Modeling VoIP Call Holding Times for Telecommunications

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Abstract

Voice over IP is one of the most popular applications in broadband access networks. It is anticipated that the characteristics of call holding times (CHTs) for VoIP calls will be quite different from traditional phone calls. This article analyzes the CHTs for mobile VoIP calls based on measured data collected from commercial operation. Previous approaches directly used the Kolmogorov-Smirnov (K-S) test to derive the CHT distributions, which may cause inaccuracy. In this article we propose a new approach to derive the CHT distributions for mobile VoIP calls and other call types. Specifically, our approach uses hazard rate to select an appropriate distribution, and then utilizes the K-S test to validate our selection. We show that the mobile VoIP CHT distribution can be accurately approximated by a mix of two lognormal distributions. Based on the derived distributions, we compare the mobile VoIP CHTs with those for non-VoIP calls and fixed-network VoIP calls. Our study indicates that the characteristics for mobile VoIP calls are quite different from those of the non-VoIP mobile phone calls and are more close to those of fixed-network phone calls.

The Federal Communications Commission (FCC) defines voice over Internet Protocol (VoIP) as a technology that allows one to make voice calls using a broadband access connection (e.g., asymmetrical digital subscriber line [ADSL] and cable modem) instead of a regular (or analog) phone line. Some VoIP services may only allow a user to call other people using the same service. Other VoIP services may allow a user to call anyone who has a telephone number, including mobile and fixed network numbers. Also, while some VoIP services only work over the user's computer or a special VoIP phone, other services allow one to use a traditional phone connected to a VoIP adapter. VoIP technology has significantly changed the billing model for telecommunications. Niklas Zennstrom stated, "The telephone is a 100-year-old technology. It's time for a change. Charging for phone calls is something you did last century." A typical VoIP service provider offers free IP-to-IP calls, and charges for IP calls to the public switched telephone network (PSTN), including mobile and fixed networks.

Figure 1 illustrates the network architecture of a Session Initiation Protocol (SIP)-based commercial VoIP system deployed by Artdio [1]. This VoIP system consists of the following components.

- A SIP user agent (UA; Fig. 1a) can be a software-based VoIP phone (e.g., a Windows Messenger or Artdio Spider UA) or a hardware-based VoIP phone (e.g., CISCO 7960, Snom 200, BCM 660 WiFi phone, or Leadtek Video phone) [2].
- The SIP server (Fig. 1b), which provides SIP registrar and proxy functions, can support 50,000 subscribers with 500 concurrent Real-Time Transport Protocol (RTP) connections.
- The Remote Authentication Dial-In User Service (RADIUS) server (Fig. 1c) provides authentication, authorization, and accounting (AAA) for VoIP users. Call related information such as call holding time (CHT) is maintained in the database of this server.
- The PSTN gateway (Fig. 1d) provides VoIP-PSTN interworking between the broadband access network (Fig. 1e) and the PSTN (Fig. 1f). The PSTN gateway is a CISCO router with several E1 interfaces. Each interface supports 30 concurrent calls.

The SIP server, RADIUS server, and PSTN gateway are located at the VoIP center in Taipei. The SIP UAs connect to the SIP server through the broadband access network connections provided by domestic Internet service providers (ISPs) such as Taiwan Academic Network (TANet) and Hinet (Fig. 1 (1)). The total bandwidth between the VoIP center and other domestic ISPs is 18,144.1 Mb/s. The SIP server connects to the RADIUS server and the PSTN gateway through 100 Mb/s Ethernet lines (Fig. 1 (2) and (3)). With SIP registration function, this VoIP system supports nomadic VoIP service. That is, subscribers can move to any place with Internet connectivity in the world and attach to the Artdio system for VoIP service.

A numbering plan similar to that for Taiwan's mobile phone service is exercised. For instance, VoIP users can dial 0936000123 to reach a FAREASTONE mobile phone user. Based on the numbering plan, the destinations of calls can be distinguished and classified into several call types.

