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Military Radio Communications Research in Australia

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ABSTRACT

An overview of recent research by the Australian Defence Science and Technology Organisation in the field of military radio communications is presented. A philosophy for improving digital radio system performance over complex, variable channels is outlined. A key breakthrough, called PDF-directed adaptive radio, which can provide substantially greater throughput over HF channels whilst minimising bit-error rate and delay, is described. Simulation results for fast adaptive schemes applied to both serial-tone and parallel-tone HF modems are presented and shown to significantly out-perform fixed rate modems and modems employing hybrid automatic-repeat-request schemes. A new detector scheme is discussed which has superior performance to conventional detectors for digital traffic in the presence of inter-symbol interference and impulsive noise.

1. INTRODUCTION

The Australian Defence Science and Technology Organisation (DSTO) is undertaking significant research which is yielding the possibility of substantial improvements in the performance, robustness and flexibility of military radio communications. This work is being carried out as a key component of the Defence organisation integrated communications (DORIC) program which is the flagship of DSTO's military communications R&D activities. DORIC is a major research program aimed at defining and demonstrating an integrated communications architecture for the Australian Defence Organisation into the twenty-first century.

This architecture will enable the integration of Defence organisation communications at three levels, namely traffic, Defence networks, and Defence/civil networks integration. The architecture research is centered around the provision of a global backbone for C^2I traffic. Hence, it takes into account the possibilities of transport of large databases, video conferencing, imagery, sensor data and intelligence traffic, in addition to the more traditional Defence traffic and it also factors in the uncertainties regarding the future volume, mix and growth rates of these various types of traffic.

In this context, it is crucial that Defence radio links and networks should be capable of carrying a widely differing range of traffic with maximum throughput and guaranteed quality of service (QoS). Nowhere is the discrepancy between these future requirements and the performance possible with current conventional technology greater than in the field of HF radio. Although the bandwidth constraints at HF will prevent its use for broadband applications, its enduring role in military communications warrants research effort to meet the DORIC goals for this important medium.

The key to a breakthrough in this area is a technology which DSTO calls probability density function (PDF) directed adaptive radio. This technology, in which DSTO has made significant advances, promises to yield substantial improvements in military radio communications throughput, reliability and availability, particulary at HF. It extends to the tactical user the same integration benefits along with efficient, cost-effective use of available transmission assets as are available to users at the fixed core of the military

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network. This paper outlines the results of current DSTO research in this field.

A PHILOSOPHY FOR IMPROVING DIGITAL RADIO COMMUNICATIONS

The optimal design of digital communications systems which are required to operate over stable, predictable channels is well documented. There are many instances, however, where radio channels do not fall into this category. Typical examples are military radio systems employing both terrestrial and satellite links, systems relying on ionospheric or tropospheric propagation, and systems which are sensitive to natural or deliberate interference. In these cases, the channel impulse response and the noise statistics are liable to be time-varying and optimisation of such links continues to be a subject for research. The traditional approaches of providing a combination of a fade margin, diversity reception, and ever-increasing amounts of signal processing, fail to achieve satisfactory performance in many cases, even though the channel may have sufficient average capacity to carry the desired traffic. There is no single panacea for optimising digital radio systems operating over complex, variable channels. So we have developed the philosophy of attacking the problem at a number of levels.

At the physical layer, communications theory has been critically re-examined to determine if the underlying assumptions, so often made to make analysis possible, are valid for the complex channels fequently encountered in military communications. The most prominent assumption in traditional communications theory is that the noise process can be adequately modelled as additive Gaussian, where the only parameter to be determined is the variance or noise power. This assumption has influenced the design of many of the signal processing stages in modems, including the filters, equalisers, symbol-timing recovery schemes, and detection. A recently conceived approach[1] to optimally estimate the channel impulse response and the data is expounded in the following section. It has been shown to significantly reduce error probability of radio systems operating in arbitrary noise and distortion environments.

