

Method for Reduction Indefinite Routing Loops Probability in the Mixed Telephone Network of Electric Power Utility

Abstract. In this paper, we present the method for reducing the probability of indefinite looping in single-layer (non-hierarchical) telephone network of Electric Power Utility. The method is based on the detection of the first loop after the reception of two initial signalling messages, which contain the same source and destination addresses. In this case the network node, where the first loop is detected, changes its routing plan. The method can't eliminate loop appearance completely, and it is not applicable when overlap signalling is used.

Streszczenie. W artykule opisano metodą redukcji prawdopodobieństwa wystąpienia pętli warunkowych w jednowarstwowej sieci telefonicznej operatora energetycznego. W metodzie dokonywana jest detekcja pierwsze pętli, o takim samym adresie źródłowym i docelowym, która wystąpi po wykryciu wiadomości inicjalizacyjnych. Następnie następuje zmiana routowania. Metoda nie eliminuje pętli w zupełności. (Redukcja prawdopodobieństwa wystąpienia pętli routowania warunkowego w mieszanej sieci telefonicznej operatora elektroenergetycznego).

Keywords: Telephone network, Signalling, Indefinite loops

Słowa kluczowe: sieć telefoniczna, sygnalizacja, pętle warunkowe.

Introduction

Routing loop is the route crossed by the signalling packet or the signalling message through network nodes and links on its way back to the node where it already was. Routing loops are always harmful and when they are established, their consequence is the loss of signalling packets or signalling messages. Two methods in loop management exist. The first one is reducing the probability of loop appearance and the second one eliminating indefinite looping in the network. Reducing the probability of loop appearance is realized in packet networks using the known routing protocols (for example: *Spanning Tree Protocol*) and in telephone networks using the carefully designed routing tables. In this paper, we present the method, applied in the mixed telephone network, which decreases the probability of indefinite loop appearance.

In section 2 we present the mixed telephone network of Electric Power Utility (EPU) with its main characteristics. Section 3 deals with the known procedures for reducing network loop influence. Section 4 presents how loops are constituted in mixed telephone EPU network. The simple method of software upgrade in network nodes, which reduces probability of indefinite looping, is presented in section 5. Section 6 deals with the simple calculation of the efficiency of this method.

Mixed EPU telephone network

In this paper we consider the EPU telephone network, which is very similar to the network presented in [1]. The main goal of EPU network configuring is to achieve as great availability as possible. That's why the network is constructed in single layer (non-hierarchical network). It has the characteristic of alternate routing and uses all available resources, regardless of technological actuality. The network consists of its IP part and TDM part. The components of IP part are IP exchanges, IP links and IP telephones. The SIP signalling system is used in the IP network part [2]. The gateways for connecting with TDM part are also the part of IP exchanges. The components of TDM part are TDM (ISDN) exchanges, TDM links and ISDN telephones. The signalling systems CCS7 [3] and QSIG [4] are used in TDM network part. The interfaces for the connection with IP network (gateways) are situated in TDM exchanges. The gateways (interfaces) are media and signalling converters. All the exchanges have the function of transit and local exchanges. It means that they can transit all connections, regardless of link type (TDM ↔ IP), because adequate interfaces (i.e. gateways) are situated in each of them. It is obvious that each exchange has the certain number of user (IP or ISDN) telephones. Fig. 1 contains the symbolic presentation of the part of mixed EPU telephone network.

