# CHAPTER 7
# ULTRASONIC TESTING

## I. HISTORY

The use of sound energy to determine the integrity of solid objects is probably as old as mankind's ability to manufacture objects in pottery and metal. The English language has many words and phrases that illustrate the acceptance of this fact and hint at the way in which sound was used in the past to test for integrity. Expressions such as "the ring of truth" or "sound as a bell" are commonplace in everyday speech to indicate quality, honesty, or good health.

Both phrases allude to the fact that a sharp tap on a solid object will set up a vibration at the natural frequency of the object; that is how a bell or any percussion instrument makes music. Any major disruption of the homogeneity of the object will distort that natural frequency and indicate that there is a problem. The instinct to tap an object is so ingrained into human nature that it probably accounts for all those people who, when viewing a prospective second-hand car purchase, unconsciously kick a tire!

The problem with this simple approach to testing an object is that it takes a relatively large imperfection to cause a significant change in sound for the human ear to detect. This is because the determining factor is the wavelength of the sound wave encountering an imperfection in relation to the size of the imperfection. Wavelength, in turn, depends on the speed of sound in the object and the frequency, or pitch, of the sound wave. Within the human range of audible sound frequencies, the wavelength is rather large in most metals. It wasn't until the ability to generate and detect sound waves at much higher frequencies existed that smaller discontinuities could be detected in metals.

The first steps toward this ability were taken in the 1870s with the publication of Lord Rayleigh's work on sound, "The Theory of Sound." This work explained the nature and properties of sound waves in solids, liquids, and gasses, which led to the development of the techniques that are currently in use in nondestructive testing.

The means for generating and detecting sound waves at frequencies above the audible range followed shortly after with the discovery of the piezoelectric effect by the Curie brothers and Lippmann. In 1880, the Curies found that an electrical potential could be generated by applying mechanical pressure to plates cut in a particular fashion from certain crystals. The following year, Lippmann discovered that the reverse was true and that the application of an electrical signal to these plates caused a mechanical distortion. Naturally occurring crystals of quartz, tourmaline, and Rochelle salt were among those materials displaying the piezoelectric effect.

Over the years, there have been many uses made of this effect, from crystal microphones and gramophone pickups to spark generators for cigarette lighters and, of course, ultrasonic transducers for NDT. However, growth in use of piezoelectricity was slow.

As early as 1912, following the Titanic disaster, it was suggested that sound waves could be used to detect icebergs at sea, an idea that received further stimulation during World War I for the detection of submarines. The pulse echo system developed for this application gave rise to peacetime uses between the two world wars in the fields of hydrographic surveys charting the ocean depths and fishing, where echo sonar was used to detect shoals of fish.

S. Y. Sokolov, in Russia, was the first to suggest using ultrasonic waves to detect discontinuities in metals. In 1929, he described some experiments in which he generated ultrasonic waves in metals including cast iron and steel samples, which were subsequently sectioned. In 1935, he described his design for piezoelectric transducers for generating and detecting ultrasound, including a method of coupling sound to the metal. His method worked through a transmission technique using continuous waves with quartz transducers and mercury as a couplant. An alternating current generator was used to drive the quartz crystal transmitter and vibrations reaching the receiver caused an alternating signal that could be measured.

Much work was carried out using this technique between the wars, particularly in Russia and Germany. However there was a problem with continuous wave testing and much of the experimental work concentrated on trying to find a solution. The principle of the continuous wave technique was that energy transmitted through the sample would generate a signal of particular amplitude when no obstruction was present. Some of the sound energy would be obstructed by a discontinuity so that the resulting received signal would be weaker.

This principle was fine as long as there was only enough energy to complete one full transition of the sample, as was the often the case with large castings, where grain size rapidly attenuated the sound. If, however, there was sufficient energy to set up multiple reflections of the sound within the sample, the reflected waves joined in with later sound waves to produce confusion. If the reflections were in phase with the continuous wave, there was an apparent increase in transmitted energy. On the other hand, if the reflections were out of phase, the signal became weaker, giving a false indication. The same effect could be observed in narrow samples, where the beam could reflect from the side walls. This reflected beam path, being longer than the straight-through path, could also create constructive or destructive interference with the main signal.

This problem limited the development and use of ultrasonic flaw detection until World War II, when several workers on either side of the combating nations adopted the technique used in echo sonar known as pulse echo. In this system, short pulses of sound are transmitted at regular intervals, the transmitted pulses and the resulting echoes being displayed on a cathode ray tube (CRT). The interval between transmitted pulses is arranged to be sufficient to allow all internal reflections to die away before the next pulse is started, thus avoiding the interference effects previously encountered. Both the transmitter and receiver crystals can be positioned on the same surface. The sweep time of the CRT can be arranged to display one or more transit periods, and the position of the initial pulse and subsequent echoes can be used to determine the depth of the various reflecting surfaces, including discontinuities.

The pulse echo system was devised around 1942. In Britain, the development was attributed to D. O. Sproule, and in the United States to F. Firestone. After the war, the two approaches were compared, together with developments that had taken place in Germany. The main difference was that Sproule had used separate transmitter and receiver elements, whereas Firestone used the transmitter element as a receiver during its quiescent periods. Since there were merits in both approaches under various circumstances, developments since World War II have tended to use either single or dual element transducers.

At this stage of the development of ultrasonic flaw detection, only "straight beam" compression wave techniques, suitable for detecting reflectors parallel to the scanning surface, were in use. Attempts to angle the beam to reflect from surfaces at other orientations gave confusing results because of the existence of mode conversion in solids. With only a small angulation of the beam at the test surface, both compression waves and shear waves were generated in the object being tested. These traveled at different angles of refraction and at different velocities in the sample, making interpretation of the displayed echoes difficult.

In 1947, Sproule described a transducer design that would generate only shear waves in the sample, and this advance allowed the development of ultrasonic techniques for many other types of discontinuities in the welding, aerospace, and foundry industries. Rapid increase in the use and application of ultrasonics followed this development.

During the next 20 years, much of the development of ultrasonic techniques, as opposed to instrumentation, centered on the accurate sizing of the reflectors detected by the beam. Various approaches using beam parameters were tried, with varying degrees of success. Some were intended to estimate actual size, some were intended to assess the minimum theoretical reflector surface area, and some were intended to provide a common "go, no-go" reporting standard. In Britain, the intensity drop technique developed by C. Abrahams used the plotted –6dB or –20dB edges of the sound beam to plot the longitudinal and vertical extents of the reflector. In Germany, Krautkramer developed the DGS (distance gain amplitude) system to compare reflector amplitude against known-diameter circular reflectors (flat-bottomed holes) at defined depths for defined transducers. In the United States, signals were compared and reported against a DAC (distance amplitude correction) curve to produce a common reproducible reporting standard.

Each of the above techniques proved to be unsatisfactory for the application of fracture mechanics to determine fitness for purpose. Only the intensity drop technique claimed to assess the critical "through-thickness" dimension and this technique was heavily dependent on operator skill and reflector profile; repeatability of results was poor.

Sproule, in the 1950s, had described the tip diffraction signal originating at the tip of a discontinuity and defined its amplitude as being 30dB smaller than a corner reflector at the same depth. Whitford, in the 1960s, developed an alternative sizing system to the intensity drop system that he called the "maximum amplitude" system, in which the last maximum, defining the edges of the reflector, was the tip diffracted shear wave echo. In theory, this diffraction signal gave a much more precise location of the limits of the reflector. However, the signal from the diffracted shear wave is weak and difficult for the ultrasonic practitioner to positively identify.

It was Silk, in 1977, who first used the time of flight diffraction (TOFD) technique to display the top and bottom edges of discontinuities in a way that would allow greater accuracy of through-thickness measurement. The method employs angled compression-wave transducers, located on the same surface, to both transmit and receive sound. Lateral wave and tip diffraction signals allow accurate triangulation of the top and bottom edges of the reflector. Recent advances in instrumentation allowing real-time and postinspection analysis of results using computer technology have increased the number of users exploiting this technique.

