SIGNAL RECOVERY

The Digital Lock-in Amplifier

TECHNICAL NOTE
TN 1003

In recent years the falling cost of digital signal processing circuitry has allowed its application in an ever wider field of instrumentation. In particular, it has become possible to use the technique in applications that hitherto have been wholly the preserve of analog electronics, such as the phase-sensitive detectors found in lock-in amplifiers.

This Technical Note describes the technology used in the **SIGNAL RECOVERY** digital lock-in amplifiers, such as the models 7220, 7220BFP, 7225, 7225BFP, 7260, 7265, 7280 and 7280BFP. Technical Note TN1002 provides similar information for instruments using analog technology.

Introduction

Lock-in amplifiers are usually only described as "digital" or DSP (Digital Signal Processing) instruments if their phase-sensitive detector is implemented in digital circuitry, even though traditional analog instruments have, for many years, used digital electronics extensively for instrument control and output processing.

Digital lock-in amplifiers have become very popular because in many cases they offer a number of advantages over analog units, including:-

- Better output stability. Since, unlike analog units, there are no DC-coupled output amplifier stages and therefore outputs are less prone to drift with time and temperature.
- Better internal oscillators. The crystal-stabilized internal oscillator is more stable with respect to time and temperature changes and hence gives better results in experiments which can be driven by it. Internal reference mode operation in DSP units also offers a very small, or even zero, lock acquisition time which is ideal for swept-frequency measurements.
- Perfect X and Y demodulator orthogonality. This improves the accuracy of measurements of weak in-phase components in the presence of strong quadrature signals.
- Better price/performance ratios. The reduced manufacturing and testing cost of DSP units delivers instruments with better price/performance ratios than the older technologies.

In spite of the above, there are still some applications for which analog techniques are better or indeed the only option. These include operating at higher frequencies, or when utilizing short time constants at mid-range frequencies (50 - 250 kHz). Requirements such as these are often encountered in dual demodulation experiments or when operating the lock-in amplifier as part of a control feedback loop. It should also be remembered that the input stages of a DSP lock-in amplifier still need to be implemented in analog technology, so that in reality the "all digital" instrument does not exist.

Instrument Description

Figure 1 shows the functional block diagram of a typical high-performance digital lock-in amplifier, such as those in the **SIGNAL RECOVERY** 72XX series.

Signal Channel

The input stage may be operated in one of three modes:-

Single-ended Voltage Mode

The signal to be measured is applied to one input connector which operates in single-ended mode and directly feeds the voltage input amplifier.

Differential Voltage Mode

Two input connectors are active and the instrument measures the difference in applied voltage between them.

Current Input Mode

A single connector is active which feeds a current-to-voltage converter, the output of which then drives the voltage input amplifier. The current conversion ratio is usually either 10^6 V/A (high bandwidth) or 10^8 V/A (low noise) although the user does not usually need to concern himself with the actual value used since when operated in the current mode the instrument directly displays the measured signals in terms of amperes.

In the current measuring mode, the input impedance is low (typically less than 100 Ω) although it does rise as the operating frequency increases, and is higher for the 10^8 V/A conversion setting than for the 10^6 V/A. If the very best performance is needed then it may be better to use a separate dedicated current preamplifier.

The current input connector is normally combined with the "B" voltage-mode input connector to simplify the layout of the front panel.

Line Frequency Notch Filter

The output of the input amplifier is optionally passed through a line notch filter. This is a band-rejection stage, designed to remove 50/60 Hz and/or 100/120 Hz interference from the input signal. Since the line frequency can vary by up to ±1% of its nominal value the filter has a low *Q*-factor to ensure satisfactory attenuation over the range of possible input frequencies. However, this does have the disadvantage of introducing significant attenuation and phase-shifts even at frequencies well removed from the set frequency.

AC Gain

The signal channel contains a number of analog filters and amplifiers whose overall gain is defined by the AC Gain function block. For each setting of AC Gain there is a corresponding level at which the instrument input will overload.

It is a basic property of the digital lock-in amplifier that the best demodulator performance is obtained by presenting as large a signal as possible to the main analog-to-digital converter (ADC). Therefore, in principle, the AC Gain setting is made as large as possible without causing amplifier or converter overload. This constraint is not too critical however and the use of a value 10 or 20 dB below the optimum makes little difference. Note that as the AC Gain value is changed, the demodulator (in-phase and quadrature multiplier) gain is also adjusted in order to maintain the selected full-scale sensitivity.

