SECTION I

INTRODUCTION TO MIXED SIGNAL PROCESSING OF REAL-WORLD SIGNALS AND SIGNAL CONDITIONING

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- ORIGINS OF REAL-WORLD SIGNALS AND THEIR UNITS OF MEASUREMENT
- REASONS FOR PROCESSING REAL-WORLD SIGNALS
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SECTION I

1

INTRODUCTION TO MIXED SIGNAL PROCESSING OF REAL-WORLD SIGNALS AND SIGNAL CONDITIONING

ORIGINS OF REAL-WORLD SIGNALS AND THEIR UNITS OF MEASUREMENT

In this seminar, we will primarily be dealing with the processing of *real-world* signals using both analog and digital techniques. Before starting, however, let's look at a few key concepts and definitions required to lay the groundwork for things to come.

Webster's *New Collegiate Dictionary* defines a *signal* as " A detectable (or measurable) physical quantity or impulse (as voltage, current, or magnetic field strength) by which messages or information can be transmitted." Key to this definition are the words: *detectable, physical quantity, and information.*

FUNDAMENTAL CHARACTERISTICS OF SIGNALS

- **Are Physical Quantities**
- **Are Measurable**
- **Contain Information**
- **All Signals Are Analog**
- **Some Signals Are Responses to Known Signals Which Act as a Stimulus (i.e. Radar and Ultrasound Return Signals)**

Figure 1.1

By their very nature, signals are analog , whether dc, ac, digital levels, or pulses. It is customary, however, to differentiate between *analog* and *digital* signals in the following manner: Analog (or real-world) variables in nature include all measurable physical quantities. In this seminar, *analog* signals are generally limited to electrical variables, their rates of change, and their associated energy or power levels. Sensors and transducers are used to convert other physical quantities (temperature, pressure, etc.) to electrical signals and vice versa. The entire subject of signal conditioning deals with preparing real-world signals for processing and includes such topics as sensors (temperature and pressure, for example), isolation and instrumentation amplifiers, etc.

Some signals result in response to other signals. A good example is a radar or ultrasound imaging return signal , both of which result from a known transmitted signal.

UNITS OF MEASUREMENT

- Temperature: °C
- Pressure: Newtons/m^2
- Mass: kg
- Voltage: Volts
- Current: Amps
- Power: Watts

Figure 1.2

On the other hand, there is another classification of signals, called *digital*, where the actual signal has been conditioned and formatted into a digit. These digital signals may or may not be related to real-world analog variables. In the specific case of Digital Signal Processing (DSP), the analog signal is converted into binary form by a device known as an analog-to-digital converter (ADC). The output of the ADC is a binary representation of the analog signal and is manipulated arithmetically by the Digital Signal Processor. After processing, the information obtained from the signal may be converted back into analog form using a digital-to-analog converter (DAC).

Another key concept embodied in the definition of *signal* is that there is some kind of *information* contained in the signal. This leads us to the key reason for processing real-world analog signals: the *extraction of information.*

REASONS FOR PROCESSING REAL-WORLD SIGNALS

The primary reason for processing real-world signals is to extract information from them. This information normally exists in the form of signal amplitude (absolute or relative), frequency or spectral content, or timing relationships with respect to other signals. Once the desired information is extracted from the signal, it may be used in a number of ways.

In some cases, it may be desirable to reformat the information contained in a signal. This would be the case in the transmission of a voice signal over a frequency division multiplexed (FDM) telephone system. In this case, analog techniques are used to "stack" voice channels in the frequency spectrum for transmission via microwave relay or coaxial cable. In the case of a digital transmission link, the analog voice information is first converted into digital using an ADC. The digital information representing the individual voice channels is multiplexed in time (time division multiplexed, or TDM) and transmitted over a serial digital transmission link (as in the T-Carrier system).

Another requirement for signal processing is to *compress* the frequency content of the signal (without losing significant information) then format and transmit the information at lower data rates, thereby achieving large reductions in required channel bandwidths. High speed modems and adaptive pulse code modulation systems (ADPCM) make extensive use of data reduction algorithms, as do digital mobile radio systems and High Definition Television (HDTV).

Industrial data acquisition and control systems make use of information extracted from sensors to develop appropriate feedback signals which in turn control the process itself. A block diagram of such a system is shown in Figure 1.3. Note that these systems require both ADCs and DACs as well as sensors, signal conditioners, and the DSP.

In some cases, the signal containing the information is buried in noise, and the primary objective of signal processing is recovery. Techniques such as filtering, autocorrelation, convolution, etc. are often used to accomplish this task in both the analog and digital domains.

TYPICAL DATA ACQUISITION AND PROCESS CONTROL SYSTEM

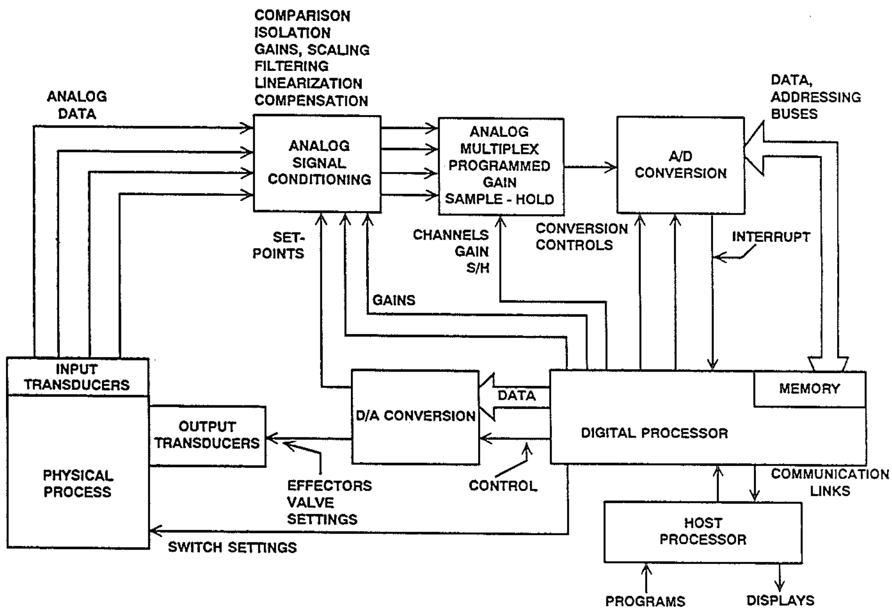


Figure 1.3

REASONS FOR PROCESSING REAL-WORLD SIGNALS

- Extract Information (Amplitude, Frequency, Spectral Content, Timing Relationships
- Reformat the Signal (FDM,TDM Systems)
- Compression of Data (Modems, Digital Mobile Radio, ADPCM, HDTV)
- Generate Feedback Control Signal (Industrial Process Control)
- Extract Signal from Noise (Filtering, Autocorrelation,Convolution)
- Store Signal Data in Digital Format for Recovery and/or Analysis using DSP Techniques (FFT Analysis)

