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Calculation of Signalling RTP Packet Error Probability in Internet

***Abstract**. We analyze the errors in the Internet packets carrying telephony signalling information. Error probability, obtained by using redundant packet sending, is also calculated. It is proved that fatal error probability, when signalling packets are sent according to standardized RTP packets, is better than it is allowed in international recommendation for analog signalling systems, but worse than it is allowed for No7 signalling systems. The improvement is achieved by using redundant packet sending, when the error probability drops below the value intended for No7 signalling.*

***Streszczenie.** Przeanalizowano błędy w pakiecie internetowym przenoszącym informację telefoniczną. Stwierdzono, że prawdopodobieństwo błędu w pakiecie zgodnym ze standardem RTP jest mniejsze w układach analogowych ale większe w systemie CCS No 7. Pomocne w tym przypadku może być przesłanie pakietu rezerwowego. (**Prawdopodobieństwo błędu sygnału w protokole RTP**)*

Keywords: Real time Transfer Protocol, telephony signalling, error probability
Słowa kluczowe: protokół RTP, telefonia internetowa.

Introduction

The multimedia connection over IP is performed in two ways: first is according to H.323 protocol suite, [1], and second according to SIP (Session Initiation Protocol, [2]). We distinguish connection establishing phase and user data transfer phase in both cases. The setting-up of the connection is performed by protocols that guaranty reliable transfer of signalling data i.e. error detection and retransmission, (TCP, Transmission Control Protocol, [3]). For transfer of user information, the RTP (Real time Transfer Protocol, [4]), is used. RTP is intended for fast data transmission, but it does not retransmit corrupted data.

The telephony over Internet (ToIP) may be viewed as simple case of multimedia connection. In the setting up phase of ToIP connection, telephone channel is used for transfer of relatively long signals. The main question that arises is: is it possible to use RTP for transfer of telephone signals i.e. to speed up the connection establishing in the network where circuit switching is dominant? The methods described in [5] give positive answer. There are two ways to transfer the telephone signals by RTP. One of them recognizes signal and transfers the signal code (event code) and description of basic signal characteristics. The second way is to transfer the signal parameters without recognition of the signal.

Due to real-time nature of RTP, retransmission of corrupted RTP packets is not provided. Since the loss of the signalling RTP packet may cause signalling error, it is important to determine probability of rejection (i.e. loss) of RTP packet.

In this paper we calculate the influence of bit errors on telephone signal transmission in both methods. The main goal is to calculate the probability of unusable signalling RTP packet in the function of bit error rate (BER) and compare it to the probability of signal error in existing telephone signalling systems.

Basic characteristics of RTP

Real-time Transport Protocol (RTP, [4]) is a protocol often used for fast transfer of user information in Internet (IP). RTP is considered to be the user level, and UDP *(User Datagram Protocol)* is used as transport protocol. The real time transport of information is possible using fast transfers (i.e. transfers without error correction}. For increase of quality of audio data, multiple sample transfer is used, as explained in [6]. Similar increase in transfer quality is accomplished by triple transfer of telephone signals in FR (*Frame Relay),* [7].

Besides transfer of user information, RTP is also used for transfer of classic telephony signalling information across IP. At the beginning of the RTP packet exists the standard header, consisted of various fields packed into the 12 octets, [4]. The fields significant for transfer of telephone signals, tones and digits, are *payload type (PT), sequence number* and *timestamp.* One RTP packet is shown in Fig. 1.

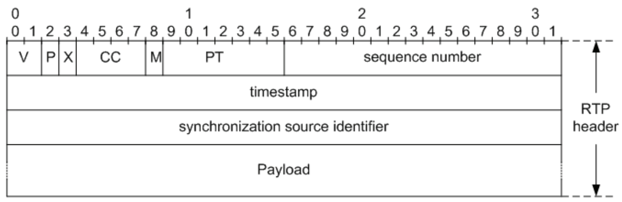

Fig.1. RTP packet

The payload type field can have only one value, because the payload of one RTP packet can be only of one kind. Some values of payload type field are given in [8]. Let us emphasize that payload type values 96-127 are planned for dynamic assignment. Sequence number of RTP packets is used for the detection of packet loss.

Timestamp is used for synchronization of payload transferred by RTP.

Payload field of RTP packet carries user payload. In this case, we are interested in part of payload carrying signalling information. Signalling information can be transferred in two ways: by payload carrying the information about signals as events, [5], section 3, and payload carrying the signal parameters, [5], section 4.

