Telecommunications Theory

The world's telecommunications network of terrestrial and undersea cables, and terrestrial and satellite radio systems is undergoing explosive growth, in terms of both the build-out of hardware infrastructure and ever-increasing loading demands. Wireline and radio networks complement each other, but whereas wireline capacity can be expanded almost indefinitely, radio spectrum is a limited resource. In response to increasing demand for radio spectrum capacity, new technologies are being developed and implemented to use spectrum more efficiently and effectively. Also, the basic paradigm of radio spectrum management is moving away from traditional, top-down frequency-assignment methods and toward autonomous, interference-limited technologies. Historically rigid spectrum band allocations on a service-by-service basis are being supplanted by more sharing of radio bands by multiple services. The new sharing schemes emphasize new capabilities in radio systems to recognize and avoid interference. But to fulfill the promise of these interference-limited schemes, the effects of noise and interference on radio receiver performance must be thoroughly understood, and such knowledge must be focused on improvements in the performance of both existing and new networks. Tools to monitor the quality of audio and video information on communication channels also must be developed and used so that audio and video quality levels can be accurately adjusted in real-time to achieve maximal quality with minimal use of available bandwidth.

To achieve these goals for the U.S. government as well as the private sector, the Institute's Telecommunications Theory Division performs research in both wireless and wireline telecommunications, seeking to understand and improve telecommunications at the most fundamental levels of physics and engineering. Strong ongoing investigations are being maintained in the major areas of broadband wireless systems performance in the presence of interference; the development of new propagation models for short-range mobile radio systems; the effects of noise and interference as critical limiting factors for advanced communication systems; automated tools for assessing audio and video quality; and the further development of advanced spectrum sharing concepts such as dynamic frequency selection.

Through technical publications, cooperative research and development agreements (CRADAs), and interagency agreements, ITS transfers the results of its work in all these technology areas to both the public and private 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.

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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.

Effects of the Channel on Radio System Performance The Institute, a recognized leader in radio channel measurement and modeling, is researching the effects of interference and noise on the performance of radio receivers and networks. Current work is focused on the effects of noise and interference as limiting factors in system performance. The project is funded by NTIA.

1.7-GHz Advanced Wireless Systems Compatibility Testing The Institute performed electromagnetic compatibility testing between newly developed advanced wireless systems (AWS) and Government communication systems. The goal of the testing was to determine the extent to which these two types of systems might be able to share spectrum at 1.7 GHz temporarily. This project was funded by NTIA.

Signal Characteristics, Spectrum Emissions, and Interference Analysis The Institute characterizes the effects of interference from radio transmitters to victim radio receivers. This work has become critically important as more emphasis is placed on radio band-sharing schemes in which services considered mutually incompatible are supposed to share spectrum on an interference-limited basis. The project is funded by NTIA.

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

Audio Quality Research

Outputs

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

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

Digital coding of speech and audio involves compromises and trade-offs among at least five basic factors: signal quality, transmission bit-rate, robustness to transmission errors and losses, coding and transmission delay, and coding and transmission algorithm complexity. More complex encoding schemes can generally encode a given signal into fewer bits. But when a signal is described by fewer bits, each bit is inherently more critical to the overall description of the signal. That is, the data stream is less robust to errors and losses. The trade-off between delay and robustness can be seen in the issues of receive-side buffering of data packets delivered by a best-effort network. Increasing the size of a buffer can reduce the number of packets lost (desirable), but this will also increase the algorithm delay (undesirable). For a complete system of coding and transmission, all five factors will generally come into play. For any given application, joint optimization with respect to these five factors can be an attractive but elusive goal.

The ITS Audio Quality Research Program identifies, develops, and characterizes new techniques that will increase relative quality or robustness or will reduce relative bit-rate, delay, or complexity of speech and audio coding and transmission algorithms. In addition, the Program seeks to develop techniques that aid in attaining a desired optimum balance between the five factors. Perhaps the most difficult of the five factors to characterize is quality. Thus a key component of our efforts is the development of more effective and efficient ways to characterize speech and audio quality. The ultimate result of these efforts would be better sounding, more reliable, more efficient telecommunications and broadcasting services.

