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Acknowledgement Technique in Communication Network

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Abstract

Broadband power line communications (BPLC) network is regarded as a promising technology to provide high quality audio/video services in digital homes. However, the frequency and time-varying characteristics of the channel in PLC network will cause unpredictable packet loss during the transmission, so that the retransmission is inevitable in order to guarantee the service integrity. This paper proposes a fast retransmission method by introducing the internal node to join the work of retransmission based on the packet loss indication from the destination node. The proposed mechanism can be deployed in the system suffered with burst or constant packet loss. Comparing with the traditional retransmission method which totally relies on the source node, the proposed mechanism can reduce the retransmission step and improve the target packet hit rate for both unicast and multicast services.

Keywords: Power Line Communication, retransmission.

Introduction

Power line communication network has proven to be a cost effective solution for the construction of in-building network like digital home to deliver broadband audio video services. One obvious advantage is the installation cost saving resulting from the usage of the existing low-voltage cable and AC outlets, furthermore, the development of power line communication standards such as Home Plug AV[1] and

Open PLC European Research Alliance(OPERA)[2] boost the achievable data rate up to 200Mbps in physical layer. However, the indoor power-line channel is a frequency selective fading channel with time-varying characteristics. One type of interference is coming from the colored and impulsive noise generated by electrical appliances and external sources [3], the other impact is the multi-path response corresponding with the power cable layout and loading conditions [4]. Such a harsh transmission environment could cause highly unpredictable interference and damage a series of consecutive packets, According to the field test in [7], the behaviour of packet loss in PLC network has two features:

a. The packet loss rate could be very small if the running traffic is light loaded;
b. The packet loss rate increases significantly once the sending rate is higher than some threshold value. Such threshold may have great variation under different connection topology or in the environment with other power appliances interference existing.

In recent years, some QoS enhancement technologies for reliable video transmission through PLC network are proposed.

Most of them focus on the forward error correction (FEC) in application layer [5], and the deployment of multiple description coding (MDC)[6][8]. Because these two approaches both require adding redundant information with the service stream to improve the robustness when part of the information is lost, but the transmission efficiency will be affected greatly due to the packet loss feature (b) mentioned above, especially under the condition when packet loss rate is increasing. The introducing of redundant data will add the burden of traffic load, which will lead to more severe packet loss. Therefore, the supplement of the redundant data is only suitable for the scenario when the average packet loss rate is low and there exists enough free bandwidth. Although retransmission for lost packet will also bring some redundancy, such mechanism is necessary for maintaining reliability, a downs flexibility to be deployed in any system suffered with burst or constant packet loss.

In this paper, we analyze the problem of loss packets retransmission in the PLC network for both unicast and multicast services. The traditional retransmission happens directly between the source and destination node. Take ARQ (Auto Repeat reQuest) for example, the receiver sends out acknowledge signal (NACK) when packet loss is found; on the other side, the source node is configured to transmit the requested data block. In order to improve the retransmission efficiency, we propose a fast retransmission mechanism by introducing the internal node to join the retransmission work, some internal nodes near to the destination node will give response when the NACK is captured. Generally speaking, the channel condition between the internal node and destination node is better than that between source node and destination node considering the factor of shorter transmission distance and lower interference, consequently, the target packet hit rate will increase and the step of retransmission will reduce. In addition, the proposed mechanism can be easily extended into multicast transmission by selecting one indication node as a agent to send NACK message, then fulfilling retransmission from one selected internal node to all of the receivers in the same branch, thus a balance of robust transmission and light network load can be achieved. The paper is organized as follows: Section II describes the packet loss pattern in the indoor power line environment. The detailed analysis and algorithm description for the proposed fast retransmission mechanism are presented in Section III. The simulation results with the comparison are shown in Section IV and a conclusion is given in Section V.

