
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 area the Institute develops perception-based measurements for multimedia services. ITS transfers all of this 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 is developing an advanced antenna testbed to be used in the investigation of "smart" antennas, which will greatly increase the capacity of wireless communications systems. The project is funded by NTIA.

Advanced Radio Technologies Symposium The Institute conducted the 2000 International Symposium on Advanced Radio Technologies. This third annual symposium focused on broadband wireless communications technologies and applications. The project is funded by the Department of Defense.

Advanced Telecommunications in Rural America The Institute conducted research for a report evaluating the technologies available to provide advanced telecommunications to rural America. The project was 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.

Augmented Global Positioning System The Institute provides technical support for the design and implementation of a nationwide differential GPS service that will provide navigation and positioning information to surface users throughout the country. The project is funded by the Federal Highway Administration (FHWA).

Mobile Network Modeling The Institute is involved in research on the performance of wireless communications networks. The project is funded by NTIA.

Narrow Pulse System Characterization The Institute conducts research to characterize and model the narrow pulse systems used in ultrawideband communications systems and radars. Projects are funded by NTIA and the Department of Defense.

Software Defined Radio Technology The Institute is involved in research on advanced radio systems including software defined radios and smart antennas. Projects are funded by NTIA and the Department of Defense.

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 involving the radio propagation channels that will be employed in new wireless communication technologies such as personal communications services. Projects are funded by NTIA and Lucent Technologies.

Adaptive Antenna Testbed

Outputs

- Wideband radio channel sounding measurements.
- Propagation loss, fading, delay, and Doppler statistics over a broad bandwidth.
- Antenna array diversity gain data.
- Angle of arrival input data for adaptive antenna schemes.

The use of wireless, mobile, personal communications services (PCS) is expanding rapidly. Multiple-access schemes based on frequency division, time division, and orthogonal codes 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-random (PN) code generator to BPSK modulate 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. A four element antenna array. The elements are spaced at $\lambda/2$ intervals (photograph by P. Papazian and E. Gray).

A recent example of an ATB application is a beam steering experiment conducted in FY 2000, which demonstrated the value of the ATB's continuous acquisition capability. A linear, four-element PCS receiving array with $\lambda/2$ element spacing is shown in Figure 1. A mobile van-mounted dipole was used to transmit a 511-bit PN code on a 1.92-GHz carrier frequency. The drive route consisted of a suburban area as well as some high rise offices in Boulder, CO. The 10 Mb/s transmitted code was then sampled at each receiving antenna at a rate of 40 MHz. This data could then be post-processed to determine the channel impulse response for each array element. A total of 2044 samples per impulse were taken, yielding an impulse duration of 51 μ s. Data were collected in the burst mode with a 5 s delay between bursts. A burst consisted of 16 impulses with a 3 ms delay between impulses.

These data were then forward/backward averaged before post-processing using several beam steering algorithms. Figure 2 shows direction-of-arrival (DOA) data using two of these algorithms: a parallelogram method (PM) and the normalized maximum likelihood method (NMLM). Results of the experiment showed that the PM better estimates the line-of-sight direction and the DOA of the main signal power, while the NMLM better identifies isolated spokes in the signal power.

Comprehensive frequency translation data were also collected for several military training ranges using this system with multi-frequency transmitters. Results of this project are described in NTIA Reports 00-380 and 00-381 (see list of Recent Publications at right).

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. 81, for more information about the ATB.)

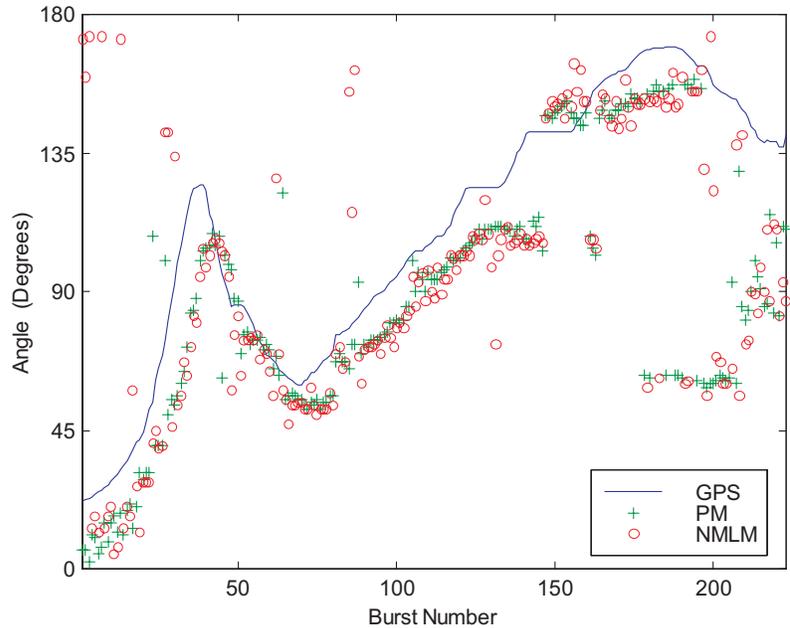


Figure 2. Direction of arrival as determined by the GPS location, the maximum of the PM estimate and the maximum of the NMLM estimate.

Recent Publications

P. Wilson and P. Papazian, "PCS band direction-of-arrival measurements using a 4 element linear array," in *Proc. Vehicular Technology 2000*, Boston, MA, Sep. 2000.

P. Papazian, P. Wilson, M. Cotton and Y. Lo, "Flexible interoperable transceiver (FIT) program test range I: Radio propagation measurements at 440, 1360 and 1920 MHz, Edwards Air Force Base, CA," NTIA Report 00-380, Oct. 2000.

P. Papazian, P. Wilson, M. Cotton and Y. Lo, "Flexible interoperable transceiver (FIT) program test range II: Radio propagation measurements at 440, 1360 and 1920 MHz, Ft. Hood, Texas," NTIA Report 00-381, Oct. 2000.

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Advanced Radio Technologies Symposium

Outputs

- Symposium proceedings.
- Exchange of ideas among leading experts in broadband wireless technologies.

The Institute hosted the Third Annual International Symposium on Advanced Radio Technologies on September 6-8, 2000 in Boulder, Colorado. This year's symposium focused on state-of-the-art and future trends in broadband wireless technologies. The symposium featured a keynote address by the Assistant Secretary of Commerce for Communications and Information, Mr. Greg Rohde, shown in Figure 1.

Session presentations by leading experts from government, academia, and industry were followed by

forward-looking open round table discussions on future directions in technologies and related issues. The symposium encouraged an interactive dialogue between the speakers and the audience so that participants could share their ideas and opinions on relevant technologies and future trends. Approximately 120 individuals from 13 countries attended the symposium.

The symposium was organized into six sessions: an opening session, two sessions on broadband wireless technologies, broadband wireless networks, broadband wireless standards, and next generation Internet. These sessions comprised 36 presentations.

After welcoming remarks by Dr. Christopher Holloway of NIST (Figure 2), the opening session, chaired by Dr. John Lemmon of ITS, began with the keynote address by Assistant Secretary Rohde, followed by an overview of broadband communica-

tions by ITS engineer Frank Sanders, discussions of spectrum issues and standards, and perspectives on broadband wireless systems by agencies from the Department of Defense.

The sessions on broadband wireless technologies included an overview presentation on broadband wireless technologies, presentations by ITS engineers Dr. Roger Dalke and Robert Achatz on performance modeling of multichannel multipoint distribution services (MMDS) and broadband wireless local area networks, and presentations on optical wireless systems, smart antennas, space-time signal processing, neighborhood local multipoint distribution services (LMDS), propagation modeling for broadband millimeter wave channels, low noise active receiver feeds, third generation wireless security, and high performance broadband spread spectrum systems.



Figure 1. Assistant Secretary Greg Rohde delivering the keynote address at the International Symposium on Advanced Radio Technologies (photograph by E. Gray).

The broadband wireless networks session consisted of talks on laws that govern the Internet, asynchronous transfer mode (ATM) traffic management in wireless LMDS networks, broadband wireless network architectures, broadband local access, and a presentation by ITS engineer Val Pietrasiewicz on the effect of evolving information technology (IT) applications on broadband wireless requirements.

The standards session comprised talks on the role of standards in advancing wireless access technologies, enabling broadband wireless through standardization, standards for multiprotocol air interfaces, fixed broadband wireless access standards, the influence of information theory on wireless architecture standards, and the business and social implications of telecommunications standards.

