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A callback mechanism for Private Telecommunications Networks

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ABSTRACT

Private Telecommunications Networks (PTNs), such as enterprise telephony and Voice over IP (VoIP), have been widely deployed. A common limitation of PTNs is that PTN users are usually not assigned with Public Switched Telephone Network (PSTN) E.164 numbers. A PTN user is usually identified by an extension number or a URI (universal resource identifier). When a PTN user initiates a call to a PSTN user, it is not possible to use the PTN calling party's identifier as the caller ID; instead, an E.164 number of the PBX or VoIP gateway is used. As a result, the PSTN called party cannot use the caller ID received to call back to the PTN calling party later, if the PTN user's extension number or URI is unknown. This paper proposes a callback table approach to solve this problem. We describe how the call-out information is stored into and retrieved from a callback table maintained in the PBXs or VoIP gateways to provide callback service. Numeric analysis has been performed to evaluate the performance of this callback solution. The numeric results indicate that the callback method provides a satisfactory solution when there is sufficient voice traffic served by a VoIP gateway.

1. Introduction

A Private Telecommunications Network (PTN) is a telecommunications network that has its private numbering plan other than the public E.164 numbering used in the PSTN (public switched telephone network) [1]. Examples of PTN are enterprise telephone systems and Voice over IP (VoIP) networks where users are not assigned with E.164 numbers. Users of enterprise systems are identified by extension numbers, while VoIP users are often identified by URIs (universal resource identifiers), such as sip:tom@nctu.edu.tw [4]. PTNs have been widely deployed in companies and the Internet/Intranet. Since PTNs are extensions of the PSTN, it is important to connect PTNs to the PSTN. Fig. 1 illustrates an abstract PTN architecture that interconnects to the PSTN. The core component in this architecture is the Private Branch Exchange (PBX) in a telephony-based PTN, or the VoIP Gateway (VPG) in an Internet-based PTN. Example Internet-based PTNs include H.323/SIP telephony networks [2–4], MGCP/MEGACO-based networks [5,6], and Skype [7]. A phone call between a VoIP user and a PSTN user consists of two connections: a VoIP connection between the VoIP user and a VPG, and a PSTN connection between the VPG and the PSTN user. This type of calls will be referred to as IP-PSTN calls. At present, a VoIP connection is free of charge or much cheaper than a PSTN connection. The cost of a PSTN connection depends on its distance. For example, an international/long-distance call costs more than a local one.

The concept of automatic callback service was presented more than 30 years ago [8]. When user A calls user B and B is already on the phone, and if A subscribes to automatic callback service, the switching office will generate a call between A and B when B finishes the conversation. The implementation of automatic callback service becomes easier and its usage becomes popular when common channel signaling is introduced [9]. Moreover, the automatic callback can also be initiated by the called party to automatically call the calling party of the last incoming call when both parties are available [10]. Callback service can also be implemented on a user's telephone device. The current PSTN uses common channel Signaling System No. 7 (SS7) for call setup, routing and control. When setting up a call, the SS7 Initial Address Message (IAM) can deliver the caller's telephone number (i.e., the caller ID) to the called party [11]. The caller ID can be automatically stored in the called party's telephone device (e.g., a mobile phone with an address book). The called party can then call back without manually dialing the caller's telephone number.

A limitation of PTNs is that from the viewpoint of the PSTN, a PTN user without an E.164 number does not have a caller ID. When a PTN user initiates a call to a PSTN user, the PTN user's identity cannot be carried by the SS7 IAM. Instead, the PBX or VPG's telephone number is used as the caller ID. Consequently, the PSTN user may not utilize the callback service to reach the PTN user later. This limitation may cause frustration to a PSTN user called by a PTN user, especially if the PSTN user is not available for the IP-PSTN call.