Packet Loss in the Power-Line Environment

In [7], only two PLC modems are placed in the test Environment, we extend the test by adding more PLC modems into the transmission path. As shown in Fig.1, one source node and three destination nodes are placed along a power line cable, each node connects to a PLC modem. In each round of test, the source node transmits data to the receiver node at designated sending rate generated by IxChariot console software. The packet size is 1024 Bytes and the UDP streams are deployed to explore the packet loss behaviour under different throughput condition.

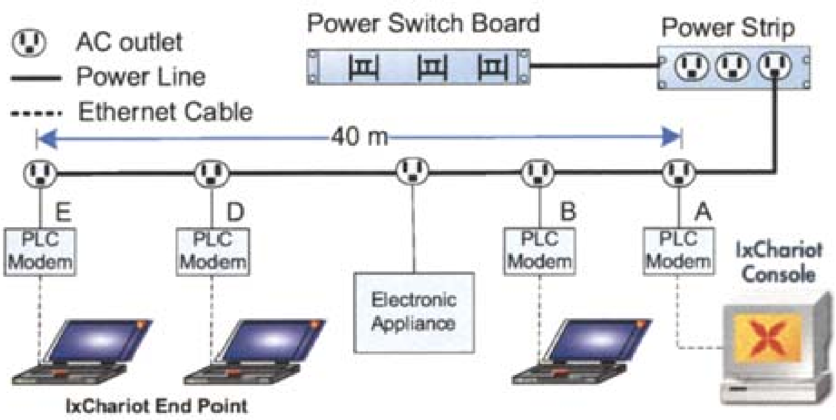


Figure 1: Test environment of PLC.

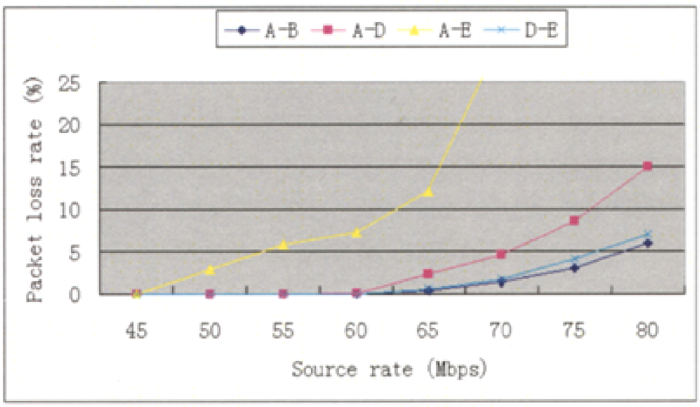


Figure 2: Packet loss rate under different source rate.

The packet loss rate between peers with different sending rate is shown in Fig.2; two conclusions can be made from the results,

1. For each transmission, the packet loss rate increase significantly when the source sending rate exceeds the threshold value, this conclusion has been mentioned in [7];
2. Under the same sending rate, the packet loss rate varies lot among different peers. For example, when sending rate is65Mbps, there is no packet loss between modem A and B, and there is nearly 2% packet loss between modem A and D, but the packet loss is higher than 10% between modem A and E. The reason is data transmission through longer path has great probability to meet packet loss by traffic congestion and noise interference.

Although some bandwidth estimation mechanism can be deployed to scan the available bandwidth before data transmission and dynamic adjust the service rate in the source node, it cannot guarantee the zero packet loss in the time-vary power line channel because of the following three constrains:

1. Bandwidth estimation is only suitable for unicast service; it cannot be deployed in multicast or broadcast service in which multiple receivers nodes exist;
2. The accuracy of bandwidth estimation based on the calculation of packet loss rate in the upper layer depends on the time duration of ACK interaction under passive mode and the response of probing packets in the active mode, packet loss still happen during the interval of source rate adjustment;
3. Many high quality video based applications require sustained bandwidth guarantee, such as HDTV. If the transmission rate for video streaming is cut down following the variation of channel condition, obviously, the visual effect will be declined.

