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Frame-Size Dependency Using Priority Service Efficiency in FR Networks

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Abstract: FR is a packet-orientated technology (with the packets of various lengths); it is impossible to give any warranty with regard to delay values, which are critical during an audio information transfer. In event of any node overfilling or fault, the packets containing the audio information can be redirected (in case of a node it can be done using technology of the commutable permanent virtual channel) that can lead to delay increase. Such delays can be escaped only by means of the traffic effective control and the entire network reliability increase. As the FR protocol assumes the various lengths of packets, the data packets buffering in the FR switches also can cause increase in delay values scatter (the frames with audio information have to wait until the longer frames are transferred to communication channel). It is possible to eliminate delay in the switches by means of the priorities application to the frames and the longer packets defragmentation. The both above mentioned methods require standardized technologies. One more potential delay source is a time necessary to fill the frame with the audio information. In order to achieve efficient use of the entire pass band and to escape the transmission of frames, which practically do not contain information, all the frames should contain information. Determined by duration of the frame filling process, delay is calculated as T = L/R (T is the time necessary to fill the frame by information; L is the frame length; R is the data receive rate). Delay exceeding 200 ms becomes appreciable and annoying to user. This problem can be solved by means of the frame length contraction[1].

Key words: Frame, relay, priority service, packets

INTRODUCTION

Frame relay is a high-speed connection-oriented standard for interface with switched packet bearer services. Frame relay, a WAN service, evolved from X.25 and ISDN, takes advantage of a more reliable, high-bandwidth infrastructure and widespread availability of intelligent customer-premise equipment (CPE). Frame relay uses statistical multiplexing, allocates bandwidth on demand up to 45-50 Mps with minimal error checking and control. Bandwidth on demand accommodates typically bursty corporate traffic.

Voice over Frame Relay, although quality of service is uneven, is rapidly gathering popularity as it reduces long-distance phone costs. Frame relay, available for national and international transmission, relies on international and proprietary standards, is non-tariffed and allows many possible configurations. It may encapsulate other protocols and may be carried by ATM, its main competitor. Frame relay's overall outstanding performance is tampered by problems in local loop access and by its very success, which introduces network congestion and delays.The research is to introduce an enhanced algorithm using priority service of short frames and the control frames.

Our approach: To eliminate the above-mentioned problems, it is suggested to apply the priority service of the short frames (the frames containing the audio information and the control frames). To analyze the suggested solution adaptability, the simulation with the device CMO application has been carried out. The main parameters being controlled during the simulation process are the time of the frame residence in the node, amount of transmitted audio information or any other "short" information within 20 sec. at the transfer intensity of 0.7 or 0.8 at the various rates of the output node line and also with the variable band width granted to transfer the audio information. The present investigation also focuses on determination of the "short" frame optimal size during the application of the frames priority service in the node[2].

The calculation of delays in the communication networks can be done using the queuing network model suggested in[3] which received widespread occurrence[4].

As the mathematical model while estimating time parameters of the considered networks in the article[5] was

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assumed the appropriateness of applying the queuing systems with the requests multivariable flow, Poisson's input flow and exponential law of the service time distribution. Every node is represented as the queuing systems with the multivariable flow of requests. Maximum value of delay is determined between the subscribers systems located on the boundary tree branches. At similar parameters of the nodes, this value equals the product of delay in one node and double value of the tree levels number.

In general case, during the priority-free requests flow, the average waiting time E(W) is determined using the following formula:

$$E(W) = \frac{\sum_{i=1}^{M} \lambda_i X_i^2}{2(1-\rho_1-\rho_2-\ldots-\rho_M)}$$

where λ_i is the i-input flow intensity, M is the number of input sources, X_i^2 is the second initial instance of the i-type requests service duration. For the service exponential law: $X_i^2 = 2X_i^2 = 1/\mu_i^2$, respectively,

$$E(W) = \frac{\sum_{i=1}^{M} \lambda_i X_i^2}{(1-\rho_1-\rho_2-\ldots-\rho_M)}$$

Let us consider the node mathematical model with two priority levels. In general case, n input channels are connected to the node that can bear n_1 priority requests and n_2 priority free ones with the sum of n requests. In this event, the node can be represented as the multi-channel queuing system with n inputs, when the input n_1 is used to receive the priority flows and the input n_2 – priority-free flows. In our case, this system can be turned into two-channel queuing system. Let us consider this subject using the example with Poisson's input flows and the exponential law of the service time. Remember that such system is reasonably sufficient for the network processes description.