Figure 1.4

GENERATION OF REAL-WORLD SIGNALS

In most of the above examples (the ones requiring DSP techniques), a DAC is required in order to generate an appropriate analog signal after the DSP has completed its processing function on the converted analog signal. In some cases, however, real world analog signals may be generated directly using DSP and DACs, while omitting the requirement for the front-end ADC. A video example of this is the digital generation of signals which drive videodacs and RAMDACs in raster scan display systems. Artificially synthesized music and speech are good low frequency examples. In reality, however, the real-world analog signals generated using purely digital techniques do rely on information previously derived from analog signals. In display systems, the data from the display must convey the appropriate information to the operator. In synthesized audio systems, the statistical properties of the sounds being generated have been previously derived using extensive DSP analysis (i.e.,sound source, microphone, preamp, ADC, etc.).

DIGITALLY SYNTHESIZED REAL-WORLD SIGNALS

- Graphics Display Systems
- Artificial Synthesis of Sound (Music, Speech)
- These only Require DSP and DACs

Figure 1.5

METHODS AND TECHNOLGIES AVAILABLE FOR PROCESSING REAL-WORLD SIGNALS

Signals may be processed using analog techniques (analog signal processing, or ASP), digital techniques (digital signal processing, or DSP), or a combination of analog and digital techniques (mixed signal processing, or MSP). In some cases, the choice of techniques is clear, in others, there is no clear cut choice, and second-order considerations may be used to make the final decision.

PROCESSING REAL-WORLD SIGNALS

- Analog Signal Processing (ASP): Filtering, Amplification, Modulation, Demodulation, Multiplication, Division, Measurement
- Digital Signal Processing (DSP): Filtering, Amplitude Scaling, Modulation, Multiplication
- Mixed Signal Processing (MSP): Analog and Digital Signal Processing Combined in the Same Function: PC Board, Hybrid, or IC, with Implicit *Real-Time* Operation

Figure 1.6

DIGITAL AUDIO STUDIO SYSTEM

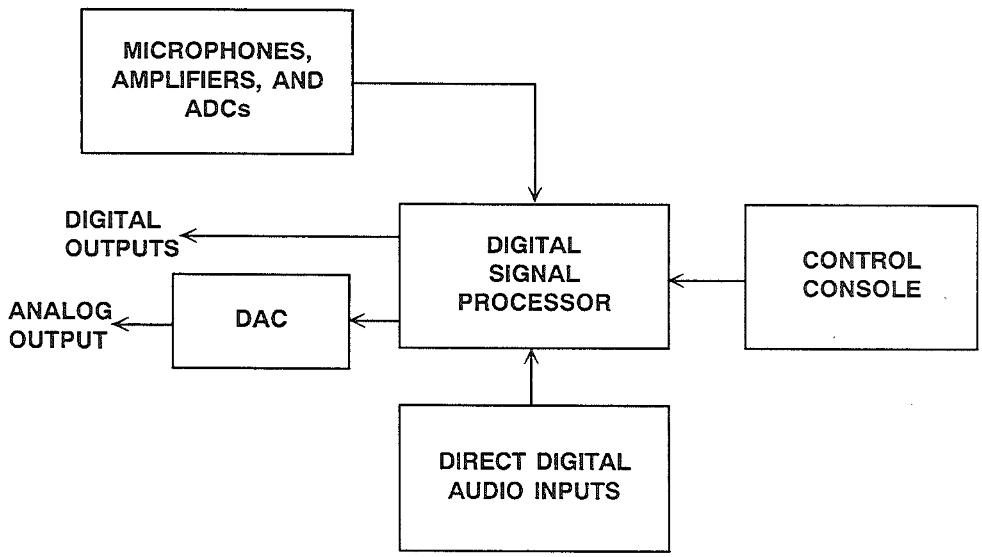


Figure 1.7

With respect to DSP, the factor that distinguishes it from traditional computer analysis of data is its speed and its ability to perform sophisticated digital processing functions. In order to understand the significance of real-time DSP, consider the much simplified digital audio system shown in Figure 1.7. After conversion, all audio processing such as mixing, equalization, filtering, dynamic range control, etc., is handled by the DSP. After processing, the signal is converted back into analog format using a DAC. The ADC sampling rate for such a system is typically 44.1 or 48kSPS. The DSP processing steps, such a digital filtering, must be completed within one cycle of the sampling clock (23µs at 44.1kSPS) in order to "keep up" with the analog signal. This is what is typically meant by *real-time* DSP operation.

The term *mixed signal processing* implies that *both* analog and digital processing is done as part of the same functional block. This functional block may be implemented in the form of a system, a printed circuit board or hybrid microcircuit, or even in the form of a single integrated circuit chip. In the context of this broad definition, ADCs and DACs are considered to be mixed signal processors, since both analog and digital functions are implemented in each. Recent advances in Very Large Scale Integration (VLSI) technology allow complex digital processing as well as analog processing to be performed on the same chip. The very nature of DSP itself implies that these functions can be performed in *real-time*.

