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An Open API Framework to Support the Capability Discovery Mechanism Using the Enhanced SIP OPTIONS Exchange

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Abstract

Mobile service providers allow their customers to use roaming services when they travel to other countries. In the standard mobile call setup procedures, multiple VPMN MSISDNs cannot be provided to the roamers. Also, call setup to the roamer is indirectly routed through the home network of the roamer, which results in the usage of expensive international carrier's trunks. In this paper, we propose a software framework that provides multiple VPMN MSISDNs in a UMTS. The proposed framework solves the indirect routing problem and can be easily deployed to mobile operators since it does not modify standard telecommunication protocols.

Keywords: *MSISDN, roaming, UMTS, tromboning, software framework*

1. Introduction

Cellular networks have been evolving very quickly in the last decade. Universal Mobile Telecommunications System (UMTS) is a third generation mobile cellular technology for networks based on the Global System for Mobile Communications (GSM) standard which was developed by the 3rd Generation Partnership Project (3GPP) [1]. UMTS phones can roam easily onto other UMTS networks if the providers have roaming agreements in place. Roaming agreements between networks allow for calls to a customer to be redirected to them while roaming. The user is uniquely identified by an International Mobile Subscriber Identity (IMSI) associated with an UMTS network mobile phone user. It is stored in the Universal Subscriber Identity Module (USIM) card inside the phone and sent by the phone to the network so that the network can identify and authenticate the IMSI. Therefore, UMTS users can use their phones while they are in foreign countries without changing their handsets or phone numbers.

Some users who roam frequently between two countries may want to have local numbers in foreign countries. The reason is that local users can call them at a cheap rate without worrying about international IDD calls. Also, local users who receive calls from them will identify calling numbers as local numbers, which are familiar to them. In addition, the users can present a multi-national appearance to local users.

However, UMTS does not provide this service. It only allows a single Mobile Subscriber Integrated Services Digital Network Number (MSISDN) with a single IMSI. This paper presents a software framework that supports providing mobile communication devices with multiple MSISDNs using one IMSI. This framework does not require a new type of USIM card in a mobile device. It can be implemented on a Service Control Point (SCP), which is a standard component of an intelligent network telephone system, so that it can be adapted to the existing UMTS easily. The standard components of UMTS and Signaling System 7 (SS7) protocols are not need to be modified and it does not affect an automatic roaming function. The framework allows operators in one country to cooperate with operators in another country to offer a local MSISDN in the visited country without changing an USIM card. Also, a mobile device must perform a location update procedure

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in the operator's network to use the service, thereby creating a monetarily beneficial relationship to both operators and the roaming users.

The next section introduces related works to provide multiple Visited Public Mobile Network (VPMN) MSISDNs to a mobile device. In Section 3, we propose the software framework to provide a local MSISDN at each partner network for a subscriber. The paper concludes with a short discussion.

2. Related Works

When a RCS user scrolls though their address book, they will see their contacts with the RCS services that are available to communicate. That is, each RCS user's device can have different versions of RCS client or can have different functionalities. Therefore, before users use a RCS service with their contacts, they should find out whether their contacts can provide the service or not. This mechanism is called as a capability discovery framework. The RCS specification provides two methods to implement the framework. One is using the Session Initiation Protocol's (SIP) OPTIONS request and the other is using a presence-based solution defined in RCS Release 1-4. First method uses an end-to-end message to query the capabilities of the target contact. Second method uses a server to manage all RCS users' capabilities and the capabilities are queried against the server. The RCS specification mentions that all terminals which want to achieve minimum conformance to this version should provide the SIP OPTIONS request method as their primary discovery mechanism. And it mentions that network service providers can provide the presence-based solution as their optional discovery mechanism. This paper covers first method because it is mandatory.

2.1. Capability Discovery Process though SIP OPTIONS Message

The default mechanism for capability discovery is based on the exchange of a SIP OPTIONS request, a peer-to-peer message exchanged between RCS clients. Figure 1 shows the flow of messages when User A want to discover User B's capabilities.

