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Effect of Load-Balancing against Disaster Congestion with Actual Subscriber Extension Telephone Numbers

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SUMMARY We demonstrated that load balancing using actual subscriber extension numbers was practical and effective against traffic congestion after a disaster based on actual data. We investigated the ratios of the same subscriber extension numbers in each prefecture and found that most of them were located almost evenly all over the country without being concentrated in a particular area. The ratio of every number except for the fourth-last digit in the last group of four numbers in a telephone number was used almost equally and located almost evenly all over the country. Tolerance against overload in the last, second-, and third-last single digits stays close to that in the ideal situation if we assume that each session initiation protocol server has a capacity in accordance with the ratio of each number on every single digit in the last group of four numbers in Japan. Although tolerance against overload in double-, triple-, and quadruple-digit numbers does not stay close to that in the ideal situation, it still remains sufficiently high in the case of double- and triple-digit numbers. Although tolerance against overload in the quadruple-digit numbers becomes low, disaster congestion is still not likely to occur in almost half of the area of Japan (23 out of 47 prefectures).

key words: load balancing, congestion, disaster, VoIP, SIP, network design, systematic sampling, subscriber extension telephone number

1. Introduction

When a disaster such as a massive earthquake occurs, people call their friends and family in the disaster-affected area out of concern for their safety. Thus, calls over the public switched telephone network (PSTN) are suddenly concentrated in the disaster-affected area. Such an abundance of calls easily overloads switches in that area. For example, about 50 times the normal number of calls in the busiest hour of the day (peak hour) in an ordinary day were directed to the Kobe area just after the Great Hanshin-Awaji earthquake in 1995 [7]. About 50 times more than the ordinary number of calls were directed to Niigata prefecture from all over Japan just after the Chuetsu earthquake in 2004 [2]. An abundance of calls was also concentrated in the disaster-affected area after the Great East Japan Earthquake struck in 2011, although people were also able to use various communication media such as social media. Research on physically resilient networks [5], [9] is important because a disaster can cause communication networks to break down. However, research on designing robust networks to cope with congestion is also important [12], [14].

Many telephone companies are planning to transform their telecommunications infrastructures into pure Internet protocol (IP) based networks because the Internet can provide wide bandwidth at low cost. However, congestion caused by disasters may still occur in these networks. Bandwidth shortage, the cause of bottlenecks in the PSTN, rarely causes disaster-related congestion in voice-over-IP (VoIP) networks. The main bottleneck factor in IP networks is the processing power of session initiation protocol (SIP) servers. The ability of SIP entities to process SIP messages quickly is crucial to achieve adequate SIP network performance since strict timing requirements are common [13]. Therefore, it is necessary to control the traffic load or to increase the processing power of SIP servers economically to prevent congestion caused by disasters.

Ejzak et al. [1] examined overload control in the SIP layer. They concluded that SIP applications have similar methods of handling overload and congestion for the Integrated Services Digital Network (ISDN) user part (ISUP); both the SIP and ISUP applications request the sending entities to slow down or stop traffic to the application that is experiencing congestion. Ohta [8] proposed an overload control method for a SIP signaling network and evaluated it using a network simulator. However, when a disaster strikes, the offered load is usually much larger than the assumed load. The load easily exceeds the capacity of a SIP server, and thus, the server is overloaded. In the situation we are focusing on, neither of these control methods work. Moreover, the additional load of requests for the sending entities to slow down or stop traffic would further overload a SIP server.

Hattori [3] emphasizes that further study should be done on networks to improve their capabilities after a disaster strikes. Satoh and Ashtagawa [10], [11] proposed a SIP telephony network design to prevent congestion caused by a disaster. This design enables load sharing without equipment for load balancing and increases the capacity of SIP servers in a network by using all the SIP servers equally, based on the assumption that users who have the same subscriber extension number are located all over the country without being concentrated in a particular area. This design is a solution for designing robust networks to cope with congestion. Designing such networks is becoming more important because the Japanese government has been encouraging research and development towards realizing a less
congestion-prone network ever since the Great East Japan Earthquake [12], [14]. A simulation indicated that this design had roughly 20 times more capacity, which is 57 times the normal load, than the conventional design in the event that a disaster occurred in the area of Niigata Prefecture that was struck by the Chuetsu earthquake in 2004 [10], [11]. However, these numbers were the results obtained assuming that every subscriber extension number was uniformly distributed in each prefecture according to its traffic ratio. If certain subscriber extension numbers are not uniformly distributed in each prefecture, the design cannot perform as effectively as in the simulation. We show in this paper that the digits of every actual subscriber extension number—except the single digit fourth from the last—are almost uniformly distributed in each prefecture. Tolerance against overload that is calculated based on this verification indicates that the proposed design performs as effectively as in the simulation.