ANALOG VERSUS DIGITAL SIGNAL PROCESSING

Today's engineer faces a challenge in selecting the proper mix of analog and digital techniques to solve the signal processing task at hand. It is impossible to process real-world analog signals using purely digital techniques, since all transducers (thermocouples, strain gages, microphones, piezoelectric crystals, disk drive heads, etc.) are inherently analog elements. Therefore, some sort of signal conditioning circuitry is required in order to prepare the transducer output for further signal processing, whether it be analog or digital. Signal conditioning circuits are, in reality, analog signal processors, performing such functions as multiplication (gain), isolation (instrumentation amplifiers and isolation amplifiers), detection in the presence of noise (high common-mode instrumentation amplifiers, line drivers, and line receivers), dynamic range compression (log amps, LOGDACs, and programmable gain amplifiers), and filtering (both passive and active).

ANALOG SIGNAL CONDITIONING AND PROCESSING

- Amplification (Gain)
- Impedance Transformation
- Removing Common Mode Noise
- Isolation
- Cable Driving and Receiving
- Multiplication of Signals
- Dynamic Range Compression
- Programmable Amplification
- Filtering (Passive and Active)

Figure 1.8

Several methods of accomplishing signal processing are shown in Figure 1.9. The top portion of the figure shows the purely analog approach. The latter two parts of the figure show the DSP approach. Note that once the decision has been made to use DSP techniques, the next decision must be where to place the ADC in the signal path. In general, as the ADC is moved closer to the actual transducer, more of the analog signal conditioning burden is placed on the ADC. This added ADC complexity may take the form of increased sampling rate, wider dynamic range, higher resolution, input noise rejection, input filtering, etc., all of which imply greater ADC costs. In fact, the probability of

ANALOG AND DIGITAL SIGNAL PROCESSING OPTIONS

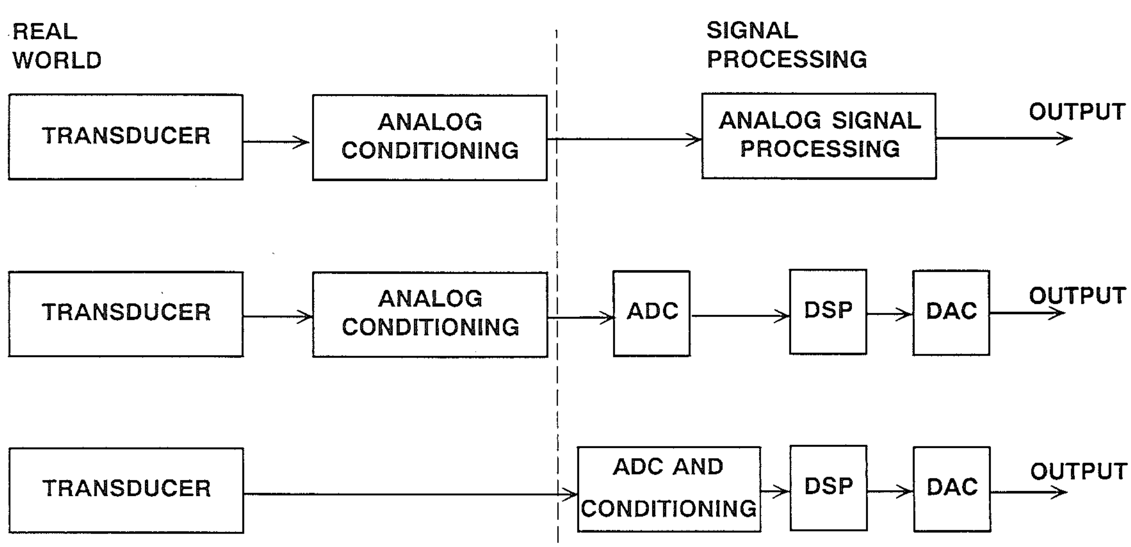


Figure 1.9

finding an ADC that is directly matched to the transducer output and has all the other desired system characteristics is indeed quite remote, except in very special cases such as the simple bimetallic thermal switch.

The system designer should face the fact that some type of signal conditioning will inevitably be required before the actual ADC. Although there are no hard and fast rules available as to where in the signal path to place the ADC, the current state-of-the-art in sampling ADCs shown in Figure 1.10 may be useful in determining the initial boundary. Usually, the performance requirements on the ADC (and hence the cost) can be relaxed at the expense of additional analog signal conditioning and processing. For instance, a programmable gain amplifier, or a logarithmic amplifier placed ahead of the ADC may reduce the ADC dynamic range requirements.

STATE OF THE ART IN ADCs

Resolution	Sampling Rate
22 bits	1 kSPS
20 bits	4 kSPS
18 bits	50 kSPS
16 bits	500 kSPS
14 bits	10 MSPS
12 bits	25 MSPS
10 bits	75 MSPS
8 bits	500 MSPS

Figure 1.10

In order to understand how to most effectively utilize either analog and/or digital signal processing, the system designer must first understand the capabilities of each. The following sections will investigate analog elements both as conditioners and as signal processors themselves.

AMPLIFIERS USED AS SIGNAL CONDITIONERS

Operational amplifiers are extremely useful devices for coupling transducer outputs to the signal processor inputs (see Figure 1.11). Op amps can provide gain, impedance transformation, filtering, and level shifting. The amplifier's input characteristics must match the output characteristics of the transducer with respect to impedance, signal level, dynamic range, bandwidth, etc. The output of the amplifier must also match the input characteristics of the signal processor with respect to the same characteristics. If the signal processing chain begins with an ADC, it is extremely important that the signal applied to the ADC adequately fills the converter's input range without overdrive. Small signals will not fully utilize the dynamic range of the ADC, while signals which are too large will overdrive the converter and cause hard-limiting. In addition, care must be taken to insure that the amplifier does not degrade the performance of the signal processor, especially with respect to dynamic performance specifications such as total harmonic distortion (THD), signal-to-noise ratio, etc. Fortunately, a wide variety of precision dc coupled high speed IC op amps are available to fit almost every application.

