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EFFICIENT ROUTING FOR INTERNATIONAL MOBILE CALL SETUP

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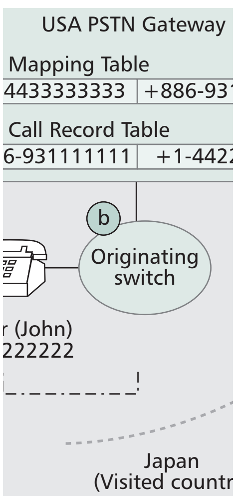


The authors propose a plug-in solution to avoid the international trunk connections. By adding Public Switched Telephone Network gateways, our solution does not need to modify the existing mobile telecommunications systems.

ABSTRACT

Mobile telecommunications operators offer telephone services when the subscribers roam to other countries. However, a call setup to a roaming mobile subscriber may result in two expensive international trunk connections. In this article we propose a plug-in solution to avoid the international trunk connections. By adding public switched telephone network gateways, our solution does not need to modify the existing mobile telecommunications systems. In this solution a timer is defined to reduce the call drop probability. A prediction method is proposed to determine the timeout period such that the call drop probability is limited within an acceptable range. Numerical experiments are conducted to indicate the performance of the prediction method. Finally, a prototype implemented at National Chiao Tung University is described.

INTRODUCTION

Global roaming of mobile telecommunications allows subscribers to receive telephone services when they travel to other countries. In telecommunications significant efforts have been devoted to optimize routing. However, call routing for international roaming mobile subscribers is definitely not optimal in the existing commercial solutions. In the standard global roaming service, a call to a roaming mobile subscriber will result in two international trunk connections, and the cost for the call is very expensive. Several commercial solutions have been proposed to provide inexpensive calls by using voice over IP (VoIP) technology [1–3]. These approaches cannot apply to mobile networks to avoid international trunk connections when subscribers with standard handsets roam to other countries.

Consider the following scenario where Jenny is a subscriber of a mobile service provider in Taiwan. She travels to Japan, and her friend John in the United States calls her. The standard roaming call setup procedure [4] results in two international trunk connections; as illustrated in Fig. 1. We briefly describe the call setup procedure as follows, and the details of signaling protocols can be referred to [5, 6].

Step A1: John dials Jenny's mobile phone number +886-931111111. The call is set up to Jenny's gateway mobile switching center (GMSC) in Taiwan through the United States and Taiwan international switching centers (ISCs); see path a → b → c → d → e in Fig. 1.

Step A2: The GMSC queries the home location Register (HLR, Fig. 1f) to identify the mobile station roaming number (MSRN) of the target mobile switching center (MSC) (Fig. 1h) visited by Jenny's mobile phone (Fig. 1j). Specifically, the HLR communicates with the visitor location register (VLR) of the target MSC (Fig. 1g) to obtain the MSRN. The signaling path is f → d → i → g → i → d → f.

Step A3: According to the MSRN (which indicates the address of the target MSC), the GMSC sets up the call to Jenny through the visited mobile network in Japan; see path e → d → i → h → j. The base station (BS) pages Jenny's mobile phone. If Jenny's mobile phone is within the radio coverage of the BS, it sends the page response signal to the BS. Jenny's mobile phone rings, and the call is connected when Jenny picks up the phone.

After the call is connected, John and Jenny start conversation through path a–b–c–d–e–d–i–h–j. The above procedure results in two international trunk connections. John pays for the international trunk connection between the United States and Taiwan (path a–b–c–d–e), and Jenny pays for the international trunk connection between Taiwan and Japan (path e–d–i–h–j). It is clear that the incoming call to a roaming mobile subscriber is expensive. Note that if John dials Jenny's mobile phone number through existing VoIP technology, only the international trunk connection between the United States and Taiwan can be avoided. Specifically, based on the dialed number, the VoIP call is set up to Jenny's GMSC in Taiwan, and then the international trunk connection from the GMSC to Jenny's mobile phone cannot be avoided.

