Telecommunications Theory

The rapid growth of telecommunications in the last 50 years has caused crowding in the radio spectrum and high levels of loading in many telecommunications networks, both wireless and wireline. New radio technologies must be developed and implemented to alleviate spectrum crowding. The parameters that limit network performance need to be thoroughly understood, and that knowledge needs to be brought to bear on improving the performance of existing and new networks. Tools to monitor the quality of audio and video information on communication channels also need to be developed and used so that network mechanisms can be adjusted in realtime 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.

ITS transfers the results of its work in all these technology areas to both public and private users, where the knowledge can be transformed into better telecommunications, new and better products, and new opportunities.

Areas of Emphasis

Advanced Antenna Testbed
The Institute has developed an advanced antenna testbed to be used in the investigation of "smart" antennas, which can greatly increase the capacity of wireless communications systems. The project is funded by NTIA.

Applied Electromagnetics
The Institute conducts research on the radio propagation channels that will be employed in new wireless communication technologies such as personal communications services and third generation (3G) wireless. Projects are funded by NTIA and DoD.

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.

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 network 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.

Advanced Antenna Testbed

Outputs

- Wideband radio channel sounding measurements.
- Antenna array diversity gain data.
- Angle of arrival input data for adaptive antenna schemes.
- 16-element MIMO response over a conductive ground plane.

The use of wireless mobile personal communications services (PCS) and wireless local area networks (WLANs) is expanding rapidly. Multiple-access schemes based on frequency division, time division, and orthogonal coding are presently used to increase channel capacity and optimize channel efficiency. Adaptive or “smart” antenna arrays can further increase channel capacity through spatial division. Antenna arrays can produce multiple beams as opposed to a simple omnidirectional antenna. Numerous narrow beams can be used to divide space, allowing the re-use of multiple-access schemes, and thereby increasing channel capacity. Adaptive antennas can also track mobile users, improving both signal range and quality. For these reasons, smart antenna systems have attracted widespread interest in the telecommunications industry for applications to third generation wireless systems.

ITS has developed an advanced antenna testbed (ATB) to serve as a common reference for testing adaptive antenna arrays and signal combining algorithms, as well as complete systems. The ATB builds on wideband channel measurement systems previously developed by ITS. These systems use a maximal length pseudo-noise (PN) code generator to apply binary phase-shift keying (BPSK) modulation to a radio channel carrier frequency at the transmitter. The received signal is correlated at the receiver with the known PN code producing an impulse-like response. The impulse response characterizes the channel over a wide bandwidth (up to 50 MHz) about the carrier frequency. Digitization of the received data allows for post-processing to examine various combining algorithms and digital beam forming schemes. Channel sounding can be done continuously or in selected bursts.

Figure 1. 16-element transmit and receive antenna arrays used for MIMO testing at the NIST open area test site. The closer array is the receiving antenna (photograph by P. Papazian).

A recent ATB application is a 16-element multiple input, multiple output (MIMO) experiment. Two 16-element MIMO arrays were fabricated and tested and then deployed at the NIST open area test site, as shown in Figure 1 on the previous page. The objective of the test was to measure the **H** matrix in a known RF environment. This allowed a comparison between the Bell Labs layered space-time (BLAST) theory and the measurement capability of a wideband system using orthogonal coding (see **Recent Publications** below).

