SUBJECTIVE EVALUATION OF DIFFERENT ERROR CORRECTION SCHEMES FOR APPLICATION WITH A 900 MHZ FREQUENCY HOPPER COMMUNICATION SYSTEM

Kimberly Braaten (1), Dean Foster (2), Bruno Korst-Fagundes (2) and Haoye Shen (2)

(1) Faculty of Engineering, University of Regina, Regina, SK, S4S 0A2
(2) Acoustics Group - ENC Nortel, 3705 35th St. N.E., Calgary, AB, T1Y 6C2

SUMMARY

This paper presents an analysis of subjective tests conducted with a 900 MHz frequency hopper. The primary purpose of such a study is to evaluate the listeners' response to radio channel interference causing loss of speech data. The pre-testing phase determined the users' preference for a 32 kbits/s Adaptive Delta Pulse Code Modulation (ADPCM) system. The three main tests performed determined the parameters of the error correction schemes most preferred by the listeners. The purpose of the tests performed was three-fold: to determine the type of correction scheme preferred by the listeners, to subjectively evaluate the performance of the preferred correction scheme, and to determine the response of the listeners to different interference scenarios. Results from the subjective testing performed are presented and analyzed in this report.

SOMMAIRE

Ce papier présente les resultats d'une étude sur la qualité de reception d'un radio 'frequency hopper' en présence de signaux parasites. Différentes methodes de corrections sur la perte de donnés fut analysées. La qualité accoustic fut mésuré subjectivement en déterminant la réaction des écouteurs en simulant la perte et correction de données en presence de signaux parasites variées.

1. INTRODUCTION

Wireless communication systems are required to communicate information quickly, efficiently and securely over selected frequency ranges. Each channel has a limited bandwidth or frequency range in which it must operate and the frequency spectrum is becoming increasingly more congested.

Spread spectrum is a modulation scheme that uses the spectrum efficiently. It can carry many uses with a reasonable level of security and operates with a minimum amount of interference [9]. In a spread spectrum system, the signals are spread over a wide range of frequencies by using a variety of broadband or frequency hopping techniques.

"Frequency hopping" is one of the spread spectrum techniques which uses special coding techniques, or pseudo-random sequences, sent between the transmitter and receiver, to determine the frequency that carries the digital information, or the carrier frequency [3] (see Figure 1).

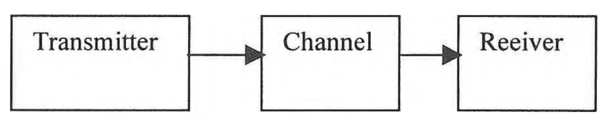

Figure 1 Block diagram of a communication system

In a frequency hopping system, the users are bounced from one available frequency to another during communication. This technique gives the most effective use of the available bandwidth as well as increased signal power without compromising capacity. However, interference is still present and subjectively noticeable in some circumstances with the use of frequency hoppers. The effect of having many users utilizing the same frequency bandwidth promotes a special problem since it becomes possible for one user to jam the signal of another. Jamming occurs when two transmissions interfere with each other on the same receiver unit (see Figure 2). This creates noise or other user perceived anomalies that considerably degrade the audio quality. Jamming can occur when the transmitted signal is interrupted, from a wall in an office building, for example, causing the receiver to pick up spurious data. In the particular case of frequency hoppers, jamming can also be caused by the carrier frequencies of two users hopping to the same frequency at the same instantaneous time. Errors, caused by jamming, can be introduced into the signal from anomalies inherent in the transmit and receive modes of a wireless communication unit transporting digital information. During the process of converting analog speech into digital data to be transmitted, redundancies or errors could become part of the speech waveform as well as errors that can be introduced through corruption in the radio transmission medium (free air). These errors are quantified through the bit error rate (BER). An error can occur in transmission from the receiver to transmitter, from transmitter to receiver or from transmitter to transmitter

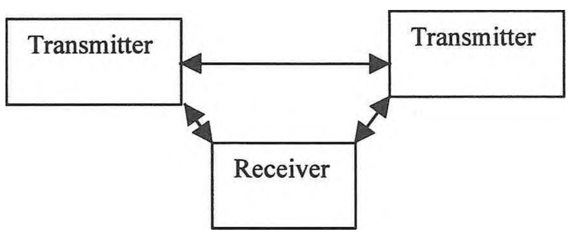

Figure 2 Jamming can occur between receiver and transmitter as well as between transmitters.

as shown in Figure 2 above, which is a real world problem. The bit error rate (BER) is the probability of an error occurring in a bit, or a change in the transmitted information. This is defined and set by the software we had written to incorporate errors in the given speech samples used for testing.

