11 shows the attack relations between the main Akogrimo actors. The most relevant attacks fields are coloured in red whereas secondary ones are coloured in green; the rest of the fields are considered out of scope and left white.

Attacker

Target

	User	Terminal	OpVO	BaseVO	Customer Domain	Network Provider	Service Provider
User	4/4/4 relevant but out of scope	4/2/2	1/2/1	2/3/2	4/2/2	4/3/3 Not specific to Akogrimo	1/1/1 Accounting; SLA
Terminal	4/3/2	4/3/2	4/3/2	4/3/2	4/3/2	4/3/2	4/3/2
OpVO	1/1/1	1/1/1	1/2/1	1/3/1	1/2/1	4/4/4	1/3/1
BaseVO	1/1/1	1/1/1	1/2/1	4/4/4	1/2/1	2/3/2	1/2/1
Customer Domain	4/1/1	2/2/2	1/3/1	1/3/1	4/3/3	4/3/2 Not specific to Akogrimo	1/3/1
Network Provider	4/3/3	4/2/2	2/3/2	2/3/2	4/4/4	4/3/2 Not specific to Akogrimo	4/4/4
Service Provider	2/2/2	3/3/3	1/1/1	1/3/1	1/2/1	1/3/1	1/3/1

Relevance in Ako/Likelihood in Ako/Impact
1 high - 4 low

Table 11 – Attack model

From this table one can infer what fields need to be tackled in WP4.1 in terms of security. The provisioning of security in WP4.1 will therefore be aligned to the following directions:

- Provisioning of security aligned to the research done in the project and therefore analyzing in detail the security issues arisen within the context of the new functionalities developed (e.g. SIP with SOAP interactivity). From the table one can easily infer that this is the issue with more preference, as most fields related to OpVO and BaseVO are considered of high priority.
- Provisioning of security as a whole, in order to provide guidelines for the security of the complete platform developed. (e.g. provide authentication and Single Sign On (SSO)) and avoid the introduction of insecurities that come to the hand of the technologies used (e.g. weaknesses introduced by the use of MIPv6).
- Provision of security on the tools used for implementation. (e.g. Java stubs). This point is of lower priority as it is not Akogrimo specific.

In general very general terms, the security mechanisms should strive to achieve the following goals:

- Provide Privacy
- Avoid Impersonation, man-in-the-middle, replay attack by means of Authentication
- Establish a robust Authorisation system.
- Provide integrity of network messages.
- Avoid DoS, DDoS attacks
- Provide Non-repudiation, and attacks using false information or internal attacks by means of Auditing.

7.3. Network substrate attack scenarios

This chapter lists relevant attacks to the Network substrate. Some of them are more relevant than others in the Akogrimo context and therefore will be treated in more detail.

7.3.1. Mobility/MIPv6

The use of MIPv6 introduces new insecurities over regular IPv6 that need to be palliated. Among them we find: MN impersonation, HA impersonation, MiM or DoS. They all exploit the fact that the mobile terminal needs to update its position when doing a handover, by manipulating, blocking this information or sending false frames.

7.3.2. Attacks to Access Router

The threats that could suffer the Access Router in the Access Network could be divided in two sections, external and internal. External refers to the AR's interface to the access network and internal refers to the interface to the core network.

7.3.2.1. External attacks

- DoS (Denial of Service)
    - There is no getting around these kind of attacks
    - If the terminal is only allowed to use "well-behaved" Akogrimo software, then DoS attacks are not possible
    - Unauthenticated users get their packets dropped. However, a big number of simultaneous attackers can be effective at DoS'ing the AR.

  - Various possibilities
    - Low level radio signal jamming
    - Unauthenticated user flooding the wireless network with bogus packets
    - Unauthenticated user sending the AR valid packets that must be analysed
- Impersonation
  - The AR works in layer 3. An attacker can spoof packets with a valid user's ip address. Authentication+authorisation+token should stop this from happening.

These attacks are not specific to Akogrimo, and consequently they are not a high priority. Some safety measures against impersonation attacks are considered, but not against DoS.

7.3.2.2. Internal Attacks

- Malicious QoS Broker sends wrong rules, for benefiting or hurting a user(s)
- Malicious A4C server can send fake user information
- Malicious QoS Broker can send fake rules; can benefit a certain user, giving him QoS level which he would not get normally
- Any component can cause DoS attacks

The AR has a few basic safety measures, such as requiring a configuration for each of the components that are able to communicate with it. That coupled with authentication of the components between them reduces risks considerably, as a bad misuse or attack can more easily be attributed to a concrete component and therefore restrict its access or take any other measure against it.

7.3.3. QoS Broker

The QoS Broker receives requests from the Access Routers and the EMS Web Service. With user profile information it gets from the A4C Server, and also information about the current network status, it decides if a user is allowed a network flow at a given QoS level.

The same than before, there are two kinds of attacks, external and internal attacks concerning the QoS Broker.

7.3.3.1. External attacks

- Authenticated user(s) causing repeated requests to the QoS Broker (DoS)
  - Since the AR gets rules installed when a new flow is requested, this will only cause problems if the user starts a large number of different flows. With typical Akogrimo software this is not possible, so the user's MT would have to have been compromised and special software installed on it.
  - Several users can cooperate, maximizing the attack

7.3.3.2. Internal attacks

- Malicious A4C server can send fake user profile
- Malicious AR can
  - use any kind of data collected from innocent users

  - Perform a DoS attack on the QoSB
- Malicious EMS can also benefit a certain user; can cause repeated QoS requests
- Any component can cause DoS attacks

As in the AR, the QoS Broker has basic safety measures that coupled with authentication of the components between them reduces these risks considerably.

7.3.4. PBNM

The PBNM is mostly sensible to internal attacks performed by another Akogrimo component such as:

- PBNM manager impersonation
- Fake Profiles obtained
- Policies not enforced correctly

7.3.5. Attacks to SIP signalling

The SIP entities that are involved in the Akogrimo SIP infrastructure are the Mobile Terminal, the Proxy SIP (in the Access Network) and finally the SIP Server and the SIP Broker in the Core Network.

The possible attacks that could suffer each one of these SIP entities could be the following.

7.3.5.1. SIP with SOAP security problem

The SIP with SOAP interaction in Akogrimo takes place in the component called "SIP Broker", which basically exposes some SIP functionalities as a WS (typically invoked by the Workflow Manager). To be precise, in the prototype version two functionalities were offered:

- Grid-initiated SIP Call, which enables the possibility to arrange a grid call between two SIP users.
- Grid-initiated SIP transfer, which enables the possibility to inform a user he/she can move an ongoing session to a most suitable device.

Other functionalities (e. g. a WS interface to expose the SIP SD facilities) are being considered to be included in next versions. So we can consider that the SIP with SOAP approach in Akogrimo mainly relays on the capabilities that this component offers.

The internal structure of the SIP Broker consists mainly on three differentiated main submodules:

- The **SOAP interface**, which accepts incoming WS requests from the "grid world".
- The **Broker Engine**. This component is the core of the SIP Broker. Its main task is to convert the WS request parameters (typically the Akogrimo IDs of the user to be put in contact, or the device URI the user is allowed to move to) into a suitable format that the SIP UA can understand. This is made through an A4CServer's user profile request (using an embedded A4C client), that returns – apart from other information - the SIP AoR of the user. For efficiency purposes, and considering that only one SIP UA and one A4C client are needed, it has been implemented as a separate Java process that can be accessed through Java RMI.

- The **SIP UA**, which handles the SIP process itself. It has been implemented as a Java library that other components can use.

In order to avoid that some non-authorised entity can make use of the facilities that the SIP Broker can provide, security mechanisms must be implemented to protect each possible access path to this component. In next section both the authorised way to use the SIP Broker and the possible attacks are described.

Apart from attacks, another crucial consideration must be taken into account. The currently implemented interactions (grid triggered SIP calls and SIP session transfers) take place within the context of a certain VO, i.e. some entity decides that two members of an opVO should have a conversation, or one of the members should be requested to transfer an ongoing session it has with another opVO component to a most suitable device. The distributed perimeter security model adopted by the opVO means that interactions between entities can only occur in the context of a specific opVO instance. This issue is considered in section 7.4.5.1.

7.3.5.1.1. SIP Broker threat model

Figure 33 shows both the authorised way to use the SIP Broker and the potential attacks we have identified.