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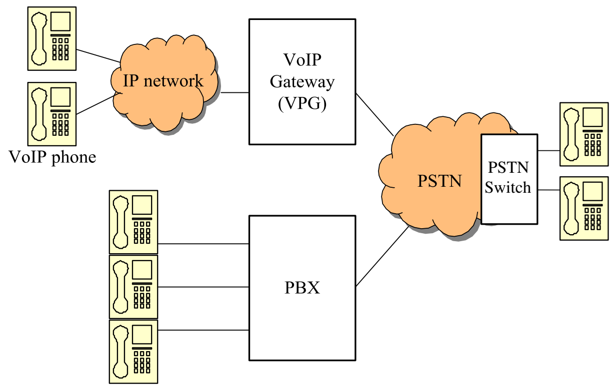


Fig. 1. Private Telecommunications Network architecture.

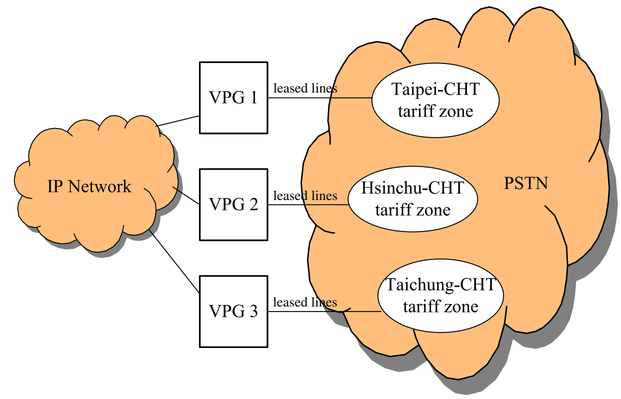


Fig. 2. The PSTN is partitioned by VPGs.

The PSTN user will not be able to call back based on the caller ID received, because he or she does not even know who made the call previously. When the PSTN user does call back using the caller ID of the missed call, the user will be connected to the PBX/VPG, but he or she is unable to reach the PTN user because the PTN user's name or number is unknown. Therefore, when a PBX/VPG dials out for a PTN call origination, the caller ID is usually blocked out, in order not to causing confusion to the called party. In this case, unanswered calls cannot be returned and important calls may be lost.

Another limitation of VoIP service is that most VoIP users, such Skype users, are usually not assigned with E.164 numbers, or even extension numbers. As a result, although it is easy for a Skype user to initiate a call to a PSTN user, it is not possible for a PSTN user to call a Skype user unless the Skype user subscribes SkypeIn service to obtain an E.164 number.

If a PTN user does have a public E.164 number, one-stage dialing is possible. A well-known example is the Direct Inward Dialing (DID) service provided by PSTN operators. When a PSTN user makes a call to a PTN user by dialing the PTN user's DID number, which is an E.164 number. The call will be routed to a designated PBX or VPG along with a signaling indicating the DID number dialed. The signaling can be an ISUP IAM, or a Q.931 setup [12]. The PBX or VPG retrieves the DID number dialed without picking up the call, and forwards the call to the PTN user called. However, if a PTN user has no public E.164 number, no one-stage dialing scheme has ever been presented.

To resolve the above issues, this paper presents a mechanism to allow a calling party outside a PTN to call back to a user within the PTN without knowing the PTN user's extension number or URI through one-stage dialing. In our solution, after calling out to a PSTN user, a PTN user, such as a Skype user without a SkypeIn number, can be called back by the PSTN user.

Without loss of generality, in this paper we consider the Internet-based PTNs only; the work can also be applied to the enterprise telephony. This paper is organized as follows. The traditional IP-PSTN call flow is presented in Section 2. Section 3 describes the PTN callback mechanism. Section 4 evaluates the performance of the callback design. Numeric results are discussed in Section 5, and conclusions are given in Section 6.

