Telecommunications Theory

The Institute is involved in research in both wireless and wireline telecommunications. The rapid growth of telecommunications in the last 50 years has caused crowding in the radio spectrum. New technology requires a new understanding of the behavior of radio waves in all parts of the radio spectrum. The Institute studies all frequencies in use, extending our understanding of how radio signal propagation is affected by the earth's surface, the atmosphere, and the ionosphere.

This work is resulting in new propagation models for the broadband signals used in new radio systems.

The Institute's historical involvement in radio-wave research and propagation prediction development provides a substantial knowledge base for the development of state-of-the-art telecommunication systems. In another research area, the Institute develops perception-based quality measures for multimedia services.

ITS transfers all of these concepts and technology to both public and private users, where knowledge is transformed into new products and new opportunities.

Areas of Emphasis

Adaptive 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.

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.

Radio Channel Effects on Networks
The Institute, a recognized leader in radio channel measurement and modeling, is involved in research to assess the effects of the wireless communication channel on communications system 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.

Wireless Propagation Research
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 wireless (3G). Projects are funded by NTIA and DoD.

Advanced Antenna Testbed

Outputs

- Analysis of some proposed antenna array comparison data.
- 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 (WLAN's) 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 omni-directional 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 arrays used for MIMO testing at the NIST open area test site (photograph by P. Papazian).

A recent ATB application is a 16-element multiple input, multiple output (MIMO), experiment conducted in FY 2001. Two 16-element MIMO arrays were fabricated and tested and then deployed at the NIST open area test site, as shown in Figures 1 and 2. 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 (Papazian et al. 2002, see **Recent Publications** below).

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

ρ = signal to noise ratio

H = complex transmission matrix

H^+ = hermetian transpose of H

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. 67, for more information about the ATB.)

Figure 2. Dipole antenna elements used by MIMO array (photograph by P. Papazian).

Recent Publications

P. Wilson, P. Papazian, and Y. Lo, "A comparison of 1920 MHz mobile channel diversity gain using horizontal and vertical arrays," *IEEE Trans. on Communications*, vol. 49, no. 12, pp. 2068-2070, Dec. 2001.

P. Papazian, M. Gans, Y. Lo, and R. Dalke, "Capacity measurements for a 16x16 BLAST array over a conducting ground plane," in *Proc. IEEE Fall VTC 2002*, Vancouver, B.C., Canada, Sep. 2002.

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Audio Quality Research

Outputs

- Technical publications documenting new research results.
- Subjective measurements and objective estimates 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 transmitted as data packets, thus sharing channel capacity (e.g., 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 bit-rate, delay, or complexity of digital speech and audio coding and transmission. The ultimate result of such progress is 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 subjects' opinions of the speech or audio quality of various coding and transmission systems. Subjects provide their opinions through electronic forms, an example of which appears in Figure 1 above.

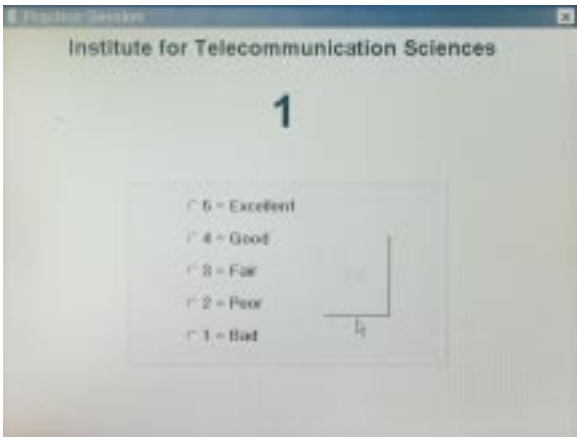

Figure 1. Example electronic response form used in subjective testing.

The Program has developed, verified, and patented tools for the objective estimation of telephone bandwidth speech quality. An example screen from one ITS-developed software tool is shown in Figure 2.

