A NEW QUALITY OF SERVICE CONTROL SCHEME FOR VOIP NETWORKS

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Key Results: A novel method for user perceived voice quality of service control, experimental results of the QoS control scheme in a real VoIP network.

How does the work advance the state-of-the-art?: The QoS control scheme makes use of objective measure of voice quality (based on the latest ITU algorithm for perceived voice quality algorithm) as well as a measure of network quality which takes into account the effects of key impairment parameters (delay, jitter, loss etc). This approach provides a superior control scheme compared to existing methods. The new scheme has implications for current best effort networks and for the design and use of emerging and future generation of IP networks.

Motivation (problems addressed): Perceived voice quality is widely recognised as a key QoS metric in VoIP applications, but existing control methods for best effort networks do not consider user perceived quality. In this project the goal is to develop an optimum QoS strategy that takes into account user perceived quality as well as network quality.

1. Introduction

IP networks are on a steep slope of innovation that will make them the long-term carriers of different types of traffic including speech, but they are not designed for real-time communication because of their variable delay and loss characteristics [1,2].

Voice over Internet Protocol (VoIP) is a new solution for the integration of existing telecommunication network and the rapidly expanding IP networks [3,4]. Due to the "Best effort" nature of the IP network, Quality of Service (QoS) has become an important issue in this field. At present, QoS control mechanisms are used to minimise the effects of network impairments on voice quality, but these do not take into account quality as perceived by the end-user [5,6]. User perceived speech quality is now widely accepted as the key QoS metric in voice communication over IP networks [3,7-10].

There is a need to develop a novel and robust strategy for the design of future QoS control mechanisms for voice service that takes into account user-perceived quality. Potentially, such a method will make it possible to achieve optimum QoS for voice communication, in terms of user perceived speech quality and resource utilisation. In this paper, we proposed a novel method to improve the over all voice quality in the IP network.

2. VoIP QoS measurement and control

A QoS control strategy that takes into account user perceived quality requires the ability to measure objectively the impact of network impairments on voice quality. In this project we have used the latest ITU-standards to measure voice quality. VoIP QoS control is different from QoS control in networking because of the subjective nature of voice quality. In VoIP, control is more application layer or user layer focused. The main goal of the VoIP QoS control is to adapt application/user layer parameters to the network or lower layers and then to control the network to delivery good voice quality.

In the proposed new scheme, the control method is separated into two aspects. The end-to-end QoS control method can alter the stream behaviour according to user perceived voice quality but have no access to other streams. This is more like application layer sender behaviour control. Another aspect is network layer QoS control. The network equipment has access to all packets streams but cannot control the sender behaviour due to the limited function. In the new approach, the network nodes have access to all sender behaviour, and so can adjust the network quality based on user perceived quality.

A scenario for the control method is as follows. Different groups of users with different Service Level Agreements (SLA) are connected to a network node. The users use bit rate variable codec to transfer speech. The node has limited network resources and the connected users have degraded voice caused by congestion in the network node. Different groups of Service Level Agreement are treated differently. If a group with a higher SLA has a poor user perceived voice quality (measured by ITU standard method), then the bit rates for all users are changed to a lower one. On the other hand, if a group with a lower SLA has poor voice quality, then only their bit rate is changed to a lower rate. When the network quality significantly improves (e.g. due to a decrease in the number of users), then let the group with higher SLA change to higher bit rate first. Thus, different group of users have different level of services. The higher-level user should have better services than the lower lever user and the lower level user should have better services than the non-controlled situation (fixed highest rate situation).

3. Simulation

Computer simulation of the proposed method has been undertaken. The basic structure of the simulation is as follows. Multi-user congestion-point (a base station or access point) with a 2M-bandwidth resource to next hop in the network is connected to users that are sharing the limited resource. The users use AMR codecs and this allows control of speech bit rate. The users' voice quality parameters (packet loss rate and AMR bit rate at this stage) are collected from the network node and the ITU algorithms are used to calculate the user perceived voice quality. The result of the user perceived voice quality is then used to change the senders' behaviour (e.g. AMR bit rate at this stage).

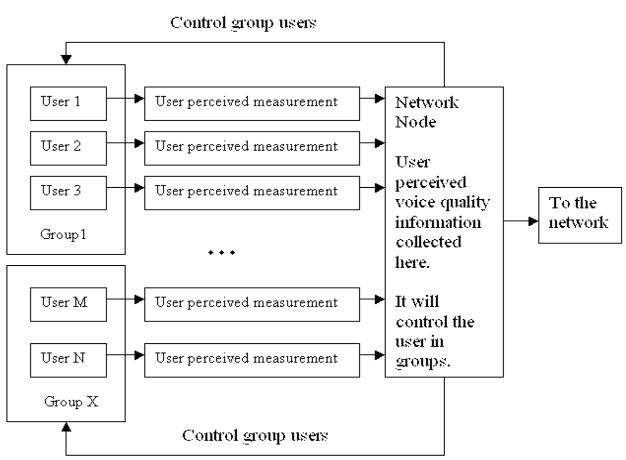

Fig.1. Network node control users in groups instead of as individuals

Results of the simulation study will be presented at the conference.

4. The test bed

The novel control method for VoIP QoS is being validated by emulation and tests in a real network situation. We have built a VoIP test bed to support the VoIP QoS control research. The first stage is to have basic VoIP connections via some industry standard equipment and latest test equipment. VoIP calls are generated in the test bed, recorded and the traffic captured for further investigation. The structure of the basic test bed is depicted in Figure 2.

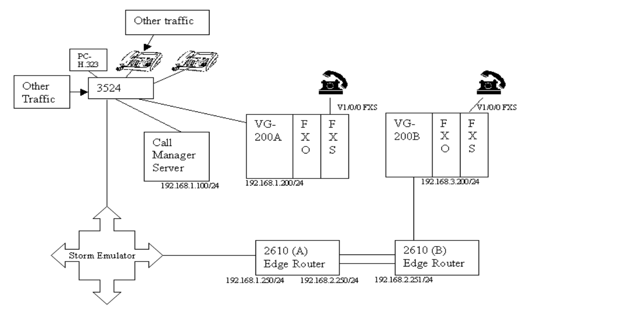

Fig. 2 The VoIP QoS control test bed

5. Further work

A further investigation involving a more extensive computer simulation and tests/emulation in the real network environment is planned.

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