2. The traditional PTN call process

To reduce the cost of serving an IP-PSTN call, a VoIP service provider may deploy a large number of VPGs, each of which is located at a tariff zone of a local exchange carrier (LEC). Fig. 2 depicts an example of three VPGs located at three different tariff zones of Chunghwa Telecom (CHT), Taiwan. To minimize the cost of an IP-PSTN call, the PSTN user should be connected to the VPG at the same tariff zone, i.e. the shortest PSTN connection is chosen. In this way, when setting up IP-PSTN calls, the PSTN is partitioned into small tariff zones, each of which is served by a VPG at the tariff zone. The number of leased lines needed at each VPG can be determined by the voice traffic served by the VPG with the constraint that the call blocking probability at busy hours should be kept below 1%. This is a common practice for telecommunication operators.

Fig. 3 depicts the traditional call flow between a VoIP user and a PSTN user. When a VoIP user attempts to make a call to a PSTN user (this procedure will be referred to as VoIP call origination), the call is first set up to the target VPG that is located at the same tariff zone as the called PSTN user (Step 1) using VoIP signaling, such as Session Initiation Protocol (SIP), H.323, or Media Gateway Control Protocol (MGCP/MEGACO). The VPG then chooses an available leased line from the leased line pool (Step 2), and connects the call to the PSTN switches (Step 3) using SS7 signaling messages.

VoIP users, such as Skype users, are usually not assigned with E.164 numbers. If a VoIP user does not have an E.164 number, the user cannot be reached directly from the PSTN. When a PSTN user attempts to call a VoIP user (this procedure will be referred to as VoIP call termination), the PSTN user first dials a telephone number of the VPG (i.e., the number is one of the leased lines owned by the VPG). Then, the PSTN switch serving the PSTN caller sets up the call to the VPG using SS7 signaling (Step 4). After receiving the call, the VPG requests the PSTN caller (e.g., through an Interactive Voice Response (IVR) system) to input the extension number of the called VoIP user (Steps 5–7). After the PSTN caller inputs the extension number (e.g., through DTMF dialing [13]), the VPG sets up the call to the called VoIP user through VoIP protocols (Step 8). In this VoIP call termination procedure, the VoIP user is connected indirectly through two-stage dialing. The PSTN user must first dial the number of the VPG, and then dials the extension number of the called VoIP user. If a VoIP user is not assigned to an extension number, the user may not be reached from the PSTN. For example, it is not possible for a PSTN user to call a Skype user if the Skype user does not subscribe SkypeIn service (i.e, is not assigned to an E.164 number), because the Skype user is not assigned to an extension number.

3. The callback mechanism for PTNs

To enable PSTN users to call back to a VoIP user through one-stage dialing, we design a callback mechanism by introducing a callback table in the VPG. Fig. 4 illustrates an abstract VPG architecture with a callback table. The callback table stores callback tags. Each callback tag represents a record of a VoIP call

