
Telecommunications Theory

The explosive growth of telecommunications traffic in recent years continues to generate ever-increasing demands for radio spectrum while greatly increasing the loading of many telecommunications networks, both wireless and wireline. Yet the radio spectrum is a limited resource. In response to these realities, new radio technologies are being developed and implemented to use spectrum more efficiently and effectively. Also, the basic paradigm of radio spectrum management is beginning to move away from traditional, top-down frequency-assignment methods and is migrating toward autonomous, interference-limited technologies that allow dynamic reassignment of radio frequencies. But to fulfill the promise of more autonomous, locally self-controlled spectrum use schemes, the effects of noise and interference on radio receiver performance must be thoroughly understood, and such knowledge must be focused on improvements in the performance of both existing and new networks. Tools to monitor the quality of audio and video information on communication channels also must be developed and used so that audio and video quality levels can be accurately adjusted in real-time to achieve maximal quality with minimal use of available bandwidth.

To achieve these goals for the U.S. Government as well as the private sector, the Telecommunications Theory Division performs research in both wireless and wireline telecommunications, seeking to understand and improve telecommunications at the most fundamental levels of physics and engineering. Strong ongoing investigations are being maintained in the major areas of broadband wireless systems performance; advanced antenna designs; noise and interference as critical limiting factors for advanced communication systems; audio and video quality assessment; advanced spectrum sharing concepts; and radio propagation.

Through technical publications, cooperative research and development agreements (CRADAs), and interagency agreements, ITS transfers the results of its work in all these technology areas to both the public and private sectors, where the knowledge is transformed into better telecommunications for the United States, new and better products for consumers and the Government, and new opportunities for economic development and growth for the economy.

Areas of Emphasis

Audio Quality Research

The Institute conducts research and development in coding, transmission, and perception-based quality measures for voice and other audio communication systems. Projects are funded by NTIA.

Broadband Radio Research and Propagation Measurements

The Institute conducts an ongoing program of radiowave propagation research and measurements, using the ITS Mobile Radio Propagation Measurement Facility and the Digital Sampling Channel Probe (DSCP). Using these facilities, researchers can determine propagation conditions and impairments which affect new digital communication systems and answer questions regarding the viability of proposed radio services. The project is funded by NTIA.

Effects of Radio Channel on Receivers

The Institute, a recognized leader in radio channel measurement and modeling, is conducting research to assess the effects of interference and noise on the performance of radio receivers and networks. Recent work has focused on the effects of noise and interference as limiting factors in system performance. The project is funded by NTIA.

Video Quality Research

The Institute develops perception-based, technology-independent video quality measures and promotes their adoption in national/international standards. Projects are funded by NTIA.

Audio Quality Research

Outputs

- Technical publications and presentations on new research results.
- Measurements and estimates of speech and audio quality and algorithm performance.
- Algorithms and data supporting speech and audio coding and quality assessment.

Digital coding and transmission of speech and audio signals are enabling technologies for many telecommunications and broadcasting services including cellular telephone services, voice over Internet protocol (VoIP) services, and digital audio broadcasting systems. Speech and audio signals can be coded and transmitted at low bit-rates with good fidelity. In addition, coded speech and audio signals can be packetized for transmission, thus sharing radio spectrum or wired network bandwidth with other data streams and hence with other users.

Innovation in digital coding and transmission involves compromises and trade-offs among speech or audio quality, transmission bit-rate, robustness to transmission errors and losses, coding and transmission delay, and coding and transmission algorithm complexity. The ITS Audio Quality Research Program works to identify and develop new techniques to increase quality or robustness, or to lower bit-rate, delay, or complexity of digital speech and audio coding and transmission algorithms. The ultimate result of such advances is better sounding, more reliable, more efficient communications and broadcasting services.

The robustness of digital coding and transmission algorithms is critical in applications that use lossy channels such as those associated with wireless systems and those provided by the Internet. In FY 2005, Program staff have continued to work towards more robust speech coding through a method called multi-descriptive coding (MDC). In MDC an encoder forms multiple partial descriptions of a speech signal and these descriptions are sent over different channels. If all descriptions arrive at the decoder intact, a higher-quality reconstruction of the speech is possible. If channel failures cause any of the descriptions to be lost, then a lower-quality reconstruction of the speech signal is still possible.