The formats for transfer of signals in these two ways are shown in [5], figures 1 and 2, respectively.

RTP payload carrying signalling events

When we study the transfer of telephone signals over Internet we observe the following chain: outgoing telephone circuits, outgoing gateway, Internet, incoming gateway, incoming telephone circuit. The transfer of telephone signals as events needs a gateway that is able to recognize the type of signal. This approach requires more complex gateway, the signal transfer starts later (due to recognition time) but gives the possibility of simpler transfer. That way, the incoming gateway receives the event code and may reconstruct the signal with prescribed characteristics regardless of the signal characteristics in incoming gateway.

The RTP payload fields significant for transfer of signal as an event are *event, volume, duration* and *end (E)*, Fig. 2. The values representing event codes are given in [5], (DTMF 0-15), [9], (data, fax and text telephony, 23-40, 49, 52-63), table 4 (E.182 line events, 64-89), table 5 (extended line events, 96-112) and [10], (trunk events, 121-137, 144-159, 174, 211). The payload of this RTP packet consists of 4 octets. This RTP packet and its payload will be called the RTP packet and RTP payload of first kind (payload 1).

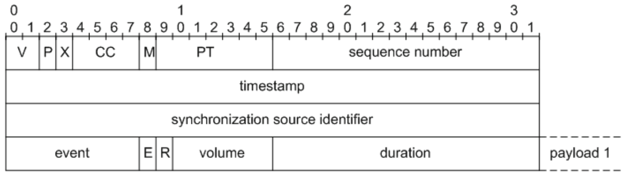

Fig.2. RTP packet of first kind

RTP payload carrying signalling parameters

All signal parameters are transferred by this payload: *modulation* frequency, indicator of frequency division by 3*(T)*, power level *(volume)*, *timestamp, duration,* end, cadence and signal *frequency.* In this type of transfer the outgoing gateway does not recognize the signal type. It is required that the transferred signal is similar to source signal as much as possible. That means the corrupted signal will be transferred regardless of the possibility of its recognition in incoming gateway. The incoming gateway may recognize the signal or discard the signalling packet. This RTP payload consists of 4 common octets and 2 octets for each signalling frequency, Fig. 3. As usual, R bits are reserved bits and are ignored at the receiver. This RTP packet and its payload will be called the RTP packet and RTP payload of second kind (payload 2). The advantage of this transfer method is the lack of waiting time for signal recognition that may last several seconds.

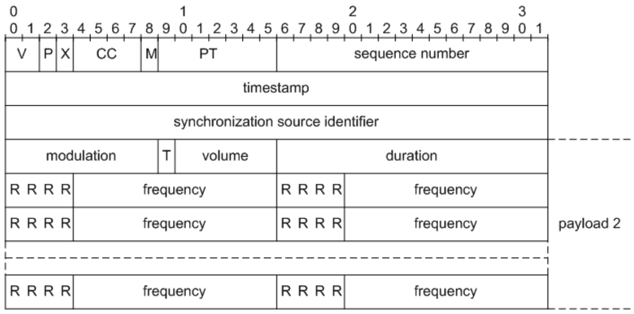

Fig.3. RTP packet of second kind

Error impact

Our main interest is to investigate the influence of bit errors (generated during transmission from sender to receiver) in both methods of transfer of signalling RTP payloads. In estimation of this error influence we distinguish two cases.

First case is appearance of so-called fatal errors. Fatal error happens when only one corrupted bit causes useless signalling RTP packet. For example, these errors are in fields: payload type, marker bit M.

The second case is sustainable error. The sustainable error is an error that changes the value of some packet field i.e. value of signal parameter, but this value may be used. For example, sustainable errors are bit errors in fields describing power level or frequency, if the wrong value is in tolerable area.

In the observation of the error impact we adopt some realistic assumptions:

a1: probability of error of one bit, *p,* is equal for all transmitted bits, (i.e. corrupted bits are randomly distributed). For each value of probability error rate, the number of corrupted messages/packets is largest in the model of channel with randomly distributed errors. It is pointed out that the concentration of errors in clusters or bursts (Gilbert-Elliott model) has a positive impact on the number of corrupted messages/packets, [11].