In FY 2007 Program staff investigated the relationship between two speech quality estimation algorithms as part of a project aimed at the development of a unified speech quality scale. Staff applied software simulations of 62 different speech coding and transmission scenarios to 128 digital speech recordings resulting in a set of 7936 data points. Figure 1 shows the relationship between two speech quality scales for these data points graphically, and a mathematical relationship, described by the solid line, can be extracted. Some clustering of data points can be seen in Figure 1, which motivated further investigation. Staff then separated the data points into two classes according to the type of filtering applied

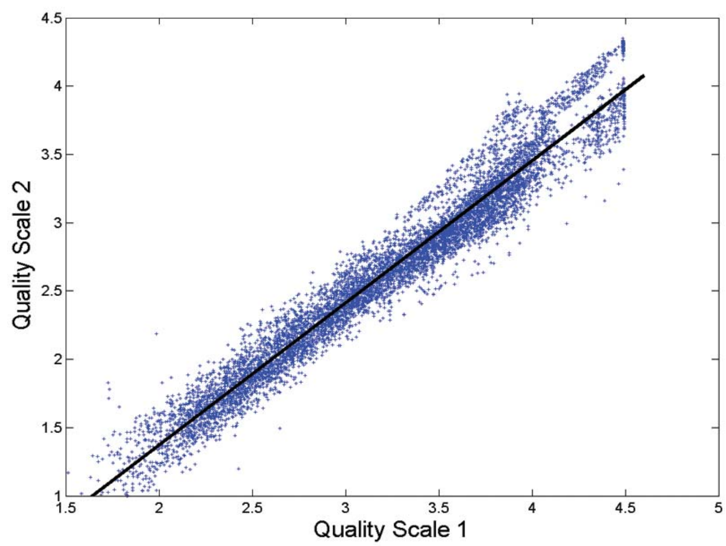

Figure 1. Graphical relationship between two speech quality scales for 7936 data points. Mathematical relationship is represented by the solid line.

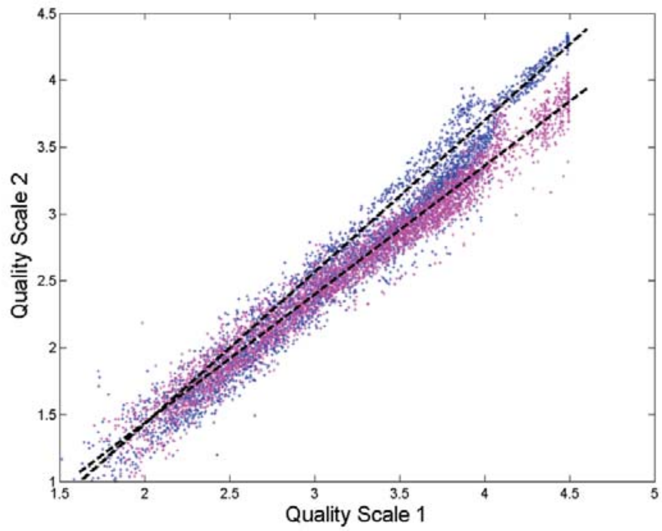

Figure 2. Data points of Figure 1 divided into two classes according to speech filtering. Now two separate mathematical relationships, represented by the broken lines, are evident.

to the speech signal. These two classes are displayed in two different colors in Figure 2. Here it is apparent that indeed, two separate mathematical relationships exist for the two classes. These relationships are described by the two broken lines in Figure 2. These results allow for a more refined and accurate approach to the unification of the two scales.

Program staff carried out significant investigations aimed at high quality audio coding at reduced bit rates. The extreme in high quality coding is lossless coding, where a bit-exact copy of the original signal is reproduced at the decoder output. Staff considered numerous alternatives to the standard transformation-based and prediction-based algorithms. Staff also developed a scheme where multiple lossless layers can be transmitted, thus allowing an efficient decomposition and reconstruction of an audio signal into the lossless representations associated with multiple related sample rates. Staff also continued with prior work related to quantization — one of the key elements in many digital speech and audio processes. This work addresses increased efficiency in quantization as well as the reduction of quantization noise.

In response to a set of important open questions, project staff developed, implemented, and verified a new subjective testing protocol. Those questions concern the extent to which various speech communication links can convey or obscure the emotion in a talker's voice, and how one would one test a speech communication link for its ability to preserve emotion. Project staff created a set of speech recordings that contained both real and dramatized emotions and set out a protocol in which listeners heard these recordings through various speech communication links in a controlled and balanced way. Listeners had the task of detecting one of two emotional states in each recording. Listeners also participated in an additional protocol that tested word intelligibility in a sentence context. The detection results and the intelligibility results were then jointly analyzed in order to identify relationships that shed light on the questions above.