Program staff developed and tested a multiple description pulse code modulation (PCM) speech coding system that exploits naturally occurring correlations between adjacent samples of speech and invokes a pair of appropriately designed vector quantizers. This system includes an aspect ratio parameter that allows one to trade off the speech quality when one channel is working against the speech quality when two channels are working. This in turn allows one to match the channel conditions to maximize speech quality.

Even when robust coding techniques such as MDC are deployed, time-varying channel conditions generally will cause end users to experience time-varying speech quality. But the perception of time-varying speech quality is not yet well-understood. For example, which would be preferable: speech that fluctuates between high quality and low quality, or speech that is consistently of medium quality? How would the preference change with the levels of high, medium, and low quality? And how would the preference change with the timing and nature of the quality fluctuations?

Program staff recently designed, conducted, and analyzed an experiment to characterize one simple yet fundamental component of time-varying speech quality. In this experiment, subjects heard recordings where speech quality changed twice, resulting in speech quality histories of the form “low, high, low” and “high, low, high.” Figure 1 below shows an example result from this experiment. The vertical

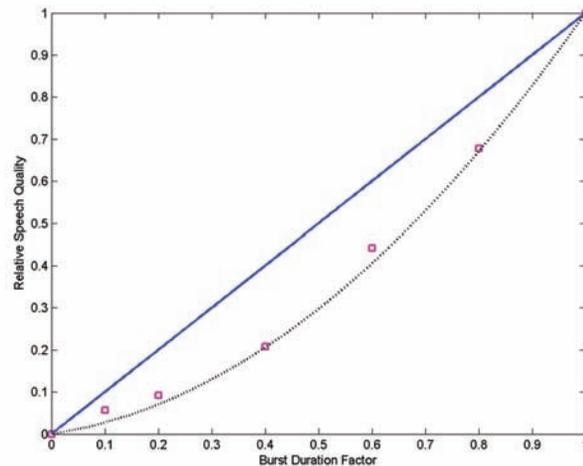


Figure 1. Example results from an experiment on time-varying speech quality. See text for details.

axis represents overall speech quality as judged by the subjects in the experiment. Here 0 represents low speech quality and 1 represents high speech quality. On the horizontal axis, the burst duration factor describes the fraction of the time that the recording has speech quality 1. (The remainder of the time the recording has speech quality 0.) If subjects judged overall speech quality by simply averaging instantaneous speech quality, then their responses would grow linearly as described by the solid blue line in the figure. Instead, subjects' responses fall below this line, as denoted by purple squares and approximated by the dotted black line in the figure. This indicates that subjects use a process that is more critical than simple averaging when judging overall speech quality. Periods of low and high quality do not balance out in the mathematical sense; rather, periods of low quality seem to carry more weight than periods of high quality. In this sense, we might say that listeners are pessimists when it comes to speech quality.



Figure 2. Subjective speech and audio testing can now be controlled by test participants using wireless PDAs (photograph by S. Wolf).

In FY 2005, Program staff designed, installed, and tested upgrades and new capabilities to the ITS Audio-Visual Laboratories. One major upgrade provides greatly enhanced flexibility for subjective test design and operation, in that subjective tests can now be controlled through a very flexible, powerful, and easy to use, high level language that is well-integrated with digital and analog audio I/O, computer graphics display, text display, text entry, and mouse input. Graphical display and mouse input is extended from a host desktop computer (operated by Program staff) to a handheld personal digital assistant (PDA) (operated by a subjective test participant). This extension is enabled by a wireless local area network (LAN) so that test participants only need to operate a single familiar, intuitive, lightweight, wireless device (see Figure 2 above).

Throughout FY 2005, program staff continued with subjective and objective audio quality testing to support this and other ITS programs. Staff continued to transfer technologies to industry, Government, and academia through numerous technical publications, presentations, guest lectures, laboratory demonstrations, and by completing peer reviews for technical journals and workshops. Program staff also incorporated recent program results into a revision of an American National Standard: ANSI T1.801-04. This telecommunications standard is titled "Multimedia Communications Delay, Synchronization, and Frame Rate," and it now includes the specification of an ITS-developed algorithm for tracking variable transmission delay across a wide range of speech coding conditions. Program publications, technical information, and other program results are available at <http://www.its.bldrdoc.gov/audio>.

