Telecommunications Theory

The worldwide telecommunications network is undergoing explosive growth, in terms of both the build-out of hardware infrastructure and ever-increasing loading demands. Wireline and radio networks generally complement each other, but whereas wireline capacity can be expanded almost indefinitely, radio spectrum is a limited resource. Ever-increasing demands for radio spectrum are driving the development of new radio technologies that promise to use spectrum more efficiently and effectively. The basic paradigm of radio spectrum management is responding to these changes by moving away from traditional, top-down frequency-assignment methods and is migrating instead toward autonomous, interference-limited technologies. Historically rigid spectrum band allocations on a service-by-service basis (i.e., differentiation of band usage on service bases) are being supplanted by more sharing of radio bands by multiple services. The new sharing schemes emphasize the need for new capabilities in radio systems to recognize and avoid interference between each other.

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 new and existing 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 continuously 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 develops and evaluates new techniques for encoding, decoding, and analyzing speech and audio signals. 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.

Interference Effect Measurements in an Allocated Radar Band In response to interference problems resulting from non-radar signals in an allocated radar band, the Institute has performed an extensive set of analyses and measurements. This problem, which had originally been predicted by ITS engineers on a theoretical basis, is now a reality. It is being assessed and solutions are being sought. 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. These 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 encoding and transmission of speech signals enables many telecommunications services including voice over Internet protocol (VoIP) services, cellular telephone services, voice messaging services, and automatic voice recognition call processing systems. Similarly, general audio signals (including music, voices, environmental sounds, and sound effects) can be digitally encoded and transmitted and this has enabled satellite and terrestrial digital audio broadcasting services and Internet digital audio streaming services. Key to all of these services is the efficient and robust encoding of speech and audio signals at low bit-rates while maintaining good fidelity.

But digital encoding of speech and audio involves compromises and trade-offs among 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 important to the description of the signal and the data stream is thus less robust to errors and losses. The trade-off between delay and robustness can be demonstrated by 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 a very elusive goal.

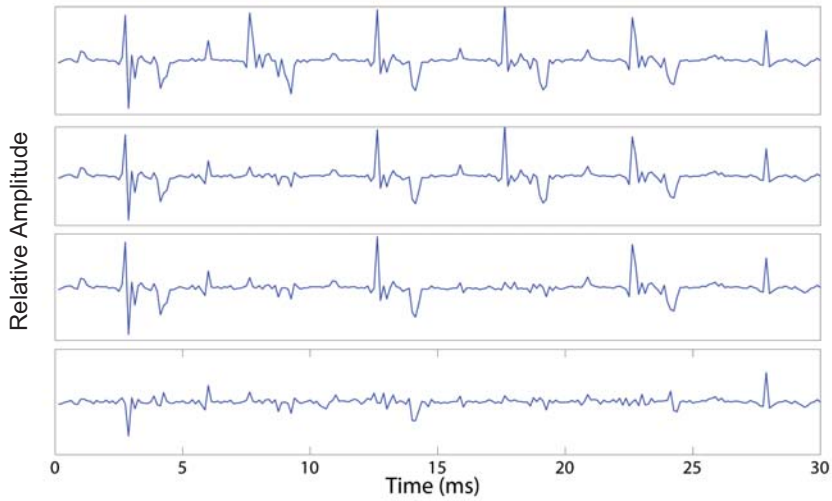
The ITS Audio Quality Research Program identifies, develops, and characterizes new techniques with potential to increase relative quality or robustness or reduce relative bit-rate, delay, or complexity of speech and audio coding and transmission algorithms. In addition, the Program seeks to develop techniques that aid in optimizing the trade-offs between these five factors. To optimize, one must be able to quantify. Quantification of speech and audio quality is a particularly difficult challenge. Thus a key component of our efforts is the development of more effective and efficient ways to characterize speech and audio quality. The ultimate goal for Program efforts is more reliable and efficient telecommunications and broadcasting services with improved speech and audio quality.

In FY 2008 Program staff performed significant work on the problem of high quality speech and audio coding at reduced bitrates. The extreme in high quality coding is lossless coding, where a bit-exact copy of the original signal is reproduced at the decoder output. Transform coding, predictive coding, and entropy coding techniques were studied, extended, and analyzed for applicability to several specific audio and speech applications.

