PTT Client in an IMS Environment

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Abstract - The IP Multimedia Subsystem (IMS) is a horizontal architecture framework that allows telecommunication operators to offer internet services with quality of service to their clients applying different charging methods. In this case, the Push-To-Talk (PTT) service is expected to be one of the first services based on IMS, available on a large amount of operators, for its reduced level of requirements, as well as its tolerance to delays and bandwidth of connections.

My assignment consisted on the development of a PTT client that runs in an IMS environment. The work started by defining the architecture of the PTT system, respecting as deep as possible the IMS principles established by the Third Generation Partnership Project (3GPP). Similarly, the Open Mobile Alliance (OMA) recommendations on the Push-To-Talk Over Cellular (PoC) system definitions were adopted for the development of the PTT client.

The PTT client consists of an application with a graphic mode, developed with Java for desktop. Several tests were carried out with the PTT system, performed in a high-bandwidth LAN connection and in a Code Division Multiple Access 2000 (CDMA2000) network available from Zapp. According to the performance parameters established by OMA it was proved that the developed PTT system can be perfectly used in both types of access networks.

1 Introduction

The Internet has spread out all over the world. It is estimated that 17.8% of world population has access to the Internet [1]. The mobile world is also in a growing expansion process over the world. Following the example of Portugal, which has approximately 10.6 million inhabitants, in the second quarter of 2007 [2], there existed more than 12 million active mobile subscribers [3].

The IMS allows telecommunication operators to offer to mobile network users all internet services worldwide, with high level of quality of service, applying different ways of charging. This will make the integration of the various services easier, regardless of the user’s access network.

The PTT service will be certainly one of the first IMS-based services supplied by a diversity of operators, because it doesn’t need high-bandwidth and low-delay links. It is according to these features that this work was developed. It includes the development of the Push-To-Talk client application in a mobile IMS Environment.

2 State of the Art

2.1 IMS

2.1.1 Definition

The IP Multimedia Subsystem is the key element in the 3G architecture, which allows access to all web services, through a mobile phone. With the IMS, operators can provide access to a wider spectrum of services, different ways of charging and Quality of Service (QoS), [4].

In mobile services the quality of service has its advantages for real time services, for example, if in a VoIP call the delay of the information in the network is over 300 milliseconds, the conversation will be very annoying.

The IMS allows operators to know which is the service being used by a certain user. This will bring to operators the capability to differentiate rate charges and adapt payment to the type of traffic, and not the quantity of transferred information.

Another reason for the existence of IMS is the integration of different services. Operators don’t need to develop all the services given to their clients, due to the standardized interfaces offered by the IMS.

The IMS allows the association between the mobile networks with the Internet. The best of the Internet is the utility of the services given to the user. The best of the mobile networks is the ubiquity of its access. The 3GPP associated the best of the two universes (Image 1), by designing the architecture known as IMS.

Image 1

2.1.2 IMS Architecture

The IMS Architecture (Image 2) is a horizontal architecture composed by three layers: transport, control and application level. The 3GPP standardized the functions of each IMS component, as well as the connection between them. The main components of this
architecture are: databases, called HSS (Home Subscriber Server) and SLF (Subscriber Location Functions); SIP servers known as CSCFs (Call/Session Control Functions); ASs (Application Servers); MRF (Media Resource Functions); one or more BGFs (Breakout Gateway Control Function); and PSTN gateways (Public Switched Telephone Network).

Image 2

2.1.3 Main Protocols
The Session Initiation Protocol (SIP), Diameter, Common Open Policy Service (COPS), Real-Time Transport Protocol (RTP) and RTP Control Protocol (RTCP), are some of the protocols chosen for the IMS. The SIP defined in the RFC 3261 is the signalling protocol of the IMS. This was specified by the Internet Engineering Task Force (IETF) as a protocol to establish and manage multimedia sessions in IP networks. It is a protocol based on text, making its extension easier, as well as the debug process. The Diameter was chosen to create the AAA protocol (Authentication, Authorization and Accounting) of the IMS. The COPS protocol (specified in RFC 2748) is used to transfer policies, as a way to control and order restrictions of the users behaviour. The RTP and RTCP specified in the RFC 3550 are used to stream Audio or Video, in real time sessions, allowing the statistic control of the packets.

