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CS 443 – Paper Summary

Title: *Quantifying Skype User Satisfaction*
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Summary: Quantifying user satisfaction of VoIP services such as Skype are challenging for various reasons – one example being that Skype calls are tricky to identify because there is no easy access to the end-to-end voice signals in a peer-to-peer network. The authors propose a quantification of Skype user satisfaction based on a rigorous analysis of the call duration from actual Skype traces.

Key Ideas

Skype, being a peer-to-peer application, does not allow one node to see traffic between any two Skype hosts. However, two Skype hosts can communicate via a relay node if they have difficulties establishing sessions. A node is more likely to be a relay node if it is a powerful machine and has been set up for a length of time. The authors set up a powerful Linux based machine to act as a relay node to collect traces between two Skype hosts. They then identify Skype calls by monitoring the dynamic port numbers used by the end hosts (keeping all <host, port> pairs in a table and monitoring their traffic). The authors collect measurements from calls that are active. Calls that are active are defined by length of flow, average packet rate and average packet size. The measurements the authors chose to collect through various non-intrusive methods were: call duration, bit rate, jitter and roundtrip times. The authors concluded that the degrees of user dissatisfaction caused by bit rate, jitter and roundtrip time are in the proportion of 46%:53%:1% which indicates that increasing the bit rate whenever appropriate would greatly enhance user satisfaction. Also, the choice of relay node should focus on network conditions – the level of congestion – rather than rely on network latency. The authors validate their choice of parameters by using speech recordings and inferring from speech patterns (responsiveness, response delay and talk burst length). The results of these validation tests proved that the authors were right in using call duration as a measure for user satisfaction.

Flaws

Too many assumptions. The authors *assume* that the collected traces are all from Skype calls. They then *assume* when users are talking and when they are silent (only background noise). They *assume* when users have to repeat themselves due to poor quality of voice transmission. They *assume* that they can infer conversation patterns by studying packet sizes – although, inferring a conversation pattern and then drawing conclusions from it are quite hard to quantify. As a user of Skype, I do not believe that their methods are sound in determining user satisfaction (especially the speech pattern validation tests). It is a very hard problem due to its complexities – therefore, the authors' biggest contributions were defining the challenges and making a first attempt to overcome them.

Relevance and Future Work

VoIP services are becoming increasingly popular. There are 200 million Skype downloads and approximately 85 million users worldwide. In order for application developers to understand whether Skype is providing a good enough voice phone service or if there is still room from improvement, quantifying Skype user satisfaction is a valuable research undertaking.

Now that the challenges are well-defined by this paper, many different parameters, algorithms and statistical tools may be explored as a means for quantifying user satisfaction. To continue the current proposed algorithm, more extensive studies can also be done using the authors' proposed method by setting up more than one relay node.