Journal of Engineering Science and Technology
Vol. 10, No. 2 (2015) 174 - 183

I-TNT: PHONE NUMBER EXPANSION AND TRANSLATION SYSTEM FOR MANAGING INTERCONNECTIVITY ADDRESSING IN SIP PEERING

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Abstract

Voice over IP (VoIP) subscribers is growing vastly in the recent years due to the ever increase in smartphones, 3G, WiFi, etc. This growth leads the VoIP service providers to peer with each other through Session Initiation Protocol (SIP) peering for low/free cost of voice communication. Naturally, this growth is not without challenges, especially in phone addressing. This paper proposes an I-TNT (Infrastructure-Phone Number Translation) numbering system to expand the range of the existing E.164 numbers and mapping between private and public number at the edge of the signalling path. As a result, I-TNT numbering system is successfully implemented and able to allocate the expanded phone numbers to end-users in one service provider.

Keywords: Session Initiation Protocol (SIP), Static indirect peering, Addressing, E.164.

1. Introduction

Voice over IP (VoIP) has been a topic of conversation for sometimes now. Meanwhile, VoIP users are increased vastly which turn the carriers' attention to improve and develop the VoIP services in order to meet users' satisfaction and requirement. Among voice communication protocols, Session Initiation Protocol (SIP) [1] is the most popular and common used. SIP has reached into a mature and stable status, and vastly adopted with multimedia applications [2]. Resulting from that, over 2000 SIP Service Providers (SSPs) over 20 countries are provides voice communication service [3].

Meanwhile, numerous SSPs are planning or already connecting with each other through a relationship what is called SIP Peering or is called Session Peering for

Abbreviations

3G	Third generation of mobile phone standards and technology
B2BUA	Back to back user agent
DID	Direct Inward Dialing
ENUM	E.164 NUmber Mapping
HTTP	Hypertext Transfer Protocol
I-E.164	Infrastructure E.164
I-SSP	Indirect SIP service provider
I-TNT	Infrastructure-Phone Number Translation
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITU	International Telecommunications Union
LRF	Location Routing Function
LUF	Lookup Function
M	Number of generated I-E.164
N	Number of digit adding at the suffix
NAPTR	Naming Authority Pointer
O-SSP	Originating SIP Service Provider
OpenSIPS	Open SIP Server
PSTN	Public System Telephone Network
RFC	Request for Comments
RTP	Real Time Transport Protocol
SBE	Signalling path Border Element
SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
SPEERMINT	Session Peering for Multimedia Interconnect
SSP	SIP Service Provider
T-SSP	Terminating SIP Service Provider
URI	Uniform Resource Identifier
VoIP	Voice over Internet Protocol
WiFi	Wireless Fidelity

Multimedia Interworking (SPEERMINT) as defined by Internet Engineering Task Force (IETF) in RFC5486. SIP peering is a layer 5 (OSI models) interconnection, established between two or more SIP domains [4]. User belong to one domain is likely to have opportunity to exchange calls with other domains with no charge or low-cost. Prior to the peering, several prerequisites, rules, and policy have to be agreed by all the peering domains in order to form peering relationship with high degree of trust among all the domains.

Generally, SIP Peering might achieved through various scenarios which can be grouped as Static and On-demand peering, each group can further sub-divided into Direct and Indirect peering [4]. However, publishing user's addresses among peering domains is an important factor in peering structure. In particular, one SSP might publish the end-users phone number (user-of-record) among the peering domains or instead only publishes the infrastructure address (carrier-of-record) and the two cases of address publishing can be resolved through E.164 NUmber Mapping (ENUM) [5]. ENUM is an application of the DNS, to perform address resolution by translating from E.164 [6] format number to SIP-URI format using Naming Authority Pointer (NAPTR) [7]. Indeed, ENUM serves SIP peering in two types of structures which are User-ENUM and Infrastructure-ENUM [8, 9] for both user-of-record and carrier-of-record respectively.

The key issue behind SIP peering is addressing phone numbers. On one hand, the shortage of E.164 numbers causes the SSP hardly to allocate unique address such as E.164 number to all users. On other hand, un-centralized address routing causes redundant operations accrued in each SSP. In this paper we propose an Infrastructure Phone Number Translation (I-TNT) system to first increase unique identity addresses by expanding one E.164 number enough to produce ten Infrastructure-E.164 numbers, and second to translate between private and public phone numbers in a real-time signalling process for inbound and outbound calls using a routing table resides at the edge of one domain.