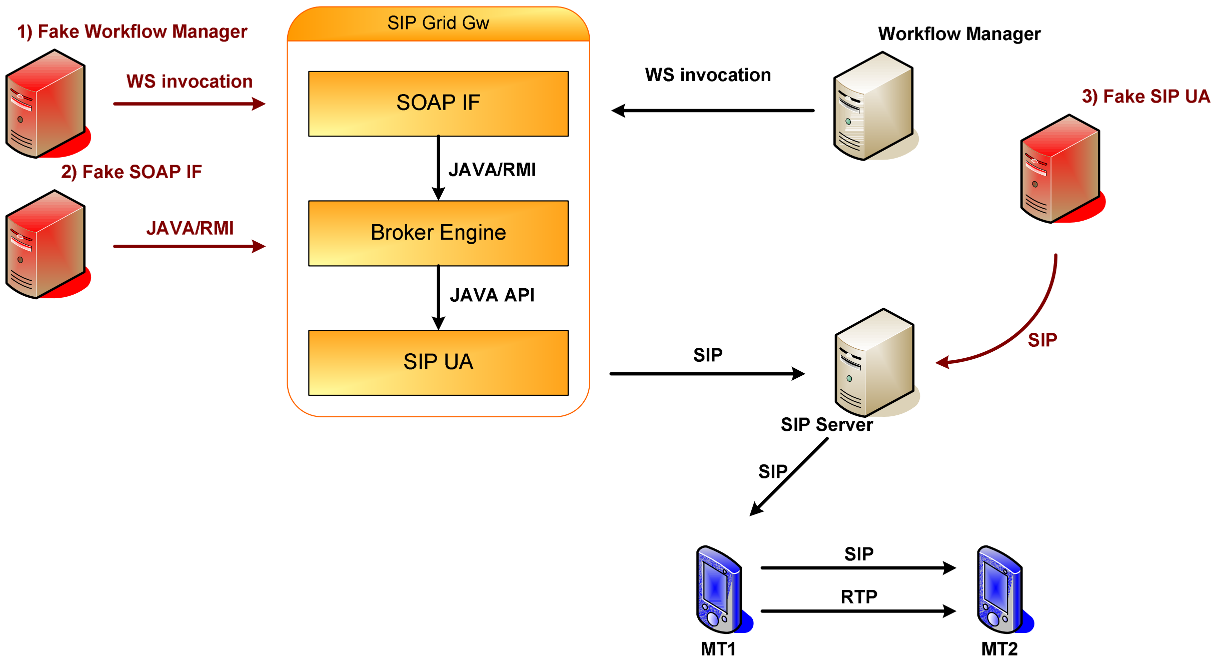


Figure 33 – "SIP with SOAP" possible attacks

During a normal (authorised) operation, the Workflow Manager will request for a SIP process using the interface that the SOAP interface subcomponent provides; then it invokes the corresponding method of the Broker Engine using Java RMI, which will invoke the Java API provided by the SIP UA to start the corresponding SIP mechanism (arrange a call or request for a transfer in the current prototype). These interactions have been depicted with black arrows in the figure.

We have identified three types of potential attacks, depending on the faked entity. Non-authorised element and interactions have been depicted in red:

1. If the SOAP interface does not check the identity of the invoking element (i. e. the Workflow Manager), it could be possible that a non-authorised element makes use of the exposed WS interface to make SIP call or SIP transfer requests. So in order to avoid this

kind of attacks, this WS interface should implement authorisation mechanisms. This is a **pure WS interface security problem**.

2. The broker engine is exposed as a RMI daemon, so it could be invoked from a remote machine. So if the identity of the calling entity is not guaranteed, some entity which knows the RMI URL could fake the identity of the SOAP interface and make non-authorised requests; to solve this, the JAVA RMI interface must guarantee that only the real SOAP interface is allowed to use the broker engine. This is a **pure Java RMI security problem**.
3. Finally, some malicious SIP entity could pose as the whole SIP Broker (to be precise, posing as the SIP UA) to send false SIP requests to end user's MTs. If neither the SIP Server nor the end user terminal checks for the request authenticity, this non-authorised entity could send a request to a MT to initiate a call or to perform a transfer. Anyway, this is a **pure SIP security problem**.

So all possible attacks in which some entity poses as the SIP Broker (of any of its components) can be reduced to a technology-related security problem (security with WS, RMI or SIP interactions). Chapters 7.4.5.2 and 7.4.5.3 describe these different security mechanisms.

7.3.5.2. Registration Hijacking

The SIP registration mechanism allows a user agent to identify itself to a registrar as a device at which a user is located, in Akogrimo project the SIP address are as *user@domain* A registrar assesses the identity asserted in the "From" header field of a REGISTER message to determine whether this request can modify the contact addresses associated with the address-of-record in the "To" header field. While these two fields are frequently the same, there are many valid deployments in which a third-party may register contacts on a user's behalf.

The "From" header field of a SIP request, however, can be modified arbitrarily by the owner of a UA, and this opens the door to malicious registrations. An attacker that successfully impersonates a party authorised to change contacts associated with an address-of-record could, for example, de-register all existing contacts for a URI and then register their own device as the appropriate contact address, thereby directing all requests for the affected user to the attacker's device.

This threat type belongs to a family of threats that rely on the absence of cryptographic assurance of a request's originator. Any SIP UAs that represents a valuable service, for instance the SIP Broker, might want to control the access to its resources by authenticating requests that it receives. Even an end-user (MT) is interested in ascertaining the identities of originator of request.

This threat demonstrates the need for security services that enable SIP entities to authenticate the originators of requests.

7.3.5.3. Impersonating a Server

The domain to which a request is destined is generally specified in the Request-URI. UAs commonly contact a server in this domain directly in order to deliver a request. However, there is always a possibility that an attacker could impersonate the remote server, and that the UA's request could be intercepted by some other party.

In Akogrimo project this attack type is especially important, in case of impersonating a SIP Broker [1], since an attacker could send a REFER message to an end user (for instance the doctor) in order to establishes a grid call with other user (the patient).

7.3.5.4. Tampering with Message Bodies

Consider a UA that is using SIP message bodies to communicate session encryption keys for a media session. Although it trusts the proxy server of the domain it is contacting to deliver signalling properly, it may not want the administrators of that domain to be capable of decrypting any subsequent media session. Worse yet, if the proxy server were actively malicious, it could modify the session key, either acting as a man-in-the-middle, or perhaps changing the security characteristics requested by the originating UA.

This family of threats applies not only to session keys, but to most conceivable forms of content carried end-to-end in SIP. These might include MIME bodies that should be rendered to the user, SDP, or encapsulated telephony signals, among others. Attackers might attempt to modify SDP bodies, for example, in order to point RTP media streams to a wiretapping device in order to eavesdrop on subsequent voice communications.
For these reasons, the UA might want to secure SIP message bodies, and in some limited cases header fields, end-to-end. However, since many header fields are legitimately inspected or altered by proxy servers as a request is routed, not all header fields can be secured end-to-end.

The security services required for message bodies include confidentiality, integrity, and authentication. These end-to-end services should be independent of the means used to secure interactions with intermediaries such as proxy servers.

7.3.5.5. Tearing Down Sessions

Once a session has been established by initial messaging, subsequent requests can be sent that modify the state of the dialog and/or session. It is critical that principal participants in a session can be certain that such requests are not forged by attackers.

Consider a case in which a third-party attacker captures some initial messages in a dialog shared by two parties in order obtain the parameters of the session ("To" header, "From" header, and so on) and then inserts a BYE request into the session. The attacker could opt to forge the request such that it seemed to come from either participant. Once the BYE is received by its target, the session will be torn down prematurely.

Similar mid-session threats include the transmission of forged re-INVITEs that alter the session.

The most effective countermeasure to this threat is the authentication of the sender of the BYE message. In this instance, the recipient needs only know that the BYE came from the same party with whom the corresponding dialog was established. Also, if the attacker is unable to learn the parameters of the session due to confidentiality, it would not be possible to forge the BYE message. However, some intermediaries (like proxy servers) will need to inspect those parameters as the session is established.

7.3.5.6. Denial of Service and Amplification

Denial-of-service attacks focus on rendering a particular network element unavailable, usually by directing an excessive amount of network traffic at its interfaces. A distributed denial-of-service attack allows one network user to cause multiple network hosts to flood a target host with a large amount of network traffic.

SIP creates a number of potential opportunities for distributed denial-of-service attacks that must be recognized and addressed by the implementers of SIP systems.

Attackers can create bogus requests that contain a falsified source IP address and a corresponding "Via" header field that identify a targeted host as the originator of the request and then send this request to a large number of SIP network elements, thereby using hapless SIP UAs or proxies to generate denial-of-service traffic aimed at the target.

Similarly, attackers might use falsified "Route" header field values in a request that identify the target host and then send such messages to forking proxies that will amplify messaging sent to the target.
"Record-Route" could be used to similar effect when the attacker is certain that the SIP dialog initiated by the request will result in numerous transactions originating in the backwards direction.