If physicists have advanced the understanding of the theory and developed more and more ways of using ultrasonics, engineers have been no less productive in improving the instrumentation over the last 50 years. Early ultrasonic flaw detectors used vacuum tubes (valves), needed generated electricity, and were heavy (see Figure 7-1). Using quartz crystals, signal amplitude was poor and resolution very poor. After a shaky start, semiconductor technology has produced flaw detectors that are light, very portable, and together with synthetic crystal materials offers performance that is greatly enhanced. Much of this had been achieved by the mid 1970s.

During the 1980s and 1990s, microchips have been incorporated into the flaw detector, allowing the operator to store calibration parameters and signal traces. This, in turn, allows off-line analysis and reevaluation at a later date. Digital technology and the use of LCD display panels instead of CRTs during the 1990s has further reduced the size and weight of the flaw detectors.

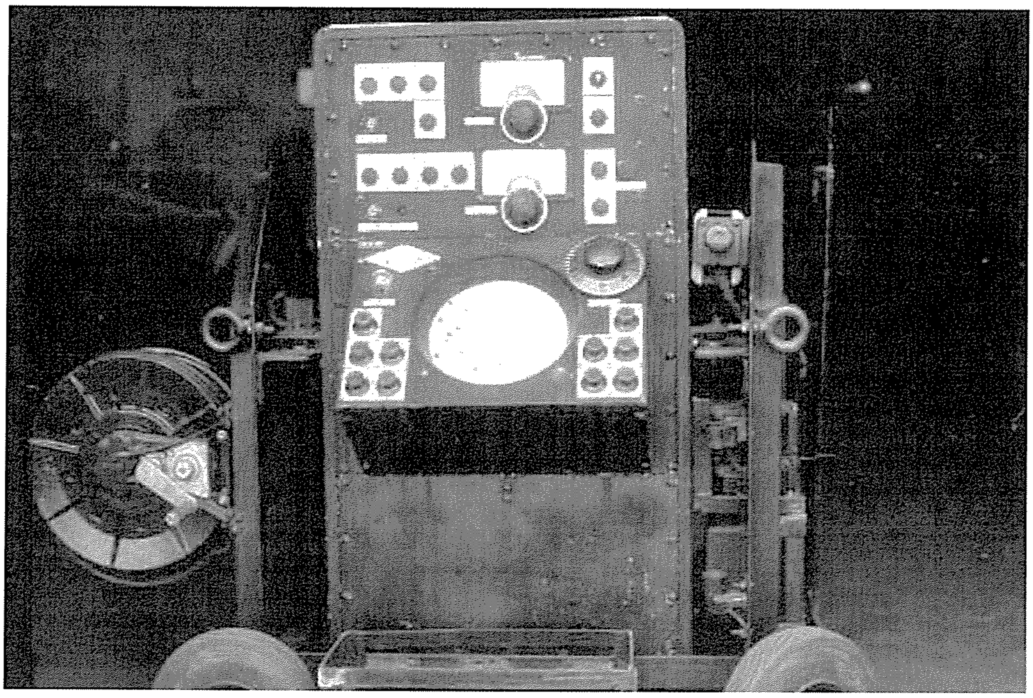


**FIGURE 7-1** Early ultrasonic instrument called the Supersonic Reflectoscope.

## II. THEORY AND PRINCIPLES

### Nature of Sound Waves

Sound waves are simply vibrations of the particles making up a solid, liquid, or gas. As an energy form they are therefore an example of mechanical energy, and it follows that, since there must be *something* to vibrate, sound waves cannot exist in a vacuum.

The only human sense that can detect sound waves is hearing, and that sense is restricted to a relatively narrow range of vibration frequencies called "the audible range". It follows that there will be vibration frequencies that are so low or so high that they cannot be detected by the human ear.

The unit of frequency is the hertz, abbreviated as Hz, defined as "one cycle of vibration per second." Sounds below approximately 16 Hz are below the limit of human hearing and are called "subsonic vibrations," and sounds above approximately 20,000 Hz are too high to be heard and are called "ultrasonic vibrations." Between those two values, in the audible range, it is more common to use the term "pitch" to refer to frequency; a high-pitched sound means high audible frequency, and low-pitched means low audible frequency. A piano key pitched at "middle C" is at a frequency of 260 Hz.

Abbreviations are used for high frequencies; 1000 Hz is shortened to 1 KHz (one kilohertz), 1,000,000 Hz becomes 1 MHz (one megahertz), and a billion cycles per second becomes 1 GHz (one gigahertz). In ultrasonic flaw detection, most testing is carried out in the MHz range (0.5 MHz to 25 MHz).

It is fortunate that there are devices called "transducers" that will change sound waves into electrical energy that can be displayed as visual signals on a cathode ray tube (CRT) or liquid crystal display (LCD) screen. This allows all sounds, including those outside the audible range, to be detected and studied. A transducer is defined as a device that will change one form of energy into another, and vice versa. Materials exhibiting the piezoelectric effect are commonly used to both generate and detect sound waves.

## Vibration and Periodic Motion

A vibration is an example of periodic motion, a term that suggests that the body or particle concerned is undergoing some repetitive change of position with time. Another example is a pendulum swinging back and forth at a steady rate or frequency. To study the essential requirements for a vibration, consider Figure 7-2. The weight, W, is suspended from a beam by a spring. At rest, two equal and opposite forces are acting on the weight—gravity (G), acting downwards, is opposed by the tension (T) in the spring. The weight is said to be in a state of equilibrium. If the weight is lifted, slackening the spring, then released, gravity will try to restore the weight to its original position. If the weight is pulled down, the tension in the spring will increase; and when the weight is released, this extra tension will try to restore the weight to its original position.

This arrangement provides all the essentials to sustain a vibration. First, there must be something (the weight) to move, and second, there must be a restoring force that will try to counteract that movement or displacement (in this case gravity and the spring).

Imagine that the weight is pulled down from its normal rest position A in Figure 7-3 to position B, and then released. The extra stretch in the spring will exert a force on W so that it will begin to accelerate back to position A. As it moves, the stretch on the spring decreases until, at position A, both force G and force T are equal again. Since the weight has been accelerating all this time, it has now reached its maximum speed.

Any mass in motion possesses inertia, and this inertia will carry the weight on past the equilibrium position A. But, of course, as soon as it passes A, the spring slackens so that T becomes less than G. In other words, gravity starts to slow the weight down. Eventually, the weight comes to rest at a new position C. Now gravity outweighs the tension in the spring, trying to accelerate W back to position A. On reaching A, inertia ensures that the weight overshoots again and the whole train of events starts again.

