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Performance Analysis, Evaluation and Up gradation of Push to Talk over Cellular

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***Abstract*—Push-to-Talk (PTT) is a half-duplex communication line in which only one party can speak at a given time. It is a quick, one-touch, conference call uses momentary button to switch between voice reception mode and transmit mode. Push-to-Talk over cellular (PoC) is a new type of communication service uses cellular network on 3G/4G network and also Wi-Fi as the underlying wireless connectivity for PTT. In this paper the efficiency of today's PoC is analyzed and results show that caller may not be able to talk for about 8 seconds after pressing the talk button. A very long waiting time may affect the use of PoC. Therefore, a performance enhanced and optimized scheme is presentedin order to reduce the waiting timeand number of transmitted messages in a group call for PoC users.**

***IndexTerms*-Push to talk (PTT); Performance analysis; Push to talk over cellular (PoC);**

I. INTRODUCTION

Push to Talk over Cellular (PoC)[1] –[4] is a wireless technology which allows subscribers to use their phone as a walkie-talkie with extended range throughout the operator's serving area. A typical Push-to-Talk (PTT) connection connects almost instantly. A significant advantage of PoC / PTT is the ability for a single user to reach an active talk group with a single button press; users need not make several calls to coordinate with a group.In PoC, speech is carried in the form of packets. Client 1, for instance, sends speech packets to the PoC server via the cellular network. The server forwards these packets on to the receiving mobiles; the performance of the service depends on the network used. The network should meet the required QoS (the network should be 3G or 4G). All these qualities make PoC particularly useful in business communications. For instance, a construction worker would find it much more convenient to hear messages burst out of a speaker on his phone, rather than having to put down his tools, take off his gloves.PoC offers many advantages when compared to walkie-talkie which is limited to parts of a city with expensive, bulky handset. But PoC is reachable in extended range as it uses 3G/4G network and No separate handset is required Personal alert can be sent. It has adhoc/predefined groups.

PoC give information about presence/ availability of its users. International organization of standardization (ISO) Open mobile alliance established PoC standards.SIP (Session Initiation Protocol) protocol [5]fromIETF (Internet Engineering Task Force) is used by PoC clients and server before establishing PoC communication to invite group members to enter into talk session, and then RTP (Real Time Transmission Protocol) protocol is used to transmit voice data and RTCP (RTP Control Protocol) protocol to coordinate the media-burst control.The OMA standards recommend some of the basic functionalities that a PoC service must provide. Firstly, it must support basic setting up of calls. Session Initiation Protocol (SIP) is used for this. It needs to control who holds the "Floor", since in a half-duplex channel only one party can talk at a time. Another module needs to carry voice packetsor "talk bursts" in the appropriate format. Yet another need is to have a document engine that can acquire, maintain, update and exchange user information documents (e.g. their registration, activation status, contacts, etc.). An Extensible Mark-up Language (XML) based module handles this part due to XML's flexibility and widespread use in IP networks.

PoC technology which is based on SIP makes users wait a period of time before initiating a session due to the much longer transmission latency of wireless network when compared to that of wired environment. It is nearly about 7-8 seconds and about 10 seconds[2] when the cell phone is in sleep mode. So such a long waiting time will be inconvenient for PoC users. So it is an important matter for PoC service to reduce the session establishment time. In previous paper[4] , method of shortening waiting time was presented but it can only be applied to one-to one PoC session but unsuitable for use on one-to-many PoC communication.

To reduce PoC waiting time in research [7], proposed a mechanism to allow the sender to start taking before the connection is installed. It can curtail waiting time with inevitable load of additional voicestorage function in PoC server and expense of more power consumption in cell phone and network bandwidth wasting from transmitting redundant Wake-Up messages and it will worsen network bandwidth usage. And there will be PoC Server burden, once established connection fails.

In this research, we first evaluate current transmission latency on SIP-based PoC service, and present an enhanced scheme without altering original SIP protocol and PoC standard, further simplifying procedures for SIP connection buildup, reducing the numbers of controlmessage, and effectively shortening waiting time before talking for PoC users.