The paper is organized as follows: we first provide a background material on SIP system, the SIP peering context and address numbering and how it affects the peering structure. Next, we describe the proposed system followed by two subsections of its methodologies. And the implementation environment is described as well to achieve the desired result. Finally, we summarize our work with a conclusion.

2. Background

In the next subsections, a background on SIP signalling for one domain is discussed. Then peering with other domains is presented with information in a common scenario along with terminology used. Last, phone numbering mapping and routing is also discussed.

2.1. Session initiation protocol

SIP is an open IETF specification. Its core functionality is described under the specification document "RFC 3261". Besides, numerous of sub-specification document related under SIP umbrella. SIP is an application layer protocol that is designed to be independent from the lower transport protocol layer. SIP mechanism used a text-based protocol that is similar to HTTP and SMTP protocols. The main aim of SIP is to initiate, negotiate, establish, change, tear-down and terminate the context of a multimedia session, not only signalling session but also handles the media transmission using RTP for carrying the data (voice) [1].

Since SIP communication uses text-based structure, a number of messages are exchange between caller and callee to establish successful dialog [1]. A simple call can be achieved between two clients via one SIP server as shown as passing message in Fig. 1.

2.2. SIP peering

SIP Peering establishes when two or more SSPs form a peering relationship with some associations established prior to the exchange of traffic [9]. SIP Peering or also known as Layer 5 Interconnectivity. Naturally, as a relationship, peering domains have to set a prerequisite rules and policies to agree by all of them. The main advantages of SIP Peering are to make the peering domains able to exchange other features from other domains such as places a free voice call among them and removes the PSTN involvement in between VoIP networks [9, 10].

One common scenario in SIP peering is the Static Indirect Peering. Here, the Originating SIP Service Provider (O-SSP) [10] and the Terminating SIP Service Provider (T-SSP) [10] are connecting with each other through one or multiple domain(s). In other words, there is no direct layer 5 connectivity between O-SSP and T-SSP [4]. The middle domain in between is an Indirect-SSP (I-SSP) where all the SIP signals and possibly the associated media are all traversed from the O-SSP to T-SSP through this I-SSP domain. A logical database entity might be also resides in this I-SSP domain to handle both Lookup Function (LUF) and Location Routing Function (LRF) for address resolution and guide the call route respectively. Worth to mention, that the I-SSP and the LUF/LRF provider can either be same provider or different provider [4]. Figure 2 presents the context and components of Static Indirect Peering.

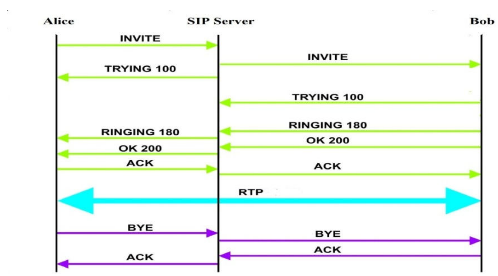

Fig. 1. Message passing for call dialog within one domain.

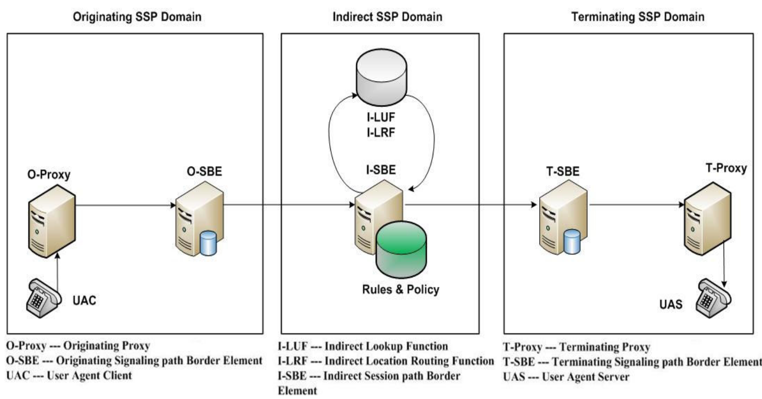

Fig. 2. Static indirect peering context and components.

2.3. Phone addressing

Generally, phone number is the address of a device/entity within the range of numbers in that domain (PSTN or VoIP) where calls can be routed [11]. In other words, one device/entity can be accessed globally by dialing its phone number merely it presents the identity of that device. SIP users can be accessed locally via its private number while globally it only can be accessed through a unique identity such as E.164 public number which is the case in SIP peering. Both private and public numbers are presented at the user part in SIP-URI (e.g., user@domain1.com).

