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Evaluating the influence of multiplexing schemes and buffer implementation on perceived VoIP conversation quality ☆

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ABSTRACT

This work presents a study of RTP multiplexing schemes, which are compared with the normal use of RTP, in terms of experienced quality. Bandwidth saving, latency and packet loss for different options are studied, and some tests of Voice over IP (VoIP) traffic are carried out in order to compare the quality obtained using different implementations of the router buffer. Voice quality is calculated using ITU *R*-factor, which is a widely accepted quality estimator. The tests show the bandwidth savings of multiplexing, and also the importance of packet size for certain buffers, as latency and packet loss may be affected. The customer's experience improvement is measured, showing that the use of multiplexing can be interesting in some scenarios, like an enterprise with different offices connected via the Internet. The system is also tested using different numbers of samples per packet, and the distribution of the flows into different tunnels is found to be an important factor in order to achieve an optimal perceived quality for each kind of buffer. Grouping all the flows into a single tunnel will not always be the best solution, as the increase of the number of flows does not improve bandwidth efficiency indefinitely. If the buffer penalizes big packets, it will be better to group the flows into a number of tunnels. The router processing capacity has to be taken into account too, as the limit of packets per second it can manage must not be exceeded. The obtained results show that multiplexing is a good way to improve customer's experience of VoIP in scenarios where many RTP flows share the same path.

1. Introduction

The use of the Internet for multimedia transmission is growing as bandwidth increases. Many of these new services, like Voice over IP (VoIP), videoconferencing, online gaming, etc. have very stringent real-time requirements, so network impairments may affect the interactivity of the service. For example, IP telephony customers expect the service to have the same interactivity as traditional telephony. As the use of IP telephony is growing, and best-effort networks without real-time delivery guarantees are often used, there is a concern regarding the quality perceived by the users of these services.

RTP is the most used protocol for real-time media transport. It has many profiles, and it is able to carry voice with different codecs, video and other real-time services. Due to real-time requirements, multimedia information has to be fragmented into small pieces of information, which are

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then transported into RTP packets using a small period. This fact implies that the overhead can be significant if the information carried by a packet is only a few tens of bytes, decreasing bandwidth efficiency. For example, a voice codec like G.729a can significantly compress information, generating a 10-byte sample every 10 ms. Thus, if two voice samples are included into an RTP packet, it will have 20 bytes of information, plus 40 bytes corresponding to IPv4/UDP/RTP headers. As a result, only one third of the bytes will carry voice information. Of course, if IPv6 is used, the efficiency becomes even worse.

There exist certain scenarios in which many RTP flows share the same path (Fig. 1): for example, a number of computers of the same office may use a PBX located at the data center of an enterprise; or different hosts of two offices of a SME (Small and Medium Enterprise) can establish simultaneous calls from one to the other. If this path includes the access network, which is normally a bottleneck, the deployment of solutions to reduce this overhead can be interesting. Two of them are header compressing, and grouping more samples into a single packet.

With regard to header compression, some schemes have been proposed, as we will see. They use a "context" shared by the sender and the receiver, which includes the protocol fields that are the same on every packet. As different flows can share the same origin and destination, each compressed packet has to include a Context Identifier (CID). The protocol also uses delta compression for the fields that increase from one packet to the next. Logically, this compression has to be applied in a hop-by-hop way.

Overhead can also be avoided by placing multiple samples into one packet [3], so as to increase the number of samples that share the same header. This can be achieved by bundling more voice samples of the same flow into a single packet (Fig. 2a), but this has a counterpart: each added sample will increase the packetization delay in the sender. There also exists the possibility of multiplexing samples of different conversations into the same RTP packet (Fig. 2b). This solution sends the same number of samples with the same frequency, so packetization delay is not increased. But it may add other delays which have to be studied.