Proposed Retransmission Mechanism

In a PLC network, a transmission path is predicted in a predetermined timing before the communication, which means the network topology will not change during the service transmission. In addition, the information of terminals aligned in a transmission path can be obtained manually by network administrator or automatically by the neighbor discovery message broadcasting. On the other hand, PLC is a shared media system, all the internal nodes located along the transmission path between the source and destination node can catch the information and signal packets in Media Access Control (MAC) layer, but none of them contribute to the transmission work in the conventional power line modems.

It is clear to show in the Fig.2, if the retransmission is happened between node D and E, the target packet hit rate,

$$\text{TPHR} = \frac{\text{Number of receiving packet at destination node}}{\text{Number of original packet at source node}}$$

will be higher than the same operation between node A and E because of the better channel condition, and the transmission with high TPHR value is regarded as good data integrity. The proposed retransmission mechanism is composed of two steps.Step1 involves the transmission sequence determination based on the topology and the service session allocation. To simplify the explanation, we define two types of internal node as following:

a. 1st node – the 1st nearest node to the receiver;
b. 2nd node – the 2nd nearest node to the receiver.

For unicast service, if there are equal or more than two internalnodes aligned in the transmission path, the 1st and 2nd node will be assigned directly, if there is one internal node existing, only1st node will be labelled. For multicast service, if all the receivers are located in the same branch, the node with the longest distance from the sender will be selected as the node to send NACK indication during transmission, the assignment of internal nodes follows the same policy for unicast service. If the receiver nodes are located in the different branch, the selection of longest distance node and the assignment will be done separately in each branch.

Step 2 includes the algorithm for retransmission processing. To constrain the latency, the request for the retransmission of one loss packet can be done at most three times. When the receiver sends out NACK signal, there is an index to tell the time sequence for the request, then different internal node or source node will send out the buffered data if necessary. As shown in the Fig 3. when the receive terminal sends out the first NACK (sequence=1) for data block #1, the1st node will be assigned to retransmit the data block #1 to the receive; if this time of retransmission is failed, the 2nd node will do the same thing as long as receiving the second NACK(sequence=2) for data block #1. The source node is in charge of the response for the NACK with sequence number equal to 3for a final try. Because in each time of retransmission, only one node (either internal or source) could send out the requested data block, there will be no interference during the time of retransmission.

Normally, in the MAC protocol of power line communication, both Time Division Multiple Access (TDMA) and Frequency Division Multiple Access (FDMA) allocation modes are supported. The retransmission can be scheduled in the contention free time slot in the TDMA or non-conflict frequency in the FDMA, so that there are no multiple copies of the same packet during retransmitted.

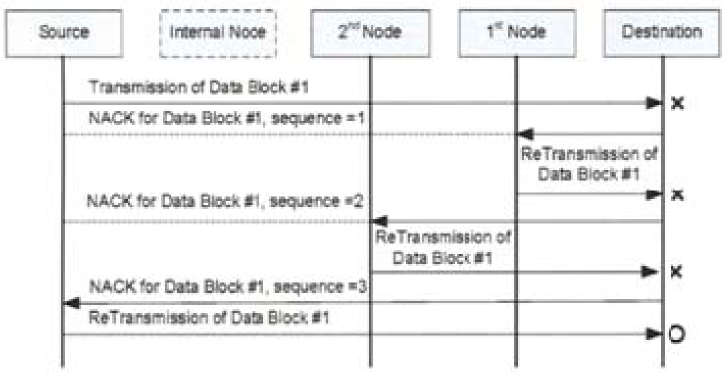


Figure 3: Process flow of the proposed mechanism.

For multicast service, most existing multicast protocols like reliable multicast transport protocol (RMTP)[9] adopt a static retransmission scheme (by creating multiple unicast sessions or a new multicast session) to retransmit lost packets; such static unicast mode may result in great network load increase, while the multicast mode will cause the accuracy variation among receivers.

To support multicast transmission, only the assigned indication node can send out the NACK signal, there transmission will be processed in each branch respectively. And one retransmission time slot will be allocated in the frame beacon for each branch if the receiver nodes are located indifferent branches to prevent interference. To simply the problem, at most 2 times request for one lost packet is allowed.

