VOICE OVER IP TO ISDN GATEWAY VIA LTE ACCESS

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ABSTRACT

The relevance of article is due to the developing and increasing of telecommunication network kinds and access technologies in railway infrastructure including Long-Term Evolution (LTE). An important practice task is to realize interaction between both developing and existing networks such as Voice over Internet Protocol (VoIP), Integrated Services Digital Network (ISDN), etc. The object of the study is access gateway to the railway operation network via LTE access for fixed and mobile subscribers. This solution might provide access to existing ISDN for a distance more than 1 km without signal regeneration and installation of additional expensive equipment, redundant in the case of access by a small number of subscribers. Thus, this study is aimed at hardware and software based solutions of the VoIP to ISDN gateway via LTE construction system. The paper proposes test of the gateway operation, as well as experiments using IP and ISDN protocol analyzers to describe the interaction of network signaling, estimated the sizes of data transmission packets for several types of codecs. Authors note that further research of the device should be carried out using the QoS criteria depending on the operating conditions of the LTE network (signal level, object speed, presence of obstacles and signal reflections).

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1 Introduction

Advanced technologies and developing of a common information space on railway transport allow making operational communication in a universal tool. VoIP technology plays an important role in telecommunication infrastructure of NGN (New Generation Networks) because it solves the problems of voice, video and data transmission. Quality of services (QoS) provided by VoIP networks becomes better every year thanks to the new types of signaling and speech coding algorithms. Currently, VoIP has a wide variety of services, integrated monitoring and management tools [1].

Because of the implementation of packet switching technology, many companies replace existing analogue equipment and digital PBX for IP one. Railway divisions also implement it as a tool of transportation management. Number of VoIP subscribers on railway infrastructure increase rapidly due to call quality and wide range of services. However, the most of services for transportation processes are still based on ISDN and analogue channels because of existing large-scale infrastructure across the railway, interaction with remote control equipment on physical layer and high structure reliability. Taking into account a transportation operation usually occurs for dispatching division or railway station selectively. To make it work, existing ISDN and analogue networks should be able to interact with the network being implemented.

Gateways are usually used to communicate with PSTN (Public Switched Telephone Network) and VoIP-networks using different signaling protocols and protocols for transmission of media flow, including VoIP networks, which are based on either open source (YATE) or proprietary (3CX) protocols [2,3].

VoIP to analogue PBX gateway solution is possible to realize via addition of ITG (Internet Telephony Gateway). Current researches discussed the analysis of voice quality in the communication process between telephone sets via ITG in a VoIP network based on the shape of the input and output, the amplitude response to frequency changes, information security, quality of service and delay [4,5]. Each terminal establishes connection to the classical time division multiplexing (TDM) via packet switching network. Terminals with packet switching are apart in the IP-network via a local area network. An important feature of VoIP is separation of connection and disconnection processions from the voice-packet transmission process. As a traditional TDM technology, VoIP communication is divided into three phases: establishment of connection, transferring of voice and disconnection. Signaling system is used to establish connection and disconnection. Furthermore, signaling system allows subscribers to use audio-, video- conferences and additional services. Signaling data provides administration system for traffic accounting, definition of services quality, calls billing and other purposes [6]. Currently, interaction already existing networks and developing ones is an important task for telecommunication infrastructure of railway transport.

One of technologies being adopted for railway systems is LTE, which implementing on railway areas in transportation processes. LTE based high-speed railway telecommunication network is more effective and reliable than GSM-R or analogue one [7]. The fast development of high-speed railway, as a high-mobility intelligent transportation system, and the growing demand of broadband services for users, introduce new challenges to wireless communication systems [8].

According to [9,10] future railways may reduce their operational costs and augment the traffic capacity by implementation, for example, Industrial Internet-of-Things (IoT) is based on developed wireless network, such as LTE. These measures are expected to improve the performance of key management systems, such as those related to traffic scheduling and transportation system capacity planning, along with railway transportation safety and energy efficiency, interoperability, and support to multimodal transportation. These systems can be integrated with the railway traffic control system. For example, European Rail Traffic Management System (ERTMS) is based on the technical specification for interoperability for "traffic-control and signal-safety" subsystems were designed by the European Union Railways Agency (ERA). IoT integration for train inspection in railway infrastructure, track monitoring and voice communication via LTE are described in [11].

It also necessary to study other possibilities of wireless telecommunications not only for railway cases, but also for public ones in open networks. One of them is an implementation LTE network for passenger comfort. It also possible to use an open system in the LTE for the transmission of signals for railway traffic control and passengers while maintaining an appropriate level of safety [12].