The session on the next generation Internet, chaired by ITS engineer Ken Allen, included presentations on large scale networking programs, the interplanetary Internet, broadband wireless access for the next generation Internet, very high spectral efficiency wireless communications, satellite and terrestrial network architectures, and optical spread spectrum for Internet operation at terabit rates.



Figure 2. Dr. Christopher Holloway welcoming symposium participants (photograph by E. Gray).

The symposium allowed participants to interact in a friendly and informal atmosphere that included numerous breaks, luncheons (Figure 3), lively dialogues among the speakers and the audience, and tours of the Boulder Laboratories.



Figure 3. Lunch on the lawn at the Boulder Laboratories (photograph by E. Gray).

Recent Publication

Symposium proceedings can be found at the web site <http://ntia.its.bldrdoc.gov/isart/>

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Advanced Telecommunications in Rural America

Outputs

- Report on status of broadband deployment in rural vs. non-rural areas in the United States.

ITS engineers worked in concert with other NTIA Offices and the Rural Utilities Service (RUS) to respond to a request by ten U.S. Senators on the status of broadband deployment in rural versus non-rural areas in the United States. The results of the analysis were published in April 2000 as a report, *Advanced Telecommunications in Rural America: The Challenge of Bringing Broadband Services to All Americans*. This report also responds to a call by President Clinton and Vice President Gore to bridge the digital divide and create digital opportunities for more Americans. The rate of deployment of broadband services will be key to the future economic growth of every region, particularly in rural areas that can benefit from high-speed connections to urban and world markets.

The results of the research show that rural areas are currently lagging far behind urban areas in broadband availability. Deployment in rural towns (populations of fewer than 2,500) is more likely to occur than in remote areas outside of towns. These latter areas present a special challenge for broadband deployment. Only two technologies, cable modem and digital subscriber line (DSL), are being deployed at a high rate, but the deployment is occurring primarily in urban markets. Broadband over cable has been deployed in large cities, suburban areas, and towns. It was found that cable modem service was offered in less than 5% of towns with 10,000 or less population, while it is offered in portions of more than 65% of cities with populations over 250,000.

DSL technology also has been deployed primarily in urban areas. The Regional Bell Operating Companies (RBOCs) are providing DSL service primarily in cities with populations above 25,000 according to public RBOC data. While more than 56% of all cities with populations exceeding 100,000 had DSL available in some areas, less than 5% of cities with populations less than 10,000 had such service. Deployment of both cable modem and DSL service in remote rural areas is far lower.

The primary reason for the slower deployment rate in rural areas is economic. For wireline construction, the cost to serve a customer increases the greater the distance among customers. Broadband service over cable and DSL is also limited by technical problems incurred with distance and service to a smaller number of customers. Both technologies, however, promise to serve certain portions of rural areas. Cable operators promise to serve smaller rural towns, and smaller, independent telecommunications companies and competitive providers may soon be able to offer DSL to remote rural customers on a broader scale.

Advanced services in rural areas are likely also to be provided through new technologies, which are still in the early stages of deployment or are in a testing and trial phase. Satellite broadband service has particular potential for rural areas as the geographic location of the customer has virtually no effect on the cost of providing service. Several broadband satellite services are planned. Their actual deployment remains uncertain, especially in light of the recent entry into Chapter 11 bankruptcy of two satellite service companies. Wireless broadband services are also planned for rural areas. More immediately, multipoint multichannel distribution service (and potentially local multipoint distribution service) fixed service capabilities may provide a solution for some rural areas. In as little as five years, third generation mobile wireless services providing data rates as high as two megabits/second may be operational.

In order to support advanced services in rural areas, NTIA and RUS recommended a number of actions. We recommended the continued support and expansion of those government programs, such as the E-rate program, that ensure access to new technologies including broadband services. We also urged the Federal Communications Commission to consider a definition of universal service and new funding mechanisms to ensure that residents in rural areas have access to telecommunications and information services comparable to those available to residents of urban areas. Support for alternative technologies will also be crucial to the deployment of advanced services in rural America. The Administration is committed to increasing investment in research and development to promote the next generation of broadband technologies.

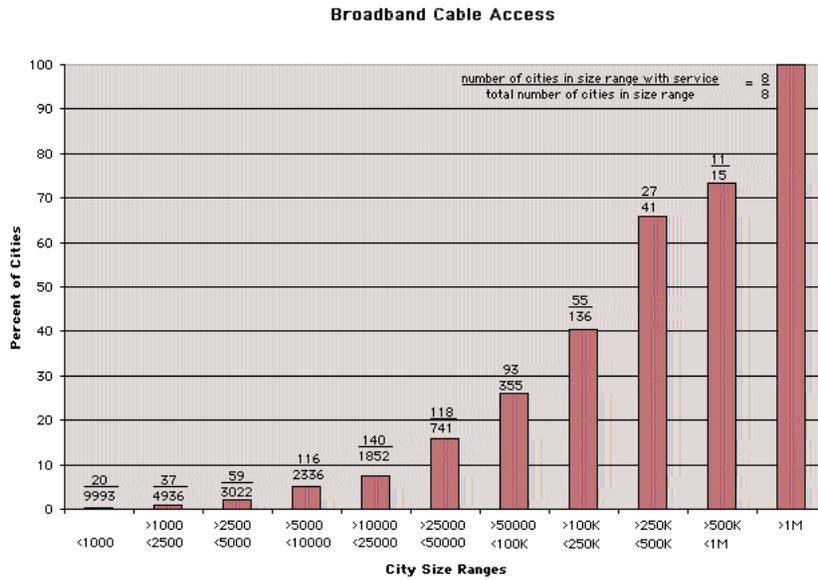


Figure 1. Percent of cities with cable modem service offered in at least some portions of the city in the early part of year 2000.

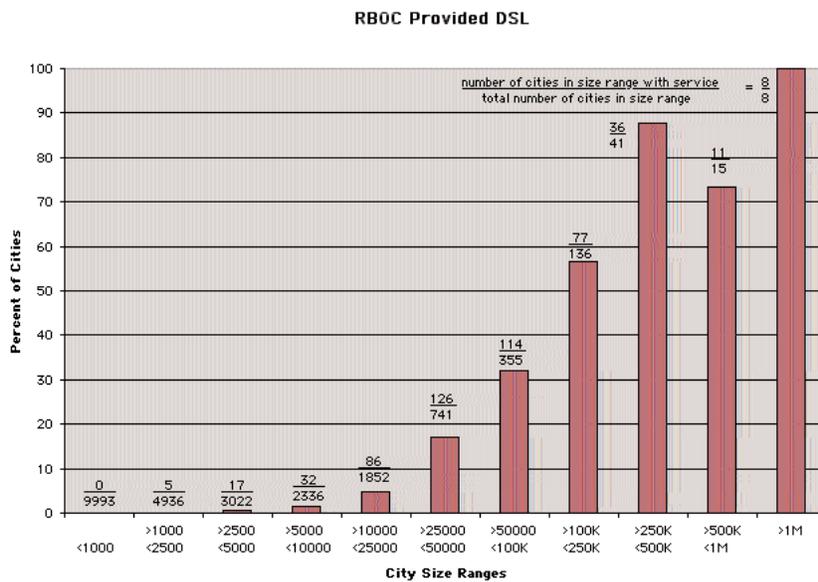


Figure 2. Percent of cities with DSL service offered in at least some portions of the city in the early part of year 2000.

Recent Publication

National Telecommunications and Information Administration and Rural Utilities Service, "Advanced telecommunications in rural America: The challenge of bringing broadband service to all Americans," April 2000.

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Audio Quality Research

Outputs

- Algorithms and software for speech and audio quality assessment and coding.
- Technical papers and reports documenting new results.
- Presentations on speech and audio quality assessment issues.

Digital compression and transmission of speech and audio signals have helped make possible the current explosion of telecommunications and broadcasting offerings, which include digital cellular telephone services, voice over Internet protocol (VoIP) services, voice messaging systems, digital audio broadcasting, Motion Picture Experts Group (MPEG) 1, Layer-3 (MP3) music files and MPEG Advanced Audio Coding systems. Digital compression allows these systems to deliver good-quality speech using bit rates between 4 and 64 kbit/s. Audio signals, including music and entertainment soundtracks, are typically delivered at rates between 16 and 256 kbit/s per channel. Compressed speech and audio signals can be transmitted as data packets, thus sharing channel capacity and possibly radio spectrum with other data streams and hence with other users.