Mathematical Analysis

A linear model is created for the theory analysis. As shown in Fig. 4, the node with label *S* is the source node, *D* represents the destination, and node *I* is the internal node, which will send out response when capturing the NACK indication from the receiver. p ,p_a,p_b are different packet loss rates among the connections. Based on our field test, there is a conclusion that

$$pa+pb<p \tag{1}$$

under the same source rate.

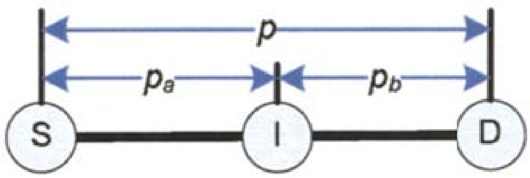


Figure 4: Linear model.

To simplify the analysis, the efficiency of one time retransmission is compared. Assuming in receiver side, there is one unit packet loss in last transmission, so that it will send out one unit NACK indication. In the traditional way, the probability of target packet hit rate (TPHR) of retransmission is *TPHR* (*old*) =(1−*p*)2 ; the TPHR of the proposed mechanism is ($TPHR\ proposed$)=$(1–P_b)^2(1\text{-}P_a)$,while $(1–P_b)^2$ is the NACK and data packet probability between node *I* and destination. $(1\text{-}P_a)$ is the matching probability of requested packet inside node I's data buffer.

To compare the variation of above TPHR we define 2ratio *x*, *y* during the calculation. $p_a = x^* p$ and $p_b = y^* p$,according to the constrain (1), we can get

$$0<x+y<1,\ and\ 0<x,y<1 \tag{2}$$

The Fig.5 shows the *TPHR* comparison under different combination of (x, *y)*. The x-label represents the packet loss rate *p* between the source and destination node. The scenarios can be divided into three categories:

a. Internal node is near to the source node, for the case when x=0.1 and y=0.8;
b. Internal node is near to the destination node, for the case when x=0.8 and y=0.1;
c. Internal node is in the middle, for the case when x=0.5 and y=0.4

It is obvious that no matter where internal node is placed, the proposed mechanism can all achieve higher *TPHR* than the traditional method.

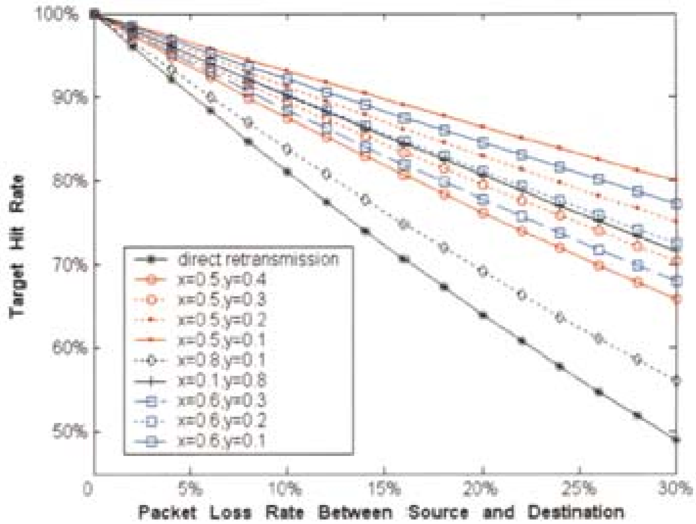


Figure 5: Target packet hit rate.

Algorithm Description

The flow chart of the retransmission algorithm is shown in Fig.6, to simply the explanation, we take unicast service for example and the 1st node is selected for case study. For multicast service, the similar processing will be done in each branch.