The International Union of Railways (UIC) is developing the Future Rail Mobile Communications System (FRMCS). The paper [13] describes the Adaptive Communications System (ACS), which is currently being developed jointly by the industry and a number of railway operators to cover all types of railway wireless networks, radio access technologies and transport infrastructure features. ACS could pave the way for innovation in the railway sector in the context of geographical and technological differences between railways in different countries and regions. LTE can be considered as a public network or a type of radio communication for train control. Article [14] describes the possibility of integrating these networks, analyzes the compatibility of their functions, and compares the requirements for developing public safety technology in LTE-R.

To organize interaction between PSTN segments and GSM wireless networks, research was carried of implementing a VoIP system using an integrated IP PBX server with telecommunication service providers. GSM VoIP gateway connection is described at article [15]. IP PBX system connected to a GSM network using a GSM VoIP gateway is designed and implemented. Another research [16] is aimed to evaluate the performance of GSM VoIP Gateway in the IP PBX system using quality of service (QoS). The SIPdroid softphone was selected as a subscriber device. The connection between the VoIP client and the GSM VoIP Gateway is classified as a good quality service, since it has an average jitter value of ≤ 5.7 ms, packet loss ≤ 0.18% and delay ≤ 9.41 ms.

Wireless data on railway infrastructure has prospects not only at train dispatching and signaling, but also it may provide access to ISDN or analog workplaces on railway stations. Already existing solutions for VoIP/LTE to ISDN gateway are based on core-PBX. The disadvantages of such solutions are excess capacity and high cost in case of installation at stations along the railway. Furthermore interaction between equipment of different vendors is not allowed or requires high costs generally. In case of organizing wireless access to a workplace within a station via LTE, distributed gateways with a small port capacity are required. Nevertheless, currently there is a problem in integrating LTE-R and existing railway telecommunication services from the perspective of functionality. That is why an actual task is to implement LTE as access network to existing ISDN.

At present, IP-based telephony networks in railway infrastructure use SIP as the main type of signaling. The signaling systems also include gateway's protocols such as MGCP (Media Getaway Control Protocol), MEGACO/H.248, SGCP (Simple Gateway Control Protocol), SIGTRAN (Signaling Transport – Protocol stack for provision reliable datagram services and user layer adaptations for Signaling System 7 (SS7) and ISDN communications protocols), STCP (Stream Control Transmission Protocol – Protocol of reliable delivery of packets in the IP-based networks), SCCP (Skinny Client Control Protocol – Corporative terminal protocol), Unistim (Private communication signaling protocol), IAX2 (Inter Asterisk eXchange protocol, v2) and others [17, 18].

The main object of the article is hardware and software solutions of VoIP-to-ISDN gateway in LTE access conditions. An important task is not only theoretical research of gateway, but also practical investigation of equipment and software used in packet-switched networks and mixed networks in order to implement it into railway telecommunication infrastructure. Since the article describes configuration of the required hard- and software of gateway solution; integration of the test system into the experimental environment; test of predefined communication scenarios including protocol analysis; traffic evaluation.

2 VoIP-to-ISDN via LTE laboratory setup

VoIP-to-ISDN via LTE gateway setup is based on IP-PBX software Asterisk. IP-PBX allows making calls using equipment that support standard SIP RFC 3261. Software implementation allows configuring all nodes of virtual PBX. It is possible to change requirements without designing or production a new equipment. Hardware requirements depend on capacity of the PBX, interaction with other PBX's or PSTN and set of additional services.

Asterisk operates as SIP-registrar and SIP proxy in VoIP-networks and also provides services of VoIP. Asterisk allows creating 5th class softswitches which are intended for work with end-users. This equipment is used for both local and long distance telephony services. Softswitches are characterized by additional services for end-users and corporate clients such as IP PBX features, call center services etc. This IP-PBX provides services for conference, group calling, voice-menu, forwarding of calls etc. [19]. It includes DNS-server, NTP-server and DHCP-server. Also, it is able to work as a platform for a gateway using additional hardware such as ISDN-card.

ISDN card HFC-2BDS0 is installed into IP-PBX in order to connect IP-PBX to ISDN one. This card has PCI interface to install at IP-PBX and BRI interface (point S/T) to interact with ISDN. S/T interface is defined by Recommendation I.430 ITU-T, the most important aspects of S/T are:

– use fo r conductors;
– the maxi um allowable distance is 1 km;
– su port configuration “point-to-point” or “point-to-multipoint”;
– data tra sfer rate is 192 kbit/s.