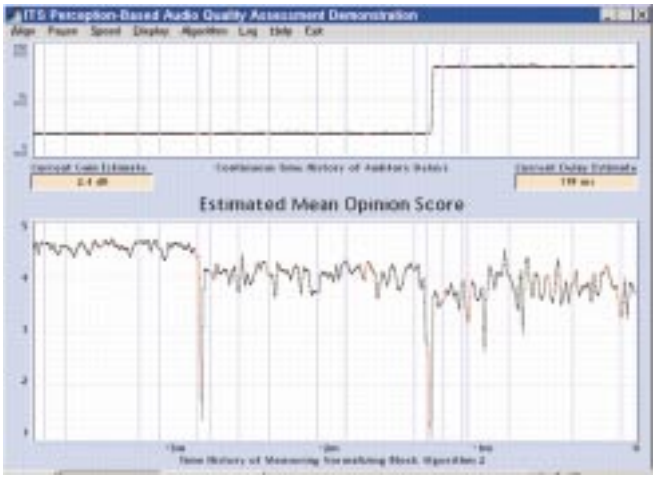

Figure 2. Example screen from ITS-developed software tool for objective estimation of perceived speech quality.

Throughout FY 2002, Audio Quality Research Program staff continued to apply and support such tools. Staff applied software tools, equipment and expertise in experiments aimed at finding relationships between the quality of speech delivered by VoIP systems and underlying network conditions.

Program staff also investigated several potential new innovations in speech and audio coding. In one investigation, the concepts of linear algebra were applied to time-frequency representations in order to generate novel decompositions of audio signals. These new decompositions display some desirable quantization properties and may ultimately lead to more efficient speech or audio coding schemes. Figure 3 uses 19 different colors to show how each of the 1024 different signal components in the decomposition can be linked to one of 19 fixed quantizers. The result is nearly optimal quantization of all 1024 signal components.

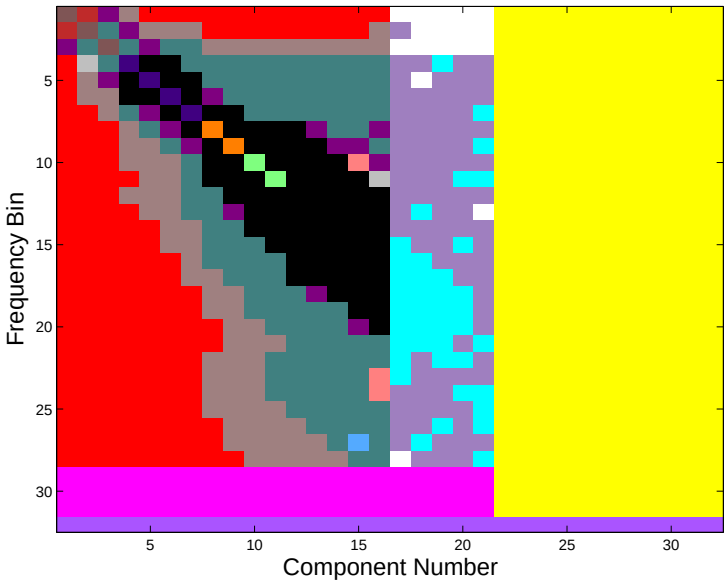

Figure 3. Example quantizer map for coding technique under study: audio signal has been decomposed into 1024 components, each component has been assigned to one of 19 quantizers indicated by one of 19 different colors.

The Audio Quality Research Program recently responded to a specific inquiry regarding the use of speech and audio coding algorithms for signals other than those they were designed for. For example, if a speech coding algorithm is given the sounds of laughter, or the sounds of a train, how badly will it distort each of these sounds? To answer the broader question in a concrete way, Program staff generated over 700 sound files and a set of tables linking to those files. Through these tables, a user can select example sounds and coders and hear first hand what a typical result might sound like.

