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COURSE ON BASIC TELECOMMUNICATIONS SCIENCE

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Communications Signal Simulator Signal-Signal

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These notes are intended for internal distribution only.

COMMUNICATIONS SIGNAL SIMULATOR SIGNAL - SIGNALS

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GETTING STARTED

On the discs you will find the SIGNAL files and other files for running the system. Make a backup copy of the discs.

The files on the discs are:

```
README        Latest details
SIGNAL.EXE    The programme for machines
              with a maths co-processor
SIGNALS.EXE   for software calculation
*.BGI         The graphics drivers
*.            Example data files for
              simulations
```

Use the version of SIGNAL/S appropriate for your machine.

COMMUNICATIONS SIGNAL SIMULATION

1 INTRODUCTION

Simulation of the signals in a communication system is a powerful tool both to understanding and for analysing the operation of a system. In the simulation a system is set up at the block diagram level, the waveforms in the system generated and observed, and the distortion and bit error rates measured. The SIGNAL programme makes such simulations practical and easy to carry out on a personal computer.

The simulation generates waveforms by stepping the system in time. At each time step a new sample of the waveform is generated and passed through the system. Signals are generated and passed through the system as real baseband signals or as the complex envelopes of bandpass signals, appendix A. The bandwidth of the signals depends on the sampling rate used. For modulated carrier signals the sampling rate has to be high if the carrier waveform is to reproduced. However for most purposes the complex envelope contains all the wanted information and so a simulation of the real and imaginary parts of this complex envelope is all that is necessary. The sampling rate then determines the bandwidth of the complex envelope of the bandpass signal.

A complex signal centred on zero frequency can also represent a real signal. Thus it is possible to use the same system for both lowpass and bandpass signals. In some processes, for example modulation both lowpass and bandpass signals are required. For this purpose real signal operations are included. These pass through the system as complex signals with only a real part, the imaginary part being set to zero.

Included in the system are modles for various communication channels which introduce noise, fading and the effects of multipath propagation. Also included are units for the analysis of the waveforms and for recording the operation of the system.

2 OPERATION

Simulations are carried out in three consecutive stages. Firstly the data about the system is entered. Secondly the simulation of the waveforms is carried out. Thirdly the waveforms are analysed.

Before carrying out a simulation a block diagram of the system to be investigated should be drawn up using the units available in the system. The units are listed in the menus in section 5. In section 6 the information required to set up the individual units is given.

In the simulation the units are connected by linking the inputs and outputs to 'buslines'. An output can only be connected to one busline. Any number of inputs can be taken from a busline. A busline acts as a voltage transfer device.

For convenience the units can be grouped in blocks, for example three blocks representing a transmitter, a channel, and a receiver. The parameters of a block can be stored on a disc file and recalled for later use. The output of one block can be connected to the input of another block. As the system operates by stepping the time sample of the waveform sequentialy through the system the blocks and units must be number in the correct sequence. The data about the units can be entered and changed in at will provided the correct block and unit numbers are maintained.

On setting up a new simulation the sampling rate must be entered. The next question is whether waveforms are to be displayed or error rates determined. If waveforms are to be displayed they are stored for a block of the system. Which block is entered when the run is started. As only a finite number of samples can be stored that number is set by the system to be 256 or 512 depending on the resolution of the display. If error rates are to be estimated the waveforms are not stored. The errors are counted as the simulation is run. In this case any number of samples up to the maximum integer handled by the computer can be used.

The parameters of units are entered in response to requests made by the system. The requests for frequencies, bit rates, and time periods are made in terms of MHz, Mb/s, and microseconds. In the system these values are normalized with respect to the sampling rate. Any consistent set can be used substituting for example KHz, Kb/s and milliseconds. Similarly when entries in Volts are requested mV can be used provided all entries are in mV.

The operation of the system is controlled by a series of menus. The choice from a menu is entered by a single keystroke. If an error is made 'Enter' will return to the previous menu.