This article investigates the CHT distributions for the various VoIP call types described above. We note that exponential distribution is often used to model the CHTs for PSTN calls. Such an exponential CHT assumption has been carried over from PSTN dimensioning to VoIP bandwidth dimensioning. Thus, the Erlang models still exert influence on VoIP. Previous studies [3–5] on PSTN showed the lack of agreement between empirical data and the exponential assumption, and

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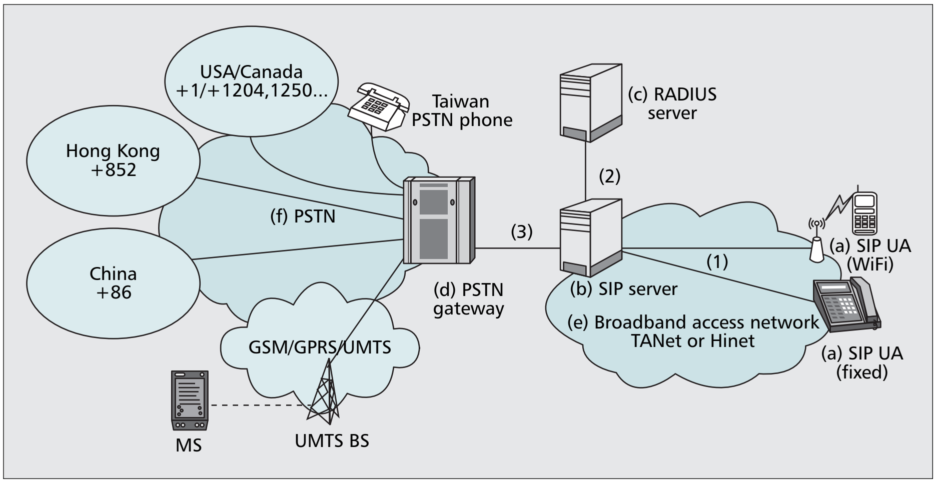


■ Figure 1. *VoIP-PSTN interworking architecture.*

network dimensioning should be conducted based on nonexponential CHTs. This article shows that VoIP CHT distributions are not exponential and are different from PSTN CHT distributions. We also note that although VoIP is an IP-based application, its CHT does not follow those for other data applications typically modeled by the Pareto distribution [6].

This article is organized as follows. We introduce call detail records for CHT collection. We describe the previous work on deriving CHT distributions. We utilize statistical tools to select specific distributions and determine the parameters that fit the measured CHT data. We analyze the characteristics for various call types.

Call Detail Records for CHT Collection

In our study the CHT information is collected using the RADIUS protocol [7]. In Fig. 1 the SIP server acts as a RADIUS client that interacts with the RADIUS server to create a call detail record (CDR) for each VoIP call. A CDR typically includes the following parameters.

The *radacctid* parameter specifies a unique record identifier (e.g., 10243), which is automatically generated by the RADIUS server. The *username* parameter specifies the calling party (e.g., 8930001). The *nasipaddress* parameter indicates the IP address of the node that requests authentication (e.g., 140.113.0.1 for the SIP server). The *acctstarttime* and *acctstoptime* parameters (e.g., 2006-06-22 09:01:32 and 2006-06-22 09:02:44, respectively) store the start and stop times of the call. The *acctsessiontime* parameter stores the CHT (e.g., 00:01:12). The *calledstationid* and *callingstationid* parameters record the telephone numbers of the called party (e.g., 886-936000123 for a mobile number in Taiwan) and the calling party (i.e., 8930001), respectively. The *acctterminatecause* parameter indicates the reason the VoIP call is terminated (e.g., user request).

Figure 2 illustrates the call setup and call release procedures where the calling party is a VoIP user and the called party is a mobile user. In the broadband access network, call control is achieved using SIP. On the other hand, the mobile network uses Signaling System Number 7 (SS7) for call control. The PSTN gateway is responsible for protocol translation between SIP and SS7.

Step 1. The calling party sends an INVITE message to the SIP server. This message includes a *Request-URI* header field designating to the called party, a *From* header field indicates the calling party (8930001), a *To* header field indicates the mobile called party (0936000123), and several authentication parameters (e.g., username, realm and nonce).

Step 2. Upon receipt of the INVITE message, the SIP server acts as a RADIUS client and sends an Access-Request message to the RADIUS server. The Access-Request message contains the authentication parameters.

Step 3. The RADIUS server retrieves the user's record from its database by using the authentication parameters. Then it replies with an Access-Accept message to the SIP server to authorize the SIP request. At this point, the *username* parameter is confirmed.

Step 4. Upon receipt of the Access-Accept message, the SIP server checks the *Request-URI*. Since 0936000123 is a mobile phone number, the SIP server forwards the INVITE message to the PSTN gateway.

Steps 5–8. The PSTN gateway generates an initial address message (IAM) and sends it to the mobile network of the called party. After the called party picks up the call, the mobile network replies with an answer message (ANM) to the PSTN gateway. The PSTN gateway generates a final response message 200 OK. This message is routed the calling party based on the reversed path of the INVITE message. This message is sent to the SIP server and then forwarded to the calling party. The calling party replies an ACK message to the PSTN gateway to confirm the receipt of the 200 OK message. The ACK message is sent to the PSTN gateway through the SIP server.