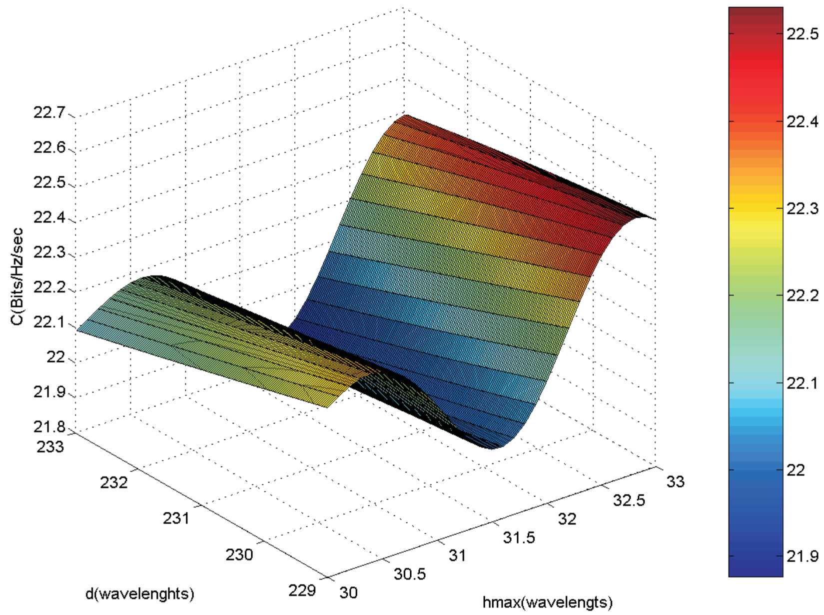

Figure 2. The capacity (C) of a 16x16 element array situated over a ground plane versus the antenna array separation (d), and the height above the ground plane of the top element of the receiving array (hmax).

A transmitter capable of generating 16 orthogonal pseudo-noise codes, one for each transmit element, was designed and fabricated using field programmable gate array (FPGA) technology. The signal received on each antenna element will then consist of the signal from all 16 transmitters after combination by the radio channel. After recording the sixteen receive channels, the 256-element channel matrix **H** can be assembled from the data. The MIMO capacity C for a communications link with n_T transmitters and n_R receivers can then be calculated using the following formula:

$$C = \log_2\left[\det\left(I + \frac{\rho}{n_T} HH^+\right)\right] bits / hz$$

$$where\ I = identity\ matrix$$

$$\rho = signal\ to\ noise\ ratio$$

$$H = complex\ transmission\ matrix$$

$$H^+ = hermetian\ transpose\ of\ H$$

Since it was known that small changes in transmitter separation and array height could change the **H** matrix, a parameter study was done to evaluate the effects of array positioning errors. Some results of this study are shown in Figure 2 above.

The ATB system is portable: both transmit and receive systems may be van-mounted. ATB measured data can be applied to the design of smart antenna PCS systems, evaluating system performance, channel model development and verification, and large communications system simulations. (See the Tools & Facilities section, p. 69, for more information about the ATB.)

Recent Publications

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.

P.B. Papazian, Y. Lo, J.J. Lemmon, and M.J. Gans, "Measurements of channel transfer functions and capacity calculations for a 16x16 BLAST array over a ground plane," NTIA Report TR-03-403, Jun. 2003.

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Applied Electromagnetics

Outputs

- Analytical models in applied electromagnetics.
- Metamaterials theoretical investigation.

ITS has a rich history of developing theoretical electromagnetic (EM) models for a wide variety of propagation scenarios. Recently, an emphasis has been placed on the EM behavior of complex structures and materials to support innovative designs of sophisticated devices for more versatile radio applications. EM theory is based on solutions to Maxwell's equations. When applied to realistic scenarios with broken symmetries and limiting dimensions, solutions involve complex variables and integrals with no closed-form solution. Although computational methods (e.g., finite-difference time-domain and finite difference) provide means to solve such problems, we focus on analytical techniques to provide intuitive understanding of physical phenomena.

Modeling the EM properties of homogeneous media gives a classic example of analytic modeling. Considering a tangible number of sources, closed-form theoretical solutions are obtainable for the electric and magnetic fields everywhere in space. However, this method is not practical for modeling of the prohibitively large number of sources that occur at the atomic level. Methods involving spatially-averaged or macroscopic field quantities are more relevant. In the presence of applied fields, macroscopic field relations are dependent on average moment densities of the medium and are derived from multipole expansions of the averaged charge and current densities. All multipole moments combine to form classical models for permittivity ($\boldsymbol{D}=\epsilon\boldsymbol{E}$) and permeability ($\boldsymbol{H}=\boldsymbol{B}/\mu$).