In this paper, subjective testing was performed on two types of interference associated with such a frequency hopping system where a number of units share the same frequency bandwidth. Software was created which simulated the methods of correcting the 'errors' or lost data due to the effects stated above. Several techniques were used to correct corrupted data. These ranged from very simple techniques to very advanced and complicated error correction techniques. In this article we analyzed two of the simplest techniques. The first correction method studied, called 'repeating', used the previously sent block of data picked up by the receiver and then repeated it. A second correction method, called 'muting', simply muted any erroneous data that was picked up by the receiver. The results of subjective testing and the test methodology for the two correction methods are presented in this paper.

2. EXPERIMENT

Digital speech transmission systems could generate degradations that involve difficulty in the listening path. These degradations could be perceived to the end user as clicks, pops, distortion, fuzziness, etc. in the receive listening audio path. To account for the listening transmission path, eight second-long high quality recordings of both male and female voices, speaking Harvard sentences, were used to effectively create the receive transmission audio. Sentences, about eight seconds long, were deemd to be appropriate for this type of subjective testing. The sentences were recorded in a soundproof room [8]. The speech recordings, originally existing on DAT tapes, were then converted to a format understood by the computer sound card. This way, every subjective listener test person would listen to the same audio file each time creating a consistent test base. All of the files required for a particular test were then loaded onto the laptop computer and modified by custom software to incorporate various degrees and types of errors. The recordings were then played from the laptop through the computer's high quality sound card to an audio handset. All subjective testing took place in a low ambient noise sound room. For additional consistency the same handset was used for each of the cases. The results from this series of tests helped the designer's choose the best error correction scheme that was available to them. To assist the designers in making the correct decision from the results, the Mean Opinion Score or MOS method [3] was used to assess the subjective listener's opinions on the various audio samples.

The Mean Opinion Score (MOS) method is a standard method used extensively for subjective listening tests. "The MOS is an opinion scores that represents a listener's assessment of the quality of a speech sample expressed over an appropriately chosen scale. CCITT recommends the use of a five-point scale {excellent, good, fair, poor, bad} which is typically numerically mapped to the decimal {5,4,3,2,1} scale" [3].

Each of the listeners judged the material on its overall quality. Test 1 and Test 2 involved comparison tests or Degradation Category Rating (DCR) MOS tests. A reference audio sample, with uncorrected errors, was played to the listener followed by the same sample using a specific error-correcting scheme (either muting or repeating) for the DCR tests. Listeners rated their perceived increase or decrease in quality level against the reference sample. Test 3 used an Absolute Category Rating (ACR) method where only one sample was heard at a time. After hearing each sample, the listener was required to record their opinion. The ACR MOS test method is appropriate for situations where a few sentences would be heard in a group and where several methods of degradation would be used in a row.

The speech samples used in the listening tests contain audible errors created by software that simulated conditions of jamming and with various levels of BER. Because channel bandwidth is at a premium, there is a definite need for speech coding at low bit rates, while maintaining acceptable fidelity or quality of reproduction. A major motivation for bit rate reduction for voice coding is to allow enough available bandwidth for the data to share the same channel. There are fundamental limits on bit rate suggested by speech perception and information theory. The standard reference for high quality transmission is a 64 kbits/s PCM communication system (which typically corresponds to 8-bit samples at an 8 kHz sampling rate). A 64 kbits/s system typically produces 4.5 or more on a MOS scale when no errors are introduced. Using this as a reference system, a frequency hopper spread spectrum radio was investigated that supported 32 kbits/s ADPCM with possible error correction. A 32 kbits/s ADPCM system would have 4-bit samples at an 8 kHz sampling rate. The fewer bits used to relay the data, the fewer would be the mistakes in terms of the BER and in jamming. It was found through previous listening tests that a 32 kbits/s ADPCM communication system offered the best audio quality for the least number of bits.

