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A VOICE CODING TECHNIQUE FOR IMPROVED HANE SURVIVABILITY

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Technical Report

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A voice coding technique based on the FFT is described. Word intelligibility tests show that the technique permits understandable speech to be reconstructed at 2400 bps, and that high intelligibility is maintained despite transmission bit errors of up to 10%.

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CONVERSION TABLE

Conversion factors for U.S. Customary to metric (SI) units of measurement.

To Convert From	To	Multiply By
angstrom	meters (m)	1.000 000 X E -10
atmosphere (normal)	kilo pascal (kPa)	1.013 25 X E +2
bar	kilo pascal (kPa)	1.000 000 X E +2
barn	meter^2 (m^2)	1.000 000 X E -28
British thermal unit (thermochemical)	joule (J)	1.054 350 X E +3
calorie (thermochemical)	joule (J)	4.184 000
cal (thermochemical)/cm^2	mega joule/m^2 (MJ/m^2)	4.184 000 X E -2
curie	*giga becquerel (GBq)	3.700 000 X E +1
degree (angle)	radian (rad)	1.745 329 X E -2
degree Fahrenheit	degree kelvin (K)	$t_K = (t^o f + 459.67)/1.8$
electron volt	joule (J)	1.602 19 X E -19
erg	joule (J)	1.000 000 X E -7
erg/second	watt (W)	1.000 000 X E -7
foot	meter (m)	3.048 000 X E -1
foot-pound-force	joule (J)	1.355 818
gallon (U.S. liquid)	meter^3 (m^3)	3.785 412 X E -3
inch	meter (m)	2.540 000 X E -2
jerk	joule (J)	1.000 000 X E +9
joule/kilogram (J/kg) (radiation dose absorbed)	Gray (Gy)	1.000 000
kilotons	terajoules	4.183
kip (1000 lbf)	newton (N)	4.448 222 X E +3
kip/inch^2 (ksi)	kilo pascal (KPa)	6.894 757 X E +3
ktap	newton-second/m^2 (N-s/m^2)	1.000 000 X E +2
micron	meter (m)	1.000 000 X E -6
mil	meter (m)	2.540 000 X E -5
mile (international)	meter (m)	1.609 344 X E +3
ounce	kilogram (kg)	2.834 952 X E -2
pound-force (lbs avoirdupois)	newton (N)	4.448 222
pound-force inch	newton-meter (N.m)	1.129 848 X E -1
pound-force/inch	newton/meter (N/m)	1.751 268 X E +2
pound-force/foot^2	kilo pascal (kPa)	4.788 026 X E -2
pound-force/inch^2 (psi)	Kilo pascal (kPa)	6.894 757
pound-mass (lbm avoirdupois)	kilogram (kg)	4.535 924 X E -1
pound-mass-foot^2 (moment of inertia)	kilogram-meter^2 (kg.m^2)	4.214 011 X E -2
pound-mass/foot^3	kilogram/meter^3 (kg/m^3)	1.601 846 X E +1
rad (radiation dose absorbed)	**Gray (Gy)	1.000 000 X E -2
roentgen	coulomb/kilogram (C/kg)	2.579 760 X E -4
shake	second (s)	1.000 000 X E -8
slug	kilogram (kg)	1.459 390 X E +1
torr (mm Hg, 0^o C)	kilo pascal (kPa)	1.333 22 X E -1

*The becquerel (Bq) is the SI unit of radioactivity; 1 Bq = 1 event/s.

**The Gray (GY) is the SI unit of absorbed radiation.

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SECTION 1
INTRODUCTION

1-1 HIGHLIGHTS OF PRIOR EFFORTS.

This report summarizes the continuation of previous studies of digital voice radio communications in the presence of fading due to high altitude nuclear device detonation [1]. Most of the effort in this study was invested in developing and testing a Fourier transform coding technique designated "Selective Frequency Coding" (SFC).

Previous efforts demonstrated that the standard 2400 baud LPC-10 algorithm degrades word intelligibility rapidly when bit error rates exceed 1-2%, and that word intelligibility suffers more in a fading environment than in a random noise environment. Furthermore, these previous efforts indicate that minor adjustments to the LPC-10 algorithm will not significantly mitigate this loss of intelligibility in a fading environment.

The current work continues the search for a voice communications system which is robust to bit errors in fading. Since the transmission channel and its protocol are fixed, modifying the channel by adding time, frequency or spatial diversity was deemed beyond the scope of this study. Instead, extensions to the previous work focused on examining alternatives to the LPC-10 algorithm which might prove more robust against typical errors in fading environments.

1-2 BASIS FOR THE FFT VOCODER.

Transform domain coders have been extensively reported in the literature as methods of achieving toll quality voice compression at medium bit rates (8 kbps - 16 kbps). These coders fall into two broad categories: those which use the slowly changing envelope of the voice spectrum to efficiently allocate bits to various spectral regions [2], and those which explicitly model voice as a sum of narrowband sinusoids [3].

MAXIM has chosen to investigate the latter sum-of-sinusoids model as a potential approach for achieving highly intelligible synthetic quality voice in a 2400 bps scheme which is more intelligible than LPC in fading environments. This approach is intuitively appealing, since the various sinusoid components when coded independently should carry comparable amounts of information. Consequently, an error in the bits representing one of these components would not completely alter the character of the voice sound as in LPC, but would instead result in some sort of background noise. One might expect that the human ear would be able to sort through the background noise, maintaining word intelligibility despite the bit errors. Experiments have verified that this conjecture holds.

The technique is conceptually simple. A 256 point FFT is performed on the input voice, and the 6 largest peaks in the spectral amplitude are found. The locations, amplitude and phase of each of these peaks are transmitted to the receiver, where a reconstruction is performed in a way that ensures waveform continuity across frame boundaries. The result is speech that sounds somewhat reverberant, but is highly intelligible, retains a high degree of talker recognizability, requires no error-prone pitch estimation, and can be implemented with low cost, current generation digital signal processing devices.

1-3 REPORT OVERVIEW.

The body of this report will present a brief summary of the characteristics of voice signals in general to provide a foundation for describing the technique, then discuss the design trade-offs and present the intelligibility test results. We will conclude with some recommendations for future study in the area of achieving survivable low bit-rate voice communications systems.

1-4 SUMMARY OF RESULTS.

The work performed under this contract has resulted in a 2400 baud voice coding technique that permits intelligible reception in the presence of 5-10% bit error rates. LPC-10 typically becomes very difficult to understand at 5-6% error rate in a comparable environment.

SECTION 2

SPECTRAL CHARACTERISTICS OF SPEECH

2-1 MODEL OF SPEECH GENERATION.

Speech sounds can be classified into two broad groups: voiced sounds such as vowels (a,e,i,o,u,y) and sibilant consonants (n,l); and unvoiced sounds such as t, ch, and p. The voiced sounds are created by periodic pops of air released by the glottis in the throat. These impulses of air are filtered by the resonances of the throat and mouth to form a ringing waveform that repeats with a basic pitch period as illustrated in Figure 2-1. Unvoiced sounds are created by suddenly releasing pressure developed behind the lips or tongue as in "p", "k", or "t", or by forcing air through a constriction as in the sounds "ch" and "s". Unvoiced sounds are not periodic, but instead resemble random noise.

Both voiced and unvoiced sounds can be modeled by factoring the sound into 2 components: an excitation function and a filter which modifies the excitation function. For voiced sounds, the excitation function is a periodic pulse train representing the glottal pulses, and the filter represents the spectral shaping imposed by the resonances of the throat and mouth. For unvoiced sounds, the excitation function is a white noise source and the filter shapes the noise spectrum appropriately for the type of noise sound.

2-2 VOCODER CONCEPT.

Most low bit rate vocoders achieve their savings in bit rate from the very simple models of the excitation functions described above: for voiced sounds, only a pitch period and amplitude need to be transmitted, while unvoiced sounds only require an excitation amplitude since the excitation function is white noise.

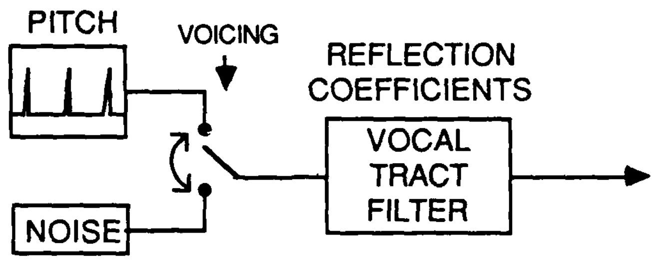

Figure 2-1. LPC vocoder structure.

This factoring of voiced sounds into two components can also be interpreted in the spectral domain. The vocal tract filter imposes a smooth overall envelope to the spectrum, while the periodicity of the waveform causes the fine structure within this envelope consisting of a series of spectral lines replicated at harmonics of the pitch period.