These digital compression and transmission techniques and the associated economic trade-offs are closely coupled to issues of speech quality and audio quality. Equipment manufacturers, service providers, and users all seek equipment and services that maximize delivered speech or audio signal quality under applicable transmission channel constraints. The complex time-varying interactions among signal content, source coding, channel coding, and channel conditions are making it increasingly difficult to define or measure speech or audio quality. The ITS Audio Quality Research Program studies quality issues in speech and audio compression and transmission, and develops and verifies tools that assist with quality estimation and optimization.

The most fundamental and correct measures of audio quality are provided by subjective listening experiments. However, properly conducted subjective listening experiments tend to be complex, time

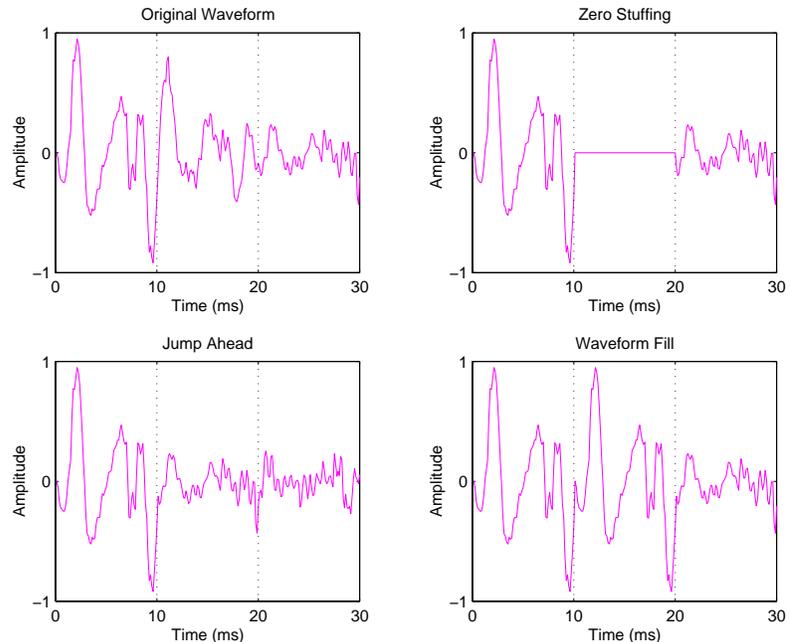
consuming, and expensive. To provide a practical alternative, the Audio Quality Research Program developed the measuring normalizing block (MNB) algorithms for estimating the perceived quality of 4 kHz bandwidth speech. The MNB algorithms include a simple hearing model followed by a more sophisticated judgment model. When speech quality estimates from the MNB algorithms are compared with the results of subjective listening experiments, a good degree of correlation is found. These algorithms furnish industry, Government, and other users with valuable tools that provide rapid and reliable quality feedback. The MNB algorithms form both the American National Standards Institute Telecommunications Standard T1.518-1998 and ITU Recommendation P.861, Appendix II, 1998. During FY 2000, program staff continued to respond to wide interest in the MNB algorithms from industry and academia. The MNB patent was issued to ITS on July 18, 2000. Much of the ongoing program work is ultimately aimed at extending the usefulness and accuracy of the speech quality estimates generated by MNB-type algorithms.

In FY 2000 the Audio Quality Research Program conducted numerous studies. One study addressed the problem of calibrating subjective test results. Subjective tests gather users' opinions of speech or audio quality, but results of different tests in different laboratories are not generally directly comparable. Program staff developed an iterated nested least-squares algorithm to map subjective test results to a common scale. Program staff also studied the causes and implications of a specific type of distortion that occurs in low-rate codebook-based speech compression devices. Extensive speech quality tests were performed on an automated two-way radio and telephone interconnection system.

Program staff also collaborated with National Institute of Standards and Technology (NIST) staff who recently developed a new magnetoresistive microscope. In support of the NIST effort, program staff prepared magnetic tape samples and formatted magnetoresistive microscope images of those samples into digital audio files for playback. Through this work it was determined that microscope images of severely abused tape fragments contain usable audio signals.

Another continuing program activity centers on the packet loss concealment (PLC) algorithms used for voice over Internet protocol (VoIP) systems. In most VoIP situations, at least some fraction of the transmitted data packets are not received in time to be decoded and played as speech waveforms. These unavailable packets are considered to be lost. PLC algorithms attempt to conceal the fact that some packets (and the corresponding portions of a speech waveform) have been lost.

Many PLC approaches have been proposed. The Figure provides highly simplified operational descriptions of three basic PLC approaches in the case where a single packet is lost. The first waveform in the Figure is a portion of an original speech waveform to be transmitted. The packet boundaries are indicated by dotted lines. The “zero stuffing” algorithm simply substitutes a string of zeros for the information in the lost packet. The “jump ahead” algorithm works only when the network behavior and the receiver jitter buffering are such that when it is time to play the lost packet N, packet N+1 is already available. The “jump ahead” algorithm simply plays packet N+1 instead of packet N. This effectively contracts the speech signal in the time dimension. Thus it is necessary to later dilate the speech signal in the time dimension by an equal amount. The “waveform fill” algorithm preserves the time dimension of the speech signal and generates an estimated waveform to fill in for the missing information. This estimate is usually based on the contents of the last received packet. Estimation may be a complex process, or it may amount to simply repeating a previously received portion of the waveform as shown in the Figure.



Simplified examples of three packet loss concealment (PLC) approaches.

Numerous variants of these three basic approaches exist, and fundamentally different approaches exist as well. The case of multiple lost packets adds additional complexity. program staff are studying PLC algorithms in terms of quality of concealment, algorithmic delay and complexity, and the amount of side information required by the algorithm. As expected, results are strongly dependent on packet size: shorter losses are easier to conceal than longer losses.

Technology transfer and literature dissemination efforts in the program were enhanced in FY 2000 through the development of a significantly expanded web presence located at www.its.bldrdoc.gov/home/programs/audio/audio.htm. Program results were also disseminated to industry, Government, and academia through technical publications and presentations, participation in workshops, conferences and symposia, and laboratory demonstrations.

Recent Publication

S. Voran, “Results on reverse water-filling, SNR, and log-spectral error in codebook-based coding,” in *Proc. 2000 IEEE Workshop on Speech Coding*, Delavan, WI, Sept. 2000.

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Augmented Global Positioning System

Outputs

- Planning of the number and location of GWEN differential GPS reference stations required to provide nationwide signal coverage.
- Recommended frequency assignments and transmitter powers for differential GPS reference stations.

The NAVSTAR global positioning system (GPS) is a space-based radionavigation system that consists of a constellation of 24 satellites in 6 orbital planes. GPS provides accurate three-dimensional position, velocity, and precise time to users worldwide, 24 hours per day. GPS was originally developed as a military force enhancement system. Although still used in this capacity, GPS also provides significant benefits to the civilian community. To make GPS service available to the greatest number of users while ensuring that national security interests are protected, two GPS services are provided. The precise positioning service (PPS) provides full system accuracy to military users. The standard positioning service (SPS) is available for civilian use but has

less accurate positioning capability than PPS, approximately 100 meters. Because the SPS accuracy of 100 m does not meet most civilian navigation and positioning requirements, various augmentations to GPS are used to provide higher accuracy positioning, as well as increased integrity and availability of the positioning information. One form of augmentation, differential GPS (DGPS), can provide 1- to 10-m accuracy for dynamic applications and better than 1-m for static users. In a 1994 report, the result of a study done for the Department of Transportation, ITS recommended implementation of a radio beacon system, operating in the 300-kHz band, modeled after the U.S. Coast Guard's (USCG) local area DGPS. This system would provide nationwide coverage of DGPS for surface applications (DeBolt et al., 1994; see Publications Cited, p. 102).