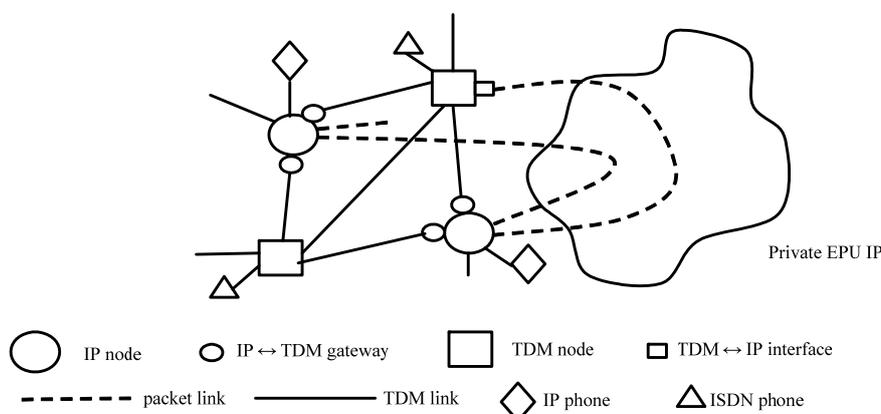


Fig.1. Symbolic presentation of the mixed EPU telephone network

Network routing loops and the used procedures

Routing loops, i.e. (signalling) packet returning in the network node where it already was, are the characteristic of single-layer networks. Routing loops creation is possible in computer networks, and the series of protocols, which reduce the possibility of routing loops creation, is developed, as, for example, in [5], [6], [7]. Several patents are devoted to solving the problem of loop creation in telecommunication networks. Patent [8] presents one solution, which uses special transfer prohibited messages between network (signalling) nodes for preventing loop forming in telecommunication network with common channel signalling. One other patent [9] presents the method of network loop detection, when the length of the route through the network is equal to or greater than the one, in advance known value. The procedure for detection of eventually created loops in routing tables of network routers is presented in [10]. Loop avoidance in the network, using MPLS (MultiProtocol Label Switching), is presented in [11]. The procedure for routing loops prevention, which inserts network authorization information allowing or prohibiting some connections, is presented in [12]. In this way, the interworking of different provider networks is enabled.

The link occupancy is the additional cause of loop forming in telephone networks. In classic telephone networks the probability of network loop forming is small, because the network is organized hierarchically. In non-hierarchical telephone network with alternate routing, which is realized in one technology, the probability of route forming is greater, and the great attention must be paid to careful designation of routing plan. In non-hierarchical telephone network, realized using several different technologies, it is practically impossible to implement the signalling protocol, which would change routing tables according to the network state.

The other way of loop influence reduction is the prevention of endless looping in the network. The methods for reducing the influence of already formed loops are based on preventing (too) long circulation of signalling packet or signalling message in the network. In header of Internet packet exists the mechanism (*Time To Live*, TTL, [13]), and its use prevents indefinite loop duration. This mechanism retains in the Multi Protocol Label Switching (MPLS, [14]). In the IPv6 header this value is called *hop limit*, [15]. The method of parameter value decrease when signalling message leaves one network node exists also in telephone networks. In ISDN signalling, [16], and in QSIG signalling, [4], this parameter is called *Transit Counter*. This mechanism is also used in CCSNo7 signalling, and the parameter name is *Hop Counter*, [3]. The common characteristic of all methods for network nodes counting is that they only prevent indefinite looping, but they can't prevent packet and signalling message loss. That's why it is crucial to avoid the loops, i.e. to reduce the probability of loop appearance. In this short paper we present the problem of loops in single-layer, i.e. non-hierarchical telephone network of Electric Power Utility (EPU) and we suggest one method for reducing the probability of indefinite loops appearance.

Loops in the telephone networks (of Electric Power Utility)

Telephone network of EPU has several specific characteristics [1], [17]. The important request in the construction of this network is the high availability. This request can be achieved using different means, and one of the main ones is that it is single-layer network, because this architecture enables the use of alternate routing. That's why

when establishing telephone connection (which is realized according to the principle link by link), in each node exists routing plan (table), which consists of the route of first, second, etc. choice for each selected number. The route of first choice is the one, which enables the fastest connection set-up and the best voice signal quality. The routes of second, third, etc. choice are used in the periods of busy or faulty links of first choice. Non-hierarchical network with alternate routing is very efficient for limited traffic values. The bad characteristic of single-layer telephone network with alternate traffic routing is the possibility of forming a loop. When creating routing tables, with several alternate routes in each node, it is difficult to predict all possible combinations of busy and faulty links. Modern networks of EPU can be mixed, composed of IP and non-IP network parts, [1]. The possibility of forming a loop in this mixed network is increased, because the routing tables are constructed in different technics.

Let us consider the part of the single-layer network in Fig. 2.a). In this figure designations 1, 2, 3, 4, 5 present network nodes and the designations A, B, C, D, E, F, G, H present links connecting network nodes. Network nodes have the function of local exchanges, but also the function of transit nodes.

