Evaluation of Main Parameters of Data Transmission for VoIP for Integration of Power Line Carrier Equipment Supporting Standard IEEE 802 in the Packet Networks for Energy Grids

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***Abstract*—**Article is dedicated to consideration of application of power line carrier equipment (PLC) supporting packet traffic transmission via high voltage (HV) lines as part of integrated packet network for energy grid for voice and data communication as part of converged packet channels. Article contents evaluation of limitation of application PLC supporting packet traffic for voice transmission, results of theoretical and practical evaluation of time delay for voice packets, link capacity and consideration about most suitable for energy grids method of queue processing.

***Keywords*—** Energy grid, packet network, time delay, VoIP

I. Introduction

NOWADAYS the most popular type of equipment using for creation the backbone channels between energy substations and dispatching centre are optical multiplexers. Wide application of overhead optical ground wire (OPGW) for high voltage lines allows creating wideband channels with bit rates from 155 Mbps and higher. Usually OPGW is using on the high voltage lines 220 and 500 kV. Such lines are the backbone transmission lines. Fiber optic also is using for 110 kV lines, but some times application of OPGW has low cost efficiency and necessity especially this actual for low population areas like mines or distant areas and for this some other type of equipment should be more suitable. This is power line carrier - PLC.Ten years ago this type of equipment was the most applicable for energy grid because it was not necessary to build any line construction to create links. The media for power used for voice and low speed data transmission. But nowadays with appearance of new types of telecontrol equipment, with requirements of files transmission between substations to the dispatching centre load capacitance of classic frequency division multiplexing power line carrier equipment is not enough. But from other hand more and more conceptions of integrated telecommunication network for energy sector contains packet network as integrated for all types of traffic transmission.

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This common tendency makes developers build-in functionalities supporting IEEE 802 standards into construction of modern PLC.

This article contains the evaluation of one of the sides application of such types of PLC for building telecommunications networks via HV lines between substations located on distant areas like mines and deposits or small towns where line have priory 110 kV class and relatively short distance.

II. The Services Using in Energy Grids and Features of Packet Data Transmission via PLC Channels

The application of wideband channels created with optical multiplexers does not invoke so many questions about packet transmission as PLC channels.

Typical types of traffic for energy networks are:

- dispatching voice channels – highest priority;
- data from SCADA systems – high priority;
- voice channels for technological purposes – middle priority;
- data from power commercial accounting systems – low priority;
- LAN, e-mail – low priority.

High bit rate and powerful routers on node substations allow using codecs like G.711 for high quality voice transmission without big time delay. Generally the same could be referring to other types of services.

The situation in the relation to backbone channels generally is clear, but integration of backbone network and the distant network using PLC channels for packet data has some difficult moments. The first one, PLC links has extremely low bit rate which usually not increase 80 kbps in 12 kHz bandwidth and could be less depends on the line conditions (noise or attenuation increase). Table I shows the maximal bit rate for

TABLE I
Maximal data Rate for PowerLink PLC

Signal/noise ratio 30 dBm	Bandwidth, kHz				
	8	12	16	24	32
	Data rate, kbps				
	48	72	96	144	192

PowerLink PLC equipment (produced by SIEMENS AG) depends on using bandwidth [1]. Because the highest limit of applicable frequency range of PLC does not increase 1000 kHz it is very difficult to create many wideband channels without reciprocal interference.

And this feature constrains the main limitation of PLC channels application – quality of voice channels depends on time delay. In turn time delay is the main problem when route from neighbor substations has few serial segments before coming to node router or IP PABX. This time could not increase defined value.

Second one significant feature is the nature of topology of power grid – it is tree with many serial segments. And nearest to the node router of the backbone network PLC link must have the highest bit rate because it is last transit segment for remote nodes and it must be able to process date from all remote substations.

Of course in narrow channels packets queues will be form. And it is necessary to define optimal algorithm of its processing. Because we have few services with different priorities it is necessary to use the most suitable algorithm for energy PLC packet network. As showed further it could be algorithm of priority queues (PQ).