(1) User A sends SIP OPTIONS message to O-CSCF to discover User B's capabilities. The message's To field is set to User B's address. Also, User A's capabilities are included in the Contact header field of this message to let User B know User A's capabilities.

(2) O-CSCF forwards the message to O-OAS by looking up initial Filter Criteria (iFC) information.

(3-4) O-OAS forwards the message to O-CSCF. And O-CSCF forwards the message to T-CSCF by looking up the routing information.

(5) T-CSCF forwards the message to T-OAS.

(6-7) T-OAS forwards the message to T-CSCF. And T-CSCF forwards the message to User B.

(8-14) User B takes out User A's capability information from the message and stores it in its local storage. User B sends SIP 200 OK message which includes User B's capabilities. When User A receives the message, User A stores the information in the local storage.

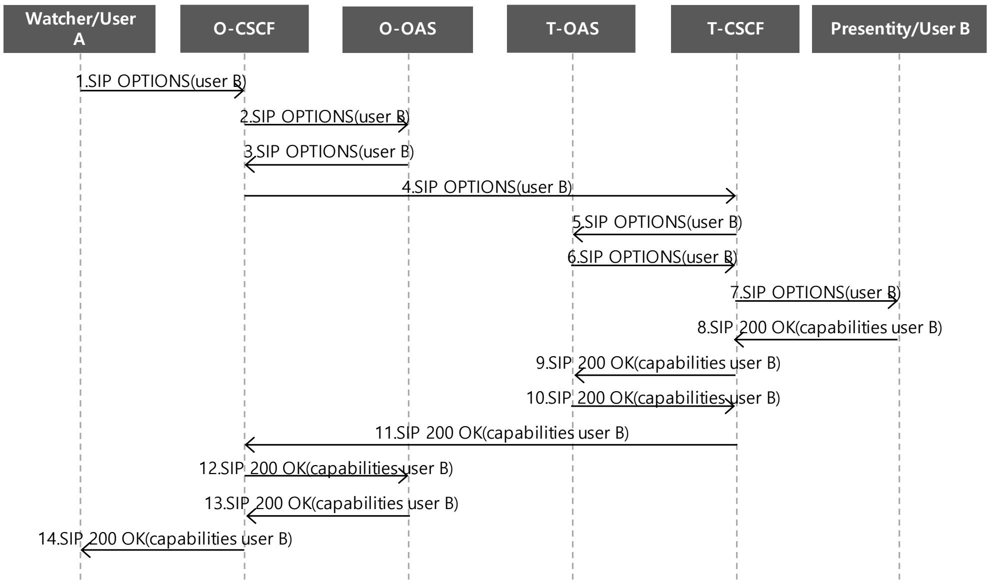

Figure 1. Flow Diagram for the Capability Discovery Mechanisms using SIP OPTIONS

2.2. OPTIONS Response Handling in Exceptional Cases

In Figure 1, User B is an RCS user and is currently registered. However, other users might not be currently registered or might not an RCS user. In these cases, users can not respond to OPTIONS request. Figure 2 shows the flow of the messages when User C is not registered currently and User D is not an RCS user.

(1-4) When User A wants to discover User C's capabilities, it follows same flows with (1-4) in Figure 1. In case of User C is an IMS user but is not currently registered at CSCF, it takes (5-8). In case of User C is currently registered but cannot respond to OPTIONS request, it takes (9-15).

(5-8) Because User C is not registered at T-CSCF, T-CSCF sends SIP 480 Temporary Unavailable message to O-CSCF. O-CSCF forwards the message to User A. Then User A notices that User C's capability discovery had failed and only the capabilities available to an offline contact are offered.

(9-11) Since User C is registered, T-CSCF forwards the OPTIONS request to User C through T-OAS.

(12-15) T-CSCF waits for a response from User C for some period. Because there was no response, it sends SIP 408 Request Timeout message to O-CSCF. O-CSCF forwards the message to User A. Then User A changes User C capability information just as (8).