Fig. 2.b) presents the routing table for the part of the network from Fig. 2.a). The designations 1x, 2x, 3x, 4x, 5x are the calling numbers, i.e. user addresses in exchanges 1, 2, 3, 4, 5, respectively. With **f**, **s**, **t** are designated the links of first, second and third choice, respectively, between particular nodes.

Example 1: If the number 1x starts from the network node (i.e. exchange) 4, then the possible most-favourable routes for signalling messages can be 4C5B1, 4G2A1, 4D3F1. It is obvious that in transit nodes messages can be also routed using different links.

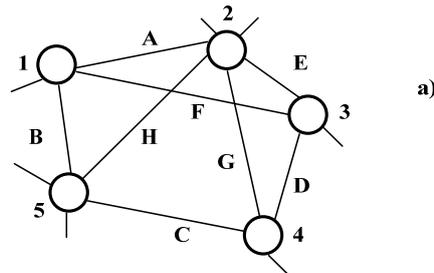
Example 2: In example 1 the route across the link of first choice 4C.... can be 4C5B1, but also 4C5H2A1 in the case of busy or faulty link B.

Let us consider connection set-up. In telephone networks routing is realized usually according to the called user address, which is named:

- *Called Party Number* in the SETUP or IAM (*Initial Address Message*) message in the access ISDN signalling, [16], and CCSNo7 signalling, [3], respectively;
- *Destination Number*, [18], in the SETUP message in QSIG signalling;
- Header Field *To:* in the INVITE message (method), [2], in SIP protocol.

Signalling messages SETUP, IAM and INVITE can be called initial signalling messages of one telephone connection.

Example 3: Let us consider the part of the network presented in Fig. 2.a) and let 3x be the called number. This called number is generated in some part of the network and has arrived in the network node 2. In the case of free and available links the call will be forwarded over the route 2E3 according to the address of called user across the considered part of network. If the link E is faulty or busy, the call will be forwarded over the route 2A1F3, Fig. 3.a), according to the routing table from the Fig. 2.b). Let us consider now rare, but possible case of busy or faulty link F. The call will be forwarded over the route 2A1B5H2, according to the routing table from Fig. 2.b). In that way the loop will be created, Fig. 3.b).



| outgoing node | 1 | | | | 2 | | | | 3 | | | | 4 | | | | 5 | | | |
|---------------|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|----|
| dialed number | 2x | 3x | 4x | 5x | 1x | 3x | 4x | 5x | 1x | 2x | 4x | 5x | 1x | 2x | 3x | 5x | 1x | 2x | 3x | 4x |
| f | A | F | B | B | A | E | G | H | F | E | D | D | C | G | D | C | B | H | H | C |
| s | B | A | F | A | H | A | E | A | E | F | E | F | G | D | G | G | H | B | C | H |
| t | F | B | A | F | E | G | H | G | D | D | F | E | D | C | C | D | C | C | B | B |

f,s,t - link of first, second, third choice

Fig.2. Alternate routing in the part of single-layer network

As the first conclusion of this section, it can be said that multiple loop is formed because the routing tables are static, i.e. they take into account only the destination address and the route state (available-not available) in the fixed routing table. In the example from Fig. 3.b) indefinite loop is formed regardless of the possibility to establish the connection using the route 2G4D3.

As the second conclusion it can be emphasized that when indefinite loop is formed, the same initial message

arrives several times to the same network node(s). It can be seen in the example from Fig. 3.b) that initial signalling message (SETUP, IAM, INVITE) appears first in the node 2 for the second time, and then in the nodes 1 and 5. The process continues in such a way that the message is repeated in these nodes until the mechanism, using *TTL*, *Hop Counter* or *Transit Counter*, interrupts the loop.