The rest of this paper is organized as follows. Section 2 explains the telephone number structure in Japan. Section 3 summarizes the SIP telephony network design proposed by Satoh and Ashitagawa [10], [11]. Sections 4 and 5 present our verification of the effect of load balancing by using actual subscriber extension numbers. Finally, Sect. 6 concludes the paper.

2. Telephone Number Structure

A national telephone number consists of the national destination code followed by the subscriber extension number [6]. In Japan, the area code and the exchange code compose the national destination code. That is, telephone numbers in Japan start with ‘0’, followed by the area code, the exchange code, and then the subscriber extension number [4], [15], [16]. The exchange code and the subscriber extension number in combination form the local number. The extension number consists of four digits. They are referred to in this paper as the last (referring to the final digit in the extension number), second-, third-, and fourth-last digits. The area code is the number that is assigned to an area where the minimal call charge is applied. The exchange code is the number assigned to each local switch (LS). There are several LSs in an area, and each has its own number. Each subscriber is accommodated by the LS that has the exchange code of that subscriber’s telephone number.

3. SIP Network Design to Prevent Disaster Congestion

A SIP telephony network can be robust against congestion caused by a disaster, whereas congestion is inevitable in the PSTN just after a disaster strikes. Users can be allocated to SIP servers independently of their location; in contrast, PSTN users have to be allocated to LSs on the basis of the user’s location. SIP network elements carry signaling packets but not voice data packets, whereas an LS in the PSTN carries both loads.

Satoh and Ashitagawa [10], [11] proposed a SIP telephony network design to prevent disaster congestion. Their SIP network design uses the one, two, three, or four-digit numbers of the subscriber extension number to assign a user to a SIP server on the condition that the Tel Uniform Resource Identifier (tel URI) is adopted as the SIP-URI. The assumed tel URI helps the migration from the PSTN to IP telephony. The area code is assigned to a user based on the user’s area as in the conventional manner, and the number given to the relevant router is used as the exchange code. The foundation of the proposed design is the simple and equal assignment of users independent of their location in relation to the SIP servers. This proposed design takes advantage of a property of VoIP networks where signaling and transmission are completely separate. That is, a SIP server only processes signaling messages, and a router only transmits but does not process packets (including both signaling packets and voice packets), whereas an LS in the PSTN carries both signaling and transmission loads.

With this design, users in an area would be allocated to several different SIP servers around the country, and the load could be distributed among the servers if a disaster were to strike that area. Moreover, routing could be done in a conventional manner on the basis of telephone numbers. The routing when users are allocated in accordance with their network scheme is shown in Fig. 1. A SIP server (1111) accommodates users from around the country whose subscriber extension numbers are 1111. Another SIP server (2222) accommodates users whose subscriber extension numbers are 2222 in a similar manner. The calling party E has the telephone number 03-3333-1111, and hence, is allocated to router 3333 and SIP server 1111. When E makes a call to party W, who has the number 06-6666-2222, an INVITE message is sent via router 3333 to SIP server 1111. Then it is sent to the called party W’s SIP server 2222, and SIP server 2222 transmits the message, which is delivered to the final destination (called party W) via router 6666. Thus, routing on the basis of telephone numbers is easily done.

4. Load-Balancing with Actual Single-Digit Numbers

Satoh and Ashtigawa [10], [11] assume that the ratio of each subscriber extension number is the same in Japan and that the ratio of each subscriber extension number is the same among prefectures in their analyses of tolerance against overload. Both assumptions are expected to be ap-
approximately held. The first assumption can be excluded if the SIP server capacity for each subscriber extension number is prepared in proportion to the ratio of each number. The second assumption is probably an accurate approximation because the user accommodation proposed by Satoh and Ashitagawa [10], [11] can be regarded as systematic samples, and systematic sampling is often used instead of random sampling. However, the ratio of each subscriber extension number cannot be completely the same among prefectures. In this section, we verified both assumptions with actual single digit numbers obtained in 2007, which were data of the number of users who had the same subscriber extension numbers in each prefecture. The users were fixed-line PSTN and SIP telephony users whose telephone numbers started with ‘0’, followed by the area code, the exchange code, and then the subscriber extension number. There were 60.8 million users, which was the number of most fixed-line PSTN and SIP telephony users in Japan.