The units which have been set up are displayed by d/D. This shows a table for a chosen block of the units in that block, the connections of units to buslines, and the parameters of the units.

Help is available from the menus by using h/H. More help will be available on-line in future issues.

When error rates are being found the occurence of errors is displayed by showing the sample count, the error count, and the probability of error, i.e. the error count divided by the sample count. If required this output can be passed to a printer.

To keep a record of the work being carried out a journal file can be opened from the main menu. The disc file has the filename SIG.JOU. In this file a record of the keystrokes entered is kept. The data in this file can be refered to during the course of a simulation. New versions of this file over write previous versions.

3 ANALYSER

When waveforms have been stored they can be analysed. The waveforms can be displayed, eye diagrams and phase plane plots produced, and the frequency spectra of waveforms generated and displayed.

When waveforms are displayed the signals on any two of the stored buslines can be shown at the same time. The real and imaginary part of each waveform is shown. For real signals the imaginary part is zero. The display is scaled in the horizontal direction so that the number of steps fills the screen. The number of steps is adjusted to be 256 or 512 depending on the resolution of the display.

When eye diagrams are displayed again two signals can be displayed and the real and imaginary part for each signal is shown. The position of the signal in the eye diagram can be shifted by altering the initial delay. This is useful for estimating the signal delay in the system and estimating the best sampling point for bit detection in digital systems.

When phase plane plots are produced again any two signals can be selected for display. The phase angle is the angle between the real and imaginary components of the signal at each sampling interval. This display is useful for showing the effect of noise on a digital signal.

When spectra are to be produced a particular signal must first be selected. Both forward and inverse transforms are available. A FFT procedure within the programme carries out the transformations from the time to the frequency domain or v.v. When the transform has been generated it can be displayed on a linear or dB scaled plot. The frequency transform is centred on the centre frequency of the simulation. The width of the spectrum in terms of frequency depends on the sampling rate. The number of lines in the spectrum and hence the resolution depends on the number of steps for which the waveform is generated. Because of the sampled form of the simulation the appearance of a spectrum depends on the ratio of the frequencies selected to the sampling frequency. To obtain clean spectra integer ratios are required between the various frequencies.

The screen display may be printed by carrying out a screen dump as provided by the computer being used.

4 ERROR COUNTING

When error rates are to be estimated a large number of samples must be passed through the system. This is because at least ten times more samples than the average number of samples between errors must be passed through the system. When samples are passed through the system a delay in the waveform occurs which for filters or a multipath channel may be several bit periods. To detect errors the transmitted signal is used as a reference signal which must be compared with the received signal. The reference signal must be delayed by the same amount as the received signal and a suitable sampling point chosen within the bit. The correct delay and sampling point can be chosen by displaying the waveforms and the eye diagrams.

It is impractical to store the waveforms for a large number of samples so the second option for running the system should be chosen when errors are to be counted. Having set the system up with the correct delay and sampling point a large number of samples can be run through the system. The error detector will show when there is disagreement between the received signal and the reference signal. The bit number at which the error occurs, the total number of errors to that time and the ratio of errors to bits is shown. At the end of the run the total number of errors will be displayed.

5 MENUS

```
      MAIN MENU

1     New simulation
2     Run simulation
3     Change system

4     Directory commands
5     List journal file

A     Analyser

J     Open journal file

F     File block
G     Get block

D     Display units
Q     Exit program

Available memory, memavail, bytes

Key choice
```

```
      DIRECTORY MENU

1     DOS command
2     Current directory
3     Change directory
4     Remove directory
5     Make directory

      Enter for last menu
```

```
      CHANGE MENU

1     Change units
2     Sampling rate
3     Steps in run
4     Blocks and units
5     Change a parameter in a unit

D     Display units
      Enter for last menu
```

SET UP UNITS

0 connectors, blank unit
1 complex signal generators
2 real signal generators
3 modulators, mixers
4 demodulators
5 filters, integrators
6 analogue units
7 digital units
8
9 error counters, display and file units
C channel units