(16-19) If User A want to discover User D's capability even though User D is not an RCS user any more, OPTIONS request is sent to T-CSCF where User D was registered.

(20-23) Because T-CSCF knows that User D is not an RCS user, it sends SIP 404 Not Found message to User A. User A excludes User D from a RCS user list.

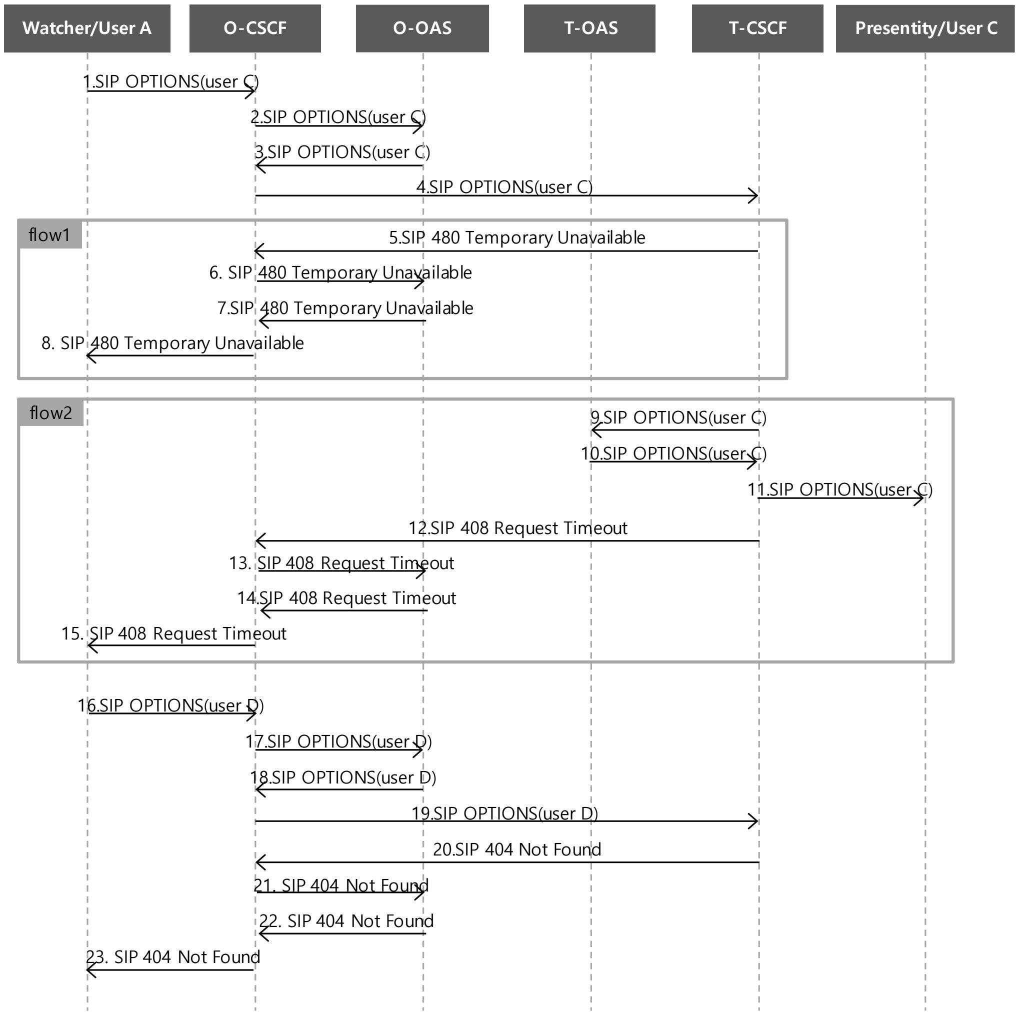

Figure 2. Flow Diagram for OPTIONS Response Handling in Exceptional Cases

3. Proposed Scheme

This paper presents a web-based open API framework to spread the use of RCS. By using the framework, web app developers can develop RCS capable apps easily without the knowledge of IMS protocols. The framework uses Representational State Transfer (REST) architecture since it is light weight and scalable. REST architecture provides interoperability between computer systems on the Internet and represents Web resources as a URL so that we can use REST-compliant services with a simple HTTP request.