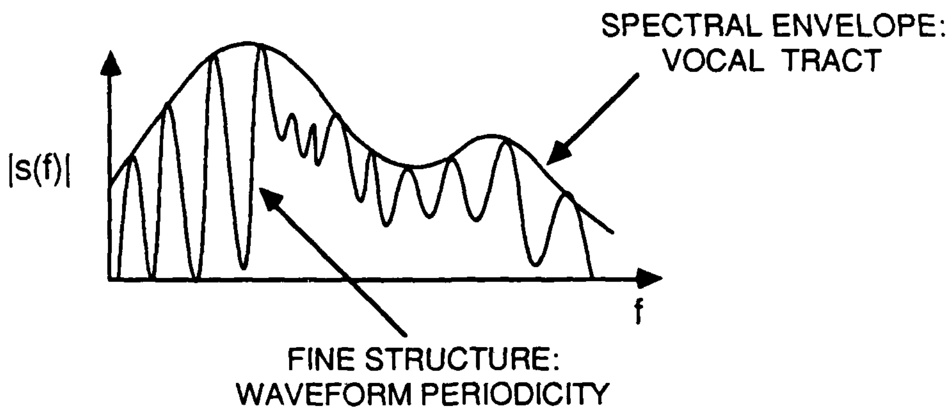

Figure 2-2. Short term voice spectrum.

2-3 SFC SPEECH MODEL.

In contrast to the pitch-excited vocoder voice model, the selective frequency coding technique does not factor voice into an excitation function followed by a slowly changing filter. Furthermore, SFC does not require an explicit pitch estimate to account for the periodicity of voiced sounds.

Instead, SFC relies on the fact that periodic signals give rise to a Fourier spectrum containing distinct spectral lines. Rather than forcing these spectral lines to be exact harmonic multiples with the specific phase relationship corresponding to a periodic waveform, SFC simply chooses the largest peaks and transmits the locations, amplitudes, and phases of these peaks.

SECTION 3
DEVELOPMENT BACKGROUND

This section will begin with a description of the hardware used to digitize and reconstruct the voice. This will be followed by brief descriptions of the various algorithms tested in the course of developing the final SFC algorithm.

3-1 DIGITIZING EQUIPMENT.

The equipment used in this study to digitize and reconstruct voice consisted of a tape deck, a stereo equalizer, a simple audio filter, a DEC LPA-11A A/D-D/A system, and a VAX 11/750 computer.

Initially, digitizing was performed at 20 kHz, permitting a 10 kHz signal bandwidth. This sample rate was later reduced to the telephone industry standard 8 kHz since peaks are rarely chosen in the 4-10 kHz region, and the 8 kHz rate yields an appropriate time span when a 256 sample frame size is used.

A National Semiconductor AF-134 filter device was used in both digitizing and reconstruction operations. The equalizer was used to provide slight pre-emphasis which was found to enhance the intelligibility of some male talkers significantly. The equalizer was also used to provide appropriate gain. Figure 3-2 shows the frequency response of the filter/equalizer combination when the equalizer is set for flat response.

The LPA-11A digitizer has an input range of +/- 5V which converts to a range of 0-4095. The output samples are offset binary with 2048 representing 0 volts. After characterizing the harmonic distortion of the digitizer as a function of input amplitude, it was determined that the converter should be driven at roughly 70% of its maximum dynamic range to minimize intermodulation distortion.

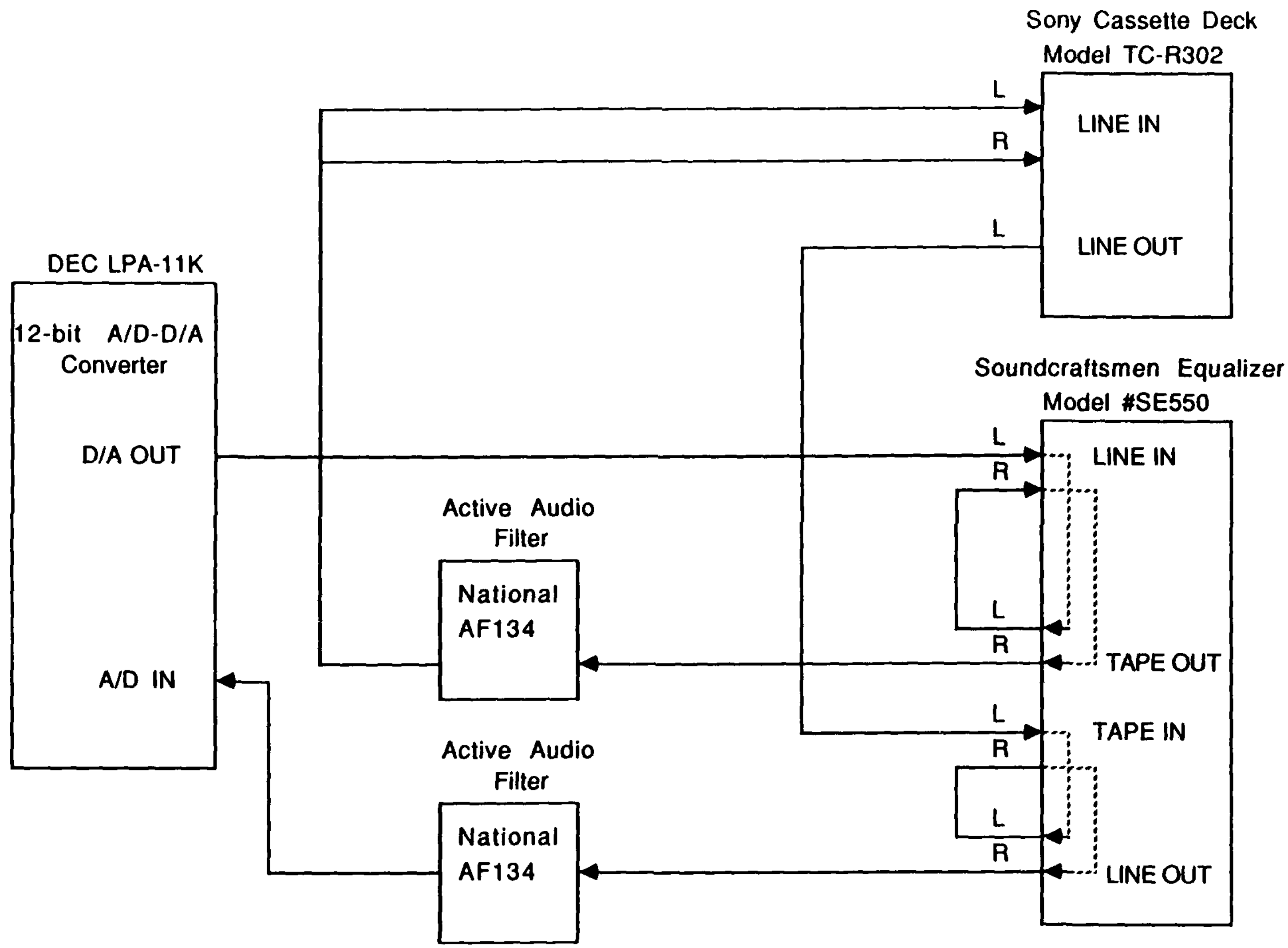

Figure 3-1. Digitizing equipment.