Efficient coding requires that redundancy be eliminated from signals as part of the coding process. One novel technique to eliminate temporal redundancy in speech prediction residuals is demonstrated in the figure on the next page. The top panel shows a speech prediction residual signal that must be transmitted. It is clear that a basic pattern (related to the glottal pulse pattern and the pitch) approximately repeats at intervals of about 5 ms. Through an iterative shifting, scaling, and adding process, the signal can be replaced with the one shown in the second panel, then the third, etc., and this process is invertible. By iterating this process we ultimately arrive at the signal shown in the bottom panel. By reversing the process we can generate the top panel signal from the bottom panel signal.

The final signal, shown in the bottom panel, can be coded at a lower rate than the original signal shown in the top panel. In order to realize this rate-reduction benefit, it is necessary to transmit the side information that will allow for the inversion of the iterative reduction process described in the figure. Thus one must weigh these two factors against each other to determine if this technique could be advantageous in a given speech or audio encoding application.

In other FY 2008 work Program staff developed, implemented, and verified a new subjective testing protocol addressing speaker identification. This protocol quantifies the extent to which various speech communication links can convey or obscure the identifying characteristics of a specific talker's voice. Project staff selected a set of speech recordings from three female and three male speakers to conform with very specific criteria of content and quality. They then processed the recordings through various speech communication links and wrote a sophisticated subjective test controller that provided listeners with a set of task-specific graphical user interfaces. After the appropriate training process, listeners heard recordings and were asked to select which of the six talkers they had just heard. When combined with parallel results from earlier intelligibility tests, staff determined that the cues associated with speaker identification can be fairly robust. In fact, even when distortions are so severe that it is not possible to determine what is being said, it can still be possible to determine who is talking.



An example speech prediction residual signal (top panel) and the results of an invertible iterative process (second, third, and fourth panels) that reduces the bit rate necessary to encode the signal, at the cost of some additional side information. Vertical axis displays relative amplitude.

Throughout FY 2008, Program staff continued with additional speech and audio quality testing using both objective and subjective techniques. These tests support both this program and other ITS programs. Some laboratory upgrades were performed 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 Publications

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

J.B. Allen, G. Chan, and S.D. Voran, "Perceptual models for speech, audio, and music processing," *EURASIP Journal on Audio Speech, and Music Processing*, Volume 2007, Nov. 2007.

S.D. Voran, "Listener detection of talker stress in low-rate coded speech," in *Proc. IEEE International Conference on Acoustics, Speech and Signal Processing,* Las Vegas, Nevada, Mar. 2008.

A.A. Catellier and S.D. Voran, "Speaker identification in low-rate coded speech," in *Proc. 7th International Measurement of Audio and Video Quality in Networks Conference*, Prague, Czech Republic, May 2008.

S.D. Voran, "Estimation of speech intelligibility and quality," Ch. 28 of *Handbook of Signal Processing in Acoustics*, D. Havelock, S. Kuwano, and M. Vorländer, Eds., Vol. I, Part V, Springer Science+Business Media, LLC, 2008, pp. 483-520.

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Effects of the Channel on Radio System Performance

Outputs

- Interference susceptibility analysis.
- Translations of channel measurements to receiver performance metrics.

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, i.e., the *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.

The channel degrades radio system performance by introducing multipath and undesired signals. Multipath 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. Undesired signals include natural noise created by phenomena such as lightning, man-made noise created by electrical devices, and signals from other radio systems.

Multipath is, in essence, channel filtering which can potentially cause intersymbol interference and attenuation. Undesired signals, as well as multipath attenuation, reduce signal power margins. Intersymbol interference and reduced signal power margins cause increased numbers of bit errors and corresponding reductions in radio system performance.

Results from our work are useful to radio system designers and radio spectrum managers. Radio system designers often want channel characteristics reported in terms of their effect on radio system performance. Radio spectrum managers need to be able to evaluate the susceptibility of legacy receivers to signals from other radio systems. This need has become more pressing as spectrum managers seek to alleviate spectrum scarcity through novel spectrum sharing techniques.

Our approach towards evaluating the effects of the channel on radio system performance emphasizes our expertise in channel characterization and radio system signal processing. This expertise allows us to pare down complex problems to simple, verifiable models and experiments with those models.

In FY 2007 we developed a computer simulated radio link model, shown in Figure 1, to evaluate the effects of the radio channel on radio system signal processing algorithms. The model is built around a radio link core (RLC) which provides reliable modulation, channel degradation, and demodulation operations. Tethered to this core are the various combinations of channel and signal processing subsystems being tested. A transmitter premodulator subsystem impresses information on source bits needed by the receiver signal processing.