At the data link layer the second tier of the philosophy is implemented through a concept called dynamic adaptivity[2,3]. In this, the configuration (i.e., modulation, coding, power output, data rate, frequency, protocols, etc.) of the radio link to suit the prevailing channel conditions and the current traffic mix is adapted. The concept is to automatically adapt the radio system parameters on three time scales—short, medium and long—which are derived from the observed rates of change of the channel state for the particular transmission medium.

The most ambitious strategy and the one with the most dramatic potential for improving radio system performance is fast adaptation. Many digital communications channels suffer fading due to propagation effects. Whether this be from ionospheric effects or multipath, errors occur due to reduced signal-to-noise ratio (SNR) and rapid phase changes in the depths of a fade. Short-term adaptation schemes rely upon the ability to determine the current channel state in a fraction of a fade period, and from this and its history to predict when such fades will occur and modify transmission during these periods of poor propagation. This modification can take the form of changing the bit rate or changing the error-control coding rate in a time frame of a few per cent of a fade, typically tens of milliseconds at HF. Two such schemes which can be used to substantially increase throughput over existing systems while guaranteeing a probability of error ceiling are described in Section 4.

Diurnal variations affect most unguided electromagnetic transmission media, in particular HF radio, meteor burst, and troposcatter systems. However, conditions can be relatively stable for periods of hours. Furthermore, natural and/or deliberate interference can be present on military radio channels for periods of minutes to hours. The adaptation process which responds to such events is termed as medium-term adaptation. Adaptation in this case is based on the selection of the most suitable link configuration. An even slower adaptation rate, long-term adaptation, takes into account the slowest changing phenomena such as the eleven year solar cycle, which heavily influences HF propagation and annual changes which influence many radio channels particularly HF and meteor burst. Long-term variations influence choice of operating frequencies, antennas, equipment configuration, and operating procedures. A knowledge-based system, currently under construction and described in Section 5, will handle overall control of the PDF-directed radio and perform adaptation at

Table 1. Summary of adaptation strategy

Adaptation	Events	Time frame	Typical changes
Long-term	Solar cycle Seasonal	Months to years	Antenna configuration Operating procedure Power output Modulation
Medium-term	Diurnal	Tens of minutes to hours	Modulation Error control Data rate Frequency
Short-term	Fade	Tens of milliseconds to minutes	Coding rate Data rate

these two slower rates. A summary of the adaptation strategy is presented in Table 1.

There is a growing trend to network military communications links in keeping with the trend in the civil arena. Networking provides greater survivability, availability, and connectivity. However, the wealth of work performed in the civil sector, particularly as embodied in civil standards is not entirely appropriate for military radio networks. This third tier of the philosophy is currently under investigation and shows promise[4].

3. PDF ESTIMATION AND SELF-ADAPTIVE CONTROL

In the field of communications over recent years, the emphasis on systems which will perform better over channels subject to fading and inter-symbol interference (ISI), and non-Gaussian noise, has led to the realisation that significant performance improvements would be possible if the detector and parameter estimation in a receiver were optimised for these conditions. Recent DSTO work[1] represents a breakthrough in the understanding of how to design suitable schemes for digital signalling over such channels, based on the estimation of PDFs.

Early papers expounding optimum detection and estimation such as those by Neyman and Pearson[5] and Woodward and Davies[6], quickly focussed onto the Gaussian noise case; a trend that has continued until now. The determination of optimal detection and estimation schemes to operate in the presence of arbitrary noise and distortion has not been pursued to date in the literature, presumably due to a lack of understanding of how to accurately estimate the parameters necessary to identify the PDF of the noise. Recently, a method of PDF estimation was invented by Scholz, Cook and Giles of DSTO as the basis for estimating error probability[7-9]. This method was based on estimating the PDF of the receiver decision variable and using this to index a lookup table to determine the error probability. From this work, it became evident that, given the noise PDF, it may be possible to design superior detection and estimation schemes.