Test 1 determined the type of correction scheme and the threshold of correction for errors preferred by listeners for corrected jammed signals. The threshold would determine the level of correction for errors used by the software. The threshold and error correction scheme (muting the error or repeating the previous block of information) preferred by listeners was established after averaging all of the scores on the MOS tests. Test 2 threshold levels were based on the results from Test 1. For Test 2, since jamming was of more concern for audio quality, the threshold parameters of Test 1 for jamming were incorporated into several selected BER's. Test 3 is based on the chosen threshold and error correction schemes determined from Tests 1 and 2. Test 3 determined when the audio quality would degrade for jamming as the numbers of users increased. It compared two different scenarios that might occur in a jamming situation. The listeners evaluated the audio quality when the jams occurred as users interfered with each other at the same time or when the interference occurred at different times. The recommended practice for subjective testing was to use at least 24 people to listen to each test [3]. For all tests presented in this paper, at least 24 people participated and reported their evaluation utilizing DCR or ACR MOS tests.

3. RESULTS AND DISCUSSIONS

3.1 Test 1

Preliminary testing determined that the subjective testing should focus more on jamming tests rather than BER tests. In this project, jamming contributed to the quality of the audio signal to a greater degree than does BER, meaning, if a signal was jammed, it ws much more noticeable to a listener than the BER factor. Therefore, Test 1 was performed to find out whether jamming using a correction scheme called muting or using the repeating method of a previous block was preferable. The listeners would find which threshold level was most acceptable using the DCR MOS subjective test method.

Each trial for this test involved comparing two speech samples derived from the same original speech sample. The original speech sample was corrupted with errors and became sample A. A second speech sample, sample B, took the A speech sample and corrected these errors with one of the error correction methods, muting or repeating, at a chosen threshold value (from 1 to 7). Each trial compared a speech sample with errors, called sample A, followed by a speech sample with the errors corrected, called sample B. Each subject gave a rating for each comparison, based on the rating schedule shown below in Figure 3 for a DCR MOS test. In this first round of tests, the data acquired from one subject was thrown out. The listener gave every speech sample the same rating. Since the degree in difference of audio quality was quite high between each sample it was thought that this particular listener had given us erroneous data. (**Note:** This was the only data for the complete set of testing, that was thrown out.)

The same	Slightly	Moderately	Much	Very much
Or poorer	better	better	better	better
Quality	quality	quality	quality	quality

1-------------2------------3-------------4------------5

Figure 3 DCR MOS Test rating schedule.

Figure 4 (shown on the next page) shows a graph describing the data for Test 1. It shows how for an increasing number of errors detected before being corrected (the threshold) the quality of the audio samples quickly degrades to less than 4.5 MOS test rating. The numbers along the bottom describe the threshold levels while the numbers along the left-hand side describe the MOS rating. The data or lines on the graph represent the score of each error correction scheme (repeating or muting) versus the threshold level.

From the results of Test 1 it was concluded that threshold level 2 plus or minus one threshold and the muting error correction scheme for jamming were the most optimal as they gave the highest MOS ratings from the 30 listeners involved in the testing.

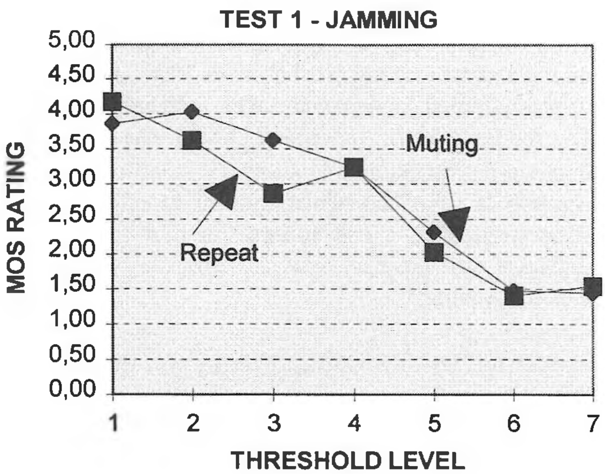

Figure 4 Test 1. Jamming from Threshold's 1 to 7.