Figure 1 provides appearance of ISDN card which can work as an ISDN card for PC through PCI-interface. It may be used as terminal adapter, network terminal, PBX or ISDN-modem [20].

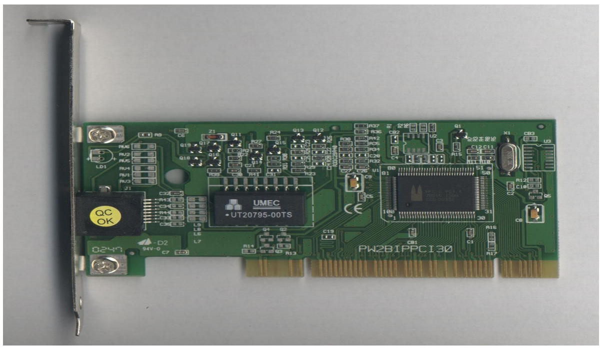


Fig. 1. ISDN card HFC-2BDS0

All necessary software (drivers) for correct work of ISDN card has been installed to IP-PBX. In this circuit, the card operates as the Network Termination of the first level – NT1 in "point-to-multipoint" mode. This mode is used to match electrical parameters of the line, control line status and conversion.

ISDN-side of complex includes ISDN-PBX with capability of 24 digital subscribers and 8 analogue subscribers. Protocol analyzer is connected to ISDN-network in order to decrypt messages of DSS-1. This device is also able to decrypt messages of 1TR6, Q-SIG, VN4, V 5.1, V 5.2, NI-1, and SS#7 [21]. ISDN analyzer could be placed between ISDN-card and ISDN PBX as well as between ISDN PBX and ISDN-phone. Figure 2 provides a schema of VoIP-to-ISDN gateway with equipment of ISDN-side and equipment for measurements in ISDN-side.

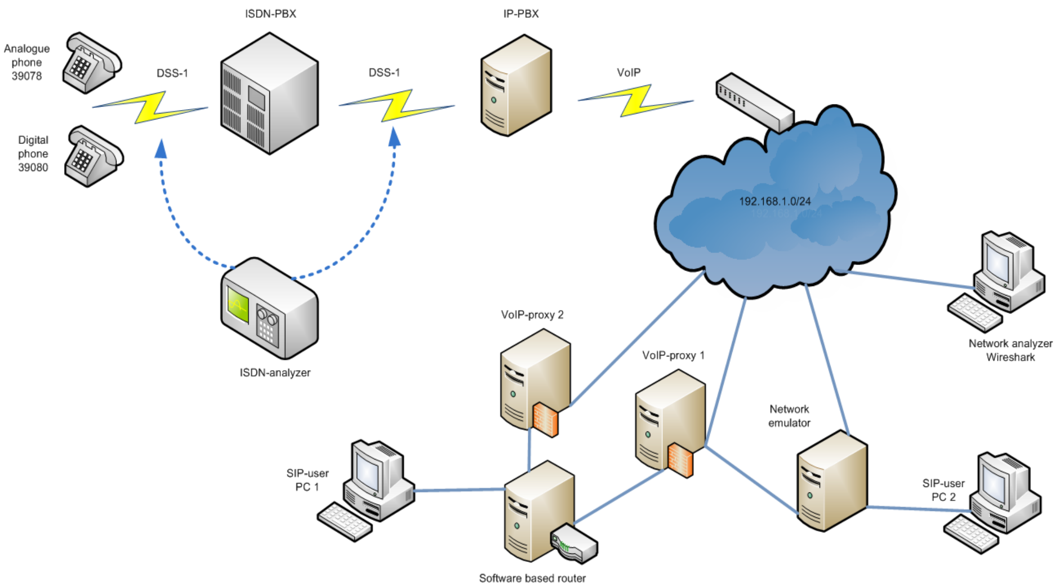


Fig. 2. Schema of VoIP-to-ISDN via LTE gateway

Network analyzer Wireshark has been also installed in the configuration. It allows capturing of IP-packets from Ethernet interface at local area network masked as /24. To achieve it all devices of this network are united by physical layer 8-port hub.

SIP-side of complex consists of two VoIP-proxies, software based router, software based network emulator and two SIP-subscribers. VoIP-proxies are based on software OpenSIP, which allows creating 4th class softswitch on PC. These softswitches are used for transit VoIP traffic between carriers. The main function of 4th class softswitch is the routing of large volumes of long distance VoIP calls. The most important characteristics are protocol support and conversion, transcoding, calls per second rate, average time of one call routing, number of concurrent calls. Software based router is used to create network layer static routes between different networks. Network emulator is used to delay and to make loss of the packets, which are flowed through emulator.