The model uses Monte Carlo techniques to generate bit error sequences. The bit error sequences are used to compute first order bit error rate (BER) and the second order error free interval (EFI) statistics. The first order BER statistic is used to evaluate how effectively the receiver rejects interference caused by the undesired signal. Briefly, a link operating at a 10^{-4} BER in the absence of the undesired signal is subjected to increasing levels of the undesired

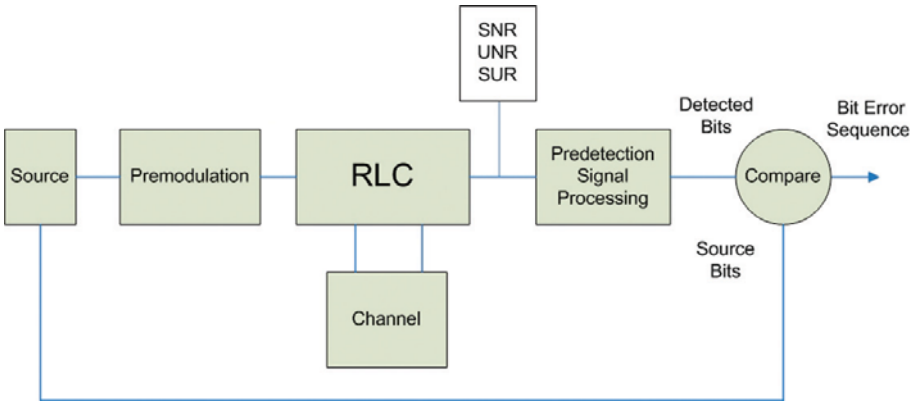

Figure 1. Block diagram of radio link model.

signal. Our performance metric is the undesired signal to noise ratio (UNR) needed to drive the link to 10^{-3} BER. Higher values of UNR correspond to greater effectiveness. In general, our interest is usually in whether or not it is greater than that of Gaussian noise.

The second order statistics are used to determine whether the bit errors are independent. If they are independent, rejection by subsequent block forward error correction can be quantified with analytic expressions much faster than they can with Monte Carlo simulation. This is extremely important at the low BER after block forward error correction (FEC) where the execution times of Monte Carlo simulation are much too long.

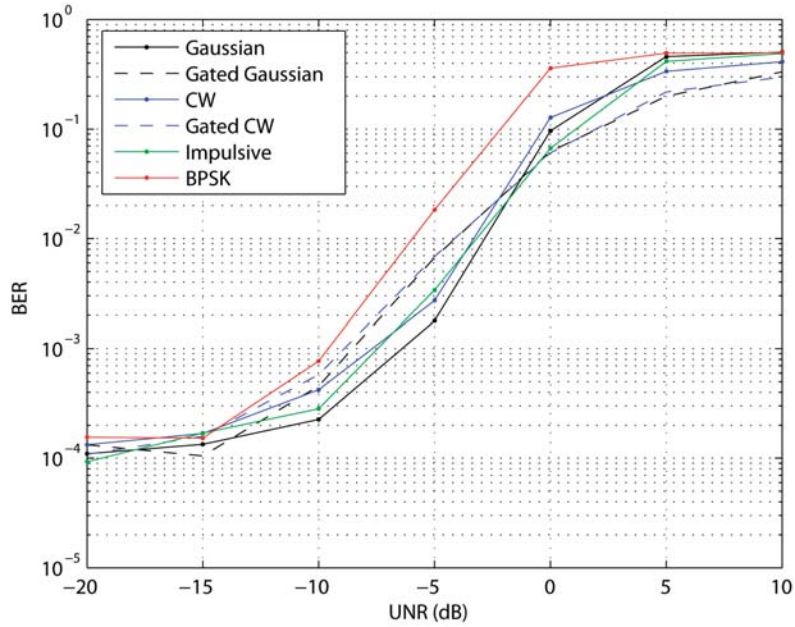

Figure 2. Bit error rate (BER) versus undesired to noise ratio (UNR) for Viterbi configuation and various undesired signals. UNR at 10^{-3} is our performance metric.

Initially the model used the simplest of signal processing algorithms — a threshold detector. In FY 2008, three new configurations were developed based on Viterbi detection, equalization, and correlation signal processing algorithms. Simulations using the three new configurations were run with continuous wave, impulsive, binary phase-shift keying, gated Gaussian, and gated continuous wave signals. Results for the Viterbi configuration, shown in Figure 2, indicate that the performance degraded to 10^{-3} with less power than Gaussian noise for all the undesired signals. In other words, the Viterbi detector was less effective in rejecting interference due to the undesired signals as compared to Gaussian noise. In fact, we found that of 15 combinations of signal processing algorithms and undesired signals, a total of 12 were less effective.