RTP multiplexing combines these two techniques: header compressing, and bundling multiple samples into the same packet, but it has some disadvantages, i.e. new delays and processing charge. Multiplexing reduces the

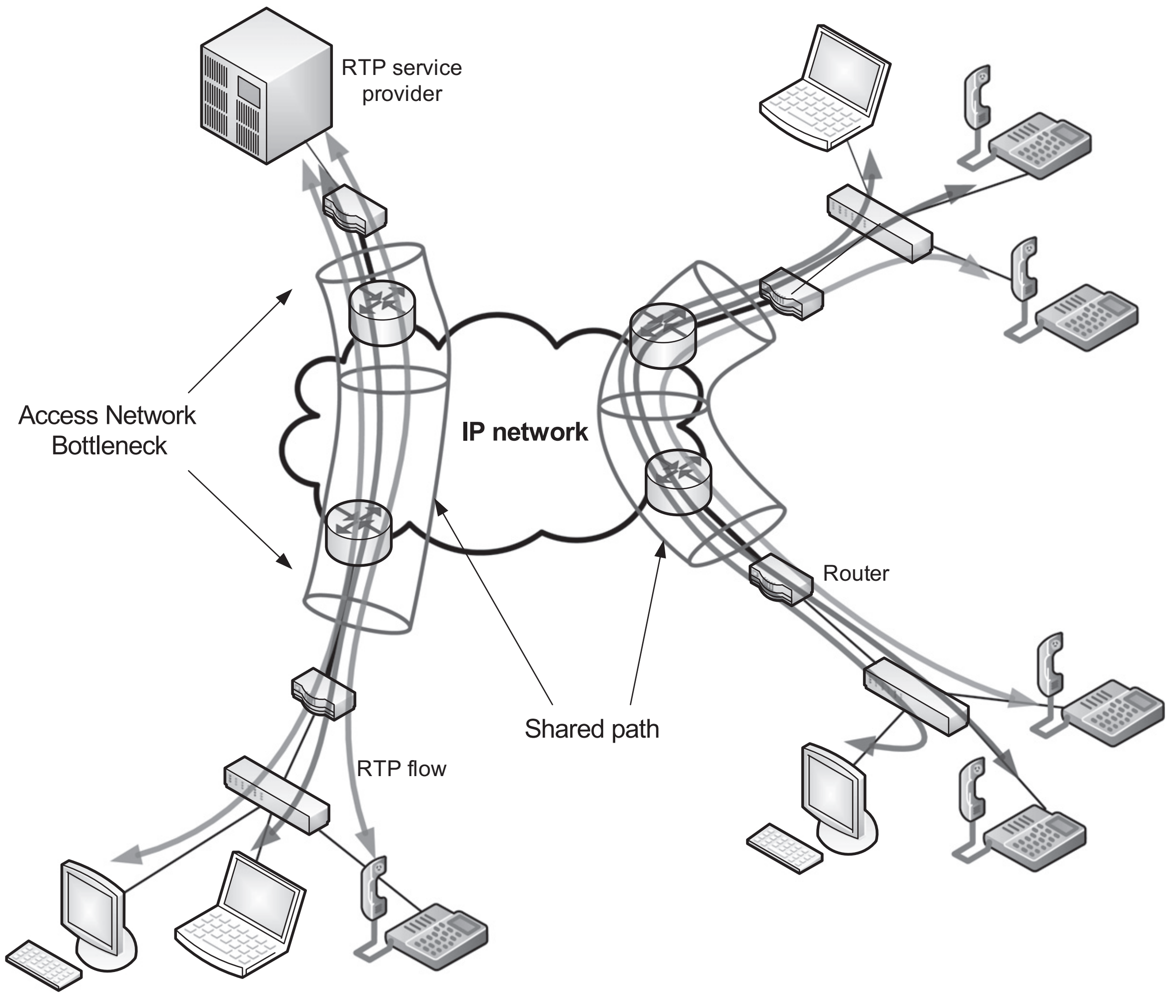

Fig. 1. RTP flows sharing the same path.