Assessment of the VoIP Transmission Quality over Digital Power Line Carrier Channels

Anton Merkulov, Viatcheslav Shuvalov
Siberian State University of Telecommunications and Informatic Sciences –
Faculty of Automatic Electrocommunication
Kyrova Str. 86, 630102 Novosibirsk, Russia
E-mail: anton-merkulov@list.ru, shwp@sibsutis.ru

***Abstract*—This article is devoted to the research of VoIP transmission quality over Digital Power Line Carrier channels. Assessment of quality transmission is performed using E-model. Paper considers the possibility of joint using of Digital Power Line carrier equipment with different architecture in one network. As a result of the research, the rule for constructing of multi-segment Digital Power Line Carrier channels was formulated. This rule allows minimizing the transmission delay and saving frequency resources of high voltage Power Line Carrier range.**

***Keywords*: E-model, high voltage Power Line Carrier communication, IP networks**

I. Introduction

The use of phase conductors of high voltage lines as a transmission medium of information signals between electrical substations began since 30th years of last century. This type of communication is widely known as Power Line Carrier communication (PLC). A little more than ten years ago, with the development of microprocessor technique, in the PLC equipment signal processors started to be used. It allowed to expand functions of devices and to create Digital Power Line Carrier (dPLC) systems. In modern dPLC equipment the methods of modulation such as QAM/TCM, MCM, OFDM, COFDM are used. DPLC channels are widely used on HVL of 35 kV, 110 kV and 220 kV. In spite of the fact that data rate in dPLC does not exceed few tens kilobit per second, low cost, fast deployment and simplicity in maintenance provide it popularity in the modern power utilities.

Talking about dPLC application the networks built by SIEMENS AG with the use of Frame Relay Access Devices (FRAD) as node equipment is the prime example of successful technical solution. DPLC Frame Relay networks were organized in the period of 2002-2009 in the countries of Latin America, Asia, Africa and CIS. In most projects, the author was personally involved. Unfortunately, with transition to IP platform the life cycle of the dPLC Frame Relay networks comes to the end. Outdated Frame Relay, removal of Frame Relay multiplexer production, and IP traffic transmission requirements updates led to the need to shift from Frame Relay technology to IP technology. For creation of convergent IP-PLC channels assessment of voice transmission quality is very important. Let's observe questions related to the assessment of VoIP transmission quality in dPLC networks taking into account characteristics of applied dPLC equipment and quantity of segments in multi-segment communication channels (MSCC).

II. Assessment of VoIP Transmission Quality over dPLC Channels with Application of the E-model

For transmission quality assessment, we use E-model. This method is described in ITU-T recommendation G.107. It is possible to find examples of using E-model in different sources. Examples of calculations of transmission rating factor R and value of mean opinion score – *MOS* applicable for dPLC channels are shown in [1].

The main difference of the materials represented in this paper is the accounting of joint use of the dPLC equipment with different architecture in one network. Calculation was made according to the last edition of the document ITU-T: G.107 (12/2011) "The E-model: computational model for use in transmission planning"

The main objective of calculations is search of value of factor *R*. According to [2] *R* is calculated by formula (1):

$$R = R_0 - I_S - I_d - I_{E-eff} + A \tag{1}$$

where

R_0 – factor represents basic signal-to-noise ratio;

I_S – factor represents combination of all impairments which occur more or less simultaneously with the voice signal;

I_d – factor represents impairments caused by delay;

I_{E-eff} – factor represents impairments caused by application of low bit-rate codecs;

A – advantage factor for compensation of voice quality degradation for specified system in comparison with conventional systems.

When we calculate R_0, using default values defined in [2], but at the same time taking into account an increased noise level on the substation, R_0 is equal to 91,2.

In dPLC networks low bit-rate codecs like ACELP and MP-MLQ are used and factor I_S characterizing quantization errors will be equal to 0.