The initial process will determine the role of retransmission node (the 1st or 2nd node) with index in the transmission path, and the service session information will be broadcasted through the path to let each selected node know which session it should work for; afterward, these active nodes should scan channel and capture passed packets. For data packets to itself, destination address will be checked and the normal packets processing such as integrity checking, refragmentation, payload filtering will be fulfilled. For those data packets in the target service session, they will be stored in the local buffer for the reservation in the queue according to its destination. The buffer depth can be set dynamically based on the type of service. For the NACK signal packet from the destination node, the sequence number is parsed to determine which time request it is. If the sequence number is larger than 2, meaning this packet has already been re-requested for many times, it would be the source node to do the retransmission; otherwise, local label will be compared with the sequence number to determine whether or not to give response. The following steps include searching in the buffer to check whether the data block indicated in the NACK is existed. If so, this block will be sent out in the next re-transmission time slot. If there is no such data block is found, a new NACK message will be generated with sequence number plus one.

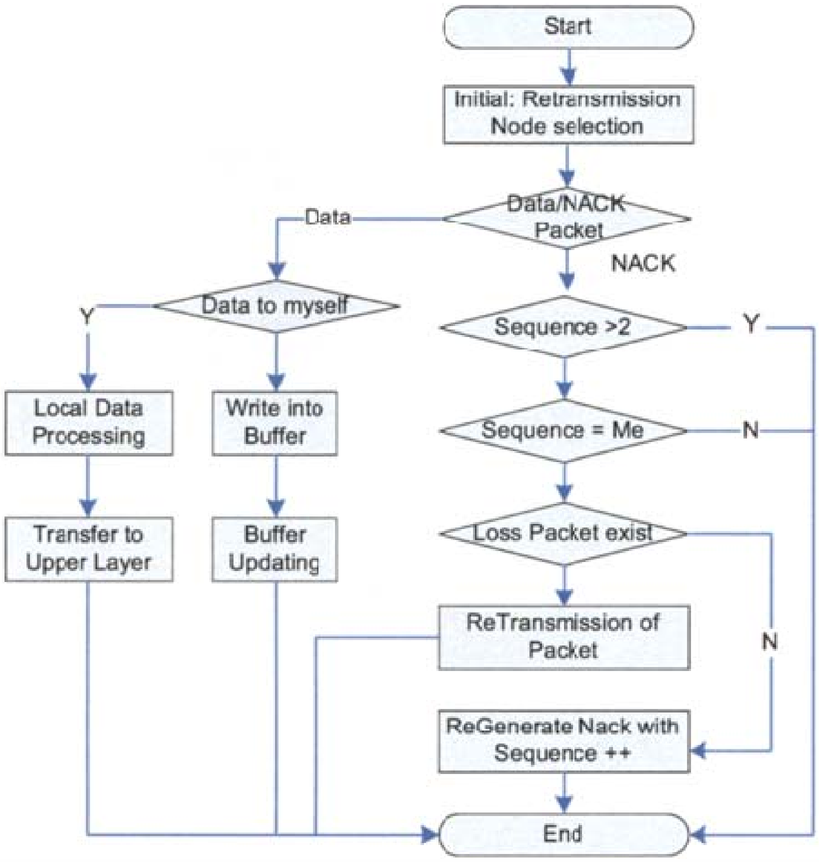


Figure 6: Flow chart of the proposed mechanism (cast study in the 1st node).

Simulation Result

In this section, we use Mat lab tool to do simulation and evaluate the performance of the proposed retransmission mechanism. Firstly, unicast service is investigated in the topology shown in Fig.1, two groups of packet loss combination are considered. In case one, there is severe packet loss from source node, representing the bad channel quality, and p=0.2,$p1$=0.05,$p2$=0.02 , here p,$p1$,$p2$are the packet loss rates from the source/1st/2nd node to the destination node respectively. Fig. 7 shows the comparison of TPHR, it is clear to see the proposed mechanism can achieve 99%+ TPHR with one time retransmission while the traditional method needs three times retransmission to achieve the same performance. In case two, p=0.1, $p1$=0.03, $p2$=0.01, which means the channel quality is better than that in case one, we can see the proposed mechanism still has better performance than the traditional method.