The probability $P_{p,n}$ that a packet, consisting of *n* bits, contains at least one corrupted bit is exactly $P_{p,n}=1-(1-p)^n$. In modern telecommunications the probability of corrupted bit *(p=BER, Bit Error Rate)* is in range 10^{-9} - 10^{-5}. In that way $P_{p,n}=n \cdot p$ is very good approximation.

a2: probability of wrong packet header is equal for both kinds of packets,

a3: probability of two or more errors in signalling RTP packet may be neglected,

a4: basic payload of first kind consists of 4 octets. Basic payload of second kind consists of 6 octets i.e. of first 6 octets from Fig. 3,

a5: time units from field *timestamp* and field *duration* are expressed in sampling periods i.e. time unit is 125 µs,

a6: one packet presents signalling time interval of t_s = 50ms = 400*timestamp units.*

According to adopted assumptions and known payload formats of first and second kind we may now determine the probability of fatal error for packets of first and second kind.

Errors in packet header

There are no redundant fields in packet header, [4]. Accordingly, each corrupted bit causes fatal error i.e. invalid packet. The probability of this event is 96·p.

Errors in payload 1

We can see from Fig. 2 that each corrupted bit in field *event* causes fatal *error* in payload 1. Namely, error causes wrong payload type i.e. wrong signal in incoming gateway. The probability of this event is 8·p.

The error of reserved bit, *R,* is without influence.

The errors in the fields *E, volume* and *duration* may cause sustainable error.

Each telephone signal is transmitted by a number *k-1* non-final packets with value of *duration* field *duration(i) = 400* (*i =1, 2, ..., k-1)* and one final packet *k, duration(k) < 400.*

The correct values of fields *E(n)* and *duration (n)* in the nth packet are

E(n)=0 and *duration(n) = duration (n-1) + 400, n = 1, 2, ..., k-1* or

E(n)=1 and *duration (n) < duration (n-1) + 400, n=k.*

We observe three cases with incorrect values of the fields *E(n)* and *duration(n)*:

c1. E(n)=0 and *duration(n) > duration (n-1) + 400*

and *E(n)=1* and *duration (n) > duration (n-1)+ 400*

It is clear that the error is in *duration (n)* field. According to assumption a3 we accept that the field *E(n)* is correct. We adopt the values

E(n)=0 and *duration(n) = duration(n-1) + 400* or

E(n)=1 and *duration(n) = duration(n-1) + 399*

and we take the packet as errorless.

c2. E(n)=1 and *duration (n) = duration(n-1) + 400*

It is obvious that one error exists. In this case we consider two possibilities. The corrupted bit is in field *E(n)* or in field *duration (n).* Let us examine the probabilities of both possibilities. For the sake of simplicity, we take *n = 1.* First possibility is that the error is in field *E(n)* and the probability of this event is p.

The second possibility is that the error is in field *duration(n)*, which means that the value $400=2^8+2^7+2^4=256+128+16$ of this field is obtained summing the correct value that is less than 400 (144 or 272 or 384) and value of one corrupted bit (which is 256 or

128 or 16). If the correct value of *E(n)* is *1* then field *duration(n)* takes the true value between 0 and 399.

The probability that the true value of field *duration (n)* is one of 144, 272, 384 is 3/400 i.e. the probability that the error is in field *duration(n)* is 3·p/400 that is negligible in comparison with p. We conclude that the case *c2* contains the error in field *E(n),* with great probability, so the RTP packet may be exploited. Similar calculation may be done for case *n>1.* We always obtain the results that show much greater probability of error in field *E(n)* than probability of error in field *duration(n).*

c3 E(n)=0 and *duration (n) < duration (n-1) + 400*

It is obvious that an error exists. If the value of field *duration (n)* is such that satisfies equation

(1) $duration(n-1) + 400 - duration(n) = 2^j, \; j = 0, 1, 2, ..., 9$

The error in field *duration(n)* is possible. If the value of field *duration(n)* does not satisfy the equation (1)*,* error is in field *E(n).* According to this conclusion we may say that, in case *c3.,* with probability between 1/400 and 9/400, the error is in field *E(n).* In that way the signalling RTP packet may be used.

We may conclude that the error in fields *E(n)* and *duration* of payload 1 is not fatal under adopted assumptions.

The field *volume* is used for transfer of DTMF signals only. This field may take value from 0 to -63dBm0, but for payload 1 the values from 0 to -35dBm0 are important. The errors giving greater value of DTMF signal are not important. Important errors are these giving power levels less than -35dBm0 (must operate limit) and -55dBm0 (must reject limit) instead of levels in the area 0 to -35dBm0. These errors happen when most significant bits, MSBs, (2^6 and 2^5) take the values 1 instead of 0. Fortunately, these bits with errors may be detected by means of fields *event (0-16)* and *duration(>0)* so the errors in field *volume* doesn't cause invalid packet.