Let us examine the possibility of the input flows group with the same priorities replacement by one flow with the total intensity. For this purpose, the average values of waiting time for two cases should be equated.

In case of the multi-channel equal-priority system, the average waiting time is determined using the formula:

$$E(W) = \frac{\sum_{i=1}^{n} \lambda_i}{\mu(\mu - \sum_{i=1}^{n} \lambda_i)}$$

where λ_i is the i-input flow intensity, μ is the intensity of the requests service by the system.

Respectively, in case of the single-channel system the average waiting time is determined by the following:

$$E(W) = \frac{1}{(\mu-\lambda)} - \frac{i}{\mu}$$

or

$$E(W) = \frac{\lambda}{\mu(\mu-\lambda)}$$

where $\lambda = \sum_{i=1}^{n} \lambda_i$

After equating of the second members of equations for the average waiting time in these two cases, the next expression is obtained:

$$\frac{\sum_{i=1}^{n} \lambda_i}{\mu(\mu - \sum_{i=1}^{n} \lambda_i)} = \frac{\lambda}{\mu(\mu-\lambda)}$$

The last equation is true in case $\lambda = \sum_{i=1}^{n} \lambda_i$

So the average waiting time for the requests multivariable flow can be replaced by those for the requests single-flow in the system with the total intensity. Reasoning by analogy relative to the second ensemble of the priority requests and taking into account that within the bounds of this ensemble the flows are equitable, it is possible to perform replacement of the multi-channel system by two-channel one. In such a way, all the further reasoning will be executed concerning two-channel system.

The requests with the same priorities are located in one queue and served in the order of the roll-call polling.

In this case, the pool time can be ignored since the requests reception from different sources is considered as those from single source with the total intensity. At node level, the network can be represented by way of the queuing system with two input flows.

In this case, the following expressions of the average waiting time $E(W)_1$ for the flow S_1 of the highest priority and the average waiting time $E(W)_2$ for the flow S_2 of the lowest priority are obtained:

$$E(W)_1 = \frac{\lambda_1+\lambda_2}{\mu(\mu-\lambda_1)}$$

and

$$E(W)_2 = \frac{\lambda_1+\lambda_2}{\mu(\mu-\lambda_1)(\mu-\lambda_1-\lambda_2)}$$

These values are considered as the components of the corporate network' total load. For an operation comparison, let's take the average value of the waiting time E(W) in event of two priority-free flows:

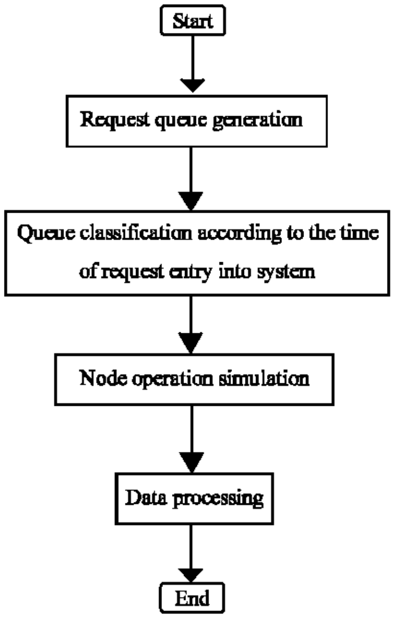


Fig. 1: The simulation algorithm flowchart

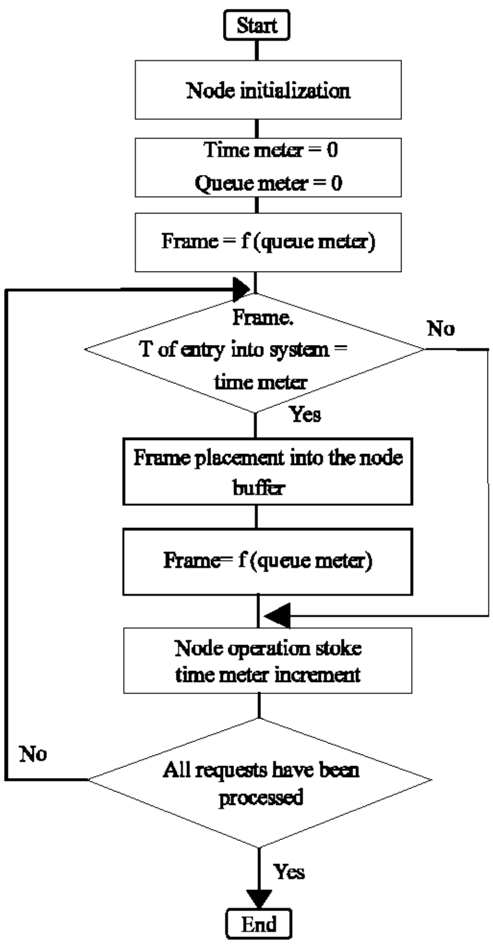


Fig. 2: Algorithm of node operation simulation

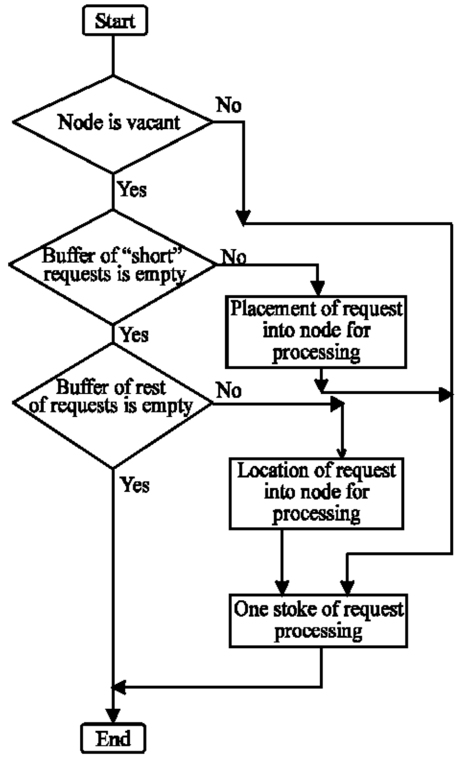


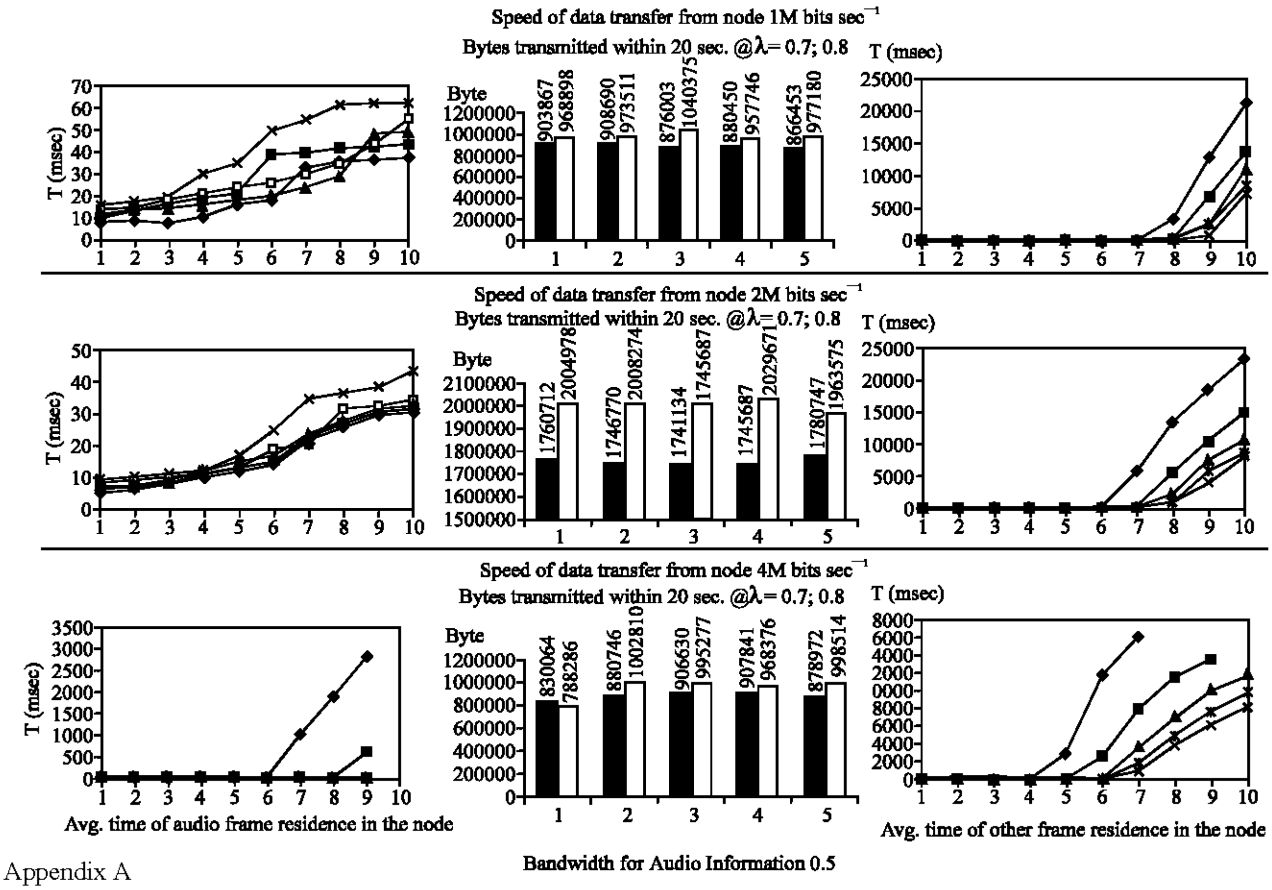
Fig. 3: Algorithm of one stroke of the node operation

