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Triple Play Services Packet Scheduling Performance Evaluation

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Introduction

Triple play networks are the next stage in development of next generation communication networks (NGN). Via the triple play service the user can have a relish for all the aforementioned services; Video on Demand (or Video Broadcasting, such as IPTV), VoIP telephony and fast internet access, all at the same time. While several ISPs have proposed their architecture and detailed specifications to support triple-plays services in NGN, they all have to deal with a critical issue: how to schedule traffic and allocate bandwidth for triple-play services at both backbone and access links.

Triple Play technology is nowadays considered as an indisputable trend. The Differentiated Service (DiffServ) architecture is preferred over “Hard Quality of Service” (QoS) Integrated Service (IntServ) architecture. Moreover it applies perfectly to triple play, as it satisfies differing QoS requirements. In addition, latest developments in the area of video and audio encoding impose new challenges in content delivery systems. The combination of these three technologies is expected to raise additional QoS related issues. Problems such as full exploitation of available bandwidth and providing adequate QoS to subscribed users by meeting the requirements of all three supported services (video, voice and data) must be addressed [1]. In this paper we are proposing the new adaptive (AFQ) scheduling model for triple play services , reasoned by the virtual queue and dynamic weighted queues service coefficient , changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load.

Triple-play services and architecture overview

Triple Play is the bundled service of Voice, Data, and Video services offered for a price that is less than the total price of the individual services. However, there is no standard for provisioning the Triple Play services, rather they are provisioned individually, since the requirements are quite different for each service. Furthermore, in our vision different services can be provided by different Service Providers, while the customers are reaching these services using the same access network. The high speed internet access is currently provided mostly based on PPPoE connections. The bandwidth provisioning is asymmetric with higher download rate. The VoIP is the most cost-effective solution for voice service. However, it has strict delay and jitter requirements, thus the provisioning of this service requires priority over the other services to achieve the best QoS guarantees possible. Call level admission control mechanism is also required to keep the guarantees. The IPTV is the most promising video service for Triple Play. The basic IPTV service is the video broadcast where streams with different resolutions can be supported [2].

The triple play service network is presented in Fig.1.

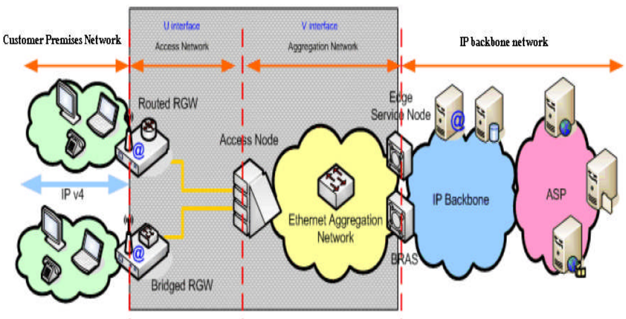

Fig. 1. The triple play service network structure

The Triple Play services require end-to-end provisioning: the services should get the proper differentiation through all domains. In the home network the home gateway distributes the services to the PC, set-top box and IP phone. The home gateway is provisioned by the service provider to differentiate the services and support three QoS channels in the first mile [3,6]. The access nodes aggregate the traffic of several customers. The role of the Metropolitan Aggregation Network (MAN) is very important, it aggregates the traffic of hundreds of thousands of customers and spans from the first mile to the service edges. In this paper we focus MAN network node aggregation model. In this paper we are proposing the new adaptive (AFQ-Adaptive Fair Queuing) scheduling for QoS management model, reasoned by the virtual queue and dynamic weighted queues service coefficient φ , changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load. This model allowing managing the packet delay in the node and network. In this paper we are proposing WFQ and LLQ (Low Latency Queue) combination for comparison with our new scheduling methods.

Triple play services packet scheduling

The provisioning of the Triple Play services requires an architecture that provides QoS guarantees while allows efficient network utilization supporting the requirements of all three service types. Furthermore, support for multiple network domains or service providers is also required for end-to-end QoS support. The QoS simulated architecture is based on the IETF Differentiated Services (DiffServ) Architecture described in RFC 2475.

