FACTA UNIVERSITATIS (NIŠ)
SER.: ELEC. ENERG. vol. 23, April 2010, 63-72

Analysis of Different Methods for Caller Identification Sending Over Internet

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Abstract: This paper describes different methods of calling subscriber number sending over Internet. The first method uses the coding of signal samples by A-law of compression. In the second one, each bit of this signal is expressed as the telephony event code according to the standardized way of Internet event code presentation. The third one is the new, original method, where the telephony event code presents the whole signal of caller identification. These three methods are compared according to the channel capacity and according to the error probability for the signal sending over Internet.

Keywords: Caller identification, internet, telephony code event, ringing signal, error probability.

1 Introduction

IN the process of integrating packet and traditional telecommunication network, the different methods of their interworking and the different places of their contact are possible. One possible place of their contact is the connection of classic telephone equipment (telephone, fax) and telephone exchange by Internet. In this case the classic telephony signals (DTMF, line signals, signals of fax, caller identification signal, etc.) are transmitted across Internet using special procedures, defined in the Internet recommendations. In the beginning the recommendation RFC2833 was used [1], but it was later divided into three recommendations [2–4]. According to these recommendations, it is, simply, possible to present the signal samples by some law of coding (for example A-law) and to send this signal in the Internet packet. The other possible method is to define signal parameters (frequency, level

Manuscript received on January 10, 2010.
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and the duration of signal), and to send these parameters by Internet packet. And, finally, it is possible to determine the meaning of the signal and to send the packet with telephone event codes, presenting this meaning.

This paper deals with caller identification (CID) signal sending across Internet, using different methods of transmission.

2 The structure of CID signal and its sending

The main application of CID signal is to send the identification of the calling subscriber to the called subscriber over traditional subscriber line. This identification, and other data, can be sent in the on-hook state associated with the ringing signal, or without it.

The CID signals are sent from the Local Exchange (LE) to the Terminal Equipment (TE) as analog signals. They are coded as Frequency Shift Keying (FSK) signals.

The CID signals are sent on the subscriber line at the rate of 1200 baud, which means that the duration of each bit is 833 μs. The signal of frequency 1300 Hz represents binary 1 (mark bit), and the signal of frequency 2100 Hz represents binary 0 (space bit) [5, 6].

The parts of the CID message are successively, without interruption in sending:

- Channel Seizure Signal (CSS): the block of 300 continuous bits of alternating ”0”s (space bit) and ”1”s (mark bit), the first bit is ”0”, and the last bit is ”1”;
- Mark Signal (MS): the block of 180 ± 25 ”1”s;
- Message type (MT): 8 bits, which define transmitted message;
- Message length (ML): 8 bits, which express the length of Presentation Layer Message;
- Presentation Layer Message (PLM): contains different parameters, as date and time of the call, calling line identity, etc., practically 75 bytes maximum;
- Frame Check Sequence (FCS) 8 bits, at the receiving end used for the control of the correctiness of the received message. It is followed by 1 to 10 ”1”s.

Each byte in MT, ML and PLM is preceded by the start bit (bit with the value ”0”), and followed by the end bit (bit with the value ”1”).

3 CID signal sending over Internet

Figure 1 presents block-diagram of CID signal sending over Internet [7].

CID signal is generated in the local exchange (LE). The content of the CID signal is formed in the Common Control Unit (CCU) of the exchange. CID signal is shaped in the modulator (MOD1) and, after that, sent on the subscriber line.

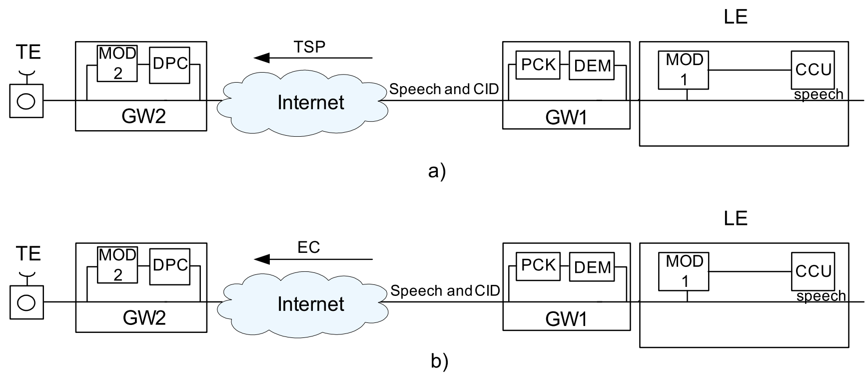

Fig. 1. The methods of CID signal sending over Internet: a) transmission of telephone signal parameters (TSP); b) transmission of event codes (EC)

The gateway (GW1) is situated at the Internet ingress port. The demodulator (DEM) in GW1 demodulates CID signal, and the packetizer (PCK) forms packets of CID signal, which are sent over Internet.