For the past four years ITS researchers have been conducting a study, sponsored by the Federal Highway Administration, to determine the optimum location and operating parameters of the DGPS reference stations required to provide this civil navigation and positioning service to all surface users across the nation. This new service will be known as

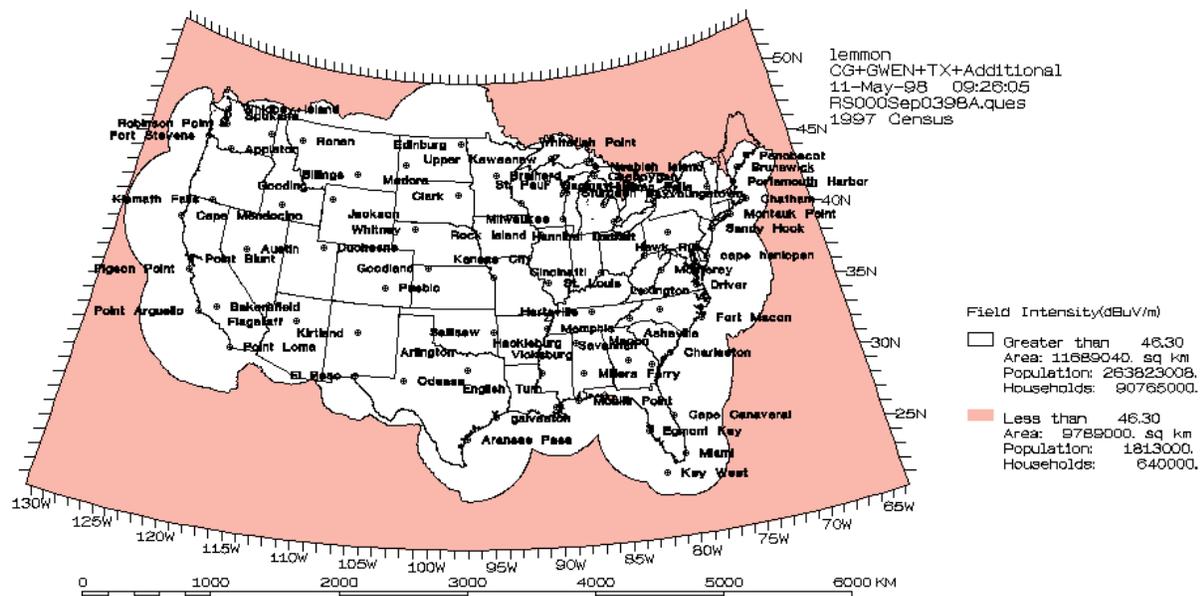


Figure 1. Signal coverage prediction plot showing nationwide coverage of the planned nDGPS service.

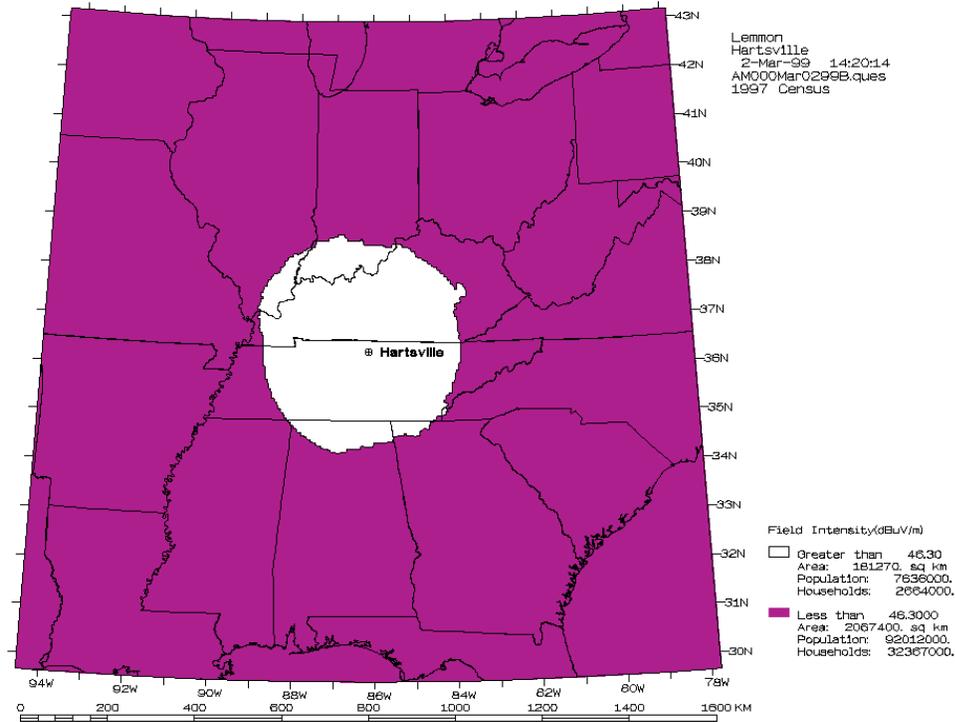


Figure 2. Signal coverage prediction plot for the DGPS reference station at Hartsville, TN.

the nationwide differential global positioning system (nDGPS). The use of this service will have an enormous impact on a diverse set of activities, including ocean and land transportation, surveying and mapping, farming, waterway dredging, recreation, emergency location and rescue operations, and many others that have not yet been identified.

The foundation of nDGPS is the DGPS reference stations currently operating or planned by the USCG and the U.S. Army Corps of Engineers; this system provides coverage of the radiobeacon DGPS signal for coastal areas, harbors, and inland waterways. ITS added additional DGPS reference stations to this foundation to provide nationwide coverage of the DGPS signal. To achieve this additional signal coverage, ITS used the Ground Wave Emergency Network (GWEN) sites, owned by the U.S. Air Force Air Combat Command. The GWEN system is an existing Federal Government asset that provides a cost-effective method of implementing nationwide coverage of the DGPS signal. The GWEN sites were used at existing locations or moved to new locations as required to complete the nDGPS signal coverage. Figure 1 shows a signal coverage prediction plot of

this nationwide coverage. Installation of the GWEN DGPS reference stations is currently underway.

In FY 2000, ITS has provided technical support to the Department of Transportation that has been required for the implementation of the nDGPS. Much of this support has been in the form of interference analyses required to assess the impact of newly installed DGPS reference stations on existing aviation beacons in the 300-kHz frequency band. These analyses are particularly important for those stations whose transmitter powers have been increased from the recommended levels to provide signal coverage in unanticipated coverage gaps. A signal coverage prediction plot for one such station, at Hartsville, TN, is shown in Figure 2.

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Mobile Network Modeling

Outputs

- Discrete radio link models.
- Markov channel models.

In the past, businesses have provided data services to their mobile workforce with proprietary, low speed mobile packet data networks. Today, businesses and individuals are accessing the information-rich worldwide web with laptop computers, personal digital assistants, and mobile phones. The mobile packet data network now consists of a mobile link terminating in a wireless local area network access point or a personal communications service base-station which interworks with the ubiquitous, standards-based Internet.

Signal distortion introduced by radio channel multipath, noise, and interference causes mobile link speed and reliability to be considerably less than those of a fixed link operating over fiber, cable, or twisted wire pairs. These effects are accentuated when the mobile terminal is used while walking or driving. Advanced signal processing techniques will undoubtedly improve mobile link speed and reliability. However, for the foreseeable future, the performance of a mobile link will always fall short of the performance of a fixed link.

Standardized network protocols such as asynchronous transfer mode (ATM) or transmission control protocol/Internet protocol (TCP/IP) used in the worldwide web were designed for routes composed of reliable fixed links. Accessing the Internet with these protocols from a mobile terminal has been found to degrade network performance. For example, packet errors are expected to be rare for reliable fixed links. Therefore these network protocols have eliminated error checking at the link level to decrease delay. As a result, mobile packets in error are retransmitted across the entire route instead of over the unreliable link alone. This decreases network efficiency.

In the future, network protocols must be developed that can remain efficient when the route includes an unreliable mobile link. Accurate mobile radio channel and radio link models are needed for this important task. In FY 2000 ITS developed custom discrete radio link simulator software to fulfill this need. The software is unique in that it easily accepts ITS radio channel measurement data and models. The software is also unique in that it provides detailed packet error statistics needed for accurate network performance prediction.

The ITS discrete radio link simulator software processes data in the following manner. The transmitter converts discrete information bits to a signal that can be propagated through the radio channel.

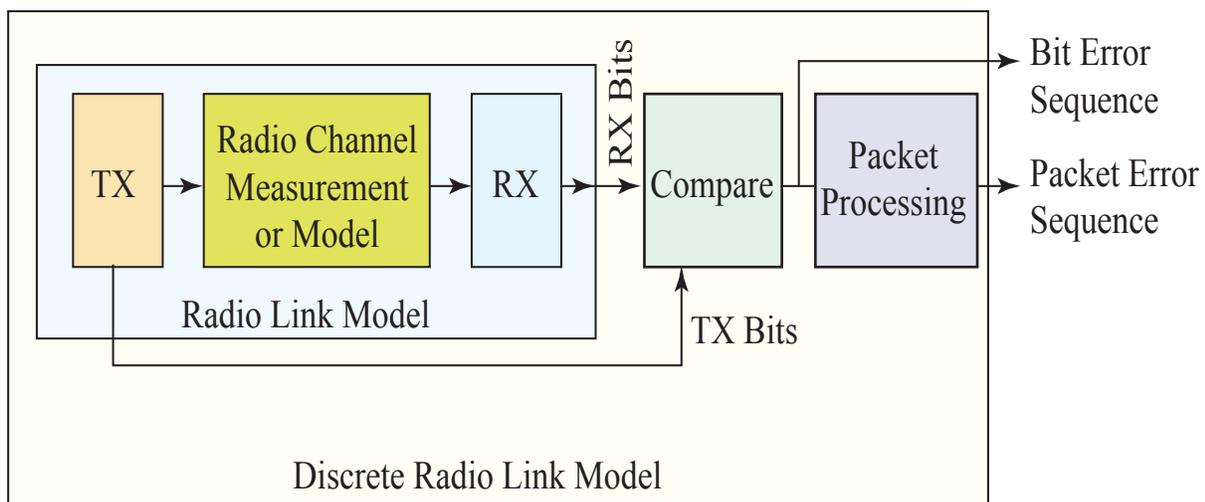


Figure 1. Block diagram of ITS discrete radio link simulation software processing.