$$E(W) = \frac{\lambda_1 + \lambda_2}{\mu(\mu - \lambda_1 - \lambda_2)}$$

The main point is as follows. In communication network with the messages (or packets) switching, there are 1, …, *N* nodes and 1, …, *W* communication channels, which are interpreted as the queuing systems *ÌÌ/1*. Both the communication channels and the nodes are perfectly reliable. The nodes execute the operations concerning the messages switching, including their editing, routing, buffering, etc. It is assumed that processing time in the node is permanent and equals pair of milliseconds according to statistics. There are the queues to the communication channels. While transmitting messages, some delays appear. Into the nodes there is transferred Poisson's requests flow, which can be determined as the flow between each nodes pair of network with the average intensity λ_{ij}, i,j = 1, ..., N, i≠j. The total flow intensity is calculated as:

$$x = \sum_{i=1}^{N} \sum_{j=1}^{N} \lambda ij$$

There are two types of requests: short requests and ordinary ones. Ordinary frames are limited by the size of 4096 bytes, short frames – 1500 bytes. To locate these messages in the network mode, the storage of unlimited



Appendix A

capacity is designed. In compliance with the problem specification, it is enough to simulate the operation of one mode. There are five branches; each one corresponds to different average length of the "short" frame (500 ± 300 bytes; 750±300 bytes; 1000±300 bytes; 12500±300 bytes; 1500±300 bytes)[6]. Average sizes of the audio frame applied during simulating are determined by different parameters of the circuits, which compress the audio information, the spread in values of ±300 bytes occurs due to time variation of the flux density of audio information.

In terms of simulating program, the average time of the frame (request) in the mode is calculated as:

$$T = \frac{\sum_{i=1}^{N} (F_i^{T\_start} - F_i^{T\_finish})}{N}$$

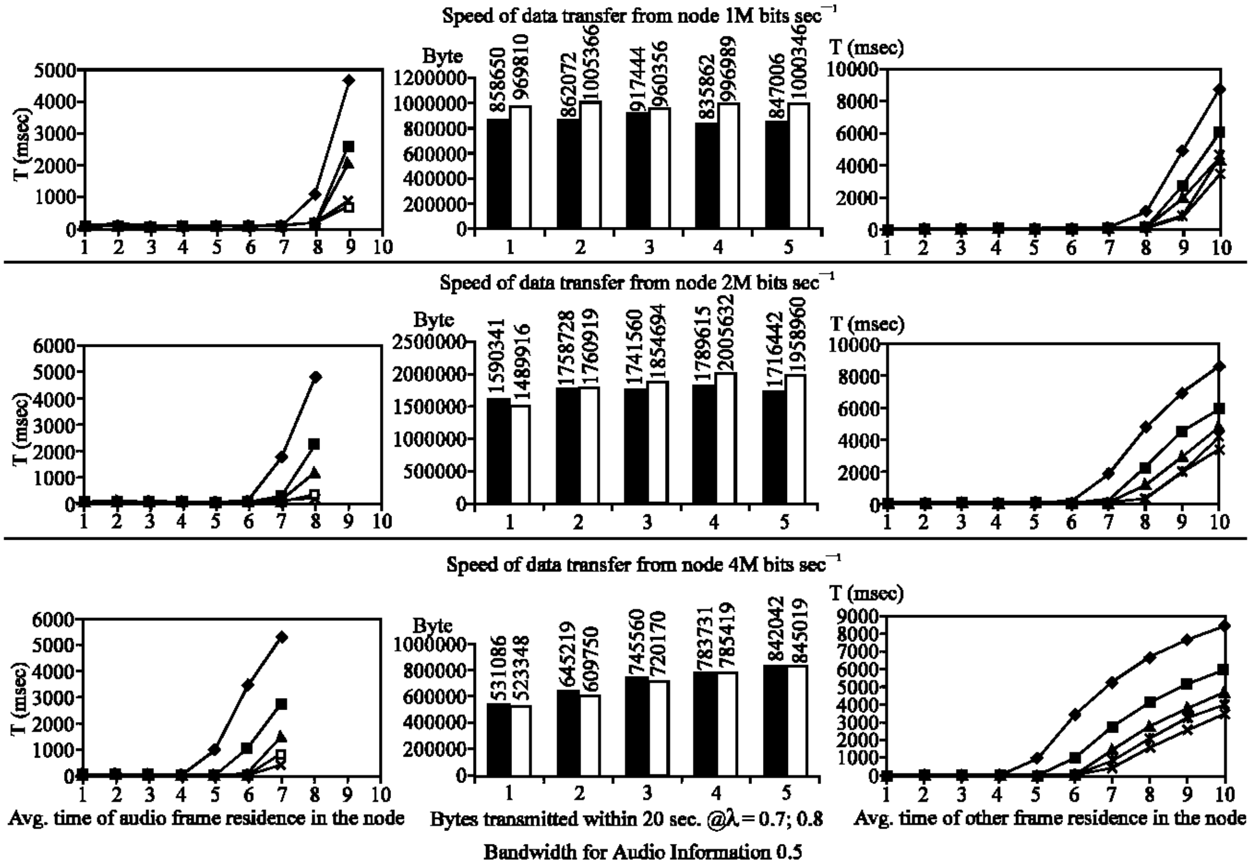
The "short" information content transmitted during 20 or 5 sec. (depending on the simulated speed and band width) is determined in the following way:

$$v = \begin{cases} \sum_i F_i^{Length}, \text{ if } F \in T_\lambda \text{ and} (\lambda = 0.7 \text{ or } \lambda = 0.8) \\ 0 \quad \text{if } F.Time \notin T_\lambda \text{ or } \lambda \neq 0.7 \text{ or } \lambda \neq 0.8 \end{cases}$$

The simulation algorithm chart and applying the priority service procedure to the frames being in queue. (Fig. 1, 2 and 3).

Experimental results: As a result of the node operation simulation, there have been obtained data allowing us to make the following conclusions:

At the band width designated for the audio information in share of 0.25 with respect to the entire pass band at the priority-free service, the value of the audio frame delay in the node sharply increases beginning from the intensity value of 0.6 - 0.7 and this factor leads to impossibility of tolerable audio information transfer in the network. While applying the service procedure with the relative priority of the "short" frames, the average time of the "short" frame residence in the node significantly decreases. This achievement allows transmitting the significantly larger information content and ensures transmission that is more reliable (as the probability of the frame extracting from the network due to the buffer overfilling and the traffic exceeding considerably decreases). Those findings are clearly stated in appendixes A and B.



Appendix B:

Research of the audio information transfer in the FR network with the help of the frames with different lengths allows making the following conclusions: in case of the priority-free service procedure at the flow intensity growth and the band width growth, the time of delay increases more rapidly in event of frames with the smaller size. However, while applying the service procedure with the relative priority the average time of the frame delay in the node increases equally slowly in comparison with the previous case and the service time of the longer frames slightly exceeds those of the shorter frames, but at the "short" frame size growth, some increase in the information content transmitted per time unit is observed. Thus, there is no single-valued answer relative to optimal size of the frame for the audio information transfer used. The audio frame size depends on the applied audio information compression circuit – in case of higher compression, the frame size is smaller, but at growth of the band width designated for information transfer, the sharp increase in the average time of residence in the node starts just for the shortest frames, as the respective value for the longer (1250, 1500 bytes) frames still remains in the tolerable zone.

As to the rest of traffic combined with the audio traffic, it is necessary to note that application of the service procedure with relative higher priority of the "short" frames does not essentially influence the average time of delay for the frames different from the "short" ones, if the bandwidth designed for audio information equals to 0.25. It is necessary to note that the average delay time of the frames in the node increases more rapidly if the shorter frames are used in combined traffic.

At growth of the band width designated for the audio information at the communication lines of the higher speeds, it is necessary to admit the sharp increase in the average time of delay of the shorter frames and in this case there, is observed the sharp increase in the average time of delay for the combined traffic frames, beginning from the input flow intensity value equal to 0.4.

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