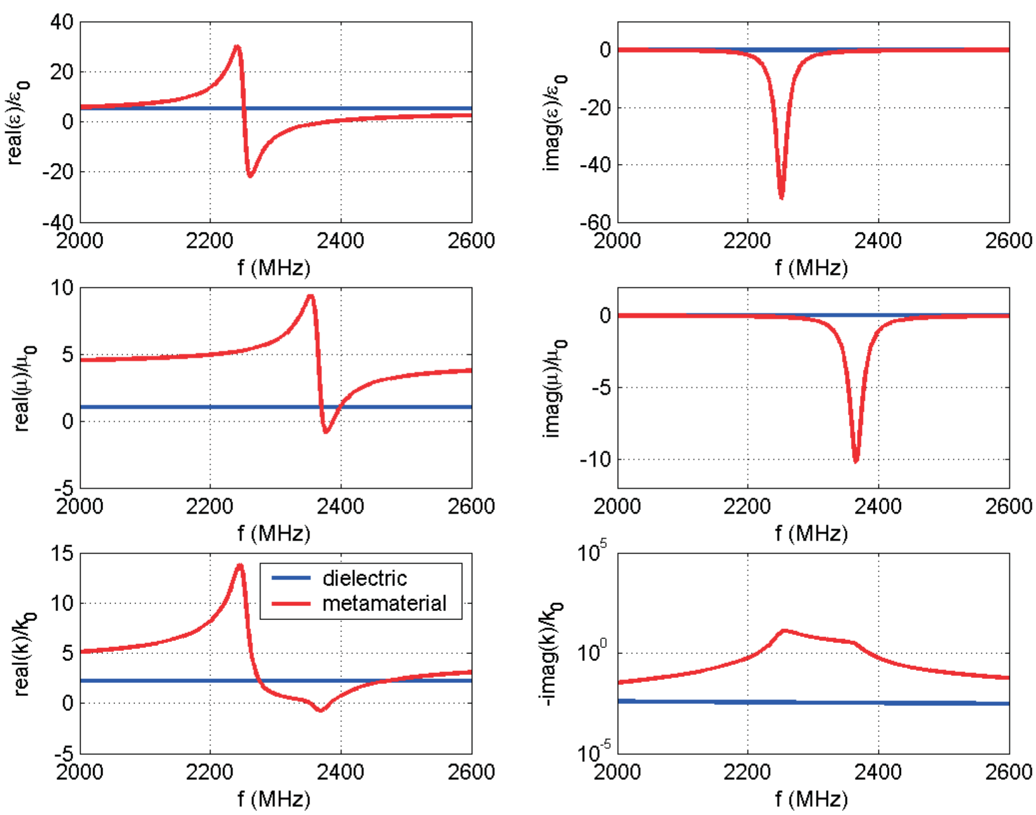

Figure 1. Double-negative material properties of a metamaterial composed of non-conducting spherical particles compared to material properties of a dielectric.

In applied EM scenarios, the scatterers inside man-made structures are typically metal or dielectric objects with dimensions ranging from relatively large to nanometer size. Whether induced moments arise from atomic-size scatterers or from macroscopic aggregates of matter, as long as wavelength is substantially larger than the dimensions and spacings of scatterers, the concept of effective medium parameters remains valid. Similarly, quasi-static approximations at relatively large wavelengths provide means to model complex structures and geometries into effective properties. For example, complicated 2d arrays and thin layers can be modeled with equivalent impedance surfaces, periodic arrays of conducting wires or small metal particles can be modeled with a sheet of average current, and arrays or random mixtures of particles in 3d can be modeled with effective medium parameters.