The mobile radio channel, consisting of ITS measurements or models, distorts the signal with multipath, noise, and interference. The receiver converts the distorted signal back to discrete bits. A bit error sequence is obtained by comparing transmitted bits to received bits. With knowledge of the packet length and error correction capabilities, the bit error sequence is converted to a packet error sequence. The relationships between radio channel model, radio link model, bit error sequences, packet error sequences, and discrete radio link model are depicted in Figure 1.

Mobility introduces time variability in the channel. The rate at which the radio channel varies determines the amount of correlation between bit errors. If the variation is faster than the transmission speed, there is little correlation between bit errors, and the bit errors are considered independent. On the other hand, if the variation is much slower than the transmission speed there is correlation in bit errors, and measures such as bit interleaving are needed to restore bit error independence. First order error statistics, such as error rate, do not convey error correlation information. Second order error statistics, such as the probability of an error burst length, do.

Many radio channel models assume that the second order bit-error statistics are determined by independent, identical, Poisson distributed random processes with exponentially distributed error burst lengths. Additive white Gaussian noise introduced by the receiver causes Poisson distributed bit errors. However, multipath attenuation and frequency selective fading and interference from other electrical or electronic devices cause the bit errors to arrive with more complex second order statistics. Bit errors may be clustered, for example, by an extended fade caused by driving behind a large building.

The bit and packet error sequences provided by the discrete radio link simulator contain all the information needed to compute first and second order error statistics. These statistics are used in tandem to develop Markov error models capable of generating

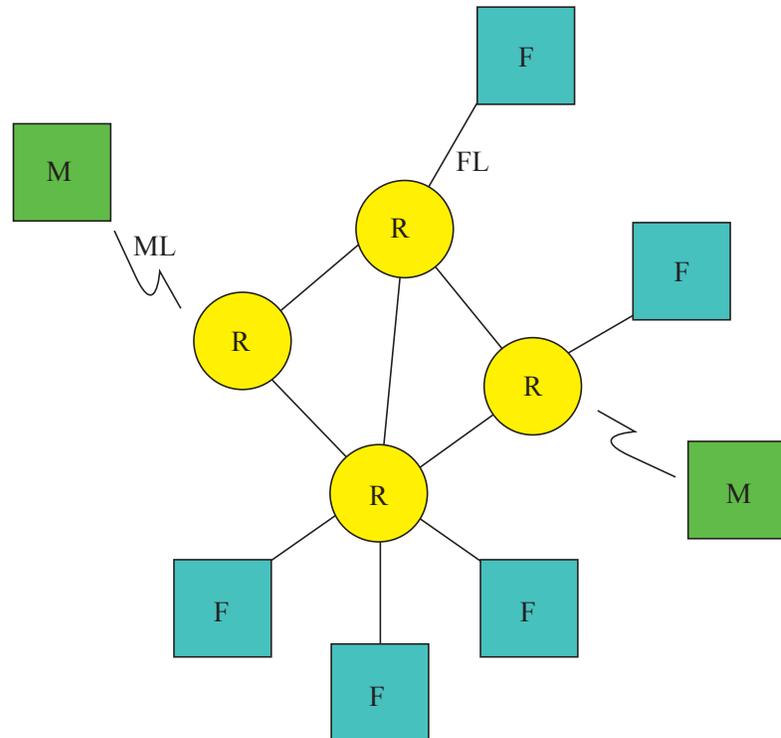


Figure 2. Mobile network.

packets with error statistics corresponding to those generated by the simulation (J.J. Lemmon, "Wireless link statistical bit error model," to be published as an NTIA Report in FY 2001).

In FY 2001 ITS will use these error models to predict the performance of the simple network shown in Figure 2. The network consists of routers (R), fixed (F) and mobile (M) terminals, and fixed (FL) and mobile (ML) links. Each router has a queuing discipline and a routing table. The terminals have traffic models and the links have packet error models. The results of this performance prediction will only be as accurate as the radio channel and corresponding error models. Because of its long history of radio channel measurement and modeling, ITS is poised to assist industry and government in designing and evaluating network protocols that optimize performance of networks with mobile links.

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Narrow Pulse System Characterization

Outputs

- Theoretical analysis of emission spectrum and signal statistics.
- Technical support for spectrum managers, regulators, and system designers.

A wide variety of existing and proposed electronic systems emit short duration pulses for purposes of radar detection, communications, and other applications. The emission spectra for such systems have an extremely wide bandwidth. Such devices are often referred to as ultrawideband (UWB) systems. The proliferation of UWB systems throughout the United States has been predicted by many industry sources. Hence, it is important that the effects of such devices on RF spectrum users be well understood by regulators, spectrum users, and system designers. ITS has analyzed the emissions spectra of a number of such systems in support of efforts to assess their effects on more traditional radio spectrum users.

The theoretical analysis of UWB communications and radar systems provides important insights into how their emissions will affect various types of RF communications devices. In addition to allowing for direct calculation of interference effects, theoretically derived results can be used to aid in the planning, design, and validation of measurements. Perhaps the most important quantity for understanding the potential for interference to RF systems is the power spectral density. The power spectral density is the average power in the signal per unit bandwidth and hence provides important information on the distribution of power over the RF spectrum. As part of an ongoing research effort, ITS has developed methods for predicting the power

spectrum for various archetypal UWB systems. Typically such systems utilize repeated random (in time) very short duration pulses for both communications and radar applications.

The theoretically derived power spectrum can be used to calculate the mean power in the bandwidth of a narrowband victim RF receiver at its operating or center frequency, and hence provides important information regarding the interference potential. For example, Figure 1 shows the power available to a receiver with a nominal 10 kHz bandwidth as a function of frequency. This result is based on a proposed UWB communications system where periodic short duration pulses are transmitted at a nominal rate. The pulses are randomized over some fraction of the pulse repetition period. As shown in Figure 1, the system emits both a discrete and a continuous spectrum. The discrete spectrum is not a factor for RF frequencies above a few hundred MHz. For narrowband victim receivers where gains due to the UWB transmitter filters/antenna, propagation channel, and receiver are fairly constant over the receiver bandwidth, the received interference power can easily be calculated by applying the appropriate gain factors to the power in the receiver bandwidth at the center frequency of the receiver.

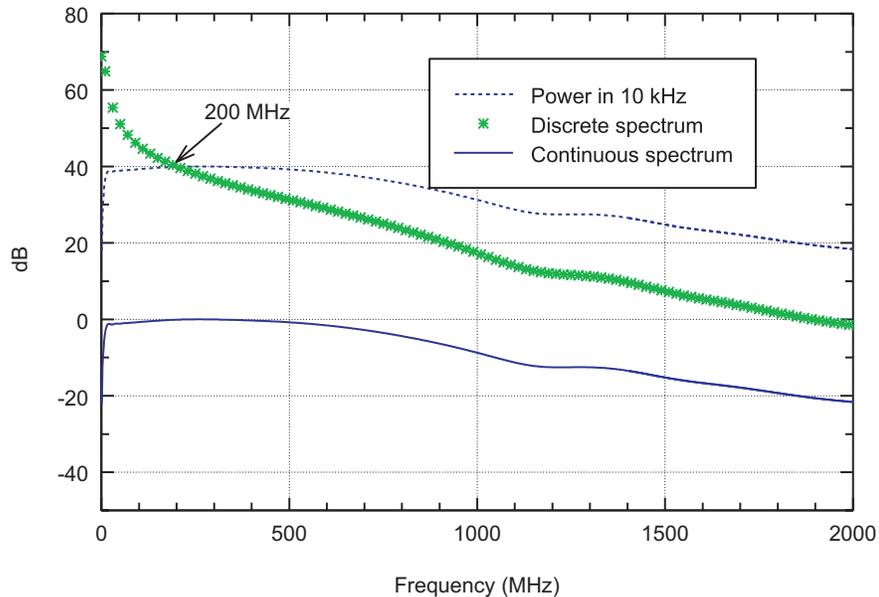


Figure 1. Calculated power spectrum for a UWB communications system.