Figure 3 shows an example of the system architecture which uses the proposed framework. Existing RCS users discover other users' capabilities using end-to-end OPTIONS message exchange through mobile access network, core network and IMS network including CSCF. When we use the framework, the client sends a request to the framework and it converts the request to a SIP or XCAP message. The framework includes two agents for protocol conversion. SIP Service Agent (SSA) deals with SIP protocol conversion and XCAP Service Agent (XSA) deals with XCAP protocol conversion.

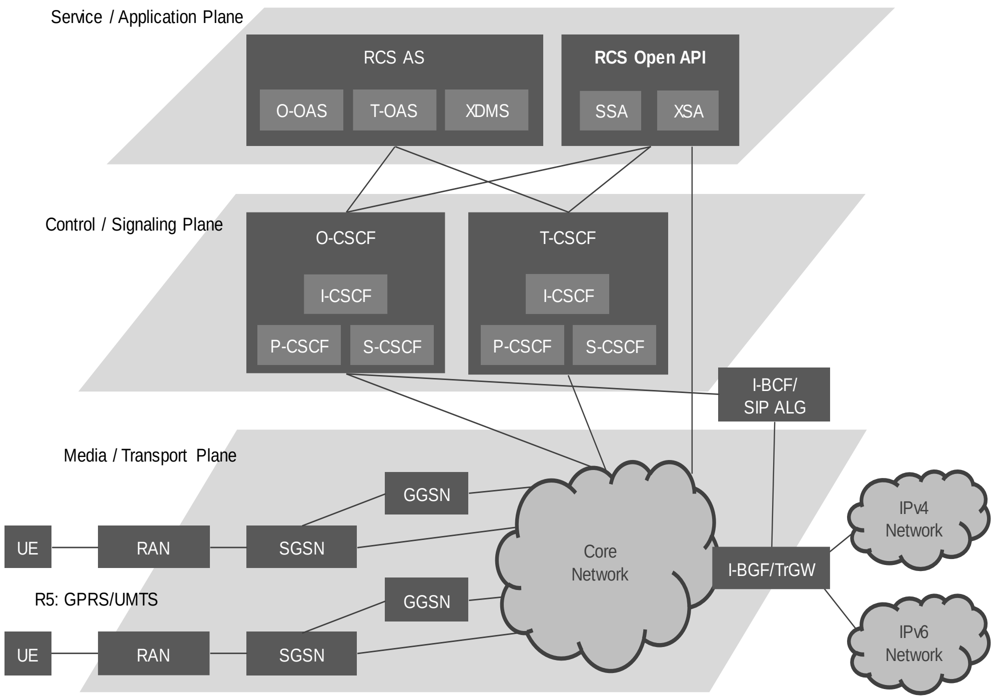

Figure 3. System Architecture for RCS Capability Discovery using the Open API Framework

3.1. Registering Capability through the Open API Framework

The open API framework does not use end-to-end transaction to perform the capability discovery. The capabilities are registered in a server and they are queried against the server. This method has a limitation that a requester cannot discover other's capabilities in real time. Generally, user's capabilities are not changing frequently. Also, real time end-to-end transaction is not suitable to web environment. Therefore, we use server-based method instead of real time end-to-end method. Figure 4 shows the flow of the messages when User B wants to register its capabilities to the server.

(1) User B sends HTTP PUT message to XSA. The message includes User B's capabilities in JavaScript Object Notation (JSON) format.

(2) XSA converts the HTTP PUT message to a XCAP PUT message and sends the XCAP message to XDMS.