Recent Publications

S.D. Voran, "A multiple-description PCM speech coder using structured dual vector quantizers," in *Proc. International Conference on Acoustics, Speech and Signal Processing*, Philadelphia, Mar. 2005.

S.D. Voran, "Multiple-description PCM speech coding by complementary asymmetric vector quantizers," in *Proc. IEEE Region 5 Conference*, Boulder, CO, Apr. 2005.

S.D. Voran, "A basic experiment on time-varying speech quality," in *Proc. 4th International MESAQIN (Measurement of Speech and Audio Quality in Networks) Conference*, Prague, Czech Republic, Jun. 2005.

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Broadband Radio Research and Propagation Measurements

Outputs

- Study of MIMO antenna systems and information theory relating to MIMO systems.
- Study of relative propagation impairments between 2.4 GHz ISM band and 5.8 GHz communications band.

An ongoing program of radiowave propagation research and measurements is supported using the ITS Mobile Radio Propagation Measurement Facility and the Digital Channel Probe. By using these facilities, researchers have the ability to determine propagation conditions and impairments which affect new digital communication systems and answer questions regarding the viability of proposed radio services.

The system has been configured to study the relative propagation impairments between the 2.4-GHz ISM band and the recently allocated 5.8-GHz communications band. Mobile data was collected in an urban environment and the relative impairments were quantified. This data is intended to help system designers of next generation systems in the 5.8-GHz band determine the relative power requirements and link budgets needed versus systems in personal communications services (PCS) and cellular bands. It will also help promote commercial high frequency spectrum use and frequency extension.

More recently the system was configured to quantify the propagation conditions in a reverberation chamber (Figures 1 and 2). These data were collected to determine the suitability of the chamber for comparative performance testing of multiple-input multiple-output (MIMO) communication systems.



Figure 1. Test equipment configured for 4 channel operation at 2.4 GHz for NIST reverberation chamber characterization (photograph by F.H. Sanders).



Figure 2. MIMO test setup in NIST reverberation chamber. Antennas configured for single input single output measurement (photograph by F.H. Sanders).

Recent Publications

J.J. Lemmon, "Radiation pattern analysis of a four-element linear array," NTIA Technical Memorandum TM-05-426, Aug. 2005.

J.J. Lemmon, "MIMO channel capacity with discrete alphabets," in *Proc. Wireless 2005*, Calgary, Jul. 2005.

P. Papazian, "Basic transmission loss and delay spread measurements for frequencies between 430 MHz and 5750 MHz," *IEEE Transactions on Antennas and Propagation*, Feb. 2005.

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Effects of Radio Channel on Receivers

Outputs

- Analysis of bit, frame, and packet transmission reliability.
- Correlations between radio channel characteristics and receiver performance.
- Radio channel measurement requirements for receiver performance analysis.

Telecommunications play a vital role in many of the services deemed essential for modern life. For many of these services, telecommunication is provided over a radio link. Land mobile, indoor, and satellite are but a few examples of radio links commonly used. The radio link consists of a transmitter, receiver, and the channel separating the two. The radio channel is often the primary impediment to fast and reliable telecommunication. Hence, the Institute has historically focused most of its efforts towards understanding the radio channel. This project builds upon these historical strengths by studying the effects of the channel on receiver performance. Figure 1 summarizes the scope of this project.

This study requires expertise in a wide range of radio engineering disciplines. For example, this study requires detailed knowledge of receiver demodulation and signal processing methods. This is a formidable challenge considering the large number of legacy receivers and the growing numbers of receivers for emerging technologies such as personal communications services (PCS), wireless

local area networks (WLAN), and global positioning system (GPS). Complexity of these receivers ranges from mere analog demodulation to digital demodulation with advanced signal processing methods including multipath equalization and error correction techniques.

This study also requires extensive knowledge of the radio channel. The radio channel is often characterized by propagation phenomena such as multipath and loss, additive noise from natural and man-made radiators, and signals from other radio links. These radio channel components affect each radio receiver uniquely. For example, PCS receivers operating in a residential neighborhood in the 1900-MHz band are primarily compromised by time-varying multipath introduced by buildings and terrain. WLAN receivers operating within buildings in the 900-, 2400-, and 5800-MHz industrial, scientific, and medical bands contend with man-made radio noise radiated by other electrical devices such as microwave ovens, in addition to multipath introduced by reflections and scattering from walls, ceilings, and objects within the room. GPS and other satellite broadcast receivers are hindered by terrestrial radio links whose signals occupy the same frequencies.