We may conclude that the probability of fatal error in payload 1 depends only on errors in field *event* and has value 8·p.

Errors in payload 2

The errors in transfer of payload 2 may or may not be fatal. Like in payload 1, sensitivity to errors depends on field containing error. (Errors in reserved fields/bits (R) are without influence).

The error in field *T* causes useless signalling RTP packet with payload 2. It should be noted that the frequencies determined by this parameter are used very rarely.

The errors in the field *volume* may be detected using fields *modulation* and *duration* in the similar way as for payload 1.

The field *duration* is very sensitive to errors due to lack of end marker. The end of signal is detected according to the value of this field (<400) in the final packet. Every telephone signal is transmitted by a number of *k-1* non-final packets with value of *duration* field *duration(i) = 400 (i = 1, 2, ..., k-1)* and one final packet.

The average number of bits with value "1" in non-final packets is 6, Fig. 4.

Final signalling packet is with value *duration(k) < 400.*

Two kinds of errors exist. First is error which causes values of *duration(j) > 400, j = l, 2,..., k.* These are not the fatal errors. We may use these packets with corrected value *duration(j) = 400.*

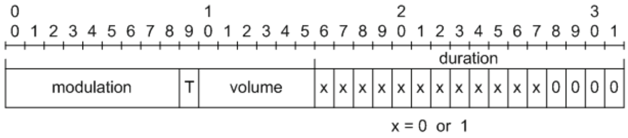

Fig.4. Duration field of non-final packet of second kind

Second kind of error is where one of "1"s in the *duration* field of non-final packet is received as "0". On the receiving end, this packet is considered as final and fatal error is caused. The probability of this error is 6·p.

The fields *modulation* and *frequency* may transmit the pulse or pause of a signal. We suppose that the probabilities of each of these events are equal (0.5).

In the case of pulse transfer, tolerable error is such one that does not change the frequency for more than 0.5% to 1.5%, [12]. That means the error of one of three least significant bits (LSBs) in field *modulation* is tolerable, Fig. 5. The error of one of six (most significant bits) MSBs is not tolerable because this error drastically changes the frequency. The probability that fatal error is in *modulation* field is 6·p. In the same manner, error in field *frequency* may be tolerated for 5 LSBs and may not be tolerated in 7 MSBs, Fig. 5. During the transfer of pulse of a signal, the probability of fatal error in fields *modulation* and *frequency* is 13·p.

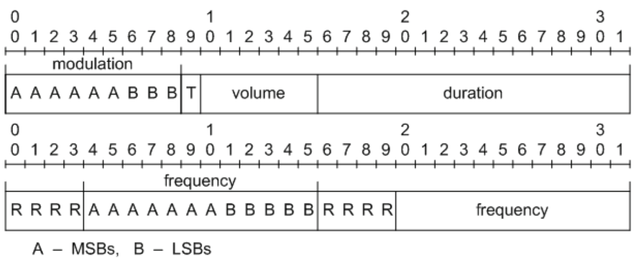

Fig.5. Allocation of MSBs (A) and LSBs (B) in modulation and frequency fields of packet of second kind

In the case of pause transfer, the errors in the fields *modulation* and *frequency* may be detected. According to assumption a3, if there are no values in fields *marker (M)* and *volume,* error is obvious and may be corrected.

The total probability that the error in fields *modulation* and *frequency* causes the useless payload 2, in the case of sending the signal, consisting of only one signalling frequency, is 0.5·13·p = 6.5·p.

Total probability that a corrupted bit causes fatal error in payload of signalling RTP packet of kind 2 is 12.5·p.

R2 signals are formed of two signalling frequencies. As for the first considered field for frequency in the payload 2, the second considered field for frequency has also the fatal error when one of 7 MSBs are with error.

Each R2 signal is sent with fatal error, if the error exists on 6MSBs of the field modulation, or on 7 MSBs in field *frequency*, or if the field *duration* is not correct. Total probability of fatal error when R2 signals are transferred using RTP packet of kind 2 is, thus, 16·p.