The factor I_d in formula (1) depends on characteristics of dPLC equipment. For dPLC terminals where QAM/TCM modulation is used the delay of information processing T_{PLC} almost does not depend on channel bandwidth. But for dPLC terminals with OFDM modulation the bandwidth is factor of high impact. It occurs due to the internal features of system architecture of OFDM devices. Typical values of T_{PLC} for modern QAM/TCM and OFDM systems taken from [3][4][5][6] are shown in Table 1. We should note that in practice dPLC links with the bandwidth of 4, 8 and 12 kHz are the most popular.

TABLE I
TYPICAL VALUES OF TPLC FOR QAM/TCM AND OFDM DPLC EQUIPMENT

Bandwidth, kHz	T_{PLC} QAM/TCM , ms	T_{PLC} OFDM, ms
4		160
8	60	120
≥12		40

I_d is calculated by formula (2) [2]:

$$I_d = I_{dte} + I_{dle} + I_{dd} \tag{2}$$

where

I_{dte} – factor of impairments due to talker echo;

I_{dle} – factor of impairments due to listener echo;

I_{dd} – factor of impairments due to long absolute delay T_a.

Application of echo cancellers in dPLC networks allows to exclude impairments (appeared because of echo) even for absolute delay more than 400 ms, therefore I_{dle} and I_{dte} are equal to 0.

The last item of I_{dd} is calculated using formulas (3) and (4) [2]:

$$I_{dd} = 25\left\{\left(1+X^6\right)^{\frac{1}{6}} - 3\left(1+\left[\frac{X}{3}\right]^6\right)^{\frac{1}{6}} + 2\right\} \tag{3}$$

$$X = \frac{\log\left(\frac{T_a}{100}\right)}{\log 2} \tag{4}$$

In order to calculate I_{dd} firstly we need find the value of T_a. The diagram of time delay sources in dPLC channel is shown in Fig. 1.

$$T_a = T_{voice} + 2 \cdot T_{Router} + T_{PLC(MD)} + \\ + T_{(link)} + T_{PLC(DMD))} + T_{serial} + T_{de\text{-}JIT} \tag{5}$$

where

T_{voice} – delay of voice processing including algorithmic delay of voice codec and packetization delay;

T_{Router} – internal delay in router;

$T_{PLC(MD)}$ – internal delay in dPLC terminal during signal processing in the transmission path;

$T_{(link)}$ – time of HF signal propagation via HVL;

$T_{PLC(DMD)}$ – internal delay in dPLC terminal during signal processing in the receiving path;

T_{serial} – serialization delay;

$T_{de\text{-}JIT}$ – delay of de-jitter buffer.

Delay of voice processing defined as (6) [7]:

$$T_{voice} = [\, T_{frame} \cdot (N+1) + T_{look-ahead} \,] \tag{6}$$

where T_{frame} – voice frame duration;

N – number of voice frames per packet;

$T_{look\text{-}ahead}$ – prediction delay.

Popular voice codec G.723.1 uses band-limited input speech signal sampled at 8000 Hz, converts it into a 16-bit linear PCM signal, at the same time operating 240 samples. Therefore, each frame corresponds to 30 milliseconds of speech signal, that is why T_{frame} = 30 ms. Also codec has a prediction delay $T_{look\text{-}ahead}$ = 7.5 ms. As result total delay for N = 1 is equal to 67.5 ms.

Basic ITU-T specification G.729 describes the voice codec operating at bit-rate 8 kbit/s. This codec uses the input frames of 10 ms corresponding to 80 samples at a sampling rate of 8000 Hz. G.729 has a prediction delay $T_{look\text{-}ahead}$ = 5 ms., the resulting algorithmic delay of G.729 is 15 ms. By the default, using G.729 codec one IP-datagram includes two speech frames, and total delay is equal to 35 ms.

The delay of packet processing in routers T_{Router} can arise only in case of appearing of queues. In absence of queues internal delay of routers does not exceed several milliseconds. Assume in calculations T_{Router} equal to 3 ms.

Delay of signal processing in dPLC $T_{PLC(MD)}$ and $T_{PLC(DMD)}$ in sum represent T_{PLC} which values was shown in Table 1.

The high frequency signal propagates in HVL approximately with light velocity, therefore value of $T_{(link)}$ can be neglected.

