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# Telecommunications Theory

The rapid growth of telecommunications in the last 50 years has caused increased demand for radio spectrum and has generated high levels of loading in many telecommunications networks, both wireless and wireline. In response to these trends, new radio technologies have been developed and implemented to use spectrum more efficiently and effectively. To meet critical national needs for better radio communication systems, the parameters that limit network performance must be thoroughly understood, and such knowledge must be focused on improvements in the performance of 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 Institute's Telecommunications Theory Division performs research in both wireless and wireline telecommunications, seeking to understand and improve telecommunications at the most fundamental level. Strong ongoing investigations are maintained in the major areas of broadband wireless systems performance; advanced antenna designs; noise as a limiting factor for advanced communication systems; audio and video quality assessment; advanced spectrum sharing concepts; and radio propagation.

Through 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 sector, 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 leading to standardization and industry implementation of perception-based, technology-independent 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 Measurements 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 Networking Performance**

The Institute, a recognized leader in radio channel measurement and modeling, is involved in research to assess the effects of the wireless communications channel on communications system network performance. Recent work has focused on 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.

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# Audio Quality Research

## Outputs

- Technical publications and presentations demonstrating new research results.
- Algorithms and data supporting speech and audio coding and quality assessment.
- Objective estimates and subjective measurements of speech and audio quality.

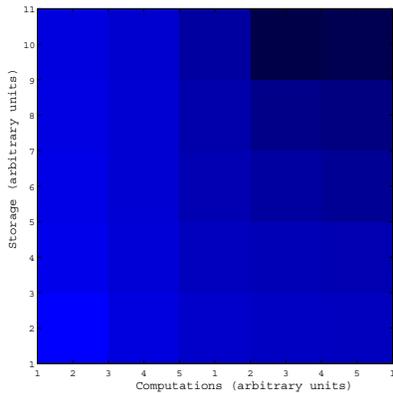


Figure 1. Bit-rate as a function of codec computations and storage. Higher rates are brighter blue.

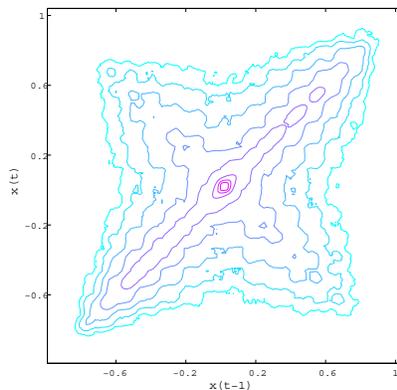


Figure 2. Contours of constant relative frequency for two-dimensional histogram of adjacent speech samples.

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 now be coded and transmitted at remarkably 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.

In digital coding and transmission there is a four-way trade-off among quality, bit-rate, delay, and complexity. The ITS Audio Quality Research Program works to identify and develop new techniques to increase quality or lower the bit-rate, delay, or complexity of digital speech and audio coding and transmission. The ultimate result of such advances is better sounding, more reliable, more efficient telecommunications and broadcasting services.

In one FY 2004 Program effort, a family of speech coders with fixed speech quality and fixed delay was developed. For this family, the four-way trade-off described above becomes a 2-way trade-off: complexity vs. bit-rate. Complexity is comprised of two major factors: a computational requirement and a storage requirement. Figure 1 shows how bit-rate can be reduced by increasing either computations or storage. For a given implementation, one would select the member of this speech coder family that gives the lowest bit rate and has computation and storage requirements consistent with the implementation platform.

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. The Program has also 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.

One MDC approach recently developed in the Program exploits the naturally occurring correlations between adjacent samples of speech. This correlation can be seen in the way probability mass is organized along the diagonals in the two-dimensional histogram in Figure 2. Working with these correlations, one can effectively decompose a stream of speech samples into two streams that provide maximum fidelity both individually and when combined.

In digital speech and audio systems, a set of complex time-varying interactions among signal content, source coding, channel coding, and channel conditions often make it difficult to define or measure speech or audio quality. The Audio Quality Research Program operates a subjective testing facility and runs controlled experiments to gather listeners' opinions of the speech or audio quality of various coding and transmission systems. The Program has also developed and verified tools for the objective estimation of telephone bandwidth speech quality.