In [5] a plug-in solution is proposed to replace international calls with local calls when both the calling party and the called party (the roaming mobile subscriber) are in the same visited country. However, the solution did not address the

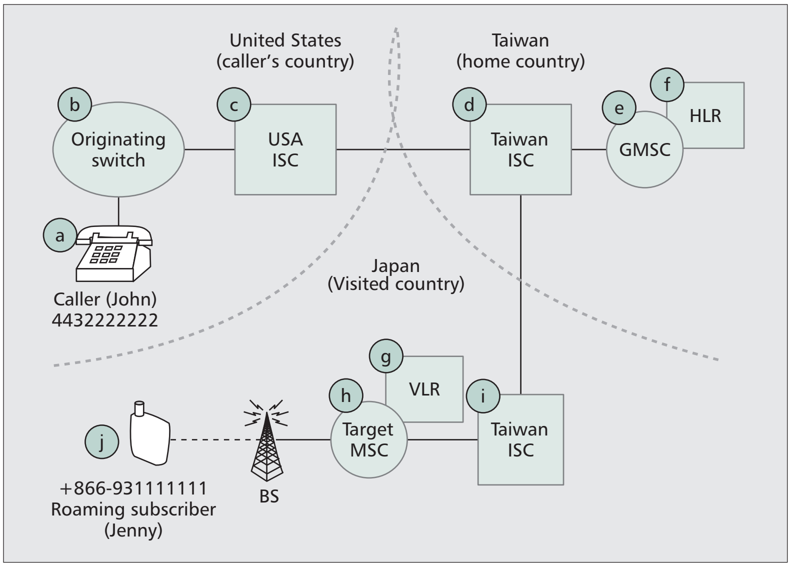


Figure 1. *International call setup in existing mobile telecommunications networks.*

issues regarding when the calling and called parties are in different countries other than the home country of the called mobile subscriber. In the next section we propose a solution called Efficient International Mobile Call Routing (EIMAR), which replaces the two international trunk connections with two local trunk connections connected by the Internet when the calling and called parties are in different countries other than the home country of the called mobile subscriber. Similar to the solution in [5], EIMAR introduces a network node called a public switched telephone network (PSTN) gateway, but does not need to modify the existing mobile telecommunications systems.

EIMAR

In this section we propose the EIMAR solution. The basic network structure of EIMAR is similar to that in [5]. We re-iterate this network structure for the reader's benefit. In the EIMAR solution, a mobile subscriber is assigned a local phone number (typically a fixed network number) for each of the countries she frequently visits. Suppose that the EIMAR solution provider is from Taiwan. A Taiwan subscriber Jenny frequently visits Japan and the USA. Therefore, Jenny has the mobile phone number +886-931111111, a Taiwan phone number, +886-5722222, a U.S. phone number, +1-4433333333, and a Japan phone number, +81-556666666. These local phone numbers are assigned by the solution provider. We note that providing local telephone numbers of multiple countries for a business person is a typical exercise now. For example, Skype provides multiple local telephone numbers to subscribers so that SkypeIn can support incoming calls from various countries [1].

The solution provider deploys a PSTN gateway [6] in each country (Fig. 2c, j, and l). A PSTN gateway can be modified from a Session Initiation Protocol (SIP) and Real-Time Transport Protocol (RTP) server [6]. Every PSTN gateway is connected to the PSTN (links c–h, j–b, and l–e in Fig. 2) and to a telecom-grade IP Network (Fig. 2k). Every PSTN gateway maintains a table that maps the local phone number (at the country where the PSTN gateway resides) to the mobile phone number of each subscriber. For example, the PSTN gateway in the United States maps Jenny's U.S. local phone number, +1-4433333333 to her mobile phone number, +886-931111111 (Fig. 2j).

The PSTN gateway also maintains a call record table. When John in the United States (with phone number +1-4432222222) calls Jenny, the PSTN gateways in the United States and the home network (i.e., Taiwan in our example) create a call record that maps the called mobile phone number to the calling phone number (+886-931111111,+1-4432222222); see Fig. 2j and 2l. Suppose that Jenny travels from Taiwan to Japan. When she arrives in Japan, she turns on her mobile phone. A small piece of roaming application software in the mobile phone switches the mobile phone to the international roaming mode, and marks the roaming country as Japan in a memory location called the *roaming country buffer*. The roaming application software can automatically obtain the roaming country identity (i.e., the country code), to be elaborated in the next section. Therefore, the implementation of the roaming application software does not need to modify the already built-in telecom software in the mobile phone. This software can be automatically downloaded to the mobile phone through over-the-air (OTA) mechanisms [7, 8]; more details are given in the next section.

When John in the USA calls Jenny, the call setup procedure works as follows.