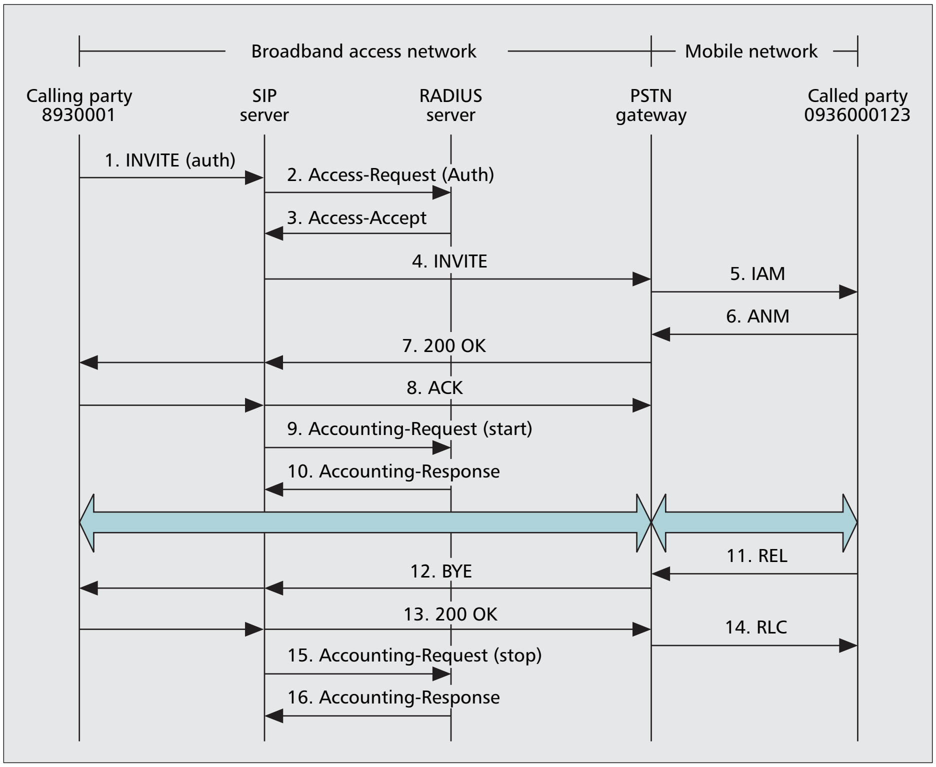
Steps 9 and 10. The SIP server sends an Accounting-Request message with status "start" to the RADIUS server to create a new CDR for this mobile VoIP call. This message includes the *username* (8930001), *nasipaddress* (140.113.0.1), *acctstarttime* (2006-06-22 09:01:32), *calledstationid* (886-936000123), and *callingstationid* (8930001) parameters. In the created CDR, the username parameter is retrieved at step 3, the *nasipaddress* parameter is the IP address of the SIP server, the *acctstarttime* parameter is retrieved from the local timer, and the *callingstationid* parameter is retrieved from the *From* header field of the Accounting-Request message. Note that the *calledstationid* parameter is retrieved from the *To* header field of the message and is inserted a prefix 886 (for Taiwan) to indicate the call type (i.e., a Taiwan-mobile call in our example). Upon receipt of the Accounting-Request message, the RADIUS server generates a new CDR in its database, fills the above parameters into the CDR and acknowledges the request message by using an Accounting-Response message containing the *radacctid* parameter (10243) to identify the CDR record.

Steps 11 and 12. Suppose that the called party hangs up the call when the conversation is complete. The mobile network sends the PSTN gateway a release (REL) message to terminate the call. The PSTN gateway then generates a BYE message and sends it to the calling party through the SIP server.

Steps 13 and 14. After the calling party terminates the call, it sends a 200 OK message to the PSTN gateway through the SIP server. The 200 OK message indicates that the call is successfully terminated at the calling party. The PSTN gateway generates a release complete (RLC) message to the called party and releases the resources reserved for this call.

Steps 15 and 16. Upon receipt of the 200 OK message, the SIP server issues an Accounting-Request message with status "stop" to the RADIUS server. This message includes the *radacctid* parameter (10243) received at step 10, the *acctstoptime* parameter (2006-06-22 09:02:44) retrieved from the local timer, and the *acctterminatecause* parameter (User-Request) that indicates the call is terminated by the user. The RADIUS server calculates the *acctsessiontime* parameter (00:01:12) and fills the parameters into the CDR. Finally, the RADIUS server replies an Accounting-Response message to the SIP server.