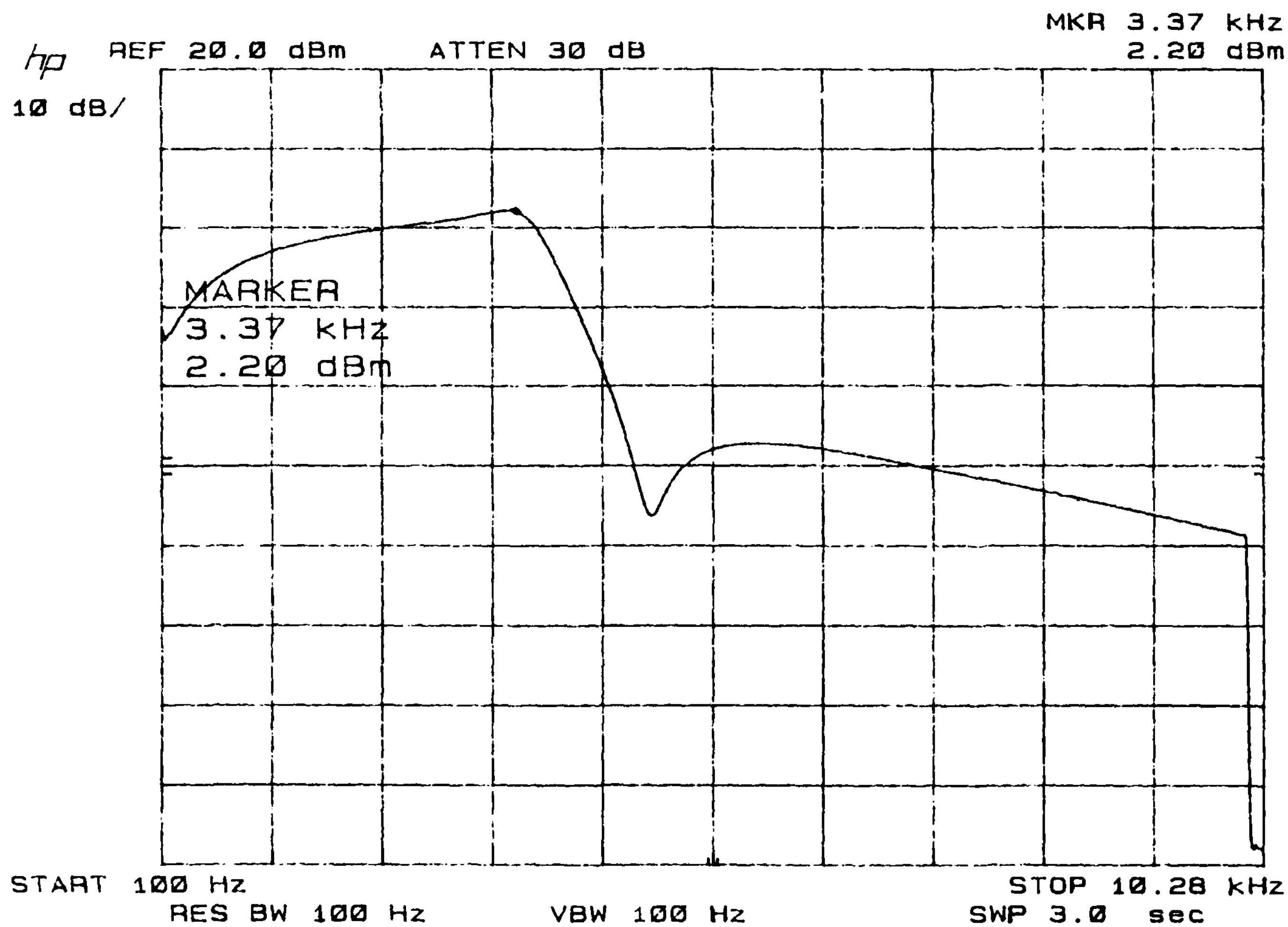

Figure 3-2. Data acquisition spectral response.

3-2 INITIAL EXPERIMENTS.

3-2.1 Algorithm Using Largest Spectral Bins.

Initial efforts for the FFT vocoder attempted to retain the most spectral power by coding the largest spectral bins of the FFT. Since the chosen frequency elements had the largest magnitudes (see Figure 3-3), they might be expected to retain most of the waveform characteristics.

Figure 3-3. Largest spectral bin analysis algorithm.

The sampled input speech was analyzed through a fast Fourier transform (FFT), and the magnitude and phase of the bins containing the largest magnitudes were transmitted. The receiver then reconstructed the waveform using an inverse FFT.

The speech quality using this technique was poor at low data rates. The frequency bin selection was concentrated largely in the low frequencies below 1 kHz, producing speech which sounded muffled due to the lack of high frequency content. At data rates above 8 kbps, very natural, highly intelligible speech was produced due to the retention of more frequency components including those in the upper passband.

In addition to the lack of high frequency information at low data rates, the reconstructions suffered from discontinuities at the frame boundaries. These discontinuities were perceived as clicks occurring at about 30 Hz.

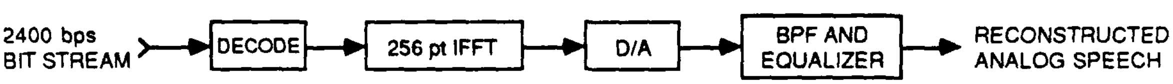

Figure 3-4. Largest spectral bin reconstruction.

3-2.2 Algorithm Using Intraframe Spectral Differences.

The previous results confirmed the intuitive notion that preserving more frequency bins improves the reconstructed speech quality, so some effort was aimed at retaining more information by relying on the fact that the spectrum often changes slowly from frame to frame. By retaining a spectrum that models the past speech frames and transmitting the bins of the current frame that create the largest differential we can preserve more information, transmit less redundant information, and produce higher quality reconstructions. This approach is depicted in Figure 3-5.

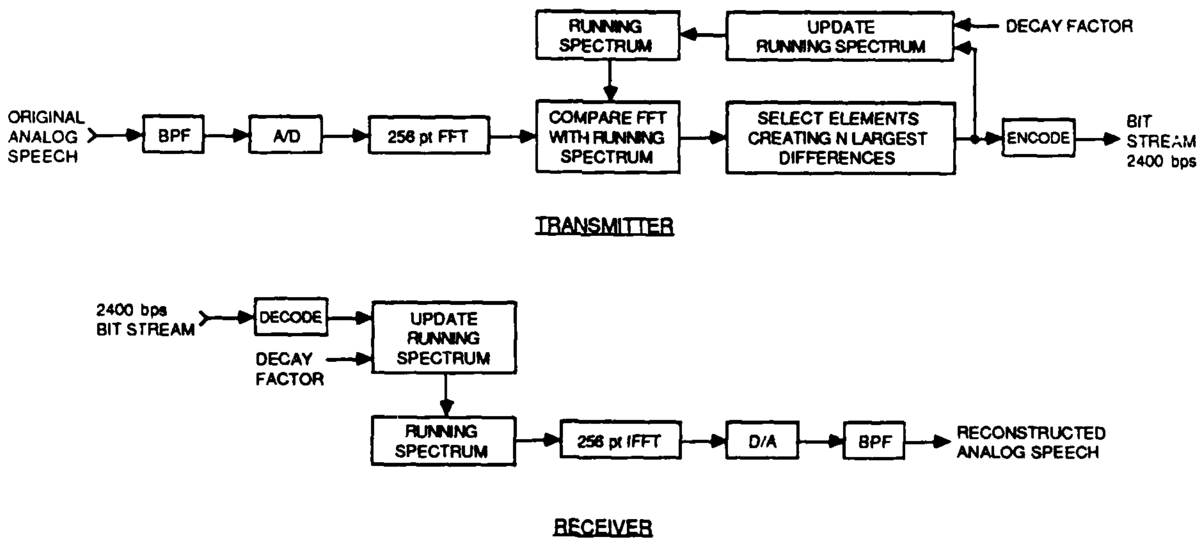

Figure 3-5. Intraframe spectral difference algorithm.

The speech is processed as before through an FFT. The resultant spectrum is compared to a running spectrum, and the magnitudes and phases of the bins creating the largest difference are coded and transmitted. The running spectrum is calculated from frequency information of previous frames, and is decayed with time.

The speech processed in this manner suffered from an unacceptable reverberance. Several attempts that were made to reduce this reverberance met with some success; however the resulting quality was ultimately not acceptable at rates below 8 kbps.

3-3 ALGORITHM USING MAXIMUM SPECTRAL PEAKS.

3-3.1 Algorithm Concept.

One of the problems with the speech quality from the previous two attempts was that the spectrum was mainly being selected in the lower end of the speech band, and the reconstructed speech consequently sounded muffled. An alternate method was attempted that considered only those FFT bins that corresponded to local maxima in the spectrum as shown in Figure 3-6. To the extent that the short term spectrum is represented by a small number of sine waves, choosing the peak spectral bins of an FFT corresponds to slightly shifting the frequencies of these sines since they would not generally fall directly on an FFT bin frequency, but would nevertheless be represented as sines at exact bin frequencies. Since the periodic nature of the voice gives rise to a line spectrum, we might expect this model to work adequately for voiced frames. With unvoiced frames, the hope was that the non-harmonic relationship of the bins chosen, together with the unstructured phases of these bins would adequately represent noise-like sounds.

In summary, the approach was to transform blocks of the input data via an FFT and transmit the amplitudes and phases of the bins with the largest local maxima in the spectrum. This frequency selection involved more of the speech band which resulted in higher quality speech than the previous two approaches: the voice was more pleasant with the muffled quality largely eliminated. While the speech sounded somewhat synthetic, high intelligibility and some degree of talker recognition was maintained.

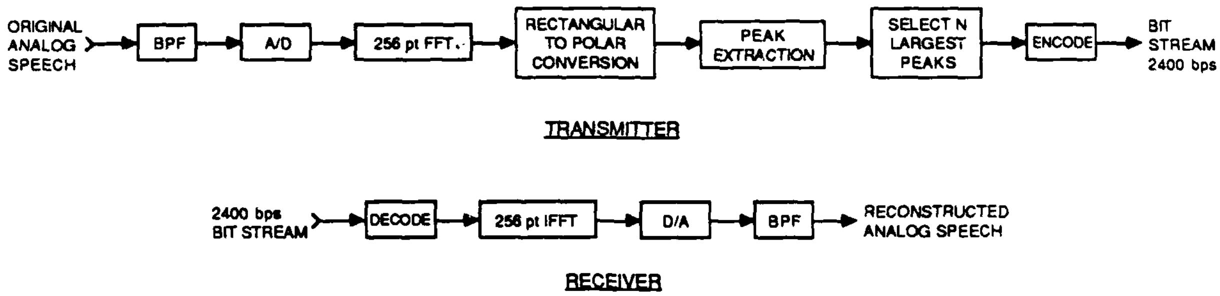

Figure 3-6. Maximum spectral peak algorithm.

3-3.2 Quantization Scheme.

At this point more attention was turned to the actual bit coding which involved choosing an ultimate compression scheme that would permit coding the maximum number of peaks, and finalize an ultimate frame size.