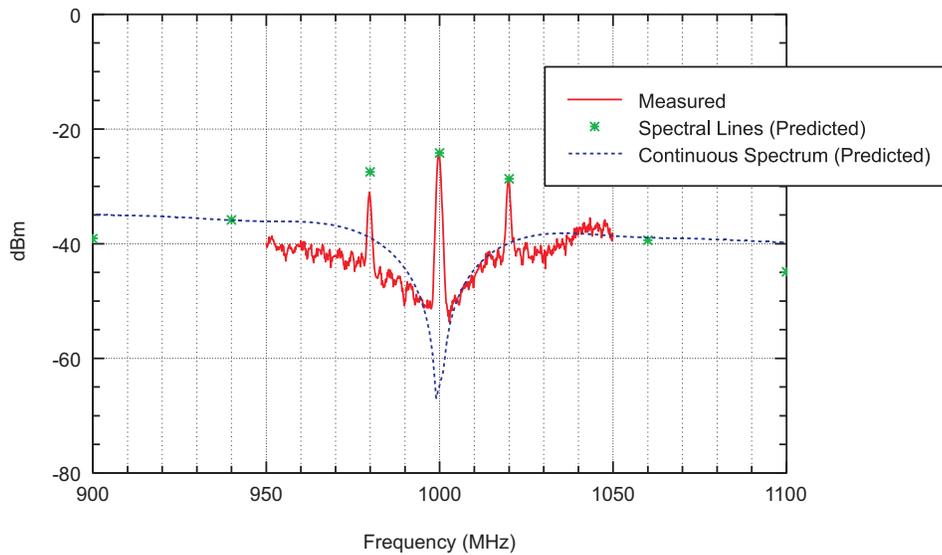


Figure 2. Comparison of measured and predicted results.

Of particular importance to spectrum managers and regulators is the fact that the spectrum contains both *continuous* and *discrete* components. The relative amplitude of both spectral components as well as a potential victim receiver's bandwidth must be used in electromagnetic compatibility (EMC) analysis. The analytical results developed at ITS are quite general and easily applied to EMC analysis. In particular, the analytical results can be used to predict the power spectral density at various points in the radio link between an interfering UWB transmitter and a victim receiver (e.g., at the output of the UWB transmitter, the UWB signal radiated from a particular antenna, or in the IF section of a narrowband RF receiver). When dealing with linear systems, the various pulse shapes are simply related by convolutions with the appropriate impulse response functions.

Since the UWB emissions are often perceived by a victim receiver as a random process, it is important to characterize the signal as viewed by a victim receiver. A knowledge of the statistics of such a process is important in predicting how interference affects the performance of a receiver. When the UWB pulse repetition frequency is larger than the receiver bandwidth, it may be expected that the received signal would appear to be indistinguishable from Gaussian noise. Since receiver performance in a Gaussian noise environment is well understood, quantifying conditions for which the received UWB interference resembles Gaussian noise is important

in predicting receiver performance and developing emissions requirements. Also, when the received signal is Gaussian, only one parameter (mean power) is required to characterize the process. Analytical methods have been developed to determine the statistical character of fixed time base dithered signals. In particular, these methods can be used to determine when UWB signals are perceived by a victim receiver as Gaussian noise, and hence, are important tools for spectrum regulators and system designers.

The analytical results developed as part of this effort are also essential to the design and validation of interference testing and measurements. Figure 2 shows a comparison of the calculated and measured power spectra for UWB emissions that were used to test interference effects for an RF communications system. The good agreement between measurements and predictions lends credibility to the testing methodology. By comparing test equipment emissions and theoretical predictions, the test procedures and methodologies can be properly validated.

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Software Defined Radio Technology

Outputs

- Survey of tools and methodologies for implementation of software defined radio (SDR) signal processing algorithms in field programmable gate arrays (FPGA).
- Development of methodology to implement system-level signal processing algorithms in FPGA's.
- Implementation of a wireless local area network (LAN) despreaders in an FPGA and performance verification at 25 Msamples/second.

A software defined radio (SDR) consists of an SDR receiver and/or an SDR transmitter. In an SDR receiver, the received signal is digitized and then processed using software-programmable digital signal processing techniques. Digitization may occur at the RF, IF, or baseband. In an SDR transmitter, the modulated signal to be transmitted is generated as a digital signal using software-programmable digital signal processing techniques. The digital signal is then converted to an analog signal for transmission. The conversion to analog may also occur at RF, IF, or baseband.

Awareness of the viability of, potential market for, and potential spectrum management impact of SDR's continues to grow as evidenced by an increasing number of SDR development efforts and actual products, the significant growth of the SDR Forum membership, and the recent Federal Communications Commission (FCC) Notice of Inquiry on Software Defined Radios. Because of the increasing importance of SDR's in wireless communications, the Institute has been involved in SDR research over the past five years.

Key areas of research in SDR's include data conversion (analog-to-digital and digital-to-analog), digital signal processing, and linear power amplification. Research in SDR's at the Institute in FY 2000 focused on the digital signal processing aspects of SDR's. Digital signal processing in SDR's can be accomplished by the use of application specific integrated circuits (ASIC), digital signal processing

integrated circuits, field programmable gate arrays (FPGA), or general purpose processors. A combination of these types of devices can also be used to perform the digital signal processing in an SDR.

Digital signal processing research in SDR's at the Institute specifically focused on implementation of system-level SDR signal processing algorithms in FPGA's. The rapid advances in the gate capacity and operating speed of FPGA's has opened up an opportunity for performing real-time digital signal processing on a single FPGA that was not possible several years ago. The impact of research on implementation of system-level signal processing algorithms in FPGA's is not only significant in its own right, it actually encompasses many other major research areas such as hardware/software co-design, rapid prototyping, system-on-a-chip (SOC), reconfigurable computing, and electronic design automation.

There is a large gap between a system level or algorithmic description of signal processing and the implementation of algorithms in an FPGA. System level signal processing is easily carried out with high-level programming languages (such as C) or with system-level simulation block diagram tools. Tools for design with FPGA's are abundant and include both schematic capture and hardware description language synthesis tools. The problem is in traversing from the world of system-level processing to the FPGA design world. Traditionally, this "bridging of the gap" has been done by using a manual translation of the system-level design into a design suitable for implementation in FPGA's.

The goal of the research at the Institute was to develop an efficient methodology (design process) for implementing SDR signal processing algorithms in FPGA's. A methodology in which a system level or algorithmic description of the processing can be implemented in an FPGA in the most efficient and automated manner was desired.

The level of abstraction at which the FPGA design is carried out is a metric for research in the field of implementation of signal processing algorithms in FPGA's. The goal is to be able to carry out the FPGA implementation at the highest level of

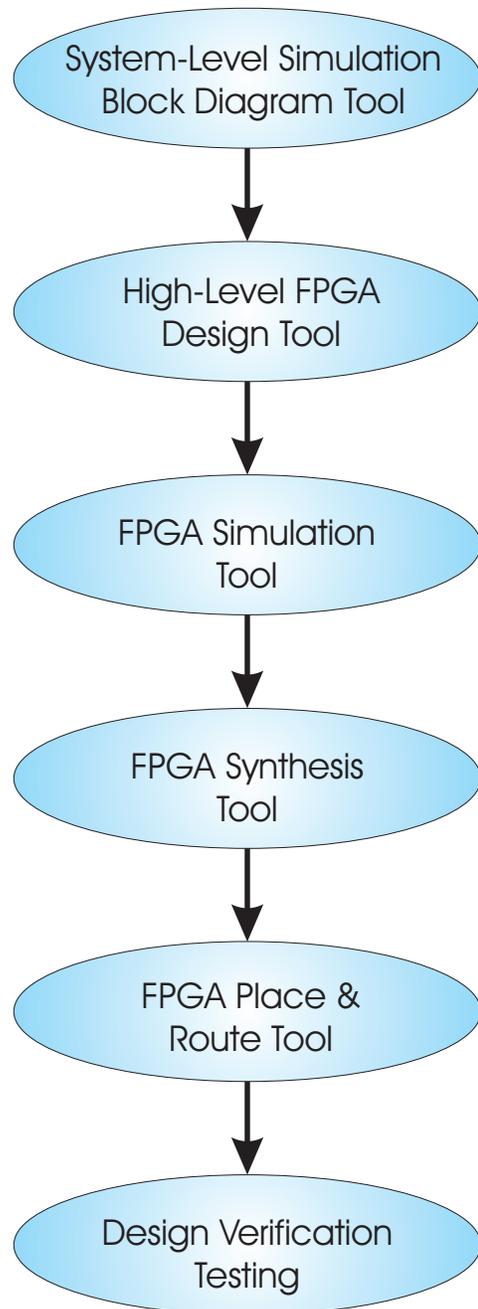
abstraction possible, ideally right at the system level. Traditional FPGA design uses a fairly low level of abstraction by designing at the hardware architecture level. Raising this level of abstraction is a goal that is being addressed by many research efforts at this time.