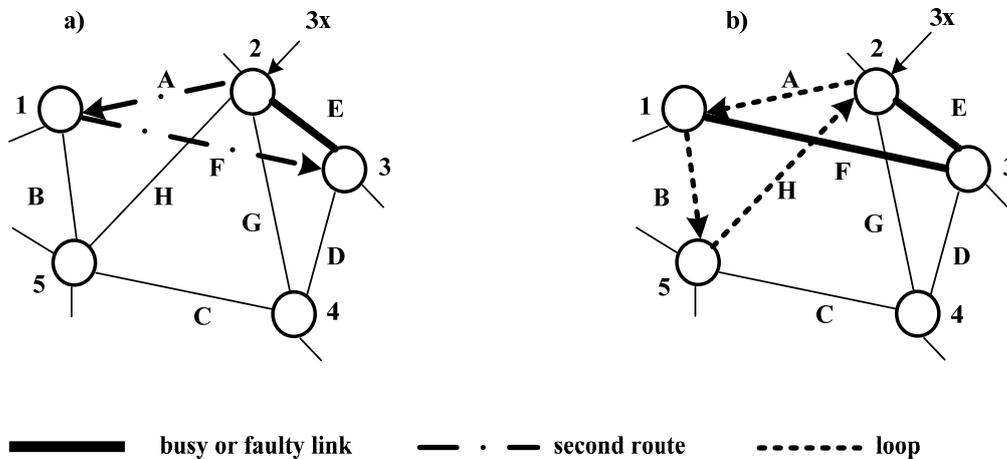


Fig.3. Call routing in the case of busy or faulty links

Reducing the probability of forming a loop

It is obvious that the probability of forming a loop can be reduced by the introduction of the changeable routing table, in which the changes of link states are promptly entered. It is impossible to carry out this complicated procedure in one mixed network, which consists of elements made in different technics. That's why we suggest the appendix of the method for signalling message routing, which takes into account also originating address, besides destination address. The additional rule for routing, according to our proposal, is:

- initial signalling message, which refers to the same connection, when comes in some network node for the

second time, will cause the different routing than the first arrival of the message.

How can we recognize that the same initial signalling message arrived in one network node for the second time? This is the message with the same originating address (OA) and the same destination address (DA) as the message, which already was in the same network node in the immediate past, and was not cancelled by one of the messages from the called user, which mean that the called user is reached (ALERTING, CONNECT, 180 RINGING, 200 OK). In modern signalling systems originating address can be found in each initial message. In access ISDN signalling, [16], and CCSNo7 signalling, [3], it is called

Calling Party Number, in QSIG signalling it is called *Originating Number*, [18], in SIP protocol it is called Header Field *From*:

Software upgrade in the network node, that reduces the probability of forming an indefinite loop, is realized of following steps:

Step 1: After the reception of the initial message store the message, and OA and DA writing them in the Temporary Table of Initial Messages (TTIM);

Step 2: Compare the values OA and DA of the received message with the values of previously received initial messages, that still exist in TTIM;

Step 3: If initial message with these values does not exist in TTIM, the message can be routed according to the order from the routing table;

Step 4: If initial message with the same values OA and DA already exists in TTIM, the message ought to be routed using the different route than the first time;

Step 5: Terminate the loop that is formed when routing the message for the first time;

Step 6: Each initial message with the received acknowledgement that the called user is reached (ALERTING, CONNECT, 180 RINGING, 200 OK) can be deleted from TTIM.

Example 4: Let us consider the part of the network where the loop is formed, Fig. 3.b). In network nodes exists software for reducing the probability for forming indefinite loops, presented in step 1-6. When the initial message appears for the second time in the node 2, forming of the loop will be detected in the step 2. Based on step 2, in step 4 initial message will be now forwarded using the route 2G4D3 and the connection will be realized, Fig. 4.a). In the step 5, the loop will be disconnected, Fig. 4.b).

It is obvious that this method can be used after the first loop is formed, and that the method practically performs rerouting of the connection. It is, also, obvious that this method can not be implemented in network node, where the route of last choice is already used.