Image 3

2.1.4 Main Advantages
The network architectures of current operators are vertical in design, and so, each network has its own control layer and associated services. When these operators decide to introduce new services in the network, they have to create all the signalling and control modules for those services. It is a long and complicated task. The IMS makes the life of operators easier due to its horizontal architecture structured in layers: transport layer, control layer, and application layer, as shown in Image 3.

The Fixed Mobile Convergence (FMC) brings the capabilities of both fixed and mobile services, accessible to terminal devices. Being independent of the access networks, IMS is a low-cost way to accelerate FMC to the operators.

Next Generation Networking (NGN) is a concept that describes some evolutions both for telecommunications networks and for access networks. The main idea is of a network that transports all types of information and services (voice, data, and video, among others) in IP packets, similarly to what is done in the Internet.

Therefore, one might say that IMS is a 3GPP standardized architecture following the principles of the NGN.

The IMS combines the best of the mobile networks and the Internet to provide a series of services that can be used independently of the access network, such as: Push-To-Talk, Push-To-Watch, videoconference, sharing all types of files/information and Instant Messaging.

2.2 PTT
PTT is a service concept based on a technology, the walkie-talkie, with several years of existence. The first walkie-talkies were developed for military usage during the Second World War, as is specified in [5]. The user that intends to talk, presses the button, keeps on holding it and can only talk when a grant signal is received, normally expressed by some “beeps”. When the conversation is over, the user releases the button allowing other users talk in turn. It is a simple service to use, as all that is needed is to press a simple button to start a session; this is a very fast process, much faster than a traditional voice call (there is no need to dial a number). This is a half-duplex service, meaning that only one user at a time has the possibility to talk. Additionally
is allows a user to talk simultaneously to many others (group call).

The PoC is a PTT service provided by a mobile network, specified by the OMA as an IMS service designed for interoperability between other PoC services in different network operators. In this document PoC and PTT are treated as synonyms, due to the usage of OMA specifications. However, PoC is not restricted for usage only in cellular/mobile networks but also on fixed networks and the Internet, provided adequate “terminals” are available (typically “soft terminals”).

PTT is probably the first IMS-based service available on a large amount of operators, not being a demanding service in what respects bandwidth and connection delays [4]. For example, some network connections are inappropriate for services such as voice or video calls, but can be appropriate for PTT.

In PTT, signalling and control sessions are based on SIP, and the voice traffic is transmitted with RTP/RTCP. Each talking client sends RTP/RTCP packets to the PTT server, and the server distributes the packets to the other session members.

2.2.1 PTT Performance Requirements

To offer useful services to clients, operators identify some factors that have impact in the users’ Quality of Experience (QoE). QoE is the term used to describe the users’ perception about the system and usability that users perceive from the system. In the Push-To-Talk service some factors have impact in the QoE: QoE1 (time to establish a PTT Session, as well as the reception of grant to talk), QoE2 (The time between pressing the PTT button and receiving the grant to talk), QoE3 (audio connection delay), QoE4 (voice quality). Other performance requirement is Turnaround Time (TaT). These performance requirements are specified by the OMA in [6].

2.2.1.1 QoE1

The established PTT session time and the reception of grant to talk should be less than two seconds.

2.2.1.2 QoE2

In a PTT session from one to one or from one to multiple users, the time between pressing the PTT button and receiving grant to talk should be less than 1.6 seconds.

2.2.1.3 QoE3

The audio delay between clients is the difference between the time the receiver starts listening the caller and the time that the caller starts speaking. This delay shouldn’t be over 1.6 seconds in a PTT Session. This delay can reach 4 seconds in the beginning of the conversation.