In FY 2004 the tools for objective estimation were further enhanced through the development, optimization, and testing of a robust technique for tracking variable transmission delay for a wide range of speech coding conditions. This is important because transmission delay can vary significantly and rapidly in packetized speech transmission systems, even within a single spoken phrase. This delay variation arises from the basic nature of packetized data networks and can be mitigated, but not eliminated, through buffering techniques. To understand the resulting speech quality, it is imperative that this continually changing delay be accurately tracked.

VoIP is presently the most prominent example of a packetized speech transmission system, but some land-mobile radio systems are also now interconnected through packetized links. Thus the Program developed a tool that is effective across the range of VoIP conditions and land-mobile radio conditions. This was accomplished through detailed analysis of a broad database that includes over 6000 different

recordings. This analysis led to a successive refinement approach that can respond appropriately to each of the different conditions of interest. Figure 3 provides an example of actual and estimated delay histories for a worst-case condition involving 2.45 kbps speech coding and a high rate of packet impairments.

Signal content plays an important role in speech quality and thus the type and amount of background noise that is combined with speech can have a significant effect on speech quality. As cellular phones have become ubiquitous, they are routinely used in locations with significant background noise. In FY 2004 Program staff developed a database of digital recordings of background noise signals. This database includes recordings from bus and car interiors, office, coffee shop, and party environments, and urban sidewalk environments as well. The recordings are used in the Program to create realistic and diverse environments for speech quality assessment.

Throughout FY 2004, subjective and objective audio quality testing was conducted to support Audio Quality Research efforts. In addition, Program staff continued with selective upgrades to the ITS Audio-Visual Laboratories to keep them abreast of the state-of-the-art. Program staff continued to transfer technologies to industry, Government, and academia throughout FY 2004, using technical publications and lectures, laboratory demonstrations, and by completing peer reviews for technical journals and workshops. Program publications and other results are available at <http://www.its.bldrdoc.gov/audio>.

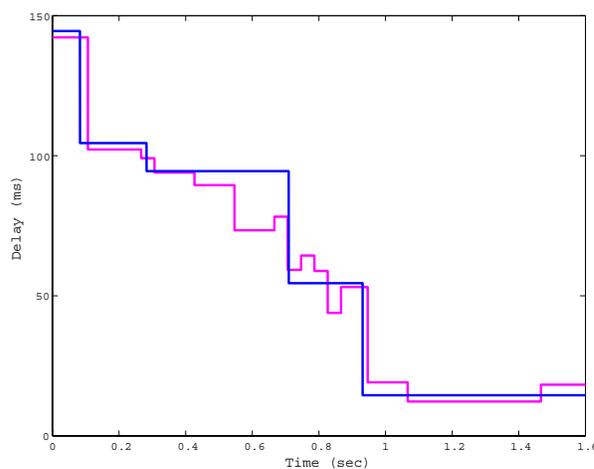


Figure 3. Actual (blue) and estimated (magenta) delay histories for 2.45 kbps speech codec with highly impaired packetized transmission.

### Recent Publications

S.D. Voran, "Compensating for gain in objective quality estimation algorithms," in *Proc. International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, Montreal, May 2004.

S.D. Voran, "A bottom-up algorithm for estimating time-varying delays in coded speech," in *Proc. 3th International Conference on Measurement of Speech and Audio Quality in Networks (MESAQIN)*, Prague, Czech Republic, May 2004.

For more information, contact:

Stephen D. Voran  
(303) 497-3839  
e-mail [svoran@its.bldrdoc.gov](mailto:svoran@its.bldrdoc.gov)

# Broadband Radio Research and Propagation Measurements

## Outputs

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

An ongoing program of radiowave propagation research and measurements is supported using the ITS Mobile Measurements Facility and the Digital Sampling Channel Probe (DSCP). 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. In the recent past these facilities were used to investigate propagation for personal communications services (PCS) at 1.85 GHz and local

multipoint distribution services (LMDS) near 30 GHz. The measurement van is capable of fixed or mobile operation and the DSCP can be configured over a wide range of bandwidths and frequencies.

Recently the system was configured at four frequencies 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 PCS and cellular bands. It will help promote commercial high frequency spectrum use and frequency extension. Figure 1 shows the routes surveyed in downtown Denver. Figure 2 shows the average path loss slopes over the measured frequency range. For more details of this work see references [1] and [2].

