Proposal of a SIP-Based Method to Supervise Free Roaming Calls

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Abstract. This paper proposes an optimised method to supervise free roaming calls. Since November 28th, 2016, West African countries and members of the Economic Community of West African States (ECOWAS) have decided to ban roaming charges among mobile phone users in these countries. Its implementation however poses a problem of performance costs of their visibility for certain actors such as regulatory authorities (R.A.). To improve this matter we proposed to set up a SIP proxy (Session Initiation Protocol) at the regulatory authorities in order to recover only the signaling of received calls in roaming. The SIP proxy is implemented by Kamailio. Compared to the literature, this article brings a new method of supervision by controlling and following the evolution of calls received in roaming. The results obtained have produced positive effects as only concerned calls will be supervised through the proxy. The proposed solution will facilitate regulators to perform their duties as taxing operators and resolving conflicts between them. It will allow end-users to have visibility into their calls.

Keywords: Free roaming · Signalization protocol · SIP · Supervision system · Kamailio

1 Introduction

Since March 31, 2017, africans traveling to some member countries of the Economic Community of West African States (ECOWAS) can call without worrying about their phone bill. Initiated on November 28th in Abidjan and put into practice on March 31st 2017, free roaming now concerns several countries in West Africa such as: Senegal, Guinea, Mali, Togo, Burkina Faso, Benin,… etc. A call from another ECOWAS member country during a trip to a “Free Roaming” country is no longer surcharged, so callers will no longer have to pay extra fees once they leave their country; despite some restrictions [13]:

- Once abroad, the subscriber will be able to receive free calls for a period of 300 min and within one month,
- Outgoing calls will be billed by the local operator as if he has billed his subscriber,

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- A call made to a country that is not a party to free roaming will be surcharged,
- Free roaming does not apply to internet data.

However the implementation of this "Free Roaming" poses a problem of performance to obtain their visibility through filtering. Visibility issues include subscribers who do not have a way to track their calls duration for the next 300 min, which can engender unexpected bills. They (visibility issues) also include the regulatory authorities, such as Senegal's regulatory authority, whose signaling tools for "Free Roaming" calls are jumbled up with that of international calls. This affects their processes performances (reporting delays and precision) in taxing operators and generates conflicts with them [11, 12].

Existing studies show generally that the implementation of Free Roaming causes many problems and stresses the importance of finding solutions. In Europe, Patrick Maillé et al. have raised two issues arising from the application of free roaming:a problem of relationship between users and service providers and economic in the operators [1, 2]. On the same continent, Jonathan Spruytte et al. have raised a lot of problems after the establishment of Roaming Like At Home (RLAH) and proposed to implement the Roaming Like At Local (RLAL) [3]. As a limit to the RLAL, they emphasized the lack of transparency for consumers. These studies did not indicate a technical solution. Other studies propose solutions but within known limits. Biswas et al. have raised the problem of free roaming by proposing cloud-based system to reduce the communication costs between users when one of the two users is in roaming [4]. By making use of technology, a robust system has been developed that automatically identifies a roaming mobile number and blocks any unknown number but notifies at either end [4, 5]. After the adoption of SIP protocol (Session initiation protocol) in WIMAX networks Cheng et al. pointed out an expensive transfer time when a subscriber is roaming and changing a network to another. When the subscriber changes a network, he sends a RE-INVITE request to register for the new network. They proposed as solution the establishment of a SIP hierarchical domain (HSIP). This means that multiple networks must be managed by a single HSIP server to form an administration domain [6]. Beaubrun et al. have detected an increase in roaming signaling traffic in Next Generation (NG) wireless networks. They proposed to set up a special gateway called the Wireless Interworking Gateway (WING) to facilitate interoperability between the heterogeneous subsystems of NG wireless systems [7].

These solutions do not control and do not track the evolution of calls in terms of their duration. However architectures they propose are to be taken into account for their robustness. Interviews show that some regulators such as Senegal have a system to oversee incoming and outgoing international communications. The defect of this system is that the signaling of free roaming calls is mixed with that of the others.