In general, the received signal may be considered to be of the form :

$$r = hx + n(\theta) \tag{1}$$

where r is the received signal vector, $r = \{r_1, r_2, \ldots, r_N\}$, x is the transmitted signal vector, h is the distortion (impulse response) matrix, $n(\theta)$ is the noise vector, and θ is a vector of parameters which describe the noise.

This model can incorporate all forms of signal distortion including ISI and fading in the h matrix. Random noise and deterministic interference are modelled in the n vector. Thus, noise samples which constitute this vector may in general be dependant as expounded in the full formulation[10], but for this discussion the simpler case of independent (but arbitrarily distributed) stationary noise is assumed.

The optimum detector for this generic model may thus be formulated as:

$$\hat{x} = \underset{x}{\operatorname{argmax}}\; p\,(x, h, \theta \,|\, r) \tag{2}$$

By application of Bayes rule it is equivalent to consider maximisation of the *a posteriori* probability $p(r\,|\,x, h, \theta)$. The joint PDF of the noise $p(n)$, reduces to the product $p(n_1)\, p(n_2) \ldots p(n_N)$ for the case of independent noise samples. It is thus equivalent to consider choosing the data x which maximises :

$$\phi = \prod_{i=1}^{N} p_n(r_i - hx \,|\, \theta \tag{3}$$

where $p_n(.)$ represents the PDF of the noise process. The noise PDF is assumed to be completely specified by the parameter vector θ.

The nature of the types of noise process to be expected is generally known, however, the particular type of noise or noise mixture needs to be estimated.

One way to estimate θ is to use a maximum likelihood (ML) estimator. Consider that a library containing L PDF models is used. Then, an ML estimate of the parameter vector θ (and hence the PDF) may be stated from Eqn (3), as :

$$\hat{\theta} = \underset{\theta}{\text{argmax}} \prod_{i=1}^{N} p_n (r_i - hx \mid \theta = \theta_j) \qquad (4)$$

$$\{j = \ ,2, \ ,L\}$$

This may be implemented by applying estimates of the channel impulse response h and data x into Eqn (4) and evaluating the PDF for each parameter vector value $\theta = \theta_i$.

Consider the simple example of detection with no ISI or fading so that $h = \{1, 0, 0, \ldots, 0\}$, but with arbitrary noise. If the *a priori* probabilities for all noise models are equally likely and all channel models are equally likely, we choose the data x which maximises :

$$\rho = \prod_{i=1}^{N} p_n (r_i - x \mid \theta) \qquad (5)$$

Since the data is uncorrelated, we can make data decisions on a symbol-by-symbol basis. So the optimum detector calculates :

$$\hat{x} = \underset{x}{\text{argmax}} \; p_n (r_i = x \mid \hat{\theta}) \qquad (6)$$

If we consider that M samples constitute one symbol (as say for FSK detection), then

$$r_i = x_i + n$$

$$r_{i+1} = x_i + n_{i+1}$$

$$r_{i+M-} = x_i + n_{i+M-1} \qquad (7)$$

The PDF in Eqn (6) becomes

$$p_n (r_i - x \mid \hat{\theta}) = \prod_{j=1}^{M} p\,(r_{i+j} - x \ \hat{\theta} \qquad (8)$$

Substituting the value of Eqn (8) into Eqn (6) gives the test

$$\hat{x} = \underset{x_i}{\text{argmax}} \prod_{j=1}^{M} p\,(r_{i+j} - x_i \mid \hat{\theta} \qquad (9)$$

or equivalently

$$\hat{x} = \underset{x_i}{\text{argmax}} \sum_{j=1}^{M} \log p\,(r_{i+j} - x_i \mid \hat{\theta}$$

For the simple case of additive white Gaussian noise (AWGN) only, the above test reduces to finding the value of x_i which produces[11] the largest correlation ζ.