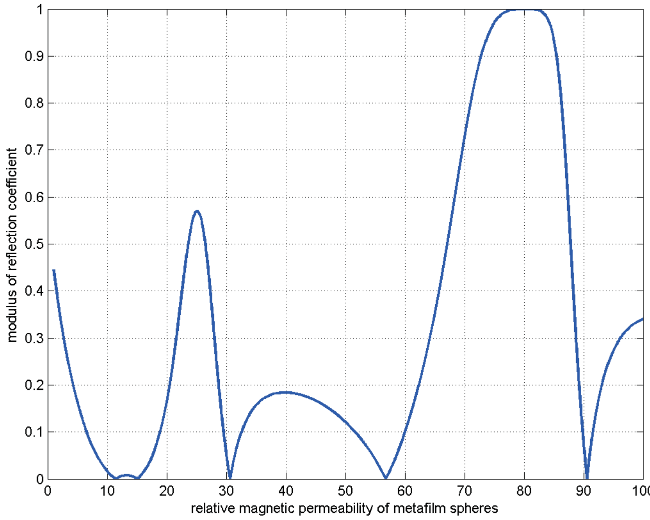

Figure 2. Metafilm reflection coefficient modulus versus magnetic permeability of spherical particles inside metafilm.

When wavelengths are comparable to and smaller than the dimensions and spacings of the scatterers, fields no longer see the composite as an effective media and more elaborate techniques to analyze the EM field interaction are necessary. Some interesting highly-dispersive EM behavior occurs. For periodic composite materials, resonances occur due to the size of the scatterer. At wavelengths near resonance, the electric and magnetic polarizations associated with individual inclusions can be simultaneously 180° out of phase with the applied **E** and **H** fields; the consequent phase velocity is in the opposite direction of the energy flow of the propagating wave in order to uphold the radiation condition. This scenario is equivalent to simultaneously negative real parts of ϵ and μ (see Figure 1 on previous page). Materials of this type have not been found in nature and have been referred to as double-negative, negative-index, and left-handed materials. They have received a great deal of attention because of their great potential for new applications.

Metamaterials are engineered composites that are designed to take advantage of such properties. These types of man-made materials are commonly engineered by designing specifically shaped scatterers embedded periodically through a volume in order to achieve a desirable bulk effect. Obviously, the more control we have over the properties of the metamaterial, the more applications we can get out of it. In fact, it has been shown that a metafilm composed of magneto-dielectric spherical particles can be designed to have total transmission or total reflection (see Figure 2 above). Further, if the inclusions were made from a material wherein its properties could be changed in real-time (e.g., with a biasing field or voltage), then a controllable surface can be realized. It is not too hard to imagine adaptable antennas and radomes that control the direction of emission, enhance emission rate, suppress interference, and perform other types of system optimization.

In our efforts we have conducted comprehensive and mathematically rigorous analyses of fundamental electromagnetic concepts applied to metamaterials. Topics include modeling the electric and magnetic polarization of metamaterials, deriving the propagation characteristics for various fundamental geometries, and exploring limitations imposed by finite dimensions of the bulk composite. Investigation into electronically controlling the electric and magnetic properties of metamaterials will also be a subject for study in pursuit of adaptable applications.

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Audio Quality Research

Outputs

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

Digital coding and transmission of speech and audio signals are enabling technologies behind many innovations in telecommunications and broadcasting including digital cellular telephone services, voice over Internet protocol (VoIP) services, and digital audio broadcasting systems. Speech signals can be coded and transmitted at rates as low as 4 kbit/s with good resulting quality. More general audio signals that include music and other sounds can be coded and transmitted with remarkably high fidelity at rates between 16 and 256 kbit/s per channel. 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, one generally must trade off quality, bit-rate, delay, and complexity. In addition, the robustness of digital coding and transmission algorithms is critical in applications that use lossy channels. Important examples of lossy channels include those provided by wireless systems and those provided by the Internet. The ITS Audio Quality Research Program seeks to identify and develop new approaches that increase quality and robustness or lower the bit-rate, delay, or complexity of digital speech and audio coding and transmission. The ultimate result of such progress should be better sounding, more reliable, more efficient telecommunications and broadcasting services at lower costs.

In most digital speech and audio coding and transmission systems, a set of complex time-varying interactions among signal content, source coding, channel coding, and channel conditions 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. Throughout FY 2003, subjective and objective audio quality testing was conducted to support Audio Quality Research efforts. In addition, customized objective speech quality estimation tools were developed to support other ITS efforts that are focused on the characterization of commercially available communications systems. Some of the laboratory equipment used to in the ITS Audio Quality Research Program is shown in Figure 1 below.