III. Evaluation of Voice Packet Transmission Time Delay

Because we have small data rate we need use voice compression codecs, more suitable are G.723.1 and G.729A. But these codecs require time for voice processing 97 and 35 milliseconds correspondingly. Also PLC equipment needs some time to prepare data for transmission via HV line. Generally time delay could be evaluated with equation (1) [2]:

$$T_A = T_{codec} + T_{frame} + T_{pred} + T_{PLC} + T_{node} + T_{line} \quad (1)$$

Where:

T_{codec} – delay in codec;

T_{frame} – delay through frame generation;

T_{pred} – prediction delay;

T_{node} – delay through of data processing in network equipment (router or IP PABX);

T_{DPLC} – delay through of data processing in PLC;

T_{line} – delay in the line.

Theoretical results of this calculation showed in Table II.

Function of quantity of nodes in network from time of delay could be evaluated with equation (2):

$$N = f(t_{\max} / T_A) \quad (2)$$

On practice results were checked with Ixea Chariot traffic

simulation application. Fig. 1 shows the scheme of test. As showed on Fig. 1 line with several serial segments was used for test.

TABLE II
RESULTS OF CALCULATION OF VOICE PACKET TRANSMISSION TIME DELAY

Parameter	Value	
	G.729A	**G.723.1**
Maximal admissible delay $\mathbf{t_{max}}$, ms	≈ 600	
Packet length, ms	20	30
Number of packets in frame	2	1
Prediction delay, ms	5	8
Codec delay, ms	35	97
Delay in node (network equipment), ms	30	
Transmission delay, ms	15	
Delay in DPLC through of data processing, ms	approx. 50 ms	
Maximal delay for one segment of network (end – to end)	255	310
1 transit node	370	450
2 transit node	515	600
3 transit node	660	745

From console terminal started voice and data pocket transmission between Endpoint 1 and remote Endpoint 2, 3, 4 and 5. Program automatically showed parameters of time delay and pocket lose. Tests were done for both types of codecs G.723.1 and G.729.

From the calculation it may be concluded that maximal numbers of nodes can not exceed three for G.729 codec and two for G.723.1 codec.

Table III shows the results of practice experiment.

TABLE III
RESULTS OF PRACTICE EVALUATION OF VOICE PACKET TRANSMISSION TIME DELAY

Data rate: 64 kbps	Average delaying of voice packets transmission, ms. Maximal admissible value: 600-650 ms	
Connection/Codec	**G.729A**	**G.723.1 (ACELP)**
EndPoint 1 – 2 (point to point)	251	303
EndPoint 1 – 3 (1 transit node)	368	445
EndPoint 1 – 4 (2 transit nodes)	512	597
EndPoint 1 – 5 (3 transit nodes)	655	739