$$\zeta = \sum_{j=1}^{M} r_{i+j}\, x_i \qquad (11)$$

In order to demonstrate the performance improvements possible in a non-Gaussian noise environment, a mixed AWGN and impulse noise model has been used to evaluate conventional and PDF-directed detectors. The impulse noise model used was that advocated by NTIA[12] and is bi-variate Gaussian distributed in the complex plane (i.e., has Raleigh distributed magnitude and uniform distributed phase). Noise impulses are typically infrequent but possess great magnitude compared to the received signal, effectively swamping the background AWGN. The resulting noise PDF may be modelled thus with $s_j = r_{i+j} - x_i$

$$p(s_j) = \alpha p_b(s_j) + (1 - \alpha) p_z(s_j)$$

where a is the probability of an impulse (uniform distribution), $p_b(.)$ is the background noise PDF, $p_z(.)$ is the impulse noise PDF, and

$$p_b(s_j) = \frac{1}{\sigma_h \sqrt{2\pi}} \exp\left(\frac{s_j}{2\sigma_b^2} \right) \qquad (13)$$

$$p_z(s_j) = \frac{1}{\sigma_{\ } \sqrt{2\pi}} \exp\left(\frac{s_j}{2\sigma_z^2} \right) \qquad (14)$$

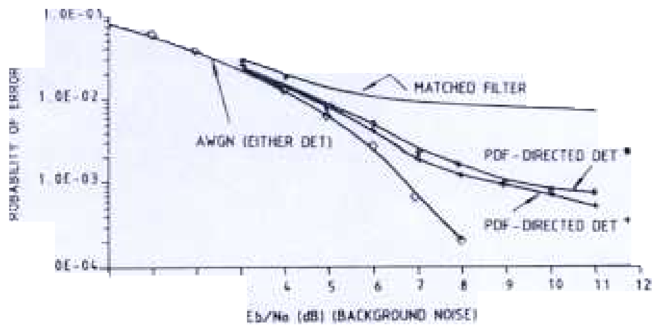

Figure 1. Detection performance, for M=2 samples per symbol in a channel with AWGN and impulsive noise, a=0.01, SIR= -20 dB.

The lower curve in Fig. 1 shows the performance of either detector in the presence of AWGN (background noise) only. The upper-most curve shows the performance of a conventional correlator in the presence of mixed AWGN and impulse noise. The centre curves show the performance of the PDF-directed detector.

The PDF-directed detector performance curves show the theoretical performance with a perfect PDF estimator and the performance of the same detector using a practical PDF estimator. In the latter case, a three-dimensional estimate of θ was used since $\theta = \{a, \sigma_b, \sigma_z\}$. The SNR and signal-to-interference ratio (SIR) were defined as follows :

$$\mathrm{SNR} = \frac{1}{2\sigma_b^2}, \quad \mathrm{SIR} = \frac{1}{2\sigma_z^2} \qquad (15)$$

Models used covered the ranges a = 0, 0.1, 0.001, and 0.0001; SNR = 0, 2, 4, ..., 20 dB; SIR = -30, -25, -20, ..., 0 dB; which gives 308 models in total (N = 200 samples were used).

Clearly the performance benefits of using a truly optimum detector structure are obvious for non-Gaussian noise. The performance benefits become even more evident with an increase in the number of samples per symbol.

The above concepts have recently been extended to allow optimum symbol timing recovery[13] and block synchronisation as well as rapid identification of modulation types[14].

4. FAST ADAPTATION

Fast adaptation is capable of providing guaranteed quality of service and improved throughput with minimal delay for a range of traffic types over complex channels. Adaptation is performed by altering the compromise between throughput, error control strategy and delay, in response to short-term channel conditions. Cavers[15] determined that to achieve optimal performance fast adaptation response needs to be no more than a few per cent of a fading period. This mandates rapid channel estimation and prediction and seamless changes in link configuration which in turn constrains the possibilities for adaptation strategies. Two feedback schemes for HF links have been examined by DSTO, one for use with serial-tone modems, the other for the parallel-tone variety.