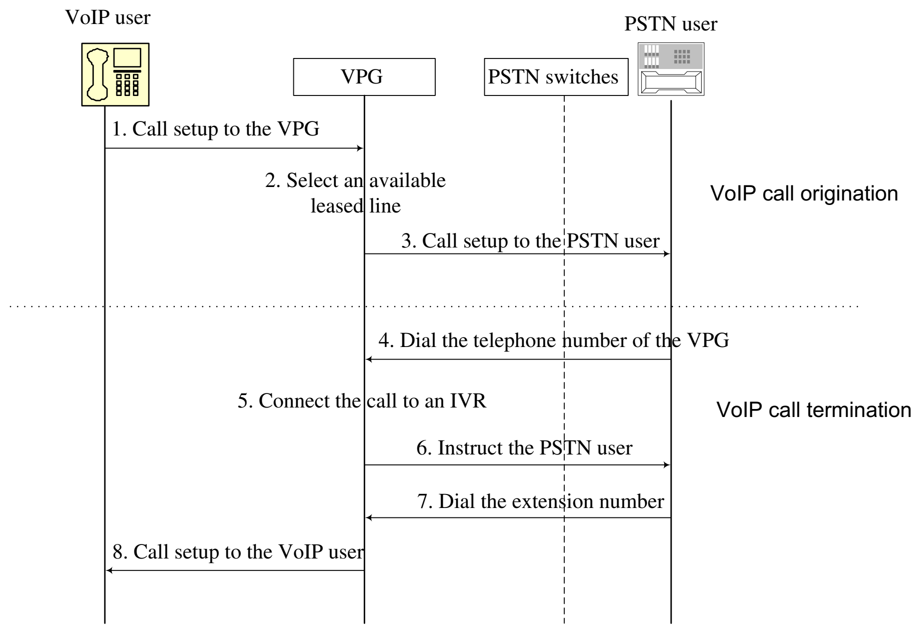
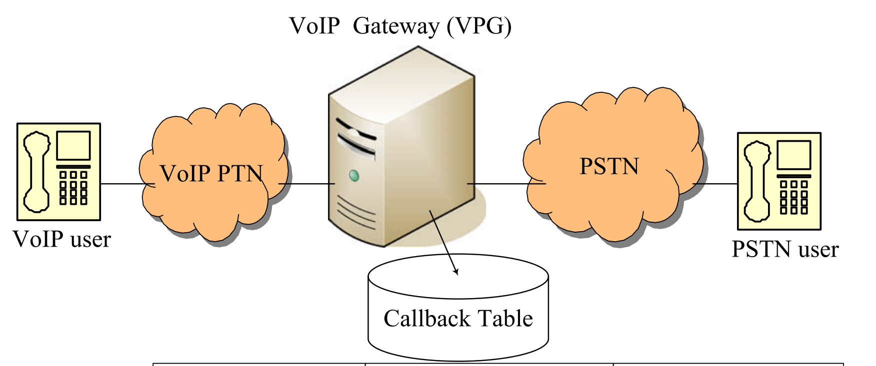


Fig. 3. The traditional call flow between a VoIP user and a PSTN user.



VoIP caller's ID	PSTN called party's tel. no.	Leased line no.
11	5752121	5720250
12	5755476	5720251

Fig. 4. A VPG with a callback table.

origination. A callback tag consists of three fields: the VoIP caller's identifier (the user's private phone number, SIP URI, or other ID representation), the PSTN called party's telephone number, and the leased line used for the VoIP call origination. A callback tag is created when a VoIP user calls a PSTN user for the first time.

Fig. 5 depicts how a VoIP call origination and a VoIP call termination work using our callback mechanism. When a VoIP user initiates a call to a PSTN user, the VPG uses the data pair (the VoIP user identifier, the PSTN called party's number) as the key to search the callback table (Step 2). If there is a matched tag, meaning that the VoIP user has called the PSTN user before, the leased line in the matched record is used to set up the call (Step 3), i.e., the same leased line used before is used again. If no matched tag is found, i.e., the VoIP user calls the PSTN user for the first time, the VPG tries to find a fresh leased line to set up the call, and inserts a callback tag (the VoIP user identifier, the called party's number, the leased line selected) in the callback table. Note that a fresh leased line is a line that has not been used by other VoIP users to call the same PSTN user. Assume that the number of leased lines on the VPG is L. If the PSTN user has been called by more than L different VoIP users, no fresh leased line can be found. This problem will be referred to as the leased line collision problem. When this happens, the call cannot be served by this VPG without deleting an old callback tag. In this situation, we route the call to a nearby VPG, and the new VPG tries to connect the call in the same way. Since a VPG at a different tariff zone is used, the call may be charged at a higher rate.

Note that in Step 2 the leased line selected may be busy, and thus the line cannot be used to call-out to the PSTN if analog signals are used between the VPG and PSTN. In our design, we assume that SS7 or Q.931 is used between the VPG and the PSTN. In this case, there is no fixed binding between a circuit and a telephone number. As a result, even when an ongoing VPG originating call is using a certain caller ID, the caller ID can still be carried in another VPG originating call request (IAM for SS7 or CONNECT for Q.931).