SIP system required both public and private number in order to perform inter-domain call such as peering. Each of which have certain position to use. For instance, private number (e.g., 1001) is used for registration, authentication, and local route only. While public number is used only for global accessibility from other region/country (e.g., +6046845022). However, the combination of use between public and private phone number can be achieved using address mapping such as Direct Inward Dialing (DID) assignment and Caller Identification (Caller ID) for Inbound and outbound call respectively [12, 13].

Routing the phone number from domain to another is achieved by classifying the full number into three tiers unlike to the IP address routing. Tier 1 represents the country code and usually consists from 2-3 digits (US uses one digit only "1"). Tier 2 represents the state or region in a said country. Last, the tier 3 is a subscriber identity. This hierarchy is used to route the call with regards of where the call destination is located [14, 15].

3. I-TNT System

The main aim of I-TNT system is to offer a better interconnectivity in SIP peering by managing phone numbers' allocation and mapping through two algorithms namely Range Expansion and Translation. The system is resides at any edge signalling path of the domain such as proxy, B2BUA, or Signalling function device [16]. In the next subsections, Range Expansion and Translation algorithms are discussed. Then, a flowchart summarizing the whole system is presented as well.

3.1. Range expansion algorithm

The main idea behind this algorithm is to expand the range of E.164 number to be enough for ten users. This achieved by appending extra one digit at the suffix of the E.164 number. For example, say +6046865022 is an E.164 number will generate ten Infrastructure-E.164 numbers (60468650220-60468650229). This however can be achieved through the Range Expansion algorithm as described in what follow.

Expansion algorithm consists from three sub-algorithms in order to achieve the main goal. The sub algorithms are; Lookup Table, Digit Increment, and Record Populating. Initially, after the call request is received the Range algorithm first extracts the user part from the Form header in a SIP URI and then to perform a Lookup table based on the User ID attribute [17]. If it finds a user record then performs a Translation algorithm (section 3.2). Else, retrieves the Port Number of that user to increment the value by one and finalizes the Range algorithm by populates all the user's new particulars into the table for further action and for subsequent calls.

3.2. Translation algorithm

After the I-E.164 number are generated and stored into the phone routing table now the translation algorithm invoked to exchange between public and private phone numbers. As we mentioned earlier, private phone number are not able to be routed globally, that is why a translation is required here to translate with a public number for outbound call. After the I-E.164 number is created in the previous algorithm (the Expansion), the Translation algorithm is able to use that number by replaced with the private number. In what follow, the Translation algorithms step is presented.

Initially, the Lookup table sub-algorithm is performed by extracting the user part from the To header in SIP URI and then performs lookup table based on the User ID attribute. If it finds a user record then retrieves its I-E.164 number from the table and overwrites user part in the SIP URI. Finally, initiates a new INVITE message considers all the new information and route the request to the peering I-SSP/domain

Worth mentioning, that the above two algorithms are served for Outbound call route. However, the Inbound call route is requires no Expansion algorithm because it already obtained an I-E.164 number upon first outbound. However, it only requires a Translation algorithm and it achieved in reverse order as outbound call. Figure 3 shows the passing messages for both algorithms.

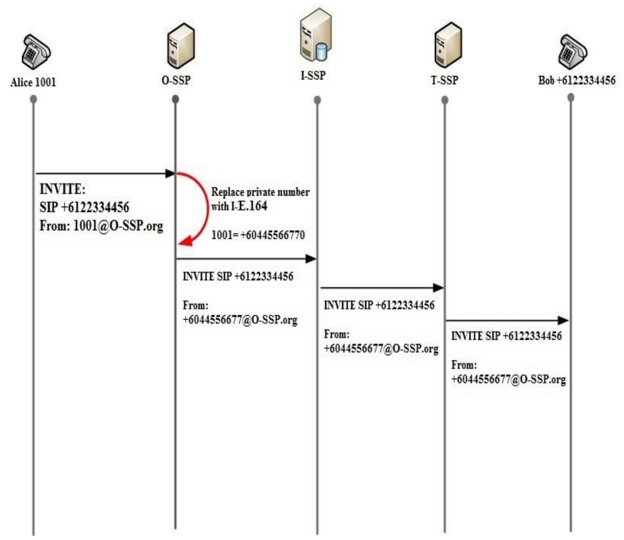

Fig. 3. Passing messages for I-TNT system for static indirect peering.

