Voice over IP Wide Area Quality Measurements

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Abstract. Users of Internet telephony applications demand good quality audio playback. This quality is largely dependent on the instantaneous network conditions as well as both the time and day. Equally, if not more important, is the reliability of using the Internet for voice communication. Calls that cannot be completed are unacceptable for all users. In this paper we describe a scheme for measuring network conditions with regard to Voice over IP. Our tests include a wide range of geographically distributed sites. A second goal was to create a measurement architecture that can be used to monitor long term characteristics of Internet links. The measurements will be plotted on Web pages to give near real-time feedback to operators *and* users of IP Telephony.

1 Problem Statement and Related Work

The traditional telephone network is carefully monitored for reliability and quality. Since Voice over IP users are more susceptible to delay and loss when using audio over packet switched networks, a wide area monitoring scheme is sensible for the Internet. The evaluation of a monitoring system comparable to the traditional telephony solution, is therefore the contribution of this paper. Making accurate measurements of voice communication when using the Internet is challenging. Call durations are not well known when there is no payment structure, sessions could exist indefinitely, the exact signalling is difficult to predict, as calls may be forwarded from server to server. There is no restriction on the voice coding schemes available, for example PCM or GSM or other standards can be used, hence a different load is created for effectively the same conversational characteristics. In the traditional telephone network these parameters are either fixed or are well known.
Measuring quality and reliability of packet audio on the Internet has attracted surprisingly little research. Bolot et. al in 1995 measured packet audio loss over the Internet [1]. They concluded that the number of consecutive losses is quite low, usually between one and five at 8am and between one and ten at 4pm. Maxemchuk, in 1997, measured the loss and delay variation for intra-state connections, within the USA and across international links [5]. Their conclusion was the quality depends on the length of the connection and the time of day. Dong Lin drew similar conclusions in [4], stating that in fact even calls within the USA could suffer

large jitter delays. Her results on packet loss also agree with those in [1], which is interesting, as the measurements were taken four years later. Our work with respect to quality measurements is similar, except we consider more complex cases. Also we include reliability measurements, which to our knowledge, has not been published for Voice over IP.

2 Sending Test Calls Between Sites

Our technique is to send pre-recorded calls between different sites on the Internet. Through modification of a VoIP tool, Sics*o*phone [3], we will gather information about loss, jitter, delay and the number of completed calls. The percentage of completed calls is our definition of reliability. Regarding the quality of calls, this work is a continuation of the measurement architecture presented in 1999 [2]. In order to gather comprehensive measurement data, we have carefully chosen ten sites, distanced by several kilometres to over twelve thousand kilometres from each other. Compared to our first work we have enhanced the measuring and monitoring facilities considerably. More specifically we have fully automated the process, hence we can send calls at any time granularity (per minute, hourly etc.). We have also included full bidirectional conversations, different duration calls, i.e. short. average and long. Whereas in the first phase we set up calls between a central site and the satellites, this time we will use a full mesh scheme, so each co-operating site can send and receive to all others. Call signalling has been added, however we use a straightforward strategy, and assume the callee is reachable via a single server access. Plots with the current quality in terms of loss, delay, jitter and reliability will be presented in real time as continuely updated Web pages. Finally since our first work concluded that Internet Telephony was feasible for most sites three years ago, an interesting result is if the same conclusion is valid.

3 Expected Results

The full version of this paper will include:

- Real time plots of the current quality and reliability
- Is VoIP still feasible in 2002 ?
- A description of the measuring and monitoring tool

References

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