### 4.1 Ratios of Single-Digit Numbers All Over Japan

We first investigated the ratios of the last, second-, third-, and fourth-last single-digit numbers all over Japan. The results are shown in Fig. 2, where A, B, . . . , J represent 0, 1, . . . , 9 but in no particular order. The average and standard deviation (SD) are given in Table 1. The ratios of single-digit numbers are almost the same except for the case of the fourth-last digit. Note that the difference in the ratios of single-digit numbers all over Japan does not affect the robustness against disaster congestion. If the capability of a SIP server is determined to meet the ratios, and all coefficients of variation (CVs) are zero, the most effective load-balancing is achieved.

![Fig. 2 Ratios of digits from 0 to 9.](image)

<table>
<thead>
<tr>
<th></th>
<th>Min</th>
<th>Average</th>
<th>Max</th>
<th>SD</th>
</tr>
</thead>
<tbody>
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<td>0.088077</td>
<td>0.1</td>
<td>0.112341</td>
<td>0.00681841</td>
</tr>
<tr>
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<td>0.086210</td>
<td>0.1</td>
<td>0.105686</td>
<td>0.00620901</td>
</tr>
<tr>
<td>Third</td>
<td>0.081288</td>
<td>0.1</td>
<td>0.112346</td>
<td>0.00851486</td>
</tr>
<tr>
<td>Fourth</td>
<td>0.052485</td>
<td>0.1</td>
<td>0.143561</td>
<td>0.0267986</td>
</tr>
</tbody>
</table>

4.2 Ratios of Single-Digit Numbers in Each Prefecture

Next, we investigated how close the ratios of single-digit numbers in individual prefectures were to those in Japan as a whole (denoted as “Japan”). If the prefecture ratios are the same as those for Japan, load-balancing is most effective, and the SIP network is robust against disaster congestion. However, the actual ratios are different among prefectures, as shown in Fig. 3, which plots the ratios of numbers for the last single digit in each prefecture. The prefecture numbers represent prefectures, but are in no particular order. The smaller the variation in the ratios among prefectures becomes, the closer the ratios are to the ratios for Japan.

We used the CV as a criterion of variation of the ratios because the ratios of single-digit numbers are not the same in Japan, as shown in Fig. 2, where $c_v(i)$ is defined as the ratio of $\sigma(i)$ to $\mu(i)$:

$$c_v(i) = \frac{\sigma(i)}{\mu(i)} \quad (i = 0, 1, \ldots, n - 1),$$

where $i$ is a subscriber extension number, $n$ is the number of subscriber extension numbers, e.g., 10 in the case of the single-digit number and 100 in the case of the double-digit number, and $\mu(i)$ and $\sigma(i)$ are the average and the SD of the ratios of subscriber extension number $i$ among prefectures, respectively. We explain how to obtain the CV in the case of the last single digit. We can obtain those of the second-, third-, and fourth-last single digit in the same manner. We calculated the ratio of users who had the same last single-digit number in each prefecture. Then we calculated the average $\mu$ and SD $\sigma$ of the ratio of each last single-digit number among all prefectures and obtained the CV of a single-digit number.

We show the CV of the numbers on the last, second-, third-, and fourth-last single digit in Fig. 4, where A, B, . . . , J represent 0, 1, . . . , 9 but in no particular order. The figure indicates that the ratios of all the single digits in each prefecture are almost the same as those in Japan except for the case of the fourth-last single digit. Table 2 lists the minimum, average, maximum, and SD of the CV.
of the numbers used as the last, second-, third-, and fourth-last single digit. The CVs of the four-last single-digit are much larger than those of the others, as seen in Fig. 4 and Table 2. The CVs of the numbers used as the last, second-, and third-last single-digit are shown in Fig. 5, where A, B, , , J represent the same numbers as those in Fig. 4. For the last, second-, and third-last single-digit number, this result shows that the variation in ratios of the last single-digit number among prefectures is small, and the ratios of single-digit numbers in each prefecture are almost the same as those in Japan. That is, the ratios of the numbers are uniformly distributed among prefectures. Load-balancing using the last, second-, and third-last single-digit numbers is therefore expected to be effective. The ratios of the fourth-last single-digit numbers are very different in Japan, and there is large variation in the ratios of the fourth-last single-digit numbers among prefectures. Consequently, load-balancing using the fourth-last single-digit numbers is expected to be less effective than the others.