II. CONVENTIONAL POC TECHNOLOGY AND PERFORMANCE EVALUATION

A. PoC technology based on SIP

Client-Server Model used in PoC is shown as Fig. 1. The entire system consists of three main elements—(1) SIP Core: before talking session starts, SIP protocol is used by PoC to ensure that PoC group members can talk successfully. Therefore, SIP Core is responsible for establishing connection via SIP protocol. (2) PoC Server: It is in charge of coordinating Media-burst control and RTP protocol to transmit voice as well voice data copy for multi-user communication through RTCP protocol.

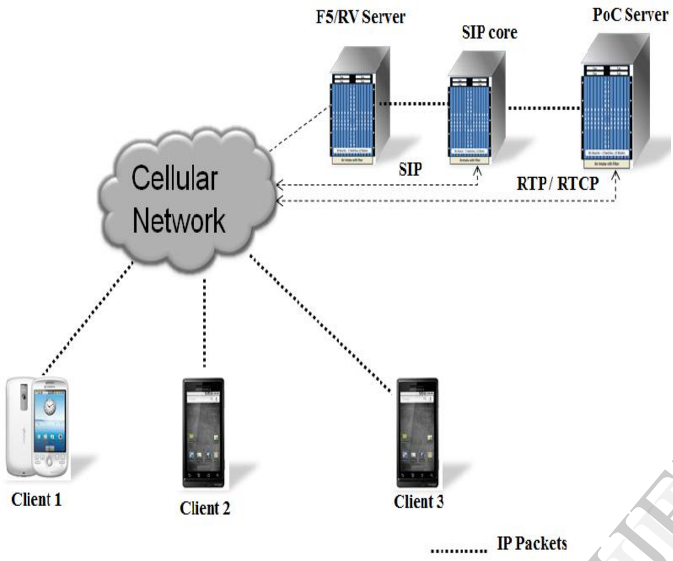

Figure 1. PoC network architecture

(3) F5/Server:The F5 server is added to the network using the EMS. In order to support FCAPS functionality for these type of nodes, a new supplementary node FCAPS framework is introduced.Fig. 2 shows the message flow when user, Client 1, initiate the PoC group Call by pressing talk button on the cell phone, with two stages of SIP session establishment and media burst control, respectively. Client 1 sets up the PoC Session with group members, Client 1 to 3. When Client 1 pressesthe PoC talk button, SIP protocol is used to establish connection from Client 1 to PoC Server before talking; then PoC Server also connects to other group members Client 2 and Client 3 by SIP protocol. SIP is a protocol regulated by IETF on internet for setting up the communication session. After SIP session is established, PoC Server employs RTCP protocol to send related MBC (Media-Burst Control) procedures, and then Client-1 uses RTP protocol to send voice data to PoC Server and after that PoC server copies the voice data and it is transmitted to all the recipients (Client-2 to 3 in Fig.2).