Two additional FY 2003 research efforts are briefly summarized here. In packetized speech transmission systems (the most prominent example is VoIP) transmission delay can vary significantly and rapidly, even within a single spoken phrase. This delay

Figure 1. Some of the laboratory equipment used to support the ITS Audio Quality Research Program (photograph by S. Wolf).

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, and program staff worked to develop an algorithm to do this.

This algorithm must compare the input and output signals of the speech transmission system under test. But many systems of interest will distort speech waveforms, so a conventional waveform correlation solution often fails. The new algorithm uses speech envelopes that are generated by rectifying and low-pass filtering speech waveforms. High frequency information is lost in this stage, but significant robustness to waveform distortion is gained, and speech envelopes have proven very useful for determining coarse estimates of delay. To refine those coarse estimates, the next stage of the algorithm compares speech power spectral densities since these representations retain important properties of the speech signal, even when the waveforms suffer significant distortion. Figure 2 (above right) shows speech waveforms, speech envelopes, and speech power spectral densities for example input and output speech signal segments.

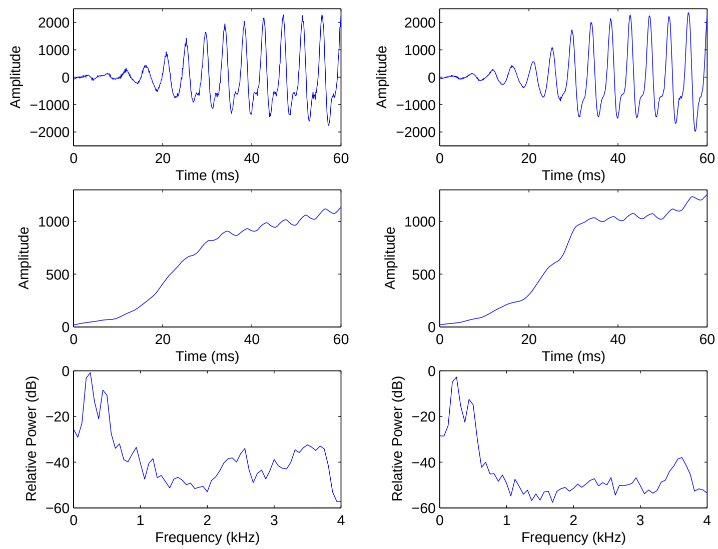

Figure 2. Speech waveforms, envelopes, and power spectral densities (top to bottom) are used to estimate the time-varying delay between a system input (on left) and output (on right).

In a separate effort, program staff worked towards more robust speech coding through the 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 physical or virtual channels. The MDC encoder does not know which of the channels are working and which of the channels have failed at any given time. On the other hand, the MDC decoder will know which of the channels have worked. 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.

Throughout FY 2003, the Audio Quality Research Program staff continued with selective upgrades to the ITS Audio-Visual Laboratories, including the introduction of a 5.1 channel digital audio system. The Audio Quality Research Program continued to transfer technology to industry, Government, and academia throughout FY 2003. Program staff prepared publications, delivered invited lectures and presentations, provided laboratory demonstrations, and completed peer reviews for journals and workshops. More detailed Program results are available at http://www.its.bldrdoc.gov/home/programs/audio/audio.htm

Recent Publications

S.D. Voran, “Channel-optimized multiple-description scalar quantizers for audio coding,” in *Proc. IEEE 10th Digital Signal Processing Workshop*, Pine Mountain, GA, Oct. 2002.

S.D. Voran, “Perception of temporal discontinuity impairments in coded speech – A proposal for objective estimators and some subjective test results,” in *Proc. MESAQIN (Measurement of Speech and Audio Quality in Networks) Conference*, Prague, Czech Republic, May 2003.