3.2 Test 2

Test 1 determined the type of error correction scheme that would be used and the threshold of correction for jamming at 1 jam/second according to the DCR MOS test. Test 2 used this chosen threshold value and error correction scheme with the selected bit error rates. Since jamming and the BER could only be corrected with one chosen threshold, there was a need to see how the parameters chosen from Test 1 for jamming compared to the selected BER's. This compariosn was performed in Test 2.

The BER's were chosen in the follwoing manner. For a BER of 0.1%, a threshold of 1 was found to be worse than the original file, and also worse than a threshold of 3 at the same BER. The repeating method was found to be worse than the muting at a threshold of 1 and 3 at 0.1% BER. Both error correction formats were worse than the original sample. It was difficult to distinguish between the thresholds at BER of 0.1%. The BER's of 0.01% and 0.001% were almost impossible to distinguish between corrected and uncorrected samples at these thresholds. Therefore, nothing was tested below 0.01% BER since anything below 0.01% BER was acceptable. The chosen BER's were 0.5%, 0.1%, 0.05% and 0.01%.

Testing was accomplished by comparing a speech sample that was corrected at each specific to the original uncorrected speech sample. All of the samples were corrected using the muting correction method at Threshold's 1, 2 and 3 chosen from Test 1. Since the threshold DCR MOS test values were so close in Test 1, it was difficult to conclude if threshold of 2 is absolutely superior. Therefore, 3 threshold values were chosen.

For this test, a non-corrected file was compared with a corrected file according to the rating system shown in Figure 5. This is the same DCR MOS rating system used for Test 1.

The test group ended up consisting of 31 people.

Moderately poorer quality	Slightly poorer quality	Same quality	Slightly better quality	Moderately better quality
1	2	3	4	5

Figure 5 DCR MOS Test rating schedule.

The different BER's are shown along the bottom with the MOS Rating along the left-hand side. The different lines within the graph itself represent the three thresholds.

Referring to Figure 6 below, it appeared that a threshold level of 2 seemed to get the best MOS rating for a BER greater than or equal to 0.1%.

From Test 2 it was evident that the optimal BER performance was with a threshold of 2 (see Figure 6) as it scored the highest overall MOS rating vs. BER.

3.3 Test 3

Test 3 was run on a different principle than the previous two tests. There were no comparisons involved for the ratings. An absolute rating schedule, as shown below in Figure 7, was used for each trial based on a single speech sample that was heard one at a time by the listener.

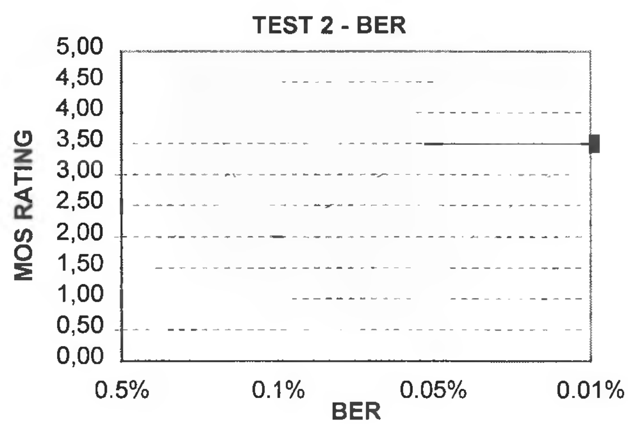

Figure 6 Test 2 for a chosen threshold of 2 with threshold's 1 and 3 using the muting correction scheme and the selected BER's.

Bad Poor Fair Good Excellent
1-----------2----------3---------4-----------5

Figure 7 ACR MOS Test rating schedule.

This test made comparisons of what it would be like to have several users jam at once or jam at a different point in time. For instance, if there were 3 jams occurring in one second (1000 ms) and they were close together, then there would be 30 ms of straight jam (since each jam equals 10 ms) with 970 ms of the regular speech sample not affected. If there were 3 jams that were far apart or dispersed, then you would hear 10 ms of jamming, then 323 ms of regular speech, then 10 ms of jamming, then 323 ms of regular speech, and finally 10 ms more of jamming followed by 323 ms of speech. The jamming was programmed into the speech samples by prewritten software. This same pattern would work for any other number of jams, except for the case of having only one jam where, of course, it cannot be dispersed. Figure 8 shows the system of jamming used for Test 3.