A number of denial-of-service attacks open up if REGISTER requests are not properly authenticated and authorised by registrars.
Attackers could de-register some or all users in an administrative domain, thereby preventing these users from being invited to new sessions. An attacker could also register a large number of contacts designating the same host for a given address-of-record in order to use the registrar and any associated proxy servers as amplifiers in a denial-of-service attack. Attackers might also attempt to deplete available memory and disk resources of a registrar by registering huge numbers of bindings.

The use of multicast to transmit SIP requests can greatly increase the potential for denial-of-service attacks.

These problems demonstrate a general need to define architectures that minimize the risks of denial-of-service, and the need to be mindful in recommendations for security mechanisms of this class of attacks.

7.4. Defences

Spotting the most important fields in which security work in akogrimo needs to be done help us focus on the targets that need to be secured, in addition to that the review of possible relevant attacks is used as a guidance to define a generic defence architecture to a broader range of attacks on the previously defined targets.

7.4.1. Assumptions

To focus even more on the most interesting parts of security we will assume the following premises:

- No attacks will be executed by an Akogrimo component from the core network towards another akogrimo component. The case of having the QoSB attacking the PBNM, although possible, does not seem to be quite probable and therefore it will not be considered, as typically network operators consider their core network secure.

- The core is considered secure, communications inside the core are not expected to be eavesdropped or manipulated as long as they don't leave the network's administration area. However, if communications transverse different administration areas, or imply two user end points or a user endpoint and the network, this is considered as communication outside of the core and therefore efficient security mechanisms should be developed.
- Misuse from part of the user that can compromise his own security is not contemplated. That is, cases like the user loosing his keychain, physical engineering, user forgetting passwords, etc… are out of scope.
- Users need to be previously registered as Akogrimo users before making use of it.
- The user's device is considered to be the security endpoint. Usually that implies not more than one *simultaneous* user per handled (device). This could be extended in the future.
- Security solutions at L2 are not considered since the Akogrimo architecture is all-IP access technology independent. So WEP, WPA or any other L2 based security mechanism is neither taken for granted nor expected and will not be considered as part of this work. Although, in principle, not necessary, its use can be optional as additional security means.
- Insecurity related to host internal software security such as software bugs, rootkits, viruses, OS exploits as well as physical access attacks fall out of scope.
- Highly sophisticated and complicated cryptographic attacks that are highly improbable are not considered. If a cryptographic algorithm is considered secure by the community so will we. Some algorithms that have been broken can be still considered secure such as the SHA-1 hash, considered broken but not necessary vulnerable. (brute force attack to find collisions would take fewer than 2^{80} operations whereas using a know attack "only" 2^{69} operations)

7.4.2. Perimeter security

In order to maintain the core secure, the perimeter of the core needs to be efficiently secured. To this end traffic coming from the Access network or the Internet will have to be correspondingly secured.

7.4.2.1. Perimeter defence

It will be enforced that all access through the core or to elements in the core, will have to be authenticated, authorised and audited. This reduces possible attacks to well known and auditable users. In order to do this, we need means to authenticate users, enforce that they communicate just with whom they are allowed to and encrypt communications between users and the network. On the other hand, the user will also want to authenticate that the network he is using is the one he thinks he is using in order to avoid attacks towards users by someone trying to pretend to be an access network. This implies then bi-directional authentication.

- For authentication, the A4C will be used as shown in the Figure 34. PANA will be used to convey the messages from the MT to the Access network and Diameter will convey the messages inside the A4C infrastructure.

- The AR is the first line of defence of the Akogrimo network. Its role is to prevent unauthorised users from accessing the Akogrimo network, and to forward valid packets from authorised users to their destination. The AR uses, at its core, Linux kernel netfilter, which given a set of rules, determines whether or not to drop (or block) a packet. A first basic rule that would need to be enforced is that no traffic without corresponding authentication or encryption will be let in.

In this section we make a short explanation of how the AR works internally, and then proceed to analyse what attacks can be made and what protections the AR has.

- **Network flows**

  A network flow is identified by protocol, source and destination addresses, their respective port numbers and DSCP code.

- **Rules**

  Netfilter is linux's packet filter. It looks at packet headers and then decides whether or not to drop packets based on rules with which can be configured

  The AR gets information about how to configure its netfilter rules from the QoS Broker.

  A rule identifies a network flow and has information on what to do with it.

- **Encryption**

  IPSec in transport mode is possible. Packets will be encrypted from the MT to the AR, where they are decrypted, allowing the AR to perform its operations. Then, after traversing the core network and reaching the AR that serves the destination MT, they are once again encrypted and finally delivered to that MT.

- **AR General Characteristics**

  - Only accepts packets from authenticated users. All other packets are dropped.
  - When a packet from a new flow arrives, it checks with the QoS Broker if it's allowed
  - When a packet from an already known flow arrives, filters are already in-place and the packet is dealt with without further requests to the QoS Broker.
  -

- Upon successful bidirectional authentication the A4C will securely convey to the AR and the MT enough cryptographic material to create corresponding Security Associations and therefore bootstrap an IPsec tunnel between both. Connections between A4C and AR are considered secured as they both are inside the core network and connections between A4C and MT are encrypted and the material is only provided after successful authentication.

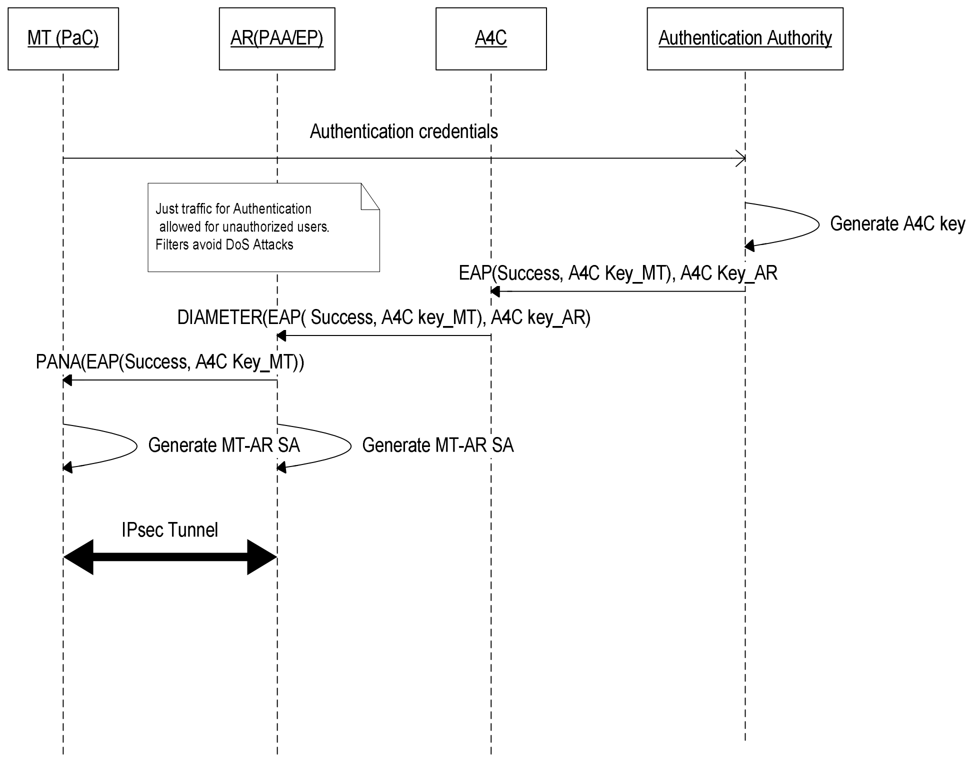

MT (PaC)
AR(PAA/EP)
A4C
Authentication Authority
Authentication credentials
Just traffic for Authentication allowed for unauthorized users. Filters avoid DoS Attacks
Generate A4C key
EAP(Success, A4C Key_MT), A4C Key_AR
DIAMETER(EAP(Success, A4C key_MT), A4C key_AR)
PANA(EAP(Success, A4C Key_MT))
Generate MT-AR SA
Generate MT-AR SA
IPsec Tunnel

Figure 34 – Network Authentication

7.4.3. Mobility/MIPv6

The use of MIPv6 in case of interdomain mobility reserves some attention in terms of security. To avoid the weak points MIPv6 introduces, the use of IPsec security mechanisms in transport mode, as described in [DuCo05, RFC3776], to ensure integrity in Binding Updates (BU) and Binding Acknowledgements (BA) is mandatory.

7.4.4. PBNM Defence

The PBNM just communicates with other components inside the same core network and therefore no special network security mechanisms are contemplated. It is suggested, nevertheless, the use of IPsec in transport mode and the use of the Akogrimo PKI infrastructure for authentication. The weak points here are not considered relevant and therefore work will be undertaken on other more important tasks instead.