THE AMPLIFIER AS A SIGNAL CONDITIONER

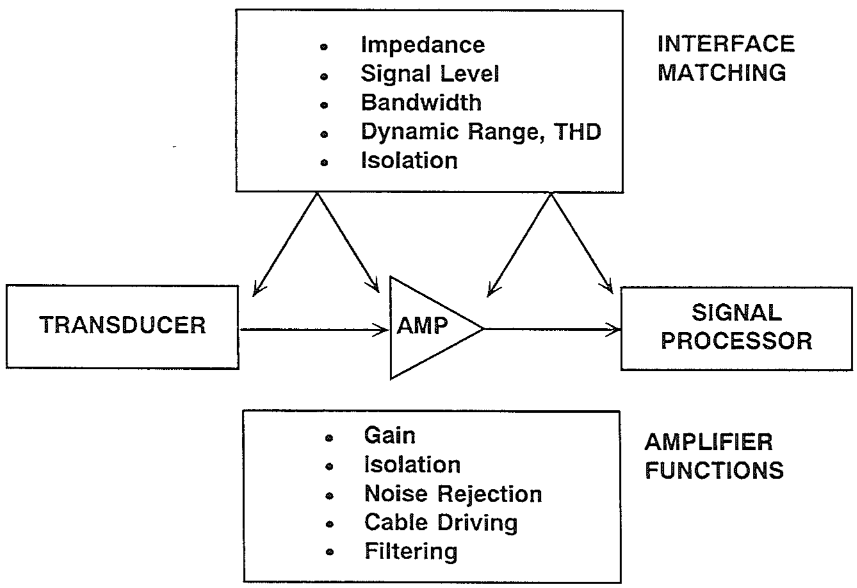


Figure 1.11

KEY OP AMP SPECS

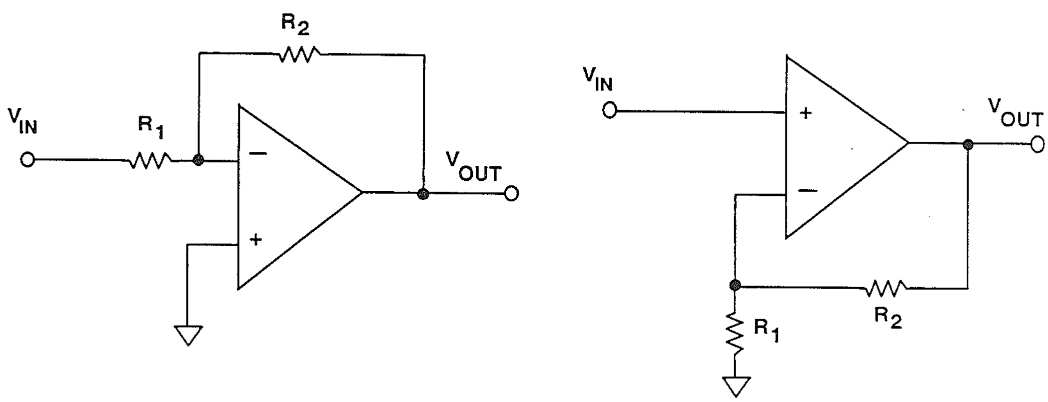
- Open Loop Gain
- Input Impedance
- Input Offset Voltage
- Input Bias Currents
- Output Voltage/Current
- Temperature Coefficients
- Long Term Drift
- Bandwidth
- Settling Time
- Distortion (THD)
- Noise: Voltage and Current
- Slewrate

Figure 1.12

Without going into op amp theory in much detail, we should, however, examine a few basic op amp configurations which are suitable to a wide variety of applications. Figure 1.13 shows the two most fundamental op amp configurations: inverting and non-inverting. Both configurations have their relative strengths and weaknesses. The inverting mode is more commonly used when level shifting is required, but presents an input impedance of R_1 to the input. The non-inverting mode has a high input impedance, but is more awkward to use if level shifting is required (see Figure 1.14).

BASIC OP AMP CONFIGURATIONS

INVERTING | **NON-INVERTING**

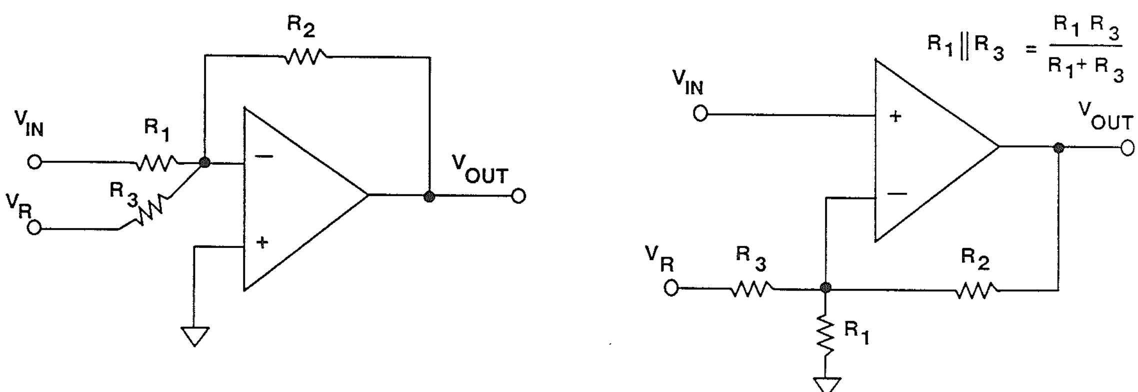


$V_{OUT} = -\frac{R_2}{R_1} V_{IN}$ **(IDEAL EQUATIONS)** $V_{OUT} = \left(1 + \frac{R_2}{R_1}\right) V_{IN}$

Figure 1.13

OP AMPS AS LEVEL SHIFTERS

INVERTING | **NON-INVERTING**



$R_1 \| R_3 = \frac{R_1 R_3}{R_1 + R_3}$

$V_{OUT} = -\frac{R_2}{R_1} V_{IN} \underbrace{- \frac{R_2}{R_3} V_R}_{\text{OFFSET}}$ **(IDEAL EQUATIONS)** $V_{OUT} = \left(1 + \frac{R_2}{R_1 \| R_3}\right) V_{IN} \underbrace{- \frac{R_2}{R_3} V_R}_{\text{OFFSET}}$