The first scheme alters the constellation-packing. That is, it maintains the maximum symbol rate whilst altering the number of bits per symbol. This scheme is well-suited to serial-tone modems and can be used to exploit periods of high SNR. Shannon[16] derived the well-known expression for the channel capacity (for AWGN) which is proportional to the logarithm of the SNR. This relation indicates that enormous peak throughput increases are possible for meteor burst and HF channels in Australia where short-term SNR has been observed to regularly exceed 30 dB[17,18], thus allowing potential increases to peaks exceeding 38.4 kbps.

A constellation-packing adaptation system for a serial-tone HF modem has been described by Jayasinghe[19] of DSTO. This system calculates the sum of the squares of the sampled channel impulse response and uses this to determine which one of three constellation-packing rates to employ, namely, 16-QAM, QPSK, or QPSK with null data. These correspond to bit rates of 9600, 4800, and 0 bps. Figure 2 contrasts the error performance of the adaptive scheme against two fixed-rate schemes operating in a severe multipath channel. Given that the adaptive scheme achieves an average throughput around 8200 bps, it can be seen that it substantially outperforms the fixed 4800 bps scheme and offers hitherto unattainable throughput at such a low average SNR.

The second scheme maintains the desired QoS by adapting the error-control coding rate to suit the short-term SNR[9,20]. This scheme is well-suited to parallel-tone modems operating over fading channels

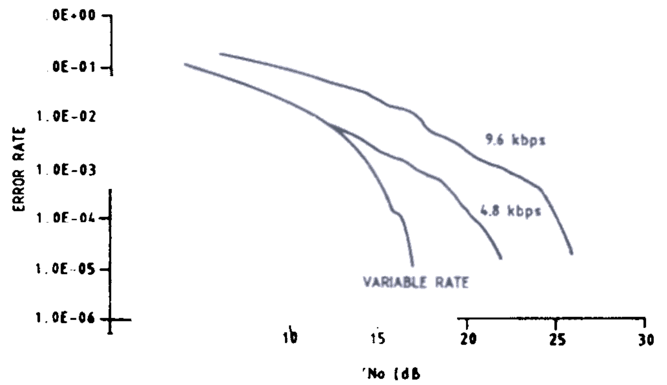

Figure 2. Performance of an HF feedback adaptive serial-tone modem.

because the large number of bits per symbol period map onto usefully-long error-control codes. Also it is relatively easy to predict the probability of error for a few symbols into the future due to their simple memoryless structure. The noise PDF is estimated using the scheme alluded to earlier[7-9]. In conjunction with this, a predictor for the fading vector is employed. The expected pre-decoder error probability for future symbols can be found by overlaying the estimated noise PDF onto the predicted vector and calculating the area of the noise PDF beyond the decision thresholds. This value is then used to calculate the expected number of post-decoder errors that would occur using each available error control code from the weakest to the strongest. The code chosen is the first to be just within the post-decoder error probability requirement. The feedback information is an index to the code rate or to the noise PDF depending on which end of the link performs the majority of the processing.

Figure 3 shows the comparative performance of this technique operating over a fading HF radio channel with eight available coding rates against

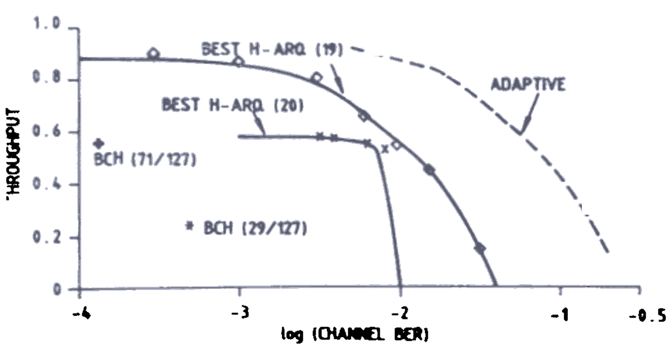

Figure 3. Comparison of a code rate adaptation scheme, H-ARQ, and fixed codes, post-decoder error probability $<10^{-4}$ over an HF channel.