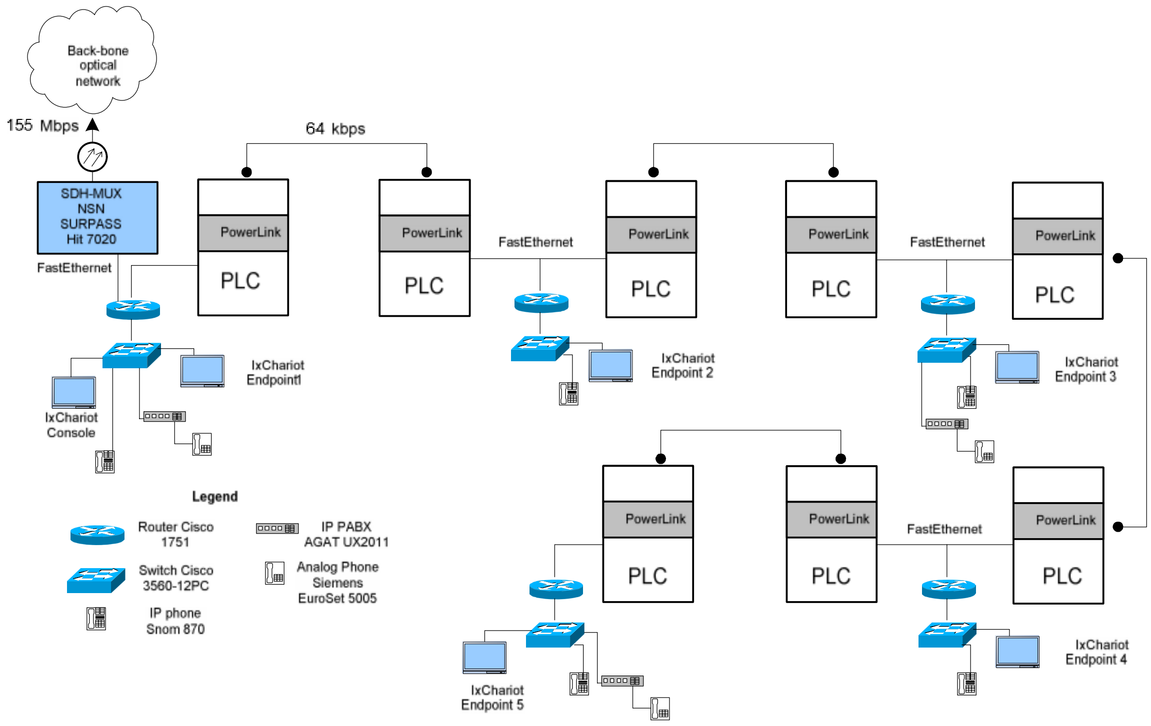

Fig. 1 Scheme of the laboratory tests for evaluation of voice packets transmission time delay

The results of theoretical calculation and practical measuring of time delay have a good correlation.

The next stage of checking was subjective evaluation of voice quality and tangible time delay with application of IP phones and IP PABX.

IP phones showed better performance in comparison of application IP PABX with analog 2-wire phones. In both case when number of transit nodes was 2 and 3 both speakers feel the time delay but it was not so critical that it could be evaluated as reason of impossibility of talking.

In case of application of PABX with analog phones echo was present and it was serious obstacle for conversation.

IV. Evaluation of Link Capacity for Voice Channels

As PLC channels have low throughput the following parameter which is necessary for considering – possible capacity of the channel.

At designing of PLC channels it is necessary to know demanded throughput for transmission of all types of services. For applications tolerant to time delay this question is not so actual, as it could be appointed a low priority. The most interesting and important question is providing quality for voice connections.

For our purposes we have two types of applicable codecs: G.729 and G.723.1. And the main question is a calculation of needed bandwidth and the number of simultaneously working voice channels.

And here is the main problem related to application VoIP. The most part of traffic in VoIP packet takes IP/UDP/RTP headers.

There are to ways to reduce the requirement to throughput of PLC channel. The first – to reduce overhead traffic, the second – to use function of voice automatic detection.

For first point the solution seems to be the application of compression of headers and using of cRTP protocol. Compressed Real – Time Transport Protocol, or cRTP, is used on a link-by-link basis to compress the IP/UDP/RTP from 40 bytes to 2-4 bytes most of the time. In packet voice environment when framing speech samples every 20 milliseconds, this scenario generates a payload of 20 bytes. The total packet size comprises an IP header (20 bytes), a UDP header (8 bytes), and an RTP header (12 bytes) combined with a payload of 20 bytes. It is evident that the size of the header is twice the size of the payload. When generating packets every 20 milliseconds on a slow link, the header consumes a large portion of the bandwidth. To avoid the unnecessary consumption of available bandwidth, CRTP is used. This compression scheme reduces the IP/UDP/RTP header to 2 bytes most of the time when no UDP check-sums are being sent, or 4 bytes when UDP check sums are used [3].

Table IV contains the data about needed bit rate for 1 voice channel with different scenario [3]:

TABLE IV
BITRATE FOR VOICE CHANNEL WITH VAD AND cRTP

Codec	Payload size, bytes	Full rate	With cRTP	Full rate with VAD	With VAD and cRTP
G.729 (8 kbps)	20	26.4	11.2	17.2	7.3
G.723.1 (6,3 kbps)	24	18.4	8.4	12.0	5.5
G.723.1 (5.3 kbps)	30	17.5	7.4	11.4	4.8

As it showed in Table IV minimal required bit rate could be obtain with application of cRTP and VAD together.