Softphones installed on PC1 and PC2 as sip-users. IP-PBX had been connected to network emulator via 8-port dual speed hub. Network-layer connection between IP-PBX-network and other networks were established by adding to main routing table on network emulator by utility “ip route add”.

IP-PBX should be registered on the proxy-voip2 as a subscriber to establish interaction with VoIP-proxies. For this purpose, is necessary to create new accounts in proxy-voip2 and define this proxy as SIP-provider for IP-PBX.

All accounts are located in the path of IP PBX server: *home/voipprak/Opensips/DB/subscriber. Accounts “1001”, “1002”*, and *“1150”* have been added to this file. Parameters of SIP-provider were determined in IP PBX.

Next step is to correspond external accounts of SIP-provider (proxy2) with internal extensions of IP-PBX (“Asterisk”). Thereby if any SIP-user from proxy2 (as well as from proxy1) calls to one of accounts “1001@voip2”, “1002@voip2” or “1150@voip2” this call will come to IP-PBX and then IP-PBX will forward this call to internal extension. But it is not possible to dial name of SIP-account from ISDN- or analog phone, because SIP- account contains letters. In order to provide calls from IP-PBX to proxy2 and proxy1 file: *etc/asterisk/extensions.conf* of IP-PBX server has been changed. This file contains dial plan configuration and defines processing and routing of incoming and outgoing calls. New scenarios of dial plan were added to make outgoing calls from IP PBX. Each extension has format shown below:

exten=>telephone number, priority, command

In this case new dial plans have been created for outgoing calls for *user1@voip2 and user2@voip1 by using command Dial()*, which can have one of two following formats:

Dial(type/identifier, timeout, options, URL)
Dial(type1/identifier1&type2/identifier2&type3/identifier3...,timeout, options, URL)

The command is trying to call on all specified channels simultaneously by type and identifier. As soon as one of the channels responds to a call for all other call is terminated. The first dial plan is shown below:

exten=>2001,1,Dial(SIP/user1@192.170.56.5)

It means, if any subscriber of IP-PBX (including ISDN-side) calls to „2001”, IP-PBX will dial subscriber user1@192.170.56.5 by channel known as SIP.

The second dial plan is used for outgoing calls to user2@192.170.34.3:

exten=>2002,1,Dial(SIP/user2@192.170.34.3,20,m)
exten=>2002,2,Dial(SIP/psgw@192.168.1.10)

This dial plan provides call to user2@192.170.34.3 for 20 seconds with music in background. If SIP-user user2@voip1 will not answer for this call during 20 seconds, IP-PBX will look at second priority of this dial plan and will call to SIP-user psgw@192.168.1.10.

Thereby, complex of gateways was successfully integrated into already existing environment of VoIP. It allows creating new scenarios for railway communication infrastructure.

3 Signaling analyzing

Analyzing of VoIP to ISDN via LTE connection signaling is based on corresponding messages between ISDN and VoIP network. In order to get DSS-1 data ISDN-analyzer is placed between IP-PBX and ISDN one. Network analyzer Wireshark is run across path of SIP-signaling via proxies to gateway. Connection between ISDN-phone and SIP-user is established to record DSS-1 data from ISDN-analyzer and SIP-messages from Wireshark. At first ISDN-analyzer is placed between ISDN-PBX and ISDN-phone, then it is replaced to ISDN-PBX gateway circuit. Picture 3 provides a schema of laboratory environment and an example screenshots of data from analyzers.

