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.

Keywords: Real time Transfer Protocol, telephony signalling, error probability

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.

Payload field of RTP packet carries user payload. In this case, we are interested in part of payload carrying signalling information 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], [7], 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).

### 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.

### 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 \( 10^{-5} - 10^{-3} \). 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 \( t_s = 50\text{ms} = 400\text{timestamp units} \).

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 \( n \)th packet are

\[
E(n)=0 \text{ and } duration(n) = duration(n-1) + 400, n = 1, 2, ..., k-1 \text{ or } \]

\[E(n)=1 \text{ and } duration(n) < duration(n-1) + 400, n=k.\]

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

\[E(n)=0 \text{ and } duration(n) = duration(n-1) + 400 \text{ or } E(n)=1 \text{ and } duration(n) = duration(n-1) + 399 \]

- **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 \( 400=2^8 \cdot 2^2 + 2^7 + 256+128+16 \) 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...
Errors in payload 2

The errors in transfer of payload 2 may or may not be fatal. If errors occur in the payload, the probability of fatal error in the duration field depends on the event of a pulse, as well as on the effect of other errors in the field. The field volume is used for transfer of DTMF signals only. This field may take value from 0 to -63dBm, but for payload 1 it can be any value from 0 to 9/400 the error in field volume. 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.

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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. Each 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(i) > 400, j = l, 2,..., k. These are not the fatal errors. We may use these packets with corrected value duration(j) = 400.
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,
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The presentations by the event code and using signal
frequencies. In our example, the first event is
presented as event code, and the second and the third
signal 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 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 \(n\cdot p\), 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 \(f\) 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 (96p), or in the part defining the redundant
packet (8p for E and block PT, and 3p for block length), or
in the part defining signalling frequencies (16p). It is
\(n=123p\) 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

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 \(4\cdot(e-1)+1\), 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 \(2\) where
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– 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
[8] Schulzrinne, H., Casner S.: RTP Profile for Audio and Video Conferences with Minimal Control, RFC 3551, July 2003

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