Figure 1.14

DIFFERENTIAL AND INSTRUMENTATION AMPLIFIERS

In many cases, the signal from the transducer is corrupted by the presence of an unwanted common-mode signal such as noise, or ac power line coupling. In these cases, a differential input configuration such as the one shown in Figure 1.15 may be desirable. Signals which are common to both V_1 and V_2 are rejected from the output by the common-mode rejection ratio (CMRR) of the op amp. Unfortunately, CMRR decreases as a function of frequency, and high frequency noise is not rejected. In addition, the CMRR of the differential input amplifier depends upon accurate ratio matching of the four resistors. A mismatch of 0.1% in any of the four resistors will produce a CMRR of approximately 66dB. Another problem with the simple differential circuit is that V_1 and V_2 see different input impedances, i.e., the input is unbalanced. Even with these sources of error, the differential configuration is quite useful as a line receiver for properly terminated cables, such as 600Ω audio. In this case, the unbalanced input is not a real problem, since the value of R_1 is chosen to be much higher (typically greater than 20kΩ) than the 600Ω line impedance. A discussion of audio line drivers and receivers will follow shortly.

DIFFERENTIAL RECEIVER INPUT CONFIGURATION

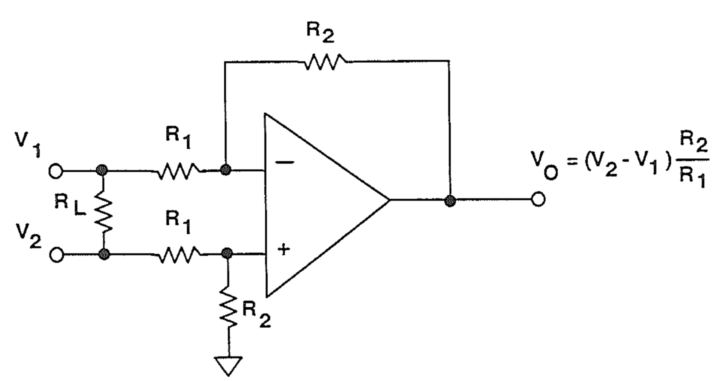
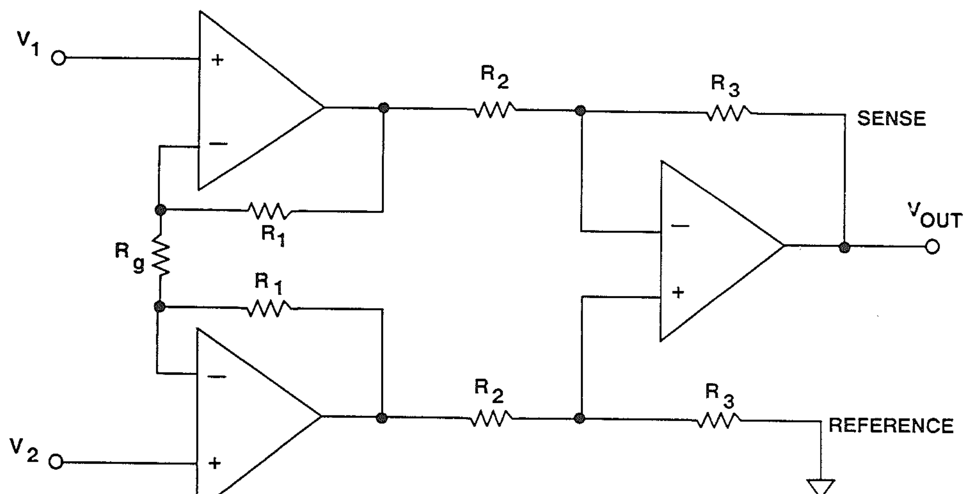


Figure 1.15

For true balanced, high impedance inputs, three op amps may be connected to form an *instrumentation amplifier* as shown in Figure 1.16. In this configuration, both V_1 and V_2 see high input impedances, and the input is balanced. The gain of the amplifier is usually set by an external resistor, R_g, as in the case of the AD521 and AD625. Such circuits are generally used for dedicated fixed-gain applications. Common mode rejection is very high as shown in Figure 1.17. Pin-programmable instrumentation amps, such as the

1

3 OP-AMP INSTRUMENTATION AMP



$$V_{OUT} = (V_2 - V_1)\left(\frac{R_3}{R_2}\right)\left(\frac{2R_1}{R_g} + 1\right)$$

Figure 1.16

INSTRUMENTATION AMPLIFIER USED TO REJECT COMMON MODE VOLTAGE

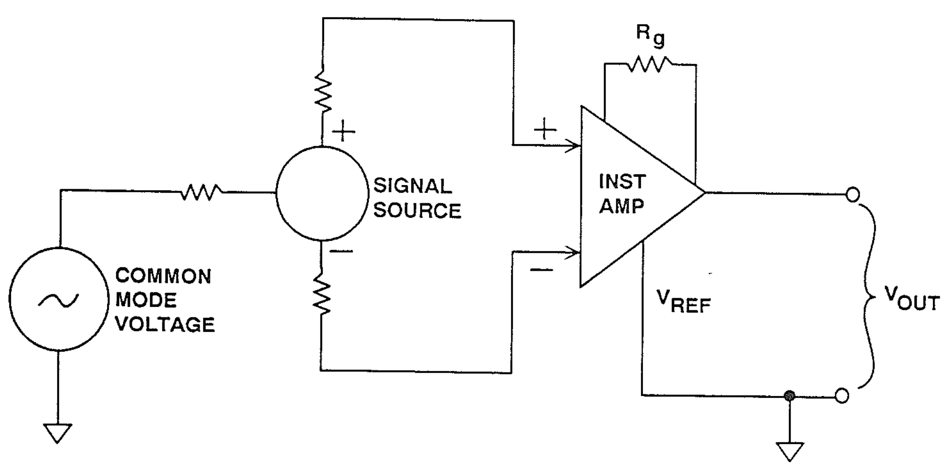
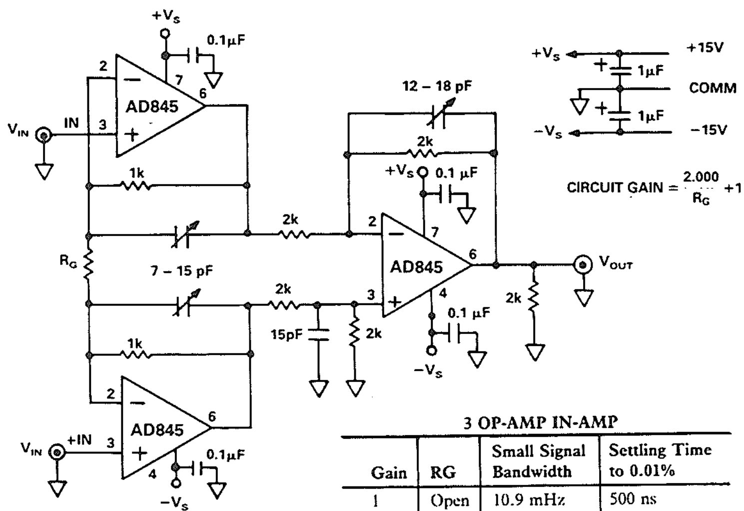