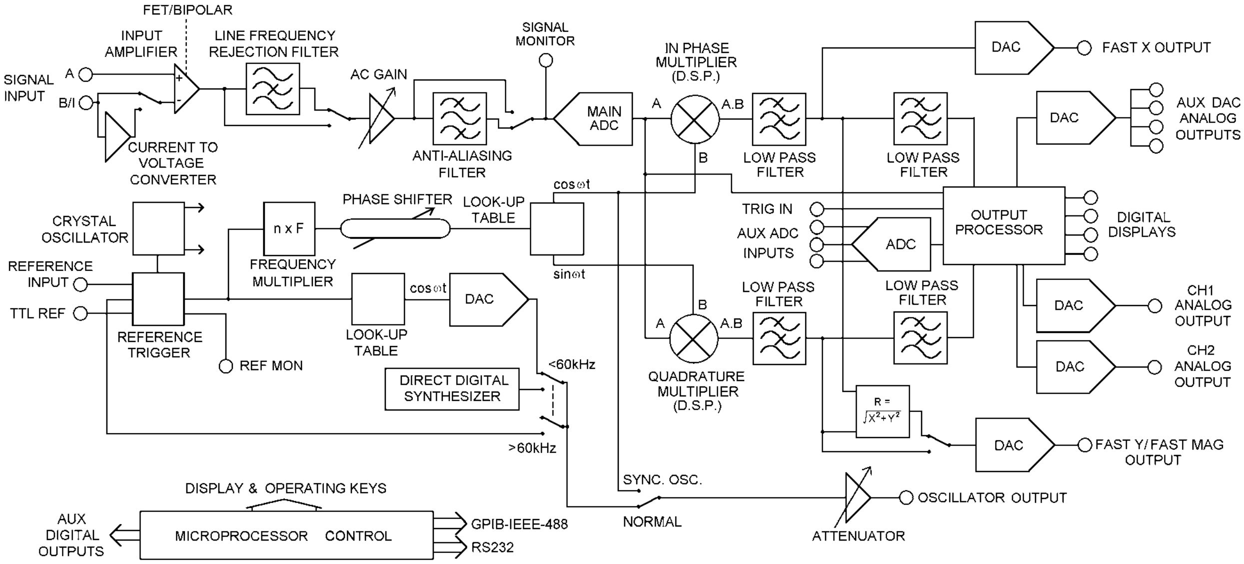

Fig 1, Block Diagram of Typical DSP Lock-in Amplifier

The instrument's full-scale sensitivity is set by a combination of analog AC Gain in the input circuits and digital output gain in the demodulator. Changes in AC Gain potentially affect other specifications, such as bandwidth and accuracy, but changes in digital gain have no such effects. Hence changes in full-scale sensitivity effected only by changes to the digital gain are free from these additional errors.

The AC Gain setting affects the instruments' dynamic reserve. This is a measure of its ability to make accurate measurements in the presence of interfering signals. If the AC gain is low but the full-scale sensitivity is high then the real signal will only occupy a few bits of the ADC's dynamic range, leaving plenty of headroom for stronger interfering signals without overload. However longer output time constant settings will be needed for a given output accuracy. Conversely, high AC Gain allows shorter time constants to be used for a given accuracy but results in lower dynamic reserve.

Anti-Aliasing Filter

Following the AC Gain amplifier stage, the signal passes to the anti-aliasing filter. This removes unwanted frequencies, which would cause a spurious output from the ADC due to the nature of the sampling process, by restricting the signal bandwidth reaching the ADC. If the instrument is being used to measure noise-free signals, as in the case of vector voltmeter measurements, then aliasing may not occur and slightly better performance can be achieved by bypassing the anti-aliasing filter.

A buffered version of the analog signal just prior to the main ADC is often made available at a connector on the rear panel of the instrument. This output can be viewed on an oscilloscope to monitor the effect of the signal channel filters and amplifiers.

Main Analog-to-Digital Converter

Following the anti-alias filter the signal passes to the main analog-to-digital converter which digitizes the input signal at the *sampling rate*. The output from this converter, which is a series of digital values representing the amplitude of the input signal, feeds the first of the digital signal processors. This implements the in-phase and quadrature digital multipliers, and the first stage of the output low-pass filtering for each of the X and Y channels.