A 32 ms frame size was found to be best for an 8 kHz sampling rate. If the frame size was less, then fewer peaks must be used to satisfy the data rate limitation. Longer frames would smear rapid changes that speech could have. The 32 ms window also gates out 256 point samples which is a convenient size appropriate for FFT processing.

With the 32 ms window chosen, 7 bits are needed for the frequency bin coding, leaving 3 bits for the magnitude, and 3 bits for the phase. The resultant array from the 256 point FFT will contain real and imaginary spectrum that are mirror images of each other. Thus only 128 frequency bins need to be searched and coded, so 7 bits suffice to uniquely specify the frequency bin number. A logarithmic ulaw compression scheme was chosen to code the magnitude. The magnitude was compressed by:

$$M = \frac{(\log(1 + u \cdot m)}{(\log(1 + u))}$$

M = compressed magnitude
m = magnitude
u = ulaw factor

The u factor was chosen experimentally and the value of 100 appeared to work best. Three bits were sufficient and resulted only in a slight amount of distortion due to the quantization.

For coding the phase a 3 bit linear compression was used, making the resolution pi/4 radians with a maximum error pi/8. With these bit allocations we could choose 6 frequency peaks if we coded the five largest elements as described, and the sixth largest with only 2 bits for magnitude, and 2 bits for phase. This was possible since the sixth element is normally lower in energy and does not require the resolution of the previous five elements. This leaves the data rate at 2375 bps with 25 bps free for synchronization.

3-4 INITIAL INTELLIGIBILITY TESTS.

In the development of a new voice coding scheme the quality must be measured to evaluate its performance. In a benign case there will be no errors and the speech will be at its best performance. To measure this performance the Diagnostic Rhyme Test (DRT) is used. This test measures the listener's ability to distinguish between two similar sounding words. In an errored case, particularly in the fading environment, the errors corrupt large portions of entire words. Since DRT scores insufficiently measure intelligibility in these cases, the Phonetic Alphabet Comprehension Test (PACT) was developed [1].

The PACT was able to measure the listeners ability to comprehend a fixed set of words in an errored environment. Consequently, PACT scores along with the DRT scores will give a good indication of the overall performance of the reproduced voice quality.

3-4.1 Diagnostic Rhyme Test (DRT).

The DRT consists of a set of words that demonstrate the phonemic features of speech (see Table 3-1): voicing, nasality, sustention, sibilation, graveness, and compactness. Pairs of words are used with one stressing the phonemic feature, and the other not. Of these two, one is chosen and processed. If the difference can be depicted, by choosing the correct word during evaluation, then the algorithm succeeds in retaining this particular phonemic feature. If the algorithm does well in all areas, then it can be determined as being completely intelligible. If not, it must be determined if it is acceptable for the specific application. We chose six speakers, three male and three female, of varying backgrounds, to randomly choose and read, from the pairs of words of each group, indicating their choices and the order they read them. The speech samples were then applied to the algorithms under test at varying data rates. For each different data rate the resultant reconstructed speech was evaluated by 6 different listeners, who were also of varying backgrounds. They would indicate their choice of the word they thought was spoken of the two choices in each word pair. This would then be compared to the actual words spoken to determine if the choice was correct and with what types of phonemic sounds the algorithm had difficulty with. The comprehension results were then compiled, corrected for guessing, and conclusions derived.

3-4.2 Phonetic Alphabet Comprehension Test (PACT).

For the nuclear stressed environment the Phonetic Alphabet Comprehension Test (PACT) was used. This test evaluates how the performance measured by the DRT is affected by an environment such as a fading nuclear environment. The PACT concentrates on the overall comprehension of a message rather than the individual sounds as with the DRT. This is applicable in the fading nuclear environment since the errors tend to be clustered. The result is the loss of complete words or large

Table 3-1. DRT word pairs.

Stimulus Words used in the DRT

VOICING	NASALITY	SUSTENTION
Voiced—Unvoiced	*Nasal—Oral*	*Sustained—Interrupted*
veal—feel	meat—beat	vee—bee
bean—peen	need—deed	sheet—cheat
gin—chin	mitt—bit	vill—bill
dint—tint	nip—dip	thick—tick
zoo—Sue	moot—boot	foo—pooh
dune—tune	news—dues	shoes—choose
voal—foal	moan—bone	those—doze
goat—coat	note—dote	though—dough
zed—said	mend—bend	then—den
dense—tense	neck—deck	fence—pence
vast—fast	mad—bad	than—Dan
gaff—calf	nab—dab	shad—chad
vault—fault	moss—boss	thong—tong
daunt—taunt	gnaw—daw	shaw—chaw
jock—chock	mom—bomb	von—bon
bond—pond	knock—dock	vox—box

SIBILATION	GRAVENESS	COMPACTNESS
Sibilated—Unsibilated	*Grave—Acute*	*Compact—Diffuse*
zee—thee	weed—reed	yield—wield
cheep—keep	peak—teak	key—tea
jilt—gilt	bid—did	hit—fit
sing—thing	fin—thin	gill—dill
juice—goose	moon—noon	coop—poop
chew—coo	pool—tool	you—rue
Joe—go	bowl—dole	ghost—boast
sole—thole	fore—thor	show—so
jest—guest	met—net	keg—peg
chair—care	pent—tent	yen—wren
jab—dab	bank—dank	gat—bat
sank—thank	fad—thad	shag—sag
jaws—gauze	fought—thought	yawl—wall
saw—thaw	bond—dong	caught—taught
jot—got	wad—rod	hop—fop
chop—cop	pot—tot	got—dot

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parts of them, so a test to measure the comprehension of a sequence of words is needed. The actual PACT considers a sequence of military phonetic words (alpha, bravo, charlie, ...yankee, zulu) (Table 3-2) that have been passed through a fading communications channel. The actual sequence is chosen randomly from the 26 phonetic words. The performance is measured by the listener's ability to detect and correctly comprehend each word in the order they were presented. The results of the PACT are used in combination with the DRT scores. Any PACT score less than 100 indicates a loss in intelligibility from the benign case measured by the DRT. A PACT score of 100 indicates a performance that is comparable to the DRT score, therefore, a 100 PACT score indicates intelligibility of the speech in a benign case.

Table 3-2. Military phonetic alphabet.

PHONETIC ALPHABET LIST

A	-	Alpha	N	-	November
B	-	Bravo	O	-	Oscar
C	-	Charlie	P	-	Papa
D	-	Delta	Q	-	Quebec
E	-	Echo	R	-	Romeo
F	-	Foxtrot	S	-	Sierra
G	-	Golf	T	-	Tango
H	-	Hotel	U	-	Uniform
I	-	India	V	-	Victor
J	-	Juliet	W	-	Whiskey
K	-	Kilo	X	-	X-ray
L	-	Lima	Y	-	Yankee
M	-	Mike	Z	-	Zulu

3-4.3 Bit Error Application.

Evaluating errors due to a noisy or a fading environment requires modeling the satellite link and the propagation channel. The link or system model includes the hardware used to protect against errors by coding, modulating, and interleaving the data. The channel model used in our studies simulates the fading due to multipath and phase distortion of high altitude nuclear effects (HANE). Coded speech data is processed through these models to measure the effects of the bit errors.

3-4.3.1 Channel Model.

The channel model simulates the effects of the environment on the data transmission, including Gaussian background noise, signal path losses, and fading due to multipath. Two groups of channel model parameters were employed in this study. In one set, a severe multipath model was applied, and in the other set, only white Gaussian noise was applied. The parameters were adjusted in the two sets to provide two cases of aggregate bit error rates: 1% and 10%. In a multipath environment, occasional deep fades result in a very high percentage of bit errors during the fade, so errors will occur in bursts. This fading case is contrasted to the case of a model with no multipath, in which the errors are distributed randomly.

The Rayleigh fading model employed allows the time scale of fading to be modified by setting the fade decorrelation time. While fade depth, duration, and separation are randomly generated by this model, a short decorrelation time will typically result in fades of short duration separated by short time periods. Conversely, a long decorrelation time will cause the multipath channel model to evolve more slowly, providing longer fade durations with longer periods between the fades on the average.

Since average fade duration is proportional to decorrelation time, and since the bit error rate during a fade is extremely high, using a long decorrelation time (say, 1 sec) will result in the loss of entire syllables or words, seriously impacting word intelligibility of the voice. With short fade durations, some frames of most words will be corrupted with errors, but enough of the frames are properly received to allow reasonable intelligibility.

3-4.3.2 System Model.