Step B1: John (with phone number 4432222222) dials Jenny's U.S. phone number, 4433333333. The originating switch (Fig. 2b) routes the call to the local PSTN gateway (path a → b → j in Fig. 2).

Step B2: The U.S. PSTN gateway uses the number 4433333333 to retrieve Jenny's mobile phone number +886-931111111 in the mapping table (Fig. 2j). Based on this mobile phone number, the U.S. PSTN gateway sets up the call to Jenny's GMSC (Fig. 2e) through the VoIP network (Fig. 2k) and the home PSTN gateway (i.e., the Taiwan PSTN gateway in Fig. 2l). These steps are the same as those for standard Skype-Out routing [1] (path j → k → l → e in Fig. 2). The call record mapping (+886-931111111, +1-4432222222) is created in both the U.S. PSTN gateway and the Taiwan PSTN gateway (Fig. 2j and 2l).

Step B3: Following the standard mobile call setup procedure (i.e., steps A2 and A3), the GMSC forwards the Signaling System Number 7 (SS7) initial address message (`IAM`) to set up the call to Jenny's mobile phone through the visited mobile network (Fig. 2, path e → f → d → g → i → g → d → f → e for retrieving the MSRN, and path e → d → g → h → m for call setup).

Step B4: When Jenny's mobile phone is paged by the BS of the target MSC, the mobile phone first checks if the international roaming mode is active. If so, steps B4a and B4b are executed. Otherwise, the mobile phone skips these steps

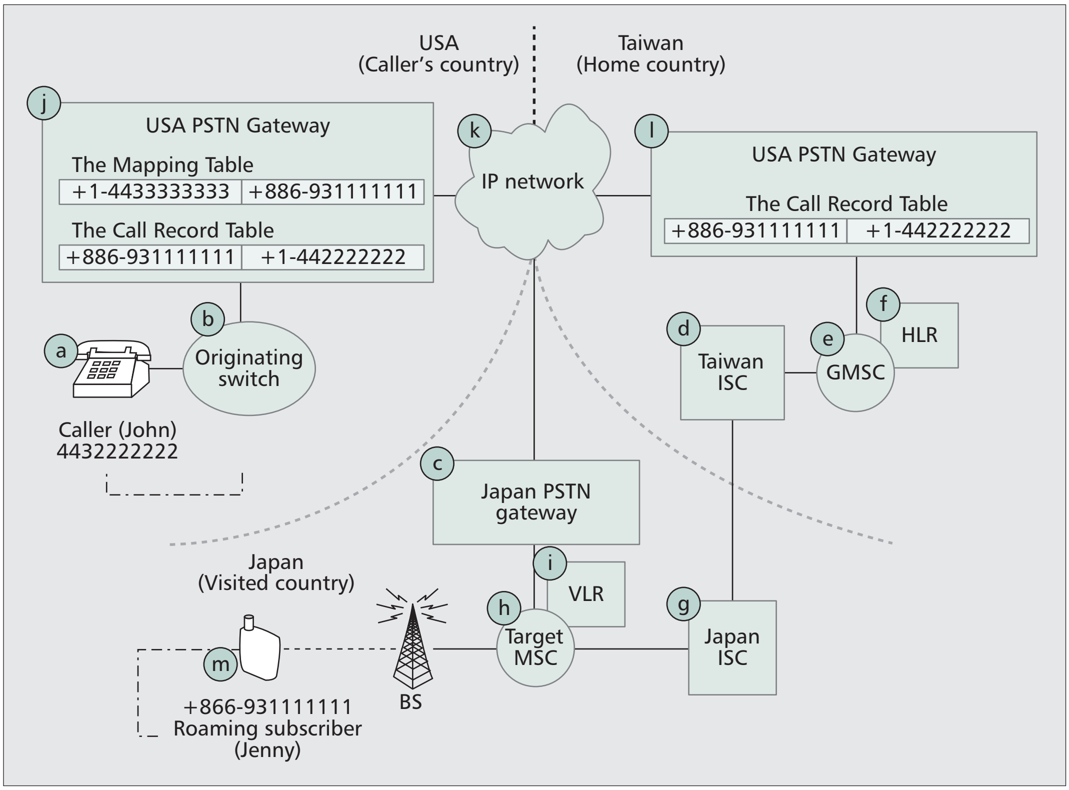


Figure 2. *International call setup in the EIMAR solution.*

Typically, the costs for local phone calls and VoIP calls are much lower than international phone calls. Therefore, the costs are significantly reduced as compared with the standard international mobile call setup procedure.