4.3 Tolerance against Overload

Satoh and Ashitagawa [10], [11] estimate tolerance against overload when every subscriber extension number, regardless of the number, is uniformly distributed among the prefectures, and each subscriber extension number has the same ratio in each prefecture. Congestion tolerance depends on only the traffic ratio, i.e., the proportion of incoming traffic to a disaster area to the total traffic in the country under the above assumption. If the proportion is smaller/larger, the load multiplication factor as the congestion tolerance is larger/smaller. When proportion r was defined as the ratio of received calls to a certain prefecture for the whole network, Satoh and Ashitagawa [10], [11] showed that the congestion tolerance load multiplication factor \( N_c(r) \) is described as

\[
N_c(r) = 1 + \frac{1.0354}{r}.
\]

(2)

Equation (2) is obtained from their simulation results and the following equation:

\[
N = 1 + \frac{47C - \frac{1}{rL}}{rL}.
\]

(3)

where \( N \) represents the load multiplication factor when the number of SIP servers is 47 (the number of prefectures), \( x \) represents the number of SIP servers in their simulation, \( C \) is the amount of load processed by one SIP server per unit time, and \( L \) is the amount of load in the whole network in a normal period per unit time.

We estimated the congestion tolerance against overload when users were accommodated in SIP servers by using the actual last, second-, and third-last single-digit numbers. We compared the load multiplication factor with that in the ideal situation as tolerance against overload. The ideal situation is where every subscriber extension number, regardless of the number, is uniformly distributed among prefectures, and the prefectures have the same ratios of users by location. We assumed that the offered load was independent of the single-digit numbers.

First, we assumed that each SIP server had a capacity in accordance with the ratio of each number used for every single digit in Japan. That is, we focused on how the congestion tolerance depended on the differences in the actual ratios among prefectures. We call this situation case 1. In case 1, \( r_p \) of prefecture \( p \) changes to \( r_p \cdot \max_i n_i^{p}/n_i^{\text{All}} \) virtually, where \( i \) represents the number of a single digit; 0, 1, , 9, \( n_i^{p} \) represents the ratio of number \( i \) in prefecture \( p \), and \( n_i^{\text{All}} \) represents the ratio of number \( i \) in Japan. As a result, the load multiplication factor \( N_{c1}(r_p) \) is written as

\[
N_{c1}(r_p) = 1 + \frac{1.0354}{r_p \cdot \max_i \left( \frac{n_i^{p}}{n_i^{\text{All}}} \right)}.
\]

(4)

The load multiplication factor in case 1 is plotted in Figs. 6--
9. The load multiplication factor decreased due to the differences in the actual ratios among prefectures as

\[
\frac{N_{c1}(r_p)}{N_c(r_p)} = \frac{1 + \frac{1.0354}{r_p \cdot \max \left(\frac{i}{n^i_p} \right)}}{1 + \frac{1.0354}{r_p}}.
\]  

(5)

The ratios of the load multiplication factor in case 1 against the ideal situation are plotted in Figs. 10–13.

Next, we assumed that each SIP server had the same capacity, regardless of the number. We call this situation case 2. In this case also, \(r_p\) of prefecture p changes to \(r_p \cdot \max, n^i_p/0.1\) virtually, where 0.1 indicates the same ratio as a single digit. As a result, the load multiplication factor \(N_{c2}(r_p)\) is written as

\[
N_{c2}(r_p) = 1 + \frac{1.0354}{r_p \cdot \max \left(\frac{i}{n^i_p} \right)}.
\]

(6)

The load multiplication factor in case 2 is also shown in Figs. 6–9. The load multiplication factor decreased due to the difference in the actual ratios among prefectures as

\[
\frac{N_{c2}(r_p)}{N_c(r_p)} = \frac{1 + \frac{1.0354}{r_p \cdot \max \left(\frac{i}{n^i_p} \right)}}{1 + \frac{1.0354}{r_p}}.
\]

(7)

The ratios of the load multiplication factor in case 2 against the ideal situation are also shown in Figs. 10–13.