For the evaluation of one segment link capacity were implemented test with application of Ixea Chariot. As software can not implement the RTP headers compression because headers are using for jitter and time delay calculation, test was done only with VAD function activation (field voice activity was set in 70 % value). During the test number of voice connections sequentially increased before value of packet loss was less than 10 %. The types of used codecs are G.729 and G.723.1 ACELP, bit rate of PLC link data flow was 64, 80 and 128 kbps. Results of this test are showed on graphs in Fig. 2 and Fig. 3.

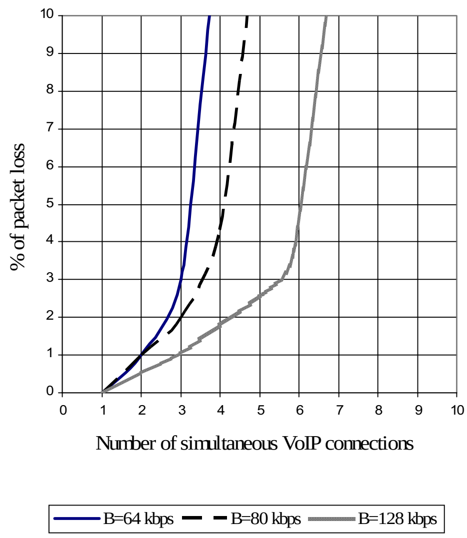

Fig. 2 Relation between number of VoIP connection and packet loss for codec G.729 for different channel bit rates

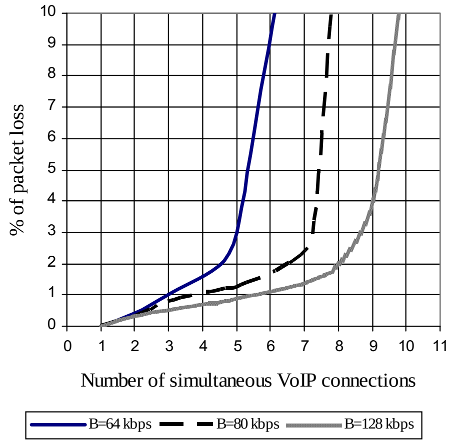

Fig. 3 Relation between number of VoIP connection and packet loss for codec G.723.1 ACELP for different channel bit rates

Results of the test show that from point of view of link capacity it is better to use codec G.723.1, but from other hand this codec has the biggest value of time delay.

V. QUEUE PROCESSING

Application of network technologies for creating of departmental communication networks for power grid utilities via high voltage power line communication brings an attention to the question about methods of queues processing in the nodes of the network. Because each type of information has its own requirements for time delay and processing and bit rate for PLC links is very low, it is necessary to define the most suitable method of packet queues processing in network nodes.

As the network for technological processes there are not many types of services. The main types are:

- voice for technology;
- data from SCADA;
- data from commercial accounting systems;
- LAN.

The voice communication in the given list has the highest priority as quality of voice channels depends on time of a delay of the voice packets transmission between source and receiver. The control system information as it is the most significant for the technological purposes further follows. Data commercial account systems and e-mail messages have the least priority as these services are tolerant to transmission delay.

Most often used methods of queues processing were considered. The first one is First Input First Output – FIFO/Last Input First Output - LIFO, second one is Priority Queuing - PQ, third is Custom Queuing - CQ, fourth is Weighted Fair Queuing - WFQ, and combined method of processing with small delay - Low Latency Queuing - LLQ.

TABLE V
COMPARATIVE ANALYSIS OF ALGORITHMS OF QUEUES PROCESSING

Parameter	FIFO/LIFO	PQ	CQ	WFQ	LLQ
Number of queues	1	4	16	256	1PQ+256WFQ
Type of distribution	absent	priority	cyclic polling	weighted	PQ+WFQ
Guaranteed distribution	no	yes (for high priority)	Yes	Yes	Yes
Guaranteed time delay	no	yes (for high priority)	No	No	Yes
Withstand network overloads	no	yes (for high priority)	Yes	No	Yes

In Table V the comparative analysis of algorithms of queues processing is presented. Proceeding from branch requirements, queues processing cannot be made by algorithms FIFO/LIFO as they do not provide the guaranteed delay and ability to withstand an overload. The Custom Queuing method supports up to 16 queues. The given method provides stable quality of service even in overloads. But CQ is not suitable for applications of real time as has a high variation of delay.