(3) XDMS takes out the User B's capabilities from the message and save them to a storage as a XML document. Then XDMS sends XCAP 200 OK message to XSA to inform that the capability was registered successfully.

(4) XSA converts XCAP 200 OK message to HTTP 200 OK message and sends the message to User B.

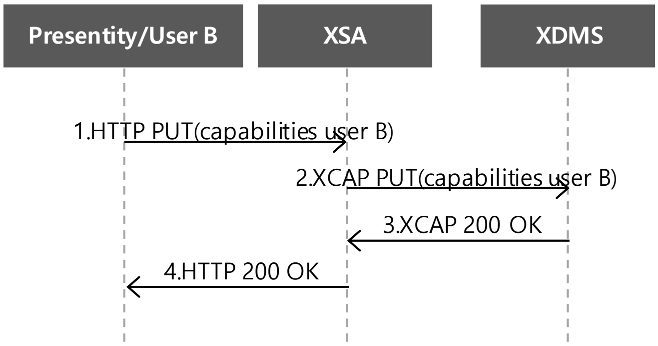

Figure 4. Flow Diagram for Registering User's Capabilities using the Open API Framework

The RSCP searches for VPMN MSISDN by using the information in the message and sends it back to the GLR on the SSI acknowledge message. The GLR sends the VPMN MSISDN and the Originating Customized Applications for Mobile Enhanced Logic Subscription Information (O-CSI) to the MSC on the ISD message. The service key for the multiple MSISDN service in the O-CSI is set to 1 and the Service Control Function address is set to the address of the RSCP. The O-CSI is used when a mobile station originates a call so that the MSC can retrieve the VPMN MSISDN from the RSCP by using the CAP Initial Detection Point (IDP) message and the CAP Connect message.

3.2. Mobile-terminated Call in HPMN

When a caller sends a call to VPMN MSISDN and the subscriber receives the call in HPMN, the procedure is performed as follows. The originating MSC (OMSC) sends the Initial Address Message (IAM) to the GMSC. The GMSC should be configured to route the Send Routing Information (SRI) message to the RSCP for the specified MSISDN range, which includes VPMN MSISDNs. Thus the RSCP emulates HLR behavior for processing SRI. The GMSC sends the SRI message to the RSCP. The RSCP changes the

