

Test and Measurement of Voice over IP Technologies

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Abstract:

The transmission of voice communication over Internet Protocol networks, or “Voice over IP” (VoIP), is the next great challenge of Internet technology. If scalable and efficient technologies can be developed to support telephony over the Internet Protocol Suite (IPS), then the holy grail of “integrated networks” can be finally achieved, but in the opposite way than was commonly predicted just a few years ago (i.e., the data networks will evolve to support voice).

There is tremendous interest and activity in the fledgling VoIP market that has emerged within the last year. Large consumers see the potential for VoIP to yield great benefits including reduced costs, simplification of the underlying infrastructure, consolidation, and the opportunity to implement advanced multimedia and multiservice applications. End-user demand is predicted to grow rapidly over the next five years: marketing and consulting company Frost & Sullivan estimated the compound annual growth rate for IP-enabled telephone equipment to be 132% between 1997 and 2002. By the year 2000 up to 70% of Fortune 1000 companies are expected to use VoIP technologyⁱ. The Internet and telephony industry is responding to this projected market with aggressive VoIP research and development efforts by router and switch vendors, Internet Service Providers, and local / long distance telephony carriers. This proposal provides the basis for building ITL competency in VoIP test and measurement and defines initial tasks that will enable ITL to make demonstrable contributions to the industry in the first year.

Technical Background:

Despite the growing interest and investment in VoIP, the basic technologies are only in their infancy and numerous key technical challenges remain to be solved. These challenges can be divided into three general areas: representation and encoding of digital speech data, enhanced network transport for VoIP, and interoperability with the existing public switched telephone network (PSTN). Representation and coding issues arise from the nature of the today’s Internet as a datagram network, offering “best effort” service, and the strict real-time demands of voice communication. Packets of speech data can be delayed or lost due to network congestion, creating an unintelligible or otherwise unacceptable reconstruction at the receiver. Coding and representation techniques that address these problems can be divided into loss reconstruction, which uses redundancy to increase the probability of successful reception of the data, and loss alleviation, which mitigates the effect of lost packets on the quality of the received signalⁱⁱ.

While current VoIP research and development is focused on adapting VoIP applications to the capabilities of today’s Internet, we must be forward thinking and begin to focus ongoing work on new, next generation internet technologies towards the problem of supporting voice and other advanced audio signalsⁱⁱⁱ (e.g., high fidelity music). There are several research efforts underway to significantly enhance the capabilities and services of the next generation IPS, including the addition of quality of service (QoS), and packet level security. To date, these efforts have not focused on the unique requirements of supporting a large scale voice infrastructure. New directions in NGI research should focus on QoS architectures, admission control and traffic shaping mechanisms, and security services specifically designed for voice and advanced audio applications.

The third area of significant challenge is the integration of new telephony applications and protocols with the existing voice infrastructure. While we expect tremendous initial growth in VoIP and Internet telephony applications and protocols, the long term success of these technologies depends upon seamless integration with the existing Public Switched Telephone Networks (PSTNs). The challenge here is to

design signaling protocols for Internet telephony that will enable new classes of multiservice applications while still maintaining the ability to interoperate with traditional telephony.

Approach:

Our approach to establishing competency in VoIP technologies is to leverage past accomplishments in areas such as testing technology for QoS sensitive applications, protocols and architecture for QoS, and the use quantitative measures of speech intelligibility. We propose to begin our work in VoIP with the construction of test and measurement systems that can be used by makers of VoIP technology to evaluate their products. Our test system will use NIST Network Emulation Tool (NIST Net), which a general-purpose tool for emulating performance dynamics in IP networks. NIST Net enables us to emulate the critical end-to-end performance characteristics imposed by various wide area network situations (e.g., congestion loss) or by various underlying subnetwork technologies^{iv}.

Central to evaluating the performance of a VoIP network is finding an easily computed quantitative measure that accurately predicts human assessment of speech quality. Subjective measures, such as Mean Opinion Score (MOS), where a panel of subjects rates speech quality on a scale from 1 to 5, are highly variable, non-repeatable, and too expensive to implement on all but the smallest scale. Existing objective measures overcome these limitations but often do not correlate well with subjective scores. Therefore, we propose the use of an automated speech recognizer to measure speech quality, building on previous ITL research^v. We will extend this initial effort to investigate the use of commercial speech recognition and transcription technology to produce a fully automated tool for assessing the quality of VoIP systems.

The construction of this initial test and measurement system will allow ITL to gain valuable competence in VoIP technologies while, at the same time, delivering useful metrology tools to the industry. Our subsequent efforts with use the test and measurement systems we have built, to conduct research in the areas of coding and representation, enhanced network transport, PSTN interoperability, and finally, transmission of other audio signals.

Proposed Deliverables

Task 1 Construct research testbed for VoIP technologies. Solicit industry involvement / contribution to VoIP test and measurement testbed.

Task 2 Research and develop prototype of a fully automated test and measurement system for voice quality over IP networks.

Task 3 Use NIST VoIP Test System to quantify existing proposed encoding and packetization techniques for loss reduction and concealment.

Future Tasks: Research advanced speech encoding and representation, and enhanced network transport, and PSTN interoperability. Expand focus to address other advanced audio signals in addition to speech.

ⁱ Voice over IP, Technology Guide Series, Applied Technologies Group, Inc., 1998

ⁱⁱ "Adaptive Loss Concealment for Internet Telephony Applications," Henning Sanneck

ⁱⁱⁱ "Networking Audio and Music Using Internet2 and Next Generation Internet Capabilities", Technical Council - Audio Engineering Society (AESW P-1001)

^{iv} Application and Protocol Testing Through Network Emulation (The NIST Network Emulation Tool), <http://www.antd.nist.gov/itg/nistnet/>.

^v "Can Speech Recognizers Measure the Effectiveness of Encoding Algorithms for Digital Speech Transmission?" C. M. Chernick, S. Leigh, K. L. Mills, and R. Toense, National Institute of Standards and Technology, December 1998.