Also in FY 2002, the Audio Quality Research Program staff performed significant upgrades to the ITS Audio-Visual Laboratories. An ad-hoc network of digital interconnections was replaced with a single unified digital audio infrastructure based on a 16 by 16 digital audio routing switcher. The Audio Quality Research Program continued to transfer technology to industry, Government, and academia throughout FY 2002. 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, “Estimation of system gain and bias using noisy observations with known noise power ratio,” NTIA Report 02-395, Sep. 2002.

S.D. Voran, “An iterated nested least-squares algorithm for fitting multiple data sets,” NTIA Technical Memorandum TR-03-397, Oct. 2002.

S.D. Voran, “Compensating for system gain: Motivations, derivations, and relations for three common solutions,” NTIA Technical Memorandum TR-03-398, Oct. 2002.

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

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Effects of Radio Channel on Networking Performance

Outputs

- Quantitative analysis of effects of radio channel on network performance.
- Bit, frame, and packet error wait time statistics.

The Institute is a recognized leader in radio channel measurement and modeling. Such knowledge is essential in the development of new, spectrally efficient radios which will operate in radio channels degraded by multipath, man-made noise, and interference. This is especially true for the mobile radio channel whose degradations vary with time. For example, development of new adaptive equalizers for modern, wide-bandwidth mobile radios would not be possible without radio channel measurement and modeling.

Systems using mobile radios to access the Internet are proliferating. IEEE 802.11 “Wi-Fi” wireless local area network (WLAN) and personal communications services (PCS) general packet radio service (GPRS) are but two examples. To reach their fullest potential these systems must reach large numbers of people and have correspondingly high bandwidth efficiencies per unit area, i.e., spectrum efficiency. In the past, enormous gains in spectrum efficiency have been found through advances in cellular frequency reuse and multiple access methodologies. In the future, equally significant gains may be found in novel queuing, routing, and retransmission algorithms.

These gains are unlikely to be realized without knowledge of the radio channel obtained through radio channel measurement and modeling. Currently, the Institute is promoting the use of this knowledge in two ways. First, it is conducting a comprehensive literature search to determine the scope of the effects of the mobile radio channel on networking tasks. Thus far, the literature search indicates that networking tasks are as sensitive to second-order statistics such as the distribution of wait time between packet errors as they are to the commonly studied first-order statistics such as packet error rate. Second, the Institute is developing methods for extracting these statistics from radio channel measurements and models.

The Gaussian Wide Sense Stationary Uncorrelated Scattering channel presented by Hufford* is often used to model the mobile radio channel. This model describes archetypal regions, i.e., urban and in-building regions, with statistically varying direct-path and multipath components. The multipath component is composed of an exponentially decaying continuum of indirect paths characterized by strength (relative to the direct component) and time constant parameters. Figure 1 shows a single realization or “snapshot” of the impulse response amplitude from an urban mobile radio channel where the strength and time constants are -5 dB and 0.5 microseconds, respectively.

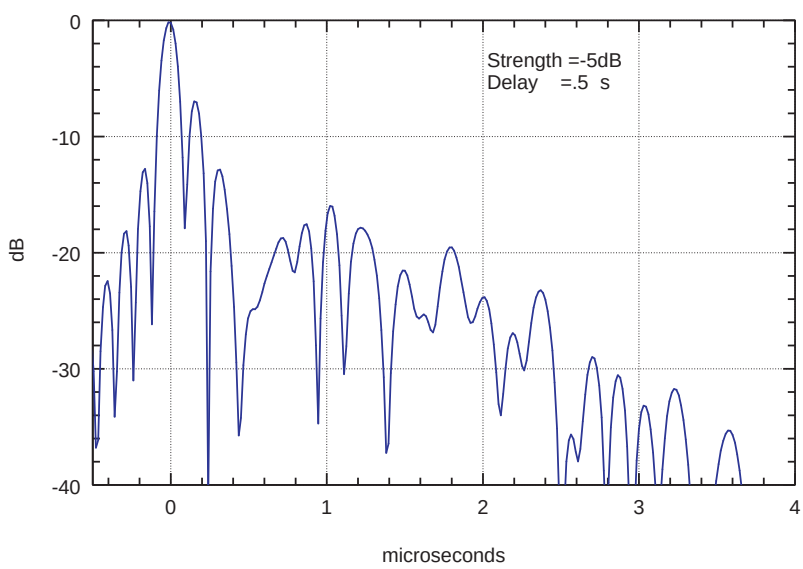

Figure 1. Radio channel impulse response amplitude.