G get block from disc

D Display units
Enter for last menu

0 SET CONNECTIONS

0 blank unit
1 block connector
2 bus connector

D Display units
Enter for last menu

1 COMPLEX SIGNAL GENERATORS

1 centre frequency oscillator
2 offset oscillator
3 voltage controlled oscillator
4 I and Q prn generator
6 bandpass noise generator

D Display units
Enter for last menu

2 REAL SIGNAL GENERATORS

1 dc level
2 sine wave
3 two tones
4 square wave
5 impulse train
6 rectangular pulse train
7 prn sequence
8 Gaussian noise
9 voltage controlled oscillator

D Display units
Enter for last menu

3 MODULATORS

1 analogue modulators and mixers
2 digital modulators

D Display units
Enter for last menu

31 ANALOGUE MODULATORS

1 a.m., full carrier
2 phase
3 frequency
4 mixer, usb
5 mixer, lsb

D Display units
Enter for last menu

32 DIGITAL MODULATORS

1 ook
2 fsk
3 psk
4 qpsk
5 oqpsk
6 msk

D Display units
Enter for last menu

4 DEMODULATORS

1 a.m. envelope detector
2 phase demodulator
3 frequency demodulator

Other demodulators can be made up by a combination of units. They should be filed under appropriate filenames for future use.

D Display units
Enter for last menu

5 FILTERS

1 centred bandpass filter
2 offset bandpass filter
3 real lowpass filter
4 integrator
5 integrator with leak

D Display units
Enter for last menu

6 ANALOGUE UNITS

1 Sum of two signals
2 Difference of two signals
3 Squarer
4 Attenuator
5 Delay
7 I/Q > Two real
8 Two real > I/Q
9 Amplitude and phase

D Display units
Enter for last menu

7 DIGITAL UNITS

1 decision sample and hold
2 sampler
3 multiplexer
4 ideal limiter for real signals
5 ideal magnitude limiter
6 negative edge detector
7 binary adder
8 analogue to digital converter
9 real signal zero crossing detector

D display units
Enter for last menu

9 ERROR COUNTERS

1 bit error counter
2
3 mean, meansquare, and r.m.s. levels
4 write to display
5 write to a file
6 read from a file

D Display units
Enter for last menu

10 CHANNELS

1 Additive white noise
2 Two path
3 Two path with noise
4 Four path
5 Two path with sliding delay
6 Rayleigh fading

D Display units
Enter for last menu

```
    ANALYSER MENU

1   Plot waveforms
2   Eye diagram
3   Phase plane plot
4   Frequency spectra

D   Display units
    Enter for last menu
```

```
    SPECTRUM CONTROLLER MENU

1   Set store for transform
2   Complex forward transform, t>f
3   Complex inverse transform, f>t
4   Plot amplitude, linear scale
5   Plot amplitude, dB scale

D   Display units
    Enter for last menu
```

7 UNITS

Units may have one or two inputs, one or two outputs and up to six parameters which describe the unit. The units which are available are as in the menus for the different types of units. The data required for each unit is shown in the following unit descriptions.

0.0 BLANK UNIT

A blank unit has no input and no output. On first setting up the system the units in the blocks requested are set to be blank units. When changing the units in a simulation which is already set up a blank unit can be inserted in place of an unwanted unit to maintain the sequence when the simulation is run.

0.1 BLOCK CONNECTOR

```
Output in current block to bus
Input from block number
Input from bus
```

Connects the curent block to the output of another block.

0.2 UNIT CONNECTOR

```
Input from busline
Output to busline
```

Links input 1 to output 1. May be useful when changing a system.

1.1 CENTRE FREQUENCY OSCILLATOR

```
Output busline
Amplitude, V
Initial phase, degrees
```

The complex envelope of a signal at the centre has constant levels in the I and Q channels. The levels depend on the amplitude and the initial phase.