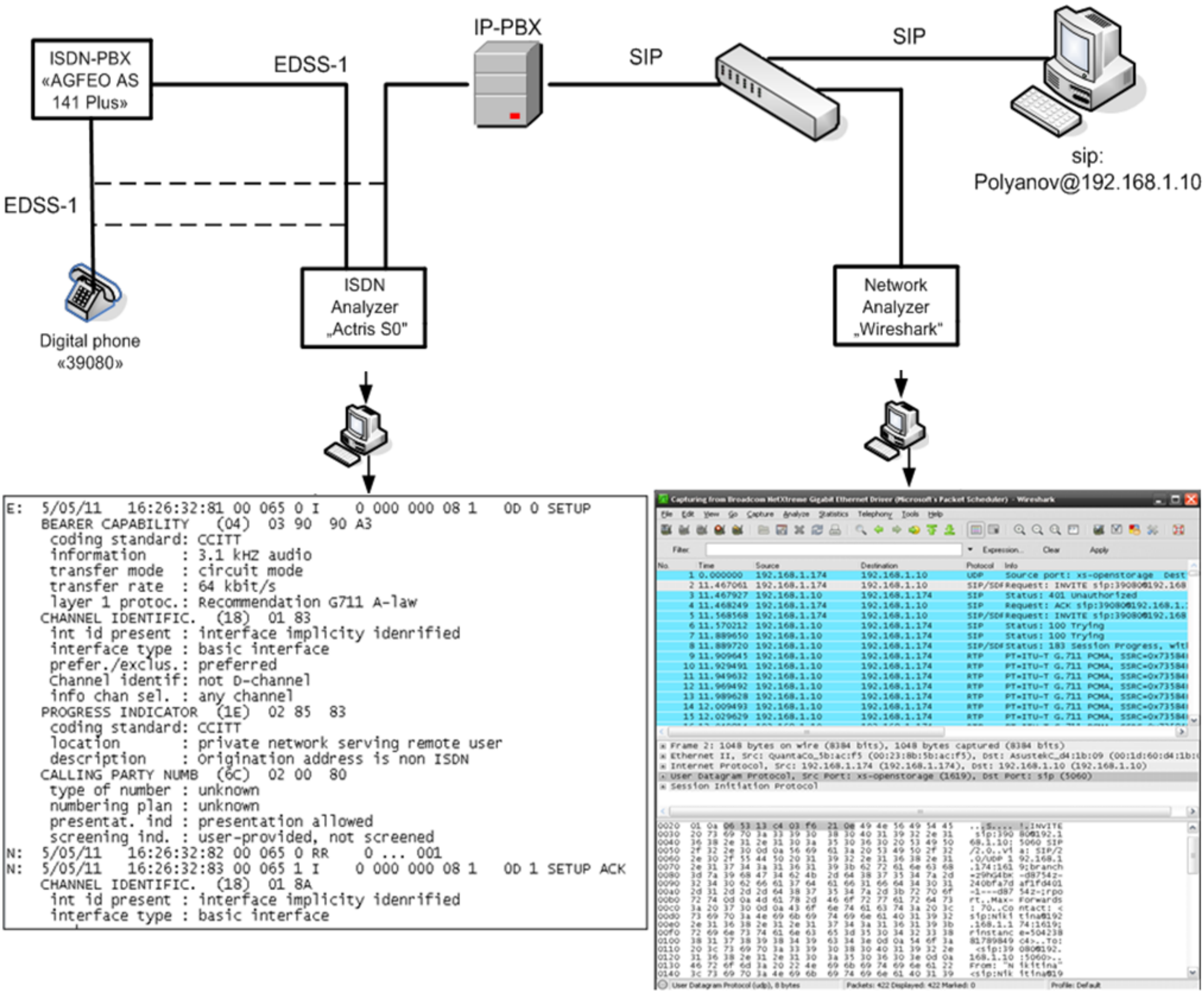


Fig. 3. Laboratory environment of signaling analysis

Figure 4 provides a graph of signaling processes between SIP-user and ISDN-subscriber. Data messages of second and third layers from ISDN analyzer were used to construct DSS-1 side of the graph. SIP-side of the graph is based on “Flow graph” function Wireshark application.