2.2.1.4 QoE4

The voice quality requirement is established, when the PTT system obtains a Mean Opinion Score (MOS) equal or over 3 and a Bit Error Rate (BER) under or equal 2 percent. The MOS is a numeric value than qualifies the received media quality, in the case of voice, this MOS is a value expressed by a unique number from one to five. In level one, the voice quality is bad and in level five the voice quality equals the original signal, according to the human ear [7]. On the other hand the BER is a percentage of bits with errors in relation to the total number of bits received in a transmission [8].

2.2.1.5 TaT – Turnaround Time

The Turnaround Time is the time elapsed from the moment when the user stops talking and releases the button to the moment when he can hear another PTT user. In the case of other PTT participant responding immediately, (about one or two seconds on average) the TaT should be less than four seconds.

2.2.1.6 Used Protocols

The OMA in the PoC specifications uses essentially five protocols in its interfaces: SIP, RTP, RTCP, Talk Burst Control Protocol (TBCP) and finally XML Configuration Access Protocol (XCAP). TBCP is a RTCP extension suggested by OMA, it uses RTCP APP messages to transmit floor information.

3 Architecture

The Image shows the architecture for the project.

3.1 Architecture Elements

3.1.1 x-CSCF

SER (SIP Express Router) was chosen for the x-CSCF components. The SER’s functions are of outbound/inbound SIP Proxy and forwards requests and responses in correct direction. It can also play the role of SIP Registrar, that is, it answers to the SIP REGISTER requests with a challenge to users. After the reception of the correct challenge response, the x-CSCF binds the user's location (in this case, the IP address) with the SIP address. The x-CSCF also plays the role of SIP Presence Server. It treats the SIP SUBSCRIBE and SIP PUBLISH
requests generating SIP NOTIFY messages whenever they are needed.

3.1.2 PTT Server
The PTT Server is an application developed in Java, and behaves like Back-To-Back User Agent (B2BUA). The Server has the function of controlling the access to the service, because only PTT Subscribers can make PTT Calls. The PTT Server is responsible for saving contacts and each PTT Subscriber's state. It also has the function of distributing audio; in a PTT Session the sender doesn't send audio directly to the other session members, but to a PTT Server that distributes to all receivers. In this way, the PTT Server plays the role of Streaming Server.

The PTT Server generates charging information to Call Detail Records (CDR). The CDR is an information registry about a call or session, for example, the information of who talked in a session should be saved as well as the conversation duration. These CDRs are saved in files and the accounting system processes these files applying the desired rates, to generate the client's bill.

The PTT Server is responsible for the co-ordination of the floor mechanism, deciding who talks and who listens in each PTT Session. The Server communicates with HSS to verify if a certain user is a PTT Subscriber. It is in the HSS that all the information about users and groups is stored. The Server has a shell that allows inserting or deleting subscribers and PTT groups, as well as adding or removing PTT Subscribers in PTT Groups. A certain session is terminated by the PTT Server if no session member speaks in a certain space of time.

3.1.3 PTT Client
This is an application with a graphic mode developed in Java for Desktop. The Application contains a list of contacts, and the user can add, modify or delete them. As a PTT application it can capture audio from the microphone and reproduce audio in the device speakers. It works as SIP User Agent, so it can register in the x-CSCF and send or receive signalling SIP messages. The PTT Subscriber has the possibility to see the PTT groups he belongs to, establish sessions with these, as well as a set of ad-hoc contacts or a single subscriber. With the PTT application, the subscriber can express his state of presence in the x-CSCF.

There is a configuration file that is loaded when the application is launched. This file contains the necessary fields including the IP Address, name, SIP Address and User's keyword, PTT Server's SIP Address, outbound/inbound SIP Proxy Address, among others. As output files, the client generates a file with statistic data where the information about the times that are useful to measure the program's performance are saved. It also keeps a log file to save all the information that can be useful for debug.