In addition, bit errors at 10^{-3} were found to be independent for the equalization and correlation configurations but not for Viterbi. Therefore rejection by subsequent block forward error correction can be determined with efficient analytic expressions for the equalizer and correlation configurations but not for the Viterbi.

Purely analytic work advanced in FY 2008 with the technical review of "Statistical considerations for noise and interference measurements" by R.A. Dalke. Methodologies in this report (published in FY 2009) show how uncertainties of statistics (mean power, APD percentiles, and power spectral density) commonly computed from noise and interference measurements can be evaluated. This work lays a firm scientific foundation to our efforts. Work has also begun on deriving an analytic expression for the bit error rate of a Viterbi detector in the presence of a gated Gaussian noise signal. This effort is critical for verifying simulation of legacy radio systems operating in the presence of ultrawideband radio signals which resemble gated Gaussian noise.

Recent Publication

M. Cotton, "A methodology for approximating BPSK demodulator performance in the presence of various undesired signals," in "Proceedings of the International Symposium on Advanced Radio Technologies/ClimDiff 2008," P.J. Raush and K.E. Davis, Eds., NTIA Special Publication SP-08-452, Jun. 2008, pp. 36-42.

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Interference Effect Measurements in an Allocated Radar Band

Outputs

- Predictions of the effects of interference from non-radar systems with air surveillance radars.
- Development of plans to measure these effects and verify the predictions of theoretical models against actual effects.
- Measurements at field sites during interference events, and injection of simulated interference signals at those sites to further verify theoretical predictions.
- Analysis of measurement data to formulate methods for coping with interference, if possible.

Contrary to popular impressions and portrayals, radar receivers are not robust against the effects of radio interference from non-radar systems. But the actual effects of interference, and the thresholds at which various types of interference cause degradation of radar system performance, have not been well understood on a technical, quantifiable basis. For the past decade, NTIA has conducted an aggressive effort to thoroughly understand these effects. The purposes of this program are four-fold: to predict the effects of new, non-radar systems that might try to share systems with radars; to provide technical advice to other agencies and the President regarding the threats that new interference sources might pose to defense and safety-of-life radars; to determine the extent to which new technologies might be able to make spectrum sharing between radars and non-radar systems more technically feasible; and to devise new technical methods that might possibly mitigate the effects of existing interference.

Past work in this area by NTIA has been broadly directed toward the characteristics and performance of the following types of radars in the presence of interference: air traffic control; airport surface search and control; air defense; ground-based and aeronautical meteorological; and maritime navigation and surface search.[1] In the past year, special attention has been focused on the effects of interference to air search radars in particular. NTIA engineers from the Institute, working closely with their colleagues at the Office of Spectrum Management (OSM) and other Federal agencies, have undertaken predictions and analyses of interference effects from non-radar systems on these radars. This effort has required careful advance work to analyze and predict the effects of the interference, followed by actual measurements of interference at operational facilities.

In FY 2008, specific radars were identified for injection of signals that have caused interference in operational systems. Both the actual interference signals and hardware replications of these signals, made with vector signal generators, or VSGs, were used during highly detailed tests and measurements at operational radar sites. The FY 2008 results have reinforced the conclusions of existing, published results from NTIA, that radars do not perform well in the presence of even low-level signals that are as much as 6-10 dB below the noise floors of radar receivers. The Institute and OSM plan to continue these analyses and measurements in FY 2009, and are incorporating the results into analyses of the overall spectrum efficiency of radars.