Throughout FY 2007, Program staff continued with significant speech quality testing using both objective and subjective techniques. These tests support both this program and other ITS programs. Numerous laboratory upgrades, including high resolution recording systems, were accomplished throughout the year. Staff continued to transfer program results to industry, Government, and academia by means of technical publications, lectures, laboratory demonstrations, and poster presentations. Staff also completed a significant number of peer reviews and associate editor functions for technical journals and conference proceedings. Program staff supported telecommunications standards development through research efforts and technical exchanges. Program publications, technical information, and other program results are available at http://www.its.bldrdoc.gov/audio.

Recent Publication

S.D. Voran, "Lossless audio coding with bandwidth extension layers," in *Proc. 2007 Workshop on Applications of Signal Processing to Audio and Acoustics*, New Paltz, New York, Oct. 2007.

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Effects of the Channel on Radio System Performance

Outputs

- Interference susceptibility analysis.
- Translation of channel measurements directly to receiver performance metrics.
- Correlations between channel characteristics and receiver performance.

Telecommunications play a vital role in providing services deemed essential for modern life. Many of these services use radio links composed of a transmitter and receiver or *radio system* and the *channel* separating the two. The channel is often the primary impediment to fast and reliable radio system performance. Understanding the channel and the effects of the channel on radio system performance are crucial to the advancement and regulation of telecommunications. ITS has historically directed considerable research towards channel characterization. Currently, ITS is developing a radio link model with computer simulation to study the effects of the channel on radio system performance.

The channel degrades radio system performance by introducing multipath and undesired signals. Multipath filters the radio signal. The filtering is due to reflection, diffraction, and scattering propagation phenomena supported by the channel media which can be as diverse as the earth's ionosphere or an urban landscape. In sufficiently narrow bandwidths this filtering is simply attenuation. The addition of undesired signals obscures the radio signal. Undesired signals include information bearing signals from other radio systems (e.g., interfering signals), natural noise created by phenomena such as atmospheric lightning, and man-made noise created by electrical devices such as automobile ignition systems and power lines.

In the past several years, spectrum managers have been evaluating new spectrum management approaches such as ultrawideband modulation and dynamic frequency selection which allow new radio systems to share spectrum occupied by older, legacy radio systems. Advocates of these new spectrum management approaches often assume that signal processing algorithms in the legacy radio systems, such as error correction, will mitigate the effects of the undesired signal from the new radio system. Others maintain that the signal processing algorithms were designed to mitigate channel characteristics different than those created by the new systems and are unlikely to be effective.

ITS has applied its extensive experience in undesired signal characterization to this problem. In addition, it has developed hardware test beds with off-the-shelf legacy radio equipment from satellite digital television, global positioning satellite, and land mobile radio systems to measure interference

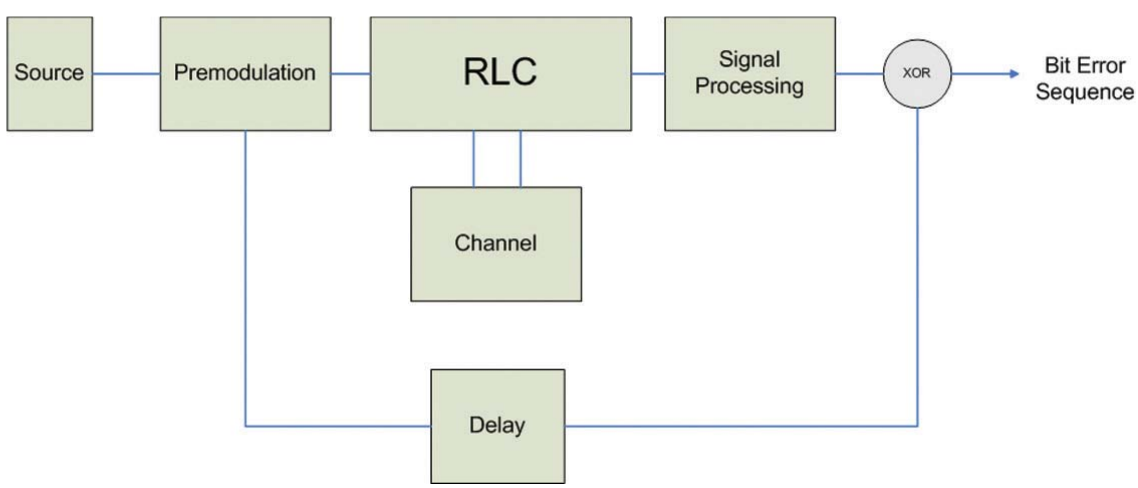

Figure 1. Radio link model with subsystems. The bit error sequence needed for receiver performance analysis is generated by comparing source bits to received bits with a logical exclusive-or (XOR) operation.

susceptibility. The tests were effective for evaluating end-to-end interference susceptibility. However, in many cases, signals within the receiver were difficult to acquire or entirely unavailable. Without these signals, it was difficult to fully evaluate the susceptibility of individual signal processing algorithms. The computer simulated radio link model was conceived to overcome this problem.