1.2 OFFSET FREQUENCY OSCILLATOR

```
Output busline
Amplitude, V
Offset frequency, MHz
Initial phase, degrees
```

The complex envelope of an offset frequency oscillator is sine waves at the offset frequency in the I and Q channels with a phase depending on the initial phase.

1.3 VOLTAGE CONTROLLED OSCILLATOR

```
Input busline, control voltage, V
Output busline, signal
Amplitude, V
Initial offset frequency of VCO, MHz
Control constant, MHz/V
```

The offset frequency of the VCO is controlled by the input control voltage.

1.4 PSEUDO-RANDOM SEQUENCE

```
Output busline
Baud rate of sequence, Mb/s
Generator length
Tap position
Uni-polar or bi-polar output
Initial delay, microsec
```

This unit generates a pseudo-random sequence with a length depending on the generator length set and the tap position. The output in the Q channel is that in the I channel delayed by one bit period. Examples of generator lengths and tap positions for maximal length sequences are given. Uni-polar or bi-polar outputs can be produced. An initial delay can be set so that the start of the sequence can can be delayed from the start of the run. This is useful if a received and delayed sequence is to be compared with an original sequence.

1.6 BANDPASS NOISE GENERATOR

```
Output busline
RMS, V
```

Uses the sum of uniformly distributed random numbers method to generate independent samples in the I and Q channels at the sampling rate. Produces a uniform spectrum over the bandwidth determined by the sampling rate.

2.1 DC LEVEL

```
Output busline
DC level, V
```

A constant level on the I channel only. Zero level on the Q channel.

2.2 SINE WAVE

```
Output busline
Amplitude, V
Frequency, MHz
Initial phase, degrees
```

A sine wave on the I channel. Zero level on the Q channel.

2.3 TWO TONES

```
Output busline
Amplitude 1, V
Frequency 1, MHz
Amplitude 2, V
Frequency 2, MHz
```

A two tone generator of sine waves. Useful when testing for distortion.

2.4 SQUARE WAVE

```
Output busline
Amplitude, V
Fundemental frequency, MHz
Polarity, uni- or bi-polar
Initial delay, microsec
```

Generates a square with the given fundemental frequency. The periodic time is rounded to the nearest integer number of sample intervals. The start of the square wave can be delayed relative to the start of the run. The delay is rounded to the nearest integer number of sample intervals. The output is either a symmetrical bi-polar or a positive uni-polar wave.

2.5 IMPULSE TRAIN

```
Output busline
Amplitude, V
Pulse rate, Mp/s
Initial delay, microsec
```

An impulse is a signal existing for a single sample time. Generates impulses at the rate entered. The impulse interval is rounded to the nearest integer number of samples.

2.6 RECTANGULAR PULSE TRAIN

```
Output busline
Amplitude, V
Pulse rate, Mp/s
Pulse length, microsec
Initial delay, microsec
```

A rectangular pulse train with an adjustable pulse length. The pulse length and the periodic time are rounded to the nearest integer number of samples.

2.7 PSEUDO-RANDOM SEQUENCE

```
Output busline
Bit rate of sequence, Mb/s
```

```
Generator length, M bits
Tap number
Polarity, uni- or bi-polar
Initial delay, microsec
```

Generates a pseudo-random sequence. The bit period is rounded to the nearest integer number of samples. Examples of generator lengths and tap positions for maximal length sequences are given.

2.8 GAUSSIAN NOISE

```
Output busline
RMS amplitude, V
```

Generates independent samples of noise with a Gaussian amplitude probability distribution. Uses a sum of random numbers algorithm.

2.9 VOLTAGE CONTROLLED OSCILLATOR

```
Input busline, control voltage, V
Output busline, signal
Amplitude, V
Initial frequency of VCO, MHz
Control constant, MHz/V
```

Generates a real signal. The output depends on the initial frequency of the v.c.o. and the elapsed time plus the integrated phase change due to the variations in frequency which follow the control voltage.