At first we calculate for each triple play service mean delay in the link (Eq.1):

$$\tau_i = \frac{\sum_{j=1}^{n} \lambda_j L_j^2}{\left(C - \sum_{j=1}^{i=1} \lambda_i L_i\right) \cdot \left(C - \sum_{j=1}^{i} \lambda_i L_i\right)} + \frac{L_i}{C}, \ \mathrm{i} = \overline{1, n}, \quad (1)$$

where λ - flow intensity of different priority service packets, L - mean value of packet length of different priority service, C - transmission channel throughput , n – number of service priority.

If we know the budget average delay, we can calculate necessary transmission channel throughput for high priority service. Let's put, That the sizes of packages of all types are identical $L_i = L$, and $\lambda_i = k\Lambda$, then necessary transmission channel throughput is calculated use formula:

$$C > \frac{L}{2}\left\{\frac{1}{\tau_1^*} + k\Lambda + \left[\left(k\Lambda - \frac{1}{\tau_1^*}\right)^2 + \frac{4\Lambda}{\tau_1^*}\right]^{\frac{1}{2}}\right\}. \quad (2)$$

The next we analyzing scheduling algorithm. The adaptive telecommunication service quality management means for differentiating stream transmitting quality ensurance model to the Internet network reasoned by the quality marginal value and evaluation of dynamic network load was proposed for solving this problem in this paper. The new M/G/1K – AFQ model for the service in the queues, reasoned by the virtual queue and weighted queues service coefficient ϕ , changing according to the evaluation of the packet state in the node and allowing to manage the packet delay in the node and network was proposed.

The structure of the analyzed node is given in Fig. 2.

Many authors use similar or underlying M/G/1 model not characterizing the queue length for the description streams of different type [3]. M/G/1/K model is used for the queue parameters evaluation as the node in the work is described by the buffers of the finite queue [4]. The proposed stream processing in the router is reasoned by the adaptive quality management changing weighting queue service coefficient ϕ that changes according to the evaluation of the state of packets in the node and allows manage the packet delay in the node and network. M/G/1 model modification with multiple vacations is used for the mathematical description of the node. This modification was chosen according to the adaptive node (router) description proposed adaptive service model M/G/1/K – AFQ reasoned by the disciplines of weighting queue service.

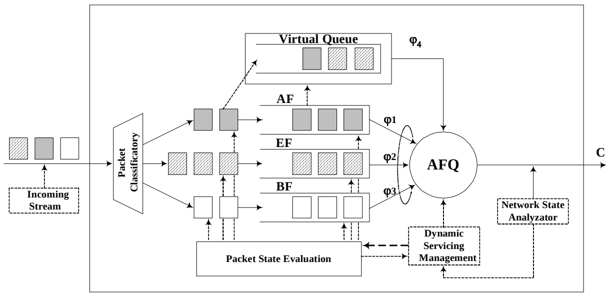

Fig. 2. The structure of the node [4]

The delay time of analyzed node is described by the equation [5]:

$$D = V + \frac{1}{\mu} \cdot \left(N_l + \sum_{j \in J} N_j\right), \quad (3)$$

where N_l is the number of packets in l-th queue; N_j is the number of packet in all the system, except k-th queue; V is the required time medium for the service of all orders in the system; $\frac{1}{\mu}$ is the average time of packet service.

The parameter V is given by expression 4:

$$V = \sum_{j=1}^{K}\left\{\frac{(1-\rho_{apt})}{\lambda_j \cdot \overline{\overline{T_{nutr_j}}}} \cdot (1+K) + \frac{\rho_{apt} \cdot (\rho - 1)}{\rho}\right\} \cdot \frac{\overline{T_{apt_j}^{\ 2}}}{2\overline{\overline{T_{nut_j}}}} =$$
$$= \frac{1}{2} \cdot \sum_{j=1}^{K}\left\{\frac{\overline{T_{apt_j}^{\ 2}} \cdot (1-\rho_{apt}) \cdot (1+K)}{\lambda_j \cdot \overline{T_{nutr_j}^2}} + \frac{\rho_{apt} \cdot (\rho - 1) \cdot \overline{T_{apt_j}^{\ 2}}}{\rho \cdot \overline{T_{nutr_k}}}\right\} = \quad . \quad (4)$$
$$= \frac{1}{2} \cdot \left\{(1-\rho_{apt}) \cdot (K+1)\sum_{j=1}^{K} \frac{\overline{T_{apt_j}^{\ 2}}}{\lambda_j \cdot \overline{\overline{T_{nutr_j}^{\ 2}}}} + \frac{\rho_{apt} \cdot (1-\rho)}{\rho}\sum_{j=1}^{K} \frac{\overline{T_{apt_j}^{\ 2}}}{\overline{\overline{T_{nutr_j}}}}\right\}$$