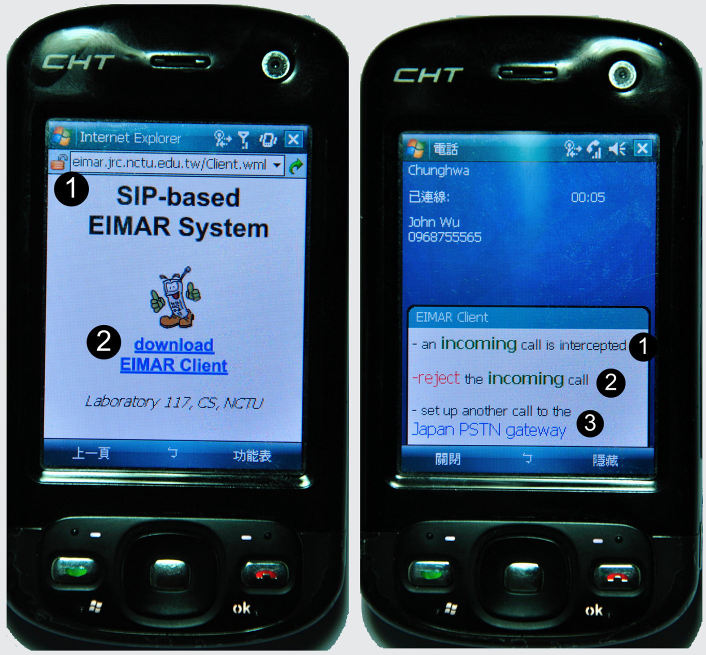
and follows the standard mobile call setup procedure:

- **Step B4a**: Jenny's mobile phone detects the international roaming mode, and automatically rejects the mobile call setup by sending the SS7 address complete message (`ACM`) with the parameter *call reject* [9]. The message is delivered through path m → h → g → d → e in Fig. 2. The GMSC forwards the SS7 `ACM`(reject) message to inform the Taiwan PSTN gateway that the call is canceled (through path e → l in Fig. 2).
- **Step B4b**: In parallel with step B4a, Jenny's mobile phone rings. From the caller ID (i.e.,+1-4432222222) carried by the `IAM` message issued at Step B3, if Jenny decides to accept this call, the mobile phone retrieves the address of the visited PSTN gateway (i.e., Japan in our example) recorded in its roaming country buffer. The mobile phone sets up the call directly to the gateway through an `IAM` message (path m → h → c in Fig. 2).

Step B5: When the Japan PSTN gateway receives the call request from Jenny's mobile phone, it checks whether there is a call record for Jenny. If so (it means that the calling party resides in Japan), the PSTN gateway bridges the call to the calling party indicated in the call record. Otherwise (i.e., the call record does not exist), from the caller ID, the Japan PSTN gateway knows that Jenny's home country is Taiwan, and routes the SIP `INVITE` message (equivalent to the `IAM` in SS7) to the Taiwan PSTN gateway (i.e., the home PSTN gateway) through the Internet path c → k → l in Fig. 2. One of the following two cases may occur in the Taiwan PSTN gateway:

- **Case 1**: The SS7 `ACM` message issued from the GMSC (step B4a) arrives at the Taiwan PSTN gateway first. The Taiwan PSTN gateway starts a callback timer. One of the following will occur:
  - –**Case 1a**: If the SIP `INVITE` (the callback request) from Jenny's mobile phone arrives at the Taiwan PSTN gateway before the callback timer expires, the gateway uses the caller ID (i.e., Jenny's mobile phone number, +886-931111111) to retrieve John's phone number from the call record table in Fig. 2l. Then the procedure proceeds to step B6.
  - –**Case 1b**: If the callback timer expires, then the Taiwan PSTN gateway clears the call record in the table, and rejects John's call request from the U.S. PSTN gateway. The U.S. PSTN gateway also clears the call record in the table. If the callback request from Jenny (step B4b) arrives after the callback timer has expired, the Taiwan PSTN gateway rejects Jenny's callback request. In this case the call is lost, and Jenny may call back again by directly dialing John's phone number.
- **Case 2**: Jenny's callback request (i.e., the SIP `INVITE` delivered at step B4b arrives at the Taiwan PSTN gateway before the call release issued from the GMSC arrives, (step B4a). The call release from the GMSC (the SS7 `ACM` message delivered at step B4a is ignored.