For a VoIP call termination, when the VPG receives a call from the PSTN (Step 4 in Fig. 5), the VPG also receives the PSTN calling party's number and the called leased line number. The VPG uses

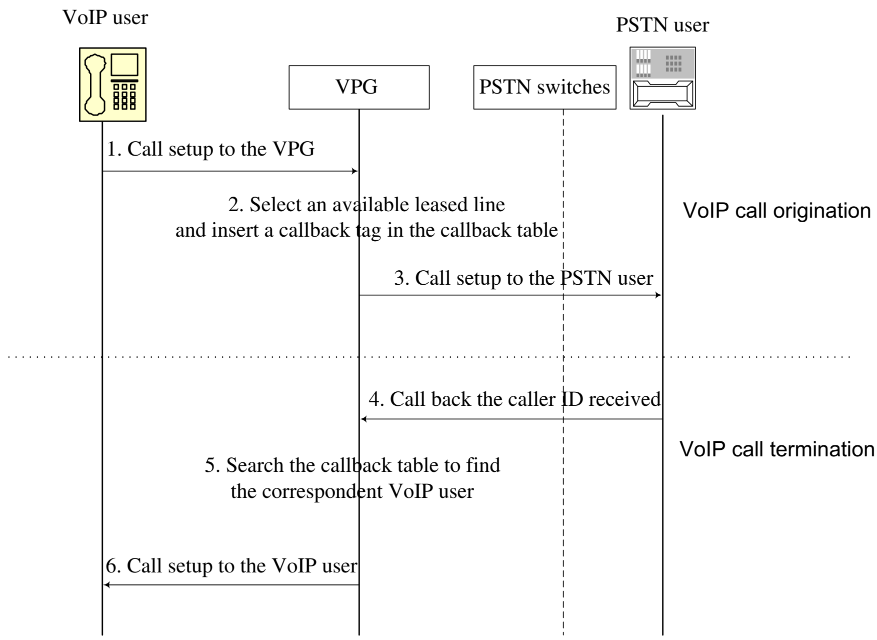


Fig. 5. The call flow of our callback mechanism.

the data pair (the PSTN calling party's number, the leased line called) as the key to search the callback table (Step 5). If a matched callback tag is found, the VPG sets up the call to the VoIP user stored in the tag (Step 6). If no matched callback tag is found, no VoIP user has used the leased line to call this PSTN user, and thus no callback service can be provided. The call can be routed to an IVR for further assistance. In this way, a PSTN user can call back to a VoIP user by one-stage dialing. Moreover, the call is charged as a local phone call, since the VPG and the PSTN caller are located at the same tariff zone. By contrast, when a PSTN user calls a SkypeIN number, the call is often a long-distance or even an international call. However, if an incoming call to the VPG from the PSTN blocks out its caller ID delivery, our callback service cannot work. In this case, the call can be routed to an IVR for further information.

The design of our callback mechanism originated from NAT (Network Address Translation) [14]. The way how a callback table works is very similar to that of a NAT mapping table. A callback tag needs to be inserted in the callback table first, before a PSTN user can call back a VoIP user through one-stage dialing. In addition, in a VPG, many VoIP users can share a leased line number just as, in a NAT gateway, many nodes with private IP addresses share a public IP address.

4. The analytic model

As we have described, when a leased line collision occurs, the incoming IP-PSTN call has to be re-routed to a nearby VPG. As a result, the call will be charged at a higher rate since the nearby VPG and the PSTN user are not at the same tariff zone. If leased line collisions occur often, the cost-saving of our callback service would be less attractive to users. One way to deal with the leased line collision problem is to increase the number of leased lines of the VPG. However, this would increase the operating cost of the VPG. The number of leased lines of a VPG is usually determined by its serving voice traffic and a desired call blocking probability at busy hours. A performance metric of our solution is the probability of leased line collision on a VoIP call origination. We study this probability when the number of leased lines is determined in the aforementioned way.