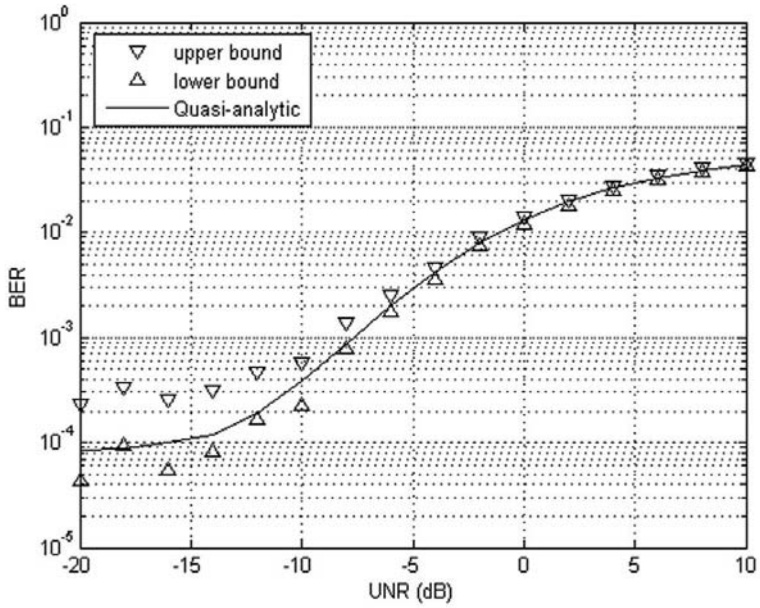

Figure 2. Upper and lower bounds of the simulated BER confidence interval are plotted with quasi-analytic BER for BPSK radio system using threshold detection and subjected to gated Gaussian noise.

The radio link model will also be used to translate new multipath and undesired signal measurements directly to receiver performance measures such as bit error rate (BER). ITS has recently performed an extensive set of mobile to mobile propagation measurements. Currently, ITS is preparing for new man-made noise measurements. Previously, these measurements were only reported in terms of parameters such as multipath root-mean-square delay spread or undesired signal mean and peak powers, which under certain conditions, can be used to estimate BER. The radio link model will allow engineers to translate these measurements directly to BER.

Conceptually, the model is built around a radio-link-core (RLC) to which various signal processing algorithms and channels are attached. The RLC, which includes the modulator and demodulator, must be highly characterized so its effects on signal processing algorithm test results are minimal. The effect of the channel on signal processing algorithm performance is evaluated by measuring BER over a range of channel model parameter values. More detailed evaluation is supported by its ability to collect various signals and sequences present while the receiver is performing poorly.

In FY 2007, the basic radio link model was designed, implemented in software, and validated. Validation compared simulated BER of a binary phase-shift keyed (BPSK) radio system using threshold detection to BER from a quasi-analytic model over a range of undesired signal power to receiver noise power (UNR) ratios. The quasi-analytic model efficiently computes BER from the first-order statistics of the undesired signal. Figure 2 shows such a comparison for a gated Gaussian-noise undesired signal. The BPSK radio system was operated at a signal to noise ratio (SNR) of 8.4 dB with a BER of 1.0e-4. The gated Gaussian noise had a 100 microsecond period with a 12.5 microsecond on time. The upper and lower bounds represent the confidence interval of the simulated BER for a 0.95 confidence level. Inclusion of the quasi-analytic BER in the simulated BER confidence interval indicates the results are statistically consistent.

In FY 2008, we will begin to test the effect of undesired signals on widely used signal processing algorithms. The signal processing algorithms will include Viterbi detection, least-mean-square equalization, and spread spectrum correlation. We are particularly interested in understanding the effect of gated Gaussian-noise on Viterbi detection. As part of this work we will investigate ways the quasi-analytic model and a hardware test bed based on a legacy satellite digital television equipment can be used to validate simulated Viterbi detection BER.

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1.7 GHz Advanced Wireless Systems Compatibility Testing

Outputs

- Measurements to determine effects of interference between proposed Advanced Wireless Systems devices and incumbent radio systems.
- Compatibility analysis between proposed devices and incumbent radio systems.