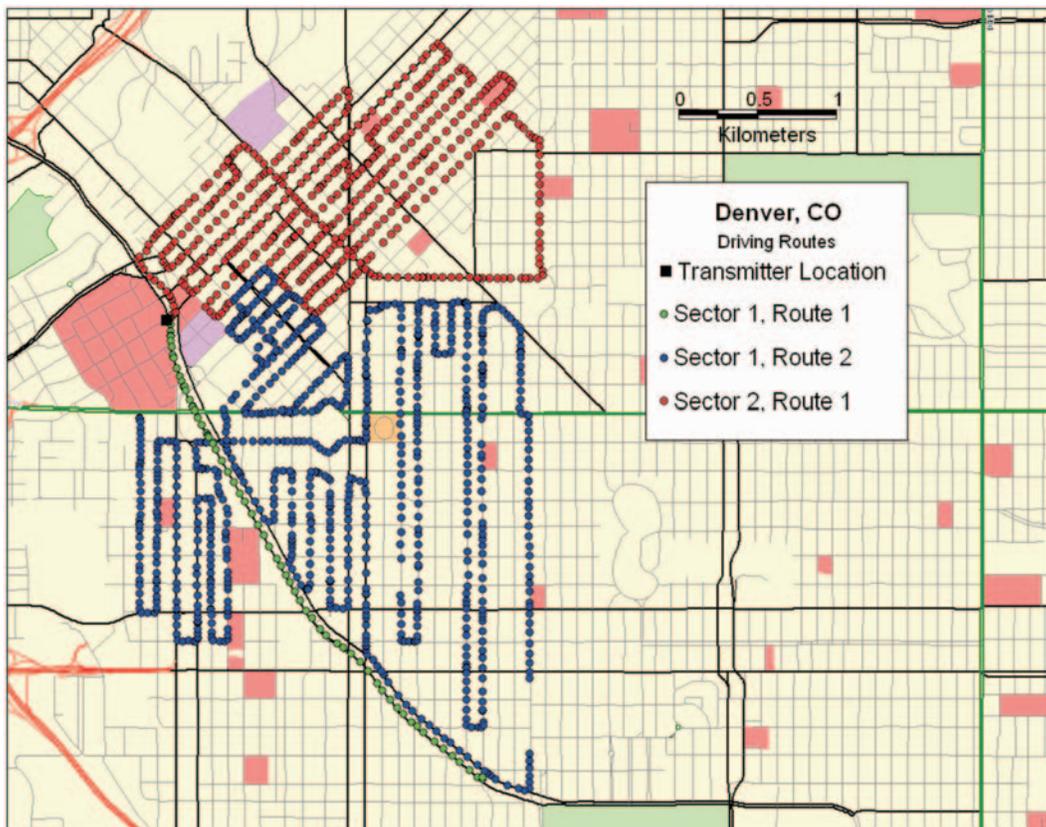


Figure 1. Drive route map for radiowave propagation survey in downtown Denver, CO.

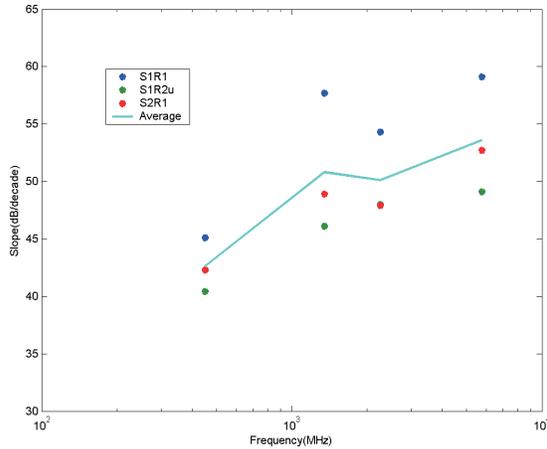


Figure 2. *L<sub>b</sub>* slope versus frequency for three urban drive routes in Denver, CO.