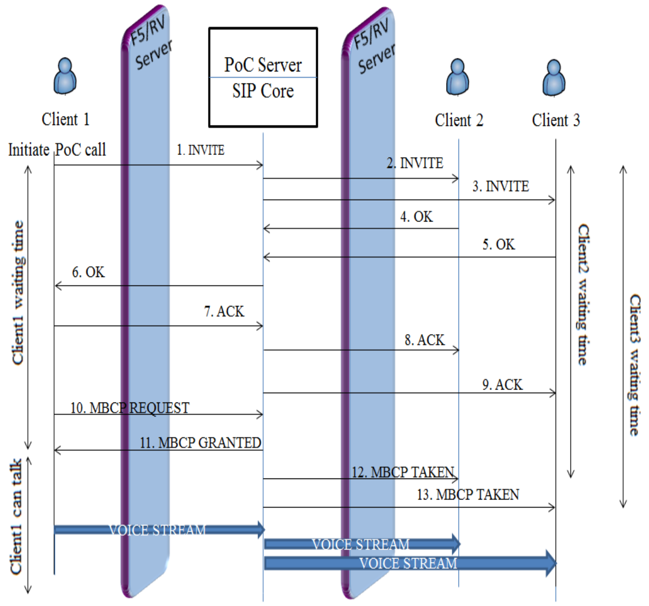

Figure2.Coventional PoC message Flow

(1) SIP session establishment stage: Client-1 wants to talk to group members of clients 2 to 3 by making group call. Firstly Client-1 pressesPoC talk button to execute SIP protocol. Client 1 sends Invite message (1. Invite in Fig. 2) to PoC Server, which in turn issues Invite message to each and every recipients (2.Invite to 3.Invite), meaning Client 1 intends to start a session with Client 2 and Client 3. However, above two classes of SIP Invite contents are different: IP address of receiver in 1. Invite message is that of PoC Server, but Invite messages from PoC Server have the recipient IP addresses of Client 2 and Client 3, respectively. Each recipient in turn send acknowledges for Invite message by sending OK message to PoC Server (4.OK and 5.OK). After receiving OK messages from all recipients PoC transmits an OK message to Client 1 (6. OK). Two stage OK messages have also different contents as follows: 6.OK responds to 1.Invite message and OK messages from recipients respond to 2.Invite to 3.Invite messages. Client 1 sends ACK to PoC Server (7.ACK) after receiving OK, and in turn PoC Server relays ACKs to all recipients (Client 2, Client3) when receiving ACK from Client-1 (8.ACK to 9.ACK). After Client 2 and Client 3 receive ACK, the complete group call session connection is establishedand ready to start communication. SIP session establishment for PoC can be categorized in two parts: one is Client 1to PoC and other is PoC to all recipients (Client 2 and Client 3)

(2) Media-burst control stage: After connection built up, PoC Session enters into Media-burst control stage. PoC Server issues a MB (Media-Burst) Granted to Client-1 and notifies Client-1 gaining Media-burst is able to talk. After that, PoC Server transmits MB Taken to every recipient (Client2, Client3) telling them Client-1 is ready to talk. Finally, Client-1 sends voice stream to PoC Server, where voice data is copied and transmitted again to other members in the same group. Noteworthy is that the first Media-burst is given to the session establishment, Client 1. If this is not the case, other members

who want to talk need to send out MB Request messages to PoC Server first. It means for the first time PoC Session is built up, MB Request can be saved to reduce the number of communication messages and the caused time delay. Otherwise, anyone which is intending to gain Media burst needs to send MB Request first. When getting MB Request, PoC Server will issue a MB Granted to the client issuing MB Request, and then they are authorized to begin talking, if Media-burst is available, or else MB Request is denied by transmitting a relevant message if media-burst is in use.

B. Performance analysis of ConventionalPoC technology

Suppose L is the message transmission latency time including wireless network and wired network, and T is Client waiting time. E.g. L Invite means the Invite message of transmission delay time, T Client 1 is the waiting time from pushing of PoC talk button to when caller is allowed to talk, and T Client 2 is the time period between receiving Invite message and the time in which Client-2 knows who is going to talk to himself. So from Fig. 2 we can deduce two formulas below:

$$T_{Client\ 1}= (3\times L_{Invite}+ 3\times L_{OK}+L_{ACK}) + (L_{MB\ Granted}) \quad (1)$$

$$T_{Client\ 2}= (2\times L_{OK}+2L_{ACK}) + (L_{MB\ Granted}) \quad (2)$$

$$T_{Client\ 3}= (3\times L_{OK}+3L_{ACK}) + (L_{MB\ Taken}) \quad (3)$$

Formula (1), (2) and (3) indicate the sum of message latency time considering two stages of SIP connection buildup and Media-burst control. For Formula (1) and (2), inside the first parentheses is the total message latency time in SIP connection establishment stage and the second parentheses is for the media-burst control stage. In multi-users session, each recipient receives SIP message and MB Taken synchronously without needing to accumulate the latency time of all recipients—the same as the one-recipient case. The message delay time which includes wireless and wired transmission delay depends on mobile station’s wireless signal strength and how many routers the message will pass through in the wired core network. It depends on the user’s geographic location,the position of server in the core network and its bandwidth.