In May 2003, the Presidential Spectrum Policy Initiative brought forth a plan to improve efficiencies in radio spectrum use and provide incumbent systems greater protection from interference. A second goal of the policy is to promote timely integration of new, emerging technologies while preserving national security, public safety and scientific research. In April 2004, a Presidential Executive Memorandum demanded that affordable broadband access be made available to all Americans by 2007. Both of these memorandums paved the way for a spectrum auction in June 2006 for 90 MHz of radio spectrum for advanced wireless (including third generation or “3G”) telecommunications services to meet the demand for new wireless services. The aforementioned 90 MHz of spectrum lies between 1710 to 1755 MHz and 2110 to 2155 MHz.

ITS was tasked by NTIA’s Office of Spectrum Management (OSM) to examine the electromagnetic compatibility between new Advanced Wireless Services (AWS) devices and incumbent radio systems in the 1710 to 1755 MHz AWS band. The primary concern was whether incumbent systems would experience interference from new AWS devices, and if such potential existed, what the coordination distances and frequency separations would need to be to allow both types of systems to share the same band.

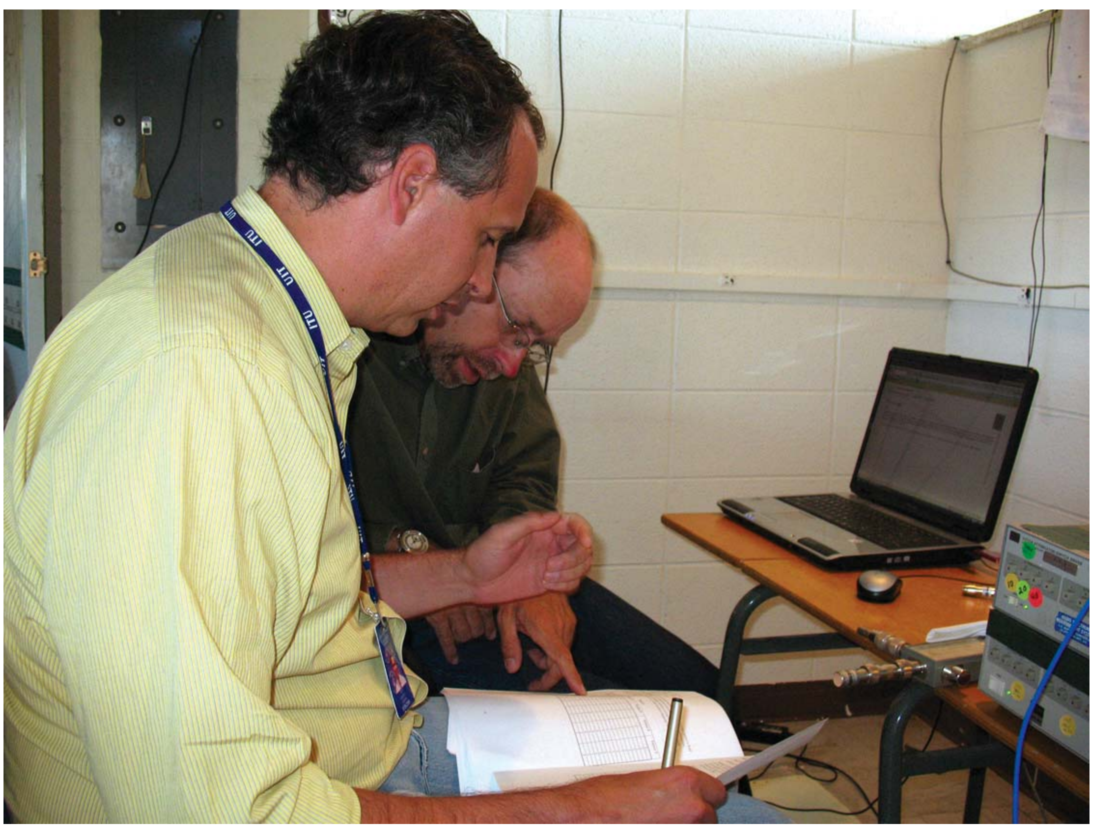

Figure 1. ITS and OSM engineers reviewing data collected during the AWS testing (photograph by J. Carroll).

Figure 2. AWS measurement participants examining the spectrum of a prototype wireless device (photograph by J. Carroll).

ITS and OSM jointly decided to perform a series of field measurements with both the incumbent systems and the new AWS devices, rather than attempting to model the problem and extrapolate the results. AWS vendors worked directly with ITS and OSM to develop a comprehensive test plan to facilitate a series of field measurements. AWS vendors supplied prototype Code Division Multiple Access (CDMA) based AWS devices. The incumbent users made available some examples of their existing communication devices. NTIA provided measurement hardware, test facilities and test and measurement personnel.