3.1.4 HSS
The HSS is a database that is running in Linux with MySQL Database Server. The PTT Server in order to have access to the data sets uses the Hibernate framework. The Hibernate is a framework that allows defining correspondence between the domain model's classes and the data sets of the relational model. It allows loading and writing objects in a persistent way, allowing the database abstraction. It generates SQL queries in an easy way for the user. In order to have access to this framework and to its benefits, it is necessary to define in a XML File the maps domain classes and the database data sets.

As it has been said, the HSS keeps the information about subscribers and PTT Groups. It also saves the information from each user belonging to each group.

3.2 The Architecture and the IMS Principals
The advantages of the IMS Architecture were already presented. However, this architecture isn't integrally IMS. The architecture includes a charging function. This function is supported by the PTT Server that generates this information to a CDR.

The IMS suggests the integration of different services and this item is demonstrated in the architecture due to the use of a Presence Service belonging to SER in an easy and efficient way, trying not to invent what already exists.

PTT Sessions are IP Sessions that have only one type of media: audio. These Sessions could be easily extended to multimedia ones. For this, it is only necessary to change the client application, trying to capture for example, video and authorising the reproduction of this media. This function is easily achieved because the PTT Server has a function of multimedia streaming server, that is, it streams whatever type of media.

The access control to the offered services is the key-point of the IMS, this mechanism is supported by the architecture, because before any user has access to the PTT Service it needs to ask for authorization to the PTT Server. The PTT Server looks for HSS if the user is or not a PTT Subscriber, controlling access to the service.

![Image 5- Tables of HSS](image-url)
3.3 The Architecture and the PoC OMA Requirements

This section presents some of the proposed requirements by OMA that were followed in the definition of the system’s architecture. The developed architecture supports communication from one to one or from one to many. The PTT Server generates the CDRs allowing the application to charge single users or groups. The OMA specifies that PTT Groups can be visible to all PTT Subscribers or to Members of the same Group only. In this project, PTT Groups should only be visible to their members, because of confidentiality. Each PTT Subscriber has its own list of contacts and can save, modify or delete contacts from that list. The client can express its status of presence in the Presence Server as well as On-line, Busy or Off-line Status.

In the end, several tests were executed to evaluate the performance of the solution, verifying its compatibility with the requirements of OMA’s performance requirements.

3.4 Architecture Protocols

Following the OMA and the 3GPP examples in the IMS definition, the architecture uses SIP and RTP/RTCP. The TBCP suggested by OMA for the floor control is achieved with SIP INFO messages and not with RTCP APP messages, following the implementation of the PTM Application of FOKUS [12] that relies also on SIP INFO Messages to transmit floor information. For the contacts, a similar approach was taken, using SIP Messages, instead of XCAP Protocol mentioned by OMA.

4 Implementation

4.1 User State Diagram

The User State Diagram suggested by OMA was used as reference [6]. Image 6 shows this diagram:

![Image 6- User State Diagram](image)

When the Client Application is started, the user is in the initial state, that is, in the state “User”. To interact with network services the user must first register, transiting to the State “Registered in the Network”. The registration in the network starts by sending a request message of type SIP REGISTER from the Client to the x-CSCF. In the “Registered in the Network” state, the user can interact with network services or express his state of presence. To change to state “Registered in the PTT Server” the client must inform the PTT Server of the intention to register. The PTT Server checks if the register request provides from a known PTT Subscriber, that is, if the User’s SIP Address is in the data set SUBSCRIBER of the HSS. Accepted in this state, the client may then invite other clients for a PTT Session or be available for an invitation. Either of these actions forces a transition to the “PTT Participant” State. In the “PTT Participant” State the client application can capture audio, encode and send it to the PTT Server or just play received audio streams. The client returns to the “Registered in the PTT Server” after leaving the session or being forced released from the session. In the “Registered in the PTT Server” State, the client may return to the “Registered in the Network” state by informing the PTT Server that he is no longer available for PTT Sessions. If the client application is in the “Registered in the Network” state, he can return back to the “User” state by informing the x-CSCF that he wants to cancel the register. In this case, the user should send a SIP REGISTER message with the field EXPIRES to 0.