Figure 1.17

HIGH-PERFORMANCE, HIGH-SPEED INSTRUMENTATION AMPLIFIER



3 OP-AMP IN-AMP

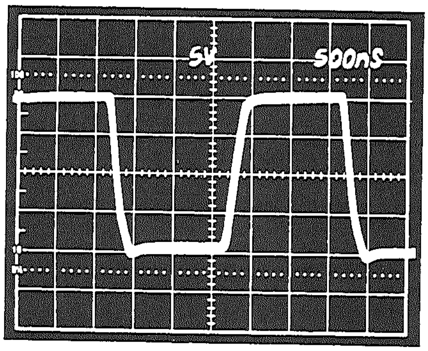
Gain	RG	Small Signal Bandwidth	Settling Time to 0.01%
1	Open	10.9 mHz	500 ns
2	2k	8.8 mHz	500 ns
10	226Ω	2.6 mHz	900 ns
100	20Ω	290 kHz	7.5 μs

Figure 1.18

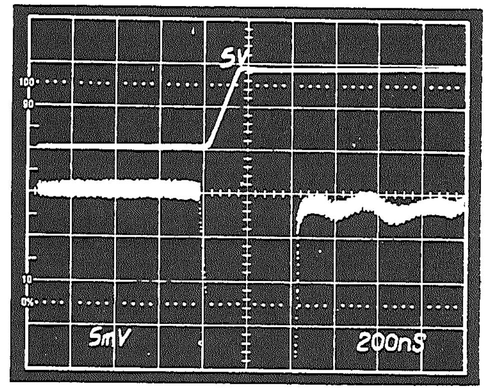
AD524 and AD624, have a set of internal resistors; a limited set of fixed gains in the range of 1 to 1000 are chosen by appropriately interconnecting the resistors via external pins. Digitally (or *software-*) programmable instrumentation amps are completely self-contained, with gains set by a 2, 3, or 4 bit digital code as in the case of the AD365 (gains of 1, 10, 100, 500).

A wideband instrumentation amplifier using three AD845 FET input op amps is shown in Figure 1.18. Gain can be adjusted from 1 to 1000. Low input bias currents and fast settling times are achieved with the AD845. The AD843 FET input op amp may also be used for even higher bandwidth. Note that the bandwidth for the circuit is 10.9MHz at a gain of 1 and 2.6MHz at a gain of 10. Settling time for the entire circuit is 500ns to 0.01% for a 10V step at a gain of 1 as shown in Figure 1.19.

HIGH SPEED INSTRUMENTATION AMP PULSE RESPONSE AND SETTLING TIME



The Pulse Response of the Three Op Amp Instrumentation Amplifier: Gain = 1, Horizontal Scale: 0.5μs/Div, Vertical Scale: 5V/Div



Settling Time of the Three Op Amp Instrumentation Amplifier: Horizontal Scale: 200 ns/Div, Vertical Scale, Pulse Input: 5V/Div; Output Settling: 1mV/Div

Figure 1.19

LINE DRIVERS AND RECEIVERS

To avoid noise pickup through the cabling, it is desirable to place the signal processor as close to the conditioning circuits as possible. If this is not possible, then the analog signal must be transmitted over cable, either coaxial or shielded twisted pair so that any noise picked up will be common-mode. This requires a balanced line driver at the sending end and a balanced line receiver at the receiving end. A good example of a balanced line driver for audio bandwidths is the Analog Devices/PMI SSM-2142. This device converts a single-ended input signal to a fully balanced, high drive, high output signal pair (see Figure 1.20). The SSM-2142 mimics the fully balanced performance of transformer-based solutions for line driving. However, the SSM-2142 maintains lower distortion and occupies much less board space than transformers, while achieving comparable common-mode rejection. Since the output stages have balanced impedances due to active laser trimming, hum and noise are rejected over the full audio bandwidth. It is suggested that a suitable differential input amplifier such as the SSM-2141 is used at the receiving end to maintain overall system performance. A functional block diagram of the device along with key specifications are given in Figure 1.21. A typical system application for the driver and receiver combination is shown in Figure 1.22. Typical THD performance at the SSM-2141 single-ended output for a 10V rms output is 0.004% at 1kHz using 500 feet of Belden 8451 cable.