This paper proposes a method to separate the signaling, to control and monitor the evolution of the meter for 300 min combining robust architecture with filtering of signalization for specific roaming calls.

In this article, Sect. 2 presents the method and architecture proposals, Sect. 3 trains results, Sect. 4 discusses the results and Sect. 5 gives a conclusion to this actual work.

2 Technical Proposition and Design

2.1 Objectives and Steps

As emphasized in the introduction, this paper aims to propose a method to supervise free roaming calls. This means a method to separate the signaling of free roaming calls from that of others; and control/monitor the evolution of their meter for 300 min. Principles of this solution are based on the implementation of a SIP proxy at each regulatory authority that will only recover and process the signaling of calls from “Free Roaming”. A program will be implemented on that proxy to process calls counter through these SIP proxies. We then propose to interconnect regulatory authorities systems.

In short the regulatory authorities will build a intra-network that will be used to route calls from “Free Foaming” to the destination.

To do so, the SIP proxy need the following functions:

- retrieve call signaling to a roaming subscriber,
- process the signaling and route the call to the authority of the country where the roaming subscriber is located,
- The proxy of the authority of the country where the subscriber is located will route the call to the visited operator.

Architecture of Proposed Solution. To design the architecture, we used the works of Cheng et al. and that of Beaubrun et al. [6, 7]. We proposed and compared the following two architectures.

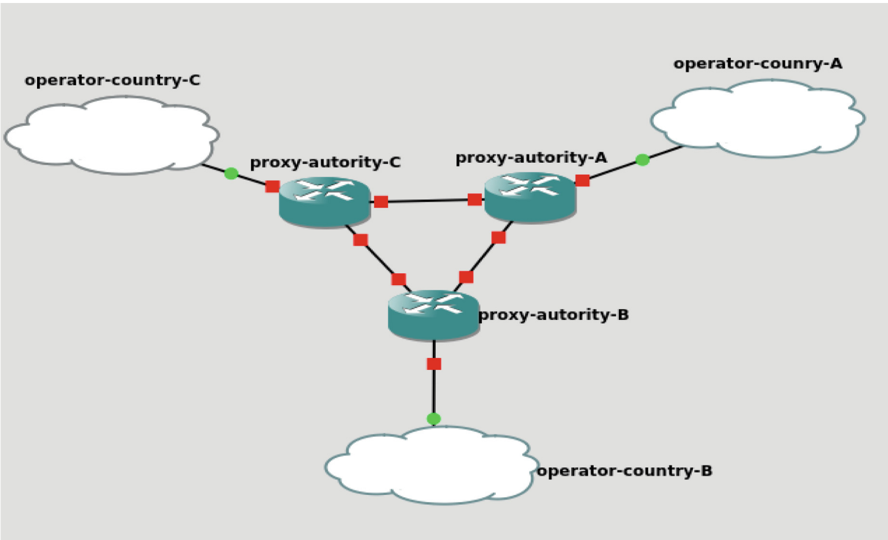


Fig. 1. Basic architecture : Regulation authorities direct proxy interconnexion

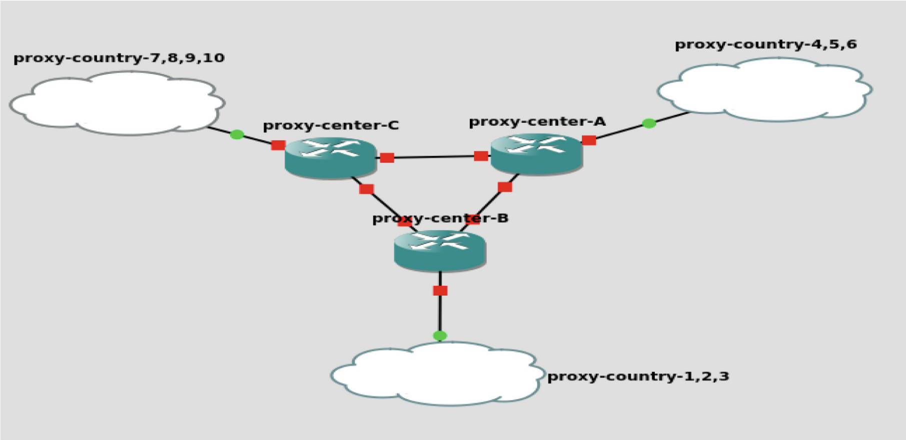


Fig. 2. Advanced architecture : Shared proxy centers interconnexion

Example of Call Processing
If for example, a call is made to a subscriber Jean of a country C op-C operator roaming in a visited network of an op-A operator in country A, the scenario below will be observed:

- The operator op-C will direct the call to the authority authority-country-C,
- Country-authority-C will perform the processing and route the call to authority authority-country-A:
  – Through a direct interconnexion with authority-country-A 's proxy (Fig. 1)
  – Through a shared proxy center network covering multiple countries (Fig. 2)
- The authority authority-country-A will route the call to the operator op-A,
- The operator op-A to make the call to the subscriber Jean,
- John will then have the choice of answering or not answering.

Entities Needed to Operate the SIP Proxy
The SIP proxy will be combined with a database system on which it will store information from the signatory countries of the "Free Roaming" (that will serve as its routing information) and the information of the recipient (for the control of the 300 min). Its tables will store the recipient information's and his calls duration sum in minutes (its values will be incremented at the end of each call) (Fig. 3).

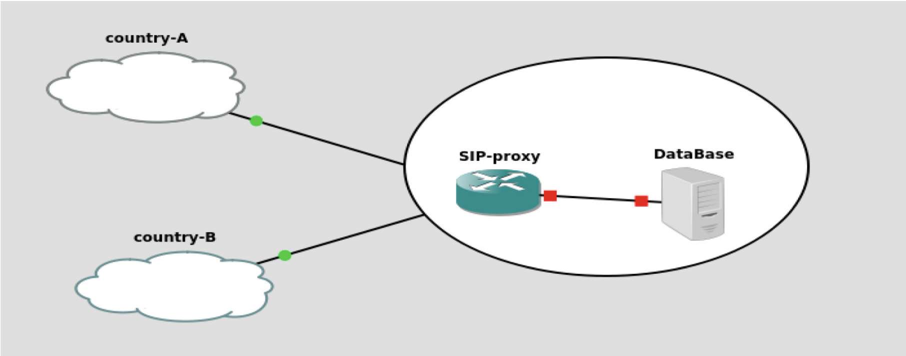


Fig. 3. Elements necessary for SIP proxy operation

2.2 SIP Protocol and Implementation Algorithm

SIP Protocol

As pointed out above, the solution is based on the SIP protocol. SIP (Session initiation Protocol) is an application layer control protocol that can establish, modify and terminate multimedia sessions [8]. SIP works in client-server mode. Like the HTTP protocol, the client sends commands and the server responds with response codes. Unlike HTTP, a SIP entity is in most cases both client and server. Between the client and the server (between the two communicators), there may be servers called proxy servers such as [8, 9]:

- Proxy server: it receives requests from clients that it processes itself or that it sends to other servers,
- the redirection server: this is a server that accepts SIP requests, translates the destination SIP address into one or more network addresses and returns them to the clients,
- The user agent "UA, User Agent": it is an application on a user equipment that sends and receives SIP requests,
- Registrar: This server registers users. It also updates the location database,
- B2BUA (Back To Back User Agent): it is like a proxy with the difference that the B2BUA establishes communication and monitors.

Consider a call between Alice and Bob. It is assumed that Alice wants to communicate with Bob (Fig. 4).

1. Alice sends a SIP INVITE message to her operators proxy (orange).
2. Alice Proxy sends the request to Bob's proxy server (tigo.sn),
3. Bob's proxy server sends the request to Bob,
4. being available Bob responds with a 200 OK response and this response goes through the same entities,
5. Alice sends an ACK confirmation message and start to communicate.