After call setup, all CDR parameters for this call are stored in the database of the RADIUS server. When the call is released, the CDR is considered closed and will be used for billing and other purposes. In our study, the CDRs were continuously collected through commercial operation of Artdio continuously over 24 hours and 63 days (30 November 2005 to 31 January 2006). Among the measured CDRs, there are about 20,000 records for mobile VoIP calls and 100,000 records for fixed VoIP calls.



■ Figure 2. *Message flow for call setup and call release from a broadband access network to a mobile network.*

Table 1 shows the charging rates for different call types in the investigated VoIP system. In this table the call destinations are distinguished by the prefix of the *calledstationid* parameter in the CDR. Based on the data retrieved from the *acctsessiontime* parameter of the CDRs, we derive the CHT distributions for various call types. We first utilize the *hazard rate* (also known as the hazard function) [8] to select a distribution. Then we use the *Kolmogorov-Smirnov* (K-S) test to validate if the selected distribution appropriately fits the measured data.

Previous Work on Deriving Call Holding Time Distributions

Previous studies [3, 4] utilized the *Kolmogorov-Smirnov* (K-S) test to verify whether the measured data come from a distribution with unknown parameters. Although useful insights were provided, the accuracy of their approach can be improved. For example, the K-S test can be more appropriately applied. In addition, they used the mean remaining holding time of the measured CHT data to justify whether the CHT follows an exponential distribution. This method is affected by a few outlier data and therefore may not be robust. To resolve the above issues, we derive the hazard rate of the measured data and select an appropriate distribution by observing the hazard rate curve in the next section [9].

Call destination	Prefix	Rate (US$)
USA/Canada	+1, +1204, +1250...	0.021/minute
Hong Kong	+852	0.024/minute
China	+86	0.027/minute
Taiwan (fixed)	+886-[1~8]	0.027/minute
Taiwan (mobile)	+886-9	0.0097/6 seconds

■ Table 1. *The rate plan from Taiwan VoIP to various destinations.*

In [3, 4] the K-S test is utilized to select a distribution and its parameters to fit the measured data through the following steps:

Step 1. Select several candidate distributions such as Erlang, normal, and exponential.

Step 2. For each of the selected distributions, apply maximum likelihood estimation (MLE) [10] on the measured CHT data to estimate the parameters of the distribution.

Step 3. For each distribution with parameters determined in step 2, use the measured data to calculate the distance D and the significance (confidence) level α of the K-S test, where