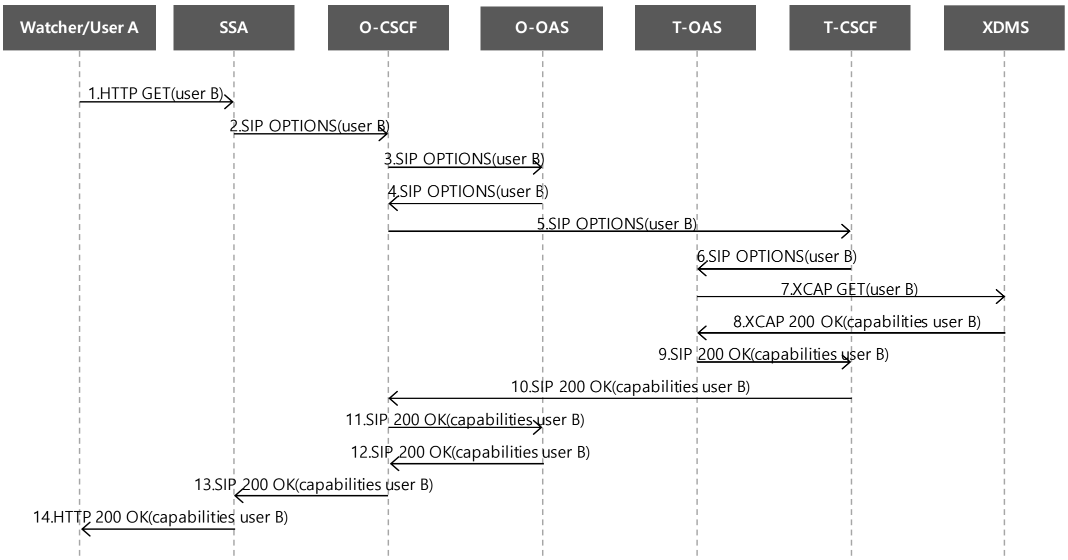

Figure 5. in VPMN

called number from VPMN MSISDN to HPMN MSISDN and adds Calling Name Presentation (CNAP) information. The CNAP is included in the DISPTEXT parameter and contains the called number so that the receiver can identify the called number. Also, the Generic Number parameter in the Additional Signal Information (ASI) contains the VPMN MSISDN. The RSCP relays the message to the GLR. Since the subscriber is in HPMN, the GLR does not have the profile. Thus the GLR rejects the SRI by setting the OR Not Supported parameter on the SRI acknowledge message. Then the RSCP sends the CSI to the GMSC on the message. The GMSC sends the IDP message to the RSCP for retrieving the HPMN MSISDN of the receiver by using the CSI. The RSCP sends the HPMN MSISDN to the GMSC on the Connect message. The GMSC sends IAM to the HPMN GMSC through international carriers. The GMSC sends the SRI message to the HPMN HLR and the HLR sends the Provide Roaming Number (PRN) message to the Terminating MSC (TMSC). The TMSC sends the Mobile Station Roaming Number (MSRN) to the GMSC on the PRN acknowledge and the SRI acknowledge message. The GMSC sends IAM to the TMSC and the TMSC pages the mobile station. Further ISDN User Part (ISUP) procedures are executed by using Address Complete Message (ACM), Answer Message (ANM), Release (REL) and Release Complete (RLC).

3.3. Mobile-terminated Call in VPMN

When a caller sends a call to VPMN MSISDN and the subscriber receives the call in the VPMN, the procedure is performed as Figure 1. The OMSC sends IAM to the GMSC. As the call flow in HPMN, the GMSC sends SRI to the RSCP. Then the RSCP forwards SRI to the HPMN GLR. Since the GLR knows that the subscriber is roaming, it sends PRN to the VPMN TMSC with the ASI and the DISPTEXT. The TMSC stores the GN and the DISPTEXT and sends the MSRN of the mobile station to the GLR on the PRN acknowledge. The GLR forwards the MSRN to the GMSC on the SRI acknowledge. Then the GMSC sends IAM to the TMSC. The TMSC pages the mobile station and sends the DISPTEXT and the VPMN MSISDN on Redirecting Party BCD Number. Then remaining ISUP procedures are executed. The voice bearer path is (C)-(E)-(D).