3 Jams/s Close Together
30 ms of Jam per 1000 ms (1 second) of speech
|XXXXXXXXX |XXXXXXXXX |
1 second 1 second

3 Jam/s Dispersed
10 ms of Jam alternating with 323 ms of speech
|XXX XXX XXX |XXX XXX XXX |
1 second 1 second

Figure 8 System 3 Jams/s close together and dispersed.

The results of Test 3 for close together jams (to simulate jamming at the same time) and jams far apart (to simulate dispersed jams) are shown in Figure 9.

From the data in Figure 9, it appears that when the jams are dispersed (highlighted as Far in the figure), most ratings are below fair (MOS < 2.5). The best scenario for dispersed jams is 4 jams since there is a drop off in quality after that. For close together jams, the ratings are fair – up to 9 jams, with an anomaly at 3 jams/s.

4. CONCLUSIONS

A 32 kbits/s ADPCM coding scheme gave the best audio quality for the lowest number of bits. For Test 1, the

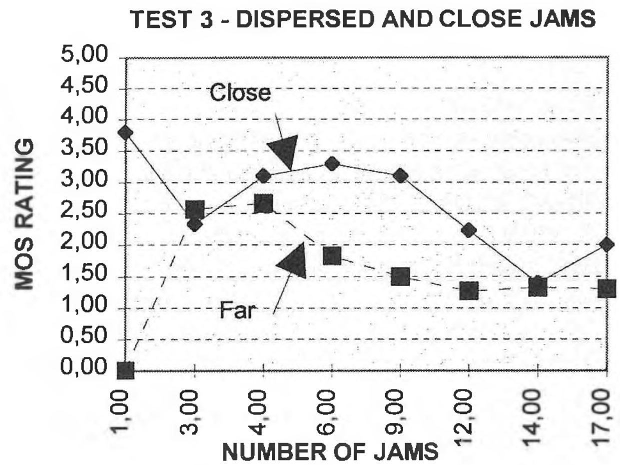

Figure 9 Test 3 for far apart and close Jams from 1 to 17 Jams with a muting error correction scheme at threshold 2.

threshold chosen was number 2. That is, the error correction scheme was invoked only after two consecutive errors were detected. These threshold levels received the highest ratings from the listeners. The correction scheme chosen was muting since the speech samples that were corrected using this scheme received higher ratings than the samples corrected with the repeating "previous block" method. The DCR MOS test rating was used since the basis of the test was to compare a reference sample to a corrected sample.

Test 2 concentrated on finding the BER with the best audio quality for the chosen threshold and error correction scheme from Test 1. To give more variety during testing, the chosen threshold from Test 1 along with the upper and lower threshold were chosen for Test 2. The level chosen from Test 2 that had the best audio quality for BER was threshold 2, which was the same as the threshold chosen for Test 1. Test 2 was conducted for a BER of 0.5%, 0.1%, 0.05% and 0.01%. Any BER meeting or exceeding 0.01% would have an acceptable level of audio quality according to our preliminary tests. Once again, the DCR MOS test for comparison ratings was used.

Test 3 incorporated parameters found from Test 1 and Test 2, which are a threshold of 2 with a muting correction scheme, to do a density evaluation for jamming. From reviewing the data, it was evident that using more than 4 dispersed jams did not have an acceptable audio quality. Between 1and occurring at the same time, or close jams, has a fair quality (except at 3 jams), but there was a drop off on either side of these values. Test 3 used the ACR MOS test method that asks for the overall opinion of each sample on its own.

Based on these findings it appears that using the muting correction method with a threshold of 2 for jamming and a

BER above 0.01% are the parameters with the best subjective audio quality for this project.

As efforts were taken to have consistency within each test it was interesting to find that a vast number of different opinions for audio quality can come from different people. It appeared that each person seemed to interpret the rating system differently. However, since the MOS system is a standardized method for performing subjective tests, it must also be standard that you can expect a certain number of people to fall within the mean and be able to expect certain deviations from the mean.

5. REFERENCES

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6. GLOSSARY

BER

Bit Error Rate. The probability (in decimals) of a bit being subject to error. There is 1 parity bit per sample and 10 parity bits per block. If the BER is 0.0001%, there is the probability of 1 in 1000 bits having an error. So in 1000 samples or in 100 blocks, there is a chance of one parity bit being wrong.