To evaluate the performance of a voice coding scheme in a satellite system, a specific transmitter and receiver must be simulated. Consequently, we must define coding schemes, modulation schemes, and other processes that may be used in a system for error correction. We used a model of an existing satellite system which is known to perform well in a white Gaussian noise environment. This link model includes convolutional coding, soft Viterbi decoding, differential phase shift keying (DPSK), and convolutional interleaving. The encode/decode process used is a rate 1/2 (R1/2) convolutional code along with a soft decision Viterbi decoder. The encoder transforms the incoming data bit with 2 symbol bits, so a 2400 bps data stream will be encoded into a 4800 bps symbol bit stream [6]. In the receiver the encoded symbols are decoded through a soft decision Viterbi decoder. The soft decision retains the quantized values of the received bit stream that indicate how the bit information varies after going through the error channel. The path metric is determined from these values and the symbols are decoded using Viterbi's maximum likelihood algorithm. The modulation/ demodulation scheme is differential phase shift keying (DPSK). Input to the modulator is the interleaved coded symbol bit stream, and the output is a waveform that shifts 180 degrees in phase whenever an input bit of logic 1 is applied. The demodulator will detect this phase shift in the waveform, and determine a quantized value in accordance with this detected information. This quantized value is a level which ultimately demonstrates the confidence in which the symbol decision can be made by the Viterbi decoder. For the fading environment an interleaver will have a large contribution in the performance of the system model. The coding described above will perform very well in a noise environment where the errors are randomly distributed. In fading, however, the errors are grouped in bursts and create bit error sequences that the decoder cannot correct. An interleaver will take its input bits and create a new bit sequence through a convolution, so that no contiguous sequence of n2 bits, of the new sequence, will contain any pair of symbols that are originally separated by n1 bits [7]. This new sequence is then modulated and demodulated along with the channel effects. The demodulated sequence will exhibit the characteristic burst of errors associated with the fading channel. But after deinterleaving, the original sequence of bits is restored and the bursty bit errors will be spread out over the whole interleaver span and appear more randomly distributed for better decoder performance. For the purpose of a demonstration of the fading environment performance a convolutional interleaver of size 24 x 384, interleaver time span (Ts) of 1.92 seconds, was used.

3-4.4 Test Results.

3-4.4.1 DRT Results.

The Diagnostic Rhyme Test (DRT) was used to measure the intelligibility of the algorithm using the spectral peak method described in Section 3.3. The talkers for the test consisted of three male talkers and three female talkers. They varied in age, nationality, and first languages. The listeners for this test also varied in age, sex, nationality, and first languages. After each test was taken, the listeners were allowed to express experiences and observations about the test. These initial tests were used on the algorithm operating at 96 kbps, 6.5 kbps, 4.6 kbps, and 2.4 kbps. The 96 kbps corresponds to the uncompressed case and will be used for a baseline. The effect of data rate on the intelligibility can be seen in Table 3-3, where the intelligibility falls steadily with decreasing data rate. The average adjusted (corrected for guessing) and unadjusted scores were as follows:

Table 3-3. Comprehensive DRT scores.

Date Rate (kbps)	Unadjusted (%)	Adjusted (%)
96	94	89
6.5	84	68
4.6	82	64
2.4	77	54

The scores show a dramatic difference between adjusted and unadjusted since there were only two choices for each word and there was a good chance of guessing the correct answer.

An interesting note is that the uncompressed speech, at 96 kbps, did not produce a 100% DRT score. This is indicative of lapses in listeners' attention, occasionally indistinct pronunciation, and the effect of the bandlimiting anti-aliasing filter. In addition, it was later discovered that the digital voice did not cover the entire dynamic range of the digitizer. This should be noted when analyzing these DRT results.

The female talkers appeared to provide the most intelligible speech, as seen in Figure 3-7a, at data rates of 2.4 kbps, and 6.5 kbps. At 4.5 kbps, and 96 kbps, the difference between male and female talkers is insignificant.

The data rate reduction also took its toll on the intelligibility as almost all the phonemic features decreased with decreasing data rate with nasality surviving the best, and graveness suffering the most (see Figure 3-7b and c).

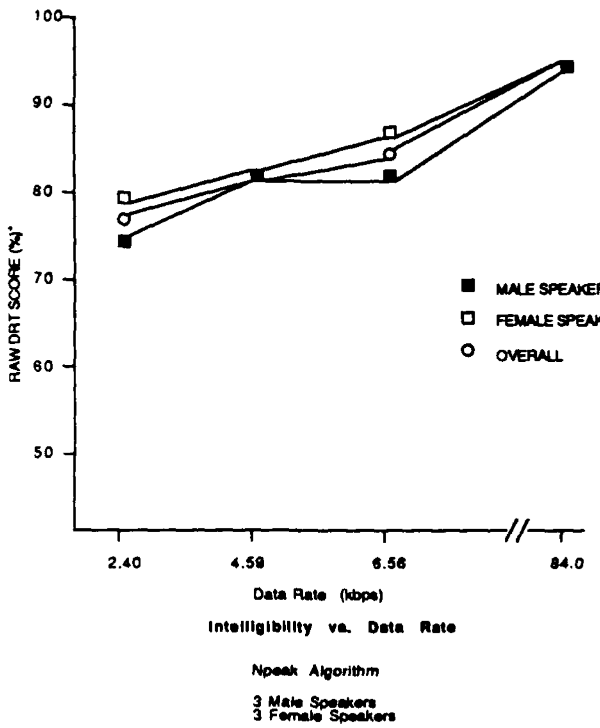

Figure 3-7a. Unadjusted DRT scores for npeak algorithm.

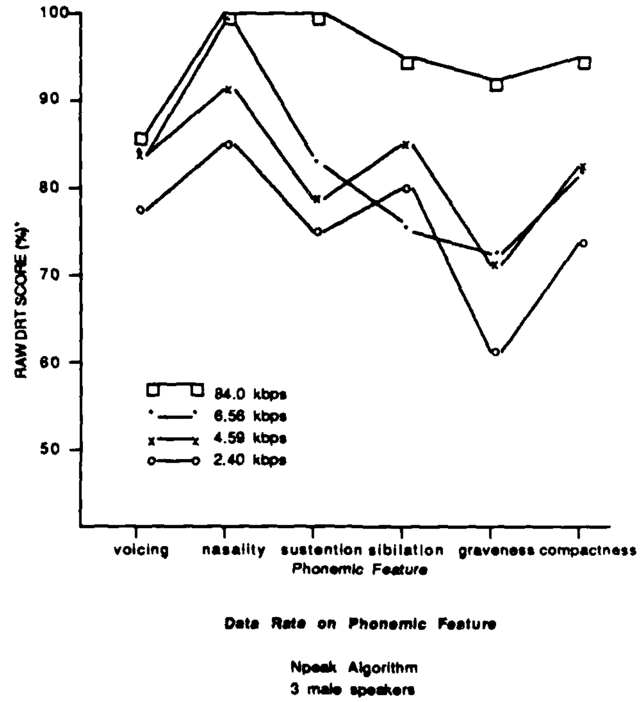

Figure 3-7b. Male speaker DRT phonemic results.

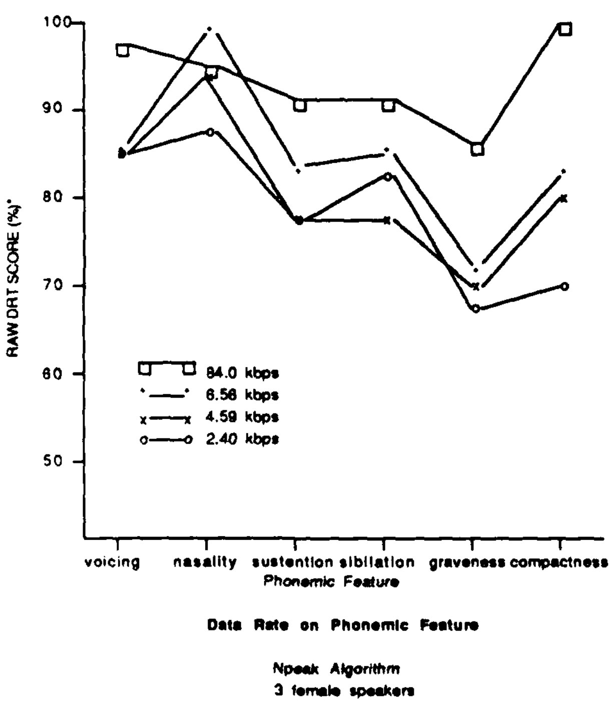

Figure 3-7c. Female speaker DRT phonemic results.

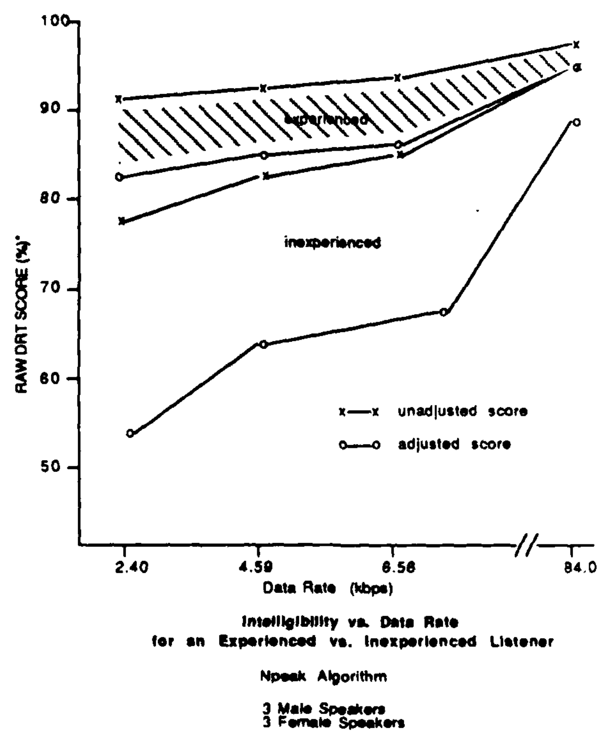

Figure 3-7d. DRT scores for the npeak algorithm with an experienced listener.