If a pen is fixed to the weight and allowed to write against a piece of chart paper pulled

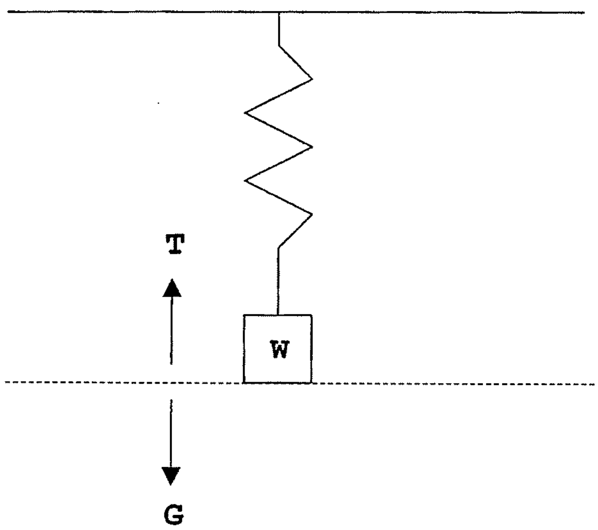



**FIGURE 7-2** Weight on spring extended.

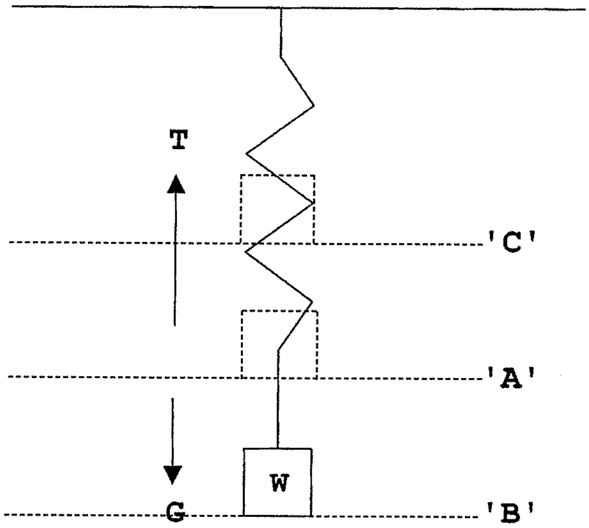



FIGURE 7-3 Weight on spring extended 2.

at a steady speed past the pen, a trace of the movement of the weight with time, as it bobs up and down, will be drawn, as shown in Figure 7-4. This trace is typical of all periodic motion and faithfully shows what has happened to the weight.

The steeper the line traced out, the faster is the movement of the weight. At the maximum displacement up or down, the trace is flat, showing that the weight has stopped briefly. As the weight passes through the normal rest position each time, the line is steepest, showing maximum speed. On the trace, $t_1$ represents the weight, traveling upwards and passing position A. The next trace position, $t_2$, shows the weight traveling down but again passing A, and $t_3$ again shows the weight traveling upward at position A.

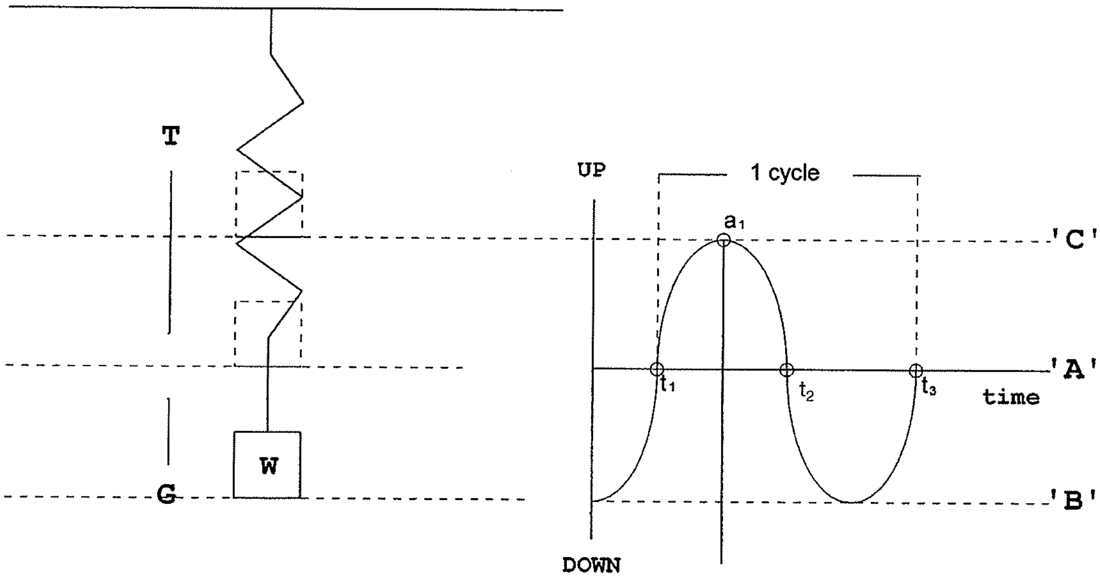



FIGURE 7-4 Weight and cycle.

Between $t_1$ and $t_3$ the weight has traveled up to its top limit, down to its bottom limit, and back to where it started. The trace between $t_1$ and $t_3$ has drawn "one cycle" of the motion of the weight in a set period. If the weight is allowed to carry on bouncing up and down until it eventually comes back to rest, the trace will only go on drawing repeats of that motion, each cycle occupying the same time span. The number of cycles completed in one second has already been defined as the frequency of the vibration.

The maximum displacement of the weight from its normal rest position is shown on the trace as $a_1$ and is known as the amplitude of the vibration. With sound vibrations, frequency is perceived as the pitch of the sound, whereas amplitude is the loudness of the sound.

## Sound Vibrations

For sound waves in solids, liquids, and gases, the vibrating bodies are the particles making up the substance, and the restoring forces are the elastic bonds holding the substance together. The particles can be imagined to be joined together by springs. If one particle moves toward its neighbor, the spring gets squashed and tends to push the invader back "home." Similarly, if it moves away from its neighbor, the spring gets stretched and the particle is pulled back into place.

Audible sound is an example of a vibration mode called a "compression wave." It travels from the source by a succession of shunting actions from one particle to the next. Each particle vibrates at the frequency of the sound, oscillating to and fro by a distance that is the amplitude or loudness of the sound. As each particle oscillates, it squashes the "spring" to the next neighbor and starts the neighbor oscillating. As the oscillation passes from one particle to the next, and the next, and so on, the sound wave is said to travel or "propagate" through the material. Note that individual particles do not migrate to another place; they only oscillate about a mean position.

## Modes of Propagation

The type, or mode, of sound wave propagation described above (compression wave) can exist in solids, liquids, or gases. Other modes of vibration can exist, but only in solids. The various ways in which sound can propagate are usually described in terms of the direction of particle motion in relation to the direction in which the sound wave travels. Compression waves can be defined on this basis as waves in which the particle motion is in the same plane as the direction of propagation.

All three media have forces that bind the particles together to resist squashing or pulling apart (compression or tension). In solids, this is provided by the modulus of elasticity, known as "Young's modulus of elasticity." The pressure of an entrapped gas as it is squashed rises to oppose the squashing force, and the pressure drops if the volume is increased, the partial vacuum applying the restoring force (see Figure 7-5).

Solids, unlike liquids and gasses, also have rigidity that is a resistance to shear loads. It is the rigidity that has to be overcome when snapping a stick, for instance. The name for this resistance to shear loads in solids is called "the modulus of rigidity," and it allows sound to propagate in a different way under certain circumstances. This new mode of propagation is known as a shear wave and is defined as a wave in which the particle motion is at right angles to the direction of propagation.

If a shear wave is set up so that it just skims along the surface of a solid, it again changes mode to one, which is contour following with a peculiar particle motion. This contour following wave is called a surface wave and is defined as a wave in which the particle motion is elliptical, with the major axis of the ellipse perpendicular to the direction of propagation.

