On the IP-based Services Distribution over Cable Television Networks

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Introduction

IP-based services are gaining ground in the cable television networks. Their specific transmission must be well studied in order to make network planning most effective.

Nowadays DOCSIS (Data over Cable Service Interface Specifications) is the only standard (protocol) for data transmission over cable television networks.

The paper aims to evaluate some of the main parameters that characterize the transmission of IP-based services over cable television networks. The peculiar characteristics of the voice telephony transmission over Internet Protocol (VoIP) are considered.

DOCSIS protocol in data transmission over cable television networks

According to the DOCSIS protocol the cable modem termination system (CMTS) co-ordinates the operation of all downstream and upstream channels and the cable modems (CM) as well.

In order to transmit information the upstream and the downstream channels of the cable network are divided into several slots of the same size. The CMTS divides the slots in the upstream channel into 2 categories: contention slots (CS) and data slots (DS). The CSs transmit the bandwidth requests from the CMs while the DSs transmit data from the CMs. The number of the DSs depends on the type of information transmitted. An upstream bandwidth allocation map (MAP) is periodically sent by the CMTS to the CMs to provide information about the usage of each slot in the upstream channel. The MAP message structure is described in details in [1].

Fig. 1 illustrates the DOCSIS operation flow when information is transmitted over CATV network [1, 2].

The DOCSIS protocol operation during data transmission is as follows:

- At time t1 the CMTS sends out a message MAP1 that describes slots allocation in the upstream channel during t3 – t8.
- With MAP1 arriving at t2 the CM is supplied with information about CSs allocation in the upstream channel. If CM is going to transmit data, it will implement the collision resolution algorithm (CRA) to determine the moment a transmission request should be sent. Finally, the request is sent out at t4.
- CMTS receives the request at t5, then it schedules all the requests received from the remaining CMs and sends out a message MAP2 at t6.
- CM gets MAP2 at t7 thus being informed about the mini-slots allocation during t8 – t11.
- At t9 CM gets information about the DSs that have been mapped by CMTS, so it starts transmitting data in these DSs. Eventually data from the protocol data unit (PDU) reach CMTS at t10.
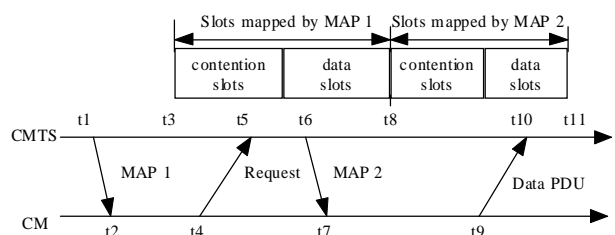

Fig. 1. Data transmission over CATV networks according to DOCSIS standard

Collisions may occur in the operation flow if more than one CM send out requests to the CMTS at one and the same time. Accordingly, the DOCSIS protocol makes use of the truncation binary exponential back-off (TBEB) algorithm [1-3] that is similar to the one used in Ethernet for contention resolution.

The data back-off start (DBS) and data back-off end (DBE) parameters are defined as the number of unsuccessful transmission attempts and the maximum possible number of unsuccessful attempts, respectively. These parameters are periodically reported by CMTS through MAP messages.

Initially the CM attempts to transmit in a slot chosen on a random basis within the interval $[0 \div (2^{DBS} - 1)]$.

When a collision is detected and the back-off window size is smaller than the maximum back-off window size 2^{DBE}, CM sets a new back-off window size 2^{DB} by multiplying the original value by 2. Then CM picks out a new random value within the redefined range and waits for re-transmission for i in number time slots, where $i = \text{random } [0 \div (2^{DB} - 1)]$. The process continues until the request is successfully transmitted or the number of retries exceeds the pre-defined maximum retry value. The number of unsuccessful transmission attempts is up to 15, according to DOCSIS 3.0.

Effectiveness evaluation of the cable television networks providing IP-based services

Modern CATV networks are based on the asymmetric approach, i.e. their downstream and upstream channels differ in parameters.

Two main problems result from the asymmetry, the first being that a large amount of symmetrical services (such as IP telephony) are hard to support by the cable network and the second being that data transmission over CATV networks can affect the normal operation of the transmission control protocol (TCP).

The TCP is based on the three-way handshake approach in order to provide reliable data transfer. With that principle the receiver, after having received the packets, returns an acknowledgement (ACK) packet to the sender and waits for a reply. The succeeding data packets in the sender's queue can be transmitted only if the sender has received the ACK packet. If the sender fails to receive the ACK packet on time, the related data packets need to be re-transmitted, so all the packets in the sender's queue will be delayed. This issue would lead to a disturbance of the TCP ACK messages transmission from CMTS to CM.