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Effects of Radio Channel on Networking Performance

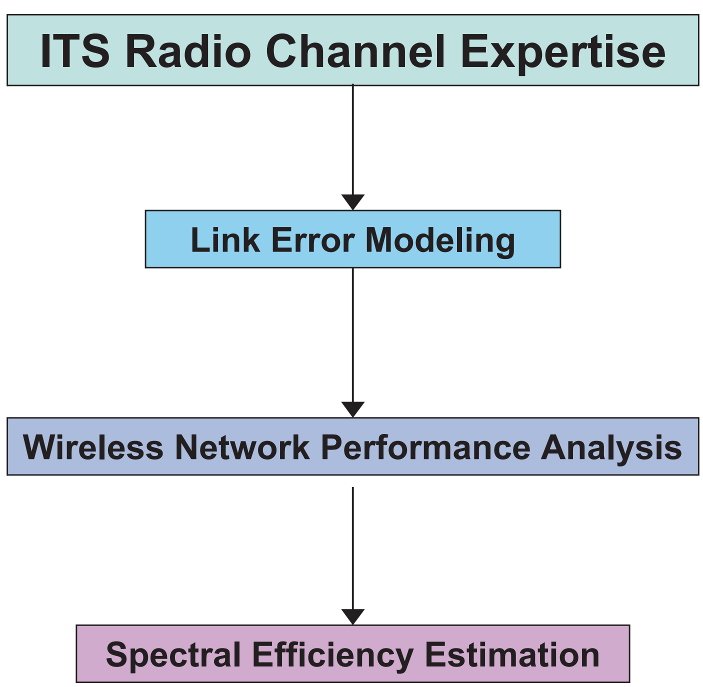
Outputs

- Models of bit, frame, and packet error random processes.
- Quantitative analysis of effects of radio channel on network performance.
- Estimation of the impact the radio channel has 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 communications 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 “Wi-Fi” 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 performance of the network measured in terms of decreased throughput, increased delay, and lost packets. This degradation ultimately limits the usefulness of allocated spectrum.

The Institute is currently striving to translate its radio channel expertise into information that helps designers analyze performance and regulators estimate spectral efficiency of wireless networks. This is being accomplished by focusing on three tasks: (1) accurate modeling of the link error random process resulting from radio channel impairments, (2) investigation of analytic techniques that correlate network performance to radio channel characteristics, and (3) the computation of wireless network spectral capacities which take radio channel impairments into account (see figure below).



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

Previous work included development of a radio link simulator incorporating multipath radio channel impairments and the analysis of bit and frame errors generated by the simulator. This work found that bit and frame errors due to frequency selective multipath had independent, geometrically distributed time intervals. This work is significant because it justifies the use of interval simulation which greatly reduces the computational burden of network simulations. In the past fiscal year we have focused on the statistical analysis of a commonly used radio channel random process, the Rayleigh fading process, in order to better understand statistical characteristics discovered through simulation.

It has been proposed that first-order Markov channel models can be used to adequately predict the behavior of a mobile "Rayleigh" fading channel and hence improve the reliability of mobile radio links. Previous research has addressed this question by applying information theory to the amplitude statistics of a stationary mobile radio channel. This approach required numerical analysis to show that for a particular covariance function and range of relevant parameters (i.e., Doppler frequency, symbol period), the channel is approximately first-order Markov. In our analysis, both amplitude and phase information are used to obtain analytic expressions that can easily be used to determine if a non-stationary arbitrary Rayleigh channel is necessarily first-order Markovian. The analytic results are given in terms of arbitrary covariance functions that can readily be applied to measurements. In particular, our results show that the Rayleigh fading channel is not first-order Markovian. In FY 2004, ITS plans to investigate the impact of this finding on characteristics of the link error processes used in network analysis and simulation.

Also, in the past fiscal year, ITS completed a comprehensive search of professional literature which defines 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. Review of research for two of the tasks, queuing and end-to-end retransmission, pointed to the need for more accurate channel modeling which included the higher-order statistical characterization of bit and frame error processes such as their correlation properties and corresponding power spectral densities.