$$D = \varepsilon\left(\sqrt{n} + 0.12 + \frac{0.11}{\sqrt{n}}\right) \tag{1}$$

and

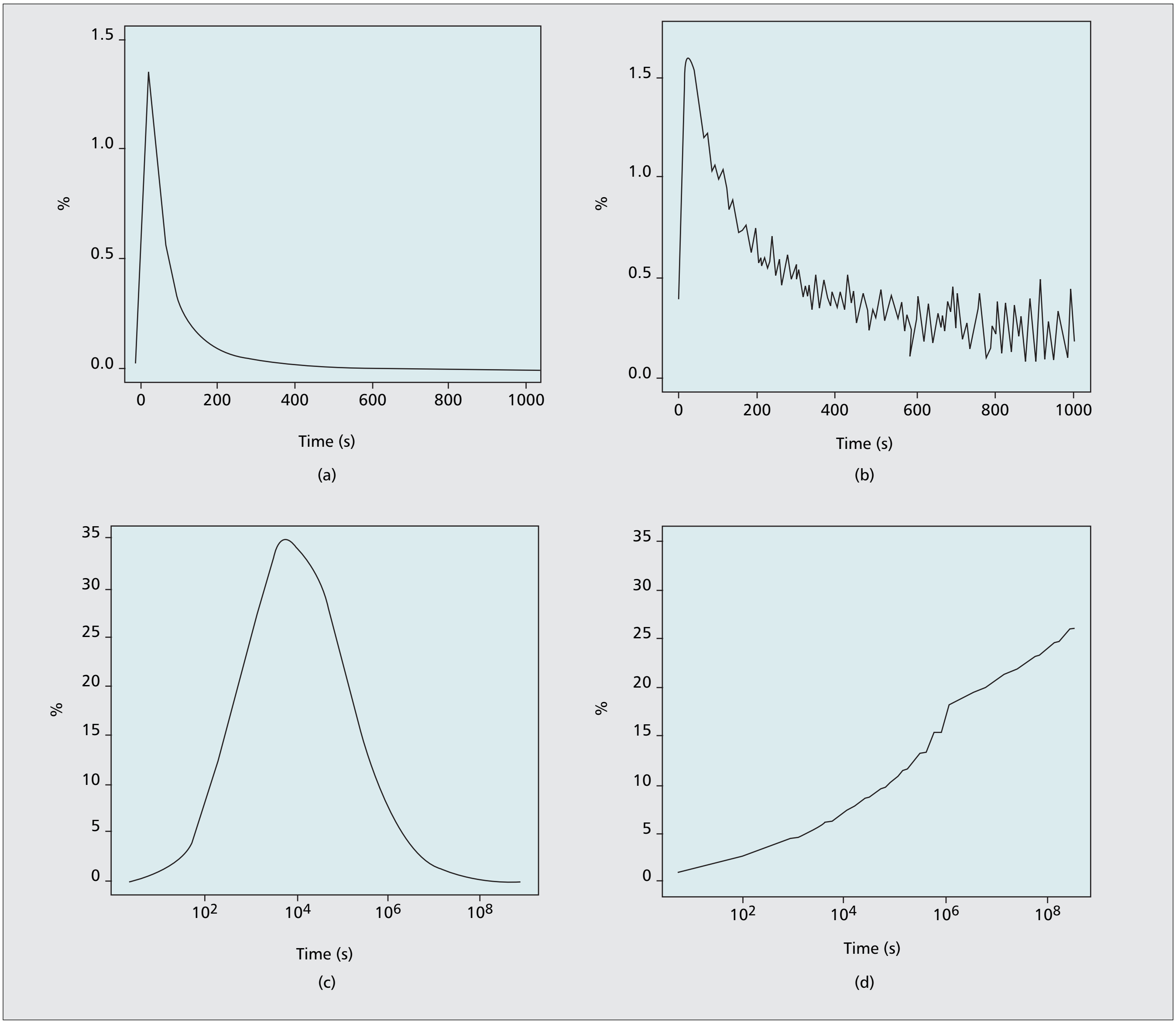
$$\alpha = 2\sum_{i=1}^{\infty}(-1)^{i-1}e^{-2i^2 n\varepsilon^2}. \tag{2}$$

In Eqs. 1 and 2, n is the number of the measured data and ε is the maximum difference between the cumulative distribution function (CDF) of the derived distribution and the measured data.

Step 4. Select the distribution with the smallest D and the largest α. This distribution is considered as the best fit to the CHT data.

The above procedure incurs two problems:

- The K-S test only applies to examining the following three cases [11–13]:
  –Whether the data set comes from a specific distribution where all parameters are known
  –Whether the data set comes from an exponential distribution with unknown mean parameter



■ Figure 3. *The CHT Histogram and the hazard rate for mobile VoIP calls in Taiwan: a) histogram (original); b) hazard rate (original); c) histogram (logarithmic); d) hazard rate (logarithmic).*

–Whether two data sets come from the same unknown distribution

The above procedure, used in [3, 4], inspected whether the data set came from a distribution with unknown parameters, which does not belong to any of the above three cases. Therefore, the K-S test cannot tell if the data set comes from the distribution selected at step 4.

- The D and α values are not suitable to identify whether the measured data come from a distribution with unknown parameters. In the real world, the data cannot be a "true" specific distribution such as normal or exponential. That is, the ε value in Eq. 1 cannot be exactly zero. In Eqs. 1 and 2, more measured data (i.e., a larger n) result in larger D and smaller α. Therefore, it is difficult to set the "threshold" values of D and α to determine if the measured data come from the distribution under test.

Based on the above discussion, it is not appropriate to use the K-S test directly to determine if the measured data come from the proposed distribution with unknown parameters. Instead, the K-S distance should be used to validate if the *already obtained* distribution can fit the measured data well after the distribution and its parameters are determined.

Deriving Mobile VoIP Call Holding Time Distribution

This section derives the call holding time (CHT) distribution for the VoIP calls to Taiwan's mobile networks (referred to as *Taiwan-mobile*). The CHT distributions for other call types (i.e., calls to Taiwan's fixed network (referred to as *Taiwan-fixed*), USA/Canada, Hong Kong and China) can be derived through the same procedure and the details are omitted. We show how to select a candidate distribution using the hazard rate, and then verify that the derived distribution fits the measured data by using the K-S test.