The approach to the testing was to couple interference signals from the CDMA devices into the communication systems of the incumbent users. The coupling was done via hardline connections, so as to ensure that the interference levels could be completely controlled. Interference was injected at carefully selected power levels and the effects of interference at each level were noted and recorded. Subsequently, the power levels at which interference effects occurred were converted into equivalent distances between the two classes of devices, so that electromagnetic compatibility studies could be completed. Interference mechanisms that were examined included: on-frequency interference, off-tuned interference and front-end overload in the victim receivers. Possible solutions, including filter-based solutions, were also examined.

The tests were performed at the ITS Table Mountain field site in August 2007 and lasted about a month. The results were collated, thoroughly analyzed by ITS and OSM engineers, and finally presented to all of the involved parties. The test data ultimately provided the key to determining how, and to what extent, this spectrum band can be shared during the transition between users.

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Signal Characteristics, Spectral Emissions, and Interference Analyses

Outputs

- Measurements and analyses showing susceptibility of a wide variety of radio systems to interference effects.
- Technical publications and presentations demonstrating research results.

Historically, radio spectrum bands have been allocated on a service-by-service basis. For example, one band might be assigned to land mobile radios while another would be assigned to radars. That paradigm is gradually changing to one in which multiple services share spectrum bands, even in cases in which such services have not been considered to be electromagnetically compatible. A case in point is the sharing of spectrum at 5 GHz between digital data links and radar systems. Sharing is often thought to provide more efficient use of spectrum, but it also means that radio systems must operate on an *interference-limited* basis. This tends to shift the onus of electromagnetic compatibility between radio services away from traditional spectrum band allocations and frequency assignments and replaces that approach with new engineering designs for entire radio services as well as individual radio systems.

This new emphasis on interference-limited spectrum sharing means that it has now become critical to understand the effects of interference from various types of transmitted signals when they are coupled into a wide variety of victim receivers. As part of its responsibilities as a Government telecommunications laboratory, ITS has undertaken multiple projects in recent years to assess, through direct tests and measurements and also through the development of complementary theoretical models and analyses, the thresholds at which various radio waveforms cause interference effects in a wide variety of radio receivers. These projects have been undertaken using the Institute's direct Government funding, as well as funding from sponsors in other Government agencies and the private sector. These studies have included: interference from ultrawideband (UWB) signals into radars, digital satellite television receivers, land mobile radios, and global positioning system (GPS) receivers; interference from radars into 5-GHz dedicated short range communication (DSRC) devices used in highway environments; interference from communication systems and digital data systems into radar receivers; and interference from radionavigation satellite signals (RNSS) into radars. In all of these studies, ITS has provided detailed technical reports describing the thresholds and manifestations of interference in victim receivers.

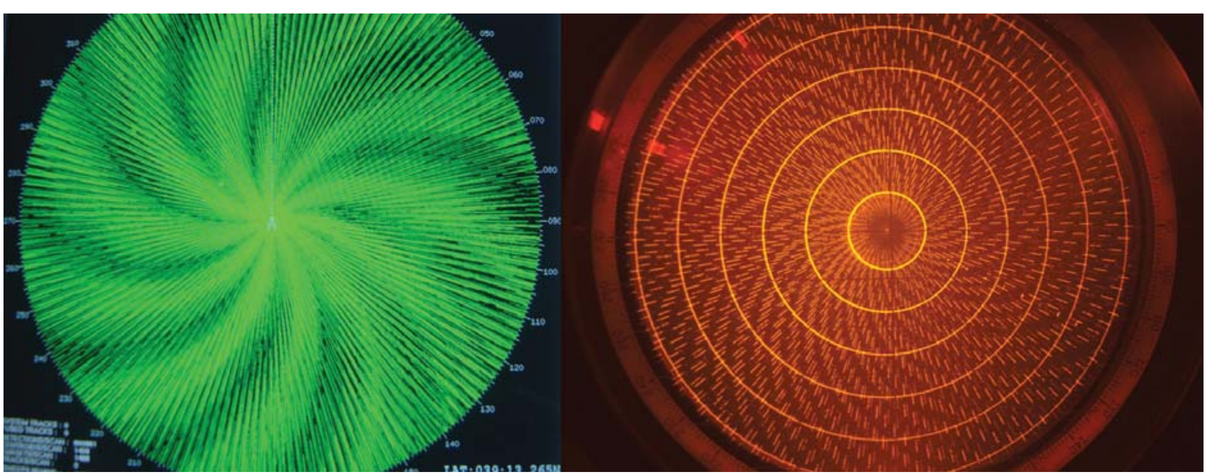

Figure 1. Air search radar display (left) and maritime surface search radar screen (right) with severe interference effects during ITS tests (photographs by F.H. Sanders.