Step B6: When the Taiwan PSTN gateway receives the call from Jenny (the SIP `INVITE` message from the Japan PSTN gateway), it uses the caller ID (i.e., Jenny's mobile phone number, +886-931111111) to retrieve the calling phone number (i.e., +1-4432222222) from the



(a) (b)

Figure 3. *The EIMAR client: a) the download page; b) execution of step B4b.*

call record table in Fig. 2l. The country code of the calling phone number (which is *one*) indicates that the caller is from the United States. Therefore, the call is routed to the US.. PSTN gateway.

Step B7: Based on the call record, the U.S. PSTN gateway bridges call a–b–j and call m–h–c–k–j. Note that the voice path m–h–c–k–j can be different from the signaling path (i.e., m–h–c–k–l–k–j) by directly routing the RTP (voice) streams among the PSTN gateways. Such flexibility is provided by SIP.

In the above procedure, the call from John to Jenny consists of two local phone calls (paths a–b–j and m–h–c) and one VoIP call (path c–k–j). These two local phone calls follow the standard mobile call setup procedures and standard billing processes. On the other hand, the VoIP call is only issued among PSTN gateways, which does not influence the mobile billing process. Therefore, the PSTN gateways can be deployed by a third party (e.g., a VoIP operator), and the third-party solution provider can charge a reasonable rate for this service. Typically, the costs for local phone calls and VoIP calls are much lower than international phone calls. Therefore, the costs are significantly reduced compared to the standard international mobile call setup procedure (which results in two international connections).

In EIMAR a timer is defined to reduce the call drop probability. In the Appendix, we propose a prediction method to determine a reasonably short timeout period such that the call drop probability is limited within an acceptable range. Then we conduct numerical experiments to indicate the performance of the prediction method.

Implementation of EIMAR

An EIMAR prototype is implemented at National Chiao Tung University (NCTU) [10]. In this prototype the client-side application software (called the EIMAR client) is implemented for mobile phones running on the Windows Mobile operating system. The EIMAR client can be downloaded over the air, and automatically installed through a standard Wireless Application Protocol (WAP) browser [7]. To download the EIMAR client software, the mobile subscriber enters the uniform resource locator (URL) of the download page into the WAP browser (1 in Fig. 3a). The download operation is only performed once at subscription time. On the download page, a hyperlink for the EIMAR client is provided (2 in Fig. 3a). The mobile subscriber clicks the hyperlink, and then the EIMAR client software is downloaded and automatically installed.

