Ns2Voip++, an enhanced module for VoIP simulations

(Poster Abstract)

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ABSTRACT
In the last ten years, many circuit-switched networks for voice have been replaced with packet-switched ones. Hence, simulating Voice over IP has become of paramount importance in assessing the performance of a network. However, a sound performance analysis should be carried out in conditions which are as close as possible to a real deployment. In this paper we present enhancements to ns2voip 6, a module for simulating realistic VoIP traffic with the ns2 simulator. In detail, we add new features, i.e.. a *correlated* model for packet generation in a two-way conversation and implementing a set of realistic playout buffers to simulate the behavior of the receiver. Our code is available at http://cng1.iet.unipi.it/wiki/index.php/Ns2voip.

Categories and Subject Descriptors
G.3 [**Mathematics of Computing**]: Probability and Statistics – s*tatistical software*. I.6.7 [**Computing Methodologies**]: Simulation Support Systems – *environments*.

General Terms
Measurement, Performance, Experimentation, Verification.

Keywords
Simulation, ns-2, VoIP, MOS, QoS.

1. INTRODUCTION
Voice over IP has become a viable alternative to the traditional circuit switched networks, both in telephone networks and over the Internet 6. It is well known that the performances of this real-time traffic heavily depends on the QoS offered by the network. As such, simulations are widely used to asses its performance to ensure in a pre-deployment stage. For this reason, offsetting up meaningful and lifelike simulation scenarios is of paramount importance to get results which the deployment of new networks can rely upon. We present enhancements to the popular ns2voip 6, a module for VoIP traffic generation. This module is developed for the ns2 simulator 6 one of the most popular simulation tools for the networking community. In its first release ns2voip allows to simulate a voice application supporting: different codecs, Voice Activity Detection (VAD), and aggregation of multiple voice frames into the same IP packet. Online collections of the data that are used to evaluate the performance is done through ns2measure 6. Among them, one of the most significant is the *Mean Opinion Score* (MOS) 6 which quantifies the quality of the conversation perceived by the end user. The two main limitations of this first release are i) the traffic generation model and ii) the playout buffers. Traffic generation at the source was driven only by the codec model and the state of VAD at the source (*on* or *off*), without considering the status of the receiver. Uncorrelated packets generation between the two sides does not consider the interplay between silence and activity periods of the two applications: most of the times the two parties will take turns in talking and listening, with relatively short transitions periods of cross-talking or mutual silence The traffic generated overlooking this interplay can be considerably different from a real conversation. The second enhancement is the implementation of playout buffers. In the first release, only an *optimal*, non causal playout buffer was implemented, which cannot be implemented in any real system. As this is a critical component for VoIP performance 6 evaluation would always yield overrated results. In our revised version, several *real* playout buffers were added, e.g. h323 6, which is used in several SIP clients (.e.g, Ekiga 6). Hereafter we briefly describe the module focusing on the enhancements introduced in this new major revision, and we illustrate some simulation results. The latter show how more reliable results can be harvested with the new version of the module/.

SIMUTools 2010 March 15–19, Torremolinos, Malaga, Spain.
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2. DESCRIPTION OF THE MODULE
The structure of the VoIP module is illustrated in Figure 1.