Tolerance against overload in case 1 is almost the same as that in the ideal situation as shown in Figs. 6–9. Table 3
lists the minimum, average, and maximum of the ratios of load multiplication factor in cases 1 and 2 against the ideal situation. It also indicates the number of prefectures that do not meet \( N_e(r_p) > 50 \) or \( N_e^2(r_p) > 50 \) by changing from the ideal situation to the actual situation. For example, the number of prefectures where the load multiplication factor became less than 50 decreased by four for the fourth-last digit in case 1, as indicated in the “Decreased prefectures” column in the table. If the load multiplication factor of a prefecture is larger than 50, disaster congestion is not likely to occur in the prefecture based on data obtained in past earthquakes [2], [7]. Of the 47 prefectures, 36 conform to \( N_e(r_p) > 50 \) in the ideal situation [11]. If users are accommodated in SIP servers by using the actual last, second-, and third-last single-digit numbers, 36 prefectures still have a load multiplication factor larger than 50, as seen in Table 3. This result is based on the small CV. Only two prefectures do not conform to \( N_e(r_p) > 50 \) by changing from the ideal situation to the actual situation by using the actual fourth-last single-digit numbers in case 1. Even if users are accommodated in SIP servers by using the fourth-last single-digit numbers, 32 prefectures still maintain the condition in which the load multiplication factor is larger than 50, although the CV of the fourth-last digit numbers is larger than those of the others. Tolerance against overload in case 2 is inferior to that in case 1. However, only two prefectures do not meet the condition of having a load multiplication factor larger than 50 in the case of the last, second-, and third-last single digits by changing from the ideal situation to the actual situation. Furthermore, case 2 is avoidable if each SIP server has a capacity in accordance with the ratio of each number on every single digit in Japan.

5. Load-Balancing with Actual Double-, Triple-, and Quadruple-Digit Numbers

We analyzed the effect of load-balancing with actual double-, triple-, and quadruple-digit numbers. The used data were the same as in Sect. 4. The double- and triple-digit numbers have several combinations, e.g., six combinations for double-digit numbers and four combinations for triple-digit numbers. We analyzed only the cases in which the last and second-last digits were used for double-digit numbers and only the last, second-, and third-last digits were used for triple-digit numbers because the average CV values became larger in the order of the last, second-, third-, and fourth-last single digits, as indicated in Table 2.

5.1 Ratios of Double-, Triple-, and Quadruple-Digit Numbers All Over Japan

We show the distributions of ratios of double-, triple-, and quadruple-digit numbers all over Japan in Figs. 14, 15, and 16, respectively. Each figure has a bell-like shape. The minimum, average, maximum, and SD of ratios of double-, triple-, and quadruple-digit numbers are listed in Table 4. We compared CVs because the ratios were different among single-, double-, triple-, and quadruple-digit numbers. The result is shown in Fig. 17. As the number of digit numbers increases, the CV increases because the population size is larger. However, the CV of the fourth-last single digit is especially large. The CV of the quadruple-digit numbers is much larger than those of the double- and triple-digit numbers because of the fourth-last single-digit number as well as the population size.

5.2 Ratios of Double-, Triple-, and Quadruple-Digit Numbers in Each Prefecture

The CVs of ratios of the double-, triple-, and quadruple-digit numbers among prefectures are shown in Figs. 18, 19, and
The CVs for most of the double-, triple-, and quadruple-digit numbers are almost the same. Only a small portion of the numbers have CVs that are much larger than the other numbers. This means that most of the double-, triple-, and quadruple-digit numbers are used evenly in every prefecture, although a small number of the double-, triple-, and quadruple-digit numbers are used often in some prefectures and are hardly used in others. Table 5 gives the minimum, average, maximum, and SD of the CVs. These values in the cases of the double- and triple-digit numbers are smaller than those of the fourth-last single-digit number among prefectures, as seen in Tables 2 and 5, although the population size is 10 and 100 times larger than that of the CV of the ratio of the fourth-last single-digit number. Therefore, the double- and triple-digit numbers are used more evenly than the fourth-last single-digit number.
5.3 Tolerance against Overload

We estimated the congestion tolerance against overload when users were accommodated in SIP servers by using the actual last double-, triple-, and quadruple-digit numbers for cases 1 and 2 in the same manner as for the single-digit number.

Figures 21, 22, and 23 plot the ratios of load multiplication factors against the ideal situation for double, triple, and quadruple digits, respectively. Table 6 lists the minimum, average, and maximum of the ratio of the load multiplication factor, and the number of decreased prefectures in both cases 1 and 2 against the ideal situation. A comparison of Tables 3 and 6 reveals that the ratios of load multiplication factor in the double- and triple-digit numbers are larger than those in the fourth-last single digit. The number of decreased prefectures is small for the double- and triple-digit numbers; they are 2 and 3 respectively in cases 1 and 4, and 5 even in case 2. If users are accommodated in SIP servers by using the actual last double- and triple-digit numbers, the congestion tolerance is nearly the same as that in the ideal situation. Although the number of decreased prefectures in the quadruple-digit numbers is relatively large, disaster congestion is not likely to occur in almost half of the area of Japan (23 of 47 prefectures) in case 1 and in one third of Japan (16 prefectures) in case 2.