7.4.5. SIP Defence

In the following the most relevant candidates for sip defence are presented.

Rather than defining new security mechanisms specific to SIP, SIP reuses wherever possible existing security models derived from the HTTP and SMTP space. Several security mechanisms are analysed and describe at Annex E of this doc, such as, IPSec, TLS, S/MIME, HTTP Digest and so on.

The implementation in Akogrimo project of some certificate-based security mechanism, like TLS [TLS], will provide data integrity and confidentiality, preventing from MiM attacks. Other authentication mechanisms, such as HTTP Digest could be used, in order to provide good DoS attacks protection.

By using IPSec security mechanisms direct attacks on the MTs could be avoided making use of peer-to-peer nature of SIP, therefore avoiding for instance that the attacker sends a request to the MT, impersonating both the SIP Server and the SIP Broker.

Finally the implementation of S/MIME provides the following cryptographic security services for electronic messaging applications:

- Authentication, message Integrity and non-repudiation of origin (using digital signatures), and Data Confidentiality (using encryption).
- Since the general use of S/MIME is to secure the message bodies, such as the SDP rather than message headers, it seems most logical, that S/MIME enables to avoid attacks like those described in chapter 7.3.5.4 "Tampering with message bodies".

It is worth noting as a drawback that S/MIME (the same as IPSec) generates considerable overhead in SIP messages.

So as conclusion, each one of these security mechanisms has some advantage and some drawbacks, therefore is possible that several of them are implemented in Akogrimo project.

Besides these mechanism related to own SIP architecture, Akogrimo will implement another security mechanism "SIP&SOAP Security Model", which will provide an additional security level.

7.4.5.1. SIP&SOAP Security Model (VO Security)

The adopted solution to take into account the opVO security perspective takes advantage on the extensibility and backward compatibility of the SIP protocol and consists on the definition of a new header field containing the opVO membership token, which will be included into the SIP signalling messages involved during the grid-triggered SIP call or SIP transfer. This new header will be processed at application level at the MTs, providing end-to-end security. Anyway, it will have a meaning only within the Akogrimo context; any other "non-Akogrimo" SIP entities are requested to ignore it, as SIP protocol specifies.

So using this approach, when a MT is requested to be the initiator of a SIP call, it can check if the request coming from the SIP Broker contains a valid opVO token prior to start the communication. In this way, the initiating MT can check that the call is authorised. The only requirement is, this information should be included somehow in the Workflow Manager request to the SIP Broker, because the SIP call / transfer is sent to the MTs by the SIP Broker. in response to a WS request from the WorkFlow Manager.

In the other hand, and in the case of a grid triggered SIP call, the initiator MT will include its opVO token in the request, so the receiving MT can be sure the call is authorised. MT will obtain its opVO tokens by connecting to the corresponding VO User Agent during the VO login. The User Agent is created by the workflow to represent a particular user to the workflow. The user "connects" to the User Agent from a particular MT (authentication, selection of opVO, authorisation, etc). At this point, the MT is given its opVO token.

The best aspect of this approach is that the SIP network entities do not need to be secured from the opVO security model perspective - being outside the opVO firewall they are not a security threat (other than from a denial of service). Of course, self protection for the SIP broker and the SIP Server would be beneficial, and these could use the same mechanism (i. e. the baseVO dynamic perimeter) to ensure they only respond to valid Akogrimo opVOs and MTs.

7.4.5.1.1. opVO's security model in Akogrimo

Essentially, any user/service participating in an opVO will be given an opVO token to prove their membership of that opVO instance. In this way, the user/service should only accept instructions/data from an entity that also possesses that token. This token can be used to provide end-to-end security for the communications.

The VO security mechanism operates as follows: all participants in a opVO are given membership tokens signed by the opVO and containing their identity. An opVO member's token will contain the following information:

OpVOManager's Token	BaseVO name BaseVO public key opVO name OpVOManager's public key
OpVO Token	OpVOManager's token member's name member's public key member's role

Table 12 – opVO member's token content

Each participant only responds to messages that contain such a token. The message content can be protected using the token and message sequencing mechanisms. Thus the opVO members communications within this opVO are secured against external attach. This basic mechanism can be extended to provide security against insider attacks by enforcing role-based authorisation, or even checking with an authorisation engine (which would have to be in the opVO).

For example, we create an eHealth opVO, O1, with doctor D, patient P, and two services S1 and S2. Now all of these are provided with an membership token for O1, containing their name. O1 (or rather its workflow instance) will instruct S1 to invoke S2. S1 checks the message came from O1 and has an O1 membership token (if not, the message is ignored). It then invokes S2, passing its own membership token. S2 checks the message is from S1 and that S1 has a membership token. It then accepts the invocation.

In the context of the SIP with SOAP interactions being considered in Akogrimo (grid triggered SIP calls and SIP session transfers), Mobile Terminals should be able to process tokens, signatures, encryption, message sequencing, etc… However, if that were not the case for a certain component, then the SIP Gateway will to take over the responsibility of operating the "distributed perimeter". On receiving an instruction to connect two Mobile Terminals, it will have to check that the message has been authorised before acting on it, i. e. that both Mobile Terminals are members of the same opVO as the sender. It will have to access the Participant's Registry to check what opVOs the MTs belongs to, so needs to be a member of the same BaseVO as the opVO. It may also have to check with the opVO that this particular communication is authorised (the insider attach protection mentioned above)

In the example, if a service were to be a doctor doctor being used by a patient, the call should only be possible if the doctor is treating the patient, i. e. in the context of an opVO. When the patient gets the call, they should be sure that the caller is their own doctor, because the call is known to belong to the opVO.

The inclusion of this mechanism in the SIP with SOAP security model has been described in section 7.4.5.1.

7.4.5.2. WS interaction path

As described earlier, in principle one attacker could impersonate the identity of the Workflow Manager and make a non-authorised use of the WS exposed by the SIP Broker. Apart from this, the WS communication path can be intercepted and the exchanged messages modified by a non-authorised entity. In order to prevent this, specific WS security mechanisms are envisaged.

WS-Security core specification [WS Sec] defines several mechanisms to provide protection against SOAP attacks. It includes also mechanisms to detect if some message has been manipulated (XML signature) and encryption mechanism as well (XML encryption).

However, the adoption of the opVO security model provides a higher level of protection, because the attacker should include a valid opVO token in the request; in any other case it will be rejected.

7.4.5.3. RMI path

Internal communications between the SOAP interface and the SIP Broker broker engine is made by means of an RMI interface. This means that remote clients could access the classes and resources that the RMI server (the broker engine in our case) makes accessible. So if some malicious entity knows how to invoke the corresponding methods, it could impersonate the identity of the real SIP Broker SOAP interface and could make unauthorised requests.

To prevent these kinds of attacks, we must take advantage on the Java Security Model [Java Sec]. Using this, you can restrict the actions that remote entities can perform. By default, an RMI program does not have a security manager installed, and no restrictions are placed on remotely loaded objects. Anyway, the java.rmi package provides a default security manager implementation that can be used, which requires the specification of a security policy file containing specific permissions regarding which operations are granted, and from which locations (machines, ports) [RMI]. So the best way to proceed is to specify that only requests from the machine in which the SIP Broker SOAP interface is running are allowed.

The current implementation of the SIP Broker enables the configuration of the policy file to be used. No security policies are applied if this file is not specified.

7.5. Security Architecture Overview

The following figure depicts the security additions to the architecture that need to be accomplished in order to realize the defence mechanisms shown above. The assumption of taking the core network as secure is taken in order to give preference to security mechanisms that revolve around innovative aspects of the project rather than standard out-of-the-box security mechanisms.