As a system limitation, the generated Infrastructure-E.164 numbers are not valid within PSTN network. In other words, the generated numbers are unique only for SIP service providers and accessible for VoIP service only.

3.3. An I-TNT system flowchart

In this section a comprehensive flowchart of the I-TNT system is presented and it also shows how is the two algorithms are interoperating together in order to achieve the main goal of the system. Figure 4 shows the flowchart.

4. Implementation

I-TNT system is implemented in Static Direct Peering scenario for testing simplicity. This implementation structured from two end clients X-lite running at two PCs to achieve end-to-end communication (caller and callee). First client (caller) is registered with private phone number (12066) under its home proxy 12066@domain1.com. This domain is an Originating SSP running with Asterisk 1.4 server using Centos 5.5 as operating system. At the other side, the callee is also uses X-lite as user agent registered under its home server 4001@domain2.com. Domian2 server also running Asterisk 1.4 using (Centos 5.5). Our system I-TNT system is implemented inside the SBE. Which is operates as B2BUA using SIP proxy opensips-1.6.1-tls installed into Debianlenny 5.0 machine.

Assume that domain1.com only has one E.164 public number (+6046845022) and it assigned to the user 12066 only one-to-one assignment used for its outbound and inbound call. At the other side, domain2.com also assigned an E.164 public number (e.g., +8099887766) and it maps to the callee user agent 4001. Initially, if 12066 sends an INVITE message it will send the request to itsproxy domain1.com. Next, the proxy will find a peering point which is here SBE to apply its local rules and forward the call to domain2 then to the callee part 4001, that was a default route used for SIP peering. However, if other user (say 12077) in domaon1.com attempts to place a call to the same distension it will pass to I-TNT system and will obtain new I-E.164 generated from the original E.164. For instance, +6046845022 will produce +60468450221 by adding one digit at the suffix of the number. Figure 5 shows the translation table and presents how one number can expand its range.

5. Results and Discussion

I-TNT system can achieve two goals from its algorithms. The first algorithm is able to provide one SSP (domain1.com) with extra 10 unique numbers used for peering. In other words, one E.164 number can produce 10 I-E.164 numbers by adding incremental digit at the suffix of the E.164. Ten numbers is because the probability of one digit range when it depend on the power of ten in a decimal presentation as shown in Eq. (1).

$$10^n = m \qquad (1)$$

where n is the number of digit adding at the suffix and (m) is the number of produced I-E.164.

In our system we add only one digit thus it produces ten numbers. A justification on why one digit used is to avoid the use of long number, still within the range of E.164 and lease routing calculation.

A translation algorithm is used to map between the private number 12077 and its public +60468450221 for outbound call and the reverse order is considered for inbound call. This means, the two operations are done using one system in one element (SBE) instead of doing so separately in each SSP. However, in one point, this can provide centralized management in term of number assignment. In other point, I-TNT system performs phone mapping automaticity during a real-time signalling instead of assigning the number manually to each user for both inbound and outbound calls. As a result, it decreases the redundant operation for each SSP and lease managing required from service administrators.