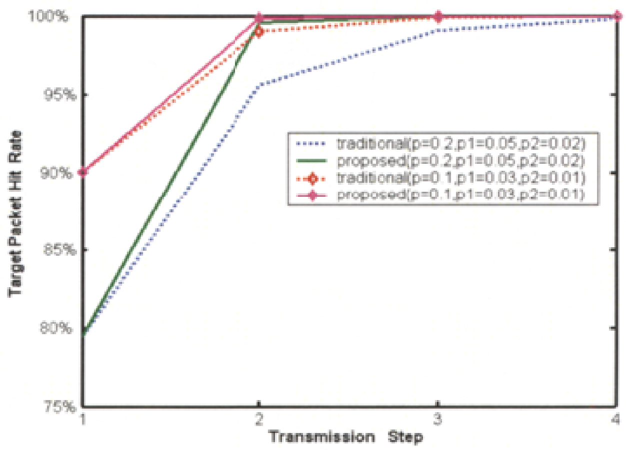


Figure 7: Target packet hit rate comparison for unicast service.

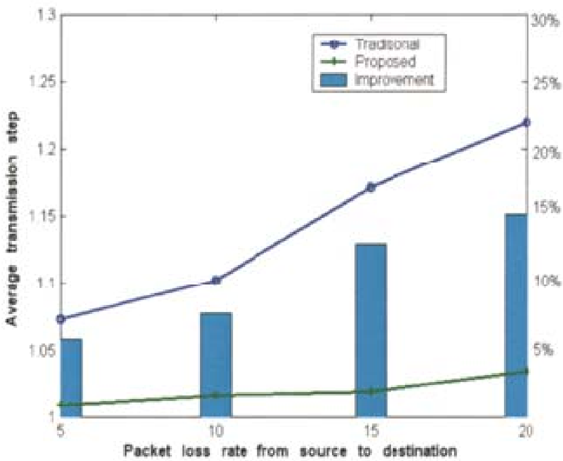
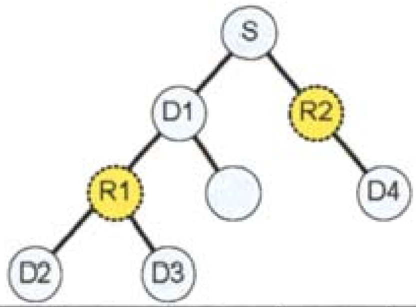


Figure 8: Average transmission step for unicast service.

Fig.8 shows the average transmission step under different packet loss from the source to destination node. We can see the proposed mechanism has 5% – 15% improvement in the aspect of step reduction. In the mean time, the transmission step is corresponding with the latency based on the interval of NACK signal message. In PLC network, the typical duration for data frame is 33ms (in North America) or 40ms (in Europe). For example, if the receiver has chance to send out ACK once every 5 frame, according to the simulation result, the average latency for retransmission data under the proposed mechanism can be less than 200ms, while the average latency under conventional method will be nearly 250ms in the worst case.

We also compare the performance for the multicast service, a parameter named average packet loss rate (APLR) is used to compare the system improvement when the proposed retransmission mechanism is deployed. The topology for multicast service is shown in the Fig.9, node with label *S* is the source node, node with label *D* is the destination node in the same multicast session, according to the policy for the 1st node selection mentioned in the Section III, On the left branch, node *D2* is selected as indication node and the *R1* is the 1st node; on the right branch, *D4* is selected as indication node and the *R2* is the 1st node. The packet loss rates among different peers are listed in the table.



	D1	D2	D3	D4	R1	R2
S	0.01	0.2	0.15	0.1	0.05	0.03
R1	0.05	0.05	0.05			
R2				0.05		

Figure 9: Topology for multicast service.

Fig. 10 shows the comparison of APLR for multicast service before and after the introduction of retransmission. If no retransmission, the APLR within four destination nodes is 11%, after one time retransmission by R1 and R2, the APLR declines to 2%.