*G. Hufford, “A characterization of the multipath in the HDTV channel,” *IEEE Trans. on Broadcasting*, vol. 38, no. 4, pp. 252-255, Dec. 1992.

It was theorized that independent and identically distributed bits transmitted over this channel would result in exponentially distributed wait times between bit errors. This hypothesis is confirmed by simulations. As an example, Figure 2 shows the complementary cumulative distribution of the wait times between bit errors which were determined from simulations of a binary phase shift keyed (BPSK) signal transmitted at 6 million bits per second at a 0.0043 bit error rate.

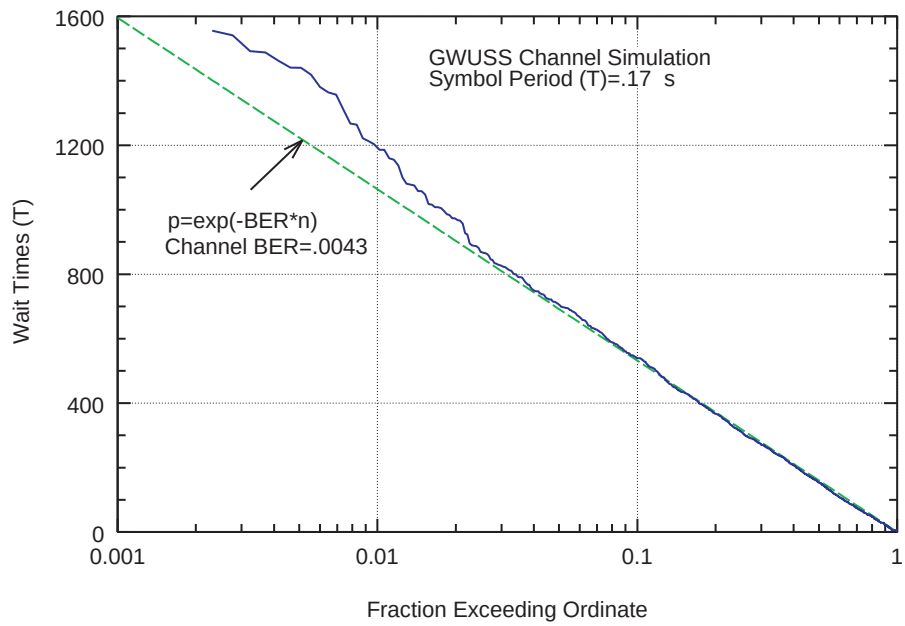

Figure 2. Complementary cumulative distribution function of bit error wait times for radio channel depicted in Figure 1.

The distribution is plotted on a log-linear graph where an exponentially distributed random variable is displayed as a negatively sloped line. The ordinate is the wait time between bit errors in units of symbol periods. The abscissa is the fraction of time the ordinate is exceeded. The mean wait time is the inverse of the bit error rate or 232.5 symbol periods. The graph shows that the wait time between bit errors exceeds 1071, 535, and 161 symbol periods for 1, 10, and 50 percent of the time, respectively.

Packet wait times are also exponentially distributed. For the above example, a packet 207 bytes long, 20 of which are error correction bytes capable of correcting 10 byte errors, would be transmitted at a rate of 3623 packets per second. The corresponding byte and packet error rates are 0.034 and 0.1, respectively. The mean wait time between packet errors is the inverse of the packet error rate or 10 packet periods. Wait time between packet errors exceeds 46, 23, and 7 packet periods for 1, 10, and 50 percent of the time, respectively.