For example, research into the effects of the radio channel on queuing used power spectral density analysis methods of the queuing process. This analysis is dependent on accurate modeling of the traffic, channel, and server random processes. Spectral analysis showed that queues could absorb rapid channel effects such as fluctuations of signal amplitude due to multipath fading but could not accommodate slower channel effects due to shadowing by a wall, building, or feature of the terrain. Queues overflowed and packets were lost when the power spectral density of the bit or frame error process had high low-frequency energy densities.

Similarly research into the effects of the radio channel on end-to-end retransmission showed that retransmission was beneficial, provided that the sender waited for channel conditions to improve. Two types of channel correlation properties are needed to analyze this: first, the correlation of the bit error process to determine when a link may be preventing end-to-end transmission, and second, the correlation of the radio channel to determine appropriate retransmission time-out thresholds. Research is quick to point out that the intimate relationship between end-to-end retransmission and network congestion control procedures, which assume any retransmission is due to congested switch queues, complicates this issue further.

At this time there is a limited set of analytic solutions that correlate the effect of a radio channel characteristic (such as Doppler frequency) on a networking task (such as queuing) through some network performance measure (such as throughput) for small networks. In FY 2004 these solutions will be evaluated to determine their usefulness in translating ITS radio channel measurements to network performance measures for analysis of queuing, routing, and end-to-end transmission methods.

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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 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.

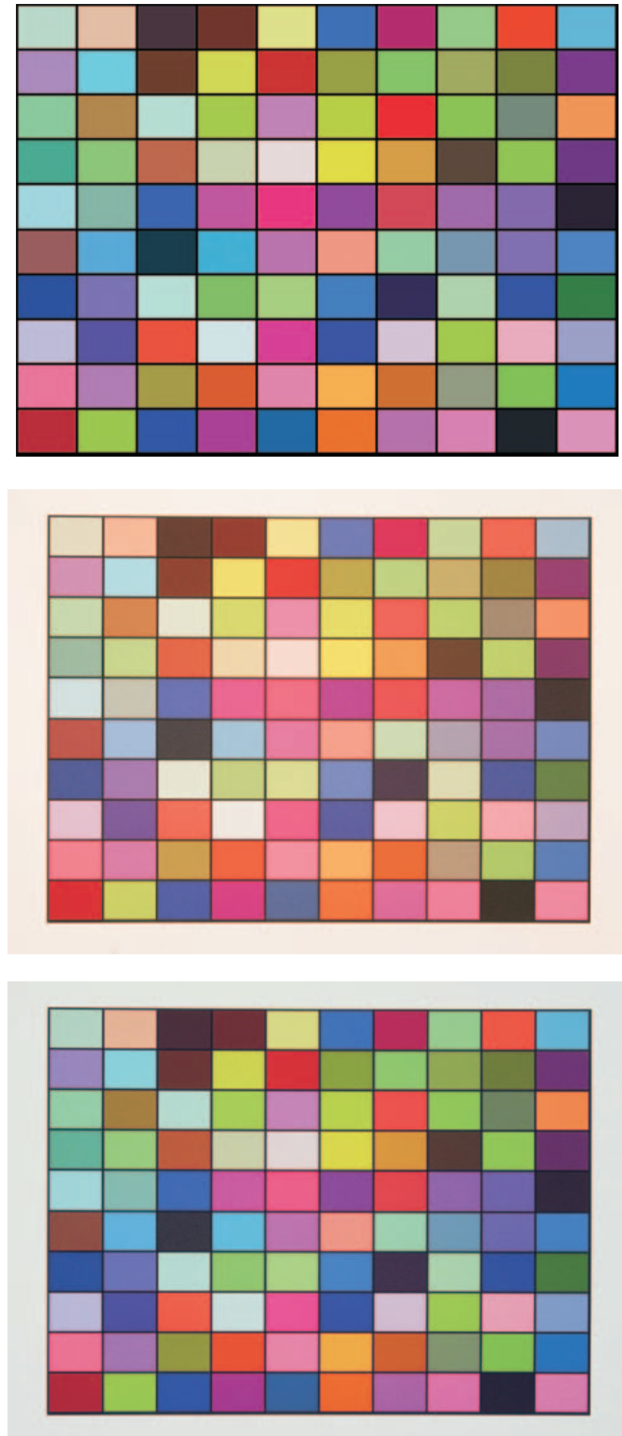
To be accurate, digital video quality measurements must be based on perceived "picture quality" and must be made in-service. This is because the performance of digital video systems is variable and depends upon the dynamic characteristics of both the input video and the digital transmission system. To 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, been accepted for inclusion in two International Recommendations, and are currently being used by hundreds of individuals and organizations worldwide.