Receiver performance evaluation includes analysis of bit and frame transmission error rates. This analysis can be extended to packet transmission error rates across a network incorporating radio links. Performance evaluation metrics such as these are correlated against radio channel parameters, e.g., multipath channel root mean square delay spread, man-made noise impulsiveness, or interfering signal

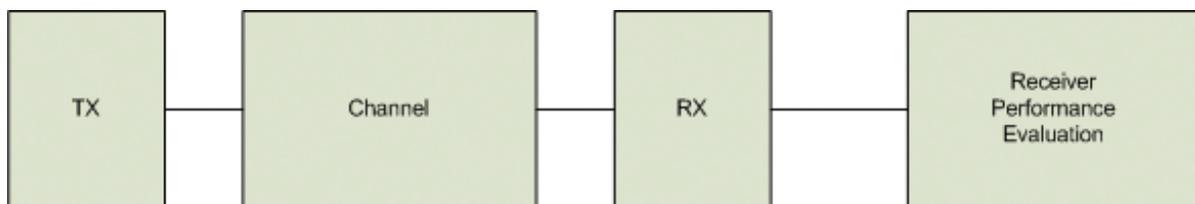


Figure 1. Scope of project includes the study of channel, receiver, and various methods of receiver performance evaluation.

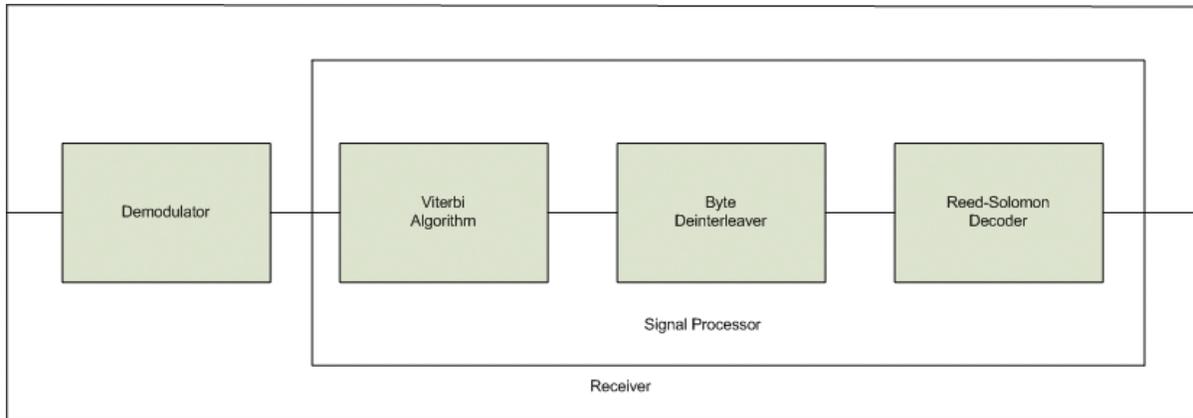


Figure 2. Block diagram of a direct broadcast satellite receiver highlighting demodulation and signal processing tasks.

level crossing rates. In addition, a considerable amount of effort is expended statistically analyzing the effect measurement uncertainties have on the performance metrics and subsequent analysis.

Success of this project is dependent on knowing the limitations of radio channel characterization measurement and analysis and how these limitations influence the study of the effects of the radio channel on the receiver. These subtleties are addressed by project personnel who have performed many of these measurements and analyses themselves and who continue to have close working relationships with those currently performing them. Project personnel are also well versed in analysis of random processes which are used to characterize radio channels and transmitted signals.

In FY 2005, project personnel focused on two primary tasks. The first task supported research within the Institute on the effects of gated Gaussian noise signals on direct broadcast satellite receiver performance. A block diagram of the direct broadcast satellite receiver with associated demodulator and signal processing components is shown in Figure 2. We supported this research by verifying measurements of the susceptibility to continuous Gaussian noise analytically and gated-Gaussian noise through simulation. We also verified gated-Gaussian noise amplitude probability distribution and the power spectral density characterization measurements. Verification of the APD measurement was critical since the effects of interfering signals on receiver performance are often correlated to the peak to average power ratio provided by the APD.