4.2 Contacts

Each PTT Subscriber has its own list of contacts that is kept in the PTT Server. This section presents in detail the interaction between the Subscriber and the PTT Server on this subject:

Image 7

After the client registers in the PTT Server, the Server will reply with the contact list using a SIP INFO BuddyList Message composed by a XML File containing the contacts. In the example above, the SIP INFO BuddyList Message contains the File “contactsIMS1.xml” in the first version. Latter on, the subscriber adds a new contact with the name “ims3” with SIP Address: sip:ims3@ims.core.org. This action originates the sending of a SIP INFO AddBuddy Message (message 3). When receiving this message, the PTT Server sends a SIP 200 OK Message to confirm the reception and updates the contact list of
PTT Subscriber changing "contactsIMS1.xml" file to version two. 
Then, the subscriber decides to remove the contact with the address: sip:ims3@ims.core.org, and this action originate a SIP INFO RemoveBuddy Message (message 5). When the PTT Server receives this messages it changes the file "contactsIMS1.xml" to version 3 and confirms the message reception sending a SIP 200 OK Message.

After this, the subscriber decides to change the name of his friend "ims3" to "JOAO". This action originates the sending of a SIP INFO RemoveBuddy Message (message 7) and after confirmation the sending of the message SIP INFO AddBuddy (message 9) with complete new user information. In this case, with the name "JOAO" and the SIP Address sip:ims3@ims.core.org. This will change the "contactsIMS1.xml" File to version 4.

4.3 Groups
The information about PTT Groups is saved in the PTT Server. PTT Subscribers have access to this information by pressing the button "Import Groups" from their application. This action originates the sending of a SIP INFO message with the field Content-Type: application/GroupsQuery and in the body the SIP Address of the PTT Subscriber, for example sip:nomeUtil@ims.core.org.

When receiving this message, the PTT Server makes a query to the database looking for PTT Groups where this Subscriber is a member. This information reaches the PTT Subscriber by a SIP INFO message with the field Content-Type: application/GroupsResponse and in the body a XML with the following structure:

```xml
<?xml version="1.0" encoding="UTF-8"?>
<Grupos>
  <Grupo nome="work">
    <Utilizador nome="worker1" Endereco="sip:worker1@ims.core.org" StatusPTT="ONLINE"/>
    <Utilizador nome="worker2" Endereco="sip:worker2@ims.core.org" StatusPTT="ONLINE"/>
    <Utilizador nome="ims1" Endereco="sip:ims1@ims.core.org" StatusPTT="ONLINE"/>
  </Grupo>
  <Grupo nome="family">
    <Utilizador nome="mother" Endereco="sip:mother@ims.core.org" StatusPTT="ONLINE"/>
    <Utilizador nome="father" Endereco="sip:father@ims.core.org" StatusPTT="ONLINE"/>
    <Utilizador nome="ims1" Endereco="sip:ims1@ims.core.org" StatusPTT="ONLINE"/>
  </Grupo>
</Grupos>
```

If subscriber "ims1" receives the above XML message, he concludes that he belongs to the groups "work" and "family". The members of the "work" group are "worker1", "worker2" and "ims1". The members of the "family" group are the users: "mother", "father" and "ims1". The field StatusPTT contains information about the User's PTT Status.

Not only the SIP INFO BuddyList Message, but also the SIP INFO GroupsResponse is sent to the client every time he registers in the PTT Servers. The SIP INFO GroupsResponse Message is sent to clients every minute because PTT Groups information can be modified in the PTT Server Shell, consequently, the update time for clients is a maximum of one minute.

4.4 Presence Services
The PTT Client has two Presence Services: Network Presence Service (optional) and PTT Server Presence Service. When using the first one it is the user who changes his status between: On-line, Busy and Off-line. The PTT Server Presence Service is maintained by the PTT Server that keeps the PTT Status of each user and is responsible for its changes: On-line, InSession, Offline, and Na (Not Available).