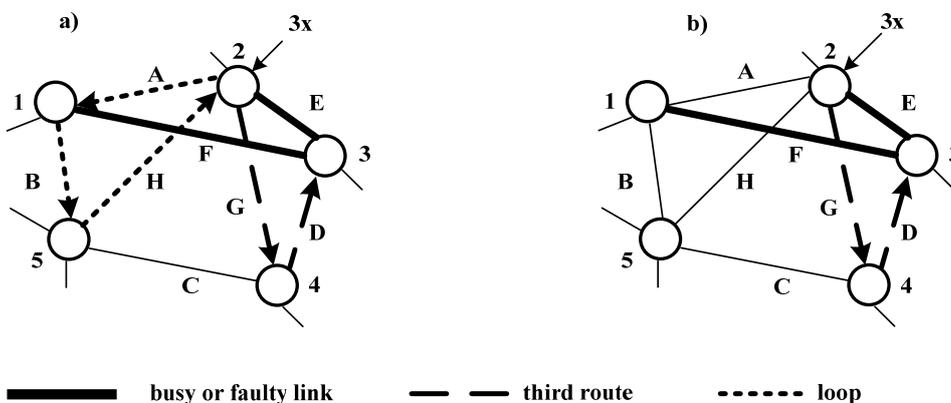


Fig.4. An example how to avoid forming an indefinite loop

Example 5: If it is necessary to send the call from network node 2 (3x) to the user, connected in node 3, and the links A, E and D are occupied or faulty, the loop 2G4C5H2 is formed, Fig. 5. We can not avoid the loop in nodes 2 and 4, because the links of last choice are already used in these nodes, Fig. 5.a). The loop can be

only avoided in node 5, where the route 5B1F3 still exists, Fig. 5.b). We can conclude that the method can not be implemented, if in all nodes, which constitute the loop, the links of last choice are already used.

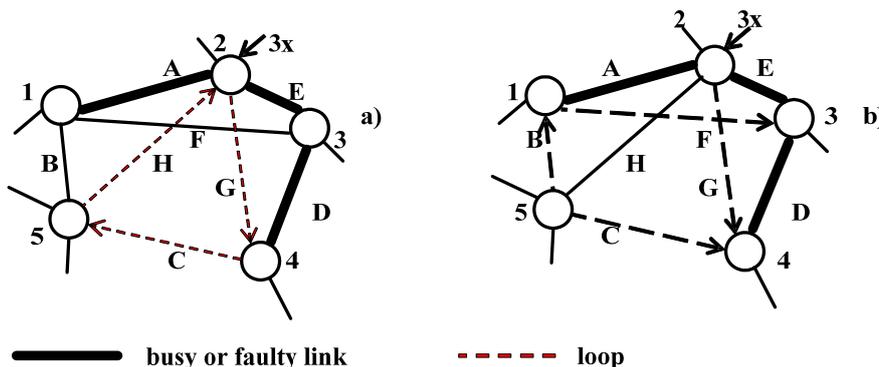


Fig.5. An example how to avoid forming the loop if in some node the link of last choice is not used

Calculation of the efficiency

The probability of forming the loop (P_L) depends on many factors: routing plans, traffic and the link failure probability. Let us calculate the probability that the loop, formed in single-layer, mixed EPU network, can not be carried over in the successful connection. For this network it

can be said that in each network node is possible to choose t links. Let us suppose that the loop, which is formed, passes n network nodes. The probability, that the connection between two network nodes is realized using the route of k^{th} ($k=1,2,...,t$) choice, is p_k . The probability, that the route, which is formed, can not be transformed in the

successful connection, P_N , is equal to the probability that the route in each network node is realized using the link of last, i.e. t^{th} choice:

$$(1) \quad P_N = (p_t)^n$$

The value $1-P_N$ presents the method efficiency for this network, because it expresses the probability that the formed loop will be transformed in the successful connection.

The total probability that the loop is formed and that continues to exist is:

$$(2) \quad P = P_L \cdot (p_t)^n$$

Example 6: For the EPU network, which is presented in [1], the experience shows that the following values can be adopted: $t=3$, $n=3$. The probabilities of taking the route of second and third choice are very small, and here we adopt the greater values than the real ones in order to illustrate method of calculation. Let us suppose that: $p_1=0.9$, $p_2=0.08$, $p_3=0.02$. According to (1), we get $P_N=0.000008$, i.e. the method efficiency is 0.999992. The probability that the loop is formed and that it continues to exist is negligible, according to (2). It is interesting to emphasise that the method efficiency is greater if the loop is realized using greater number of network nodes.

Conclusion

The probability of forming indefinite loops in single-layer telephone network of EPU can be reduced by the adaptation of the software for call routing in network nodes. The method cannot eliminate loops completely. The main condition for its successful operation is that the loop hasn't appeared using the route of last choice in all network nodes making loop. The method cannot be used in networks, where overlap signalling is used, because of the lack of complete information about the destination address. The time used for TTIM table look up is short, because telephone network of EPU has relatively limited capacity.