When setting up IP-PSTN calls, the PSTN is partitioned by the VPGs. In our analysis, we consider a VPG and the PSTN users served by the VPG. Let L denote the number of leased lines at the VPG, and G denote the number of PSTN users served by the VPG. In addition, let k denote the total number of VoIP users calling a specific PSTN user; k is assumed to be normally distributed with mean $\overline{K}$ and standard deviation σ. The calls, originating and terminating, between a PSTN user and one of the PSTN user's correspondent VoIP users form a Poisson process with rate λ at busy hours. The service time of a voice call is assumed to be exponentially distributed with mean $1/\mu$ seconds. Let P_B denote the fraction of time that all leased lines on the VPG are busy at busy hours, i.e., the probability that an incoming call to the VPG is blocked. Using Erlang's B formula in Eq. (1), we can obtain P_B, where E_b is the Erlang B value and can be obtained from Eq. (2). To provide an acceptable call blocking probability ($P_B < 1\%$), L has to be large enough, and its value can be determined from Eq. (1).

$$P_B = \frac{(E_b)^L/L!}{\sum_{i=0}^{L}(E_b)^i/i!} \tag{1}$$

$$E_b = \frac{(\overline{K}G\lambda)/\mu}{3600} \tag{2}$$

When a PSTN user has received calls from more than L different VoIP users and receives a call from a new VoIP user, the VPG encounters a leased line collision problem because all L leased lines have been used. In this case, the call is re-routed to a nearby VPG. Let P_r denote the probability that a call from a VoIP user to a PSTN user encounters a leased line collision problem and needs to be re-routed. We assume that the calls to a PSTN user are uniformly distributed among the user's correspondent k VoIP users. As a result, P_r equals to $E(k-L, \forall k > L)/\overline{K}$, and after corrected for continuity, it can be expressed in Eq. (3).

$$P_r = \frac{\sum_{i=1}^{\infty} i \times \left(prob\left(k > L + i - \frac{1}{2}\right) - prob\left(k \geqslant L + i + \frac{1}{2}\right)\right)}{\overline{K}} \tag{3}$$

Since the VPGs are the required components to provide interconnection functions between VoIP users and the PSTN, the feasibility of our callback mechanism depends on the storage and computation overhead on the VPGs needed to support the callback service. For each PSTN user served by a VTG, the VTG keeps at most L callback tags. The total number of callback tags is less than $L \times G$, and the size of the callback table is $O(L \times G)$. To serve a PTN call origination, the VTG needs to search the callback table with key pair (the VoIP user identifier, the PSTN called party's number); to serve a VoIP call termination, the VTG searches with key pair (the PSTN calling party's number, the leased line called). Therefore, the callback table can be sorted according to the PSTN telephone numbers of the callback tags. In this way, the search and the insertion of a callback tag can be done in $O(ln(L \times G))$. Given the cost of memory and the computation power of processors at the present time, the storage and computation overhead is easily manageable.

5. The numeric results

In general, G, the number of users in a PSTN tariff zone, can be as large as millions. When G is large, the number of leased lines required on the VPG (L) is also large. As a result, the leased line collision problems do not occur because L is larger than k. To study the leased line collision problems, in our experiments G varies in the range of 1000–16,000. For each PSTN user, the number of correspondent VoIP users, k, is assumed to be normally distributed. It is clear that as k increases the leased line collisions occurs more often. However, its mean value, $\overline{K}$, is difficult to determine because no such data have been collected. In our experiments, $\overline{K}$ varies in the range of 50–300 and the standard deviation equals to $\frac{1}{4}\overline{K}$. Considering the total number of different VoIP users calling the same PSTN user, we believe an average value of 300 is large enough to represent the worst case realistic situation. The mean service time of a call ($1/\mu$) is assumed to be 60 s, and the arrival rate of voice calls (λ) between two correspondent VoIP and PSTN users varies in the range of 0.005–0.025 calls/h at busy hours.