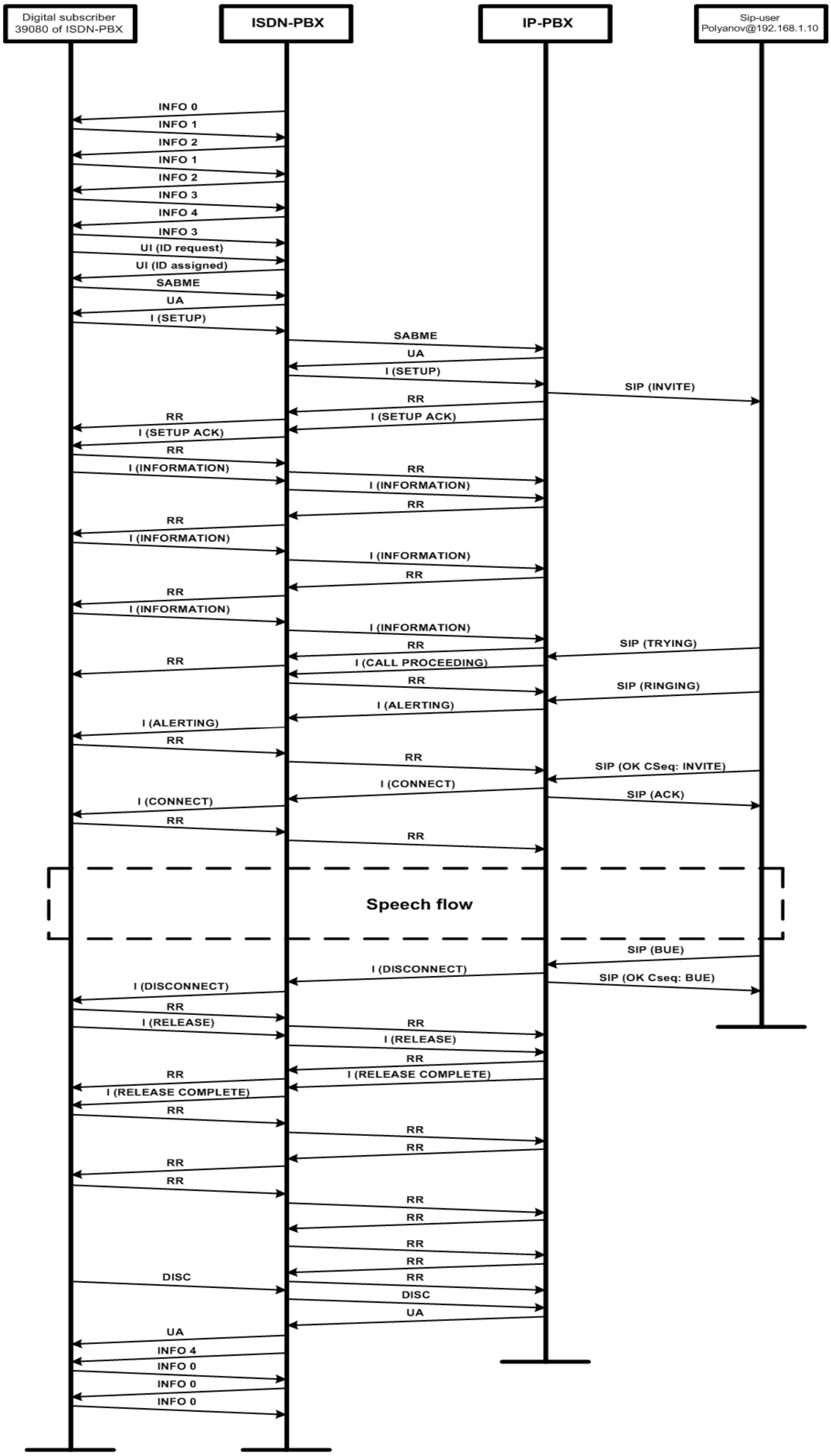


Fig. 4. Graph of signalling processes between SIP-user and ISDN-subscriber

The caller is ISDN-subscriber with extension “39080” and the callee is SIP-user Polyanov@192.168.1.10. Session between ISDN-phone and the PBX was started when subscriber “39080” pick headset up. Session connection to ISDN-network begins with conflict resolution access to the terminal. For this purpose ISDN-phone and PBX exchange INFO-messages (INFO0, INFO1, INFO2, INFO3, and INFO4). Each session ends by exchanging of INFO-messages.

The Terminal Equipment (TE) sends a request “IDrequest” to identify itself and then TE receives an answer with the ID of the terminal - TEI by unnumbered message. Before establishment of LAPD-connection (Link Access Procedure for D-channel), only unnumbered frames may be used for data-transmission. This process involves the transfer of command to Set Asynchronous Balanced Mode Extended (SABME). The recipient must respond by sending Unnumbered Acknowledgment (UA). Command DISC is used for LAPD. Second layer of DSS-1 can carriers messages of third layer after establishing of LAPD.

There two types of messages are defined in LAPD as reliable information (“I”) and unnumbered information (“UI”). Acknowledgments messages of type “I” are Receiver Ready “RR”.

Protocol Q.931 is used at the third level of DSS-1 to define the meaning and content of the signaling messages, in logical sequence of events and in the process of existence and the terminations of connections. Messages of the third layer of DSS-1 are transmitted by “I” - messages of LAPD. Some messages of the third layer of DSS-1 are corresponded with SIP-requests and SIP-answers in Table 1.