Telephone signals in Internet packets can be represented by telephone signal parameters (TSP) (frequency, modulation frequency, level, duration) ([2], *section 4. RTP Payload Format for Telephony Tones)*, as it is presented on figure 1a). They can be also defined as event codes (EC) ([2], *section 2. RTP Payload Format for Named Telephone Events)*, as it is presented on figure 1b).

On the other side of packet connection, i.e. at the Internet egress port to the classic telephone network, depacketizer (DPC) in gateway (GW2) converts Internet packets in the content of CID message. This content is then sent to the modulator (MOD2), which forms the CID message towards the terminal equipment (TE).

The characteristics of the TSP usage are faster transmission, simpler realization, but also the possibility to transmit invalid signal. For example, the signal on the subscriber line can be attenuated more than it is allowed. The parameters of this attenuated signal will be measured in the gateway and sent over Internet [7].

The second method requires more complicate gateway and more time to recognize the signal, because the meaning of the signal must be detected, but the (tone) signals are replayed in standard shape and level.

The gateway, which sends the signal samples without detecting the signal parameters, or its meaning, is the simplest for realization, guarantees the faster transmission, but the needed channel capacity is greater than in the other two other explained cases.

4 The structure of Internet message packets

The structure of one Internet message packet (their parts and the duration of these parts) is shown in the figure 2. The RTP payload length depends on the content which is sent, and the length of the other packet parts is precisely defined, as it is presented in figure 2.

Ethernet header (14 bytes)	IP header (20 bytes)	UDP header (8 bytes)	RTP header (12 bytes)	RTP payload	FCS check-sum (4 bytes)

Fig. 2. The structure of Internet message packet

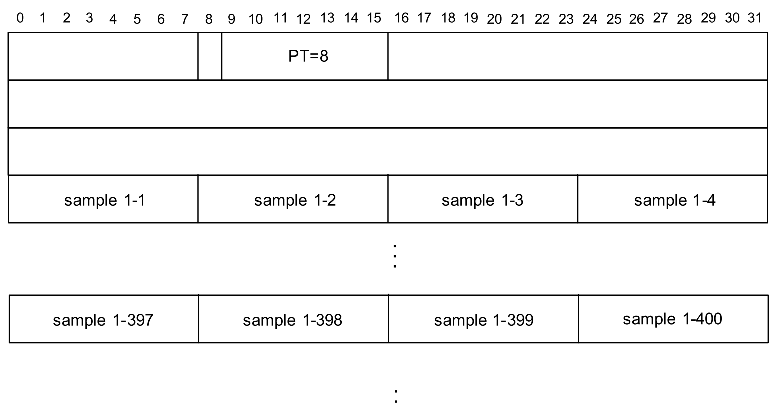

Fig. 3. CID signal coding using A-lawt

Figure 3 presents RTP part of Internet packet when A-law of compression is used for coding of CID signal. The part of the CID signal in figure 3 has 400 samples (it lasts 50 ms). Each row in figure 3 has 4 bytes. The 3 rows at the top (12 bytes) are the RTP header, and the remaining rows are the RTP payload (samples from 1-1 to 1-400). After that, the next part of the CID signal may be sent using the similar packet, or some other method of transmission. In the RTP header, it is important to mention the payload type (PT), which defines the data type in the RTP payload. $PT = 8$ means that the RTP payload content are the samples coded using A-law of compression. This PT is static, meaning that the value 8 at this place in RTP header always defines samples coded using A-law of compression.