Serialization delay T_{serial} is the time that is necessary for processing of layer 2 frame from the first to the last bit and defines by formula (7) [8]:

$$T_{serial} = \frac{S}{C} \tag{7}$$

where S – size of L2 frame, bits;

C – data rate, bps.

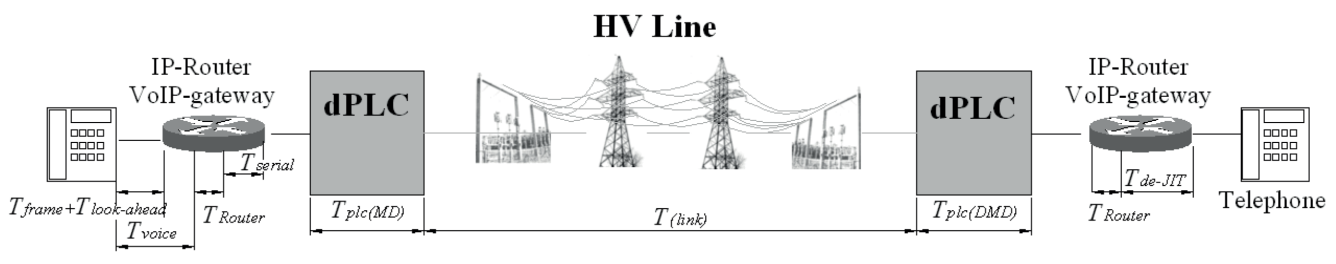

Fig.1. Diagram of time delay sources in dPLC link.

T_{serial} depends on data rate of the communication channel. In T_a calculations we will assume that data rate of the communication channel allows to provide a serialization delay no more than 20 ms while recommended value is 10-15 ms.

In calculations of T_a it is necessary to consider the size of de-jitter buffer $T_{de\text{-}JIT}$. . The volume of the buffer shouldn't be too big in vain not to increase a time delay, but also shouldn't be less than required minimum $T_{de\text{-}JIT(min)}$. We assume $T_{de\text{-}JIT(min)}$ = 30 ms.

According to [9], the value of factor $I_{E\text{-}eff}$ for codec G.729A+VAD is equal to 11, and for codec G.723.1+VAD is equal to 15.

The last term of (1) is an advantage factor *A*. It is easy to show that the using data given in [2], the value of the coefficient *A* can be approximated by the expression (8):

$$A = 0{,}0444 \cdot T_a - 6{,}6667 \qquad (8)$$

Calculation of *MOS* is performed with use of (9) [2]:

$$MOS = \begin{cases} 1, \text{for } R < 0, \\ 1 + 0{,}035 \cdot R + R \cdot (R - 60) \cdot (100 - R) \cdot 7 \cdot 10^{-6}, \text{for } 0 < R < 100 \\ 4{,}5, \text{for } R > 4{,}5 \end{cases} \qquad (9)$$

Values of *R* and *MOS* was derived for G.729A+VAD codec as it has smaller value of $I_{E\text{-}eff}$ and algorithmic delay in comparison with G.723.1. The results of calculations are shown in Table 2. The best transmission quality of voice signal is provided using OFDM equipment with bandwidth of 12 kHz and more. For channels with a bandwidth of 4 and 8 kHz for minimization T_{PLC}, it is expedient to use QAM/TCM equipment.

TABLE III
RESULTS OF *R* AND *MOS* CALCULATION

dPLC equipment	T_a, ms	T_{PLC}, ms	*Id*	*A*	*R*	*MOS*
OFDM (bandwidth 4 kHz)	251	160	9,04	4,48	75,47	3,84
OFDM (bandwidth 8 kHz)	211	120	4,22	2,70	78,51	3,97
OFDM (bandwidth 12 kHz)	131	40	0,01	0,00	80,02	4,02
QAM/TCM (any bandwidth)	151	60	0,18	0,04	79,89	4,02

III. VOICE TRANSMISSION VIA MULTISEGMENT DPLC CHANNELS

For one-segment channels it is possible to provide good quality of speech with dPLC equipment of any configuration and architecture. Difficulties start to arise in case of appearance of transit nodes. In most cases the dPLC network consists of several multi-segment channels. In practice number of hops in MSCC rarely exceeds 4.