Redundant packet sending

Redundant signal sending is used in order to improve the reliability in message sending over Internet. The method can be applied by presenting one signal in two different fashions, or by sending the message, which presents the signal, two (or more) times over Internet. Also, it is possible to send the detected signal first as the primary content together with the signals, which are retransmitted, because they are detected in the previous detection intervals. The detected signal, which we consider, will be retransmitted in the following packet together with the signal, which will be detected then. In all these cases, if one of the packets, representing the considered signal, is not transmitted correctly to the receiving side, it remains the second representation. The detected signal can be reproduced correctly according to this, second, representation.

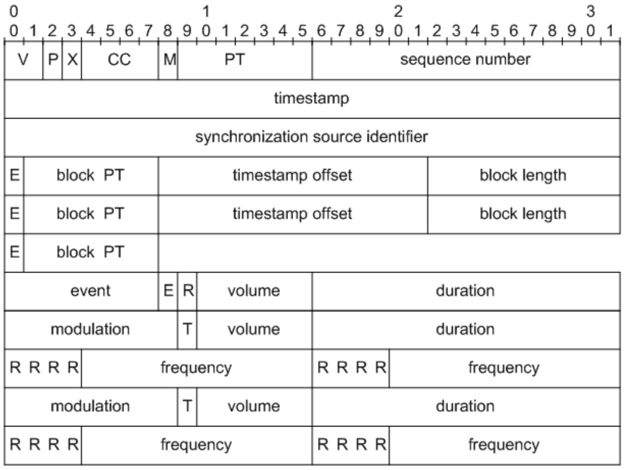

Fig.6. The structure of the redundant packet

The structure of one, redundant packet, is presented in figure 6. This packet represents three events: the last one is a new event (primary data sending), and the previous two are older events, i.e. events detected in two previous time intervals (data retransmission).

After the standard RTP heading, follows the part of the packet, defining that the packet is of the redundant type and that it consists of three events. First two groups of 4 bytes define the kind of data for first two events (redundant description), and the data for the third event are defined by only one byte (primary description). The number of bytes in this part of packet is 4·(e-1)+1, where e is the total number of events presented in this packet.

The meaning of the data in this part of packet is as follows:

- E – information is the concerned event the last event in the set of events, presented in that packet (E=0) or not (E=1);
- block PT – definition of the kind of presentation for that event. It can be the presentation in the form of payload 1 (using event code) or in the form of payload 2 (using signal frequencies). In our example, the first event is presented as event code, and the second and the third event by signal frequencies;
- timestamp offset – time shift of the signal beginning from the last signal, presented in that packet;
- block length – the number of bytes, which are, in the packet continuation, in more detail presenting that event.

The presentations by the event code and using signal frequencies are already presented in the paper.

Error analysis in redundant packet

The errors in the fields E and block PT, which define the structure of the redundant packet, are, mainly, fatal. Concerning the field E, the problem is that this part of packet for the last event in the sequence is consisted of only one byte. If this bit is changed from 1 to 0 in the part of payload, which is not intended for the last event in the packet, the beginning of the field timestamp offset would be interpreted as the beginning of the presentation for the event, i.e. would be interpreted as the field event. If, in the part of the payload intended for the last event, E is changed from 0 to 1, the beginning of the field event would be interpreted, in our example, as the beginning of the field timestamp offset for the third event. Concerning the error in the field block PT, it can be said that this part of the packet defines in the wrong way what kind of presentation is used and, also, disables the further separating parts of the packet. Therefore, the error probability, caused by this part of the packet, is 8·p.

The length of the field block length is 10 bits, but, in the case that the signal is presented as the event, the value 4 (2^2) is only possible in this field. The errors on all other bits of this field can be reconstructed in order to give the value 4. Therefore, the error in the field block length is sustainable when the signal is presented as the event.

In the case that signal is presented using signalling frequencies, from which it is constituted, the value in the field block length will be 4+2f, where f is the number of signalling frequencies. Usually, each signal is constituted of one or two signalling frequencies, i.e. usually the number of bytes is 6 or 8. The error is fatal for such values of 2^j where is *j=1,2,3*. (Even more, the signals constituted of 5 signalling frequencies maximum can be represented when these 3 bits are used). The other errors in the field block length in this case are sustainable errors. Thus, the probability of fatal error is, for this situation, 3·p.

The field timestamp offset is constituted of 14 bits, so it can present time duration till 2s. For longer time duration, this field is not important. This is the redundant field, i.e. the values in the field are related to the values in the field duration for the signal presentation by the event code or by the signal frequencies. Thus, if some inconsequence in signal duration exists, this field can be ignored, i.e. it does not produce fatal error.