two contemporary hybrid-automatic-repeat-request (H-ARQ) schemes[21,22] advocated for HF radio links. It also shows that the adaptive code rate system out-performs the other techniques, particularly at high error rates. The technique achieves a guaranteed QoS with minimal delay and high throughput for an HF link, extending the range of useful operation to previously inadequate SNRs. The system described outputs the data in the order it was sent, obviating the need for packet numbering, repeat requests, and packet sorting at the receiver, thus reducing the time delay. The concepts used here may be easily extended to other fading environments such as mobile radio, mobile satellite services or troposcatter.

5. INTELLIGENT CONTROLLER

Fast adaptation can optimise the error performance versus throughput for a particular link configuration (i.e., modem type, error-control coding type, baud rate, power output and frequency of transmission). However, it cannot alter the link configuration if that configuration is no longer optimum for the traffic type being transmitted and the prevailing channel conditions. This can be crucial, for example, on a fading channel where it is necessary to match the modem in use to the average SNR ratio. It is well-known that serial-tone modems are the best choice when good SNR is available whereas parallel-tone modems would be the best option for noisy or flutter-fading channels[23,24]. For channels which are dominated by co-channel interference, low-rate communications can be achieved with a chirp modem[25]. When a number of modems are available, the optimal selection depends on the expected quality of service of each for the current average channel conditions. Similarly, the selection of the optimal error-control strategy (i.e, the type of code, the depth of interleaving, and the length of the code word or, alternatively, the constraint length of the code) is sensitive to the delay the traffic can tolerate and the expected error burst characteristics of the chosen modem operating in the current channel.

Conventional control engineering techniques are not well-suited to the task of controlling the radio configuration because the link transfer function changes significantly with configuration changes. There is, however, an established body of knowledge concerning appropriate configurations for given channel states and traffic requirements. This knowledge can be found in the literature, from analysis and simulation, and from experienced radio operators, indicating the use of knowledge-based techniques. Fortunately, alterations in radio link configuration need to occur at much slower rates than fast adaptation because the configuration must match the average channel statistics which can only be ascertained over many fading cycles. Thus, the slow response often associated with artificial intelligence software is not of concern.

A prototype adaptive link experimental controller (ALEC) is currently under construction within DSTO[26], based on the real-time problem-solving blackboard system concept popularised by. Nii[27]. Blackboard systems comprise three major components: a number of independent agents, a blackboard data structure which is used to hold problem-solving state data and provide for interaction between the agents, and a control entity which co-ordinates the functions of the system. The agents can be thought of as independent knowledge-based systems each of which in turn comprises a knowledge source and an inferencing engine. A blackboard system is well-suited to the task as the control problem can be modularised into interacting agents which synergistically determine a suitable radio configuration. The adoption of an architecture which uses distributed knowledge sources has the distinct advantage of avoiding the construction and maintenance of a single large knowledge base. It also facilitates the possibility of replacing knowledge-based processing with algorithmic techniques at a later date.

It can be seen from Fig. 4, a block diagram of ALEC, that there are three inputs: channel-related measures, traffic and network requirements, and knowledge of the capabilities of the radio. The raw input information is processed by the three input agents into an intermediate form suitable for placing on the blackboard. Thus, a traffic input such as voice is translated into say 2400 bps, error rate less than 5×10^{-3}, less than 300 ms delay, continuous transmission and moderate tolerance to error bursts. Similarly, the platform input is translated from the list of modules available in the radio into capabilities supported; this feature makes the control software generic for a range of radio platforms.