Let's calculate the maximum number of segments *M* in MSCC depending on used equipment. Calculation will be done considering that fact that VoIP transmission quality for any segment shall not be worse than fair, i.e. *R* = 60 and *MOS* > 3, with T_a not more than 400 ms. Transmission delay for MSCC with number of segment equals *M* can be found as (10):

$$T_a = T_{voice} + T_{Router} + \sum_{i=1}^{M} \left(T^{(i)}_{Router} + T^{(i)}_{plc(MD)} + T^{(i)}_{link} + T^{(i)}_{plc(DMD)} + T^{(i)}_{serial} \right) + T_{de-JIT} \qquad (10)$$

After *R* and *MOS* calculation was done with the use of (10) we can make the following conclusion:

1) with the given characteristics, in case of using OFDM equipment with the bandwidth of 4 kHz, it is possible to provide only 1 segment channel;

2) using only OFDM equipment with bandwidth of 8 kHz, it is possible to create 2 segment MSCC with characteristics of the most distant segment *M*=2: T_a = 354 ms, *R* =68,9 and *MOS* =3,55;

3) the use of QAM/TCM equipment allows to create 4 segment MSCC with characteristics of the most distant segment *M* =4: T_a = 400 ms, *R* =67,05 and *MOS* =3,46;

4) the best results could be obtained with application of OFDM equipment with bandwidth 12 kHz and more. This solution allows to create MSCC with *M* =5;

5) During planning of dPLC network the following question ought to be considered: what type of dPLC architecture should be in MSCC segments in order to reduce voice transmission delay.

IV. RULE OF CREATION OF MULTISEGMENT DPLC CHANNELS

In conclusion we formulate the rule of choice of dPLC equipment for MSCC segments. This rule allows minimizing the transmission delay and saving the frequency resources: *"For minimizations of transmission delay if QAM/TCM equipment provides sufficient capacity for defined value of SNR (signal to noise ratio), it needs to be used in segments with bandwidth of B=4 and B=8 kHz. In other cases it is necessary to use OFDM equipment. For transmission of voice signal through the maximum number of segments OFDM equipment with bandwidth ≥ 12 kHz has to be used".*

For applied tasks this rule means: in one dPLC network both OFDM and QAM/TCM equipment should be used. For the most distant segments of MSCC QAM/TCM equipment should be used in channels with bandwidth of 4 and 8 kHz. OFDM equipment should be used in segments with bandwidth of 12 kHz and more. Joint use of the equipment of different architecture provides minimization of transmission delay and higher transmission quality of voice consequently, also saving frequency resource of the high voltage PLC range.

References

[1] S.E. Romanov, “Transmission of Voice Signals over dPLC channels.” [Online]. Available: http://romvchvlcomm.blogspot.com/

[2] Recommendation ITU-T G.107 – 2011 The E-model: a computational model for use in transmission planning. 18 p.

[3] Power Line Carrier PowerLink 100 CSPi Manual. Germany: Siemens AG, 2011. 620 p.

[4] Power Line Carrier ETL 600 Manual. Switzerland:ABB, 2011. 440 p. Digital Power Line Carrier STE-D technical description. Italy: SELTA, 2011. 38 p.

[5] Product description of Digital Power Line Carrier Equipment CVK-16. Saints-Petersburg, Russia: NPF MODEM 2012. 83 p.

[6] Recommendation ITU-T G.108 – 1999 Application of E-model: A planning guide. 130 p

[7] Recommendation ITU-T G.108 – 1999 Application of E-model: A planning guide. 130 p

[8] Design Best Practices for Latency Optimization. CISCO, 2006. 8 p.

[9] Recommendation ITU-T G.109 Amendment 1 – 2007 New Appendix I – The E-model-based quality contours for predicting speech transmission quality and user satisfaction from time-varying transmission impairments. 9 p.