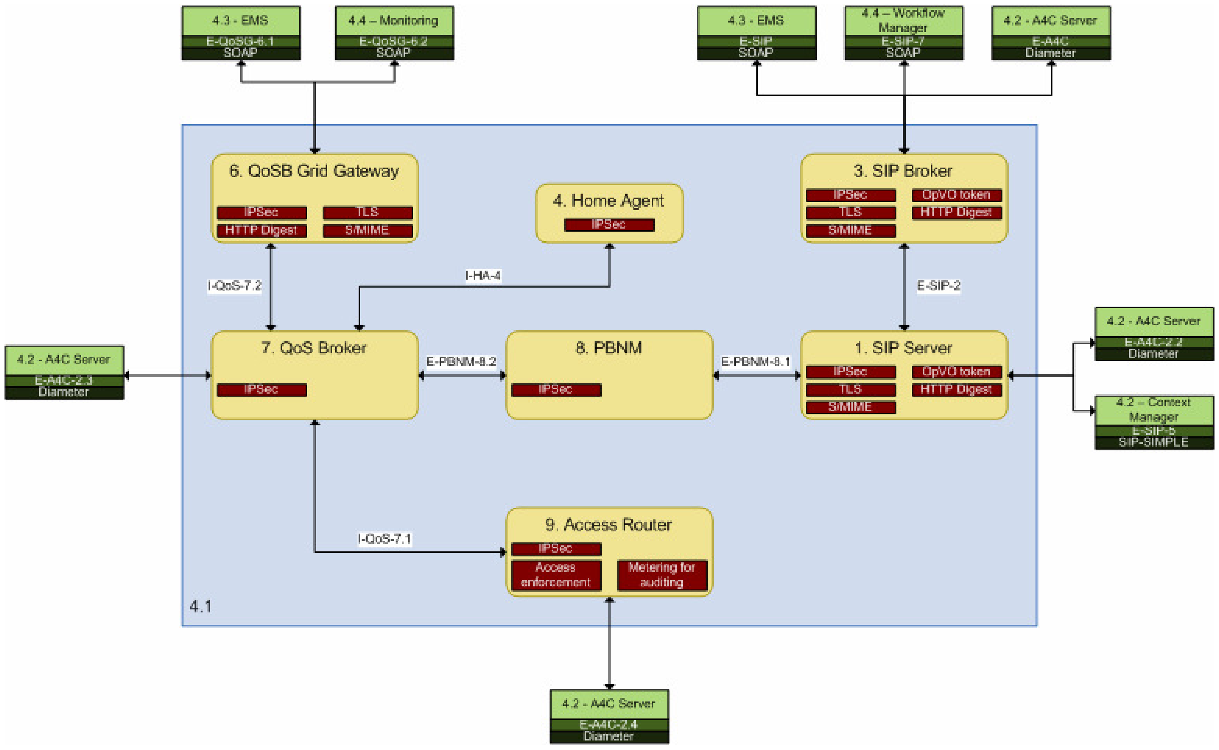


Figure 35 – General architecture

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Annex A. Access security at L3 example

```
Secure wireless access at 3-layer (IPsec). Just access with IPsec is
allowed. This is an example implemented using IPv4 to show the
functionality. The future implementation in IPv6 is analogous.

MN: 192.108.37.99
AR:  192.108.37.97

# apt-get install racoon   # Automatic keying
Choose direct mode in the menu
# modprobe crc32c
# modprobe ipcomp6

/etc/racoon/setkey.conf
-------------------------------
#!/usr/sbin/setkey -f

flush;
spdflush;

# server out to anything wireless: unrestricted
spdadd 192.108.37.97/32 192.168.37.96/29 any -P in none;

# anything wireless into server: unrestricted
spdadd 192.168.37.96/29 192.108.37.97/32 any -P out none;

# anything else going to laptop must tunnel from server to laptop
spdadd 0.0.0.0/0 192.108.37.99/32 any -P in ipsec
        esp/tunnel/192.108.37.97-192.108.37.99/require;

# anything else coming from laptop must tunnel from laptop to server
spdadd 192.108.37.99/32 0.0.0.0/0 any -P out ipsec
        esp/tunnel/192.108.37.99-192.108.37.97/require;
------------------------------------
In the server is the same but changing all "in and outs"


/etc/racoon.conf
-----------------------------------------------
path pre_shared_key "/etc/racoon/psk.txt";
path certificate "/etc/racoon/certs";
remote anonymous {
        exchange_mode main;
#        generate_policy off;
#        certificate_type x509 "my-cert.pem" "private.pem";
#        my_identifier asn1dn;
#        peers_identifier asn1dn;
        proposal {
               encryption_algorithm 3des;
               hash_algorithm md5;
#              authentication_method rsasig;
               authentication_method pre_shared_key;
               dh_group modp1024;
        }
}
sainfo anonymous {
        pfs_group modp768;
        encryption_algorithm 3des;
        authentication_algorithm hmac_md5;
        compression_algorithm deflate;
}
---------------------------------------

/etc/racoon/psk.txt
# IPv4/v6 addresses
192.108.37.97   I did it my way
#Add here the rest of the nodes in the wireless link if you want to
communicate with them...

#172.16.1.133   0x12345678
#194.100.55.1   whatcertificatereally
#3ffe:501:410:ffff:200:86ff:fe05:80fa   mekmitasdigoat
#3ffe:501:410:ffff:210:4bff:fea2:8baa   mekmitasdigoat
# USER_FQDN
#foo@kame.net   mekmitasdigoat
# FQDN
#foo.kame.net   hoge
-----------------------------

On both machines:
# setkey -f /etc/racoon/setkey.conf
# racoon -F -f /etc/racoon/racoon.conf

----------------------------------

Filter non authorized access to the network i.e. everything that is
not a esp tunnel.
(AR)
#!/bin/bash

EXT_IF="wlan0"
INT_IF="eth0"
IPTABLES="/sbin/iptables"

$IPTABLES -F
$IPTABLES -t nat -F
$IPTABLES -t mangle -F

$IPTABLES -P INPUT DROP
$IPTABLES -P FORWARD DROP
$IPTABLES -P OUTPUT ACCEPT

# Mark VPN packets
$IPTABLES -t mangle -A PREROUTING -i $EXT_IF -p esp -j MARK --set-mark
1 #VPN

$IPTABLES -A INPUT -m state --state RELATED,ESTABLISHED -j ACCEPT
$IPTABLES -A INPUT -i $INT_IF -j ACCEPT
$IPTABLES -A INPUT -p icmp -j ACCEPT
$IPTABLES -A INPUT -p tcp -m tcp --dport 22 -j ACCEPT #SSH
$IPTABLES -A INPUT -i $EXT_IF -p udp -m udp --dport 500 -j ACCEPT #VPN
$IPTABLES -A INPUT -i $EXT_IF -m mark --mark 1 -j ACCEPT

$IPTABLES -A FORWARD -m state --state RELATED,ESTABLISHED -j ACCEPT
$IPTABLES -A FORWARD -i $EXT_IF -m mark --mark 1 -j ACCEPT
$IPTABLES -A FORWARD -i $INT_IF -j ACCEPT
```