The TCP effectiveness in the asymmetrical networks is evaluated with the following coefficient (asymmetric ratio):

$$K = \frac{Q_d L_{ACK}}{Q_u L_{data}} \ , \qquad (1)$$

where Q_d and Q_u – the capacities of the downstream and the upstream channels respectively, L_{ACK} – the ACK message length in bytes and L_{data} – the data packet length in bytes.

The TCP behaves normally when K is less than or equal to 1 [4-5]. When the data transmission rates in the downstream and the upstream directions are different (i.e. $K > 1$), ACK packets arrive at the bottleneck link in the upstream direction at a rate faster than the bottleneck link can support. As a result, the sender clocks out data at a slower rate and slows down the growth of the congestion

window, which in turn lowers the throughput in the downstream direction. From (1) a conclusion can be made that the asymmetry problem can be solved in two ways:

- by varying the packets lengths L_{ACK} and L_{data}, which however directly refers to the type of information transmitted, hence can not be carried out within large limits.

- by using the bonded channels approach for some upstream channels in accordance with DOCSIS 3.0. Thus, additional upstream channels can be adopted until the required value of Q_u is achieved.

Another parameter of great significance for the network performance is the round trip delay (RTD). The following formulae can be used to determine its value while sending a packet in the downstream (RTD_d) and the upstream (RTD_u) direction [4]:

$$RTD_d = 2T + (\frac{L_{data}}{Q_d} + \frac{L_{ACK}}{Q_u}) + B_{CM}T_u , \qquad (2)$$

$$RTD_u = 2T + (\frac{L_{data}}{Q_u} + \frac{L_{ACK}}{Q_d}) + B_{CM}T_u , \qquad (3)$$

where B_{CM} – the CM buffer size (the maximum number of packets to be stored within the memory), T_u – the average time between two consecutive packets sent into the CM buffer and T – the delay constant component which is assumed to be 0,5 ms. From Table 1 the dependences of RTD_d and RTD_u on T_u can be determined, the value of T_u being chosen to vary at a constant step within the interval 0-300 ms. According to DOCSIS 3.0 when single upstream and downstream channels are considered and the Reed Solomon and FEC coding methods are applied the downstream and the upstream channel capacity values are assumed to be as follows:

Q_d = 50 Mbps with QAM-256;
Q_u = 9 Mbps with QPSK.

The parameters values to investigate the dependences of RTD_d and RTD_u on T_u are chosen according to DOCSIS 3.0. They are shown on Table 1.

Table 1. DOCSIS parameters

Parameter	Value
Q_d	50 Mbps
Q_u	9 Mbps
T	0,5 ms
T_u	0-300 ms
L_{data}	1024 bytes
L_{ACK}	64 bytes
B_{CM}	10 (packets)

When the necessary values are introduced in formula (1) the result $K = 0,347$ is obtained, which satisfies the condition for the TCP normal operation in CATV networks.
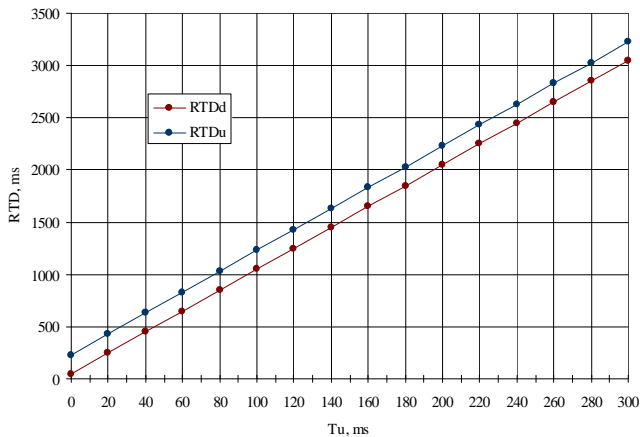
Fig. 2. Dependence of RTD_d and RTD_u on T_u

On Fig. 2 the dependence of the average delay while sending a packet in the downstream (RTD_d) and the upstream (RTD_u) direction on the average time T_u between two consecutive packets transmitted in the CM buffer is shown.

It can be seen from the diagram that the dependence of RTD_d and RTD_u on T_u is a linear function, the average delay while sending a packet being higher in the upstream direction than that in the downstream direction.