3.1.1 AM FULL CARRIER MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
```

Generates a full carrier a.m. signal. The amplitude of the modulating signal should be less than the carrier amplitude to prevent over modulation.

3.1.2 PHASE MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
Phase deviation, degrees/V
```

Generates a phase modulated wave. The phase of the carrier is proportional to the input modulating voltage.

3.1.3 FREQUENCY MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
Peak frequency deviation, MHz
```

```
Maximum amplitude of the modulating signal, V
```

Generates a frequency modulated wave. The instantaneous frequency is controlled by the modulating signal.

3.1.4 MIXER, USB

```
Input 1 busline
Input 2 busline
Output busline
```

Gives the upper sideband complex output of the product of the two complex input signals

3.1.5 MIXER, LSB

```
Input 1 busline
Input 2 busline
Output busline
```

Gives the lower sideband complex output of the product of the two complex input signals

3.2.1 OOK MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
```

The complex output signal has the carrier amplitude if the input modulation is greater than zero, else the output is zero.

3.2.2 FSK MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
Deviation from carrier, MHz
```

Produces a fsk modulated carrier wave. The modulation will have either a non-coherent phase or a coherent phase characteristic depending on the ratios of bit rate to deviation chosen.

3.2.3 PSK MODULATOR

```
Input busline for modulation
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
Phase shift for (1), degrees
Phase shift for (0) or (-1), degrees
```

The phase of the output carrier wave is shifted by the angles entered.

3.2.4 QPSK MODULATOR

```
Input busline, I and Q inputs
```

```
Output busline
Carrier amplitude, V
Carrier offset frequency, MHz
Initial phase of carrier, degrees
```

The input modulating signal is two independent components on the I and Q channels of the input busline. The two components must have the same bit rate. The input signals produce a modulation of the in-phase and quadrature components of the carrier wave.

3.2.5 OQPSK MODULATOR

```
Not yet programmed
```

An o.q.p.s.k. modulator can be made up from a q.p.s.k. modulator fed by an input signal for which the I and Q channels have a half bit time displacement.

3.2.6 MSK MODULATOR

```
Input busline for modulation, real data
Output busline
Bit rate, Mb/s
Carrier amplitude, V
Carrier offset frequency, MHz
```

Produces an m.s.k. signal. The input busline is the real data signal.

4.1 AMPLITUDE DEMODULATOR ENVELOPE DETECTOR

```
Input busline
Output busline
```

This produces an output which is the amplitude of the complex envelope including the d.c. level depending on the constant carrier component.

4.2 PHASE DEMODULATOR

```
Input busline
Output busline
```

Gives an output proportional to the phase of the complex envelope. The phase is calculated mod 2π. This may cause problems in interpreting the output if the phase deviation produced by a modulator is more than 2π.

4.3 FREQUENCY DEMODULATOR

```
Input busline
Output busline
Initial phase of carrier, degrees
Peak frequency deviation, MHz
```

The instantaneous frequency is found as the slope of the phase difference between succesive samples. There may be difficulties in interpreting the output when sucesive phase angles jump by 2π.

5.1 CENTRED BANDPASS FILTER

```
Input busline
Output busline
Filter type, Butterworth or Chebyschev
Cut-off frequency, MHz
Stop-band frequency, MHz
Attenuation at stop-band frequency, dB
Ripple amplitude, %, for Chebyshev filters
```

This unit is for a filter acting on the I and Q channel signals. It has a symmetrical response about the centre frequency. The order of the filter is calculated for an even order filtered. The order is displayed. The filter coefficients are calculated for the 2nd order sections of the filter.

5.2 OFFSET BANDPASS FILTER