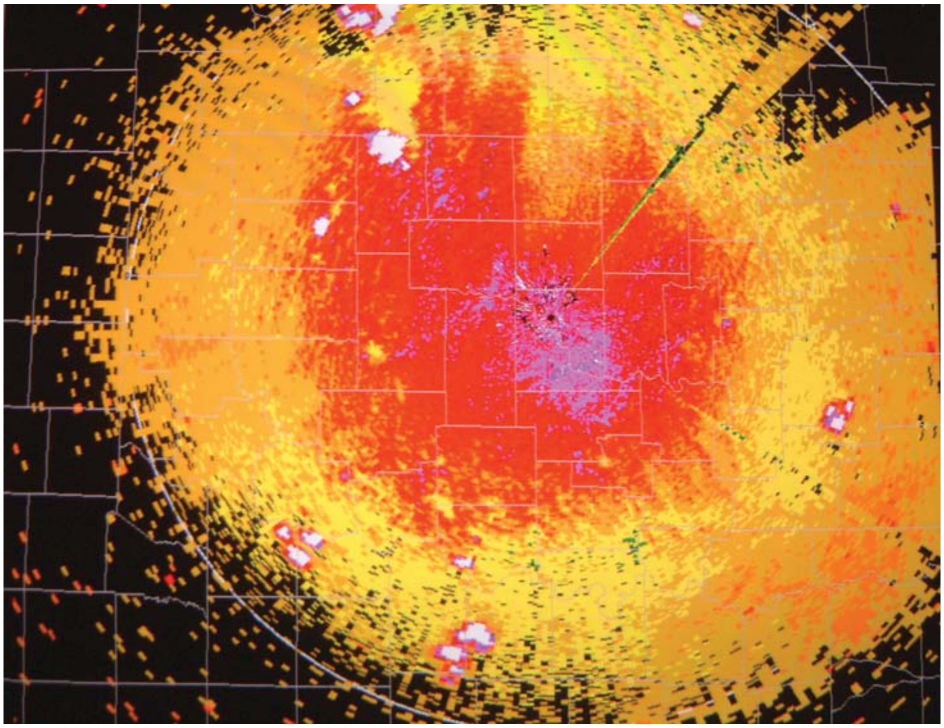

Figure 2. A weather radar display with interference effects at about 1:00 and 4:00 during ITS tests (photograph by F.H. Sanders).

Engineers from throughout the Institute are involved in these studies. The Institute is continually developing new methodologies for generating interference waveforms and testing the effects of the waveforms on victim receivers. The problem is complicated because, on the one hand, test and measurement data are needed to determine the effects of interference on receiver performance, while on the other hand, resource limitations make it impossible to perform hardware tests on all possible interference waveforms into all possible victim receivers. The approach that ITS is taking to solve this problem is to develop theoretical models of receiver performance in the presence of interference, and to validate and correct the models using strategically selected sets of tests and measurements on hardware-generated interference signals that are coupled into key types of victim receivers.

To cite some examples, UWB signals have been mathematically simulated, downloaded into a vector signal generator (VSG), and injected into a digital satellite television receiver. The receiver's performance has then been assessed as a function of the interference levels and pulse parameters. In another ongoing effort, wireless local area network (WLAN) signals have been recorded from actual WLAN transmitters using a vector signal analyzer (VSA) digitizer. The recorded waveforms are then downloaded into a VSG, and the VSG is used as a transmitter that injects WLAN interference into radar receivers.

Interference at high levels may produce easily observed effects that effectively terminate the successful operation of victim receivers, as shown in Figure 1. As devastating as these effects are, they at least are obvious to operators. At lower interference levels, however, the effects of interference may be much more subtle and thus more insidious; interference at these low (but important) levels is an ongoing area of research. Current work in this area includes the effects of interference from digital communication signals on both existing and evolving types of radar receivers and the effects of interference on software-simulated digital radio victim receivers.

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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 measurement 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 four U.S. patents (with one additional patent pending), been adopted as the North American Standard for measuring digital video quality (ANSI T1.801.03-2003), been included in two International Recommendations (ITU-T Recommendation J.144, and ITU-R Recommendation BT.1683), and are currently being used by hundreds of individuals and organizations worldwide.