The improved upper and lower bounds of number of packets in node are respectively:

$$N_j \le \min\left(\sum_{\substack{k=1 \\ j=1}}^{J}\left(\left(N_k + 1\right) \cdot \frac{\phi_j}{\phi_k}\right) + \frac{\phi_j}{T_{serv} \cdot C}; \qquad N_j + \frac{M}{C} \cdot \lambda_v\right), \quad (5)$$

$$N_j \ge \min\left(\max\left(\sum_{\substack{k=1 \\ j=1}}^{J}\left(\left(N_k + 1\right) \cdot \frac{\phi_j}{\phi_k}\right) + \frac{\phi_j}{T_{serv} \cdot C} - 1,\ 0\right); \quad N_j + \frac{M \cdot \lambda_v}{C}\right), \quad (6)$$

where: φ_j – weight coefficient of queue j; φ_k – weight coefficient of queue k; T_{serv} – packet service tme; C –

link throughput; M – packet size; λ - packet arrive intensity.

The queue scheduling weight coefficient φ are being changed dynamically. The new set of queue scheduling weighting coefficients for *i-th* session is based by the evaluation results of the packet state in the node. The *k-th* queue weight coefficient change can be written as [4]:

$$\varphi_k(t)=\begin{cases}\phi_k^{'}(t)=\varphi_k(t), & \text{if } \forall N_{i+k}\in Q^{+}\\ \phi_k^{'}(t)=\varphi_k(t)-\Delta\phi(t), & \text{if } \forall N_k\in Q^{+}, \text{ any one } N_i\in Q^{+-}\\ \phi_k^{'}(t)=\varphi_k(t)+\Delta\phi(t), & \text{if any one } N_k\in Q^{+-} \text{ and } \forall N_i\in Q^{+}\\ \phi_k^{'}(t)=0, & \text{if any one } N_{i+k}\in Q^{+-}\{W_\Lambda<W_k<W_C\}\end{cases}. \quad (7)$$

The change of queue weight coefficient $\Delta\varphi$ is calculated according the equations:

$$\Delta\phi^{+}(t)\geq\frac{N_k\{Q^{+-}\}}{\underset{i\neq k}{N_i}};\ \Delta\phi^{-}(t)\geq\frac{N_k\{Q^{+}\}}{\underset{i\neq k}{N_i\{Q^{+-}\}}}. \quad (8)$$

Using those parameters, EIGRP protocol calculates three metrics: Route weight metrics W_k; route weight metrics, using static (supporting) delay W_c; route weight metrics, using residual delay W_Δ. The evaluation of packet state is performed according to the obtained metrics means.

Evaluating possible packet states, three gradations are used:

- “good” state Q^+;
- “satisfactory” state Q^{+-};
- “bad” state Q^-.

Those packet state conditions are the function from the weight metrics $Q=f(W_i,W_c,W_\Lambda)$.

$$Q=\begin{cases}Q^{+} \text{ if } W_k<W_C<W_\Lambda,\\ Q^{+-} \text{ if } W_C<W_k<W_\Lambda\cup W_C<W_\Lambda<W_k\cup W_k<W_\Lambda<W_C\cup W_\Lambda<W_k<W_C,\\ Q^{-} \text{ if } W_\Lambda<W_C<W_k.\end{cases} \quad (9)$$

Every packet state is converted into the composed binary code that is included into the free places of DSCP. All this formula are used for modify edge node performance.

Simulation model and results

Simulation was provided using telecommunication network program package Opnet Modeler. Simulation network structure is shown in the figure 3. The 100 Mb/s transmission speed is set between the final network nodes and edge routers. The speed is limited to 2 Mb/s in user network . The initial parameters for service determination are given in Table 1. Simulation was made in three cases, when the load of the network (network use) changed: ρ = 0,2 to 0,3; ρ = 0,5 to 0,7; ρ = 0,75 to 0,9. The main parameters given in Table 2 were set during the imitative simulation using the proposed adaptive quality managing means AFQ. Taking into account the proposed SLA service parameter for quality evaluation in ITU-T and ETSI recommendations.