Annex B. SER (SIP Express Router) configuration file (ser.cfg)

```
#
# $Id: ser.cfg,v 1.4 2006/03/08 12:22:25 jebl231 Exp $
#
# simple quick-start config script
#

# ----------- global configuration parameters ------------------------

#debug=         # debug level (cmd line: -dddddddddd)
#fork=yes
#log_stderror=no       # (cmd line: -E)

# Uncomment this line to enter debugging mode
debug = 3

# If log_stderror is set to 'yes', traces are sent to the console; if set
to 'no', traces are sent
# to the corresponding trace file in folder /temp
log_stderror = yes

# Uncomment these lines for multithreaded
fork = yes
children = 3

# Here goes the real IP of the machine hosting the SIP server
listen = 192.108.37.72
listen = 2001:638:202:e00:213:d4ff:fe36:dab1
port = 5060

check_via=no    # (cmd. line: -v)
dns=no           # (cmd. line: -r)
rev_dns=no      # (cmd. line: -R)

fifo="/tmp/ser_fifo"
alias = "akogrimo.org"
alias = "test.akogrimo.org"

# ------------------ module loading ----------------------------------
# Uncomment this if you want to use SQL database
#loadmodule "/usr/local/lib/ser/modules/mysql.so"

loadmodule "/usr/lib/ser/modules/sl.so"
loadmodule "/usr/lib/ser/modules/exec.so"
loadmodule "/usr/lib/ser/modules/tm.so"
loadmodule "/usr/lib/ser/modules/rr.so"
loadmodule "/usr/lib/ser/modules/maxfwd.so"
loadmodule "/usr/lib/ser/modules/usrloc.so"
loadmodule "/usr/lib/ser/modules/registrar.so"
loadmodule "/usr/lib/ser/modules/esc.so"
loadmodule "/usr/lib/ser/modules/a4c.so"

# Uncomment this if you want digest authentication
# mysql.so must be loaded !
#loadmodule "/usr/lib/ser/modules/auth.so"
```

```
#loadmodule "/usr/lib/ser/modules/auth_db.so"

# ----------------- setting module-specific parameters ---------------

modparam("registrar","default_expires",86400)
# -- esc params (presence agent module) --
modparam("esc", "trace_level",   2)
modparam("esc", "check_expires_timer_interval",   600)
modparam("esc", "max_subscription_expires_value",   3600)
modparam("esc", "max_publish_expires_value",   3600)
modparam("esc", "auth_mode",   0)
#modparam("esc", "single_authorised_subscriber_uri",
"contextmanager@akogrimo.org")

# -- a4c params --
modparam("a4c", "trace_level",   2)
modparam("a4c", "AC4_token_hf", "X-A4C_Token")
modparam("a4c", "A4C_client_command",   "/usr/lib/ser/modules/A4CIFmodule")

# -- usrloc params --
modparam("usrloc", "db_mode",   0)

# Uncomment this if you want to use SQL database
# for persistent storage and comment the previous line
#modparam("usrloc", "db_mode", 2)

# -- auth params --
# Uncomment if you are using auth module
#
#modparam("auth_db", "calculate_ha1", yes)
#
# If you set "calculate_ha1" parameter to yes (which true in this config),
# uncomment also the following parameter)
#
#modparam("auth_db", "password_column", "password")

# -- rr params --
# add value to ;lr param to make some broken UAs happy
modparam("rr", "enable_full_lr", 1)

# -------------------------  request routing logic -------------------

# main routing logic

route
{

        # initial sanity checks -- messages with
        # max_forwards==0, or excessively long requests
        if (!mf_process_maxfwd_header("10"))
        {
                sl_send_reply("483","Too Many Hops");
                break;
        };
        if ( msg:len > max_len )
        {
                sl_send_reply("513", "Message too big");
                break;
        };

        # we record-route all messages -- to make sure that
        # subsequent messages will go through our proxy; that's
        # particularly good if upstream and downstream entities
        # use different transport protocol
        record_route();
        # loose-route processing
        if (loose_route())
        {
                t_relay();
                break;
        };

        # if the request is for other domain use UsrLoc
        # (in case, it does not work, use the following command
        # with proper names and addresses in it)
        if (uri==myself)
        {

                if (method=="REGISTER")
                {

                        # Uncomment this if you want to use digest
                        # authentication
                        /*
                        if (!www_authorize("hi.inet", "subscriber")) {
                                www_challenge("hi.inet", "0");
                                break;
                        };
                        */
                        # Uncoment this if A4C authorisation will not be
                        # used and comment lines below
                        /*
                        log(1, "REGISTER transaction\n");
                        save("location");
                        break;
                        */
                        # Uncoment this to use A4C authorisation and comment
                        # lines above
                        if (aa_request())
                        {
                                log(1, "REGISTER transaction\n");
                                save("location");
                        }
                        break;


                };

                # Presence agent behaviour
                if (method=="SUBSCRIBE")
                {
                        if (!t_newtran())
                        {
                                log(1, "newtran error\n");
                                sl_reply_error();
                        };
                        handle_subscription();
                        break;
                };
                # Inform about not registered destinations
                if (!lookup("location"))
                {
                        log(1, "user not found\n");
                        sl_send_reply("404", "Not Found");
```

```
                                break;
                        };
                        if (method=="PUBLISH")
                        {
                                # The following piece of code creates a new
                                # transaction.
                                # MANDATORY IF THE MODULE PROCESSING THIS METHOD
                                if (!t_newtran())
                                {
                                        log(1, "newtran error\n");
                                        sl_reply_error();
                                };
                                handle_publish();

                                break;
                        };
                }
                # Proxy behaviour (forward to current URI using stateful
                # forwarding)
                log(1, "Proxy transaction\n");
                if (!t_relay())
                {
                        log(1, "newtran error\n");
                        sl_reply_error();
                };

        }
```

Annex C. Network User Profile

This is a skeleton XML file of the Akogrimo Network User Profile.

```
<?xml version="1.0" encoding="UTF-8" ?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema"
elementFormDefault="qualified" attributeFormDefault="unqualified">
  <xs:element name="ANUP">
    <xs:annotation>
      <xs:documentation>Akogrimo Network User Profile</xs:documentation>
    </xs:annotation>
    <xs:complexType>
      <xs:sequence>
        <xs:element name="UID" type="xs:string">
        </xs:element>
        <xs:element name="PRIO" type="xs:short">
        </xs:element>
        <xs:element name="MAC" type="xs:string">
        </xs:element>
        <xs:element name="DSCP" maxOccurs="unbounded">
          <xs:complexType>
            <xs:sequence>
              <xs:element name="ID" type="xs:unsignedInt">
              </xs:element>
              <xs:element name="Macro">
                <xs:complexType>
                  <xs:sequence>
                    <xs:element name="MaxUpBW" type="xs:unsignedLong">
                    </xs:element>
                    <xs:element name="MaxDownBW" type="xs:unsignedLong">
                    </xs:element>
                  </xs:sequence>
                </xs:complexType>
              </xs:element>
              <xs:element name="Layout" maxOccurs="unbounded">
                <xs:complexType>
                  <xs:sequence>
                    <xs:element name="Filter">
                      <xs:complexType>
                        <xs:sequence>
                          <xs:element name="FromAddr" type="xs:string">
                          </xs:element>
                          <xs:element name="ToAddr" type="xs:string">
                          </xs:element>
                          <xs:element name="FromPort" type="xs:unsignedInt">
                          </xs:element>
                          <xs:element name="ToPort" type="xs:unsignedInt">
                          </xs:element>
                          <xs:element name="Protocol" type="xs:string">
                          </xs:element>
                        </xs:sequence>
                      </xs:complexType>
                    </xs:element>
                    <xs:element name="QoSBundle" type="xs:unsignedLong">
                    </xs:element>
                    <xs:element name="PeriodOfDay" maxOccurs="unbounded">
                      <xs:complexType>
                        <xs:sequence>
```

```
                          <xs:element name="Start" type="xs:string">
                          </xs:element>
                          <xs:element name="End" type="xs:string">
                          </xs:element>
                        </xs:sequence>
                      </xs:complexType>
                    </xs:element>
                  </xs:sequence>
                </xs:complexType>
              </xs:element>
            </xs:sequence>
          </xs:complexType>
        </xs:element>
      </xs:sequence>
    </xs:complexType>
  </xs:element>
</xs:schema>
```

Annex D. gSDP formats

This annex describes the different formats being used for the services SIPification framework

D.1. gSDP session

The gSDP sessions are described using an xml document. The related xml schema is:

```
<?xml version="1.0" encoding="UTF-8" ?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema">
<!-- definition of simple types -->
<xs:simpleType name="avstatus">
      <xs:restriction base="xs:NMTOKEN">
            <xs:enumeration value="AVAILABLE"/>
            <xs:enumeration value="UNAVAILABLE"/>
      </xs:restriction>
</xs:simpleType>
<!-- definition of attributes -->
<xs:attribute name="id" type="xs:string" use="required"/>
<xs:attribute name="reference" type="xs:string" use="required"/>
<xs:attribute name="availability" type="avstatus"/>
<!-- definition of complex types -->
<xs:element name="service">
      <complexType>
            <xs:attribute ref="id"/>
            <xs:attribute ref="reference"/>
            <xs:attribute ref="availability"/>
      </complexType>
</xs:element>
<xs:element name="services">
      <complexType>
            <sequence>
                  <xs:element ref="service" maxOccurs="unbounded"/>
            </sequence>
      </complexType>
</xs:element>
</xs:schema>
```

D.2. gSDP notification

The content of the gSDP notifications are also based on xml. Its related xml schema is:

```
<?xml version="1.0" encoding="UTF-8" ?>
<xs:schema xmlns:xs="http://www.w3.org/2001/XMLSchema">
<!-- definition of simple types -->
<xs:simpleType name="avstatus">
      <xs:restriction base="xs:NMTOKEN">
            <xs:enumeration value="AVAILABLE"/>
            <xs:enumeration value="UNAVAILABLE"/>
      </xs:restriction>
</xs:simpleType>
<!-- definition of attributes -->
<xs:attribute name="mobileuser" type="xs:string" use="required"/>
<xs:attribute name="service" type="xs:string" use="required"/>
<xs:attribute name="status" type="avstatus" use="required"/>
<xs:element name="message">
      <complexType>
            <xs:attribute ref="mobileuser/>
            <xs:attribute ref="service/>
            <xs:attribute ref="status/>
      </complexType>
</xs:element>
</xs:schema>
```

Annex E. SIP security mechanisms

From the threats described above, the fundamental security services required for the SIP protocol are: preserving the confidentiality and integrity of messaging, preventing replay attacks or message spoofing, providing for the authentication and privacy of the participants in a session, and preventing denial-of-service attacks. Bodies within SIP messages separately require the security services of confidentiality, integrity, and authentication.

Rather than defining new security mechanisms specific to SIP, SIP reuses wherever possible existing security models derived from the HTTP and SMTP space.