Fig.11 shows the comparison of traffic load generated by one time retransmission between the static unicast retransmission mode and the proposed mechanism. The same amount of NACK is sent out from the receiver node, and in static unicast mode, multiple sessions from source to destination node will be built during retransmission. It is clear to see that the proposed mechanism can reduce the additional traffic load over the network and avoid the duplicate traffic during retransmission.

Conclusion

This paper presents a fast retransmission mechanism which can be deployed in the indoor power line communication system. Some internal nodes sharing with the transmission path are introduced into the retransmission processing. Comparing with the traditional method that totally depends on the source node to do such work, the proposed method will greatly improve the target packet hit rate when the channel condition between source and destination node is bad, and the retransmission step and latency will be reduced at the same time. Furthermore, it can also enhance the data integrity for multicast service by one time retransmission without introducing duplicate data. In the future work, the repeat function of internal node can be added into the mechanism to guarantee end-to-end bandwidth and decrease the packet loss.

Figure 10: Average packet loss rate for multicast service.

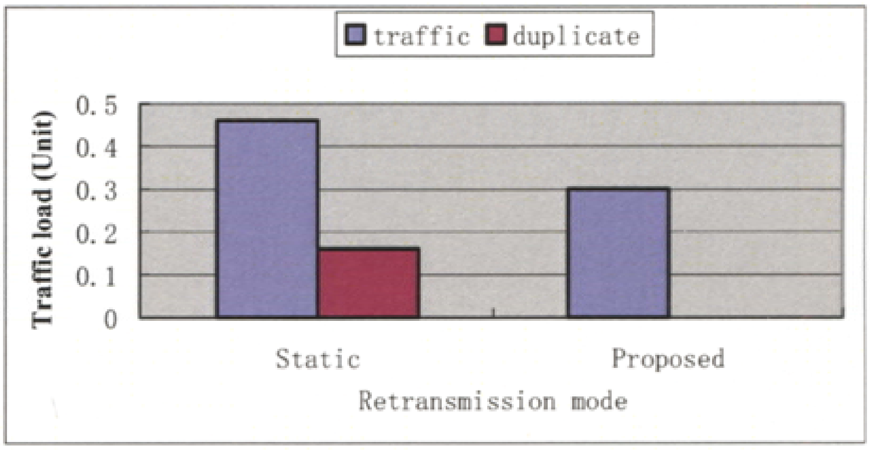


Figure 11: Traffic load comparison for multicast service.

Reference

[1] Home Plug Powerline Alliance http://www.homeplug.org

[2] Open PLC European Research Alliance http:// www.ist-opera.org/

[3] H. Meng, Y. L. Guan, and S. Chen, “Modeling and analysis of noise effects on broadband power-line communications,” IEEE Trans. Power Del., vol. 20, no. 2, pt. 1, pp. 630–637, Apr. 2005.

[4] D. Anastasiadou and T. Antonakopoulos, “Multipath characterization of indoor power line networks,” IEEE Trans. Power Del., vol. 9, no. 1, pp. 90–99, Jan. 2005.

[5] M. Luby, M. Watson, etc., "High-Quality Video Distribution using Power Line Communication and Application Layer Forward Error Correction", IEEE ISPLC 2007. pp. 431-436, Mar. 2007
[6] K.L. Chang, C.Y. Shiann, etc, "Robust Video Streaming over Power Lines", IEEE, ISPLC 2006, pp.196- 201, March 2006
[7] K.L. Chang, C.Y. Shiann, etc, "Bandwidth Estimation of in-Home Power Line Networks", IEEE, ISPLC 2007, pp.413-418, March 2007
[8] R. Bernardini, M. Durigon, etc."Robust transmission of multimedia data over power-lines" IEEE ISPLC 2005, pp.295-199, April 2005
[9] J. C. Lin and S. Paul, “RMTP: A Reliable Multicast Transport Protocol”, IEEE INFOCOM 1996, San Francisco, California, March 1996.