Figure 4 presents the RTP part of the Internet packet if each bit of CID signal

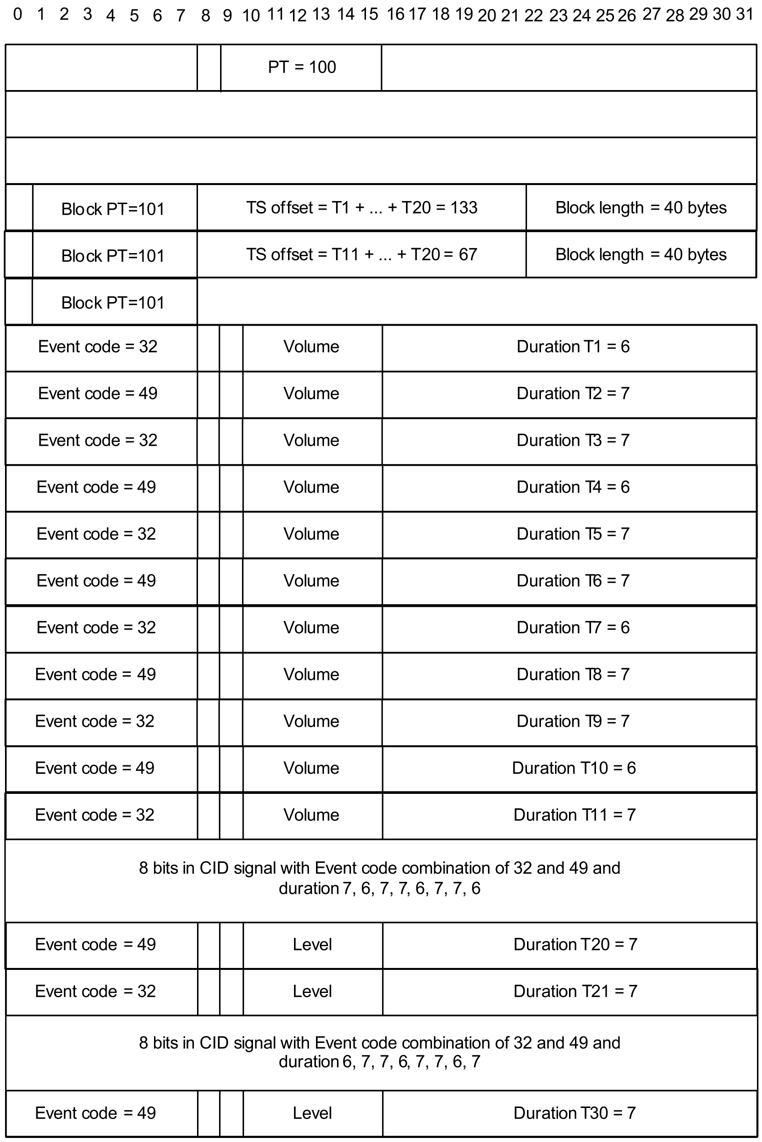

Fig. 4. The message presenting each CID signal bit as the separated event

is presented as the event code [8]. Dynamic $PT = 100$ is used in RTP header. (Dynamic PTs take the values from the range 96-127). The meaning of this PT is not *a priori* defined; it is agreed between two gateways by special messages (SDP messages) before signal sending. In this case, it is agreed by SDP messages that the PT is intended for redundant packet, composed of telephone events ($PT = 101$).

The payload format for each event has 4 bytes, with the first byte meaning the event code [9], and the other 3 bytes defining the volume and the duration of the signal. RTP payload is composed of 3 blocks with 10 events each, so each block length with event code descriptions is 40 bytes. The values of event codes are 32 (space, or 2100 Hz signal) and 49 (mark, or 1300 Hz). The duration of these signals is 6 or 7 sampling interval, because the duration of each bit in CID signal is 833 μs. The example in figure 4 presents the packet for sending 3 bytes of CID signal (30 bits including start and stop bit), i.e. 25 ms of CID signal.

Figure 5 presents the message with the event code for CID signal (90) [8]. It is the first byte in the RTP payload. After that the value of the second byte defines the remaining byte number in the RTP payload ($n+2$, n being the length of the RTP content which codes CID signal). The third byte is the number of 01 pairs in CSS part of CID signal (150), and the fourth one is the length of the MS part in CID signal (180). The remaining part of this message is the content of MT, ML and PLM (bytes $1,2,\ldots,n$).

This message presents the complete content of the CID signal.