Results such as these are immediately useful to network designers. For example, since the packet error rate is high, the network designer may ask for a lower BER, drop queued packets from this link first when switches are congested, reroute packets away from this link if possible, or deny retransmission of packets from this link.

Ultimately, results such as these will benefit the public and private sectors by facilitating the integration of new wireless links into existing legacy networks, reducing the proliferation of redundant wireless network protocols, and decreasing the pressure for additional spectrum by using existing allocations more efficiently. These benefits will become more meaningful as the convergence of voice and data as well as fixed and mobile users onto the Internet continues.

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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.

Digital video systems are creating many new telecommunication services and industries (e.g., direct broadcast satellite, digital television, high definition television, video teleconferencing, telemedicine, e-commerce) that are becoming an essential part of the U.S. and world economy. Objective metrics for measuring the performance of these systems are required 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 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 developed a new measurement paradigm that is based upon extraction and comparison of low bandwidth perception-based features that can be easily communicated across the telecommunications network. This new measurement paradigm has received three U.S. patents, been adopted as an ANSI standard, and is being used by organizations worldwide. Figure 1 demonstrates the ability of the ITS measurements to perform continuous in-service quality monitoring. This graph shows an example time history of single stimulus continuous quality evaluation (SSCQE) data from an MPEG-2 test and corresponding quality predictions from a 10 kbit/s reduced reference quality monitoring system (i.e., the extracted features have 10 kbit/s of total bandwidth). A 100 on the scale is excellent quality while a 0 is bad quality. During FY 2002, two SSCQE experiments were conducted and this subjective data was used to optimize the ITS metrics for an automatic quality monitoring of MPEG-2 digital television systems.

Also during FY 2002, ITS completed development of a UNIX-based and Windows®-based automated video quality measurement (VQM) tool. This VQM tool can be downloaded from the Video Quality Research home page given below. Version 2.0 of the VQM software includes detailed documentation of the technical algorithms, a user's manual, and test video sequences to verify proper software installation. The VQM software performs all necessary processing steps on the sampled input and output video streams, including:

1. Calibration. A video system may introduce spatial shifts of the picture (both horizontal and vertical), a reduction of the picture area (valid region), changes in gain (i.e., contrast) and offset (i.e., brightness), and temporal shifts of the video stream (i.e., video delay). The process of estimating and removing these four items from the destination

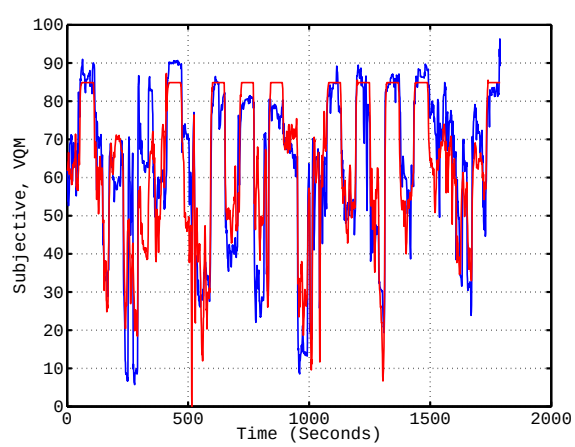

Figure 1. SSCQE subjective data (blue) and corresponding objective quality predictions (red).

video stream is referred to as calibration or normalization. Many deployed video systems have been observed to have poor system calibration. If not properly accounted for, this can create erroneous results with very poor correlation to subjective quality ratings.