The inclusion of Presence Services is used not only to inform the users but also to define policies of PTT Session Establishment. When a PTT Client needs to make a PTT Session for one or more users (Ad-hoc), he may only make it if all selected users are in appropriate situation to start a PTT Session. A PTT User is in an appropriate situation if his Network Presence Status and PTT Server Presence Status are On-line or when having a PTT Server Presence Status On-line and the Network Presence Status disabled. If any selected users is in an inappropriate situation, than the PTT Subscriber that is trying to establish the PTT Session will be notified and the session will not be allowed.

On the other hand, if the PTT Client pretends to establish a PTT Session with a group, there has to be at least two users of the group in an appropriate situation to start a PTT Session, otherwise the client will be notified with an error message.
PTT CLIENT in an IMS Environment

Image 8
4.5 Floor Control
The Floor Control or Talk Burst Control (OMA designation) is the protocol that arbitrates which user should talk (take the floor) at any moment. To apply floor control, some SIP INFO Messages were created, such as:
- SIP INFO TBCP Granted
- SIP INFO TBCP Deny
- SIP INFO TBCP Idle
- SIP INFO TBCP Taken
- SIP INFO TBCP Revoke
- SIP INFO TBCP Request
- SIP INFO TBCP Release.

The first five Messages are sent by the PTT Server to a client, while last two (Request and Release) are sent by the PTT Client to the Server.

The OMA suggests that TBCP Connect, Disconnect and TBCP Talk Burst Acknowledgement Messages should be used in pre-established sessions, but as this function wasn’t implemented it wasn’t necessary to apply these messages.

As the queuing mechanism wasn’t implemented to floor requests, it wasn’t necessary to implement TBCP Talk Burst Request Queue Status Request and TBCP Talk Burst Request Queue Status Response Messages.

4.6 Audio
Audio is the most important part in the PTT Client. With no audio all of the other application elements: interface, network registry, PTT registry, TBCP States, Presence, Contact List and PTT Groups would make no sense.

In a PTT Service there are two audio components: send and receive. To treat all audio details, Java Media Framework (JMF) was used. The PTT Server plays the role of streaming server.

4.7 General Interaction Example
In this section, a general example of interaction of the PTT Service between 3 participants is presented. It is a combination of concepts explained earlier in this paper, showing all the messages that are exchanged.

The diagram in Image 8 does not show the messages associated with the Network Presence Service or PTT Server Presence Service for simplicity.

In that diagram may observe the message sequence of a dialog between 3 PTT Terminals, the x-CSCF and the PTT Server. The 3 terminals register in the network (x-CSCF) (messages 1 to 12). After the network register, it is necessary to register in the PTT Server, (messages 13 to 24). When the register in the PTT Server is finished, the server will send information to the terminals about their contact list (messages 25 to 36), and their PTT Groups (messages 37 to 48).

Terminal 1 decides to establish a session with its buddies, terminal 2 and 3 (messages 49 to 78). When terminal 1 receives floor grant to start talking (message 76), it starts sending audio to the PTT Server which will distribute for the remaining group members (in this case, terminal 2 and 3). Terminal 1 sends message 79 informing the PTT Server that it released the PTT Button. It is the PTT Server’s obligation to inform terminal 2 and 3 that terminal 1 isn’t talking and that the channel is free (between messages 87 and 94).

After a certain amount of time, terminal 3 decides to abandon the session, informs the PTT Server of this intention, in message 127, and obtains confirmation in message 130. Terminal 2 also decides to abandon session and as it makes no sense that terminal 1 is alone in session, the PTT Server sends it a SIP BYE message indicating that the session is over (message 135).

The terminals decide to end the registry of the PTT Server and the x-CSCF, (messages 139 to 156), starting by terminal 3, then terminal 2 and ending with terminal 1.

5 Tests
In order to verify the correct operation of the PTT System some application tests were run in an independent mode, and other test interacting with the PTT system. These tests aimed to confirm the system functionality, as well as its performance against the requirements suggested by OMA, such as, QoE1, QoE2, QoE3, QoE4 and TaT. The codec used was the G.723.1 (of 6.3 kbps) which has a MOS over three units (3.9, as it can be verified in [7]). In order to respect the QoE4 requirement BER was measured during the tests, verifying if it reached values less or equal to two percent.