A wide variety of tools were examined to assess their potential usefulness in the process of implementing signal processing in FPGA's. The tools examined fell into the following general categories: 1) system-level simulation block diagram tools, 2) automatic translation tools that convert code from high-level programming languages into a hardware description language such as VHDL, 3) object-oriented system design tools such as the specification and description language (SDL), 4) a high-level FPGA design environment tool, and 5) an object-oriented hardware description language.

While many tools exist to aid in the process of implementing system-level signal processing in FPGA's, and there have been and continue to be efforts to automate the process, our research revealed that automation has only been successful under certain limited and simple circumstances. Full automation of the process is indeed in its infancy.

Since an efficient, fully-automated process was not yet available, a design methodology was developed that utilizes a commercially available, high-level FPGA design environment tool and a commercially available system-level simulation block diagram tool. The combination of these tools along with FPGA simulation, synthesis, and place and route tools provides an efficient semi-automated design methodology. A high-speed pattern generator and logic analyzer are used for design verification by providing real-time stimulus and response testing of the FPGA. The block diagram in the Figure summarizes the design methodology and verification.

Verification of the design methodology was initially accomplished by demonstrating the implementation of a matched filter correlator for a wireless LAN in an FPGA. Real-time performance of the matched filter correlator FPGA implementation at 25 Msamples/second was demonstrated. A wireless LAN despreader was also implemented in an FPGA and successfully tested at 25 Msamples/second.



Design methodology and verification for implementation of system-level signal processing algorithms in FPGA's.

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Video Quality Research

Outputs

- Digital video quality measurement technology.
- Journal papers and 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 replacing all existing analog video systems and making possible the creation of many new telecommunication services (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 video performance of these systems are required by government and industry for specification of system performance requirements, comparison of competing service offerings, network maintenance, and optimization of the use of limited network resources such as transmission bandwidth. 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 U.S. telecommunications industry. The increases in quality of service made possible with the new measurement technology benefit both the end-users and the providers of telecommunication services and equipment.

To be accurate, digital video quality measurements have to be based on perceived “picture quality” and have to be made in-service using the actual video being sent by the users of the digital video system. The primary reason for these requirements is that the performance of digital video systems is variable and depends upon the dynamic characteristics of both the input video (e.g., spatial detail, motion) and the digital transmission system (e.g., bit-rate, error-rate). To address this problem, ITS developed the revolutionary approach shown in Figure 1, based upon extraction and comparison of low bandwidth perception-based features (e.g., edges, motion) that can be easily communicated throughout the broadcast network. The new measurement paradigm has received two U.S. patents, been adopted as an ANSI standard (ANSI T1.801.03-1996; see Publications Cited, p. 102), and is being used by organizations worldwide.

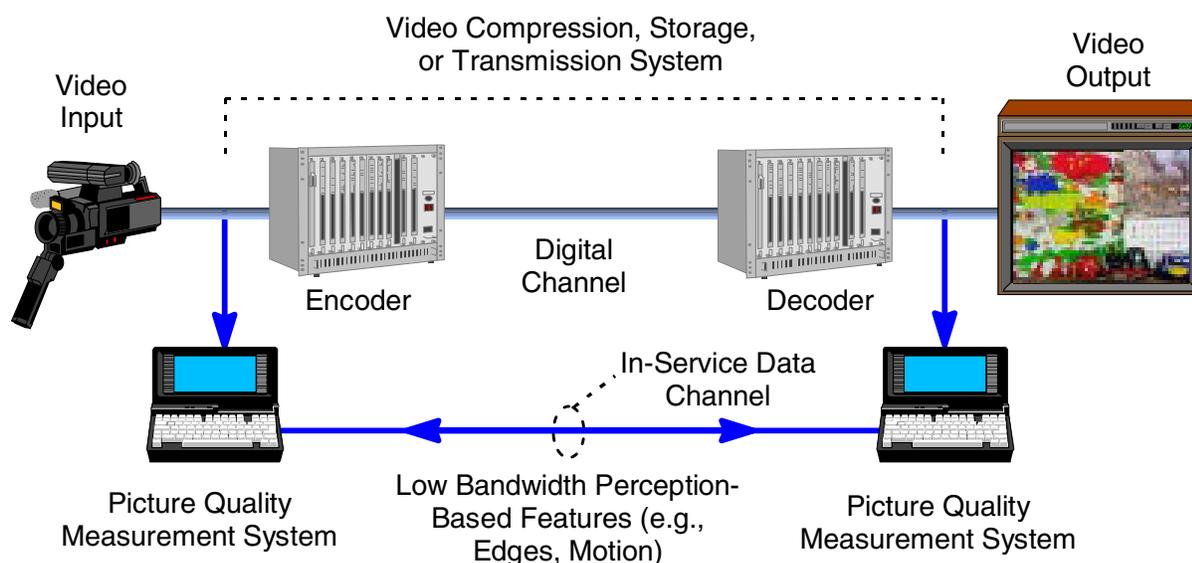


Figure 1. In-service perceptual picture quality measurement system.

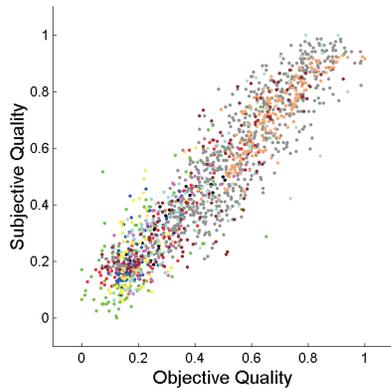


Figure 2. Objective predictions versus subjective data for each video clip.

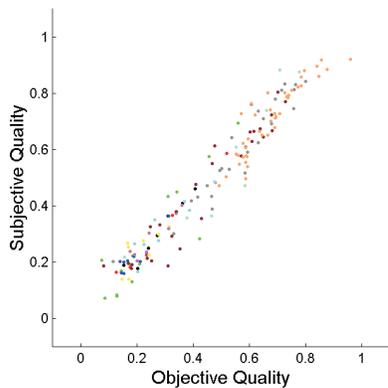


Figure 3. Objective predictions versus subjective data for each video system.

ITS continues to make refinements to the technology and to apply it to an ever-wider range of video scenes and systems. In FY 2000, eleven subjective (i.e., rated by viewers) and objective (i.e., rated by quality metrics) data sets were combined using a newly developed least squares fitting algorithm. The eleven video data sets covered 158 video systems operating at bit rates from 10 kbits/sec to 45 Mbits/sec and 115 video test scenes. ITS objective quality metrics produced the scatter plots shown in Figures 2 and 3. The horizontal axis gives the objective quality predictions while the vertical axis gives the subjective mean opinion scores. Each data set is plotted in a unique color and the scale of the plot is normalized such that 0 is no perceived impairment while 1 is maximum perceived impairment. Figure 2 gives the results for each clip (i.e., a given scene sent through a given video system) while Figure 3 gives the results for each video system (i.e., averaged results for a given video system). Figures 2 and 3 achieved correlation coefficients between the objective data and the subjective data of 0.94 and 0.98, respectively.

To facilitate the transfer of this promising technology to private industry, both UNIX-based and Windows®-based automated video quality measurement software has been developed. This software includes (1) video calibration, including video system gain/level correction and spatial/temporal registration of the input and output video streams, (2) four objective video quality metric calculations, optimized for specific digital video applications (TV, Videoconferencing, General – wide range of quality, Developer – wide range of quality + fast computation), and (3) root cause analysis summaries that give the perceptual relations between the objective metrics and specific digital coding artifacts (e.g., blurring, block distortion, jerky or unnatural motion, added noise, error blocks). Figure 4

is a photograph of the graphical display presented to the user by the Windows®-based automated video quality measurement software.