The responses from the listeners also showed that their performance improved as they became more experienced. Demonstrating this fact are the results from the experienced listeners who have several hours of listening time. The isolated experienced listener scores were as follows:

Table 3-4. Experienced listener DRT scores.

Data Rate (kbps)	Unadjusted (%)	Adjusted (%)
96.0	98	96
6.5	93	87
4.6	92	84
2.4	91	82

Comparing these scores to those of the LPC DRT of 88 [1], we can see that the experienced listener using the maximum spectral peak algorithm can perform as well as LPC.

3-4.4.2 PACT Results.

The purpose of these initial tests including bit errors was to compare the performance of the SFC and the LPC in similar environments. Bit errors were simulated by the system model and a channel model with randomly distributed noise or noise and fading. The bit error rates applied were 1% or 10% and in the noise and fading cases had decorrelation times of 1s (tau0 = 1s). The word sets were those described in the PACT description with three sets of the random alphabet sequence to prevent the listener from learning a sequence. Both the SFC and the LPC-10 were run through the same errored environment so the results are directly comparable.

Figure 3-8 displays the results from the initial PACT.

The results show that the SFC was just as survivable as the LPC-10 in all cases, and slightly better in the case of 10%

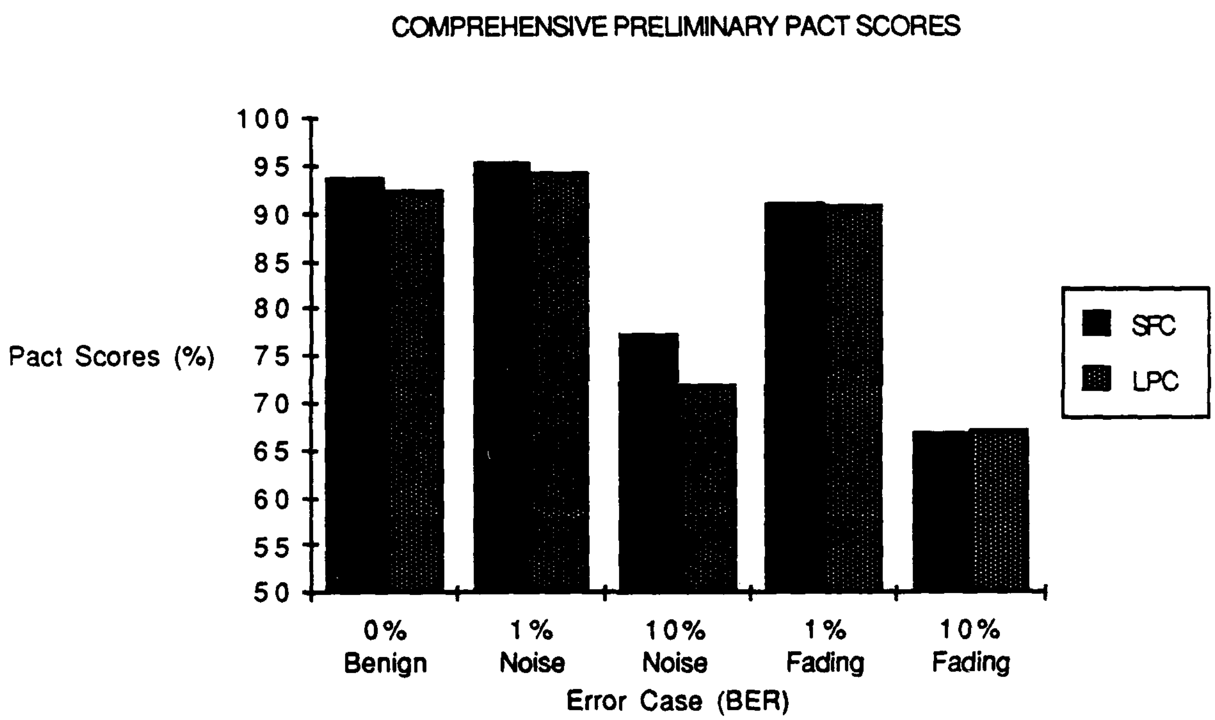

Figure 3-8. Comprehensive PACT scores.

random noise. In the benign case (no bit errors) both demonstrate higher intelligibility but the scores are lower than the 1% noise case. This is most likely due to the learning curve associated with the synthetic quality of the speech and with experience in taking the tests. In 1% noise both SFC and LPC perform reasonably well, maintaining better then 90% intelligibility. In the 10% noise, the intelligibility of both coding schemes suffers significantly. The LPC performed more poorly than the SFC; however the 70-80% scores indicate that roughly 1 of 4 words do not survive the bit errors. In the fading cases both the LPC and the SFC survive the 1% bit error rate, but both perform poorly at the 10% bit error rates.

For an experienced listener there was only a slight variation. Figure 3-9 displays the results for the experienced listeners.

The 10% noise case for the SFC survives the errors with nearly 90% word recovery rate where as LPC could only produce an 85% word recovery rate. In the fading case the errors still seem to be very severe and neither LPC or SFC can recover from the bit errors.

3-4.5 Conclusions.

The SFC voice using the simple initial algorithm sounds rather synthetic with some prominent, annoying clicks caused by the frame discontinuities at each boundary. Despite these artifacts, experienced listeners are able to filter out these artifacts. Apparently, the experienced listener was much more familiar with the quality of the synthetic speech, and wasn't distracted by the unusual character of the voice.

When bit errors are applied, the listeners perform about the same with both the LPC and the SFC. With high bit error rates LPC was prone to dropping out completely, resulting in losses of whole words, or parts of words making the phonetic

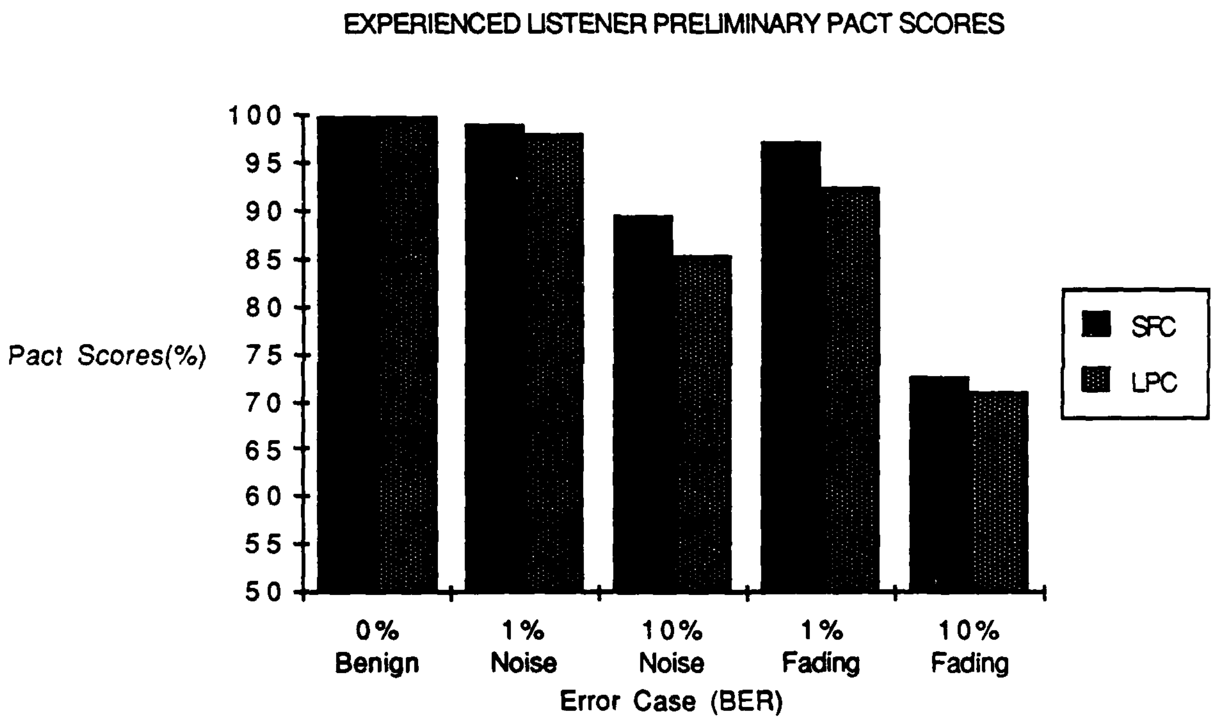

Figure 3-9. Experienced listener PACT results.

word spoken unrecognizable. With the SFC, the bit errors resulted in stray tones or inappropriate magnitudes of the frequency components chosen. Consequently, a frequency or magnitude could be in error for a particular frame, but there still remained the surviving frequency components containing valid information. Although part of the errored word may have corrupted frequency information, enough valid frequency information survives the errors to make the reconstructed word recognizable. The recognizability relied strongly on the listener's ability to mentally filter out the stray tones caused by the errors. The nature of the artifacts from the errors, are such that they may be corrected in most cases. These correction schemes, discussed in the next section, are able to recognize most decoded frequency and magnitude errors leaving only the valid frequency information for the reconstruction, resulting in significantly better performance for SFC.