The obtained simulation results using standard queue service discipline WFQ and proposed adaptive quality managing means AFQ are given. Coding/encoding, spreading and bufferization delay time-frames are not included performing the simulation and analyzing the packet delay time between the final users. The components of delay time-frames are allocated to the fixed delay and do not depend on dynamic load; so during the simulation we analyze only the variable delay (packet delay in queues of the node). So, the allowed delay margin for the voice is not 150 ms, but 40 ms.

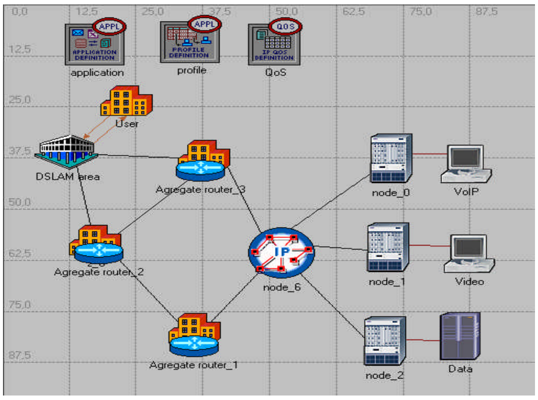

Fig. 3. Simulation network structure

Table 1. Service parameters determination

Service	Stream intensity			Prior.
	Codec	File size in bytes	Traffic intensity	
Voice	G.729	-	Poiss (30)	EF
Video	H.261	-	Poiss (20)	AF43
FTP	-	700000	Exp (36)	BE
Data base	-	327680	Exp(12)	BE
E-mail	-	20000	Exp (36)	BE
HTTP	-	10000	Exp (12)	BE

Table 2. The parameters of AFQ queues service methods

Service	**Allowed delay margin T_{max}**	**The set static medium delay time meaning of one node T_s**	**Service coefficient ϕ of initial weight queues**
Voice	40 ms	2ms	20
Video	80ms	4ms	15
Other	250ms	15ms	5

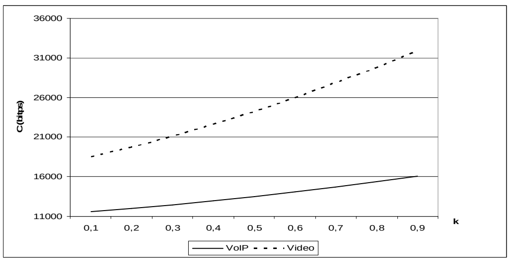

Fig. 4. Necessary transmission channel throughput ρ = 0,2

In the case ρ = 0,20 to 0,30, analyzing the obtained results of the simulation we can observe that all the provided service provided in the network does not exceed posed allowed delay target time when the loading of the network is small (ρ = 0,20 to 0,30) using standard quality methods (WFQ) (voice – 40 ms; video – 80 ms; HTTP, FTP – 250 ms).

The service of real time (voice and video) not always meets the allowed quality demands when there is medium network use (ρ = 0,50 to 0,70). The simulation result are shown in Fig. 5 – 6.

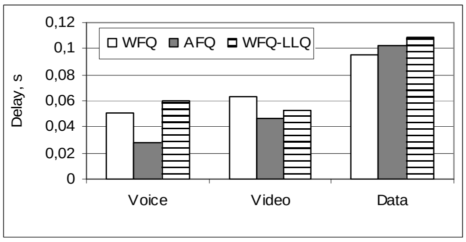

Fig. 5. Average delay time (ρ = 0,50 to 0,70)

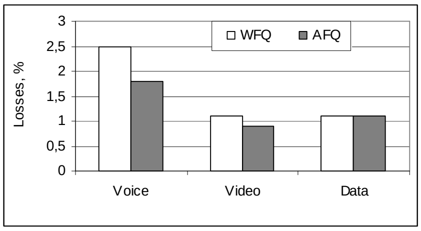

Fig. 6. Packet loses (ρ = 0,50 to 0,70)

In the case □ = 0,75 to 0,9 the obtained results are given in Fig. 7

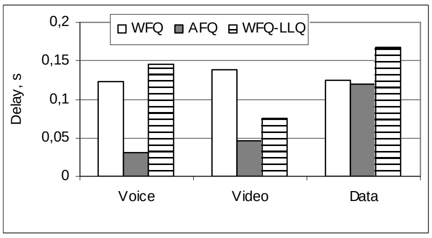

Fig. 7. Average delay time (ρ= 0,75 to 0,9)