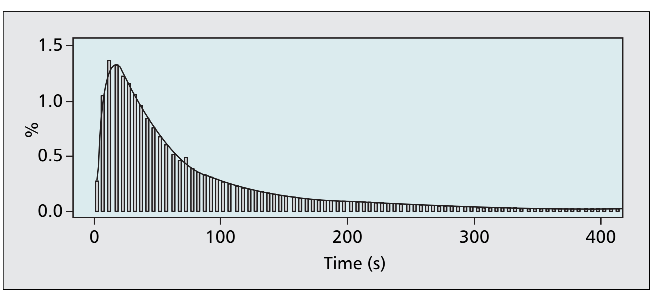
For a probability distribution with the probability distribution function (PDF) $f(t)$ and CDF $F(t)$, the hazard function is defined as

$$h(t) = \frac{f(t)}{1 - F(t)}.$$

Figure 3 illustrates the histogram and hazard rate for 20,000 measured CHT samples. The histogram in Fig. 3a is similar to

Call type	Taiwan-mobile	Taiwan-fixed	Hong Kong-mixed	China-mixed	USA/Canada-mixed
K-S distance	0.0033	0.0063	0.0101	0.0226	0.0117
p_1	0.6419	0.0349	0.1116	0.0359	0.1521
p_2	0.3581	0.4665	0.8884	0.9641	0.2640
p_3	N/A	0.4987	N/A	N/A	0.5839
μ_1	3.6067	1.2065	2.3810	2.5196	1.4795
μ_2	4.6174	3.3832	4.5534	4.5124	3.0626
μ_3	N/A	4.4937	N/A	N/A	5.3206
σ_1	0.9816	0.2906	0.7947	0.2642	0.4456
σ_2	1.1508	0.7276	0.9593	1.2355	0.9016
σ_3	N/A	1.1538	N/A	N/A	1.2993
Mean (seconds)	110	107	138	185	306
Coefficient of variance	2.07	2.16	1.47	1.89	2.83

■ Table 2. *Parameters of the derived lognormal distributions.*



■ Figure 4. *The histogram of the measured CHTs of mobile VoIP calls in Taiwan (the bars) and its fitting PDF (the curve).*

the PDF of a gamma or lognormal distribution. Figure 3b shows the hazard rate of the measured data, which does not fit the gamma distribution. From the hazard rate curve in Fig. 3b, we are not able to identify its distribution. Therefore, we explore more characteristics of the measured CHTs by investigating the logarithm of the measured data (because the measured data have a long tail). Figures 3c and 3d illustrate the histogram and hazard rate of the logarithm of the measured data (referred to as the logarithm measured data).

Figure 3c shows that the histogram of the logarithm measured data is similar to that of a normal distribution. Figure 3d shows that the hazard rate is an increasing function, which fits the hazard function of a normal distribution. Based on the above observations, we confirm that the logarithm measured data match a normal distribution, and therefore, the measured data match a lognormal distribution. In addition, we observe that the curve in Fig. 3c contains an asymmetric peak, which implies that this curve may be mixed by two normal distributions. The mixed distribution has the following PDF:

$$f(x)=\frac{p_1}{\sqrt{2\pi\sigma_1}}e^{-\frac{(x-\mu_1)^2}{2\sigma_1^2}}+\frac{p_2}{\sqrt{2\pi\sigma_2}}e^{-\frac{(x-\mu_2)^2}{2\sigma_2^2}}\ (\text{where } p_2=1-p_1).$$