In order to satisfy Nyquist's sampling theorem, the sampling rate generally needs to be at least twice the bandwidth of the anti-aliasing filter. If this bandwidth starts at DC, i.e. the filter is of a low-pass design, then this equates to a requirement that the sampling rate be at least twice the maximum reference frequency. However, the sampling rate can be reduced if the anti-aliasing filter can be operated in bandpass mode as in ***SIGNAL RECOVERY*** DSP lock-in amplifiers.

Reference Channel

The reference channel serves to provide the demodulator DSP with a stream of "phase values" at the sampling frequency of the main ADC. Each of these values represents the instantaneous phase angle of the reference frequency waveform at the sampling time.

For example, consider a 1 kHz reference frequency, either internally generated or from the reference input connector, with the unit set to fundamental reference mode (n = 1). If the main ADC sampling rate is 180 kHz then there will be 180 samples of the applied input signal for each reference period, which is one millisecond. Each one of these samples needs to be multiplied by the value of a cosine wave of unit amplitude at the corresponding phase position. Consequently, the reference DSP measures the reference frequency (1 kHz) and outputs 180 phase values during each reference period, which in this case would be at phase angles 0 to 360° in 2° (360/180) increments.

In external reference mode a second DSP, operating as a digital phase-locked loop (PLL), is used to measure the period (or frequency) of the signal applied to the TTL or analog reference inputs very accurately and generate the stream of phase values.

In internal reference mode the ideal situation would be to allow the reference processor to generate both the amplifier's internal oscillator output signal and the phase values for the demodulator at the selected reference frequency. In this case the reference channel is not dependent on a PLL, unlike the situation with analog lock-in amplifiers. Consequently the phase noise is extremely low and, because no time is required for a PLL to acquire lock, reference channel lock is immediate. However, technical limitations mean this is not yet possible at all reference frequencies, so that at the higher end of the frequency range the reference trigger input is provided by an internal link from the output of a separate direct digital synthesizer and the reference channel then operates as if in external reference mode.

The reference channel DSP is also utilized for implementing reference frequency multiplication, as is required for measurements made on the harmonics of the reference frequency. Normally, a lock-in amplifier measures the applied signal at the reference frequency but in some applications, such as Auger Spectroscopy or amplifier characterization, it is useful to be able to make measurements at some multiple "*n*", or harmonic, of the reference frequency "F". Digital lock-in amplifiers allow this multiple to be set to any value between 2 (i.e. the second harmonic) and 65535 (depending on which model is being used), as well as unity, which is the normal mode. The only restriction is that the product "$n \times$ F" cannot exceed the maximum normal reference frequency.

Phase-Shifter

The reference DSP also implements a digital reference phase-shifter, which adds or subtracts the set reference phase angle from the phase values being sent to the demodulator DSP. By adjusting this reference phase control, which has a typical phase-shift resolution of ten millidegrees, the signals at the reference and signal channel inputs to the X-channel demodulator can be brought into phase. The phase-shifter also incorporates the facility to add 90° steps, which is especially useful when the phase is adjusted by first looking for a null at the X-channel output and then adding 90° or 270° to maximize it.

SIGNAL RECOVERY lock-in amplifiers provide a TTL logic signal, at the reference frequency, at the REF MON connector, which allows the user to check that the reference channel is operating correctly.

Demodulator DSP

The main DSP takes each phase value from the reference DSP and uses it to find the amplitude of a cosine wave at the corresponding angle by means of a look-up table. It then takes this value and digitally multiplies it by the signal sample, the resulting number being the X-channel output sample. A second part of the DSP carries out a similar calculation, but uses the value of a sinewave at the same angle. This gives the Y-channel output sample.

First Stage Output Filters

The output from the multiplication process is a stream of digital X and Y channel output samples, at the sampling rate. These feed the first stage of the X channel and Y channel digital output filters, which implement the conventional output low-pass filter function of the lock-in amplifier. The filtered outputs are fed into the output processor which carries out further filtering and processing. In ***SIGNAL RECOVERY*** models these outputs are also used to drive two fast 16-bit digital-to-analog converters (DACs) to generate the signals at the instrument's FASTX and FASTY analog output connectors.

The demodulator output is digitally scaled to provide the demodulator gain control. As discussed earlier this gain is adjusted as the AC Gain is varied to maintain the overall full-scale sensitivity.