The third input agent, the channel state interpreter, is a key feature of the system. This agent accepts input from various parts of the radio to determine its actual measured or estimated characteristics as well as the generic channel type. The prime measurement required is the channel error probability and this is obtained using the extremely versatile error rate estimator (EVEREST)[7-9] technique operating on the PDF extracted from the receiver decision variable. The technique matches the received PDF against a library of values and selects the best match using a maximum likelihood test. The error probability is then set to that determined for the library model. In all statistical estimators there is a compromise between measurement variance and the sample size. This is a major issue where the measurand is time-varying since the simple expedient of taking many samples does not yield a more accurate result. It has been shown that the EVEREST technique is a most efficient estimator of probability of error in that its variance is the smallest achievable for a given number of samples[28]. The measurement variance can be found from :

$$\sigma_{\rho}^{2} = \frac{0.0300\,\gamma}{NP_{e}^{2}\,e^{2\gamma}} \qquad 16)$$

where N is the number of samples, γ is SNR (E_b/N_o), and P_e is the probability of error as estimated by the technique.

The long-term adaptation agent asserts knowledge of expected propagation conditions, path geography, and time-related phenomena of interest during the proposed link lifetime, on the blackboard. This achieves long-term adaptation. The valid configuration generator examines the contents of the blackboard and creates a list of possible configurations which are, in turn, placed on the blackboard. The configuration evaluator examines each of the proposed configurations, including the current one and predicts their performance on the prevailing channel. A cost metric is generated for each which takes into account the relative performance of the proposed configurations and the disruption that would result from changing the configuration. In the event that the current configuration no longer meets the traffic performance requirements, the proposed configuration which exhibits the lowest cost is selected as its replacement.

Once a configuration change is signalled to the blackboard, the output agents go to work. The radio configuration generator reads the new configuration and