The Video Quality Research Project accomplished several highly significant milestones in FY 2003. An ITS-developed video quality metric (VQM) was prepared and submitted to the ITU Video Quality Experts Group (VQEG) for independent testing and verification. A total of 8 international VQM proponent submissions were evaluated for two different video tests (525-line U.S. video standard and 625-line European video standard) during the Dec. 2002 to Feb. 2003 time frame. Impaired video files (unknown to all proponents) were processed through the ITS VQM software and the results were returned to VQEG for independent analysis. Of the 8 submissions, the ITS VQM submission was the clear "winner" of the competition. The ITS submission was the only VQM in the top performing group for both the 525-line and 625-line video tests. For the U.S. standard 525-line video test, the ITS VQM achieved a correlation coefficient to the subjective data of almost 95%, near the theoretical limit. These test results are even more remarkable because the ITS VQM submission was a "reduced-reference" measurement system, whereas the other proponents submitted "full-reference" measurement systems. The ITS VQM submission only requires approximately 1/100 of the reference data to make a measurement. As a result of these international achievements in 2003, the ITS VQM was standardized by ANSI in July 2003 (ANSI T1.801.03-2003). ITU-T Study Group 9 and ITU-R Working Party 6Q have also included the ITS VQM in their upcoming Draft Recommendations that will be finalized next fiscal year. To assist companies and potential licensees in the deployment and use of the patented ITS VQM technology, evaluation software that implements the above national and international standards was posted on the ITS web site. Since the evaluation software was posted, nearly 200 U.S. and 100 international individuals and companies have downloaded it. ITS staff members received a Department of Commerce Silver Medal for this work.

Significant progress was also made in FY 2003 in the following areas: developing and validating Single Stimulus Continuous Quality Evaluation (SSCQE) subjective testing methods, developing methods to combine multiple subjective data sets into one large coherent data set (required to effectively utilize ITS's massive subjective data base for

VQM development), construction of a subjective and objective high definition television (HDTV) laboratory, and development of color calibration algorithms for digital still and video imaging systems. Since the limited space provided here is not sufficient to describe these research activities, the reader is encouraged to examine the publications below for further details.

The figure (right) demonstrates application of the color correction algorithms that were developed to remove linear and non-linear color distortions. The linear correction involves the use of a color correction matrix that allows each color component in the corrected image (e.g., red) to be calculated as a linear summation of a DC component and all the color components in the uncorrected image (e.g., red, green, and blue). This algorithm can correct for color distortions that are more complicated than a simple gain and DC shift in each of the color components.



Demonstration of color correction: Original (Top), Camera (Middle), Calibrated (Bottom).

Recent Publications

ANSI T1.801.03-2003, “Digital Transport of One-Way Video Signals — Parameters for Objective Performance Assessment.”

M. Brill et al., “Accuracy and cross-calibration of video quality metrics: New methods from ATIS/T1A1,” *Signal Processing: Image Communication Journal*, special issue on Video Quality, Nov. 2003.

M. H. Pinson and S. Wolf, “Comparing subjective video quality testing methodologies,” in *Proc. SPIE Video Communications and Image Processing Conference*, Lugano, Switzerland, Jul. 2003.

M. H. Pinson and S. Wolf, “An objective method for combining multiple subjective data sets,” in *Proc. SPIE Video Communications and Image Processing Conference*, Lugano, Switzerland, Jul. 2003.

S. Wolf, “Color correction matrix for digital still and video imaging systems,” NTIA Technical Memorandum TM-04-406, Dec. 2003.

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

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