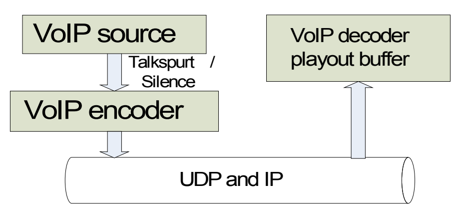
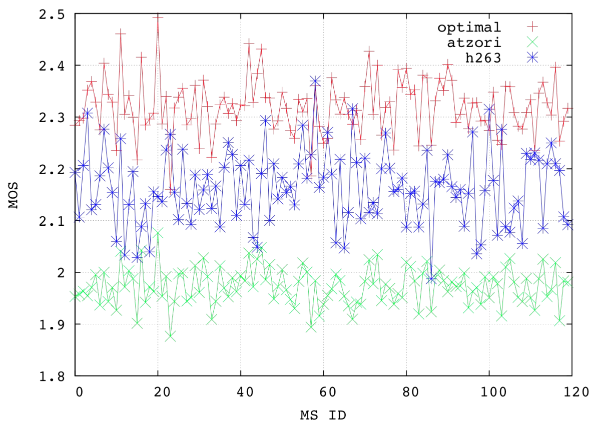
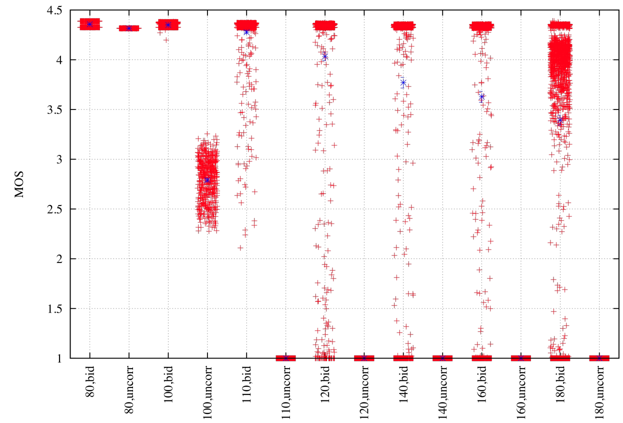


Figure 1. VoIP module structure

A *VoIP source* models the user activity, deciding whether a user is in a *talkspurt* or in a *silence period*. In the former case, the source signals to the encoder the beginning and the (randomly selected) talkspurt duration. During a talkspurt, the encoder generates data according to the chosen codec. The latter defines the size of each packet and the frequency of their generation. Many popular VoIP codecs are implemented (e.g., G.711 and GSM.AMR), and more can be added at negligible implementation cost. At the receiver side, a VoIP *decoder* receives packets from





the network and employs a playout buffer to pace their playout. the MOS, as well as other performance metrics, are evaluated at the decoder. In the following, we describe in details the two enhancements of this new version.

2.1 VoIP correlated traffic

In order to model the Voice Activity Detection (VAD) we implement the model described in6. In this model the whole conversation is simulated through an eight-state Markov chain in which each state represents one of the following situation: mutual silence, in which both speakers are silent, active talking periods: either only one side of the conversation is active or the two parties are talking at the same time.

2.2 Playout buffers

The original release included only the *optimal* non-causal playout buffer. In this new release we add more realistic and well-known playout buffers, i.e.: h323 6 and eEM (causal) 6.The former is well known dynamic playout buffer used in a wide variety of well-knows softwares such as the open source VoIP client Ekiga.

We imported the real H323 codec taken from the Ekiga application source code into the ns2voip framework, and made it capable of handling ns2voip talk frames and collecting ns2measure statistics.

The latter implemented playout buffer works just like the Optmal one, so it is capable of computing on a given talkspurt the best playout delay which leads to the best achievable MOS for that talkspurt. However, instead of being non-causal as the Optimal playout buffer, it waits for a whole talkspurt to arrive at the receiver, and *then* it exhaustively computes what would have been the best possible playout instant for that talkspurt, i.e. the one that would maximize its MOS. This computed delay is used as playout delay for the next to come talkspurt, by supposing variation in best playout delays of two consecutive talkspurts is small enaugh to consider them as having the same value. This playout buffer obviously can also be seen as the causal version of the optimal one.

3. SIMULATION RESULTS

In order to demonstrate how a playout buffer influences the perceived performance we simulated a wireless networks with an access point, scheduling downlink traffic according to the Earliest Due Date (EDD) policy. Figure 2, illustrates the average MOS versus the mobile station ID. As expected, the optimal playout buffer offers the highest performance while the causal playout buffer provides the worst. The results show that the difference between these two solutions is of 20% on average, which is a relevant figure. Figure 3 shows differences in evaluated MOS for different set of users making use of uncorrelated and correlated VoIP streams. In the latter case the model takes into account mutual silences states and avoids mutual talk states. Talk activity of correlated flows is lower than uncorrelated ones, which leads to different and more accurate simulation results.

4. CONCLUSIONS

In this work we present a new revision for the ns2VoIP software package. A new set of playout buffers and a correlated traffic source model has been implemented in order to enhance the simulations reliability.

5. ACKNOWLEDGMENTS

The authors would like to thank Luca Niccolini for his valuable help in implementing the new VoIP source model.

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