REFERENCES

- [1] Krajnović, N.: „The Design of a Highly Available Enterprise IP Telephony Network for the Power Utility of Serbia Company“, IEEE Communications Magazine, vol. 47, no 4, April 2009, pp. 118-122.
- [2] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J., Sparks, R., Handley, M., Schooler, E.: „RFC3261: SIP: Session Initiation Protocol“, June 2002.
- [3] ITU-T: „Recommendation Q.763: Signalling system No7 – ISDN user part formats and codes“, December 1999.
- [4] Standardizing information and communication systems: „Standard ECMA-252: Broadband Private Integrated Services Network (B-PISN) – Interexchange Signalling

- Protocol – Transit Counter Additional Network Feature (B-QSIG-TC)“, December 1996.
- [5] Moy, J.: „RFC2328: OSPF Version 2“, April 1998.
- [6] Rekhter, Y., Li, T., Hares, S.: „RFC4271: A Border Gateway Protocol 4 (BGP-4)“, January 2006.
- [7] Chen, E., Yuan, J.: „RFC6286: Autonomous-System-Wide Unique BGP Identifier for BGP-4“, June 2011.
- [8] Havansi, K.: „Patent Number 5,930,236: Method for Preventing Circular Routing in a Telecommunication Network“, United States Patent, July 1999.
- [9] Brocken, F. W. A., Houben, C. G. J.: „Patent Number 5,371,732: Method of Detecting a Routing Loop in a Telecommunication Network, Telecommunication Network for Using the Method, and Detection Means for Use in the Telecommunication Network“, United States Patent, December 1994.
- [10] Talur, D., Fernando, R., Sachdev, A., Yeung, D. M.-K.: „Patent Application Publication Number US 2006/0159034 A1: Method for Providing Loop Free Routing Table in a Router“, United States Patent Application Publication, July 2006.
- [11] Lee, C.-Y., Anderson, L.: „Patent Number 6,879,594 B1: System and Method for Loop Avoidance in Multi-Protocol Label Switching“, United States Patent, April 2005.
- [12] Stampfi, R.: „Patent Number 5,705,998: Method for Routing Telecommunication Calls in a Network“, United States Patent, January 1998.
- [13] Postel, J.: „RFC791: Internet Protocol, Darpa Internet Program Protocol Specification“, September 1981.
- [14] Rosen, E., Viswanathan, A., Callon, R.: „RFC3031: Multiprotocol Label Switching Architecture“, January 2001.
- [15] Deering, S., Hinden, R.: „RFC2460: Internet Protocol, Version 6 (IPv6) Specification“, December 1998.
- [16] ITU-T: „Recommendation Q.931: Digital subscriber signalling system No. 1 (DSS 1) – ISDN user-network interface layer 3 specification for basic call control“, May 1998.
- [17] Lebl, A., Mitić, D., Markov, Ž.: „Influence of Connection Length on Speech Signal Quality in Packet Network of Electric Power Utility“, Revue Roumaine des Sciences Techniques, vol. 56, no. 3, September 2011, pp. 295-304.
- [18] Standardizing information and communication systems: „Standard ECMA-142: Private Integrated Services Network (PISN) – Circuit Mode 64kbit/s Bearer Services – Service Description, Functional Capabilities and Information Flows“, December 2001.

Authors: Mr Mladen Mileusnić dipl.Ing., IRITEL A.D., Batajnički put 23, 11080 Belgrade, Serbia, phone 381-11-3073480; e-mail: mladenmi@iritel.com; dr Aleksandar Lebl dipl.ing., IRITEL A.D., Batajnički put 23, 11080 Belgrade, Serbia, (phone 381-11-3073403; e-mail: lebl@iritel.com; dr Dragan Mitić dipl.ing., IRITEL A.D., Batajnički put 23, 11080 Belgrade, Serbia, phone 381-11-3073425; e-mail: mita@iritel.com; prof. dr Žarko Markov dipl.ing., IRITEL A.D., Batajnički put 23, 11080 Belgrade, Serbia, phone 381-11-3073403; e-mail: Zarko.Markov@iritel.com.