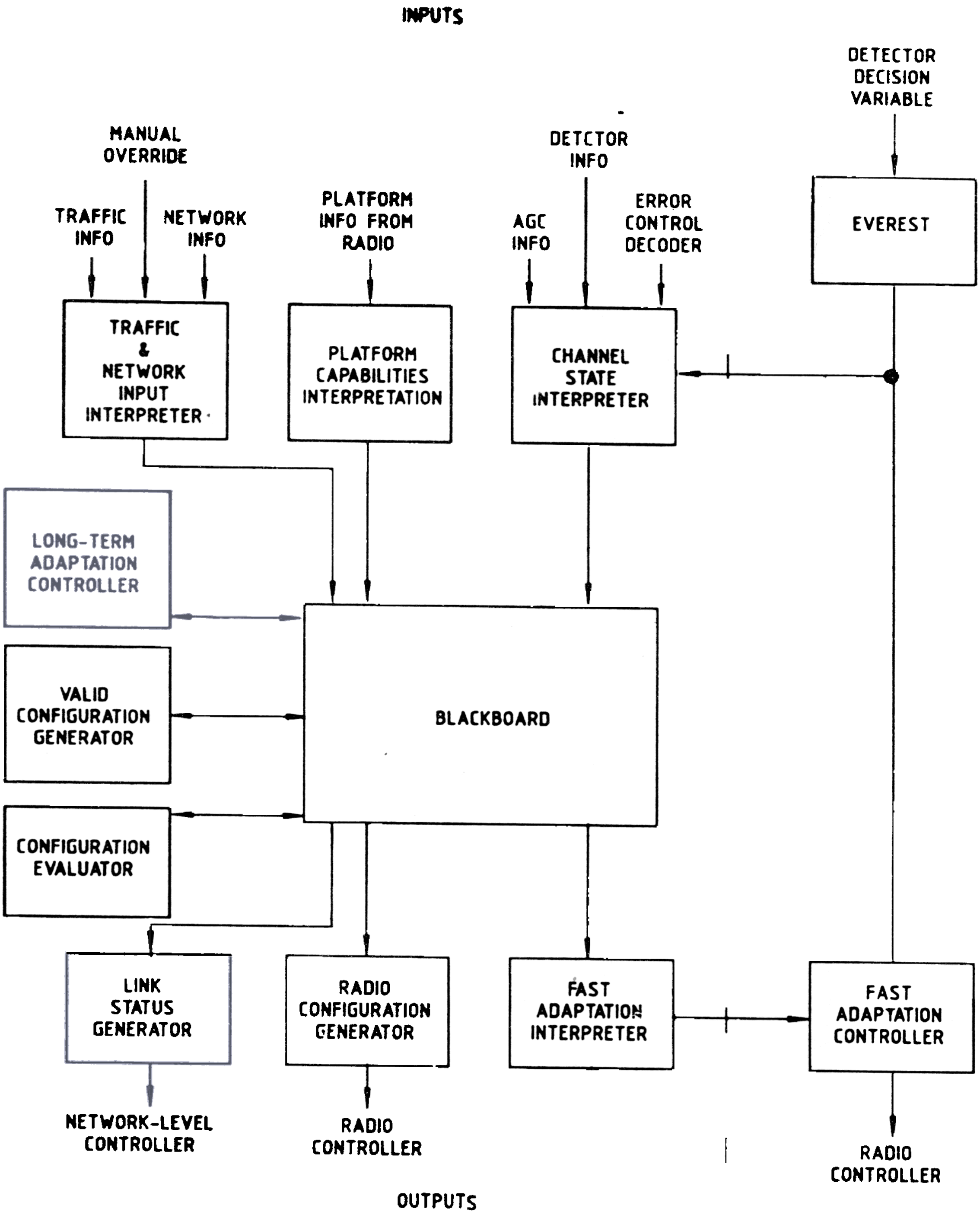

Figure 4. ALEC architecture.

translates it into low-level hardware- and software-specific instructions for the radio platform. The fast adaptation interpreter determines the parameter (for example, symbol packing rate, error-control coding rate, baud rate, etc) which should be controlled via fast adaptation to maintain error performance. It should be noted that once this parameter is chosen, the algorithmic fast adaptation controller operates without interaction with the knowledge-based system.

The last agent, the link status generator, supplies real-time quality of service information (for example, average throughput, error performance attained, transmit delay, estimated link lifetime, the presence of security devices) to higher level communication entities. This agent is also capable of responding to user or maintenance diagnostics. Thus errors in reasoning and incorrect entries in the knowledge base can be made readily apparent and maintenance and troubleshooting, the bane of the knowledge-based system designers, is ameliorated.

6. REALISATION

The dynamic adaptivity techniques and strategies are being implemented on what we term an uncommitted radio[29]. By this it is meant that the radio parameters (modulation type, coding type and rate, data rate, link protocol, etc) are not pre-defined to a limited set as in a conventional radio. Rather, the radio is fully programmable and can be configured automatically to any of a wide range of communication link configurations limited only by the availability of software modules.

The radio consists of a minimal analog RF front-end and associated analog/digital and digital/analog converters ahead of a powerful parallel processing engine. Through the use of DSP techniques the parameters and functionality of the radio are then implemented in software on this engine. We estimate that the full functionality to meet our goals can be achieved with a processing power of up to 10^{10} operations per second. This includes the capability to simultaneously process at least two information streams. Current technology projections predict the required power should be available on five chips by turn of the twenty-first century[30].

An interesting property of uncommitted radios is that, by virtue of their parametric freedom and multi-processing capability, they can be configured to emulate other radios, yielding essential interoperability and can also act as gateways between incompatible networks, for example, cellular mobile and HF.

7. CONCLUSION

A philosophy for improving the performance of military radio communications over complex, variable channels as typified by HF, has been outlined. Key breakthroughs based around a new technology called PDF-directed adaptive radio show that significant performance improvements over conventional digital HF radios in terms of throughput, bit-error rate and delay can be achieved. A burgeoning experimental program is underway to validate the theoretical and simulation results within the next year.

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