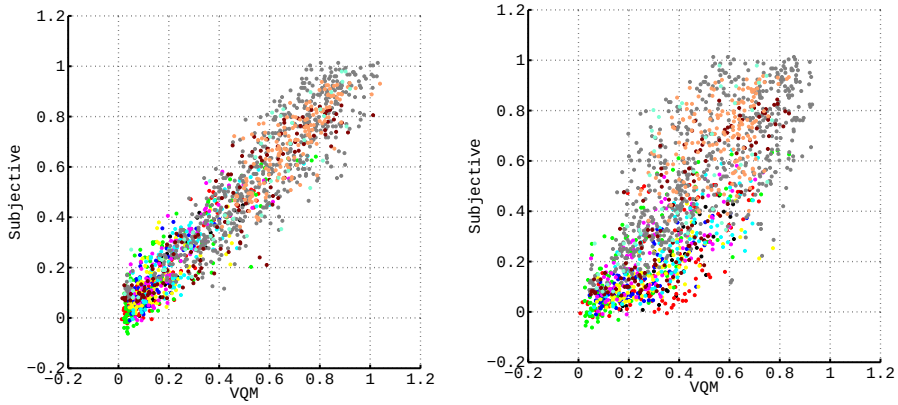

Figure 2. Performance of two video quality mappings: General Purpose – left, PSNR– right.

2. Feature Extraction. The input and output video streams are divided into spatial-temporal (S-T) regions from which features, or summary statistics, are extracted that quantify the perceptual aspects of video quality. Two types of features are extracted from the luminance signal Y: features based on spatial gradients and features based on temporal gradients. Features derived from spatial gradients are used to characterize perceptual distortions of edges and detail, while features derived from temporal gradients quantify distortions in the flow of motion. Some coding and transmission impairments create special color impairments (e.g., a colored error block, localized color shifts). A useful feature to quantify these types of impairments is based on the mean two-dimensional chrominance vector (Cb, Cr) computed over each S-T region.

3. Distance Measures. The perceptual impairment at each S-T region is calculated using functions that model visual masking of the spatial and temporal impairments. Loss and gain are normally examined separately, since they produce fundamentally different effects on quality perception (e.g., loss of spatial detail due to blurring and gain of spatial noise due to blocking). Of the many comparison functions that have been evaluated, two forms have consistently produced the best correlation to subjective ratings. These visual masking functions imply that impairment perception is inversely proportional to the amount of localized spatial or temporal activity that is present.

4. Quality Mapping. A video quality mapping, or model, is a particular algorithm that is used to combine a set of distance measures into an overall estimate of quality. The quality mapping can depend upon many things, including the type of subjective experiment and video application. Currently, fifteen subjective data sets that span a very wide range of scenes and video systems are used as truth data for the development and testing of objective video quality mappings. The VQM software that is available for download implements five different quality mappings. Figure 2 gives the performance of two of these quality mappings: a “General Purpose” mapping using ITS-developed metrics and a traditional Peak Signal to Noise Ratio (PSNR) mapping. Eleven data sets are shown where each data set is plotted in a different color. The Y-axis is the actual subjective score while the X-axis is the quality mapping. The data has been scaled such that “0” is mapped to no perceived impairment and “1” is mapped to nominally maximum perceived impairment. The correlation coefficients for the left and right scatter plots are 0.95 and 0.78, respectively.

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

Recent Publications:

S. Wolf and M. Pinson, “Video quality measurement techniques,” NTIA Report 02-392, Jun. 2002

M. Pinson and S. Wolf, “Video quality measurement user’s manual,” NTIA Handbook 02-01, Feb. 2002.

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Wireless Propagation Research

Outputs

- Survey and implementation of data analysis and modeling techniques for multipath channel estimation.
- Methods for the prediction and measurement of wireless indoor radio channel characteristics.
- Assessment of geometric optics approximation for indoor ray-trace models.

The Institute has been involved in research efforts related to wireless communication applications and theory. More specifically, ITS develops models and measurement systems to estimate propagation characteristics of various multipath environments. The objectives of these efforts are to support new wireless technology development and help U.S. industry compete in the worldwide telecommunications marketplace. In the past, a majority of the wireless research has been related to the outdoor propagation environment. Recently, with the emergence of new indoor wireless local area networks and wireless local campus networks, the research has been extended to indoor scenarios.