The second task concerned the calculation of the uncertainties of radio channel characterization measurements. These uncertainties are rarely reported in professional journal articles. Our goal is to describe how the uncertainties are computed from a finite sample set and show how they can be used in multipath, man-made noise, and signal characterization measurement analysis. In the future, these uncertainties will be translated to receiver performance parameters.

Recent Publications

M. Cotton, R. Achatz, J. Wepman, and P. Runkle, "Appendix B: Verification of susceptibility results" in "Interference potential of ultrawideband signals - Part 2: Measurement of gated-noise interference to C-band satellite digital television receivers," NTIA Report TR-05-429, Aug. 2005.

R. Dalke, "Statistical considerations for noise and interfering characterization measurements," NTIA Report, in progress.

R. Dalke and G. Hufford, "Analysis of the Markov character of a general Rayleigh fading channel," NTIA Technical Memorandum TM-05-423, Apr. 2005.

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Video Quality Research

Outputs

- Digital video quality measurement technology.
- Journal papers and national/international video quality measurement standards.
- Technical input to development of U.S. policies on advanced video technologies.
- A national objective and subjective digital video quality testing laboratory.

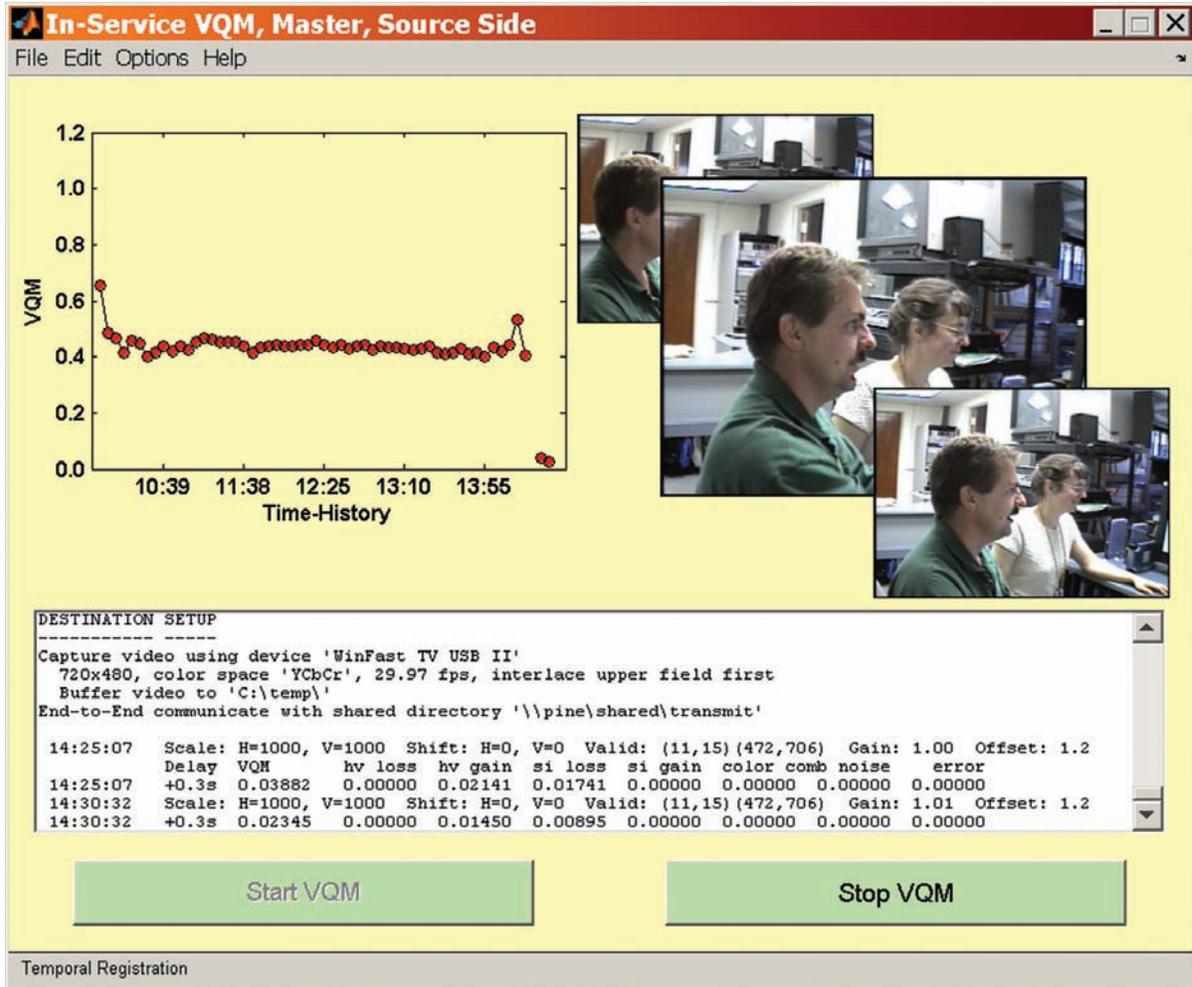
Objective metrics for quantifying the performance of digital video systems (e.g., direct broadcast satellite, digital television, high definition television, video teleconferencing, telemedicine, internet and cell phone video) are required by end-users and service providers for specification of system performance, comparison of competing service offerings, network maintenance, and use optimization of limited network resources. The goal of the ITS Video Quality Research project is to develop the required technology for assessing the performance of these new digital video systems and to actively transfer this technology to other government agencies, end users, standards bodies, and the telecommunications industry, thereby producing increases in quality of service that benefit all end users and service providers.