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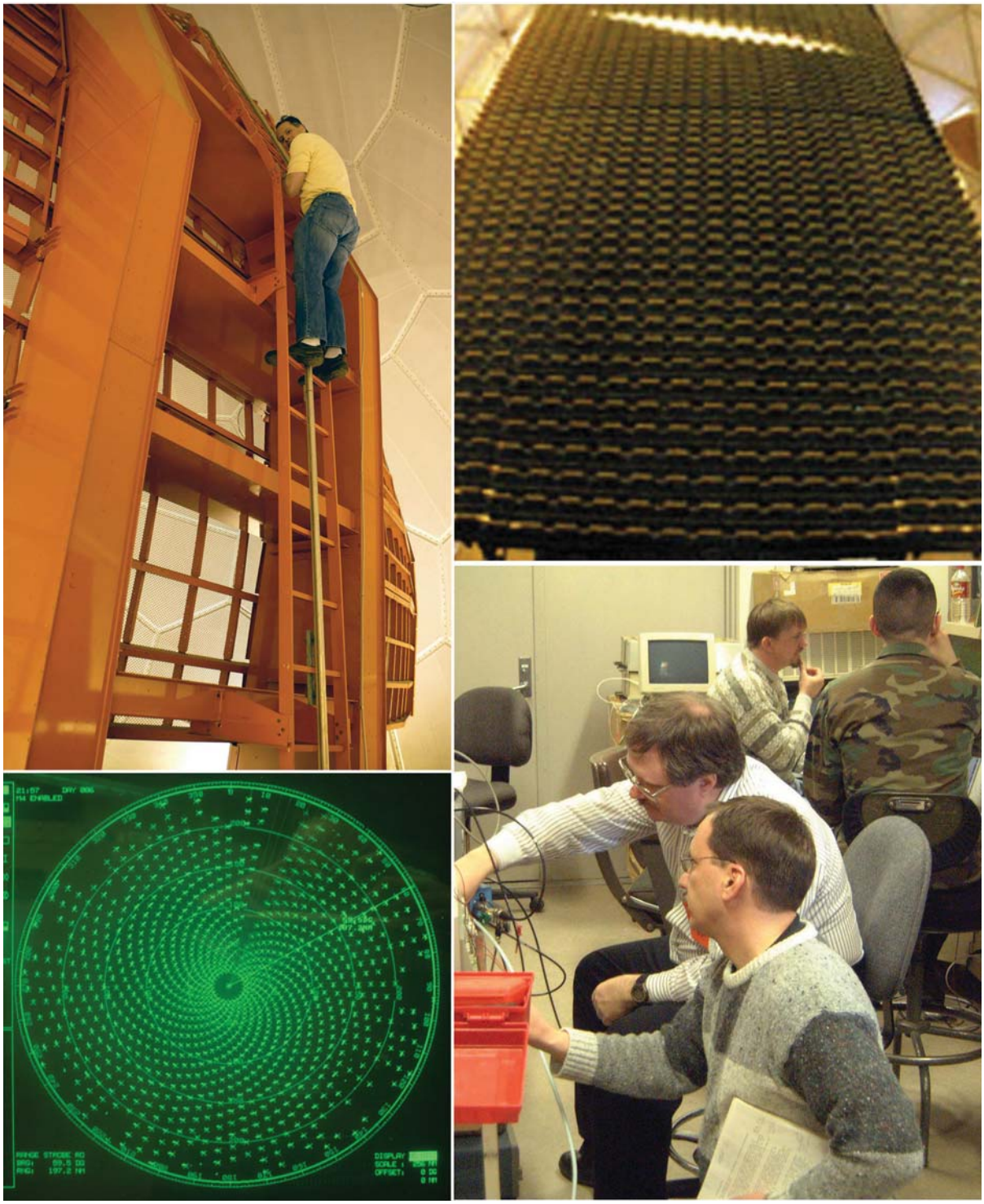
1 F.H. Sanders, R.L. Sole, B.L. Bedford, D. Franc, and T. Pawlowitz, "Effects of RF interference on radar receivers," NTIA Technical Report TR-06-444, Sep. 2006.



Clockwise from the upper left: An NTIA engineer climbing a radar antenna; a phased-array air surveillance radar antenna; a team of ITS engineers and engineers from other agencies during interference tests and measurements; a radar data display.

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 RR or "Reduced Reference" measurements) have received four U.S. patents, been adopted as the North American Standard for measuring digital video quality (ANSI T1.801.03-2003), been included in three International Recommendations (ITU-T Recommendation J.144, J.244, 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 the Windows and Linux operating systems. Using these software tools, users and service providers can quantify the digital video quality of their networks using methods standardized by ANSI and the ITU. The first tool, called Command-Line VQM (CVQM), provides a simple command