The quality of the SFC benign speech, and of its performance in errors demonstrated the possibility of developing a low bit rate, survivable voice coding scheme. From here we decided to take a closer look at techniques for improving the quality of the speech with bit errors using straightforward error correction schemes uniquely suited to this SFC coding approach.

3-5 ERROR CORRECTION SCHEMES.

The majority of the stray tones in SFC due to bit errors can be eliminated by monitoring the decoded magnitude and frequency data and exploiting the structure of the transmitted data. By ignoring or correcting received frequency components which are clearly incorrect, stray tones at inappropriate frequencies can be minimized.

Normally in speech a smooth transition should occur if the frequency peaks drift, thus monitoring the peak selection from frame to frame can help correct errors in frequency identification. Whenever a decoded frequency bin is beyond the maximum deviation allowed from frame to frame the bin decoded is assumed to be in error and is deleted. Magnitude correction is obtained by sending the magnitudes for each frame in descending order; i.e., the first frequency bin sent has the largest magnitude, the second has the next largest, etc.. Should there be a decoded magnitude at the receiver that does not fall in proper order, then the magnitude in error is interpolated between the surrounding magnitudes.

The result from these two error correction schemes eliminated a very large portion of the stray tones due to errors. The reconstructed speech in many cases still had enough valid frequency information so that word recognition was maintained. The listener no longer had to filter out the stray tones and was able to concentrate more on the surviving information in the effort to recognize the word.

3-6 WAVEFORM CONTINUOUS ALGORITHM.

The success of the intelligibility experiments described in the previous section led us to seek a method of reducing the most distracting element of the SFC voice reconstructions: the discontinuities at frame boundaries. We felt that eliminating these clicks at the frame rate would not only improve the overall perceived quality of the reconstructions, but could also add to the intelligibility scores.

In searching for methods to eliminate frame discontinuities, several references were encountered which addressed this issue from the perspective of high quality, higher data rate coding [4,5]. Since the the techniques described in these references were similar to techniques we had begun pursuing, we applied modified versions of the reported methods to achieve waveform continuity for the low rate, synthetic quality coding technique being studied.

The basic idea underlying this approach is to synthesize the voice using sine wave generation rather than an inverse FFT on the frame, and allow the frequency, phase and amplitude of each of the 6 sine wave components to vary smoothly across the frame to match with corresponding components in the following frame. Components of the two frames which aren't close enough in frequency are not forced into phase continuity. Thus, components which are part of an extended vowel sound will remain phase continuous. The result of this synthesis is a reconstructed waveform with smooth but reverberant and mildly gurgly vowel sounds.

The specific interpolation procedure employed connects the amplitudes of the sine wave with a simple linear interpolation across the frame, and the phase is modeled as a cubic function of time. This cubic polynomial phase representation allows the phase to change smoothly across the frame as described in [4].

A quadratic phase function was also tried and very similar qualitative results were obtained, indicating that a simpler quadratic phase function could be used in a real-time implementation of the technique. The quadratic function results from allowing both the frequency and the phase to change linearly across the frame to match the initial samples of the next frame; i.e.,

$$f(i) = (a_1 + \Delta_a * i) * \sin((w_1 + \Delta_w * i) * i + p_1 + \Delta_p * i)$$

w_1 = frequency at frame 1

p_1 = phase at frame 1

Δ_w = (w2-w1)/N (with frame size N), and

Δ_p = (p2-p1)/N.

When voice is synthesized with the interpolation described above applied across the entire frame, the smooth frame transitions are achieved at the expense of more robotic voice quality. Furthermore, it's clear from examining plots of the voice data that frames synthesized without interpolation more closely match the initial waveforms. Consequently, we sought a method of ensuring frame continuity, but without disrupting the waveform over the entire frame. As a compromise, we allowed amplitude, frequency, and phase interpolation over the final quarter of each frame, but preserved the fixed sinusoids during the first three quarters of the frame. The result is a reasonable trade-off between the extremes of frame clicks with no interpolation and robotic voice that lacks sharp transitions with full-frame interpolation.

To determine the effect of this modification to the algorithm on word intelligibility in various noise and fading conditions, the PACT tests were administered again. The results of these tests are summarized in Figures 3-10 and 3-11. In these histograms, the benign case involved no bit errors, the cases labeled "noise" included randomly distributed errors at the bit error rates indicated, and the cases labeled "fade" included bit errors obtained from the link model in a fading channel with a 1.9 sec interleaver at the indicated error rate. Inspection of this histogram indicates that forcing waveform continuity improved the intelligibility several percentage points.

Moreover, this version of SFC (with the error correction) outperformed standard LPC in all cases when bit errors were present.

The PACT scores do not reflect the fact that the reconstructed female voice is significantly less warbly sounding than the male voice. This is attributed to higher pitch of the female talker, resulting in fewer total spectral peaks in the analyzed band. This in turn increases the likelihood that corresponding peak locations will be tracked for the entire span of a particular type of sound. For the male talker, a peak might be lower in amplitude for a particular frame as a result

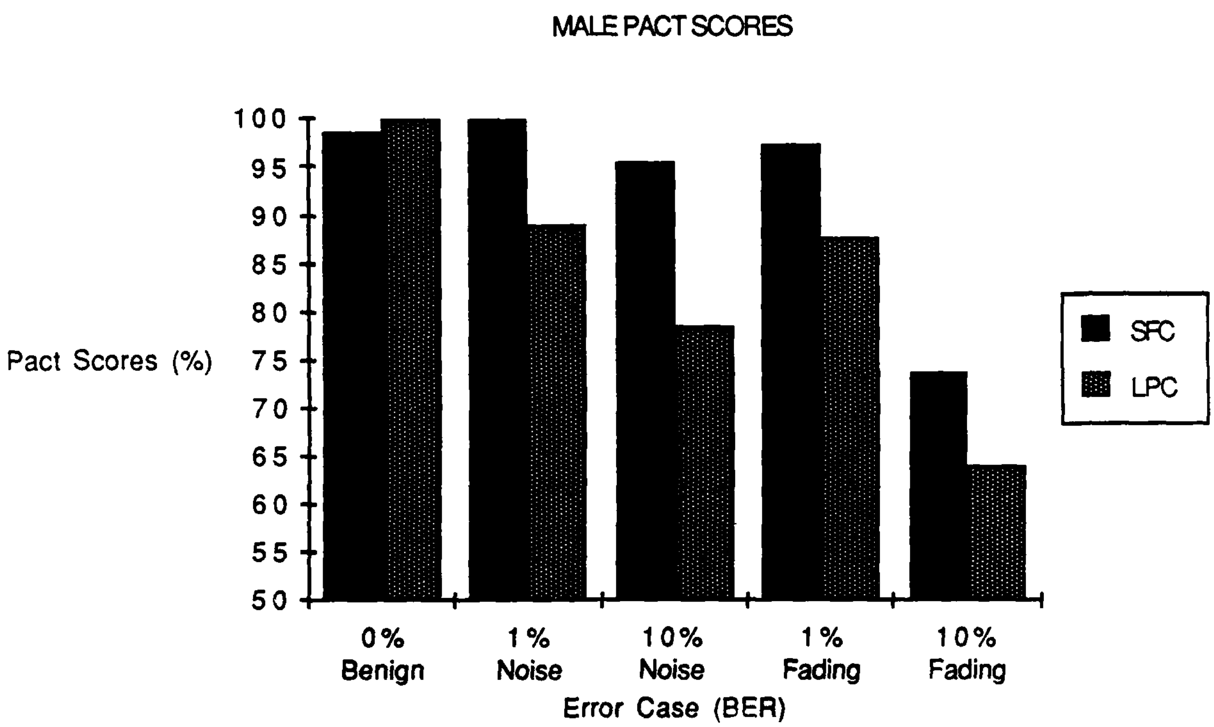

Figure 3-10. PACT scores for final SFC - male talker.