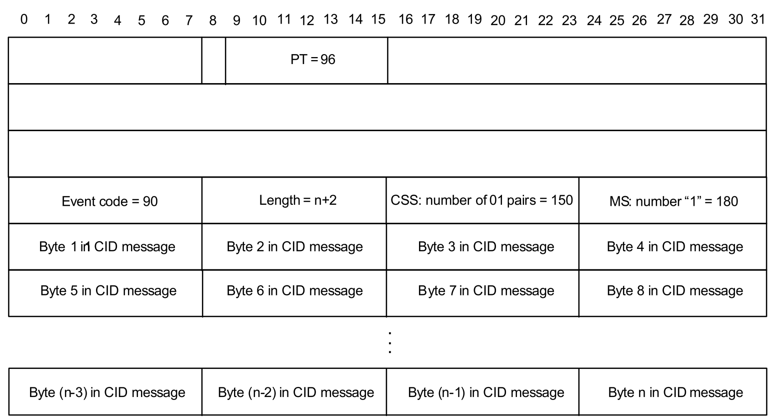

Fig. 5. The message with the event code for the whole CID signal

5 Comparison of different methods for CID signal sending over Internet

The methods for CID signal sending over Internet can be compared according to different criteria: the delay of the reproduced signal relative to original signal, gateway complexity for different methods of realization, channel capacity for CID signal transmission over Internet, error probability in CID signal transmission. In this paper the comparison is made according to last two of these four criteria.

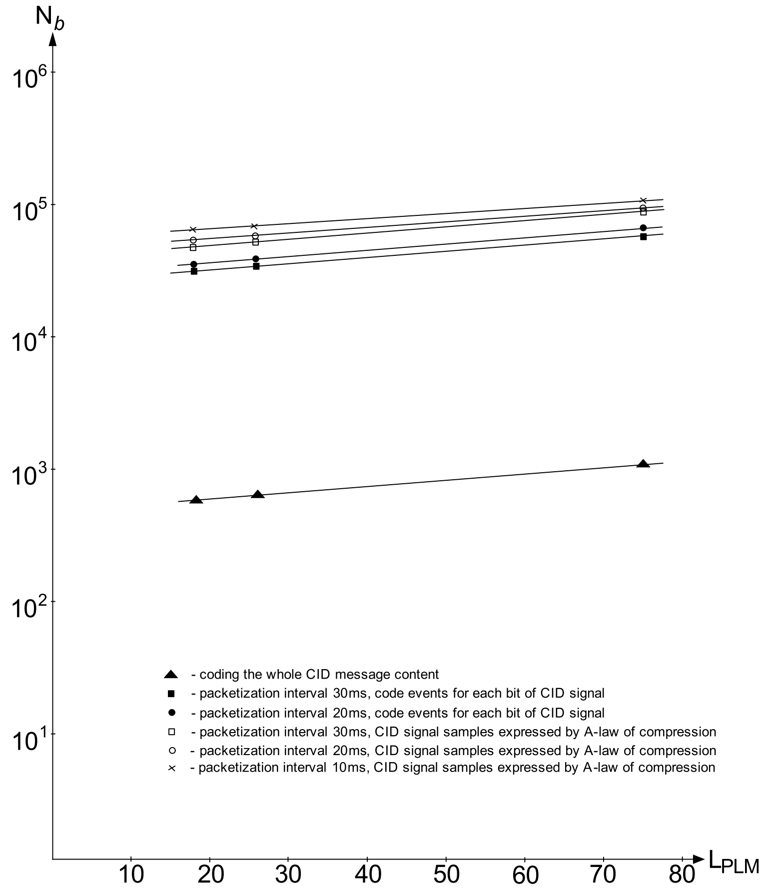

Fig. 6. The channel capacity for CID signal transmission using different coding methods

Figure 6 presents channel capacity for different methods of CID signal transmission over Internet. The channel capacity is expressed in the function of PLM length. Three characteristic lengths of PLM in figure 6 are:

- 18 bytes for the subscriber number of 6 digits;
- 26 bytes for the subscriber number of 14 digits;
- 75 bytes, the maximum expected PLM length.

For CID signal sending as samples coded by A-law of compression and for each CID signal bit coding as the event code, the channel capacity is calculated for different time intervals (10 ms, 20 ms, 30 ms), packetized in one Internet packet.

Presenting each CID signal bit by the event code doesn't decrease noticeably the channel capacity, comparing to sending CID signal samples coded by A-law of compression. In the first case each CID signal bit is presented by 4 bytes as the event code, and for the second case each signal bit has the duration of 833 μs, or 6.66 sampling intervals at average. Comparing to these two methods, when the packet, which expresses the CID signal by the special code for the whole signal, is used, the channel capacity is significantly less. This capacity is 1–1.5% of the capacity needed for sending CID signal samples coded by A-law of compression.