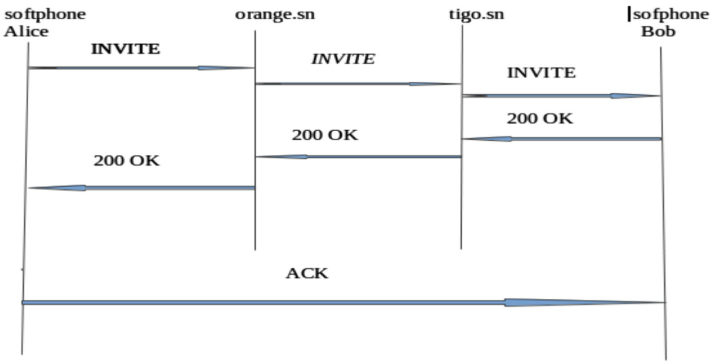
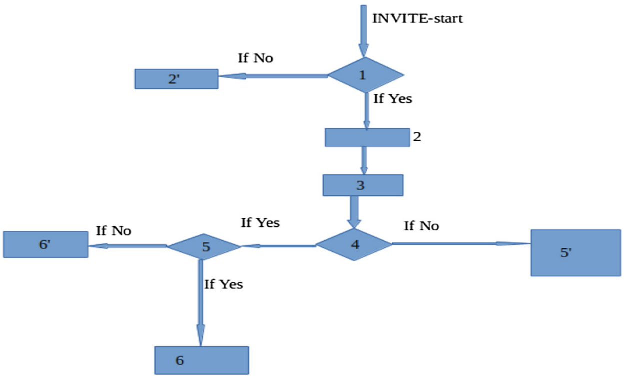


Fig. 4. Session setup between two users who do not have the same provider

Algorithm for Setting up the SIP Proxy

The SIP proxy will process a call in three phases: when receiving SIP INVITE, ACK and BYE commands. Let the table APPEL (id, uri-dest, counter) the table used to store the recipient's information, the INVITE, ACK and BYE messages will be processed as follows: (a) processing an INVITE, (b) then an ACK and (c) finally a BYE.

Processing an INVITE



1: check if the country of destination is part of the signatory countries of "Free Roaming"
2 ′: reject the call
2: continue treatment
3: retrieve the recipient's UR
4: check if the user is in the database (table CALL)
5 ′: create an entry in the CALL table and route the call
5: check if the meter is less than 300 min
6 ′: redirect the call
6: get the counter and route the call.

Processing the ACK command

After receiving an ACK, the SIP proxy will process the message as follows:

- recover the TIMESTAMP T1,
- calculate the possible duration of the call,
- set the duration of the call,
- Continue the call.

Processing a BYE query

After receiving a BYE, the SIP proxy will trigger the following processes:

- recover the TIMESTAMP T2,
- calculate the duration T of the communication: T = T2-T1,
- Update the counter: counter = counter + T.

3 Implementation and Results

3.1 Implementation

We propose the Kamailio SIP proxy as an implementation. Kamailio is a telephony server based on the SIP signaling protocol [10]. To work, Kamailio uses several modules and those we will use are:

- RTPPROXY: to ensure the proxy function of Kamailio,
- SQLOP: for interaction with databases,
- MYSQL: to implement the MYSQL database,
- DIALOG: to set the duration of the dialogue,
- CARRIERROUTE AND RR: for routing.

We have used also Ubuntu.14 and MYSQL database. We used the language of the Kamailio configuration file.

3.2 Results

When a subscriber is roaming, he has the ability to receive free calls for the first 300 min and within a month. We just simulated the 300 min counter. In the simulation, we used two SIP servers and the proxy in their middle (Fig. 5).

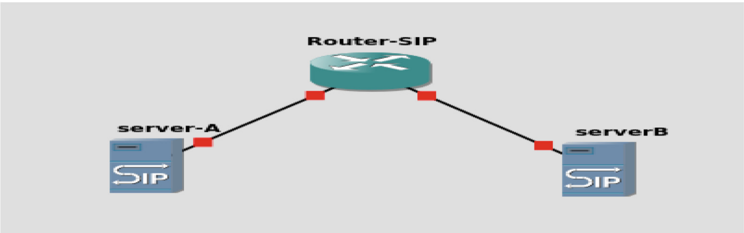


Fig. 5. Architecture simulation

In the architecture above, the servers simulate the operators and R-SIP simulates the SIP proxy. We will begin by viewing the description of the CALL table on which the recipient's call information is stored (Fig. 6).

```
mysql> desc `CALL`;
+---------+-------------+------+-----+---------+----------------+
| Field   | Type        | Null | Key | Default | Extra          |
+---------+-------------+------+-----+---------+----------------+
| id      | smallint(6) | NO   | PRI | NULL    | auto_increment |
| login   | varchar(60) | NO   |     | NULL    |                |
| counter | varchar(60) | NO   |     | NULL    |                |
+---------+-------------+------+-----+---------+----------------+
3 rows in set (0.00 sec)
```