The tests were carried out in a high-bandwidth link and in a CDMA2000 network. In each test several samples were taken for each performance parameter. The following table resumes the average, median, standard deviation, minimum and maximum sample values for each test carried out.

<table>
<thead>
<tr>
<th>Access Network</th>
<th>Functions</th>
<th>QoE1 (ms)</th>
<th>QoE2 (ms)</th>
<th>QoE3 (ms)</th>
<th>BER (%)</th>
<th>TaT (ms)</th>
</tr>
</thead>
<tbody>
<tr>
<td>High-Bandwidth</td>
<td>Average</td>
<td>745.00</td>
<td>79.07</td>
<td>59.39</td>
<td>0.42</td>
<td>484.22</td>
</tr>
<tr>
<td></td>
<td>Median</td>
<td>940.00</td>
<td>79.00</td>
<td>1.00</td>
<td>0.57</td>
<td>335.09</td>
</tr>
<tr>
<td></td>
<td>Standard Deviation</td>
<td>206.35</td>
<td>24.43</td>
<td>275.36</td>
<td>0.32</td>
<td>342.14</td>
</tr>
<tr>
<td></td>
<td>Minimum</td>
<td>610.00</td>
<td>47.00</td>
<td>0.00</td>
<td>0.00</td>
<td>275.00</td>
</tr>
<tr>
<td></td>
<td>Maximum</td>
<td>965.00</td>
<td>125.05</td>
<td>920.00</td>
<td>0.00</td>
<td>1555.00</td>
</tr>
<tr>
<td>CDMA2000</td>
<td>Average</td>
<td>3113.33</td>
<td>710.47</td>
<td>486.44</td>
<td>2.49</td>
<td>1091.33</td>
</tr>
<tr>
<td></td>
<td>Median</td>
<td>2594.00</td>
<td>571.00</td>
<td>413.00</td>
<td>2.45</td>
<td>1084.00</td>
</tr>
<tr>
<td></td>
<td>Standard Deviation</td>
<td>2193.47</td>
<td>394.17</td>
<td>299.99</td>
<td>0.32</td>
<td>586.42</td>
</tr>
<tr>
<td></td>
<td>Minimum</td>
<td>1312.08</td>
<td>328.09</td>
<td>173.00</td>
<td>2.10</td>
<td>1264.00</td>
</tr>
<tr>
<td></td>
<td>Maximum</td>
<td>3489.00</td>
<td>864.00</td>
<td>1000.00</td>
<td>2.89</td>
<td>2659.00</td>
</tr>
</tbody>
</table>

OMA Limit Values:
- 2000.00 in 1600.00 in 1600.00 in 2.00 in 4000.00.
After the analysis of the table (Image 9), it may be concluded that the PTT system respected all of the performance parameter limits for the high-bandwidth network.

With the CDMA2000 network, we verified that the QoE2, QoE3 e TaT parameters have the average values according to the OMA limits. However, only the QoE3 e TaT parameters have their values according to the limits. In the tests with the CDMA2000 network the PTT server was not located at the Core of the network but on the access network, resulting in a reduction of performance values. If it would be located at the core, the PTT server would probably respect all the performance parameters of the OMA.

6 Conclusion

The main objective of this assignment was to conceive or adapt a PTT Client application, in order to work in an IMS mobile environment, according to the OMA Performance Parameters. We may conclude that the goals were achieved because the defined architecture suggests that the IMS principals and performance parameters were practically fulfilled in both access networks: high-bandwidth connection and CDMA2000.

As a future work it would be important to test the PTT System with a high number of users in order to test the system scalability. On the other hand, it would be also interesting to deploy the PTT Server at the CDMA2000 core to eliminate the variability nature of the access network.

References


[2] Instituto Nacional de Estatística, Estatísticas Portugal População Residente por Local de Residência, Sexo e Grupo Etário