SSM-2142 BALANCED AUDIO LINE DRIVER

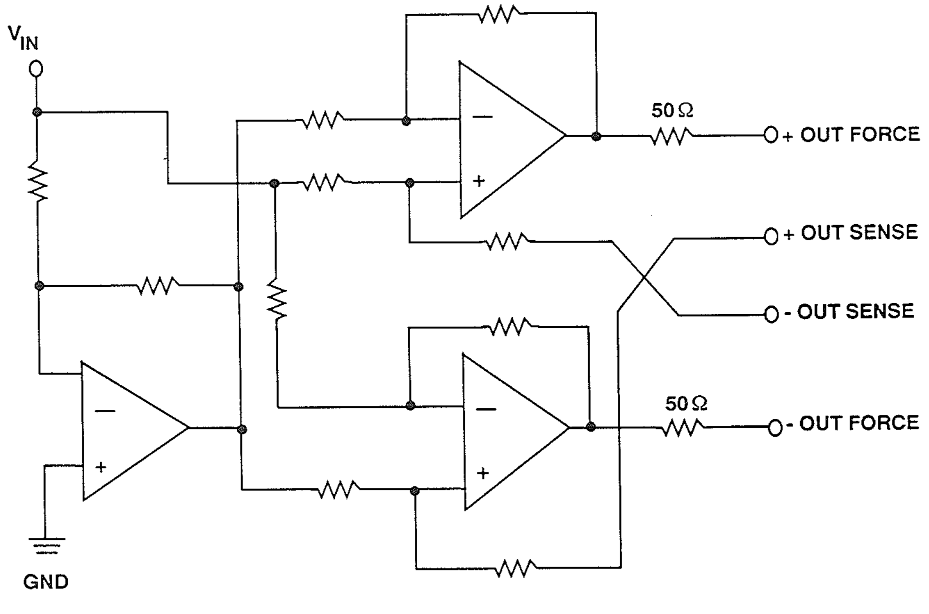
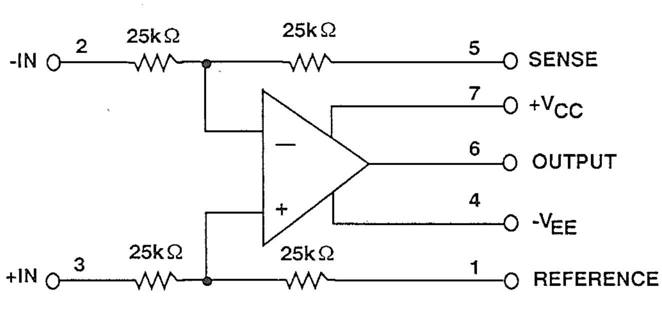


Figure 1.20

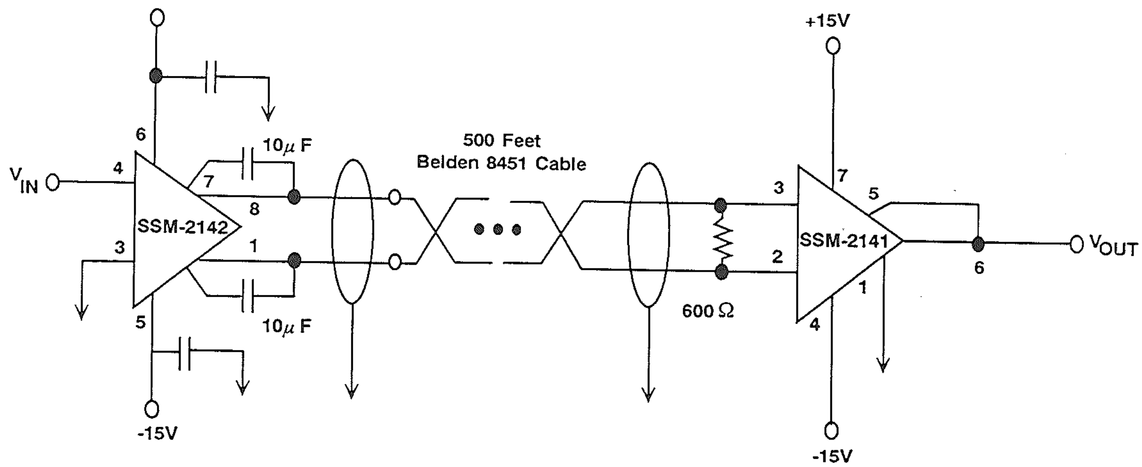
SM-2141 HIGH COMMON-MODE REJECTION DIFFERENTIAL AUDIO LINE RECEIVER



- CMRR: 100dB @ 60 Hz
  70dB @ 20 kHz
  62dB @ 40 kHz
- THD: 0.001%
- BANDWIDTH: 3MHz

Figure 1.21

AUDIO LINE DRIVER AND RECEIVER (SSM-2142, SSM-2141)



■ THD: 0.004% FOR V_{OUT} = 10Vrms

Figure 1.22

Differential techniques such as those described above become difficult if not impossible at video frequencies primarily because of the lack of suitable line drivers and receivers having good common mode rejection and low distortion at high frequencies. Furthermore, video signals must be transmitted over properly terminated cable in order to avoid standing waves and distortion. Typical configurations for high speed cable driving are shown in Figure 1.23. Note that the cables are terminated both at the source and at the load to minimize reflections. This implies that a gain of at least two is required in order to restore the signal to its original level at the receiving end of the cable. Figure 1.24 shows a video line driver circuit using the AD829 which has been optimized for bandwidth flatness and low differential gain and phase. This circuit will drive reverse-terminated 75Ω video cable to standard video levels (1V p-p) with 0.1dB gain flatness to 30MHz with only 0.02° and 0.02% differential phase and gain at the 4.43MHz PAL color subcarrier frequency. This level of performance meets the requirements for high-definition video displays and test equipment.