Full encryption of messages provides the best means to preserve the confidentiality of signalling - it can also guarantee that messages are not modified by any malicious intermediaries. However, SIP requests and responses cannot be naively encrypted end-to-end in their entirety because message fields such as the "Request-URI", "Route", and "Via" need to be visible to proxies in most network architectures so that SIP requests are routed correctly. For instance the Proxy Servers need to modify some features of messages as well, such as adding values to "Via" header

To this purpose, low-layer security mechanisms for SIP are recommended, which encrypt the entire SIP requests or responses on the wire on a hop-by-hop basis, and that allow endpoints to verify the identity of proxy servers to whom they send requests.

SIP entities also have a need to identify one another in a secure fashion A cryptographic authentication mechanism is provided in SIP to address this requirement.

An independent security mechanism for SIP message bodies supplies an alternative means of end-to-end mutual authentication, as well as providing a limit on the degree to which user agents must trust intermediaries.

E.1. Transport and Network Layer Security

Transport or network layer security encrypts signalling traffic, guaranteeing message confidentiality and integrity.

Oftentimes, certificates are used in the establishment of lower-layer security, and these certificates can also be used to provide a means of authentication in many architectures.

Two popular alternatives for providing security at the transport and network layer are, respectively, TLS [RFC2246] and IPSec [RFC2401].

E.1.1. IPSec

IPSec is a set of network-layer protocol tools that collectively can be used as a secure replacement for traditional IP. IPSec is most commonly used in architectures in which a set of hosts or administrative domains have an existing trust relationship with one another. IPSec is usually implemented at the operating system level in a host, or on a security gateway that provides confidentiality and integrity for all traffic it receives from a particular interface (as in a VPN architecture). It can also be used on a hop-by-hop basis.

In many architectures IPSec does not require integration with SIP applications; It is perhaps best suited to deployments in which adding security directly to SIP hosts would be arduous. UAs that have a pre-shared keying relationship with their first-hop proxy server are also good candidates to use IPSec. Any deployment of IPSec for SIP would require an IPSec profile describing the protocol tools that would be required to secure SIP.

The IP security (IPsec) provides confidentiality, integrity, data origin authentication services as well as traffic analysis protection.

IPsec was introduced to provide security services such as:

- Encrypting traffic (So it can not be read in its transmission)
- Integrity validation (Ensuring traffic has not been modified along its path)
- Authenticating the Peers (Both ends are sure they are communicating with a trusted entity the traffic is intended for)
- Anti-Replay (Protect against session replay)

For more detail see related documentation [RFC2401]

E.1.2. TLS

TLS provides transport-layer security over connection-oriented protocols, for the purposes of this document, so signifying TLS over TCP. The primary goal of the TLS Protocol is to provide privacy and data integrity between two communicating applications

TLS is most suited to architectures in which hop-by-hop security is required between hosts with no pre-existing trust association. For example, Alice trusts her local Proxy Server, which after a certificate exchange decides to trust Bob's local Proxy Server, which Bob trusts, hence Bob and Alice can communicate securely.

TLS must be tightly coupled with a SIP application. Note that transport mechanisms are specified on a hop-by-hop basis in SIP, thus a UA that sends requests over TLS to a proxy server has no assurance that TLS will be used end-to-end.

SSL/TLS has many of the advantages of IPsec and the successful introduction of the protocol in the wired Internet has proved its usability and effectiveness. Likewise, SSL/TLS runs above TCP/IP and below higher-level protocols such as HTTP or FTP and consequently the TCP header is not encrypted.

In contrast to IPsec, TLS does not assume any trust relation among communication parties, so, it is more suitable for Inter Domain Authentication.

SSL/TLS provides protection against several known attacks (including man in the middle attacks), prevents the eavesdropping, the tampering, and message forgery. However is susceptible to a number of denial-of-service (DoS) attacks. In particular, an attacker who initiates a large number of TCP connections can cause a server to consume large amounts of CPU doing RSA decryption.

Because TLS runs over TCP, it is also susceptible to a number of denial-of-service attacks on individual connections. In particular, attackers can forge RSTs, thereby terminating connections, or forge partial TLS records, thereby causing the connection to stall. These attacks cannot in general be defended against by a TCP-using protocol. Implementors or users who are concerned with this class of attack should use IPsec.

On the other hand, the biggest drawback with SSL/TLS is does not run over UDP. In addition, keeping up many TCP connections simultaneously may be too heavy for Proxy Servers.

E.2. SIP URI Scheme

The use of a SIPS URI of the form "sips:alice@akogrimo.org " an INVITE message requires that TLS should be used on the whole path to the destination. Since each hop may add route information to the SIP message header, TLS protection must be realized on a hop-by-hop basis on each segment of the path. The use of TLS requires the use of TCP as a transport protocol (TCP/SIP) and necessitates a public key infrastructure.

E.3. HTTP Digest

SIP provides a challenge capability, based on HTTP authentication that relies on the 401 and 407 response codes and header fields for carrying challenges and credentials as well. Without significant modification, the reuse of the HTTP Digest authentication [RFC 2617] scheme in SIP allows for replay protection and one-way authentication.

In the SIP context, the "server" entity authenticating the "client" entity can be a proxy, registrar, a redirect server or another user agent. The basic mechanism is similar in all cases: both client and server have a shared secret (password), which must be known in advance (e.g. using out of bands mechanisms). When the server receives a request, it challenges the client to provide valid credentials; then the client provides a valid name and demonstrates it knows the password.

One of the advantages of the HTTP Digest mechanism is that the password does not have to be sent in clear text, like in HTTP Basic Authentication. For this purpose the Digest mechanism uses hash algorithms like MD5 or SHA1 (a hash function is a one way functions that provide a fixed length output) and nonces (an arbitrary and random value which is used only once).

Next figure describe how this mechanism works. Note that the name of the header to be include and the challenge number depends on the nature of the server entity: Registrar Servers and UAS use the "WWW-Authenticate" and "Authorisation" header, and the 401 reply, while proxies use the "Proxy-Authenticate",and "Proxy-Authorisation" headers, and 407 reply.

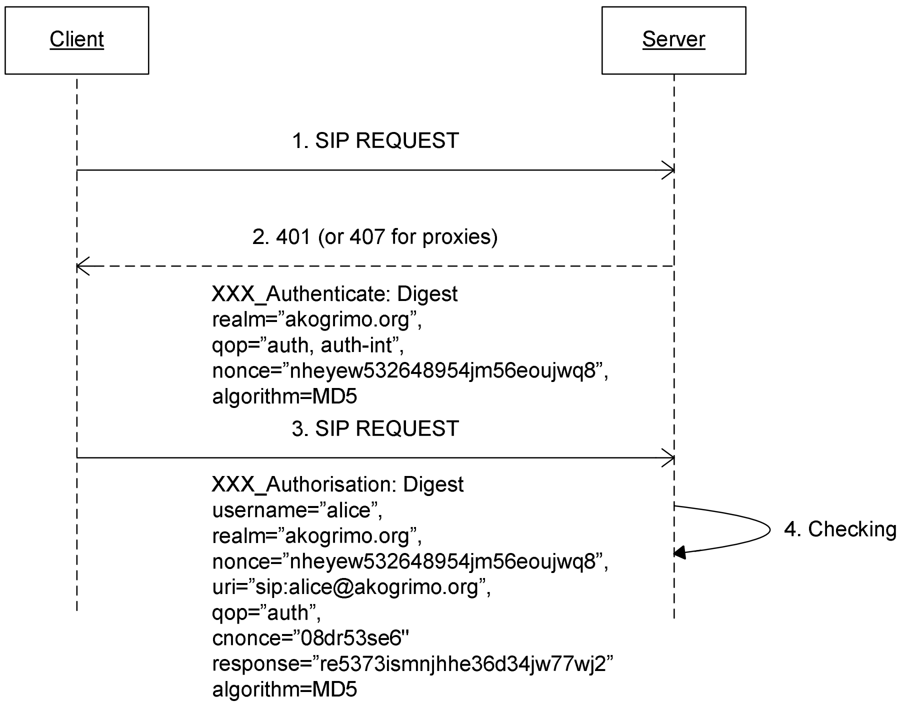

Figure 36 – HTTP Digest security mechanism.

Client sends a SIP request to the Server (REGISTER, INVITE…)

Server challenges the request with a 401 (407) reply, including a "XXX_Authenticate" header field with several parameters:

- realm: the domain of which the share secret (password) is required.

- Qop (quality of protection): indicates if integrity protection applies for either the request line (auth) or both request line and message body (auth-int).
- nonce, algorithm: hash function and random nonce to be used in the client side for credential presentation (will be also used in the server side to check the reply)

The client replies with a new request with a "XXX_Authorisation" header field containing several parameters. The most important is response, which contains a hash comprising the username, password, the server's nonce, the client's nonce (specified in the cnonce parameter) and the request line (also the body if auth-int was chosen).