```
Input busline
Output busline
Filter offset frequency, MHz
Filter type, Butterworth or Chebyschev
Cut-off frequency, MHz
Stop-band frequency, MHz
Attenuation at stop-band frequency, dB
Ripple amplitude, %, for Chebyshev filters
```

This filter has the same response as the centre frequency filter but is centred on the offset frequency.

5.3 REAL LOWPASS FILTER

```
Input busline
Output busline
Filter type, Butterworth or Chebyschev
Cut-off frequency, MHz
Stop-band frequency, MHz
Attenuation at stop-band frequency, dB
Ripple amplitude, %, for Chebyshev filters
```

This filter has the same response as the centre frequency filter but acts on the real component of the signal only. Runs faster than the centre frequency filter.

5.4 INTEGRATOR

```
Input busline
Output busline
```

The integrator sums succesive samples of a real signal.

5.5 INTEGRATOR WITH LEAK

```
Input busline
Output busline
Leak factor < 1
```

The integrator with leak sums the current sample value with a fraction of the previous sum. The fraction depends on the leak factor.

6.1 SUM OF TWO SIGNALS

```
Input 1 busline
```

```
Input 2 busline
Output busline
```

Sums the I and Q components of the two complex input signals. The output is a complex signal.

6.2 DIFFERENCE OF TWO SIGNALS

```
Input 1 busline
Input 2 busline
Output busline
```

Takes the diference of the two I inputs and the two Q inputs. Input 2 is subtracted from input 1. Outputs the complex difference.

6.3 SQUARER

```
Input busline
Output busline for double frequency component
Output busline for the zero frequency component
```

Gives the square of the input complex signal. The double frequency component is a complex signal. The zero frequency component is a real signal.

6.4 ATTENUATOR

```
Input busline
Output busline
Attenuation, dB
```

Attenuates the complex input signal by the ratio entered.

6.5 DELAY

```
Input busline
Output busline
Delay, microsec
```

Delays the complex input signal. The delay is rounded to the nearest integer number of sample intervals.

6.7 I/Q TO TWO REAL

```
Input busline
Output 1 busline
Output 2 busline
```

Separates the two components of the complex input signal into two real signal components.

6.8 TWO REAL TO I/Q

```
Input 1 busline
Input 2 busline
Output busline
```

Combines the two real input signals to one complex output. Input 1 becomes the real component of the output, input 2 the imaginary part of the output.

6.9 AMPLITUDE AND PHASE

```
Input busline, complex
```

```
Output 1 busline, amplitude
Output 2 busline, phase
```

Forms the amplitude and phase of the complex input signal. Output 1 is the amplitude. Output 2 is the phase mod 2π.

7.1 DECISION SAMPLE AND HOLD

```
Input busline
Output busline
Decision level, V
Data bit rate, Mb/s
High output state, V
Low ouput state, V
Delay, microsec
```

The input signal is a real signal. The output is a real signal at a level equal to the high output state if the input is above the decision level else it is at the low output state. The bit period is truncated to the lower number of sample periods. The delay is rounded to the nearest integer number of sample intervals.

7.2 SAMPLER

```
Input busline
Output busline
Data bit rate, Mb/s
Initial delay, microsec
Sampling interval, microsec
```

The input is a real signal. The output is a real signal at a level equal to the input level at the sampling time. The bit period is truncated to the lower number of sample periods. The initial delay and the sampling interval are rounded to the nearest integer number of sample intervals.

7.3 MULTIPLEXER

```
Input 1 busline
Input 2 busline
Output busline
Input bit rate, Mb/s
Initial delay, microsec
```

Time division multiplexes the two real input signals into a single real output signal. The two input signals should have the same input bit rate and should be synchronised. The output bit rate is double the input bit rate. The output bit rate is half the input bit rate truncated to the lower integer number of sample intervals. The initial delay is rounded to the nearest integer number of sampling intervals.

7.5 IDEAL MAGNITUDE LIMITER