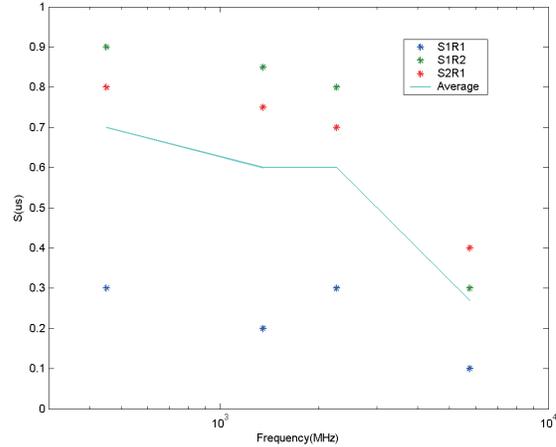


Figure 3. Average results for delay spread exceeded 50% of the time ( $S_{50\%}$ ) for 3 cells versus transmission frequency.

In addition to the path loss data, the relative delay spread versus frequency was also measured. In Figures 3 and 4 we see that delay spread decreased versus transmission frequency. More detailed analysis of these effects can be found in references [1] and [2].

In the coming year the Broadband Radio project will continue its work in the area of multiple input multiple output (MIMO) antenna systems. The purpose of this work is to advance spectrum efficiency using new high capacity radio technology. The project plans to work cooperatively with NIST to characterize a reverberation chamber for use in testing MIMO systems. ITS will also continue its study of information theory relating to MIMO systems.

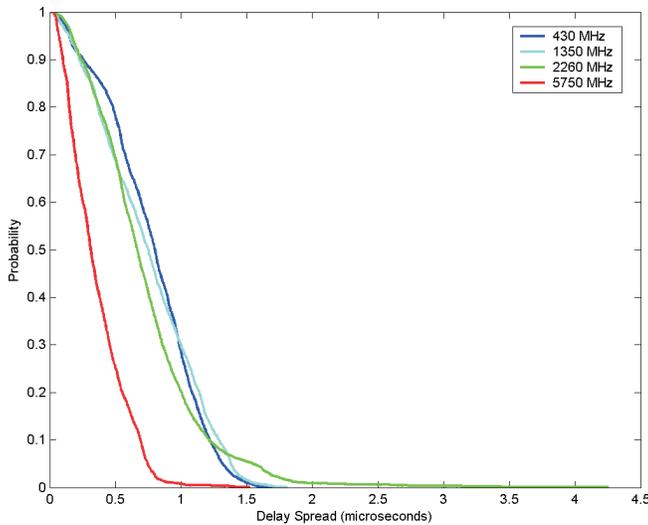


Figure 4. The cumulative distribution function of delay spread versus transmission frequency for cell 2 route 1.

### Recent Publications

- [1] P. Papazian and M. Cotton, "Relative propagation impairments between 430 MHz and 5750 MHz for mobile communication systems in urban environments," NTIA Report TR-04-407, Dec. 2003.
- [2] P. Papazian, "Basic transmission loss and delay spread measurements for frequencies between 430 MHz and 5750 MHz," to be published in *IEEE Transactions on Antennas and Propagation*, Feb. 2005.

For more information, contact:

Peter B. Papazian  
 (303) 497-5369  
 e-mail ppapazian@its.bldrdoc.gov

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# Effects of Radio Channel on Networking Performance

## Outputs

- Models of bit, frame, and packet error random processes.
- Quantitative analysis of effects of radio channel on network performance.
- Estimation of the radio channel's impact on spectral capacity.

The Institute is a recognized leader in radio channel measurement, modeling, and analysis. In the past 10 years this leadership has included work in characterizing multipath in personal communication services (PCS) and wireless local area network (WLAN) frequency bands as well as man-made noise at VHF and UHF frequencies. Such knowledge is essential for the development of robust mobile radio links. For example, development of new adaptive equalizers for modern, wide-bandwidth mobile radio links would not be possible without radio channel multipath measurement, modeling, and analysis.

Wireless network hosts that access the Internet are proliferating. IEEE 802.11 "WiFi" WLAN and 2.5/3rd generation PCS general packet radio service (GPRS) are but two examples. Recent research has shown that the radio channel can significantly degrade network performance by decreasing throughput, increasing delay, and losing packets. This degradation ultimately limits the usefulness of allocated spectrum.