To be accurate, digital video quality measurements must be based on perceived “picture quality” and must be made in service. This is because the performance of digital video systems is variable and depends upon the dynamic characteristics of both the input video and the digital transmission system. To solve this problem, ITS has continued to develop new measurement paradigms based upon extraction and comparison of low-bandwidth, perception-based features that can be easily communicated across the telecommunications network. These new measurement paradigms (now commonly known throughout the world as “reduced reference” measurements) have received three U.S. patents, been adopted as the North American Standard for measuring digital video quality (ANSI T1.801.03-2003), been included in two International Telecommunication Union Recommendations (ITU-T Recommendation J.144Revised, ITU-R Recommendation BT.1683), and are currently being used by hundreds of individuals and organizations worldwide.

To facilitate the transfer of ITS-developed video quality metrics (VQMs) into the private sector, ITS has developed two software tools. The first tool, called the “Laboratory VQM Tool,” is useful for bench testing of video systems. For this tool, video from the source and destination ends of a video system must be present at a single personal computer (PC). Work was completed in FY 2005 to expand the existing VQM software tools to include new end-to-end video quality monitoring capabilities. This new software tool, called the “In-Service VQM (IVQM) Tool,” runs on two PCs, one located at the source end and the other located at the destination end. The two PCs communicate their reduced reference features via the Internet. Using these new software tools, users and service providers can quantify the digital video quality of their networks using methods standardized by ANSI and the ITU.

The figure on the next page gives a screen snapshot of the IVQM monitoring screen (master controlling computer at video source). The IVQM monitoring screen contains a menu bar, a time history plot of the selected VQM, three sample frames from the last video capture (first, middle, and last), a text box listing detailed calibration and VQM results, two buttons (“Start VQM” and “Stop VQM”), and a status bar at the bottom of the screen. VQM estimates are reported on a scale from zero to one, where zero means that no impairment is visible and one means that the video clip has reached the maximum impairment level. The capture time of the sequences is displayed on the X-axis. Results and captured video sequences can be saved for later analysis.

During FY 2005, 176 new Cooperative Research and Development Agreements (CRADAs) were implemented with U.S. companies/individuals and 111 new Evaluation License Agreements (EVAs) were implemented with foreign companies/individuals. These CRADAs and EVAs provide companies with an easy mechanism for evaluating ITS video quality measurement technology and software before signing commercial licensing agreements. As a result of this arrangement, a leading provider of telecommunication performance measurement equipment and services signed a commercial licensing agreement with ITS in FY 2005.



Snapshot of IVQM monitoring screen (master controlling computer at video source).

Recent Publications

M.H. Pinson and S. Wolf, "In-service video quality metric (IVQM) user's manual," NTIA Handbook HB-05-424, Apr. 2005.

S. Wolf and M.H. Pinson, "Low bandwidth reduced reference video quality monitoring system," in *Proc. of First International Workshop on Video Processing and Quality Metrics for Consumer Electronics*, Scottsdale, Arizona, Jan. 2005.

M.H. Pinson and S. Wolf, "Video scaling estimation technique," NTIA Technical Memorandum TM-05-417, Jan. 2005.

Further information can be found on the Video Quality Research home page at <http://www.its.bldrdoc.gov/n3/video>.

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