CID signal is sent over Internet using RTP protocol. This protocol is designed to transmit signals in real time and it hasn't possibility to retransmit the errored messages. The errored packets are rejected. The error is determined by calculating checksum of the message and comparing this calculated value with the data in the FCS part of packet.

The error probability in message transmission over Internet is:

$$P_E = 1 - P_{CT} = 1 - (1 - BER)^n \tag{1}$$

where P_{CT} is the probability of correct message transmission, *BER* is the error probability on one bit in Internet message (*Bit Error Rate*), and *n* is the number of bits in Internet message

For the small values of BER (practically, the value of BER is not expected to be greater than 10^{-6}), from (1) follows:

$$P_E \approx n \cdot BER = (L_{HEADER} + L_{RTPPAY}) \cdot BER \tag{2}$$

where the new abbreviations are: L_{HEADER} is the length of all headers in the message and L_{RTPPAY} is the length of the RTP payload.

From (2) it can be seen that the overall bit number in Internet message is calculated as the sum of lengths of all headers (*Ethernet*, IP, UDP, RTP) and RTP payload in RTP part of the packet.

The error probability in CID signal transmission is calculated according to these formulas. Figure 7 presents the error probability in CID signal transmission when each CID signal bit is coded as the event code and when the whole CID signal is transmitted in one packet, using the special code for CID signal. The significant error probability decrease when CID signal is transmitted using the special code for the whole CID signal is the result of significant decrease in required channel capacity for transmission.

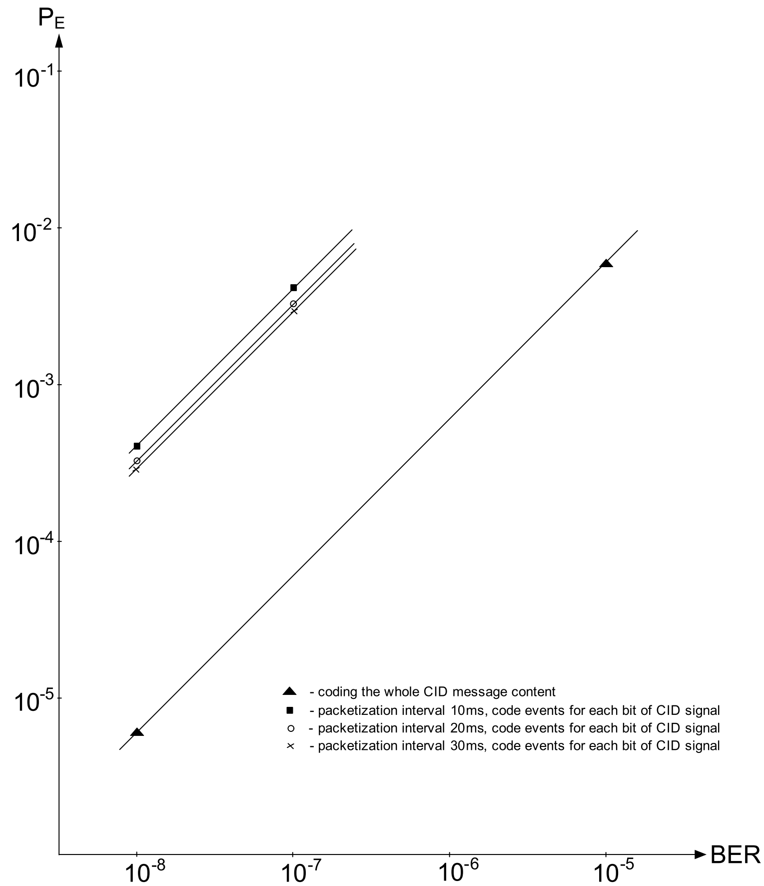

Fig. 7. The comparison of error probability when CID signals are transmitted using different coding methods

6 Conclusion

This paper presents different possibilities for CID signal transmission over Internet. Coding of each bit in CID signal as the event code doesn't decrease significantly the channel capacity and the error probability comparing to sending CID signal samples. Significant decrease in channel capacity and error probability is obtained by sending the complete CID signal content in one packet, using special event code for the whole CID signal. But, the gateway for this method of sending is more complicate than in two previous cases.

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