The number of leased lines required on a VPG can be determined from Eq. (1); L is chosen to be large enough so that $P_B \leqslant 1\%$. Fig. 6 depicts the number of leased lines (L) required on the VPG. The arrival rate of voice calls (λ) is fixed at 0.012 calls/h, i.e., the expected number of IP-PSTN calls to and from a PSTN user is 0.6–3.6 calls/h ($\overline{K}\lambda$) during busy hours. The results indicate that L increases almost linearly as $\overline{K}$ and G increase, i.e., a larger $\overline{K}$ or a lager G indeed needs more leased lines. Note that L equals to $\overline{K}$ when G is in the range of 4000–5000. When the number of correspondent VoIP users of a PSTN user (k) is less than L, the PSTN user's correspondent VoIP users experience no leased line collision. This suggests that the leased line collisions are very common when G is less than 4000.

Fig. 7 depicts the effects of $\overline{K}$ and G on P_r, the probability that a VoIP originating call needs to be re-routed to a nearby VPG, i.e., the call encounters a leased line collision problem. The arrival rate of voice calls (λ) is fixed at 0.012 calls/h. The results indicate that P_r

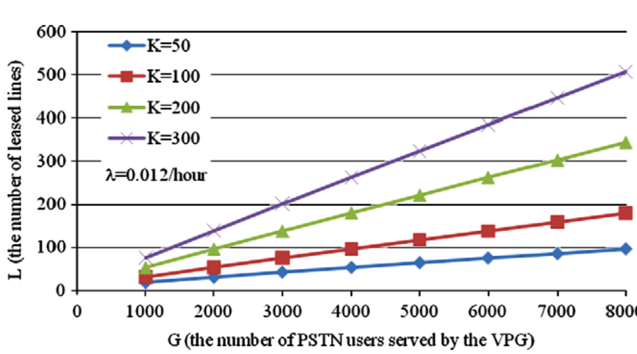


Fig. 6. The number of leased lines required on a VPG when $\lambda = 0.012$/h.

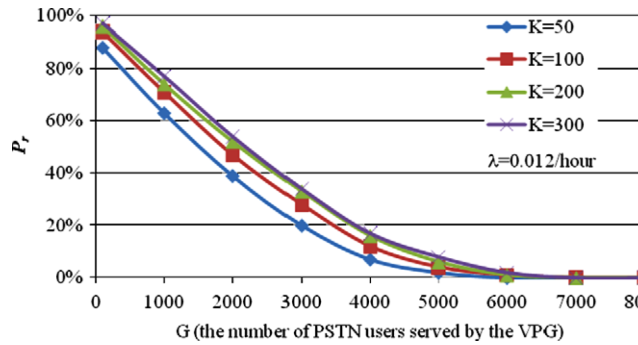


Fig. 7. The probabilities of leased line collisions when $\lambda = 0.012$/h.

increases as $\overline{K}$ increases, i.e., a larger $\overline{K}$ causes more leased line collisions. However, the difference between the cases $\overline{K} = 200$ and 300 is very small. On the other hand, P_r decreases as G increase; P_r drops nearly to 0% when G is larger than 7000. Since G, the number of PSTN user at a tariff zone, is in general much larger than 7000, we conclude that leased line collision problems seldom occur if λ is more than 0.012 calls/h.