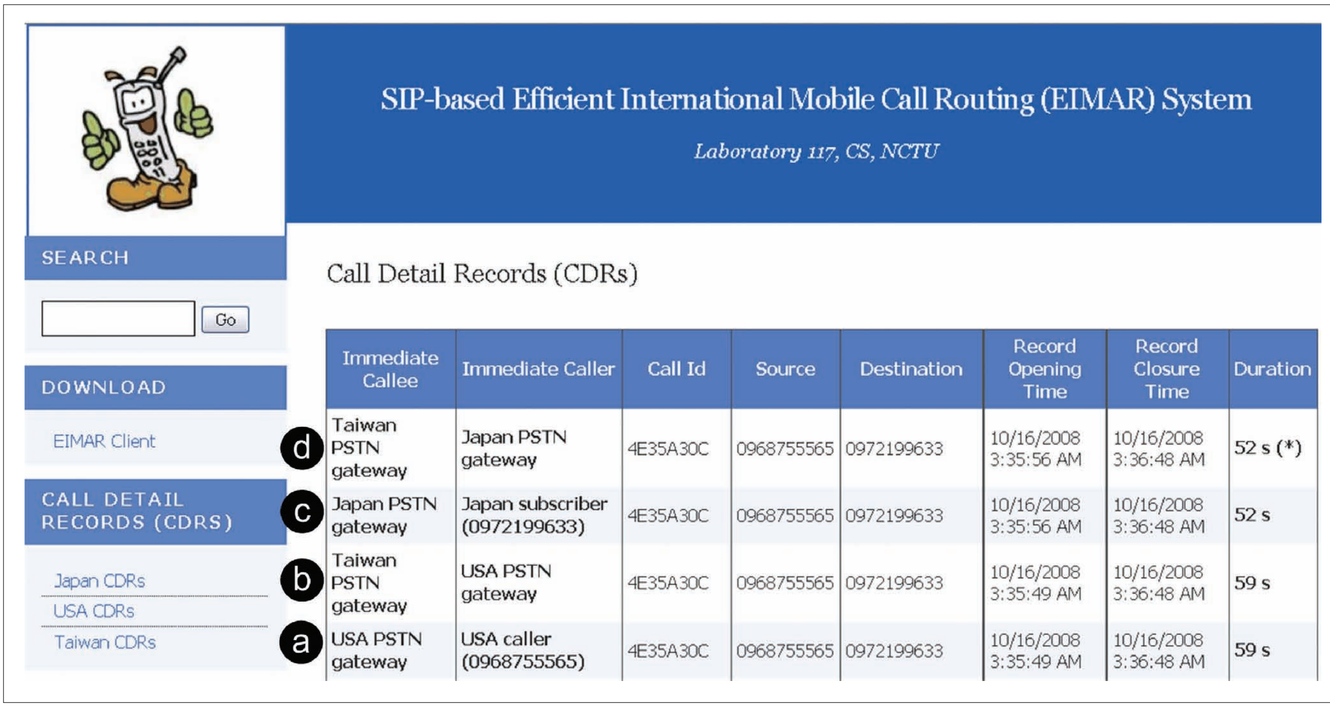
The EIMAR client checks the roaming status of the mobile phone and records the visited country code through the `lineGetCurrentOperator` function provided in the Windows Mobile Extended Telephony Application Programming Interface (TAPI) [11]. Alternatively, the roaming status can be obtained through the `AT+COPS` command of the AT command set [12] provided in a legacy Global System for Mobile Communications (GSM) handset.

To support call release and callback at step B4, the EIMAR client implements a callback function InterceptCall and configures this function into the Windows Mobile Radio Interface Layer (RIL) [13] for intercepting incoming calls (1 in Fig. 3b). This callback function utilizes `RIL_Hangup` (2 in Fig. 3b) and `RIL_Dial` (3 in Fig. 3b) to automatically reject and set up mobile calls at step B4.

In our EIMAR prototype the PSTN gateway is implemented in Java based on the BEA WebLogic SIP library [14]. The PSTN gateway implements the `processCallSetup` function (for call setup at step B2), the `processCallRelease` function (to release the call at step B5), and the `processCallbackRequest` function (to handle the callback request in the visited country at step B5 and the home country at step B6).

After a call completes, the call detail records (CDRs) of this call in the PSTN gateways are sent to an operation management center (OMC), and are shown on the administration webpage (sorted by the *Record Opening Time* field in descending order) in Fig. 4. In this figure the *Immediate Callee* and *Immediate Caller* fields indicate the immediate call parties of the CDR; the Immediate Callee field also indicates the PSTN gateway that generates the CDR. The *Call Id* field uniquely identifies a call. Four CDRs (a–d) in this figure are generated for call ID 4E35A30C; this call is set up from phone number 0968755565 (indicated in the *Source* field) to phone number 0972199633 (indicated in the *Destination* field). In the scenario described earlier, the signaling paths are partitioned into five segments:

- John dials Jenny's U.S. phone number at step B1 (Fig. 2, path a → b → j): This segment is



	Immediate Callee	Immediate Caller	Call Id	Source	Destination	Record Opening Time	Record Closure Time	Duration
d	Taiwan PSTN gateway	Japan PSTN gateway	4E35A30C	0968755565	0972199633	10/16/2008 3:35:56 AM	10/16/2008 3:36:48 AM	52 s (*)
c	Japan PSTN gateway	Japan subscriber (0972199633)	4E35A30C	0968755565	0972199633	10/16/2008 3:35:56 AM	10/16/2008 3:36:48 AM	52 s
b	Taiwan PSTN gateway	USA PSTN gateway	4E35A30C	0968755565	0972199633	10/16/2008 3:35:49 AM	10/16/2008 3:36:48 AM	59 s
a	USA PSTN gateway	USA caller (0968755565)	4E35A30C	0968755565	0972199633	10/16/2008 3:35:49 AM	10/16/2008 3:36:48 AM	59 s

Figure 4. *The EIMAR administration webpage.*

recorded by the U.S. PSTN gateway as CDR a in Fig. 4. This CDR is opened when the USA PSTN gateway receives the call setup signal from John (with phone number 0968755565, shown in the Immediate Caller field).