<table>
<thead>
<tr>
<th>Digit</th>
<th>Case</th>
<th>Min</th>
<th>Average</th>
<th>Max</th>
<th>Decreased prefectures</th>
</tr>
</thead>
<tbody>
<tr>
<td>Double</td>
<td>1</td>
<td>0.8091</td>
<td>0.9034</td>
<td>0.9727</td>
<td>2</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>0.6966</td>
<td>0.8059</td>
<td>0.9012</td>
<td>4</td>
</tr>
<tr>
<td>Triple</td>
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<td>0.6366</td>
<td>0.8218</td>
<td>0.9088</td>
<td>3</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>0.5944</td>
<td>0.6985</td>
<td>0.8452</td>
<td>5</td>
</tr>
<tr>
<td>Quadruple</td>
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<td>0.253</td>
<td>0.5472</td>
<td>0.7258</td>
<td>13</td>
</tr>
<tr>
<td></td>
<td>2</td>
<td>0.2495</td>
<td>0.4467</td>
<td>0.6473</td>
<td>20</td>
</tr>
</tbody>
</table>

6. Conclusion

We demonstrated that the SIP telephony network design proposed by Satoh and Ashitagawa is effective against congestion caused by a disaster based on actual data. Tolerance against overload in the last, second-, and third-last single-digit number is maintained in the ideal situation if we assume that each SIP server has a capacity in accordance with the ratio of each number used for every single digit in Japan. Although tolerance against overload in the double-, triple-, and quadruple-digit numbers is not maintained in the ideal situation, it still remains high in the case of double- and triple-digit numbers.

The numbers for the last, second-, and third-last single digits are used almost evenly all over Japan, whereas the
ratio of the numbers used for the fourth-last single digit is different all over Japan. The ratios of numbers used for the last, second-, and third-last single digits in each prefecture are almost uniformly distributed among prefectures. Consequently, if users are accommodated in SIP servers by using the last three single-digit numbers in the subscriber extension numbers, the distribution of user location is almost the same. The distribution of user location based on the fourth-last single-digit number is less uniformly distributed among prefectures compared with the other single-digit numbers. The ratios of numbers used on the double- and triple-digit numbers are almost uniformly distributed among prefectures. If users are accommodated in SIP servers by using the double- and triple-digit numbers in the subscriber extension numbers, the distribution of user location is almost the same. The distribution of user location based on the quadruple-digit numbers is less uniformly distributed among prefectures compared with the double- and triple-digit numbers.

Tolerance against overload in the last, second-, and third-last single-digit number is maintained in the ideal situation if we assume that each SIP server has capacity in accordance with the ratio of each number for every single digit in Japan (case 1). Disaster congestion is not likely to occur in 36 out of 47 prefectures. This number is the same as that in the ideal situation. Disaster congestion is not likely to occur in 32 prefectures even in the case of the fourth-last single-digit number. Tolerance against overload in the last, second-, and third-last single-digit numbers decreases by two prefectures from that in the ideal situation if we assume that each SIP server has the same capacity (case 2). Ten prefectures decrease in the case of the fourth-last single digit. Disaster congestion is not likely to occur in 26 prefectures in that case. Although tolerance against overload in the double-, triple-, and quadruple-digit numbers does not match that in the ideal situation, it still remains high in the case of the double- and triple-digit numbers. Disaster congestion is not likely to occur in 34 and 33 prefectures out of 47 prefectures in the case of the double- and triple-digit numbers in case 1, respectively. The numbers of prefectures where disaster congestion is not likely to occur in the case of the double- and triple-digit numbers are 32 and 31 in case 2, respectively. It is true that tolerance against overload in the quadruple-digit numbers becomes low, but the disaster congestion is not likely to occur in almost half the area of Japan (23 of 47 prefectures) in case 1 and in one-third of Japan (16 prefectures) in case 2.

A bandwidth shortage may cause a bottleneck when damage occurs due to a disaster. Further study remains necessary to control bandwidth in such a situation by controlling signaling.

References

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