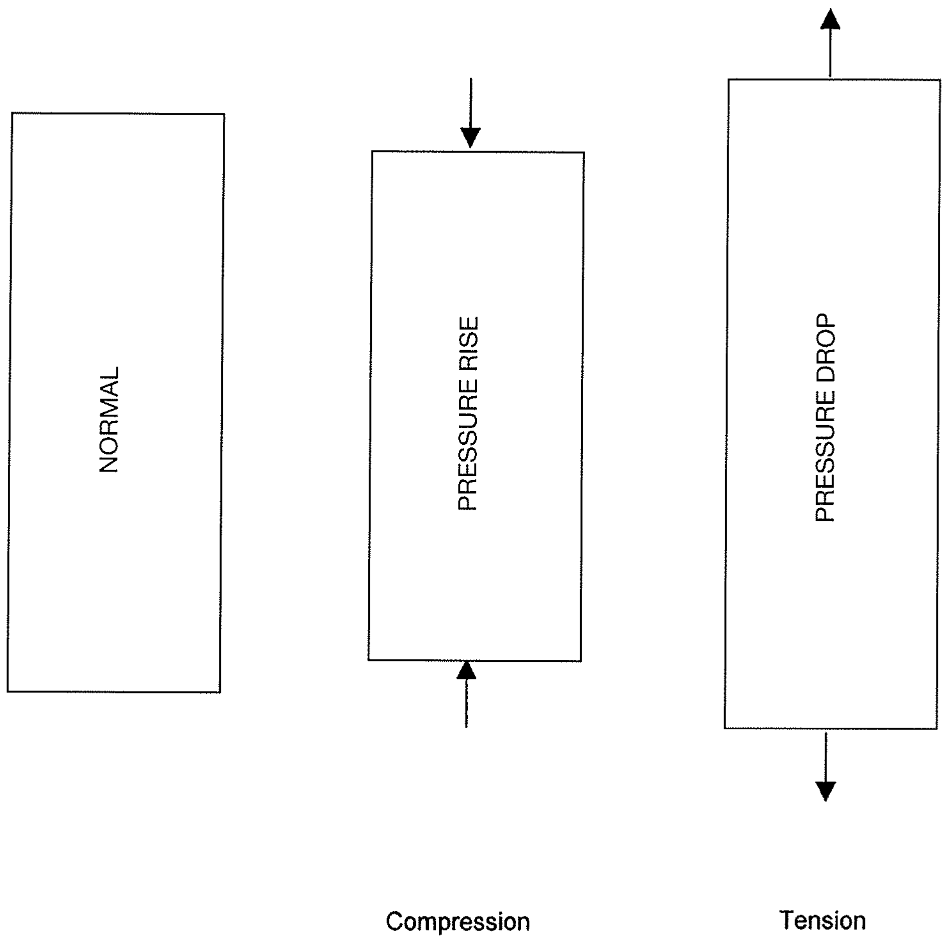



**FIGURE 7-5** Cylinders.

Lamb waves, like surface waves, propagate parallel to the test surface and have a particle motion that is elliptical. They occur when the thickness of the test material is only a few wavelengths at the test frequency and where the test piece is of uniform thickness. In other words, they can exist best in plate, tube, and wire.

Finally, there is a special type of compression wave that skims along the surface rather like a surface wave and is called a creeping or lateral wave. Its use is described under TOFD techniques.

The four main modes of propagation are the compression wave, the shear wave, and the surface wave. Each of these has an alternative name that is sometimes used. These alternative names are:

Compression waves are sometimes called longitudinal waves

Shear waves are sometimes called transverse waves

Surface waves are sometimes called Rayleigh waves

Lamb waves are sometimes called plate waves

## Properties of Sound Waves

### *Velocity*

Sound travels at different speeds through different materials. This is noticeable when, for example, a railroad worker is observed from a distance striking a rail with a hammer. Since the speed of light is much faster than that of sound, the observer first sees the hammer strike the rail. If the observer is standing close to the rail, he or she next hears the sound of the blow coming from the rail. Finally, the airborne sound resulting from the blow is heard.

This shows us that the speed of sound in the rail is faster than the speed of sound in air. It is true that sound travels faster in liquids than in gasses and faster in metals than in liquids. However, it is also true that sound travels at different speeds in different metals. There is a distinct speed of sound for each material. In ultrasonics, this is called the velocity of sound for that material. This being so, it would be useful to have an understanding of the reasons for the difference.

### *Factors Affecting Velocity*

The two main factors affecting velocity are the density and the elasticity of the material. To grasp a logical explanation for this, imagine that the molecules of any material are balls whose weight is analogous to the material density. So for lead, the balls would be heavy; they would be lighter for aluminum, and featherweight for air. Also imagine that these balls are joined together by springs representing the elasticity, or strength, of the material. For steel, the springs would be strong; they would be weaker for lead, and very weak for air. With these two concepts in mind, the scene is set.

The speed with which sound propagates through a material depends on how quickly one ball can get its neighbor to take up the vibration; in other words, to pass on the message. To get its neighbor moving, it has to overcome the inertia of that neighbor. Suppose two balls of a given weight are joined by strong springs and two more of equal weight are joined by weak springs. If the first balls in each pair are moved simultaneously, the ball facing the strong spring will quickly build up enough force to overcome the inertia of its neighbor. On the other hand, the ball facing the weak spring will have to move further and thus take longer to build up the equivalent force. From this, the logic tells us that it is reasonable to expect materials with a high value for Young's modulus of elasticity to have a high velocity of sound, as is the case, for example, with steel.

Consider two more pairs of balls. This time the springs are all the same strength but one pair of balls is light and the other heavy. If, again, the first balls in each pair are moved simultaneously, the lightweight pair quickly exchange messages but this takes longer in the heavy pair. Again, the general rule is indicated: the higher the density of a material, the lower the velocity of sound. Lead, for example, has a lower velocity than steel.

Density and elasticity are the dominant factors affecting velocity, but there is another one that plays a relatively minor, but nonetheless significant, role, and it is called Poisson's ratio. It is easy to see that when an elastic band is stretched it also gets thinner. The more it is stretched, the thinner it gets. Poisson's ratio relates the thinning to the stretching and can be calculated by dividing the change in diameter of the elastic band by the change in length.

The velocity of the compression wave for a given material can be calculated from the equation

$$V_c = \sqrt{\frac{E}{\rho} \cdot \frac{1 - \sigma}{(1 + \sigma)(1 - 2\sigma)}}$$

where
$V_c$ = compression wave velocity
$E$ = Young's modulus of elasticity
$\rho$ = material density
$\sigma$ = Poisson's ratio

Shear waves are able to exist in solids and they do not travel at the same velocity as the compression wave in a given material. This is because it is the modulus of rigidity, rather than Young's modulus, that dictates the velocity, and the modulus of rigidity is lower than the modulus of elasticity. This means that the shear wave velocity is always slower than the compression wave velocity in a material. As a rule of thumb, the shear wave velocity is roughly half the compression wave velocity. The velocity can be calculated from

$$V_s = \sqrt{\frac{E}{\rho} \cdot \frac{1}{2(1+\sigma)}} \qquad \text{or alternatively} \qquad V_s = \sqrt{\frac{G}{\rho}}$$

where
$V_s$ = shear wave velocity
$G$ = modulus of rigidity
$\rho$ = material density
$\sigma$ = Poisson's ratio

Surface (Rayleigh) waves also have their own particular velocity, which is generally taken to be approximately 90% of the shear wave velocity.

Although the velocity for each of these modes of propagation can be calculated, it requires a precise knowledge of all the parameters, and these are not usually available to the ultrasonic practitioner. Parameters such as density and strength vary with alloying, heat treatment, casting, rolling, and forging processes, all of which make it difficult to know that the correct values are being used. Instead, it is more normal to carry out a routine called "calibration" during the setting up procedure, for an ultrasonic inspection. In the calibration procedure the flaw detector timebase is adjusted to give a convenient scale against a calibration sample of known thickness and made of the same material as the work to be tested.

### *Wavelength*

The distinction between the oscillating motion of the particles making up a solid, liquid, or gas and the velocity of the sound moving through the substance has already been made. As the particles are completing each cycle of their vibration, the sound wave is moving on in the direction of propagation at the characteristic velocity for that material. It follows that during the time taken to complete one cycle of vibration, the sound wave will move a certain distance depending on the velocity in that material. For gasses with low velocities, that distance is small compared to the distance in metals, which have high velocities. This distance for a given material and sound vibration frequency is called the wavelength.