Output Processor

Although shown on the block diagram as a separate entity, the output processor is typically part of the instrument's main microprocessor. It provides more digital filtering of the X channel and Y channel signals, as well as carrying out vector magnitude and phase, noise, ratio and other calculations.

Second Stage Output Filters

Generally, digital lock-in amplifiers use Finite Impulse Response (FIR) low-pass filters offering 6, 12, 18 and 24 dB/octave roll-off with increasing frequency. These filters offer a substantial advantage in response time compared with analog filters or digital infinite impulse response (IIR) filters.

The 6 dB/octave filters are not satisfactory for most purposes because they do not give good rejection of non-random interfering signals, which can cause aliasing problems as a result of the sampling process in the main ADC. However, the 6 dB/octave filter finds use where the lock-in amplifier is incorporated in a feedback control loop, and in some situations where the form of the time-domain response is critical. Normally, the 12 dB/octave setting is used unless there is some definite reason for not doing so.

Following the output filters, an output offset facility is provided to enable an offset (up to ±300 % full-scale in ***SIGNAL RECOVERY*** units) to be applied to the X, Y or both outputs.

Magnitude and Phase Outputs

The processor calculates the vector magnitude and signal phase of the input signal. If the input signal, $V_s(t)$, is a sinusoid of constant amplitude at the reference frequency and the output filters are set to a sufficiently long time constant, then the filtered demodulator outputs are constant levels V_x and V_y. The vector magnitude $\sqrt{(V_x^2+V_y^2)}$ is dependent only on the amplitude of the required signal, $V_s(t)$ (i.e. it is not dependent on the phase of $V_s(t)$ with respect to the reference input) and is computed by the output processor in the lock-in amplifier. The phase angle between $V_s(t)$ and the applied reference signal is called the signal phase, it is the arctangent of the ratio V_y/V_x and is also computed by the processor.

Noise Calculation

The noise measurement facility uses the output processor to perform a noise computation on the Y output where it is assumed that the amplitude distribution of the waveform at this point is Gaussian with a mean value of zero. The zero mean is usually obtained by using the reference phase control or the Auto-Phase function with a comparatively long time constant (say 1 s). The time constant is then reduced (to say 10 ms) for the noise measurement.

The instrument takes into account the equivalent noise bandwidth of the measurement, which is set by the output time constant and slope, and displays the noise value directly in V/√Hz or A/√Hz either on the digital panel meters or via the computer interfaces.

Analog Outputs

Any of the outputs which are digitally displayed are also available in analog form at connectors on the rear panel. These analog outputs are generated by using two further 16-bit digital-to-analog converters (DACs).

Internal Oscillator

As mentioned earlier, when used in internal reference mode, the reference DSP generates phase values, representing the required reference frequency, to drive the main DSP. The sinusoidal values from the look-up table are applied to a fast DAC and the resulting analog signal, after filtering, is the unit's "internal oscillator" output. The great advantage of this technique is that lock acquisition is instantaneous, since no time is needed for a reference channel PLL to acquire lock.

However, as the oscillator frequency is increased towards the instrument's sampling frequency, the stepped approximation to a sinewave, that is generated by this technique, becomes more obvious. To overcome this effect a dedicated direct digital synthesizer running at a much higher sampling frequency is used at higher frequencies to generate the oscillator signal. This is then internally coupled back to the "external" reference input of the lock-in amplifier when internal reference mode is selected.

Microprocessor Control

In addition to the digital electronics used for the phase-sensitive detector, the digital lock-in amplifier incorporates a microprocessor which, as has been seen, is used to implement a number of signal processing functions. It also allows the instrument to be operated via a standard computer interface such as a GPIB (IEEE-488) or RS232 link.

The command set used in ***SIGNAL RECOVERY*** units is based on simple mnemonics which generally operate either to set or interrogate instrument settings depending on whether an optional parameter is sent. As an example, to set the full-scale sensitivity of the model 7260 or 7265 to the 20 nV setting requires the user to send the following ASCII character string:- SEN 4 <CR>

where <CR>, the *terminator*, represents the carriage return character. To read the instrument's present sensitivity setting, the user would send:-

SEN <CR>and the instrument would respond with:-

4 <CR>

Ease of use is further improved over earlier analog instruments by offering the "floating point" mode for reading instrument outputs. Conventionally, for example, determining the true level of the in-phase component of an input signal in volts rms has required the programmer to know the present sensitivity setting and then to send a command to the instrument to read the output value. The response is sent as a percentage of the present full-scale sensitivity setting, and so a look-up table is needed to convert this response and the sensitivity setting into a value expressed in volts.