Jamming

Jamming consists of period and duration. The period is the periodic time (in milli-seconds) for jamming to be active. The duration is the time (in milli-seconds) for the length of the jamming. For example, if you want to jam a signal 3 times far apart in one second, you simply enter a 10 ms jam at intervals of 323 ms. If you want the jams to be close together, you would enter 30 ms of jamming with 970 ms remaining since 1 second consists of 1000 ms. Each jam consists of 10 ms of jamming per second. A practical example of jamming is when a frequency hopping radio hops between pseudo-random frequencies at 10 milliseconds and at a certain time interval hops to a frequency that is occupied by another signal. The radio hopper will experience 'jamming' for 10 milliseconds until it hops to a new frequency.

Threshold

The threshold levels in this paper range from 1 to 10. A block will be replaced by all 1's (muting) or the previous block if the number of bit errors in the block (40 bits with 10 parity bits) equals the error threshold. So, if there are 2 errors in a block and you are correcting these errors with a threshold of 2, then this particular block will be replaced according to the error correction scheme you have prescribed.

Block Contains 10 samples, which are 40 bits and 10 parity bits.

Sample Contains 4 bits and 1 parity bit.

Parity Bit Determines whether the sample contains an error or not. The parity bits are introduced after ADPCM encoding as a way to introduce errors into the samples. The bit itself is not corrected.

Muting An error correction scheme where an entire block is muted.

Repeating An error correction scheme where an entire block is replaced by the data from the block immediately before it.

1 block = 10 samples with 10 parity bits.
1 sample = 4 bits with 1 parity bit.
1 block = 40 bits plus 10 parity bits or 50 bits in total.
1 burst = 4 blocks = 40 samples = 200 bits.
1 hop = 2 Tx and 2 Rx bursts = 800 bits = 10 ms
1 burst = 2.5 ms
10 ms = 8 blocks of Tx, 8 blocks of Rx.

The Canadian Acoustical Association
l'Association Canadienne d'Acoustique

PRIZE ANNOUNCEMENT

A number of prizes, whose general objectives are described below, are offered by the Canadian Acoustical Association. As to the first four prizes, applicants must submit an application form and supporting documentation to the prize coordinator before the end of February of the year the award is to be made. Applications are reviewed by subcommittees named by the President and Board of Directors of the Association. Decisions are final and cannot be appealed. The Association reserves the right not to make the awards in any given year. Applicants must be members of the Canadian Acoustical Association. Preference will be given to citizens and permanent residents of Canada. Potential applicants can obtain full details, eligibility conditions and application forms from the appropriate prize coordinator.

EDGAR AND MILLICENT SHAW POSTDOCTORAL PRIZE IN ACOUSTICS

This prize is made to a highly qualified candidate holding a Ph.D. degree or the equivalent, who has completed all formal academic and research training and who wishes to acquire up to two years supervised research training in an established setting. The proposed research must be related to some area of acoustics, psychoacoustics, speech communication or noise. The research must be carried out in a setting other than the one in which the Ph.D. degree was earned. The prize is for $3000 for full-time research for twelve months, and may be renewed for a second year. Coordinator: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Past recipients are:

1990	*Li Cheng*	*Université de Sherbrooke*	*1995*	*Jing-Fang Li*	*University of British Columbia*
1993	*Roland Woodcock*	*University of British Columbia*	*1996*	*Vijay Parsa*	*University of Western Ontario*
1994	*John Osler*	*Defense Research Estab. Atlantic*			

ALEXANDER GRAHAM BELL GRADUATE STUDENT PRIZE IN SPEECH COMMUNICATION AND BEHAVIOURAL ACOUSTICS

The prize is made to a graduate student enrolled at a Canadian academic institution and conducting research in the field of speech communication or behavioural acoustics. It consists of an $800 cash prize to be awarded annually. Coordinator: Don Jamieson, Department of Communicative Disorders, University of Western Ontario, London, ON N6G 1H1. Past recipients are:

1990	*Bradley Frankland*	*Dalhousie University*	*1994*	*Michael Lantz*	*Queen's University*
1991	*Steven D. Turnbull*	*University of New Brunswick*	*1995*	*Kristina Greenwood*	*University of Western Ontario*
	Fangxin Chen	*University of Alberta*	*1996*	*Mark Pell*	*McGill University*
	Leonard E. Cornelisse	*University of Western Ontario*	*1997*	*Monica Rohlfs*	*University of Alberta*
1993	*Aloknath De*	*McGill University*	*1998*	*Marlene Bagatto*	*University of Western Ontario*

FESSENDEN STUDENT PRIZE IN UNDERWATER ACOUSTICS

The prize is made to a graduate student enrolled at a Canadian university and conducting research in underwater acoustics or in a branch of science closely connected to underwater acoustics. It consists of $500 cash prize to be awarded annually. Coordinator: David Chapman, DREA, PO Box 1012, Dartmouth, NS B2Y 3Z7.

1992	*Daniela Dilorio*	*University of Victoria*	*1994*	*Craig L. McNeil*	*University of Victoria*
1993	*Douglas J. Wilson*	*Memorial University*	*1996*	*Dean Addison*	*University of Victoria*

ECKEL STUDENT PRIZE IN NOISE CONTROL

The prize is made to a graduate student enrolled at a Canadian academic institution pursuing studies in any discipline of acoustics and conducting research related to the advancement of the practice of noise control. It consists of a $500 cash prize to be awarded annually. The prize was inaugurated in 1991. Coordinator: Murray Hodgson, Occupational Hygiene Programme, University of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3.

1994	*Todd Busch*	*University of British Columbia*	*1996*	*Nelson Heerema*	*University of British Columbia*
1995	*Raymond Panneton*	*Université de Sherbrooke*	*1997*	*Andrew Wareing*	*University of British Columbia*

DIRECTORS' AWARDS

Three awards are made annually to the authors of the best papers published in *Canadian Acoustics*. All papers reporting new results as well as review and tutorial papers are eligible; technical notes are not. The first award, for $500, is made to a graduate student author. The second and third awards, each for $250, are made to professional authors under 30 years of age and 30 years of age or older, respectively. Coordinator: Delila Giusti, Jade Acoustics, Concord, ON L4K 4H1.

STUDENT PRESENTATION AWARDS

Three awards of $500 each are made annually to the undergraduate or graduate students making the best presentations during the technical sessions of Acoustics Week in Canada. Application must be made at the time of submission of the abstract. Coordinator: Ramani Ramakrishnan, Aiolos Engineering, Toronto ON M9W 1K4, Tel: (416) 674-3017.

The Canadian Acoustical Association
l'Association Canadienne d'Acoustique

ANNONCE DE PRIX

Plusieurs prix, dont les objectifs généraux sont décrits ci-dessous, sont décernés par l'Association Canadienne d'Acoustique. Pour les quatre premiers prix, les candidats doivent soumettre un formulaire de demande ainsi que la documentation associée au coordonnateur de prix avant le dernier jour de février de l'année durant laquelle le prix sera décerné. Toutes les demandes seront analysées par des sous-comités nommés par le président et la chambre des directeurs de l'Association. Les décisions seront finales et sans appel. L'Association se réserve le droit de ne pas décerner les prix une année donnée. Les candidats doivent être membres de l'Association. La préférence sera donnée aux citoyens et aux résidents permanents du Canada. Les candidats potentiels peuvent se procurer de plus amples détails sur les prix, leurs conditions d'éligibilité, ainsi que des formulaires de demande auprès du coordonnateur de prix.

PRIX POST-DOCTORAL EDGAR ET MILLICENT SHAW EN ACOUSTIQUE

Ce prix est attribué à un(e) candidat(e) hautement qualifié(e) et détenteur(rice) d'un doctorat ou l'équivalent, qui a complété(e) ses études et sa formation de chercheur, et qui désire acquérir jusqu'à deux années de formation supervisée de recherche dans un établissement reconnu. Le thème de recherche proposée doit être relié à un domaine de l'acoustique, de la psycho-acoustique, de la communication verbale ou du bruit. La recherche doit être menée dans un autre milieu que celui où le candidat a obtenu son doctorat. Le prix est de $3000 pour une recherche plein temps de 12 mois avec possibilité de renouvellement pour une deuxième année. Coordonnatrice: Sharon Abel, Mount Sinai Hospital, 600 University Avenue, Toronto, ON M5G 1X6. Les récipiendaires antérieur(e)s sont:

1990	*Li Cheng*	*Université de Sherbrooke*	*1995*	*Jing-Fang Li*	*University of British Columbia*
1993	*Roland Woodcock*	*University of British Columbia*	*1996*	*Vijay Parsa*	*University of Western Ontario*
1994	*John Osler*	*Defense Research Estab. Atlantic*			

PRIX ÉTUDIANT ALEXANDER GRAHAM BELL EN COMMUNICATION VERBALE ET ACOUSTIQUE COMPORTEMENTALE

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne et menant un projet de recherche en communication verbale ou acoustique comportementale. Il consiste en un montant en argent de $800 qui sera décerné annuellement. Coordonnateur: Don Jamieson, Department of Communicative Disorders, University of Western Ontario, London, ON N6G 1H1. Les récipiendaires antérieur(e)s sont:

1990	*Bradley Frankland*	*Dalhousie University*	*1994*	*Michael Lantz*	*Queen's University*
1991	*Steven D. Turnbull*	*University of New Brunswick*	*1995*	*Kristina Greenwood*	*University of Western Ontario*
	Fangxin Chen	*University of Alberta*	*1996*	*Mark Pell*	*McGill University*
	Leonard E. Cornelisse	*University of Western Ontario*	*1997*	*Monica Rohlfs*	*University of Alberta*
1993	*Aloknath De*	*McGill University*	*1998*	*Marlene Bagatto*	*University of Western Ontario*

PRIX ÉTUDIANT FESSENDEN EN ACOUSTIQUE SOUS-MARINE

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne et menant un projet de recherche en acoustique sous-marine ou dans une discipline scientifique reliée à l'acoustique sous-marine. Il consiste en un montant en argent de $500 qui sera décerné annuellement. Coordonnateur: David Chapman, DREA, PO Box 1012, Dartmouth, NS B2Y 3Z7.

1992	*Daniela Dilorio*	*University of Victoria*	*1994*	*Craig L. McNeil*	*University of Victoria*
1993	*Douglas J. Wilson*	*Memorial University*	*1996*	*Dean Addison*	*University of Victoria*

PRIX ÉTUDIANT ECKEL EN CONTRÔLE DU BRUIT

Ce prix sera décerné à un(e) étudiant(e) inscrit(e) dans une institution académique canadienne dans n'importe quelle discipline de l'acoustique et menant un projet de recherche relié à l'avancement de la pratique en contrôle du bruit. Il consiste en un montant en argent de $500 qui sera décerné annuellement. Ce prix a été inauguré en 1991. Coordonnateur: Murray Hodgson, Occupational Hygiene Programme, University of British Columbia, 2206 East Mall, Vancouver, BC V6T 1Z3.

1994	*Todd Busch*	*University of British Columbia*	*1996*	*Nelson Heerema*	*University of British Columbia*
1995	*Raymond Panneton*	*Université de Sherbrooke*	*1997*	*Andrew Wareing*	*University of British Columbia*

PRIX DES DIRECTEURS

Trois prix sont décernés, à tous les ans, aux auteurs des trois meilleurs articles publiés dans l'*Acoustique Canadienne*. Tout manuscrit rapportant des résultats originaux ou faisant le point sur l'état des connaissances dans un domaine particulier sont éligibles; les notes techniques ne le sont pas. Le premier prix, de $500, est décerné à un(e) étudiant(e) gradué(e). Le deuxième et le troisième prix, de $250 chacun, sont décernés à des auteurs professionnels âgés de moins de 30 ans et de 30 ans et plus, respectivement. Coordonnateur: Delila Giusti, Jade Acoustics, Concord, ON L4K 4H1.

PRIX DE PRÉSENTATION ÉTUDIANT

Trois prix, de $500 chacun, sont décernés annuellement aux étudiant(e)s sous-gradué(e)s ou gradué(e)s présentant les meilleures communications lors de la Semaine de l'Acoustique Canadienne. La demande doit se faire lors de la soumission du résumé. Coordonnateur: Ramani Ramakrishnan, Aiolos Engineering, Toronto ON M9W 1K4, Tel: (416) 674-3017.