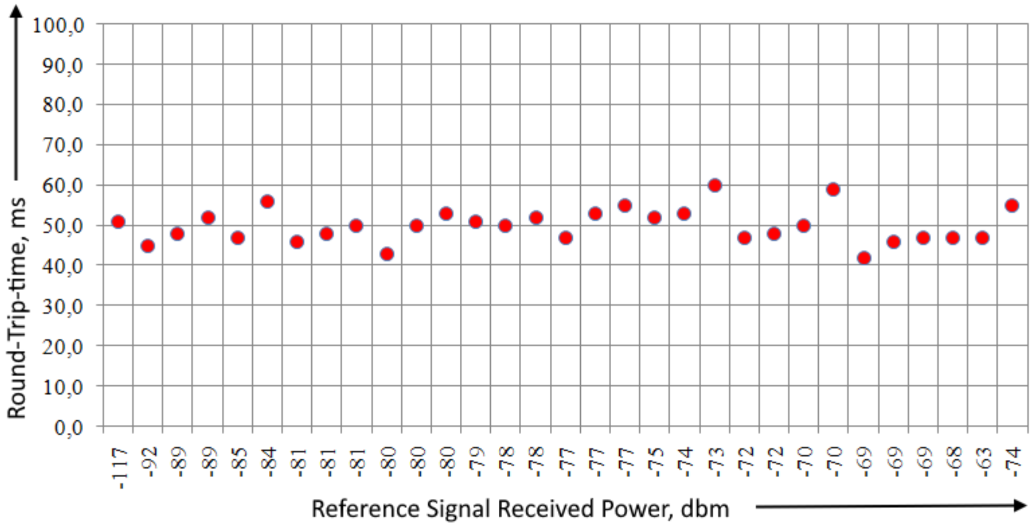
Table 1

Comparison of SIP- and DSS-1 messages purposes

Message of DSS-1	SIP-request (answer)	Purpose
SETUP	INVITE	Request to establish connection
SETUP ACK	OK CSeq: INVITE	Confirmation of request’s delivery
CALL PROCEEDING	TRYING	Extended search proceeding
ALERTING	RINGING	Sending ring-signal to subscriber
CONNECT	OK CSeq: INVITE	Subscriber answering
CONNECT ACK	ACK	Confirmation of answer’s request
DISCONNECT	BUE	Clear’s request
REALISE		Disconnection
REALISE COMPLETE	OK CSeq: BUE	Disconnection, clearing of resources

4 Estimation of minimum RTP-packet length

In order to implement the LTE network to ISDN access, the Round-Trip-Time (RTT) and downlink rate (DLR) parameters were estimated depending on the radio coverage level within four Base Transceiver Stations (BTS) in total at one railway station. Figure 5 shows results of RTT and DLR measurements. It was found that the RTT parameter is stable and varies within 40-60 ms when the radio signal level from the BS ranges from -70 to -120 dBm. There is no explicit correlation between the DLR and the coverage. However, at a critically low signal level, the speed decreases in an avalanche-like manner (for example, to a level of 0.5 Mbit/s at a signal level of -117 dBm).