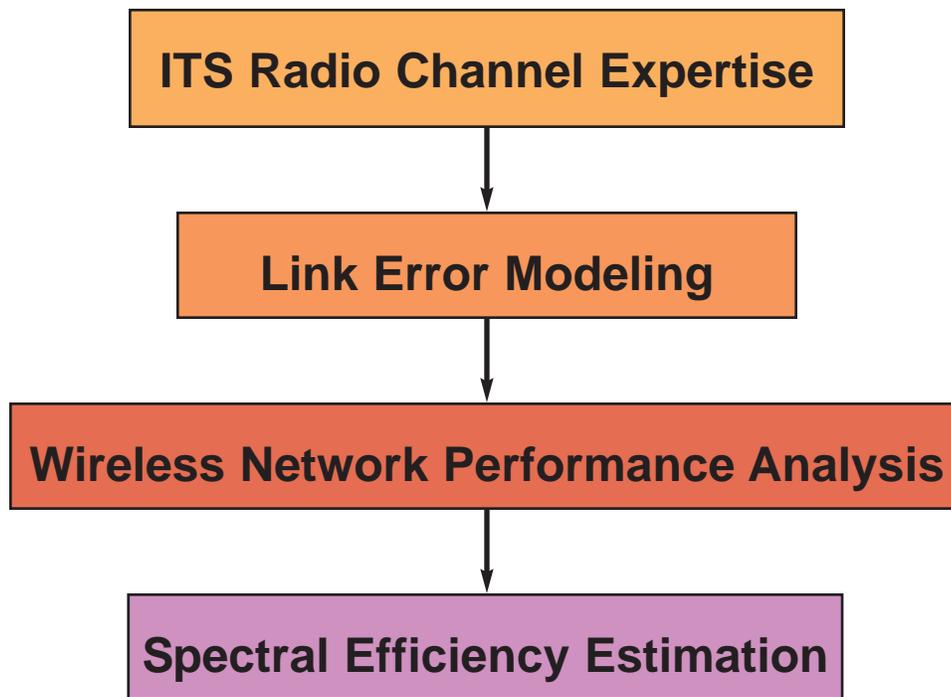
ITS is currently striving to translate its radio channel expertise into information that helps designers to improve the reliability and regulators to estimate the spectral efficiency of wireless networks. This is being accomplished by focusing on three tasks — (1) accurate modeling of the radio link bit, frame, and packet error processes resulting from radio channel impairments, (2) investigation of analytic techniques that correlate network performance to these error processes, and (3) computation of wireless network spectral capacities that account for these error processes.

Previous work included development of a radio link simulator incorporating multipath radio channel impairments. In FY 2004, this simulator was used to model bit and frame error processes caused by radio channel multipath. Initially, two frequency selective multipath radio channels were investigated. The first radio channel, referred to as the Hufford channel, had a direct path in addition to diffuse multipath created by a number of paths with independent Rayleigh fading processes. The second radio channel, referred to as the Gaussian wide sense stationary uncorrelated scattering (GWSSUS) channel, had only diffuse multipath. Bit and frame error processes due to the Hufford channel had independent, geometrically distributed time intervals. The error process due to the GWSSUS channel was markedly different.

In order to understand these differences we turned the investigation towards the analysis of the Rayleigh fading channel. Previous research has shown that the amplitudes of the Rayleigh fading channel can be modeled as a first-order Markov process where the current amplitude is dependent only on the amplitude of the previous sample. We hypothesized that the differences between the error processes may be due to the memory introduced by the first order Markov process.

We began our research by investigating the claim that the amplitude process was a first-order Markov process. The results of this investigation, currently in IEEE review, clearly demonstrated that only a fraction of the information needed to predict the current amplitude is in the previous sample and a more complex model is required to include information in earlier samples. In FY 2005, ITS plans to investigate the impact of this finding on bit and frame error processes used in network analysis and simulation.

In FY 2003, ITS completed a comprehensive search of professional literature which defined the scope of the effects of the radio channel on network tasks. This search indicated that queuing, routing, and end-to-end transmission tasks were the most severely compromised by the effects of the radio channel.



*ITS is striving to translate its radio channel expertise into information that helps designers analyze performance and regulators estimate spectral efficiency of wireless networks.*

Today, simulation is often used to correlate the effects of the radio channel on the performance of these tasks. However, our search also pointed out that validation of simulation results, either through experimentation or theoretical analysis, is difficult and not commonly done.

In FY 2005, we will explore new methods for validating wireless network performance simulations. One approach to this problem is to find ways of incorporating the effects of the link error process into theoretical expressions which are commonly used to predict wired network throughput and delay. These analytic expressions include Burke's theorem which is commonly used to investigate the behavior of two queues, or Jackson's theorem which is commonly used to investigate the behavior of a group of queues.