- The VoIP call routing between the U.S. and Taiwan PSTN gateways at step B2 (Fig. 2, path j → k → l): This segment is recorded by the Taiwan PSTN gateway as CDR b in Fig. 4. This CDR is opened when the Taiwan PSTN gateway receives the SIP `INVITE` message from the U.S. PSTN gateway.
- The standard mobile call set up at steps B2 and B3 (Fig. 2, path l → e for call setup to Jenny's GMSC, path e → f → d → g → i → g → d → f → e for retrieving the MSRN, and path e → d → g → h → m for call setup to Jenny in Japan): Since the immediate callee of this segment is not a PSTN gateway, no CDR is created for this segment in Fig. 4
- The call set up from Jenny to the Japan PSTN gateway at step B4 (Fig. 2, path m → h → c): This segment is recorded by the Japan PSTN gateway as CDR c in Fig. 4. This CDR is opened when the Japan PSTN gateway receives the callback request signal from Jenny's mobile phone (with phone number 0972199633 shown in the Immediate Caller field).
- The VoIP call routing between the Japan and Taiwan PSTN gateways at step B5 (Fig. 2, path c → k → l): This segment is recorded by the Taiwan PSTN gateway as CDR d in Fig. 4. This CDR is opened when the Taiwan PSTN gateway receives the callback request signal from the Japan PSTN gateway. At this moment, the conversation starts.

When the call completes, these four CDRs are closed. The call holding time is equal to the duration of the last CDR (see * in Fig. 4d; the call holding time is 52 s). From the CDRs collected in the OMC (Fig. 4), John pays for the local call to his mobile operator in the United States (CDR a), and Jenny pays for the local call in Japan to her mobile operator in Taiwan through a global roaming agreement (CDR c). CDRs b and d indicate a VoIP call, which is paid for by Jenny to the solution provider. It is clear that our solution saves money for both John and Jenny.

Conclusions

In this article we propose a plug-in solution, EIMAR, for reducing the costs of incoming calls to an international roaming mobile subscriber. EIMAR replaces two international trunk connections with two local trunks and one VoIP connection when the calling and called parties are in countries other than the home country of the called mobile subscriber. The costs of local phone calls and VoIP calls are much lower than those of international phone calls. Therefore, the costs are significantly reduced from those of the standard international mobile call setup procedure. In EIMAR local PSTN gateways are deployed in different countries without modifying the existing mobile telecommunications systems, so this solution can be deployed by a third party.

In EIMAR a timer is defined to reduce the call drop probability. We proposed a prediction method to determine a reasonably short timeout period so that the call drop probability is limited within an acceptable range. Numerical experiments indicate that our prediction method can appropriately predict the call drop probability. Finally, a prototype implemented at NCTU is described.

EIMAR reduces the usage of international

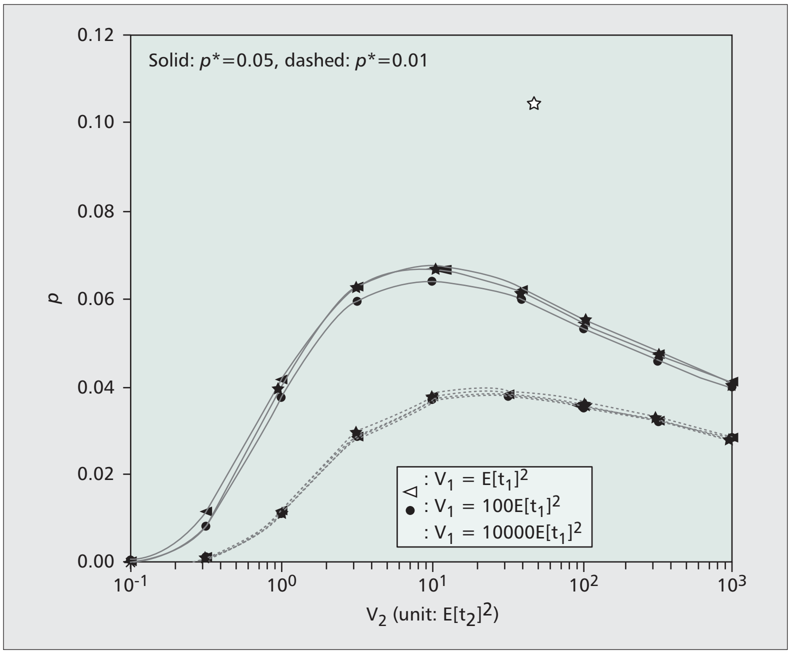


Figure 5. *Predicting p by using Eq. 1 (*$E[t_1] = E[t_2]$; t_1 *is Weibull distributed, and* t_2 *is Lognormal distributed).*