Weighted Fair Queuing algorithm allows distributing throughput of the channel between several queues, being based on weight factors V1, V2..., Vn, ∑ Vi, =1 and an arrival time of packets. WFQ supports to 256 queues. It guarantees the minimum time of the response for applications critical to delay. However at operating ratio of the channel 0.5 that is especially essential to the considered channels, the given method reduces the general quality of service of all streams.

Two methods - PQ and LLQ provide priority processing of packets. The PQ method supports four classes of services: highest, high, normal and low. Quality of service is regulated with buffer length. Such method provides guaranteed QoS of highest priority traffic even with in overloaded channel. But with high coefficient of channel utilization low priority traffic is almost blocked. Method LLQ consists of two algorithms: WFQ which provides distribution of the allocated throughput of the channel between various applications and PQ which provides immediate processing of the traffic critical to delay.

Thus the most suitable methods of queues processing could be PQ and LLQ. But as method LLQ has only one priority queue, and other information is processed by cyclic polling, it concedes to method PQ, where we can appoint a rigid binding on priorities for various applications.

Processing information could be divided into four classes and as method PQ supports 4 priority queues, it is the best for the considered scenario. In Fig. 4 division of types of the transferred information according to priorities is presented.

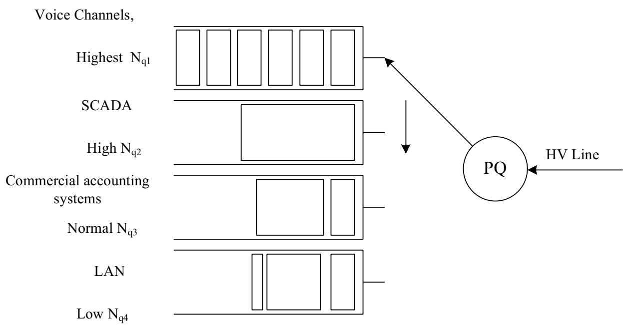

Fig. 4 Example of queue processing for services in power grids

Thereby implemented tests show the limitation of application PLC supporting packet traffic transmission about value of voice packet transmission time delay, but in spite of it such solution could be applicable for creating integrated packet network for energy grid.

The main benefits of application power line carrier equipment supporting IEEE 802 standard are:

- not necessary to use boards with VF interfaces;
- universal interface for all services;
- reducing the cost of PLC;
- wide range of VoIP and network equipment manufacturers;
- simple interconnection with network or backbone equipment;
- application of common principles of LAN architecture for network building and routing;
- cost efficiency of networks;
- possibility of transmission protocols for SCADA – IEC 61850 and IEC 60870-5-104 via PLC.

Also some difficulties and limitations are present, but generally it is covered by DPLC as class of equipment:

- commonly needs wide bandwidth;
- limitation about DPLC technical parameters, as service

availability depends on SNR;

- big time of delay inserts limitations in the number of nodes for voice transmission;

The possibility of extension of the number of nodes could be obtained from application of mixed scheme – voice channel using analog mode and data from services not critical to delay goes in packet traffic.

REFERENCES

[1] PowerLink 50/100 Equipment Manual. Munchen: SIEMENS AG, 2010, chapter 2, p.26.

[2] Merkulov A. “VoIP over DPLC” article from collected papers of AIPET science conference 2009. Almaty: AIPET 2009 – pp 39-41.

[3] Lievin V. “The system of data interaction on the rail way with IP telephony application”: Candidate’s dissertation, Moscow, 2003.

[4] Application note “VoIP bandwidth consumption”: Ascom Business Systems, 2001, p.3.

[5] ITU-T Recommendation G.114. One-way transmission time – 1996.

[6] ITU-T Recommendation G.729. Speech codec for multimedia telecommunications transmitting at 8 / 13 kbps. - 1996.

[7] ITU-T Recommendation G.723.1. Dual Rate speech coder for multimedia communication transmitting at 5.3 and 6.3 kbps. - 1996.

[8] Olifer V., Olifer N. The New Technologies and Equipment for IP Networks. St. Petersburg.: BVH- Petersburg, 2001. – 512 p.