In general and from former practical estimation [1], cellular network has message latency time between 300ms and 1000ms; therefore we calculate the range of client waiting time T using these two bounds. The deduced maximum waiting time for Client 1, Client 2 and Client 3are:

$$\text{Max } D_{Client\ 1}= 8\times L=8\times 1000ms=8000ms \quad (4)$$

$$\text{Max } D_{Client\ 2}= 5\times L=5\times 1000ms=5000ms \quad (5)$$

$$\text{Max } D_{Client\ 3}= 7\times L=7\times 1000ms=7000ms \quad (6)$$

Whereas the shortest waiting time for Client 1, Client2 and Client 3 are:

$$\text{Min } D_{Client\ 1}= 8\times L=8\times 300ms=2400ms \quad (7)$$

$$\text{Min } D_{Client\ 2}= 5\times L=5\times 300ms=1500ms \quad (8)$$

$$\text{Min } D_{Client\ 3}= 7\times L=7\times 300ms=2100ms \quad (9)$$

It is clear that, under very high delay time condition, thecaller cannot talk after pressing talk button until waiting about eight seconds and also the recipient Client 2 cannot know who prepares to talk until five seconds and Client 3 until 7 seconds. In this analysis, average waiting time for the sender is (8000ms + 2400ms)/2= 4.6 seconds, matching well to the practically measured average value of 4.5 seconds in [4]. Deserving more attention is that above value excludes the situation cell phone is in Sleep mode during connection establishment, in which waiting time more than 10 seconds will be necessary for caller and recipient [8]. For those PoC users who want to establish session to talk quickly, enough patience is required. And it is possible that users maybe reduce the interest of using on PoC service.
Much limited bandwidth for wireless network compared with the wired network introduces the fact that the number of messages in building up the connection has influence on not only waiting time before talking but also the inadequate bandwidth resource of wireless network. Total number of necessary messages transmitted for PoC Session buildup is determined as follows:

$$N_{message}=N_{Client} \times(M_{Invite}+ M_{ACK}+M_{OK})$$

$$+ [M_{MB\ Granted} + (N_{Client} -1) \times M_{MB\ taken}]$$

$$= 4 \times N_{Client} \quad (10)$$

Where N message signifies the total amount of messages required for PoC connection to be established, Nclient is the number of participants in the same PoC session, and M means type of message, e.g. Minvite is Invite message. From formula (9) based on Fig. 2, total number in PoC session established is four times the number of members in the session.

III. PERFORMANCE ENHANCING OF POC TECHNOLOGY

A. Performance improvement of PoC technology

Fig. 3 shows our revision of PoC message flow, inwhich amount of message is the same as Fig. 2 and most important is that all ACK messages are omitted (7.ACK, 8.ACK and 9.ACK). With ACK message, MB Granted is also omitted in that. MB Granted is not really requiredand the reason as below. PoC Server sends to Client-1 not only OK (6.OK) but also MB Granted (11.TB Granted), meaning Client-1 gains the Media-burst ; however, bidirectional transmission becomes confirmed and justified for Client-1when he receives OK message from PoC Server, and the first media burst will be obviously given to Client-1 after PoC Session connection. Both pointing out MB granted is redundant, and only to add more delay time for building connection,particularly considering the much less bandwidth of wireless network than wired core network. So it is sufficient that PoC Server to send only MB Taken to other group members confirming them talk-burst is in the hand of Client-1.

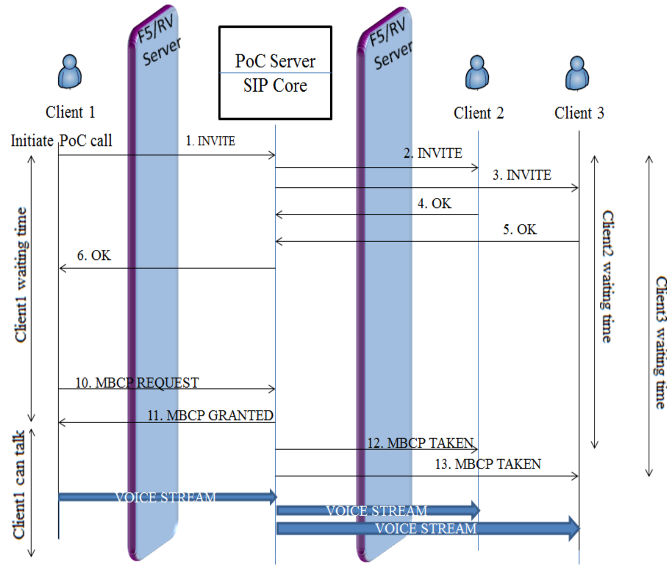

Figure 3.Enhanced PoC message flow diagram

Our aim is to reduce waiting time of PoC users without altering SIP protocol and PoC standard. SIP and PoC belongs to application layer following TCP/IP 4-layer network model, and any modification of their original operations is not our intention. Therefore, our optimised scheme is shown as Fig.4 where the amount of message is the same as Fig. 2.