trunk resources, and therefore reduces the network (and social) costs. With this approach, roaming mobile subscribers can be charged much lower rates for incoming international calls, while the telecom operators can utilize the saved trunk resources to accommodate more real international calls.

Appendix: Selection of callback Timeout Period

In this appendix we show how to select a proper value of the callback timer. Let p be the probability that case 1b occurs at step B5 in our approach. A small p means that the call is unlikely to be dropped. In Fig. 2, let t_1 be the delay of the call release for path m → h → g → d → e → l). Let t_2 be the delay of the callback request for path m → h → c → k → l.

Let constant T be the timeout period of the callback timer in the Taiwan PSTN gateway. Then p is the probability that $t_2 > t_1 + T$. An important issue in network operation of our approach is to determine a reasonably short timeout period T such that p is limited within an acceptable range. Assume that t_1 is a Gamma random variable with the shape parameter α and the scale parameter β, and t_2 has the exponential distribution with mean $1/\lambda$. Note that the Gamma distribution is widely used in modeling telecommunications related delays [15]. Following the same analytic analysis in [5], we can predict p by using the following formula:

$$T = \left(\frac{1}{\lambda}\right)\left\{\ln\left[\frac{\beta^\alpha}{(\beta+\lambda)^\alpha p^*}\right]\right\}, \qquad (1)$$

where p^* represents the predicted value of p. During commercial operation, we would like to select a T value such that the call drop probability p is roughly equal to p^*.

By using Eq. 1, we compute T as follows. The solution provider measures the message delays t_1 and t_2 from the commercial operations, and calculates the α, β, and λ values from the measured data. The solution provider selects a p^* value, and substitutes p^*, α, β, and λ into Eq. 1 to obtain the estimated T. For example, if $\lambda = \beta$ and $\alpha = 1$, to predict that $p \approx 0.05$, we set $p^* = 0.05$ and apply Eq. 1 to yield the estimated $T = 2.3/\lambda$.

Does Eq. 1 work when t_1 is not Gamma distributed, or when t_2 is not exponentially distributed? For demonstration purpose, suppose that t_1 has the Weibull distribution with variance V_1, and t_2 has the lognormal distribution with variance V_2. Figure 5 illustrates the actual p curves where T is calculated by Eq. 1. This figure shows that the p values are roughly the same under all V_1 values. Similar results are observed when t_1 and t_2 have other distributions, which will is presented in this article.

In Fig. 5, when $p^* = 0.05$, $p > p^*$ when $E[t_2]^2 < V_2 < 200E[t_2]^2$. In this V_2 range, we need to select a smaller p^* value for Eq. 1 so that the derived T value ensures that p is within an acceptable range. Let p^+ be the maximum value of the p curve for a particular V_1 value in Fig. 5. In this figure, as V_2 increases, p increases and then decreases; p^+ occurs in the V_2 range $(E[t_2]^2, 200E[t_2]^2)$. When $p^* = 0.05$ and $V_1 = 10{,}000E[t_1]^2$, we have $p \le p^+ = 0.068$ in Fig. 5. We observe that p^+ is a roughly linear function of p^*; specifically, we have

$$p^+ = 0.695 \times p^* + 0.033.$$

By re-writing Eq. 1, we have

$$T = \left(\frac{1}{\lambda}\right)\left\{\ln\left[\frac{\beta^\alpha}{1.439\left(p^+ - 0.033\right)(\beta+\lambda)^\alpha}\right]\right\}. \qquad (2)$$

To ensure that $p \le p^+$, we select T by using Eq. 2. For example, to predict that $p \le 0.068$, we set $p^+ = 0.068$ and apply Eq. 2 to yield the T value when $E[t_2]^2 < V_2 < 200E[t_2]^2$. When V_2 is very large (i.e., $V_2 > 200E[t_2]^2$) or very small (i.e., $V_2 < E[t_2]^2$), Eq. 1 is used to compute T.

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