Conclusions

Triple play services impose heavy bandwidth requirements for video and latency constraints for voice. Using proposed adaptive AFQ scheduling the average delay of packets with the highest priorities decrease from 51 ms to 28 ms and losses from 2,5 % to 1,8 % during the imitation simulation when the network is loaded (ρ = 0,50 to 0,70). In this way, the mean delay of video packet of lower priority decreases from 63 ms to 46 ms having used the proposed adaptive quality management means. The losses because of allowed delay limit having used the proposed adaptive quality management means decreases from 1,1 % to 0,9 %

References

1. **Jha Sajay.** Engineering Internet QoS.- Artech house.,2002 - 314p.
2. **Kern Andras.** On the Optimal Configuration of Metro Ethernet for Triple Play // IEEE Communication., 2006.-P. 334-341.
3. **Dekeris B., Narbutaitė L., Adomkus T.** Packet delay evaluation using M/G/WFQ model // WSEAS Transactions on communications, Issue 2, Volume 4, 2005. – P. 51-56.
4. **Dekeris B; Narbutaitė, L; Adomkus, T.** A new adaptive fair queueing (AFQ) scheduler for support SLA // ITI 2007 ,- ISBN 978-953-7138-09-7. - Zagreb, 2007, P. 597-602
5. **Villy B. Iversen**. Teletraffic engineering and network planning. DTU. 2005.-350p.
6. **Vishal Phirke**. Traffic Sensitive Active Queue Management for Improved Quality of Service. Report. Worcester Polytechnic Institute. 2002.- 61p.

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Triple Play technology is nowadays considered as an indisputable trend. Triple Play is the bundled service of Voice, Data, and Video services. Moreover it applies perfectly to triple play, as it satisfies differing QoS requirements. Problems such as full exploitation of available bandwidth and providing adequate QoS to subscribed users by meeting the requirements of all three supported services (video, voice and data) must be addressed . In this paper we are proposing the new adaptive (AFQ) scheduling for triple play services QoS management model, reasoned by the virtual queue and dynamic weighted queues service coefficient ϕ, changing according to the evaluation of the packet state in the node, reasoned by delay target time and present network load. Ill. 7, bibl. 6 (in English; summaries in English, Russian and Lithuanian).

Л. Нарбутайте, Б. Декерис. Оценка алгоритма обслуживания очередей для услуг Triple Play // Электроника и электротехника. – Каунас: Технология, 2008. – № 6(86). – С. 85–88.

Услуга triple play, т.е. предоставление единого канала связи для передачи голоса, видео и данных. Техническое обеспечения заданного QoS – это в первую очередь реализация необходимой полосы пропускания сети абонентского доступа. Важным аспектом сети передачи для архитектуры Triple Play является то, насколько полно сеть обеспечивает поддержку различной производительности и функций QoS, требуемых для каждого типа услуг. В этой статье предлагается адаптивная (AFQ) модель для управления QoS очередью в маршрутизаторе. В этом методе введен динамический коэффициент обслуживания очередей, изменяющийся согласно оценке QoS состояния пакета в узле, временем задержки и нагрузкой сети. Ил. 7, библ. 6 (на английском языке; рефераты на английском, русском и литовском яз.).

L. Narbutaitė, B. Dekeris. Triple Play paslaugų srauto aptarnavimo metodo įvertinimas // Elektronika ir elektrotechnika. – Kaunas: Technologija, 2008. – Nr. 6(86). – P. 85–88.

Triple Play“ – tai paslaugų komplektas, susidedantis iš trijų paslaugų: duomenų perdavimo, IP telefonijos ir interaktyviosios televizijos. Kiekvienai iš šių paslaugų būtina užtikrinti reikiamus QoS parametrus perduodant tuo pačiu metu. Straipsnyje pateikiamas modifikuotas eilių aptarnavimo maršrutizatoriuose M/G/1/K – AFQ modelis, pagrįstas virtualia eile bei svoriniu eilių aptarnavimo koeficientu, kintančiu priklausomai nuo mazge esančių paketų būklės įvertinimo ir leidžiantis valdyti paketų vėlinimą tinkle. Pateikiami modeliavimo rezultatai. Il. 7, bibl. 6 (anglų kalba; santraukos anglų, rusų ir lietuvių k.).