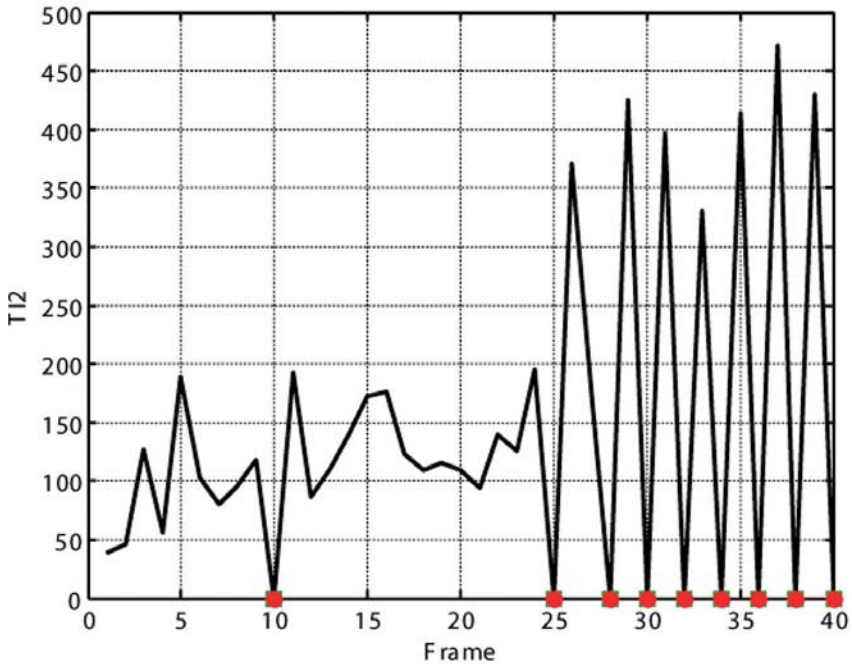


Figure 1. NR metric detecting dropped video frames.

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. Source code and executable binaries of these tools are available royalty free to all interested parties.

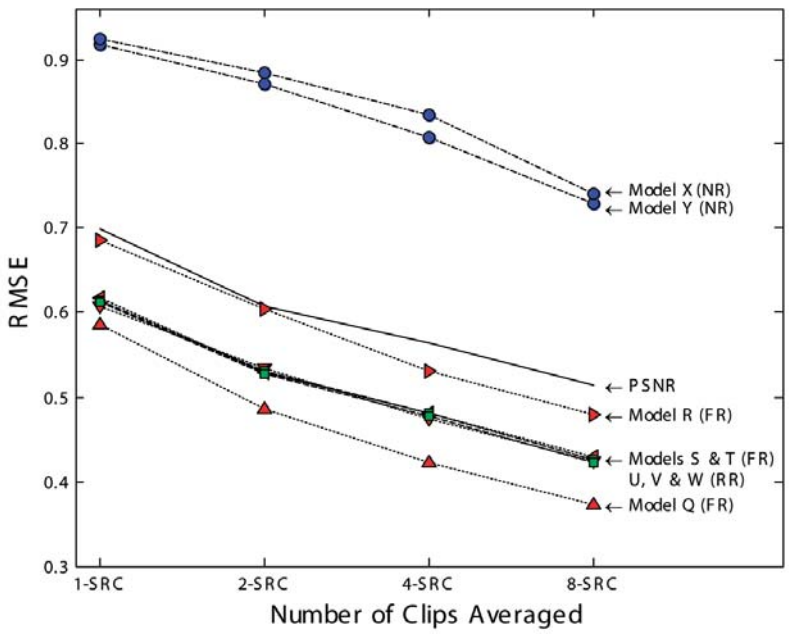


Figure 2. Video quality model RMSE vs. number of clips averaged.

In FY 2008, a No Reference (NR) video quality metric was developed that detects dropped or repeated video frames, which visually appear as pauses in the flow of motion. NR metrics do not require any reference information from the source video stream and are thus easy to implement for in-service quality monitoring. There are two primary reasons why digital video systems drop frames: (1) the video encoder may decide to reduce the video frame transmission rate in order to save bits, and (2) the video decoder may decide to freeze the last good video frame when transmission errors are detected. Figure 1 shows a plot of the scene motion energy (i.e., the square of the Temporal Information, denoted as TI2) and the detected frame drops (shown in red) where the video coder performs dynamic frame dropping that depends upon the amount of motion in the video scene. Low motion portions of the scene are transmitted at full or near full frame rates (from frames 1 to 24) while high motion portions are transmitted at reduced frame rates by dropping selected video frames (from frames 25 to 40).

ITS worked as an independent laboratory within the Video Quality Experts Group (VQEG) to validate objective video quality models for mobile/PDA, and internet communication services. One of ITS' investigations examined how the models' accuracy changes when several scenes are passed through a single system, and the scores for those scenes are averaged to produce an average quality score for the system. Figure 2 depicts the accuracy of ten models as an increasing number of scenes are averaged. The root mean squared error (RMSE) accuracy improved steadily for all models as the number of scenes averaged increased from 1 to 2 to 4 to 8.

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

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