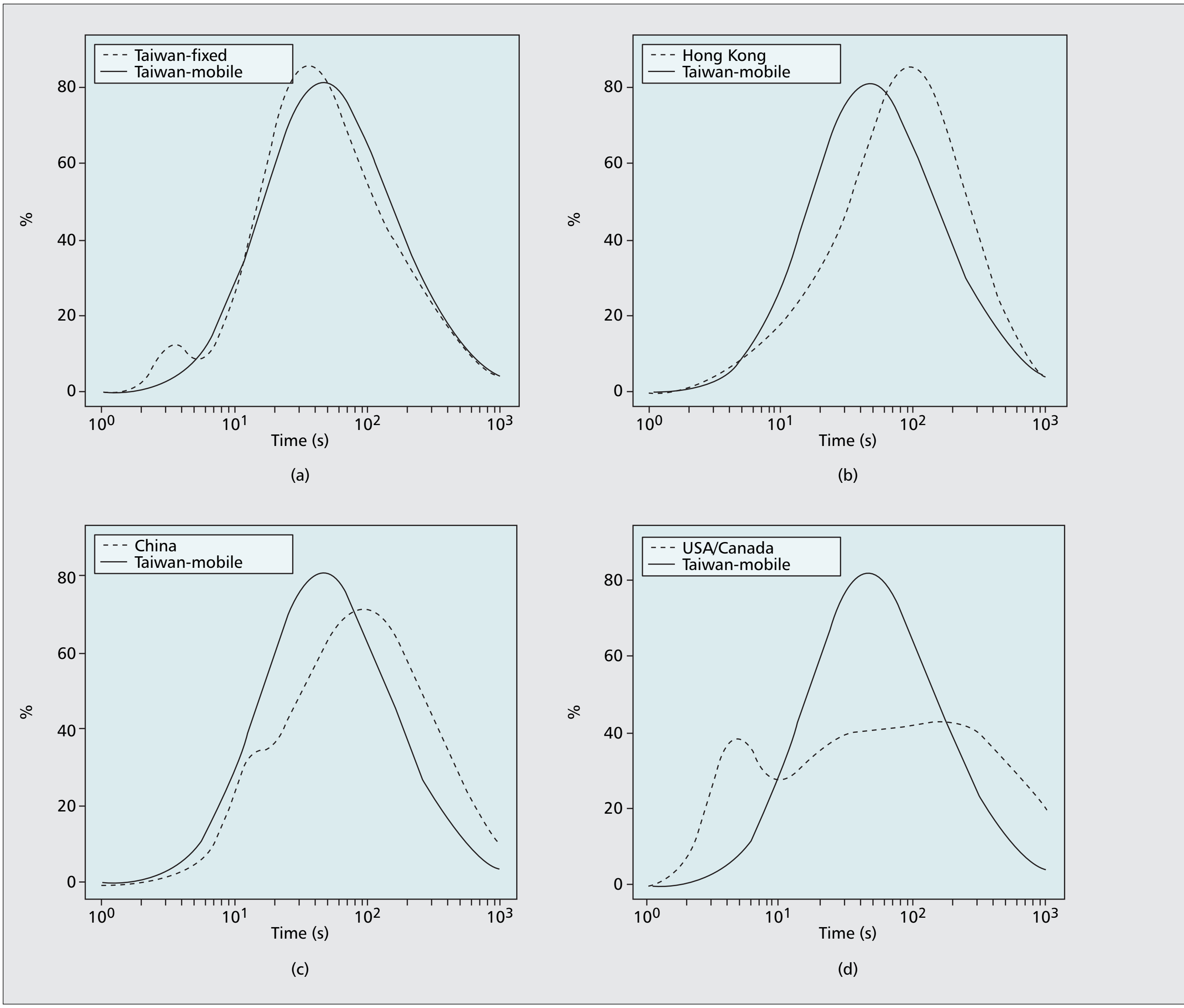
To determine the parameter values for this mixed distribution, we use the Expectation-Maximization (EM) algorithm to find the MLE [14, 15]. From the statistic tool R [16], we obtain the parameters μ_1 = 3.606694, σ_1 = 0.9816, μ_2 = 4.617415, σ_2 = 1.150835, p_1 = 0.6418951, and p_2 = 0.3581094. Therefore, the CHT distribution has a mixed lognormal PDF:

$$f(x)=\frac{0.6418951}{x\sqrt{1.9270788\pi}}e^{-\frac{(\ln x-3.606694)^2}{1.9270788}}+\frac{0.3581049}{x\sqrt{2.648844\pi}}e^{-\frac{(\ln x-4.617415)^2}{2.648844}}.$$

To verify the derived distribution, we plot the histogram for the measured data (i.e., the bars) and the derived lognormal distribution (i.e., the curve) in Fig. 4. Through the K-S test, the derived distribution has K-S distance D = 0.00325696. Since the K-S distance is short, it indicates that the derived distribution provides good approximations to the measured data.

By applying the same procedure described above, we obtain the parameters of the derived mixed lognormal distributions for other call types (Table 2). The CHT distributions for *Taiwan-mobile*, *China-mixed* (mixing of mobile and fixed calls to China), and *Hong Kong-mixed* (mixing of mobile and fixed calls to Hong Kong) are mixed from two lognormal distributions. The CHT distributions for Taiwan-fixed and *USA/Canada-mixed* (mixed mobile and fixed calls to the United States and Canada) are mixed of three lognormal distributions. Among the 120,000 CHT samples investigated, the proportions of Taiwan-mobile, Taiwan-fixed, Hong Kong-mixed, China-mixed, and USA/Canada-mixed are 17 percent, 30 percent, 2 percent, 49 percent, and 2 percent, respectively. In Table 2 the K-S distances show that the derived distributions provide good approximations of the measured data.

Based on the CHT statistics, the next section compares the characteristics among different call types.