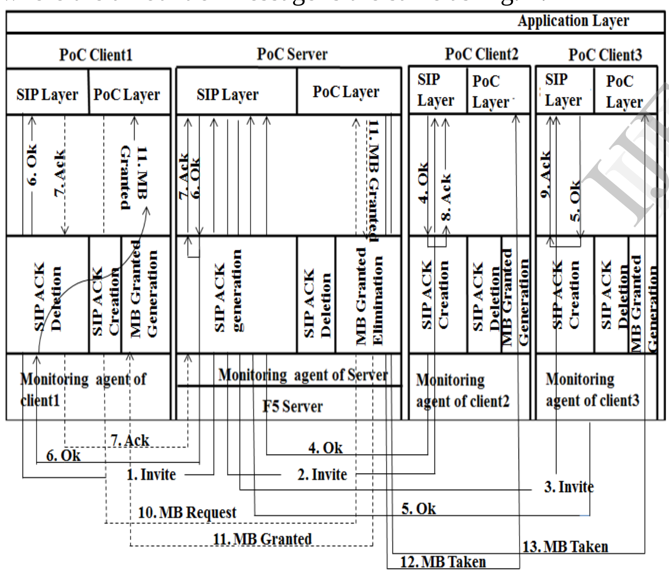

Figure 4.Application layer architecture after addition of Monitoring Agent

In the application layer for both Client and Server, wedesign different monitoring Agents monitoring in/out information of SIP and PoC. For Client, Monitoring agent isactivated after PoC talk button being pushed and stopsoperation following the end of PoC session, whereas monitoring agent of Server is always active for monitoring all sessions of SIP and PoC. Embedded in monitoring agent of Client are three components—SIP ACK Deletion, SIP ACK Creation and MB Granted Generation—and those in Server are SIP ACK Deletion, SIP ACK Creation and MB Granted Elimination.

B. PoC performance evaluation after improvement

This section holds the same assumptions as section II.B.$T_{Client\text{-}1}$ is the waiting time of Client 1 from the push of PoC talk button to the time caller can talk. $T_{Client\text{-}2}$ is the time period between Client-2 receiving Invite message and the time Client-2 knows who is talking. $T_{Client\text{-}3}$ is the time period between Client-3 receiving Invite message and the time Client-3 get to know who is talking. Three formulas are deduced from Fig. 3.

$T_{Client\ 1} = 3\times L_{Invite} + 3\times L_{OK}$ (11)

$T_{Client\ 2} = (2\times L_{OK} + L_{MB\ taken})$ (12)

$T_{Client\ 3} = (2\times L_{OK} + L_{MB\ taken}) + (L_{MB\ Taken})$ (13)

According to practical calculation transmission delay time of 300ms to 1000ms [2], we can infer that the maximum and minimum waiting time after enhanced scheme for Client-1, Client-2 and Client-3 are listed separately as below:

Max $D_{Client\ 1} = 6\times L = 6\times 1000ms = 6000ms$ (14)

Max $D_{Client\ 2} = 3\times L = 3\times 1000ms = 3000ms$ (15)

Max $D_{Client\ 3} = 4\times L = 4\times 1000ms = 4000ms$ (16)

Min $D_{Client\ 1} = 6\times L = 6\times 300ms = 1800ms$ (17)

Min $D_{Client\ 2} = 3\times L = 3\times 300ms = 900ms$ (18)

Min $D_{Client\ 3} = 4\times L = 4\times 300ms = 1200ms$ (19)

Referring to PoC session buildup process after Enhanced scheme in Fig. 3, number of required messages on wireless network is inferred as below:

$N_{message} = N_{Client} \times (M_{Invite} + M_{OK}) + (N_{Client} - 1) \times M_{MB\ taken}$

$= 3 \times N_{Client} - 1$ (20)

Performance comparison of original PoC procedureand that after improvement using enhanced scheme are shown in `Fig. 5 and table 1, in which waiting time of the caller establishing PoCsession and that of the recipient are correlated and number of messages required for establishing PoC session is distinguished. Compared with conventional PoC procedures, the caller can cut back approximately 33% waiting time in connection establishment, the Client 2 decreases the waiting time of 25% and Client 3 by 20%. Total number of messages reduced around 25%, so enhanced PoC procedure can effectively decrease required resource waste in limited wireless bandwidth.

TABLE I. NUMBER OF MESSAGES TO ESTABLISH POC CONNECTION

PoC users	Number of messages	
	Conventional PoC	Enhanced PoC
3	12	8
5	20	13
10	40	25
15	60	38
18	72	45
20	80	50

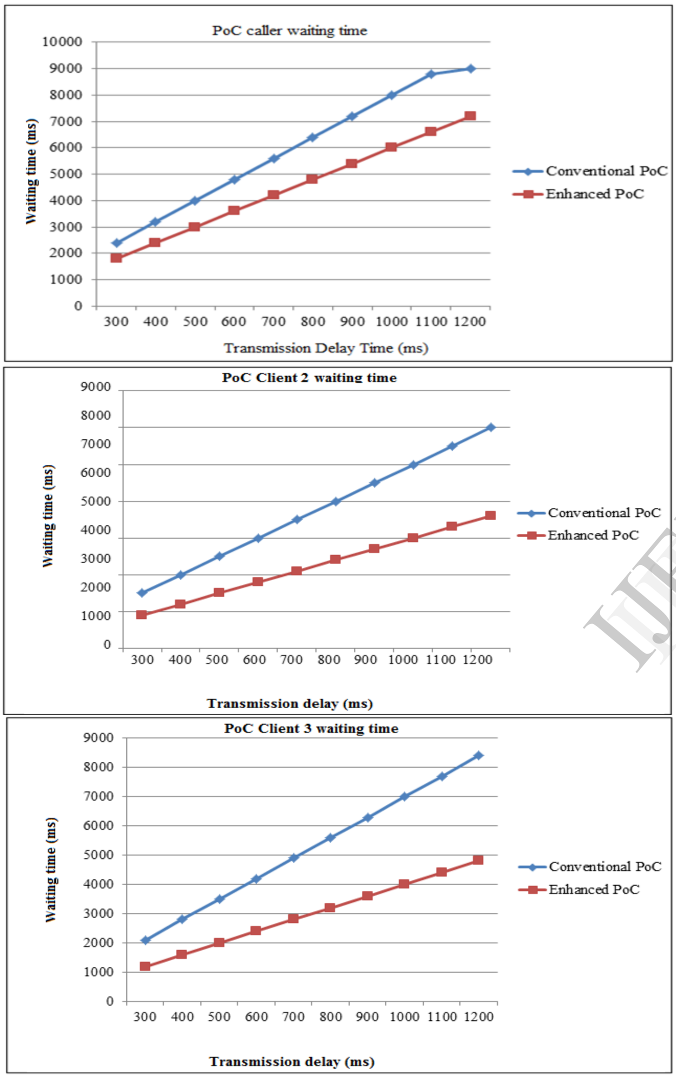

Figure 5.Comparisons of waiting time between Conventional and Enhanced PoC

IV. CONCLUSION

A value-added service in mobile telecommunication network, PoC service having many advantages of the walkie-talkie discards inconvenience of one-to-one communication and geographic distance coverage limitation. In this research, SIP connection procedures for PoC are analyzed to find which steps can be deleted in standard SIP; and Monitoring Agent is used on the application layer for more enhancements. Improved PoC provides reduced waiting time with lesser number of message transmissions. In addition, Monitoring Agent proposed here changes neither standard SIP nor PoC protocol, and offers compatibility with current PoC architecture for curtailing waiting time of session establishment.

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