Further information and a list of recent publications can be found on the Video Quality Research home page at

<http://www.its.bldrdoc.gov/n3/video>

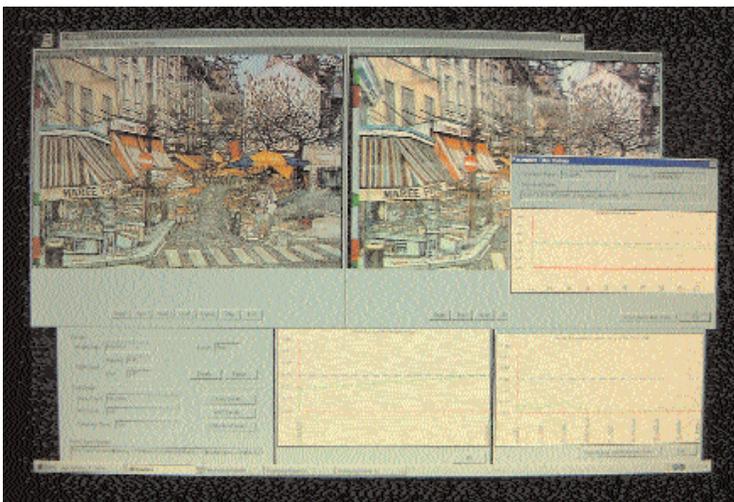


Figure 4. User display presented by Windows-based video quality measurement system (photograph by F.H. Sanders).

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Wireless Propagation Research

Outputs

- Methods of predicting the impulse response for an indoor propagation channel and the delay spread of the multipath channel.
- Assessment of the geometric optics approximation for indoor ray-trace models.
- Identification of a pseudo-lateral wave phenomenon.

For several years the Institute has been involved in research efforts related to wireless communication applications and theory. The majority of this work has been related to the outdoor propagation environment. Recently, with the emergence of new indoor wireless local area networks and wireless local campus networks, we have concentrated our efforts on investigating the indoor propagation environment. The objective of this effort is to support new wireless technology development and help U.S. industry compete in the worldwide telecommunications marketplace. More specifically, ITS develops models and measurement systems to predict and measure propagation characteristics of various multipath environments. This work supports the advancement of new techniques and technologies (e.g., smart antennas and diversity) to overcome limiting factors for indoor communication systems.

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, characterization of an anechoic chamber, and analyzing the coupling mechanisms between rooms. Figure 1 shows a typical calculated impulse response from this model. While the ray-trace technique is accurate, it can be time consuming. The Institute has also developed a simplified model for calculating the impulse response and delay spread for the indoor channel in a matter of seconds on a personal computer. Also shown in Figure 1

are results for this simple model. Notice that the simple model captures the delay characteristics of the ray-trace model.

In an attempt to assess the validity of the ray-trace model, we have investigated the accuracy of some assumptions used in ray-tracing. 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 2 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.

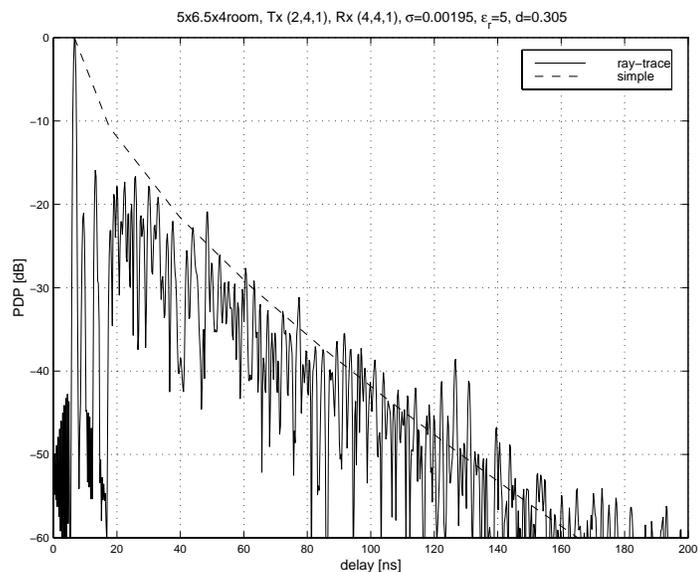


Figure 1. Calculated power delay profile from both ray-tracing and the simple model.

The Institute is also involved in measurement efforts for various indoor propagation scenarios. In an attempt to understand antenna polarization and directivity effects indoors, impulse response measurements were acquired in a wide range of indoor environments employing different types of antennas. The results of one set of these measurements are presented in Figure 3.

Results indicated less linearly-polarized (LP) basic transmission loss than circularly-polarized (CP) basic transmission loss for both line-of-sight and obstructed channels. Also, LP rms delay spread was similar to CP rms delay spread in both line of sight (LOS) and obstructed (OBS) paths. The apparent advantage of using LP signals over CP signals indoors may be attributed to the relatively high degree of circular depolarization measured. Results also supported the use of omnidirectional antennas indoors to improve signal coverage. Omnidirectional measurements, however, demonstrated large delay spreads for some extraneous cases.

Recent Publications

M.G. Cotton, E.F. Kuester, and C.L. Holloway, "A frequency- and time-domain investigation into the geometric optics approximation for wireless indoor applications," NTIA Report 00-379, Jun. 2000.

M.G. Cotton, R.J. Achatz, Y. Lo, and C.L. Holloway, "Indoor polarization and directivity measurements at 5.8 GHz," NTIA Report 00-372, Nov. 1999.

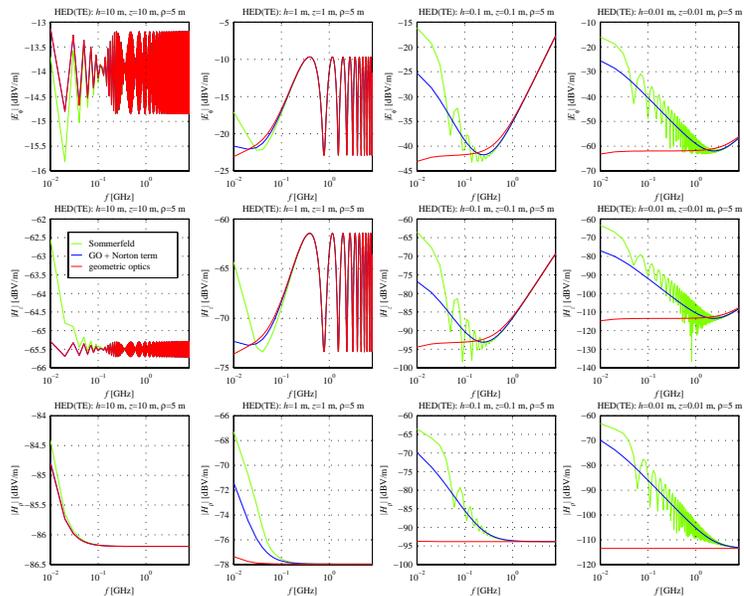


Figure 2. Near-surface effects on field strength of an elementary horizontal electric dipole above a concrete half space.

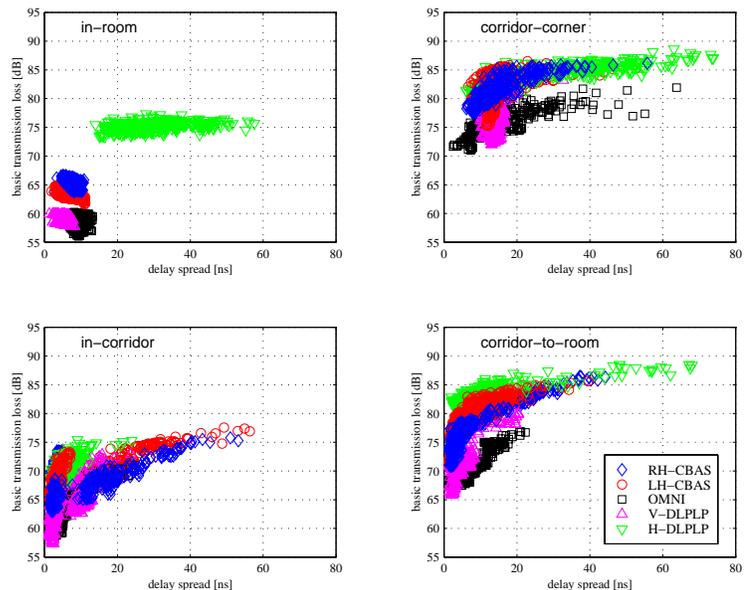


Figure 3. Scatter plots of basic transmission loss versus delay spread of individual impulses for a V-OMNI transmit antenna and various receive antennas.

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