■ Figure 5. *The PDFs of various call types: a) Taiwan-fixed vs. Taiwan-mobile; b) Hong Kong-mixed vs. Taiwan-mobile; c) China-mixed vs. Taiwan-mobile; d) USA/Canada-mixed vs. Taiwan-mobile.*

Comparison and Conclusion

Previous studies indicated that the mean CHT for non-VoIP fixed phone calls is about 3 min [17] or 113 s [5]. The non-VoIP fixed phone calls to a call center investigated in [15] are 201 s with coefficient of variance (*cv*) of 1.23. Studies on non-VoIP mobile phone calls [3, 4] indicated that the mean CHT is 40.6 s with $cv = 1.7$ during working hours and 63.3 s with $cv = 2.91$ during non-working hours. Measured data from Taiwan's mobile operators indicate that the mean CHT is 45 s [18]. Table 2 lists the means and *cv*s of the VoIP CHT distributions. Our study indicates that the mean CHT of Taiwan-mobile is 110 s (with $cv = 2.07$), which is much longer than the expected CHT of the non-VoIP mobile calls [3, 4, 18] and close to that of the non-VoIP fixed phone calls [5].

The mean CHT for all VoIP call samples measured in our study is 148 s, which is between the values measured in [15] and [5] for PSTN-based fixed network calls. Moreover, the *cv*s of all call types are larger than 1. This phenomenon indicates that the CHT distributions of all call types have high variations and cannot be approximated by the exponential distribution. We also note that most calls to Hong Kong and China are business calls, where the *cv*s are less than 2. Calls of other call types are mixed business and residential calls, and the *cv*s are larger than 2. That is, the CHTs for business calls are more regular than for residential calls. This phenomenon is also observed in [3, 4].

It is interesting to note that in the investigated VoIP system, the farther the distances to the destinations, the longer the CHTs. This is due to the fact that, compared to the charges for pure PSTN calls, more savings are expected for VoIP calls over farther distances. Also, the charges for VoIP calls to the United States and Canada are lower than those for other destinations (Table 1), which is a major reason CHTs for USA/Canada-mixed are longer than those for other destinations.

Figure 5a plots the PDFs for Taiwan-mobile and Taiwan-fixed. For CHTs of lengths ranging from 5 to 10 s and from 50 to 400 s, there are more mobile calls than fixed calls. For other CHT values, more fixed calls are observed than mobile calls. In particular, there is a peak for fixed CHT PDF at 3–4 s, which means that a group of VoIP subscribers tend to make short calls to Taiwan-fixed. It is interesting to note that although the charge for Taiwan-mobile is 3.6 times that for Taiwan-fixed (Table 1), the average CHT for Taiwan-mobile is longer than that for Taiwan-fixed (Table 2). As explained earlier, this is due to the fact that the charges of mobile calls for pure PSTN are much higher than for mobile VoIP calls. Therefore, users tend to make long mobile calls through the VoIP system. On the other hand, the charges for non-VoIP fixed calls are not much higher than for VoIP calls, and users tend to make long calls through pure PSTN for better voice quality.

Figure 5b indicates that for CHT lengths shorter than 70 s, more calls for Taiwan-mobile are observed than for Hong Kong-mixed. Figure 5c indicates that for CHT lengths shorter than 80 s, more calls for Taiwan-mobile are observed than for China-mixed. Figures 5b and 5c show that the CHT PDFs for Hong Kong and China are shifted to the right, which means that the CHT lengths for Hong Kong and China are longer than those for Taiwan-mobile. The variances of CHTs for Hong Kong-mixed and China-mixed are smaller than for Taiwan-mobile. As mentioned before, this is due to the fact that most calls to Hong Kong and China are for business purposes. Therefore, these CHTs are more regular. Figure 5d indicates that for CHT lengths ranging from 10 to 200 s, more calls for Taiwan-mobile are observed than calls for USA/Canada-mixed. The *cv* statistics in Table 2 indicate that calls to the United States and Canada are more irregular than mobile calls within Taiwan.

In summary, our study indicates that the characteristics of mobile VoIP calls are quite different from those of non-VoIP mobile phone calls and are close to those of fixed-network phone calls.

Acknowledgment

We would like to thank Editor-in-Chief Dr. Ioanis Nikolaidis and the anonymous reviewers for their valuable comments, which significantly improved the quality of this article. The work of Y.-B. Lin and W.-E. Chen was sponsored in part by NSC Excellence Project NSC 95-2752-E-009-005-PAE, NSC 95-2221-E-009-024, NTP IMS Project under grant number NSC 95-2219-E-009-019, NICI IPv6 Project, IIS/Academia Sinica, and ITRI/NCTU Joint Research Center. The work of H.-N. Hung was supported in part by the National Science Council of Taiwan under grant NSC-95-2118-M-009-004-MY2.

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