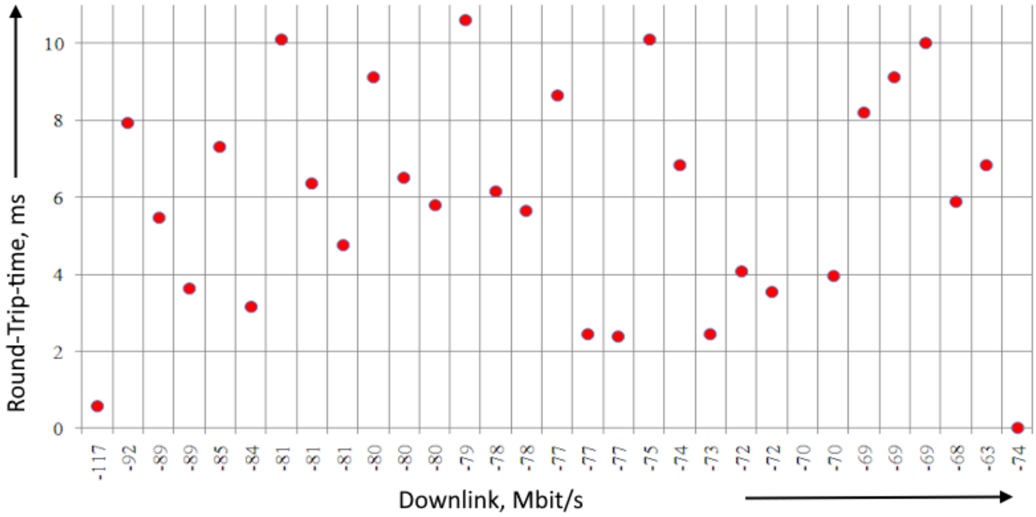


Fig. 5. Results of RTT and DLR measurements

VoIP-to-ISDN gateway based on IP PBX allows optimizing network traffic by adjusting of packet size depending on codecs. The most software based PBX's use RTP protocol for transport voice packets. There codecs G.711 (A-law for Europe and Russia and u-law for USA and Japan) and G.729 are most popular in software solutions of IP-PBX. They provide good call quality in various network conditions. For an example, G.729 requires low bandwidth and has proven to be effective when network resources need to be optimized. But G.711 consumes more bandwidth and provides high voice quality where bandwidth availability has been less of an issue [22, 23]. Solutions of IP-PBX allow a list of codecs for speech encoding and decoding; flexible numbers of subscriber and set of services, because IP-PBX should be able to extend and interact with other systems.

Codec's spec in VoIP-side of the gateway determines number of speech frames, overall packet length of RTP-packet and ratio of useful part of the package to the overall RTP packet length.

Calculating of speech frame numbers, overall RTP-packet length and ratio of useful part of the package to overall RTP packet length, for each codec (G.711, G. 726 (32kbit/s) and GSM) are possible due to choose one of the codecs in IP-PBX. It also possible to compare results of calculation with results from network analyzer.

To calculate the number of speech frames transmitted per second the formula is used:

$$n = \frac{1}{T_{sample}} \quad (1)$$

n – number of speech frames per second, frames/s; T_{sample} – duration of voice sample, s.

Voice sample duration depends on used encryption technology. The next step is to calculate the size of the packetized data according to the formula:

$$H = v \cdot T_{sample} \quad (2)$$

H – the size of packetized data, bytes; v – the encoding rate, bytes/s.

Encoding rate is calculated by the formula:

$$v = \frac{r}{8} \quad (3)$$

r – bit rate, bits/s.

Bit rate is different for each codec and depends on encryption technology. Headers of RTP and TCP/IP stack should be considered in order to determine full size of RTP packet. The total length of headers is 40 bytes (RTP – 12 bytes, UDP – 8 bytes, IP – 20 bytes). The total length of RTP packets encapsulated into stack UDP/IP is defined by sum of length of packetized data, IP-header, UDP header and RTP header:

$$H_{total} = H + H_{ipheader} + H_{udpheader} + H_{rtpheader} \quad (4)$$

Useful part ratio of the package to total size of RTP packet ratio for each codec is defined using the formula:

$$\mu = \frac{H}{H_{total}} \cdot 100\% \quad (5)$$

The parameter values for some of the codecs and calculation results are shown in the Table 2.

Table 2

Calculation results

Type of codec	Bit rate r , kbit/s	Duration of voice sample Tsample , ms	RTP packet size, bytes	μ, %
G.711 (A-law)	64	20	200	80
G.726 (32,0 kbit/s)	32	20	120	67
GSM	13	20	73	45
iLBC (13,33 kbit/s)	13,33	30	90	56
iLBC (15,2 kbit/s)	15,2	20	78	49
G.723.1	5,3	30	60	33
G.723.1	6,3	30	64	37
G.729 A	8	10	50	20
Silk	40	20	140	71

Thus, when G.711 codec is used for encoding speech on the VoIP area of LTE access to ISDN, voice is transmitted over 50 frames per second and packet size is 160 bytes. Each frame is encapsulated in the protocols RTP, UDP and IP, resulting in the packet size increases to 200 bytes, and useful part of the package is 80% of the total. It is possible to use, for example G.729 A with minimal RTP packet size, but ratio of useful parts for this codec is 20%.

Still, in the conditions of using LTE as an access network, it is important to take into account the operating conditions of the radio interface. Public wireless networks demonstrate excellent performance for VoIP in terms of throughput, packet loss, latency and jitter [24,25]. However, a closed LTE-R network requires QoS analyzing depending on radio coverage, electromagnetic compatibility, subscriber speed etc.

Conclusion

This article describes a method for creating a VoIP ISDN gateway over LTE using software and hardware solutions and the main aspects of implementing laboratory setup. An analysis of network boundary signaling for some communication scenarios is carried out. The laboratory facility can be integrated into existing railway communication networks as a diagnostic tool of operation analysis of LTE access in different conditions including interface disturbances.

The main advantage of the proposed gateway is based on existing hardware and freely licensed software. Analysis of the gateway operation in the LTE network showed that the quality of the wireless network is sufficient for its successful operation. Current work is continued as improving the gateway for performance experiments according QoS management and to improving system efficiency, bandwidth and LTE criteria.

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