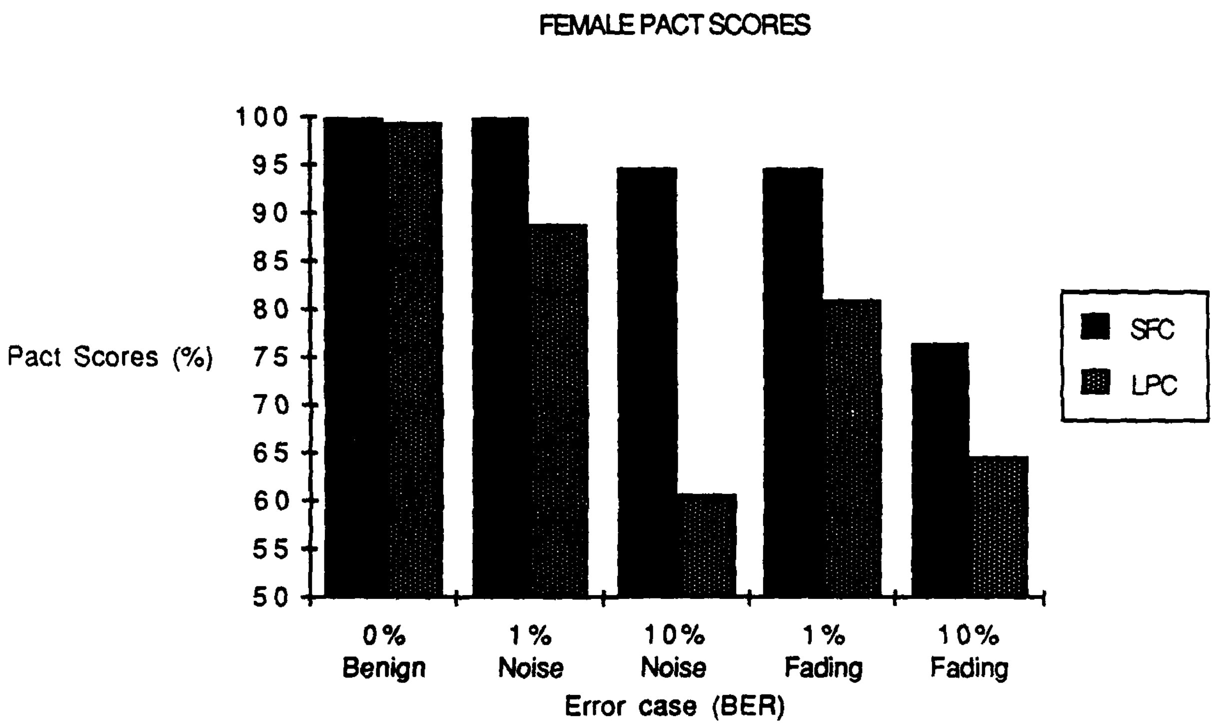

Figure 3-11. PACT scores for final SFC - female talker.

of the pitch pulse placement within a frame, for example. This temporarily lower peak amplitude can allow some other peak to be chosen occasionally within a given type of sound, causing a rough or warbling effect for the male talker.

Conclusions that can be drawn from this data are presented in section 4.2.

3-7 ALGORITHM COMPLEXITY AND IMPLEMENTATION.

In evaluating the proposed algorithm, it is worthwhile to consider the computational complexity to determine whether the technique can be implemented at reasonable cost. The computations involved are summarized in Table 3-5, which provides rough estimates of the computation time required by each block of processing using a typical, readily available and inexpensive signal processing microprocessor, the TMS 32010. As can be seen from this table, all the required processing could be performed by one or two such devices within the frame time of 32 msec. To focus on the signal processing requirements, this table assumes that line protocol issues such as signalling bits and frame synchronization are handled by a separate single chip microcontroller.

Table 3-5. Estimated SFC computation time for TMS 32010.

Analysis:	256 pt FFT (real input)	4.0 msec
	Compute Magnitude Squared	.2
	Pick Peaks	.2
	Compute Magnitude & Phase	.6
	Quantize Peaks	.3
	Pack data	.5
Synthesis:	Unpack data	.5
	Linearize magnitude	.3
	Match phase to next frame	.3
	Generate sines	2.0
	TOTAL	8.9 msec
	Time available (256 samples)	32.0 msec

SECTION 4
CONCLUSIONS AND FUTURE DIRECTIONS

4-1 ALGORITHM SUMMARY.

The final algorithm for voice compression resulting from this study can be summarized as follows:

1. Section the input voice into 256-sample blocks. As each block is digitized, apply an FFT.

2. Convert the {Re,Im} pairs of transformed data into magnitude squared (Re*Re + Im*Im).

3. Find the FFT bin numbers of the local peaks in spectral magnitude. A peak bin is simply a bin with an amplitude greater than the amplitudes in both adjacent bins.

4. Sort the peaks to determine the bin numbers of the 6 largest peaks.

5. Determine phase of the 6 peak bins.

6. Match the peaks to corresponding peaks in the previous frame by determining the bin in the current frame closest to each bin of the previous frame. If corresponding peaks are more than 2 bins apart (62 Hz), then disqualify the match. If 2 bins of the previous frame link to the same bin of the current frame, preserve the link with the largest amplitude.

7. Linearly quantize the amplitude and phase to 3 bits for each of the largest 5 peak bins. These are transmitted in order of descending peak amplitude to permit error correction at the receiver. The final (smallest) peak is transmitted with amplitude quantization of 2 bits

over half the range of the first peak, and with phase quantization of 2 bits.

8. Transmit the bin number (7 bits) and quantized amplitude and phase for each bin.

9. At the receiver, apply data checking:

   a. Ensure that peaks are specified in decreasing amplitude order. Apply smoothing to ensure this order if necessary.

   b. If any peak locations are separated by more than 4 bins from peaks in either the previous frame or the next frame, do not synthesize them. This reduces the energy.

10. For peaks that survive the data checking, synthesize 192 samples of the voice using a sum of constant phase and amplitude sine waves. This can be done either using an inverse FFT or directly generating the sinusoids through table lookup, weighting them appropriately and adding.

11. The final 64 samples of the frame are generated by determining the phase at sample 192, then computing for each peak frequency:
    at = (new amp - old amp)/64
    dph = (new phase - old phase)/64
    dk = TWOPI*(new bin # - old bin #)/(256*64)
    Then, for all remaining 64 samples compute
    s(i) = s(i)+(a1+at*im)*sin((w1+dk*im)*i+dph*im+t1)
    where i=192..256, and im=i-192, and a1,w1, and t1 are amplitude, frequency, and phase of the corresponding frequency component in the previous frame.

All 256 samples of the frame have now been computed.

4-2 CONCLUSIONS.

Inspection of Figures 3-10 and 3-11 clearly demonstrate four significant results:

1. Word intelligibility of SFC with no bit errors is about 100%.

2. Word intelligibility of SFC in a 10% randomly distributed bit error rate remains high (95%) while word intelligibility of LPC-10 drops dramatically (60-80%).

3. In fading with a 1.9 sec interleaver length, SFC intelligibility degrades much more slowly than LPC.

4. As a consequence of result 2, intelligibility loss from fading with SFC can be avoided at up to 10% average bit error rate by using an interleaver. This strategy does not work for LPC.

Not apparent from the PACT scores is the reverberant quality of the voice. This aspect of the speech quality is initially distracting, but within a short listening period the listener adapts to the reverberance.

4-3 FUTURE DIRECTIONS.

The positive results of this development indicate that several issues should be studied further before arriving at a specific recommendation for techniques to improve the survivability of low data rate voice communications systems against atmospheric nuclear device effects.

4-3.1 Real-time SFC Implementation.

First, the SFC technique should be implemented in a real-time test set to permit a more comprehensive evaluation of the voice quality acceptability in fading and non-fading environments. Due to the simplicity of the technique, this implementation could be efficiently accomplished using off-the-shelf digital signal processor cards compatible with an IBM/AT computer. Further improvements to the coding technique might also be considered at this time. For example:

- the encoder should consider the peak locations of both the next frame and the previous frame to choose the peak set for any specific frame.

- more thought might be given to coding the peak locations efficiently, then adding redundant coding to protect against transmission errors;

- RELP (residual-excited linear prediction) vocoder techniques could be applied to the received data which might permit smooth frame transitions without the robotic quality of full frame interpolation;

- the possibility of coding z-plane pole locations rather than FFT bin number and phase should be examined.

4-3.2 Redundantly Coded LPC-VQ.

In addition, other possible techniques should be surveyed. In particular, the technique of using vector quantization in conjunction with conventional LPC vocoders should be considered. Vector quantization is a technique for reducing the number of bits required to represent a frame of LPC voice, and has typically been applied to the problem of transmitting voice at data rates below 1000 bits/second. By reducing the basic data rate, then adding error protection to bring the data rate back to 2400 baud, it may be that LPC can be made more effective in a fading environment.

4-3.3 Real-time Error Simulator.

Finally, some mechanism is needed to tie PACT intelligibility scores or DRT scores to the actual impact on typical operations for currently deployed AN-DVT units. This could be efficiently accomplished with an inexpensive unit which corrupts communications with bit errors corresponding to typical atmospheric nuclear effects for specific link models during an operational exercise. Such a unit could be assembled from a standard personal computer with a data base of bit error patterns for various combinations of decorrelation times and interleaver buffer sizes. An operator could specify a particular scenario, then the appropriate bit errors would be injected into the data stream. Implementing an error pattern at the modulation baseband is dramatically more cost effective than performing such a test with an RF atmospheric effects simulator, permitting several such units to be used during the tests or as training devices to prepare personnel and adjust communications protocols to cope with channels degraded by atmospheric effects.

SECTION 5
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