Wavelength is given the Greek symbol λ (lambda) and for any material and sound frequency, can be calculated from the equation

$$\lambda = \frac{V}{f}$$

where
$\lambda$ = wavelength

$V$ = Velocity
$f$ = frequency

*Example 1.* Calculate the wavelength of a 5 MHz compression wave in steel, given that the velocity of sound in mild steel is 5960 meters per second (m/sec).

$$\lambda = \frac{V}{f}$$

$$\therefore \lambda = \frac{5960}{5{,}000{,}000} \text{ meters (m)}$$

$$\therefore \lambda = 0{\cdot}00192 \text{ m}$$

It would be better to express such a small distance in millimeters (mm) by multiplying the answer by 1000:

$$\lambda = 0.00192 \times 1000$$

$$\lambda = 1.192 \text{ mm}$$

At ultrasonic frequencies, the wavelength of sound in metals is relatively short and so it is usual to express the wavelength in millimeters. This is done at the start of the calculation by changing the velocity from meters to millimeters per second by multiplying by 1000.

*Example 2.* Calculate the wavelength of a 5 MHz compression wave in aluminum, given that the velocity is 0.252″/μ sec (6400 m/sec).

$$\lambda = \frac{V}{f}$$

$$\lambda = \frac{6{,}400 \times 1000}{5{,}000{,}000}$$

$$\lambda = 1.28 \text{ mm}$$

Wavelength is useful in many ways in ultrasonic flaw detection. In the first place, the smallest reflector that can be detected must have a major dimension of at least half a wavelength at the test frequency. If the critical size of the discontinuity that must be detected is known, the knowledge helps with selection of an appropriate test frequency. Wavelength is also used in the calculation of the sound beam shape and the near-field distance. The significance of these will be discussed later.

***Reflection***

The boundary between one medium and another—for instance, steel to air at the far side of a steel plate—is called an "interface." At an interface, a proportion of the sound may be transmitted to the next medium and the remainder reflected back to the first medium. In the case of a steel to air interface, almost all the energy reflects and virtually none goes into the air. If the steel is under water, so that there is a steel to water interface, 88% of the energy is reflected and 12% is transmitted into the water. The proportions that will be reflected or transmitted depend upon the properties of the materials on either side of the interface.

In order to understand this phenomenon, consider again the balls and springs. This time, consider a train of tightly packed heavy balls joined by strong springs, representing steel. The steel train leads to a train of widely spaced light balls attached to weak springs, representing the air. If a compression wave is initiated at the start of the steel train, the message is passed on along the train, with each ball limited in its oscillation by its neighbor until the last ball in the steel train. It finds more space and lighter neighbors and is able to move a greater distance from its mean position in the direction of the air. In doing this, it stretches the spring joining it to the last but one steel ball until the spring tension arrests its motion. The last steel ball now starts to accelerate back into position and is so fast when it gets there that it overshoots, crashes into the last but one ball and starts a compression wave going back the other way. This is a reflection.

The two big differences between air and steel are density and elasticity, and these are the factors that decide how much energy is reflected and how much is transmitted at the interface. Each material is given a factor that is used to calculate reflectivity at an interface. This factor is called the "acoustic impedance" and given the symbol $Z$. Acoustic impedance is the product of density and velocity for that material. Stated mathematically

$$Z = \rho \times V$$

where
$Z$ = acoustic impedance
$\rho$ = material density
$V$ = material velocity

To calculate the percentage of energy reflected at an interface between any two materials, the following formula is used:

$$\text{Reflected energy} = \left(\frac{Z_1 - Z_2}{Z_1 + Z_2}\right)^2 \times 100\%$$

Where $Z_1$ & $Z_2$ are the acoustic impedance of the materials on either side of the interface.

*Example 3.* Calculate the percentage energy reflected at a steel to water interface, given that the acoustic impedance for steel is 46.7 and that for water is 1.48.

$$\text{Reflected energy} = \left(\frac{Z_1 - Z_2}{Z_1 + Z_2}\right)^2 \times 100\%$$

$$\text{Reflected energy} = \left(\frac{46.7 - 1.48}{46.7 + 1.48}\right)^2 \times 100\%$$

$$\text{Reflected energy} = \left(\frac{45.22}{48.18}\right)^2 \times 100\%$$

$$\text{Reflected energy} = (0.93856)^2 \times 100\%$$

$$\text{Reflected energy} = 0.8809 \times 100\%$$

$$\text{Reflected energy} = 88.09\%$$

*Example 4.* Calculate the percentage energy reflected at a steel to air interface, given that the acoustic impedance for steel is 46.7 and that for air is 0.0004.

$$\text{Reflected energy} = \left(\frac{Z_1 - Z_2}{Z_1 + Z_2}\right)^2 \times 100\%$$

$$\text{Reflected energy} = \left(\frac{46.7 - 0.0004}{46.7 + 0.0004}\right)^2 \times 100\%$$

$$\text{Reflected energy} = \left(\frac{46.6996}{46.7004}\right)^2 \times 100\%$$

$$\text{Reflected energy} = (0.99998)^2 \times 100\%$$

$$\text{Reflected energy} = 1.0000 \times 100\% \textit{ approx}$$

$$\text{Reflected energy} = 100\%$$

When a beam of sound traveling through a metal sample encounters a discontinuity such as a crack, lamination, void or nonmetallic inclusion there is an interface. On one side is the sound metal and on the other the discontinuity. At this interface, some energy will be reflected and some transmitted. If the discontinuity side of the interface is air then the reflection is total; but even for a nonmetallic inclusion, most of the energy will be reflected. This property of sound waves allows for the detection of discontinuities in materials.

### *Couplant*

The property of reflection can also be a problem because if a transducer is simply placed on a part there must be an air gap, however small. But a solid to air interface creates 100% reflection, so the sound goes straight back into the transducer without transmitting into the metal! To overcome this problem there has to be some way to exclude the air using a medium that will match the acoustic impedance of the transducer to the metal. Since this medium must also allow the transducer to be scanned over the surface of the metal it needs to be a liquid, grease, or paste. Such a substance is called a "couplant."

There are many suitable substances that can be used as couplants, the main criteria being the best possible match and no adverse chemical reaction between the couplant and the metal. Most couplants only allow limited matching because liquids in general have a low acoustic impedance. In immersion testing, the couplant is usually water, which only allows about 12% of the energy into the steel, and, of course, only 12% of any echoes to pass back across the interface and back to the receiving transducer. Most couplants permit between 10% and 15% sound transmission. The best of these is glycerin at around 15%. Commonly used couplants are:

- Water
- Kerosene
- Oil
- Grease
- Wallpaper paste
- Glycerin
- Special gels designed for the purpose

### *Refraction and Mode Conversion*

So far only sound entering the metal perpendicular to the surface has been discussed. When the sound is introduced at an angle to the surface called the "angle of incidence," several things may happen, depending on the actual angle of incidence.

Figure 7-6 shows a beam of sound traveling toward an interface at an angle of incidence $i°$ to the perpendicular that is usually called the "normal." The velocity of sound in Medium 1 is $V_1$ and in Medium 2, on the other side of the interface, the velocity is $V_2$.

Assume for this example that $V_1$ is slower than $V_2$, as would be the case if Medium 1 had been water and Medium 2 steel. As the beam travels toward the interface, the whole beam is moving at the same speed until the left-hand edge first strikes the interface. The moment the edge of the beam reaches Medium 2, it speeds up. But the sound still in Medium 1 stays at the old speed. Gradually, as the entire wave-front sweeps the interface, it speeds up until at last the right hand edge passes across the interface and the entire beam travels on in Medium 2 at the new speed.