ITS is able to measure and model the transfer function of a wide variety of wireless radio links. The data is non-deterministic and there is significant complexity in reducing that data into a concise set of metrics with limited ambiguity. Signal attenuation is a performance metric that can be expressed statistically and straightforwardly. However, time-dispersion parameters that reflect the bandwidth limitation imposed by the channel are not so clear cut. Historically, researchers have used RMS (root mean square) delay spread as a dispersion metric because of the derived correlation with bit error ratio (BER) for channels with Gaussian, wide-sense stationary, uncorrelated-scattering characteristics. There are a number of adverse wireless radio environments, however, where the frequency-domain statistics are dynamic over small distances and the justification for using rms delay spread is compromised.

Over the course of the last year an extensive literature search was conducted on characterization for wireless channels. The resulting article database provided numerous processing and modeling techniques for consideration. Those procedures were incorporated into a channel characterization toolbox for MATLAB, which has proved to be a useful tool for system evaluation, data comparison, statistical analyses, model development, presentation of results, and distribution of data.

In an attempt to understand antenna polarization and directivity effects indoors, impulse response measurements were acquired in a number of scenarios (e.g., in-room, in-corridor, obstructed line-of-site) employing different types of antennas. An effective method of presentation for this data is shown in Figure 1. The left plot shows time dispersion versus excess loss; this provides a clear comparison between different channels and demonstrates a correlation between the two metrics. The right plot is the cumulative distribution function of maximum delay on Gaussian paper, which provides an intuitive view of the distribution for that random variable.

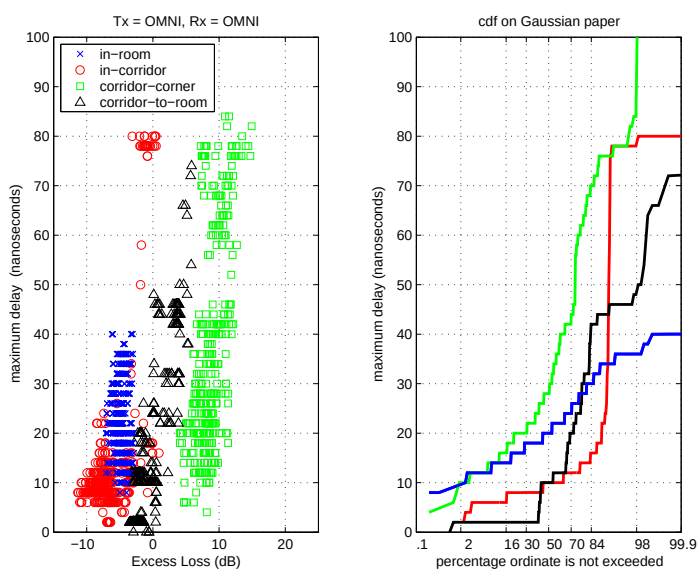

Figure 1. Maximum delay versus excess loss from four canonical indoor scenarios measured with linearly polarized omnidirectional receive and transmit antennas.

Figure 2 statistically demonstrates the different effects of the channels in the frequency domain.

The Institute has developed a geometric optics (or ray-tracing) model for calculating the field strength and impulse response of an indoor radio propagation channel, characterizing an anechoic chamber, and analyzing the coupling mechanisms between rooms. In an attempt to assess the validity of the ray-trace model, we have investigated the accuracy of some of the inherent assumptions. Using the exact Sommerfeld formulation for a source above a dielectric half space, a thorough investigation into the geometric optics (GO) approximation was performed. This study demonstrated discrepancies associated with surface-wave and near-field effects and the use of plane-wave Fresnel reflection coefficients, as is common in ray-trace models. Figure 3 shows fields from an elementary horizontal dipole close to a dielectric surface calculated from the GO approximation (with and without the Norton surface-wave term added) and numerical evaluation of Sommerfeld's formulation. A discernable pseudo-lateral wave phenomenon was identified that produces an interference pattern in the Sommerfeld solution with respect to the GO plus Norton term approximation at relatively high frequencies when the source and observation points are near the surface.