To facilitate the transfer of ITS-developed video quality metrics (VQMs) into the private sector, ITS has developed and maintains three software tools that run under both the Windows and Linux operating systems. Using these new software tools, users and service providers can quantify the digital video quality of their networks using methods standardized by the American National Standards Institute (ANSI) and the ITU.

Figure 1. SSCQE-HRR subjective rating device used for HDTV experiment (photograph by S. Wolf).

The first tool, called Command VQM (CVQM), provides a simple command line interface for processing (i.e., calibration and video quality measurements) a pair of video files that have been captured from the source and destination ends of a video transmission system. The second tool, called Batch VQM (BVQM), allows the user to perform Graphical User Interface (GUI) based batch mode processing of many captured video streams, or files. The third tool, called In-Service VQM (IVQM), requires two PCs, one located at the source end and the other located at the destination end of a video transmission system. The two PCs communicate their reduced reference features via the Internet, producing in-service end-to-end video quality monitoring results. Beginning in late FY 2007, source code and executable binaries of these tools were made available royalty free to all interested parties.

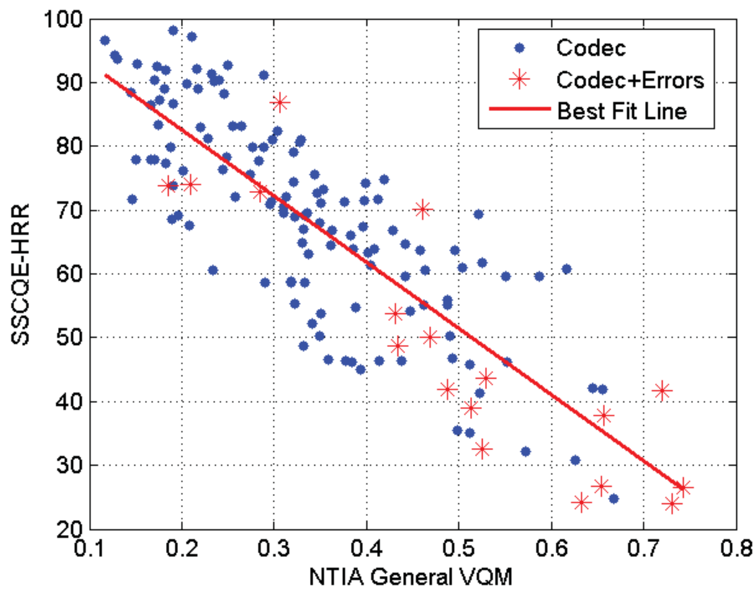

Figure 2. Subjective to objective correlation results for HDTV experiment.

Work is continuing on extending the measurement methodologies to High Definition TV (HDTV) systems. In FY 2007, results were analyzed and published from an experiment whose goal was to assess whether the NTIA General VQM (i.e., the metric standardized by ANSI and the ITU) is an acceptable objective metric for measuring HDTV quality. The HDTV subjective experiment that was performed to evaluate the NTIA General VQM contained sixty 30-second video clips that were rated using the Single Stimulus Continuous Quality Evaluation with Hidden Reference Removal (SSCQE-HRR) method. In the SSCQE-HRR method, viewers move a quality slider in real time (see Figure 1) to express their opinion of video quality. The 60 clips used in the experiment included twelve 1080i HDTV originals and 48 processed versions of these originals from 16 different video systems. The video systems included 5 different HDTV codecs (coder/decoders) running at bit rates from 2 to 19 Mbps and broadcast transmission errors (i.e., RF transmission with poor signal-to-noise-ratio). Excellent objective-to-subjective correlation results for this experiment (see Figure 2) demonstrate the potential application of the NTIA General VQM to HDTV quality monitoring. The graph in Figure 2 plots the slider samples at 10 second intervals versus the corresponding NTIA General VQM for the preceding 10 second interval.

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

Recent Publications

M.H. Pinson, S. Wolf, and R.B. Stafford, “Video performance requirements for tactical video applications,” in *Proc. IEEE Conference on Technologies for Homeland Security*, Woburn, Massachusetts, May 16-27, 2007.

S. Wolf and M.H. Pinson, “Application of the NTIA general video quality metric (VQM) to HDTV quality monitoring,” in *Proc. Third International Workshop on Video Processing and Quality Metrics for Consumer Electronics (VPQM-07)*, Scottsdale, Arizona, Jan. 25-26, 2007.

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