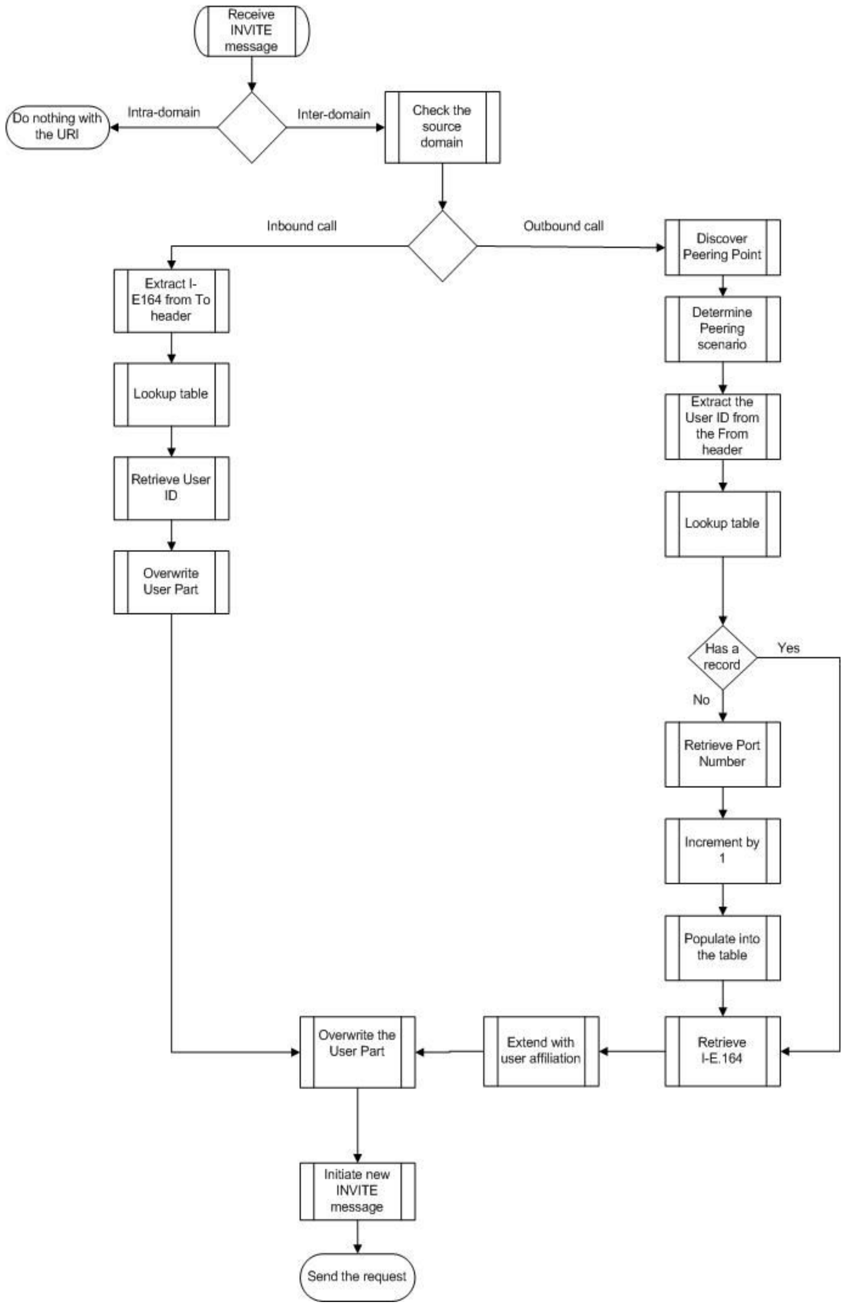

Fig. 4. I-TNT system flowchart.

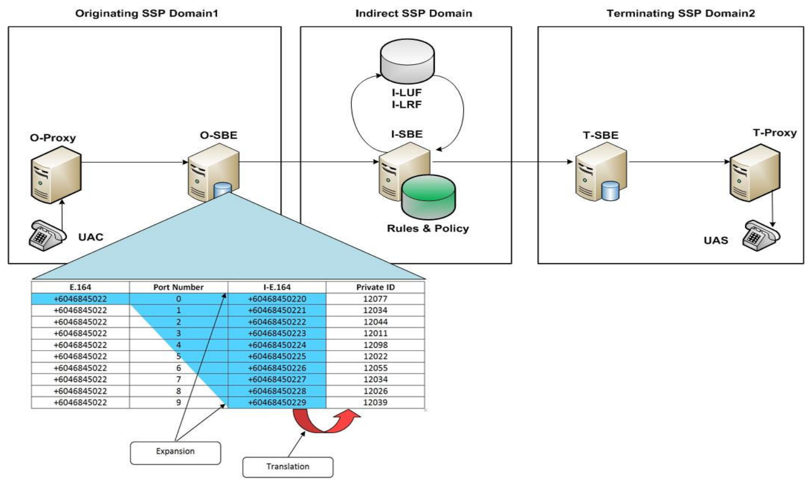

Fig. 5. Phone number range expansion and a translation table.

6. Conclusion

SIP peering between SSPs has a promise future in VoIP industry [18, 19]. SSPs have already or planned to peer with each other for free communication. However, phone addressing is a critical problem for big service providers as it is hard to allocate public phone number for all users. An I-TNT system is proposed to expand the E.164 range number to produce I-E.164 numbers sufficient to be unique within inter-domain infrastructure peering, as well as real-time number translation for mapping between public and private phone numbers implemented at the gateway of the said domain. The quest for a cost-effective phone number addressing solution has begun and might be required by big enterprises such as universities, organisations, companies etc. Such solution might meet their requirements for their vast users and their cooperation with other via voice communication. It also decreases PSTN involvement between service providers.

A future work can be done with big scale of implementation. The system can serve multiple domains for both inbound and outbound call route and to be tested with massive SIP traffic for evaluation.

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