The server calculates another hash value using the same input. If the result matches with the response parameter of step 3, the request is processed; otherwise, the request will be challenge again.

The security properties of this mechanism are the following:

- The usage of random nonces prevents again reply attacks (if some malicious entity obtains the correct has value for a particular nonce, it will not be able to access the server because a different one will be used in a different challenge).
- Provides user authentication.
- Provides limited integrity protection (i.e. the request line – method and URI – and the body, if the proper qop parameter is selected). But considering that not all headers fields are protected, this schema is sensible to MitM attacks.
- DoS attack protection (during the challenge period, the server remain stateless, so an attacker can not block resources for a long time).

This strategy is commonly used in UAC because typically they will not implement advanced and secure certificate-based security mechanism. These site certificates are commonly used for network entities.

The HTTP Digest authentication drawbacks are as follows:

- Does not support message Integrity and Confidentiality.

E.4. S/MIME

S/MIME can also be used to ensure both integrity and confidentiality of SIP messages. It basically consists on a certificate-based mechanism than can be used to sign or encrypt some parts of SIP messages, typically SIP bodies. Encryption of an entire SIP message to provide end-to-end confidentiality is not appropriate because some network entities need to view certain headers (as shows Table 13) to properly route the message; however, S/MIME can provide some kind of protection for headers fields, using a mechanism named "SIP message tunnelling", which basically consists on the encapsulation of a SIP message as the body of an "outer" SIP message. This body, containing the whole message, is then encrypted using S/MIME.

Header	**Request URI**	**From**	**To**	**Via**	**Record Route**	**Record**	**Call Id**	**Cseq**
Modification Allow	YES	NO	NO	YES	YES	YES	NO	NO

Table 13 – Encrypted headers with S/MIME

As mentioned earlier, S/MIME is a certificate-based mechanism. These certificates are associated with the keys that are used to sign or encrypt message bodies. Currently there is no a consolidated authority today that provides certificates for end user applications, so most usual solutions are:

- Usage of pre-configured certificates previously trusted.
- Usage of self-signed certificates (that is, a certificated that cannot be verified by the recipient), which are also sent in the requests as a multipart MIME body (different sets of data combined in a single body, as defined in RFC 2046 [RFC2046]). In this case, the recipient end user application is responsible for accepting or rejecting the request, based on the information the certificate carries.

Each user agent supporting S/MIME must have a key ring for end-user certificates in order to map between AoRs and their corresponding certificates. Anyway, SIP can also be used as a mean to distribute these certificates. The S/MIME specification enable the inclusion of the certificate itself together with the proper content to be protected as part of a multipart MIME body, so when a new request including a certificate arrives, the user agent can validate it and add the new certificate to its local database.

The security that this process provides relies on the usage of a random key which is sent together with the protected content. This random key is encrypted using the public key of the recipient, so when receiving a message, it will make use of its private key to a) decrypt the content encryption key and b) use this decrypted key to decrypt the message itself. This process is depicted in the figure below:

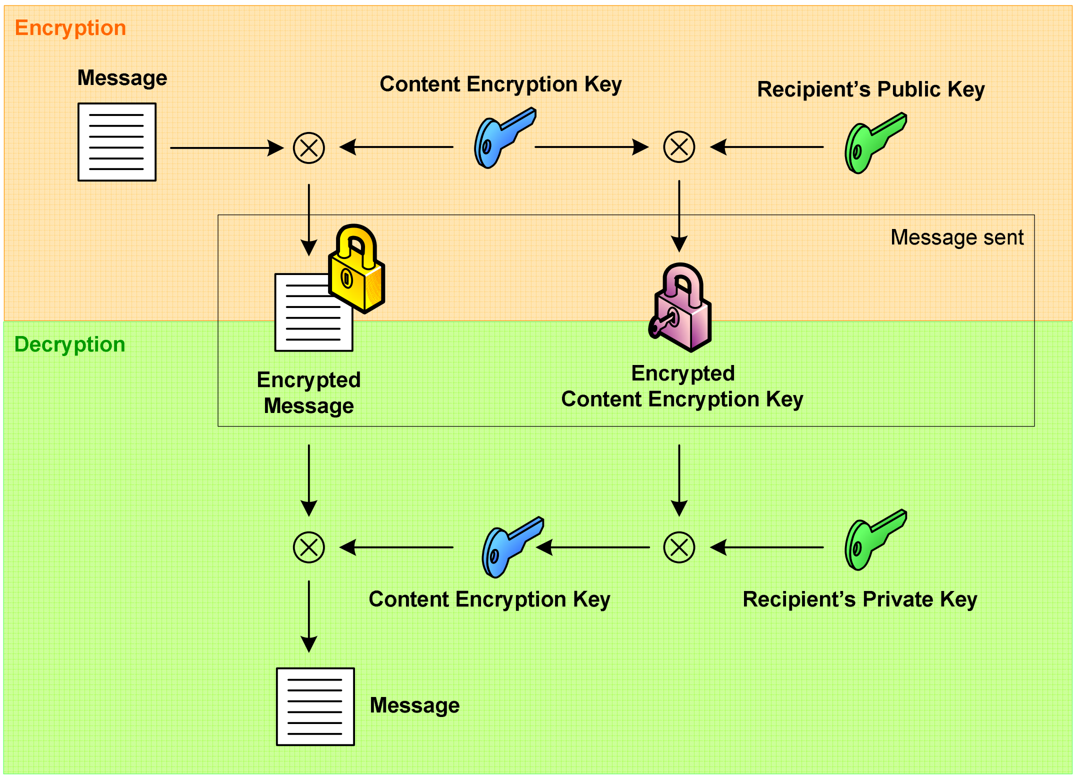

Figure 37 – S/MIME encryption and decryption

The usages of a symmetric key (the same key will be used in both sides) have a significant impact on the speed of the encryption/decryption process. Obviously, senders must have foreknowledge of the public key of recipients, so mechanisms to access this information must be implemented.

In order to provide sender's authentication, messages can be also signed. However, in this case the sender's private key is used, as next figure describes. Again, the recipient needs to know in advance the public key of the opposite side, in order to compare the decrypted message hash it has received with the calculated one by itself; but as mentioned before, senders may include it with the message as appropriate.

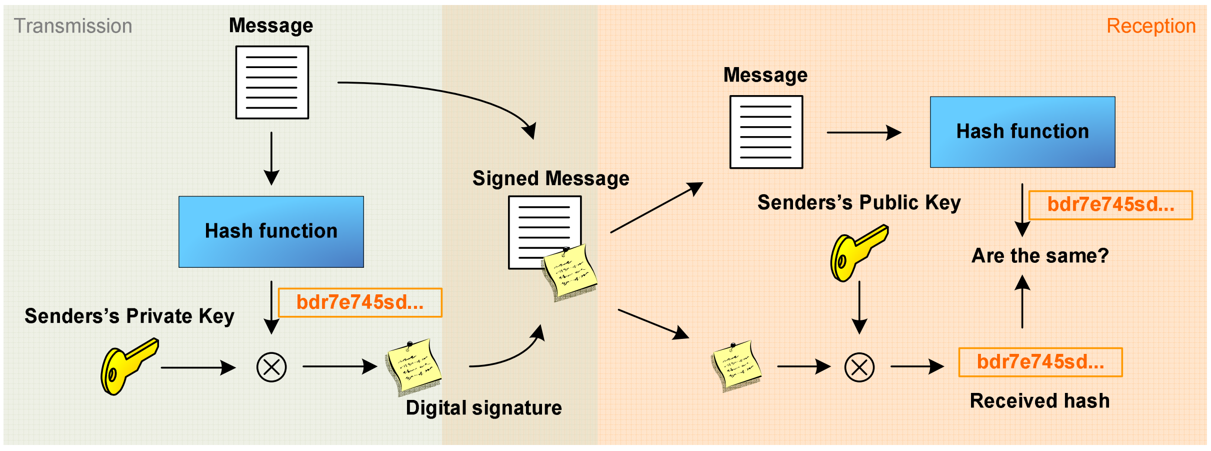

Figure 38 – S/MIME signature

Finally, just to mention that both processes can be nested: encrypted messages can be signed, and signed messages can be encrypted.

S/MIME provides the following cryptographic security services for electronic messaging applications:

- Authentication, message Integrity and non-repudiation of origin (using digital signatures), and Data Confidentiality (using encryption).
- Since S/MIME is mostly used for securing message bodies, such us the SDP rather than message headers, S/MIME avoids tampering with message bodies' attacks.

It is worth noting as drawback that S/MIME (the same than IPSec) generates considerable overhead in SIP messages, and the lack of PKI is an additional restriction for the operation of S/MIME in SIP.