During this transition, the beam slues around to a new angle in Medium 2 called the "angle of refraction." It is a bit like somebody passing through a doorway, catching their pocket on the door handle, and being diverted by the braking action on that side. The reason for this refraction is the velocity difference on either side of the interface. Snell's law allows the new angle to be calculated if the two velocities and the angle of incidence are known. Snell's law states that the sine of the angle of incidence divided by the velocity in Medium 1 equals the sine of the angle of refraction divided by the velocity in Medium 2. Stated mathematically:

$$\frac{\sin i^\circ}{V_1} = \frac{\sin R^\circ}{V_2}$$

where

$\sin i^\circ$ = The sine of the angle of incidence
$\sin R^\circ$ = The sine of the angle of refraction
$V_1$ = The velocity in Medium 1
$V_2$ = The velocity in Medium 2

Refraction refers to the transmitted portion of the sound energy at the interface; the proportion of energy reflected is the same as before but the reflected energy leaves the interface at an angle of reflection equal to the angle of incidence, as shown in Figure 7-7. This diagram shows the angle of incidence (i°), angle of reflection (r°), and angle of refraction (R°).

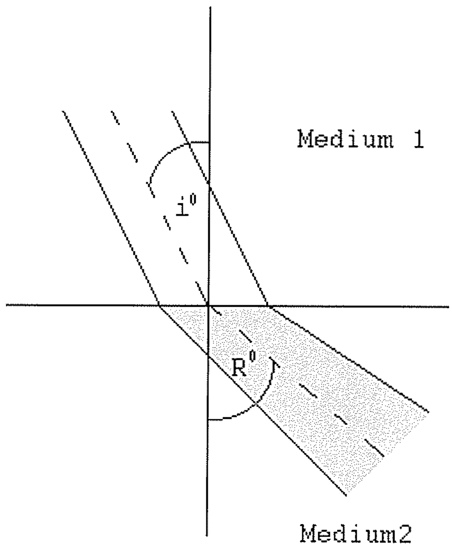



**FIGURE 7-6** Refraction.

*Example 5.* Calculate the angle of refraction in steel for an incident angle in water of 10° given that the compression wave velocity of sound in water is 1480 m/sec and in steel 5960 m/sec.

$$\frac{\sin 10°}{1480} = \frac{\sin R°}{5960}$$

$$\frac{5960 \times \sin 10°}{1480} = \sin R°$$

$$\frac{5960 \times 0.1736}{1480} = \sin R°$$

$$0.6993 = \sin R°$$

$$R° = 44.37°$$

The relationship between velocity and refraction can be seen in the above example, the compression wave velocity of sound in steel is roughly four times that in water, and the refracted compression wave angle is roughly four times the incident angle.

***Mode Conversion***

As the beam of sound is introduced at an angle of incidence to a solid, another phenomenon begins to arise, and that is mode conversion. Although the incident beam is a compression wave, a refracted shear wave begins to develop in the solid as the sound crosses the interface, in addition to the refracted compression wave. For small angles of incidence, the amplitude of the shear wave is small and can be ignored, but as the angle of incidence increases, so does the amplitude of the shear wave. Eventually, both the shear wave and the compression wave are about equal in amplitude. Snell's law shows us that the two modes will not refract through the same angle because the velocity of the shear wave is less than the compression wave.

*Example 6.* Calculate the angle of refraction of the shear wave in steel for an incident compression wave of 10° in water, given that the shear wave velocity in steel is 3240 m/sec.

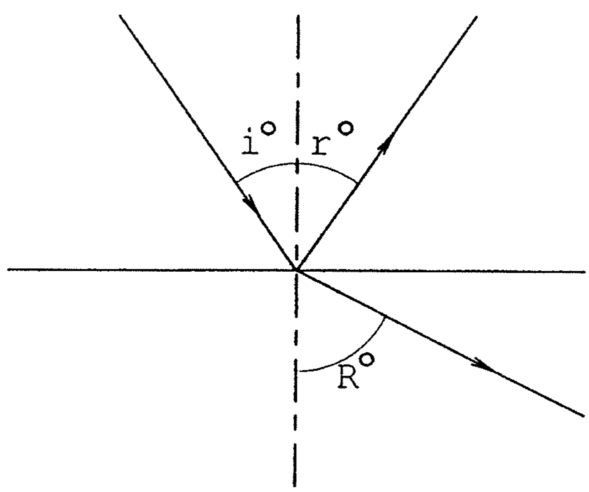



**FIGURE 7-7** Snell's law.

$$\frac{\sin i^\circ}{V_1} = \frac{\sin R^\circ}{V_2}$$

$$\frac{\sin 10^\circ}{1480} = \frac{\sin R^\circ}{3240}$$

$$\frac{3240 \times \sin 10^\circ}{1480} = \sin R^\circ$$

$$\frac{3240 \times 0.1736}{1480} = \sin R^\circ$$

$$0.3801 = \sin R^\circ$$

$$R^\circ = 22.34^\circ$$

The problem for ultrasonic flaw detection is immediately obvious—two beams traveling at different speeds and in different directions spells chaos! This was a problem that beset the early practitioners until Sproule came up with a solution in 1947. What he did was to get rid of one of the beams. He did this by increasing the angle of incidence until the refracted compression wave refracted to 90°. Any further increase in incident angle leaves only a refracted shear wave. The compression wave is said to have undergone total internal reflection in Medium 1.

The angle of incidence giving a 90° refracted angle for the compression wave is called the "first critical angle." The first critical angle for a water to steel interface is about 15°, and for Plexiglas (Lucite or Perspex) to steel the first critical angle is about 28°. Above the first critical angle of incidence, only a shear wave remains. By suitable choice of an incident angle above the first critical angle, a shear wave beam of any desired angle can be achieved. For immersion testing, the transducer is simply tilted through the calculated angle of incidence in water. For manual scanning, the transducer is mounted on a Plexiglas wedge angled to the desired incident angle.

If the incident angle is increased more and more beyond the first critical angle, eventually the shear wave will also be refracted to 90°. The angle of incidence to achieve this is called the "second critical angle." At the second critical angle, the shear wave undergoes another mode conversion; this time it becomes a surface (Rayleigh) wave, which is the contour-following wave. Any increase in angle of incidence beyond the second critical angle leaves no sound in Medium 2 at all; there is total internal reflection in Medium 1. For water to steel, the second critical angle is about 27°, and for Plexiglas to steel about 58°. Figure 7-8 shows the angle of refraction in steel obtained for increasing angles of incidence in Plexiglas at a Plexiglas to steel interface, and Figure 7-9 shows the same for a water to steel interface.

### *Reflective Mode Conversion*

Mode conversion also takes place inside a solid when an ultrasonic beam strikes a reflector at an angle of incidence other than perpendicular. Figure 7-10 shows a compression wave, C, striking a steel to air interface at an angle of incidence, i°, to the normal. The reflected compression wave, $C_r$, is at a reflected angle r° equal to the angle of incidence. However, there is also a mode converted shear wave, S, at an angle derived from Snell's Law, s°. In ultrasonic flaw detection, this mode conversion can cause confusion, depending on the relative amplitudes of the reflected compression wave, and the mode converted shear wave.