```
Input busline
Output busline
Limit level, V
```

The input is a complex signal. The phase of the input the input signal is calculated. The output is a complex signal with an amplitude equal to the limit level and a phase equal to the input signal phase. The zero crosing points are preserved. If the input is real, i.e. zero imaginary component, the phase is $\pm\pi$ 2 and the output is a real signal at the limit levels.

7.7 BINARY ADDER

```
Input 1 busline
Input 2 busline
Output busline
```

The binary adder operates on bi-polar real signals. It produces a bi-polar binary sum as the real output signal.

7.8 ANALOGUE TO DIGITAL CONVERTOR

```
Input busline
Output busline
Maximum amplitude of input signal, V
Conversion sampling rate, Mb/s
Number of levels, 2^M, enter M
Initial delay in sampling, microsec
```

Converts the real input signal into one of 16 different integer levels at the sampling intervals.

7.9 ZERO CROSSING DETECTOR

```
Input busline
Output busline
```

Produces an impulse when the input real signal waveform passes through zero.

9.1 BIT ERROR COUNTER

```
Input 1 busline, reference sequence
Input 2 busline, signal
Output busline
Message baud rate, Mb/s
Sampling point delay, microsec
```

Compares the sign of the input real signal with that of the reference sequence at the sampling point within the bit. The correct sampling point must be found by observing the delay in the eye of the signal. If an error occurs an error counter is incremented. The level on the I component of the output is the error count. The bit number at which an error occurs, the error count, and the error/bit ratio is displayed on the screen.

9.3 MEAN, MEAN SQUARE, AND RMS LEVELS

```
Input busline
```

Sums the sample and the square of the sample values and displays on the screen at the end of the run the mean, mean square, and r.m.s. levels. These values are written to the journal file if that file is open.

9.4 ON-LINE DISPLAY

```
Input busline
Choose display type
Y scale factor
```

Displays one parameter of a complex input signal on the screen. For a complex signal the parameters which may be displayed are the real part, the imaginary part, the magnitude, and the phase.

The phase is displayed mod 2π.

The Y scale factor adjusts the Y scale on the display. The length of the display is adjusted so that the number of steps requested fills the screen.

9.5 WRITE TO A FILE

```
Input busline
Filename
```

Writes the input complex data to a named text file.

9.6 READ FROM A FILE

```
Output busline
Filename
```

Reads the complex data from a named text file. Outputs the data to a busline.

10.1 ADDITIVE WHITE NOISE CHANEL

```
Input busline
Output busline
Noise level, RMS V
```

Adds independent Gaussianly distributed noise samples to the I and Q channels. The satandard deviation of the Gaussian distributions depends on the r.m.s. noise level.

10.2 TWO PATH CHANNEL

```
Input busline
Output busline
Second path delay, microsec
Second path relative amplitude, dB
```

The output is the sum of the input and a delayed and attenuated version of the input signal.

10.3 TWO PATH CHANNEL WITH ADDED NOISE

```
Input busline
Output busline
Second path delay, microsec
Second path relative amplitude, dB
Noise level, RMS V
```

Combines the effects of the previous two models.

10.4 FOUR PATH CHANNEL

```
Input busline
Output busline
Second path delay, microsec
Second path relative amplitude, dB
Third path delay, microsec
Third path relative amplitude, dB
Fourth path delay, microsec
Fourth path relative amplitude, dB
```

The data for the paths should be entered with the paths in the order of increasing delay.

10.5 TWO PATH CHANNEL WITH SLIDING DELAY

```
Input busline
Output busline
Second path initial delay, microsec
Second path final delay, microsec
First path attenuation, dB
Second path attenuation, dB
```

The delay is incremented from the initial value at the start of the run to the final value at the end of the run. Shows the bandwidth expansion or contraction due to Doppler.

10.6 RAYLEIGH FADING

```
Input busline
Output busline
Correlation time, microsec
Standard deviation
```

The output amplitude fades during the run with a Rayleigh amplitude probability distribution. Shows the spectral broadening due to fading. The fading rate depends on the correletion time.