The floating point response mode causes the instrument to carry out this translation, although the older technique is retained for compatibility.

Other Features

SIGNAL RECOVERY digital lock-in amplifiers offer many features beyond those strictly needed for signal recovery but which nonetheless make the experimenter's job easier. These include the following:-

Auxiliary ADC, DAC and Digital Outputs

Many computer-controlled experiments require one or two analog or digital control signals and may generate analog signals which need recording at the same time as the output from the lock-in amplifier. The instruments therefore include analog and digital auxiliary inputs and outputs taking advantage of the presence of the ADCs, DACs and computer interface electronics already in the instrument and often avoiding the need to use separate units.

Data Storage

A 32k, 16-bit data buffer allows selected instrument outputs to be stored prior to transfer to an external computer, or displayed on the graphical display panel where available.

Transient Recorder

The auxiliary ADC inputs may be used in conjunction with the curve storage capability to allow the units to function as 16-bit, 40 kSa/s transient recorders.

Frequency Measurement

The internal crystal clock and the microprocessor allow the units to measure the present reference frequency very accurately and thereby offer the capability of acting as a frequency meter over their operating frequency range.

Unique Features

The ***SIGNAL RECOVERY*** DSP lock-in amplifiers provide the following three modes of operation, believed to be unique.

Virtual Reference Mode™

In the virtual reference mode, the Y channel output is used to make continuous adjustments to the internal oscillator frequency and phase to achieve phase-lock with the applied signal, such that the X channel output is maximized and the Y channel output zeroed. This mode of operation allows signal recovery measurements to be made without the use of a reference signal.

Dual Reference Mode

The dual reference mode allows the instrument to make simultaneous measurements at two different reference frequencies, an ability that previously required two lock-in amplifiers. This flexibility imposes a few restrictions, such as the requirement that one of the reference signals be external and the other is derived from the internal oscillator, a limit on the maximum operating frequency and that both signals be passed through the same input signal channel. This last restriction implies either that both signals are derived from the same detector (for example two chopped light beams falling on a single photodiode) or that they can be summed prior to measurement, either externally or by using the differential input mode of the instrument. Nevertheless, the mode proves invaluable in many experiments.

Dual Harmonic Mode

Dual harmonic mode allows the simultaneous measurement of two different harmonics of the input signal. As with dual reference mode, there are a few restrictions, such as a limit on the maximum operating frequency.

Further Information

This Technical Note is intended to describe the overall structure of a modern digital lock-in amplifier. Additional information may be found in further ***SIGNAL RECOVERY*** publications, which may be downloaded from our website at **www.signalrecovery.com**

TN 1000	What is a Lock-in Amplifier?
TN 1001	Specifying Lock-in Amplifiers
TN 1002	The Analog Lock-in Amplifier
TN 1004	How to Use Noise Figure Contours
AN 1000	Dual-Channel Absorption Measurement with Source Intensity Compensation
TN 1005	What is a Boxcar Averager?
TN 1006	Boxcar Averager Specification Comparison
TN 1007	The Incredible Story of Dr D.P. Freeze
AN 1001	Input Offset Reduction using the Model 7265/7260/7225/7220 Synchronous Oscillator/Demodulator Monitor Output
AN 1002	Using the Model 7225 and 7265 Lock-in Amplifiers with software written for the SR830
AN 1003	Low Level Optical Detection using Lock-in Amplifier Techniques
AN 1004	Multiplexed Measurements using the 7225, 7265 and 7280 Lock-in Amplifier
AN 1005	Dual Beam Ratiometric Measurements using the Model 198A Mixed Beam Light Chopper

SIGNAL RECOVERY

SIGNAL RECOVERY is part of **AMETEK** Advanced Measurement Technology, Inc

801 SOUTH ILLINOIS AVENUE
OAK RIDGE
TN 37831-2011
USA
Phone: +1 865 482 4411
Fax: +1 865 483 0396

SPECTRUM HOUSE
1 MILLARS BUSINESS CENTRE, FISHPONDS CLOSE
WOKINGHAM, BERKS RG41 2TZ
UNITED KINGDOM
Phone: +44 (0)118 936 1210
Fax: +44 (0)118 936 1211

E-mail: info@signalrecovery.com **Web Site: www.signalrecovery.com**