NONINVERTING AND INVERTING CABLE DRIVERS

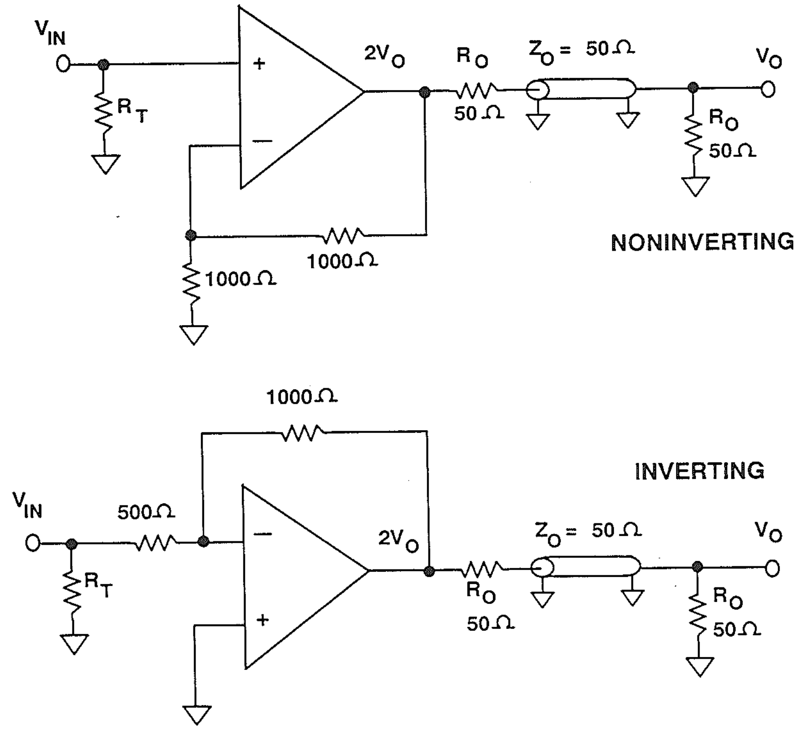


Figure 1.23

LOW DIFFERENTIAL GAIN AND PHASE VIDEO LINE DRIVER: AD829

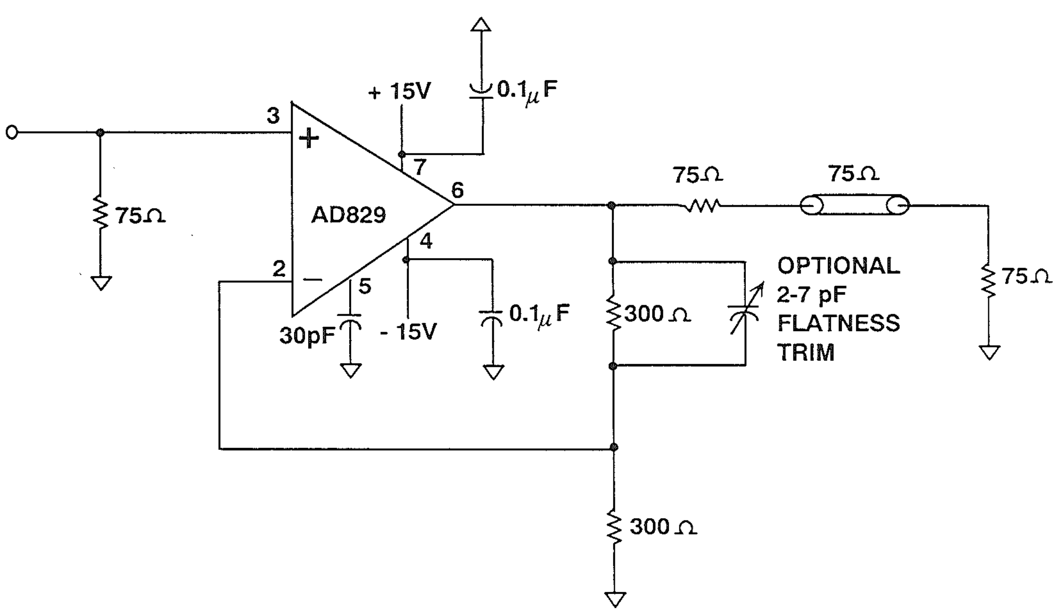


Figure 1.24

ISOLATION AMPLIFIERS

The isolation amplifier has an input circuit that is galvanically isolated from the power supply and the output circuit. In addition, there is minimal capacitance between the input and the rest of the device. Therefore, there is no possibility for dc current flow, and minimum ac coupling. Isolation amplifiers are intended for applications requiring safe, accurate measurement of low frequency voltage or current (up to about 100kHz) in the presence of high common-mode voltage (to thousands of volts) with high common mode rejection; line-receiving of signals transmitted at high impedance in noisy environments; and for safety in general-purpose measurements where dc and line-frequency leakage must be maintained at levels well below certain mandated minima. Principle applications are in electrical environments of the kind associated with medical equipment, conventional and nuclear power plants, automatic test equipment, and industrial process control systems.

In the basic two-port form, the output and power circuits are not isolated from one another. In the three-port isolator shown in Figure 1.25, the input circuits, output circuits, and power source are all isolated from one another. The figure shows the circuit architecture of a self-contained isolator, the AD210. An isolator of this type requires power from a two-terminal dc power supply. An internal oscillator (50kHz) converts the dc power to ac, which is transformer-coupled to the shielded input section, then converted to dc for the input stage and the auxiliary power output. The ac carrier is also modulated by the amplifier output, transformer-coupled to the output stage, demodulated by a phase-sensitive demodulator (using the carrier as the reference), filtered and buffered, using isolated dc power derived from the carrier. The AD210 allows the user to select gains from 1 to 100 using an external resistor. Bandwidth is 20kHz, and voltage isolation is 2500V rms (continuous) and ± 3500V peak (continuous).

AD210 THREE-PORT ISOLATION AMPLIFIER

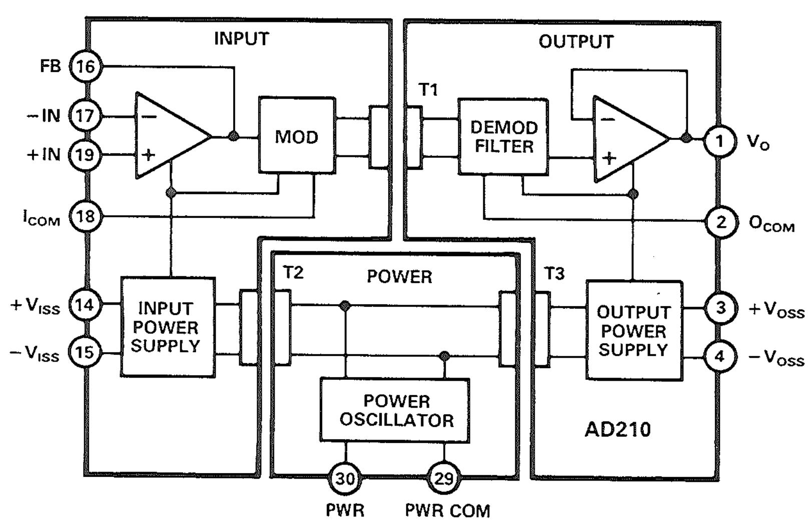


Figure 1.25

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