Fig. 6. Structure of the CALL table

At the counter time is expressed in seconds.

We will now view the evolution of the counter after two successive calls.

```
mysql> select * from `CALL`;
+----+-------------------------+---------+
| id | login                   | counter |
+----+-------------------------+---------+
| 46 | sip:1002@192.168.43.65  | 35      |
+----+-------------------------+---------+
1 row in set (0.00 sec)

mysql> select * from `CALL`;
+----+-------------------------+---------+
| id | login                   | counter |
+----+-------------------------+---------+
| 46 | sip:1002@192.168.43.65  | 98      |
+----+-------------------------+---------+
1 row in set (0.00 sec)
```

Fig. 7. Evolution of the counter

We see that the first call lasted 35 s and this duration was recorded in the CALL table. After the second call, the counter was incremented to 98 s, which meant the call lasted 63 s.

To simulate the call control, we will now view the behavior of the KAMAILIO router when receiving a call to a recipient who has consumed its 300 min. We simulated the 300 min at 5.

```
09:41:42.669 GRAVE: [569] impl.protocol.sip.OperationSetBasicTelephonySipImpl.pr
ocessResponse().589 Received error: 403 your corespondent has consumed his 5mn
```

Fig. 8. Rejection of an INVITE

We sent a message to the caller informing him that the called party had consumed his 300 min of receiving the call. It was just to show that we can reject or redirect the call.

To conclude, this section has presented the proposed solution and its implementation. The next part will discuss the obtained results.

4 Discussion

Our main objectives are: to separate the signaling of the calls, to control and to follow the evolution of the calls (evolution of the counter) in the case of roaming situations. The implementation of a SIP proxy at the regulatory authorities seems to be a functioning control and supervision solution.

Its deployment through the proposed architectures (Figs. 1 and 2) allows to physically separate the signaling of the calls as shown in the results (Fig. 7). It also facilitates calls control (Fig. 8) in a separate infrastructure, distributing performances costs in regulator authorities' infrastructures. By this mean a subnetwork linking countries members of the free roaming zones is also created with costs reduction in case of central proxy sharing (Fig. 2).

Compared to the literature [6, 7], the similarities are on the proposed architecture (Figs. 1 and 2). We therefor beneficiate of the robustness induced by the flow calls separation. We also added monitoring and tracking calls support for the regulators; and notification functionalities for callers as our main contributions (visibility).

This proposal allows regulatory authorities to perform their duties such as taxation and management of conflicts between operators (Fig. 7) by automated data processing (mainly semi-automated in case of non filtering solution). It also allows roaming end-users to follow the evolution of the meter (Fig. 8) in real-time.

However a study must be made to evaluate routing and By Passing problems.

5 Conclusion

The purpose of this article was to propose a method to supervise free roaming calls in west of Africa. We can conclude from the results obtained that setting up a SIP proxy, on either the basic (Fig. 1) or advanced (Fig. 2) architecture is a good method to fulfill those goals within that specific context.

The results obtained show that the proposed algorithm allows to separate call flows processing in a specific server, reducing the impact on the regulator authorities infrastructure with a possible cost reducing through sharable SIP proxy centers. Compared to previous works, the proposed solution improves the calls visibility and allows regulatory authorities to perform their obligations as taxation operator and conflict manager between operators by providing efficient tools. It also allows consumers to have visibility on their calls in a roaming context.

However a study must be made to avoid routing and By Passing security risks.

The proposed architecture seems also to provide a deployable service in a single (sub)regional network context.

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