Architectures to improve the quality of the provided IP-based services

The use of broadcast approaches for sending information from/to cable television network subscribers is not effective in respect to both the necessary channel capacity and the communications reliability. On Fig. 3 an architecture solution is considered for network segmentation in the downstream direction. In that case switches are used at the head-end and in the distribution hubs. Owing to segmentation the packets in the given sequence are sent to subscribers, each subscriber receiving only the packets that are addressed to him. Thus communications reliability is provided and the channel capacity is used in a more effective way - if compared with the broadcast approach where information is transmitted to all the subscribers and each subscriber has to pick out the packets addressed to him.
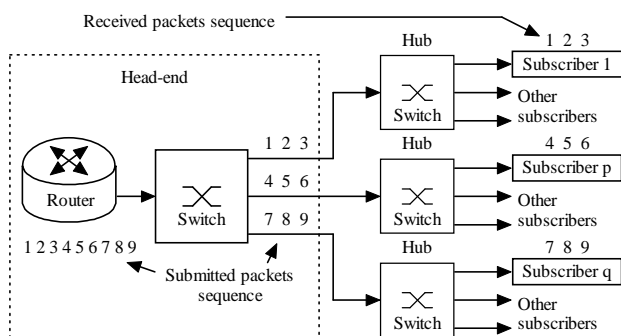

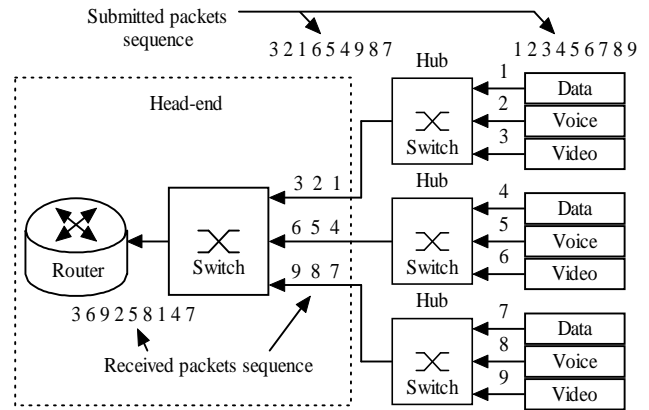**Fig. 3**. Network segmentation in the downstream direction

Fig. 4. Network segmentation in the upstream direction in terms of QoS

Segmentation can be implemented in the upstream direction as well by grouping the services according to their quality (QoS – Quality of Service), as shown on Fig. 4. In this case an architecture solution is suggested for network segmentation in the downstream direction for three types of services – data, voice and video. Packets priority is introduced according to the type of information transmitted in each packet. If numbers from 1 to 9 are assigned to each packet as shown on the Fig. 4, the received packets sequence at the head-end will correspond to the sequence of transmitted services classified by quality, i.e. video information at first, then voice services, then data.

Bit rate determination in VoIP services over cable television networks

In VoIP communications with silence suppression (UGS-AD service according to the DOCSIS standard) the voice traffic that enters the boundary routers can be represented as an on-off (or birth-death) Markov process. For each individual flow of packets transmitting VoIP the alternating periods of activity and silence are exponentially distributed random values of duration $1/\mu$ and $1/\lambda$, respectively. The time interval within which the voice source is "on" is $\lambda/(\lambda+\mu)$. We assume that when the source is "on" fixed-size packets are generated at constant intervals. On the other hand, packets are not transmitted when the source is "off". The packets' size and transmission rate depend on the voice codec chosen.

Let us consider a fluid model that is a single queue of voice packets (PQ) with N in number potential incoming voice traffics and a bit rate BR.

$$BR = \frac{BR_{voice}}{BR_{voice} + BR_{data}} Q \ . \qquad (4)$$

In this formula Q – the channel capacity and BR_{voice} and BR_{data} – the necessary bit rates for voice and data transfers respectively.

From [6] and by using the inverse Q-table [7] we can determine the value of BR for a given value of packet loss rate δ_{max}:

$$BR \geq Nm + Q^{-1}(\delta_{max})\sqrt{N}\sigma , \qquad (5)$$

where Nm – a mean and $N\sigma^2$ – a variance of the voice traffic. The packet loss rate should not exceed 3 % from the total number of packets sent in order to preserve sufficient voice quality [6]. This value can be even less than 1 %, according to other information sources. We'll consider a third case as well (when $\delta_{max} = 10\,\%$) in order to obtain a more accurate idea about the system operation. We obtain the value of $Q^{-1}(\delta_{max})$ for these 3 cases from a table given in [7]. After substitution in (5) we receive the expression for channel capacity allocation with maximum admissible loss ratio as follows:

$$BR(\delta_{max} = 0{,}01) \geq Nm + 2{,}32\sqrt{N}\sigma , \qquad (6)$$

$$BR(\delta_{max} = 0{,}03) \geq Nm + 1{,}88\sqrt{N}\sigma , \qquad (7)$$

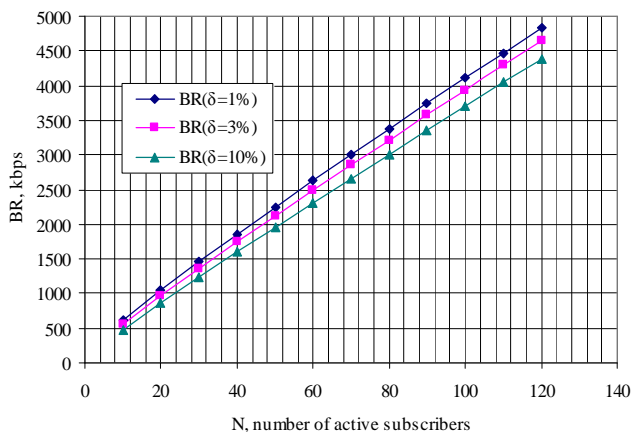$$BR(\delta_{max} = 0{,}1) \geq Nm + 1{,}28\sqrt{N}\sigma . \qquad (8)$$

Fig. 5. Dependences of BR_{min} on N and δ_{max}

We assume the maximum number of active subscribers be 120, and the values of m and σ be m = 32 kbps and σ = 39.2 kbps, respectively. After substitution in (6), (7) and (8) we get a diagram (Fig. 5) which shows the dependence of the necessary minimum bit rate BR_{min} on the number of subscribers N for three different values of the packet loss ratio δ_{max}.

Conclusions

1. The investigations here described show that when TCP is applied in CATV networks which are asymmetric in terms of the upstream and the downstream transmission speed the average delay time between two consecutive packets transmitted is higher in the upstream than in the downstream direction.

2. The necessary minimum serving bit rate has been evaluated for different values of the packet loss ratio when providing the VoIP service to cable television networks subscribers. This dependence turns out to be quite linear. In order to provide a higher service quality i.e. a minimum packet loss ratio the serving bit rate must be increased.

References

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L. T. Jordanova, J. N. Nenkov. On the IP-based Services Distribution over Cable Television Networks // Electronics and Electrical Engineering. – Kaunas: Technologija, 2009. – No. 7(95). – P. 77–81.

The peculiarities of transmitting IP-based services over DOCSIS-based CATV/HFC networks are presented. The way DOCSIS protocol acts in data transmission and in collisions solving is discussed in details. The TCP effectiveness is evaluated for CATV / HFC networks that are asymmetrical in terms of channel capacity in the upstream and the downstream direction. Architectures for the upstream and the downstream segmentation are suggested that implement switches to improve the quality of the subscriber services provided. The minimum bit rate is determined that makes it possible for the VoIP service with silence suppression (UGS-AD according to the DOCSIS standard) to be provided to the cable television network subscribers. Ill. 5, bibl. 7 (in English; summaries in English, Russian and Lithuanian).

Л. Т. Йорданова, Й. Н. Ненков. Транслирование IP-базированных услуг по CATV/HFC сети // Электроника и электротехника. – Каунас: Технология, 2009. – № 7(95). – С. 77–81.

Представлены особенности транслирования IP услуг по CATV/HFC сети в соответствии с DOCSIS стандартом. Детально обсуждается действие DOCSIS протокола при передачи данных для решения проблемы коллизий. Оценена эффективность TCP протокола при его использовании в асимметричных CATV/HFC сетях. Представлены архитектурные решения сегментации сети в восходящем и нисходящем направлениях с комутаторов, улучшающие качество абонентам предоставленных услуг. Определена скорость передачи данных при предоставлении сервиса VoIP (UGS-AD согласно DOCSIS стандарту). Ил. 5, библ. 7 (на английском языке; рефераты на английском, русском и литовском яз.).

L. T. Jordanova, J. N. Nenkov. IP paslaugų teikimas kabelinės televizijos tinklais // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2009. – Nr. 7(95). – P. 77–81.

Nagrinėjami IP paslaugų teikimo DOBSIS pagrindu sudarytuose CATV/HFC tinkluose ypatumai. Detaliau aptariamas DOBSIS protokolas ir jo poveikis duomenų perdavimui ir kolizijoms. Įvertinamas asimetrinių CATV/HFC tinklų TCP efektyvumas. Pasiūlytos siuntimo ir priėmimo kanalų segmentavimo architektūros, kurios pagerina vartotojui teikiamų paslaugų kokybę. Nustatyta minimali perdavimo sparta, užtikrinanti galimybę teikti VoIP paslaugą su tylos panaikinimo funkcija (UGS-AD pagal DOCSIS standartą). Il. 5, bibl. 7 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).