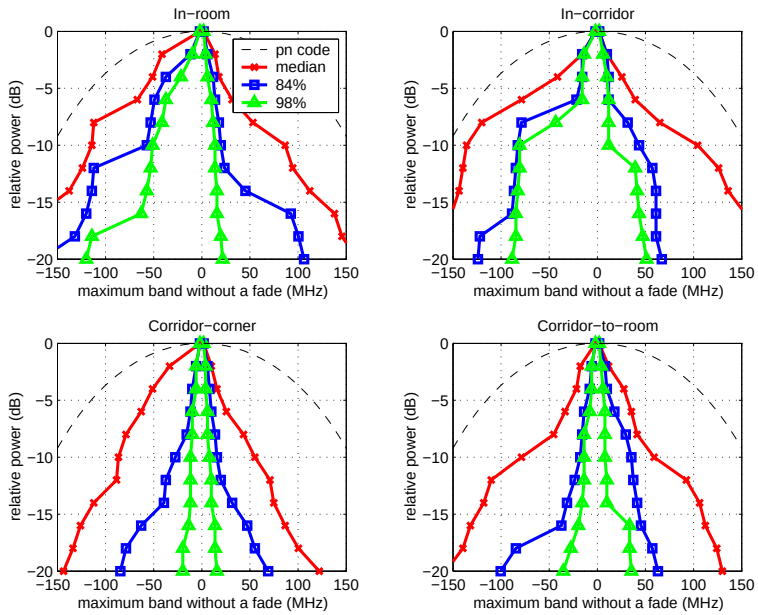

Figure 2. Channel bandwidth statistics from four canonical indoor scenarios measured with linearly polarized omnidirectional receive and directional transmit antennas.

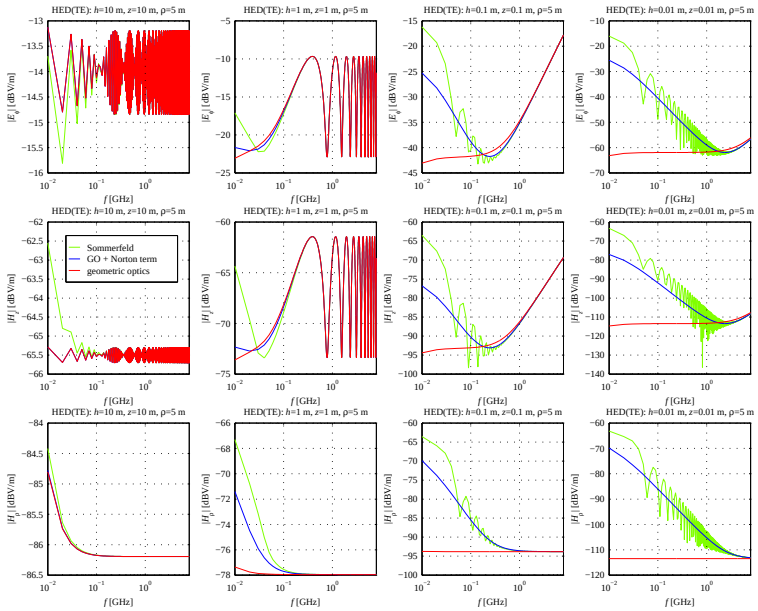

Figure 3. Near-surface effects on field strength of an elementary horizontal electric dipole above a concrete half space.

Recent Publication

M.G. Cotton, E.F. Kuester, and C.L. Holloway, "An investigation into the geometric optics approximation for indoor propagation models," *Radio Science*, vol. 37, no. 4, pp. 1-1 – 1-22, Jul.-Aug. 2002.

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