Error probability analysis in redundant sending of signal

In the previous chapters, we have seen that the error probability in packet sending can be expressed (approximately) by adding error probabilities for message parts, whose wrong transmission produce fatal errors. If the error probability for each bit is p, then the error probability for the whole packet is n·p, where n is the number of bits, whose irregular transmission produce fatal error. For signal transmission as the event code, n=104 (the error is fatal, when it is in header, or in field event), and for signal transmission using signal frequencies, n=112.

Let us suppose, at first, that the redundancy is achieved only by repeating two same messages. The transmission is not correct, if both packets are transferred with fatal error, i.e. the error probability is $n^2 \cdot p^2$. The concrete value for the transmission using the event code is $10816 \cdot p^2$, and for using signalling frequencies $12544 \cdot p^2$.

Let us consider now the signal sending by redundant packet, where we use one primary content and one retransmission, and the signal is represented using signalling frequencies. The error is fatal in the case of the error in heading (96·p), or in the part defining the redundant packet (8·p for E and block PT, and 3·p for block length), or in the part defining signalling frequencies (16·p). It is n=123·p altogether for the fatal error. For two consecutive packets, the error is, approximately, $15129 \cdot p^2$.

Conclusion

The transfer of signalling telephone signals by RTP may be recommended for lines of good quality. The probability of invalid signalling RTP packet is about 100·BER. For modern transmission media, where the value of BER is 10^{-9} – 10^{-8}, the probability of invalid signalling RTP packet may be estimated as 10^{-7} – 10^{-6}. The errors in the packet header are the main cause of invalid packets.

From simple calculation we may see that the reliability of signal transfer is approximately same in both methods of transfer. The redundancy of packet fields in payloads diminishes the probability of useless packet.

The comparison of invalid RTP packet probability to the largest allowed error probability in transmission of R2 signals, (10^{-5} to 10^{-4}, [13]), and to the largest allowed probability of undetected error in CCS No 7, (10^{-10}, [14]), shows that the probability of invalid (useless) signalling RTP packets is between these two values.

The probability of R2 signalling improvement can be achieved by redundant signal transferring. The signals are repeated twice. We analyzed the fatal error probability for two special cases. The fatal error probability in those situations for the BER of 10^{-9} – 10^{-8} is less than $1.6 \cdot 10^{-12}$, which is better than the largest allowed error probability for CCS No 7.

REFERENCES

[1] ITU-T: H.323 - Packet based multimedia communications systems, 1999

[2] Rosenberg, J., Schulzrinne, H., Camarillo, G., Johnston, A., Peterson, J, Sparks, R, Handley, M., Schooler, E.: SIP: Session Initiation Protocol, RFC 3261, June 2002

[3] IETF: Transmission Control Protocol, RFC 793, September 1981

[4] Schulzrinne, H., Casner, S., Frederick, R. and Jacobson, V.: RTP: A Transport Protocol for Real - Time Application, RFC 3550, July 2003

[5] Schulzrinne, H. and Petrack, S.: RTP Payload for DTMF Digits, Telephone Tones and Telephony Signals, RFC 4733, December 2006

[6] Perkins, C., Kouvelas, I., Hodson, O., Hardman, V., Handley, M., Bolot, J. C. Vega-Garcia, A., Fosse-Parisis, S.: RTP Payload for Redundant Audio Data, RFC 2198, September 1997

[7] FRF.11.1: Voice over Frame Relay Implementation Agreement, Frame Relay Forum Technical Committee, December, 1998

[8] Schulzrine, H., Casner S.: RTP Profile for Audio and Video Conferences with Minimal Control, RFC 3551, July 2003

[9] Schuzrinne, H., Taylor, T.: Definition of Events for Modem, Fax and Text Telephony Signals, RFC 4734, December 2006

[10] Schuzrinne, H., Taylor, T.: Definition of Events for Channel – Oriented Telephony Signalling, RFC 5244, June 2008

[11] Markov, Z. and Mitic, D.: The influence of short error clusters on CCS No 7 link availability, Int. J. Electron. Commun. (AEU) 56(2002), No 3, pp 205-207

[12] ITU-T: Recommendation E.180/Q35: Technical characteristics of tones for the telephone service, March 1998

[13] ITU-T: Recommendation Q.458 - Reliability of interregister signalling, 1993

[14] ITU-T: Recommendation Q.706: Specifications of signalling system No. 7: Message transfer part signalling performance, March 1993

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