Spectrum capacity is often measured in terms of the number of voice channels which can be supported per unit area. With the proliferation of the Internet, it will become important to measure spectrum capacity

in terms of the number of end-to-end packet transmission circuits the spectrum can support. This estimate will clearly be dependent on the effects of the radio channel on network tasks such as queuing, routing, and end-to-end transmission. In FY 2005, the impact radio channel impairments have on computing the spectrum capacity of packet transmission circuits will be investigated.

#### **Recent Publication**

R. Dalke and G. Hufford, "Analysis of the Markov character of a general Rayleigh fading channel," submitted to *IEEE Transactions on Vehicular Technology*.

*For more information, contact:*  
Robert J. Achatz  
(303) 497-3498  
e-mail [rachat@its.bldrdoc.gov](mailto:rachat@its.bldrdoc.gov)

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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, have been adopted as the North American Standard for measuring digital video quality (ANSI T1.801.03-2003), have been included in two International Recommendations (see Recent Publications below), and are currently being used by hundreds of individuals and organizations worldwide.

During FY 2004, international standardization of the ITS General video quality metric (VQM) was completed with the publication of ITU-R Recommendation BT.1683 and ITU-T Recommendation J.144R. These new international recommendations provide end-users and service providers with standardized methods for measuring the video quality of standard definition television (SDTV) systems. One hundred and seventy-two new Cooperative Research and Development Agreements (CRADAs) were implemented with U.S. companies/individuals and 91 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.

During FY 2004, ITS worked to extend the above patented and standardized video quality measurement techniques to two new areas; high-definition



*Figure 1. HDTV subjective viewing room.*

TV (HDTV) and multimedia (MM) systems. HDTV and MM differ from SDTV in image resolution, viewing distances, display type, and user expectations. The differences between SDTV and HDTV/MM necessitate the creation of new subjective and objective testing facilities and procedures. Figure 1 shows a new HDTV subjective viewing room that was used to conduct ITS's first HDTV subjective quality experiment. Figure 2 shows a new HDTV transmission/reception system that was used to generate transmission impairments for this HDTV experiment. Viewer ratings of HDTV compression and transmission quality will be used to see if ITS's SDTV technology scales to HDTV (HDTV has approximately 4 times the resolution of SDTV). Similar investigations are being performed for MM resolution systems, which typically have only 1/4 to 1/16 the resolution of SDTV.



Figure 2. HDTV transmission/reception system.

The current VQM software for the CRADAs and EVAs mentioned above is limited to bench testing, where video from the source and destination ends of a video system under test must be present at a single PC. Work began in FY 2004 to expand the existing VQM software tools to include new end-to-end video quality monitoring capabilities. This new software 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 the new software tools, users and service providers will be able to monitor their end-to-end digital video quality.

### Recent Publications

M. Pinson and S. Wolf, "A new standardized method for objectively measuring video quality," *IEEE Transactions on Broadcasting*, v. 50, n. 3, pp. 312-322, Sep. 2004.

ITU-R Recommendation BT.1683, "Objective perceptual video quality measurement techniques for standard definition digital broadcast television in the presence of a full reference," approved Jun. 2004.

ITU-T Recommendation J.144R, "Objective perceptual video quality measurement techniques for digital cable television in the presence of a full reference," approved Mar. 2004.

ITU-T Recommendation J.149, "Methodological framework for specifying accuracy and cross-calibration of video quality metrics (VQM)," approved Mar. 2004.

M. Pinson and S. Wolf, "The impact of monitor resolution and type on subjective video quality testing," NTIA Technical Memorandum TM-04-412, Mar. 2004.

M.H. Brill, J. Lubin, P. Costa, S. Wolf, and J. Pearson, "Accuracy and cross-calibration of video quality metrics: New methods from ATIS/T1A1," *Signal Processing: Image Communication*, v. 19, pp 101-107, Feb. 2004.

S. Wolf, "Color correction matrix for digital still and video imaging systems," NTIA Technical Memorandum TM-04-412, Dec. 2003.

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

For more information, contact:  
Stephen Wolf  
(303) 497-3771  
e-mail [swolf@its.bldrdoc.gov](mailto:swolf@its.bldrdoc.gov)