However, at the initial stage after the VPGs are installed, the users may call less often, i.e., the call arrival rates are small. Therefore, we studied the effects of call arrival rates on P_r. Note that at this stage, k tends to be small too, because the service is not popular yet. The average number of correspondent VoIP users of each PSTN user, $\overline{K}$, is assumed to be 100 to represent the average number of active contacts in one's phone book. Figs. 8 and 9 depict our experiment results. Fig. 8 depicts the number of leased lines, L, required on a VPG when $\overline{K}$ is fixed at 100. The results indicate that L

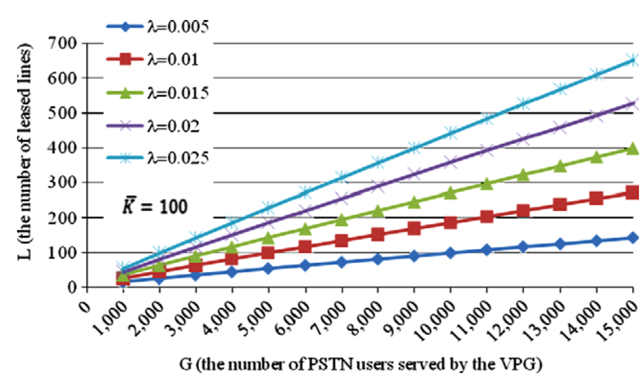


Fig. 8. The number of leased lines required on the VPG when $\overline{K} = 100$.

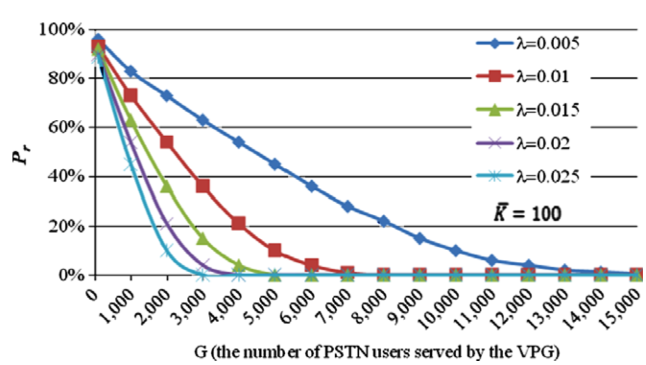


Fig. 9. The probabilities of leased line collisions when $\overline{K} = 100$.

increases as λ and G increase. Note that when the call arrival rate is as low as 0.005 calls/h, G has to be as large as 11,000 so that L equals to $\overline{K}$, and thus P_r is expected to drop rapidly as G increases. However, when λ is high (e.g., 0.025 calls/h), the transition point is when G equals to 2500.

The results in Fig. 9 indicate that when the call arrival rate is low, we need more PSTN users to bring P_r down. For example, when the call arrival rate is 0.005 calls/h, G has to be larger than 15,000 so that P_r is close to 0%. On the other hand, when λ is 0.025 calls/h, the transition point is when G equals to 2500. The results confirm what we observed in Fig. 8. However, at the initial stage after the VPGs are installed, we expect neither λ nor G is large. Therefore, the system has to be built-ahead with capacity buffer for future expansion and for reducing the leased line collision problems.

6. Conclusions

Since PTN users are usually not assigned with E.164 numbers, it is difficult to provide callback service to PTN users. This paper investigates this limitation issue. Based on VoIP protocols and PSTN signaling, we propose a callback table approach to resolve this problem. We describe how the call-out information of PTN users is stored into and retrieved from the callback table on VoIP gateways or PBXs to provide callback service. PTN users can be called back by the PSTN users through one-stage dialing. This is a plug-in solution that does not affect the existing call setup message flow or users' dialing behavior.

An analytic model has been developed to evaluate the leased line collision problem. The numeric results are consistent with real world experience. The results indicate that our callback mechanism provides an effective solution when the traffic served by a VPG is above a threshold. In our design, when a leased line collision occurs, the IP-PSTN call is re-routed to another VPG. However, the VPG can also choose to serve the call by deleting an old callback tag that is least recently used (LRU). The replacement strategies of callback tags may merit further study.

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