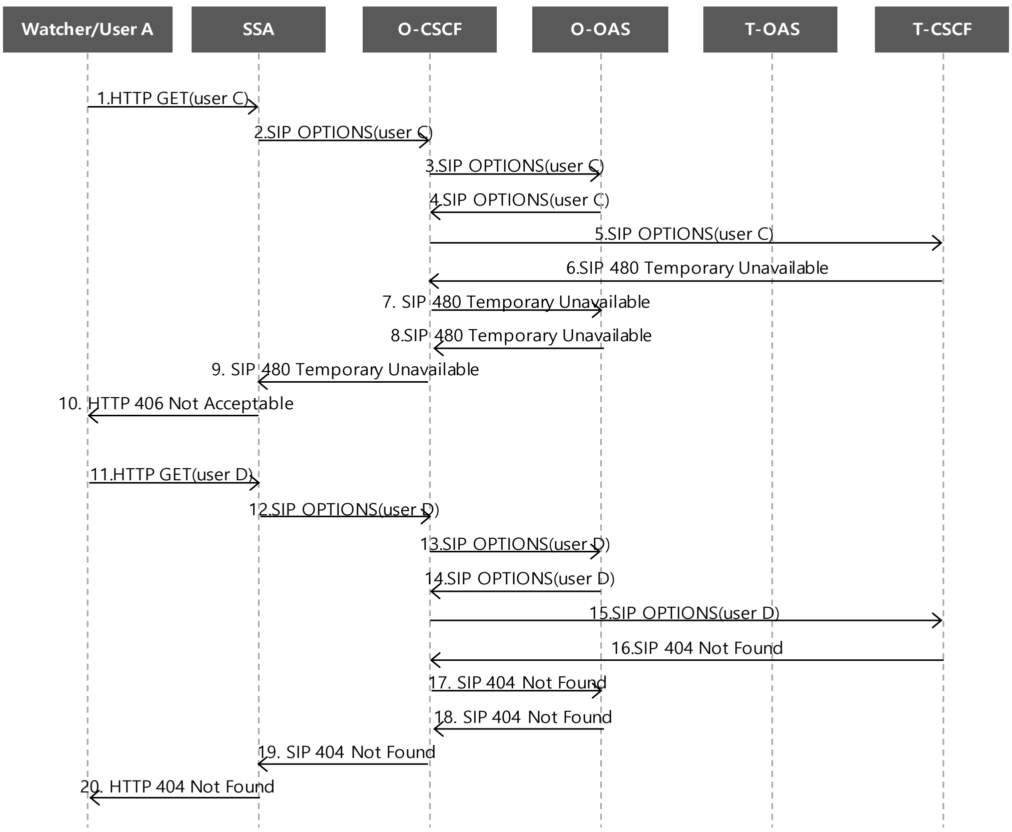

Figure 6. in VPMN

If we compare the mobile-terminated call procedures of the schemes in VPMN, the USIM-based scheme has the longest bearer path. Furthermore, the path consists of international voice trunks, which cause the tromboning problem. The MAP-based scheme and the proposed scheme have the identical voice bearer path and do not have the tromboning problem. However, their signaling paths are different.

4. Conclusions

This paper introduces a new scheme for providing multiple VPMN MSISDNs to roamers and compares with other related schemes. Proposed scheme uses the RSCP software frame work to provide the service. It does not need a new type of a USIM card and eliminates the tromboning mobile call setup for international roaming users. Also, since it uses standard telecommunication protocols, such as MAP, CAP, and ISUP, without modification, adopting this framework to existing mobile telecommunications systems of mobile operators becomes simple. Future work will include the analysis of the computational complexity of the execution of signaling in 3GPP wireless networks, and how it impacts the cost of the network.

References

[1] A. Rico-Alvarino, M. Vajapeyam, H. Xu, X. Wang, Y. Blankenship, J. Bergman, T. Tirronen and E. Yavuz, "An overview of 3GPP enhancements on machine to machine communications", IEEE Communications Magazine, vol. 54, iss. 6, **(2016)**, pp. 14-21.

[2] S. P. Rao, S. Holtmanns, I. Oliver and T. Aura, "Unblocking Stolen Mobile Devices Using SS7-MAP Vulnerabilities: Exploiting the Relationship between IMEI and IMSI for EIR Access", Proceedings of the IEEE Trustcom, vol. 1, **(2015)**, pp. 1171-1176.

[3] L. Chin and J. Chen, "SIM card based e-cash applications in the mobile communication system using OTA and STK technology", Proceedings of the IET International Conference on Wireless, Mobile and Multimedia Networks, **(2006)**, pp. 1-3.

[4] Y. Lin, "Eliminating tromboning mobile call setup for international roaming users", IEEE Transactions on Wireless Communications, vol. 8, no. 1, **(2009)**, pp. 320-325.

[5] H. P. Singh, S. Singh and J. Singh, "Performance Analysis of Interleaving Scheme in Wideband VoIP System under Different Strategic Conditions", IJAST, vol 44, **(2012)**, pp. 49-60.

[6] ITU-T, "The international public telecommunication numbering plan," Recommendation E.164, **(2010)**. pp. 8-13.

[7] 3GPP, "Group services and system aspects; support of optimal routing, service definition," TS 22.079, **(2008)**, pp. 6-14.

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