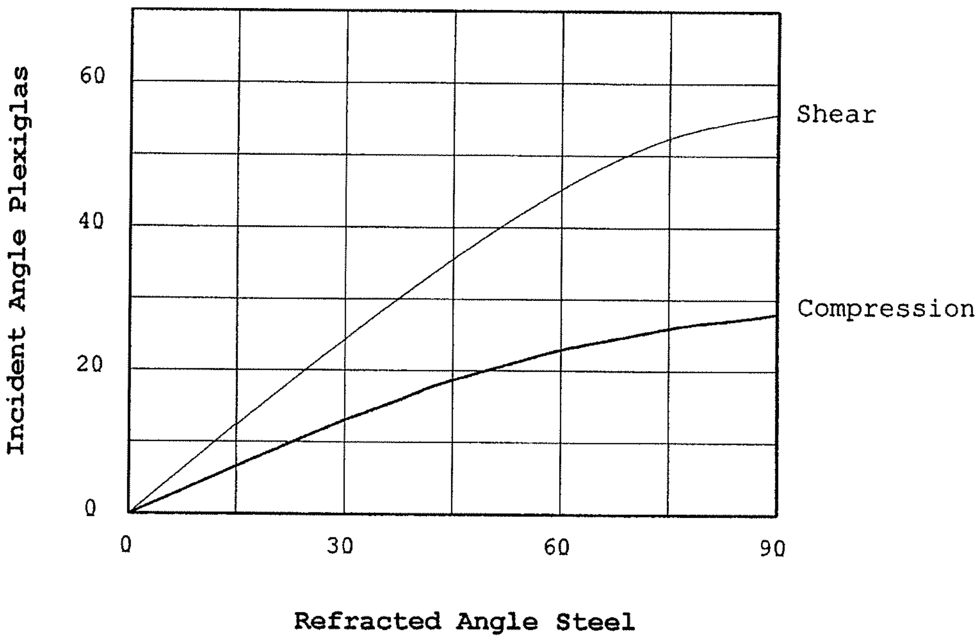



**FIGURE 7-8.**

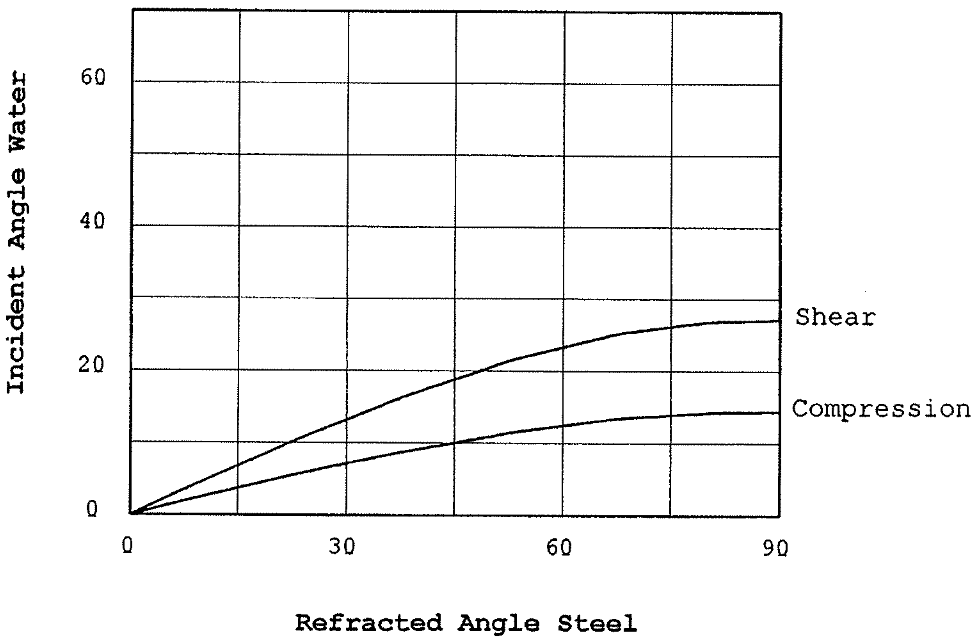



**FIGURE 7-9.**

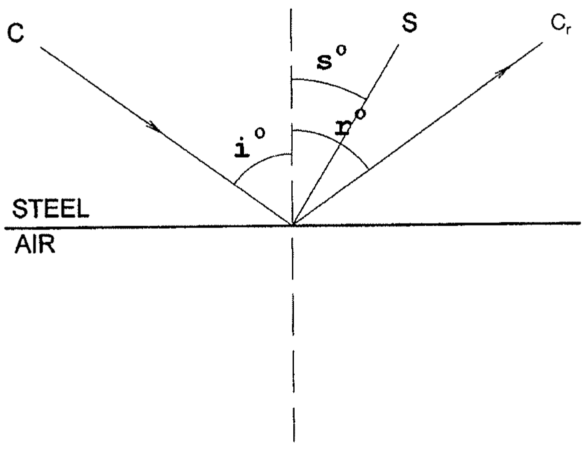



**FIGURE 7-10** Incident and reflected angles.

Figure 7-11 shows the relative amplitudes of the reflected compression wave and the mode conversion shear wave for increasing angles of incidence, of the compression wave at a steel to air interface. It can be seen that at low angles of incidence the shear wave is weak and can be ignored. At an angle of incidence of about 25°, the reflected compression wave and the shear wave are at the same amplitude, and at about 70°, the reflected compression wave is very weak whereas the shear wave is still very strong.

Figure 7-12 shows the angle of the mode-converted shear wave with respect to the normal for increasing angles of incidence for the compression wave at the steel to air interface. These two graphs show the strength and direction of the mode conversion. The practitioner may encounter these circumstances when carrying out a compression wave (straight beam) test if a reflecting surface is not parallel to the scanning surface.

Mode conversion can also take place when a shear wave meets a reflecting surface, as shown in Figure 7-13. In this case, the reflected shear wave is again at the same angle as the angle of incidence, but the mode-converted compression wave is at an angle $\alpha°$, which can be calculated from Snell's law.

Figure 7-14 shows the amplitudes of the reflected shear wave and the mode-converted compression wave relative to the amplitude of the incident shear wave.

Figure 7-15 shows the angle of the mode-converted compression wave. It can be seen from the graph that when a shear wave is incident to a reflecting surface at about 30°, less than 10% of the shear wave is reflected, but the mode-converted compression wave amplitude rises steeply, and is far greater than the reflected shear wave. Figure 7-15a shows that for this incident angle of shear wave, the mode-converted compression wave will be at an angle of about 65° to the normal. This situation will occur if a 60° angle beam transducer is chosen to examine a weld with a vertical fusion face, such as an electron-beam weld, or the root face of a double "V" weld preparation. As shown in Figure 7-15b, the shear wave will meet any vertical nonfusion at an incident angle of 30°, and the strong mode-converted wave will reach the transducer. The path taken by this wave and its velocity will cause confusion if the practitioner is unaware of the problem. Obviously, a 60° beam angle is a wrong choice in those circumstances.

### *Beam Characteristics*

Many of the illustrations so far used have treated the sound as if it were a single ray, but in fact, the sound propagates as a beam. Within the beam, intensity or amplitude of the

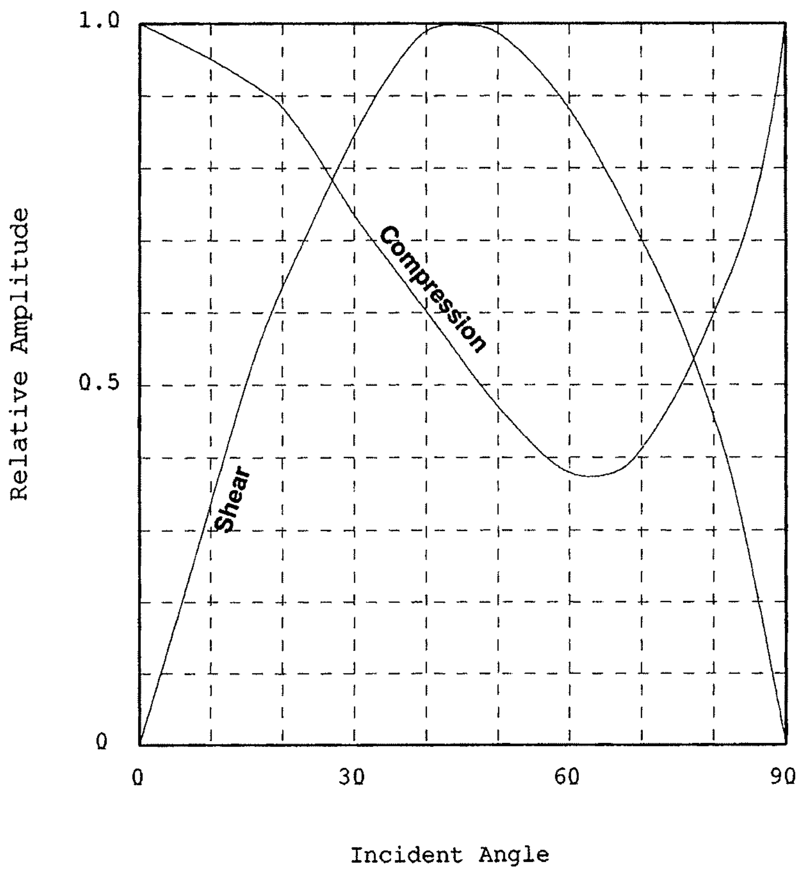



**FIGURE 7-11** Incident angle graph.

sound energy varies. The following paragraphs deal with those variations and with the shape of the beam. For convenience, the beam is divided into two distinct zones called the "near field" and the "far field." In these two fields, different mechanisms are at work to vary the sound intensity. The word used to describe what effectively is a gradual loss of sound energy is "attenuation." Attenuation is the combined effect of a number of parameters:

- Interference and diffraction effects
- Interference Absorption (friction and heat)
- Interference Scatter
- Interference Beam spread

*Interference and Diffraction Effects.* Huygens developed a convenient way of looking at wave energy propagating from a source. He said that a point source was rather like dropping a stone into a pond; the disturbance moves out as an expanding circle on the pond, but from a sound source the circle becomes an expanding sphere—it moves out in all directions. A sphere is a three dimensional object that is difficult to portray on a sheet of paper, so for this exercise a circle will have to do.

Figure 7-16 shows a point source surrounded by concentric circles representing successive pressure waves of sound frozen in time a short time after the sound starts. The

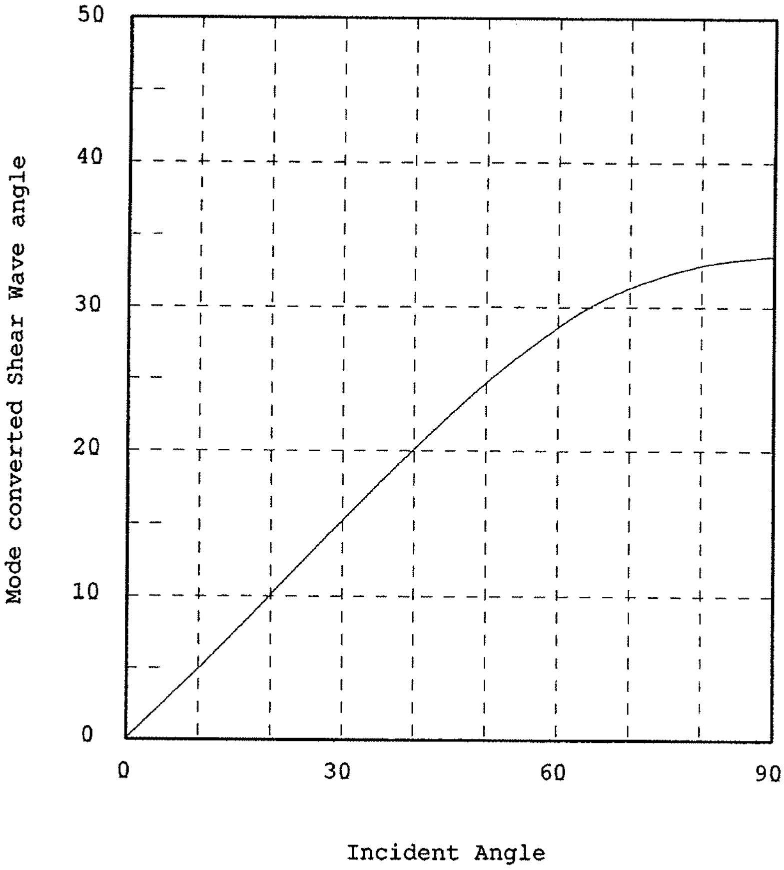



**FIGURE 7-12** Incident angle graph 2.

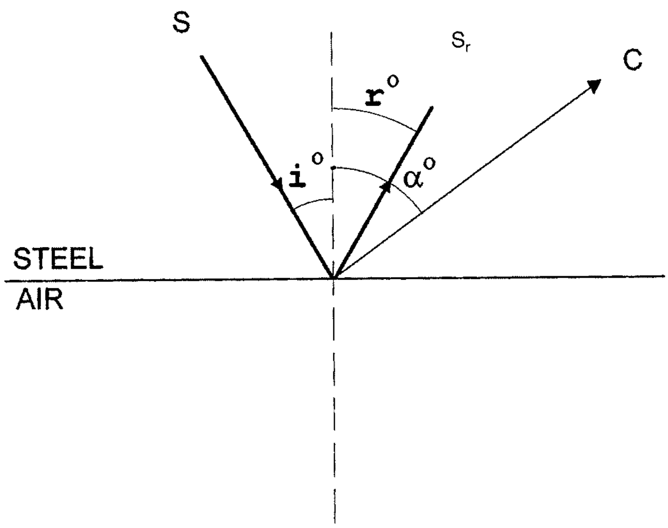



**FIGURE 7-13** Incident and reflected angles 2.

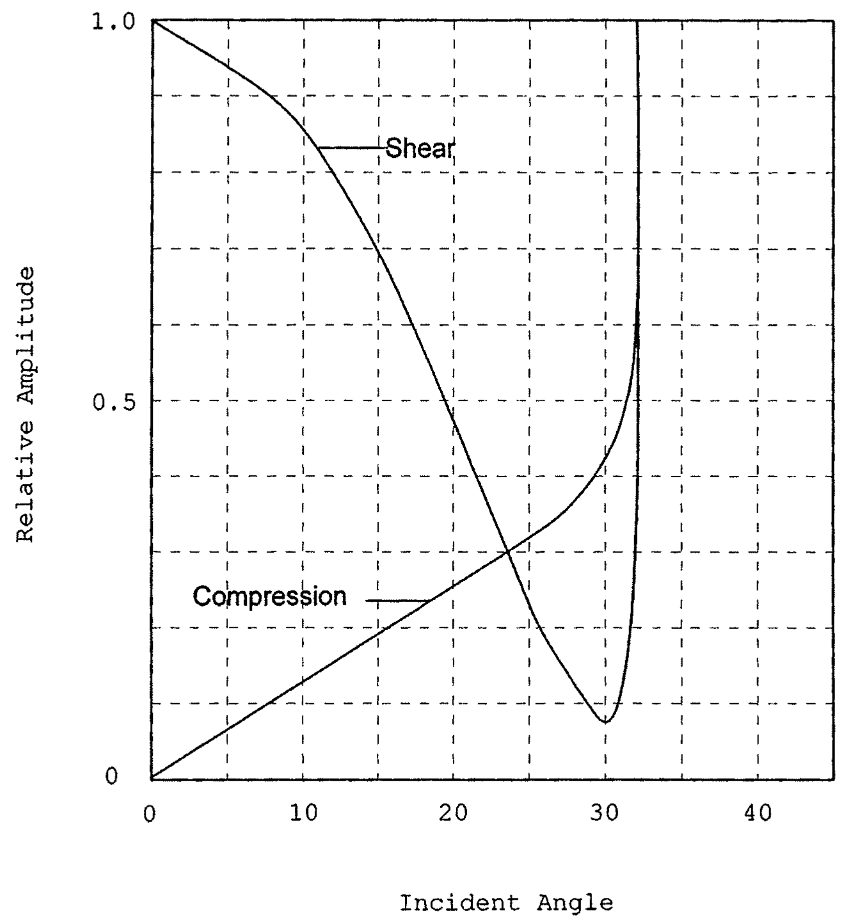
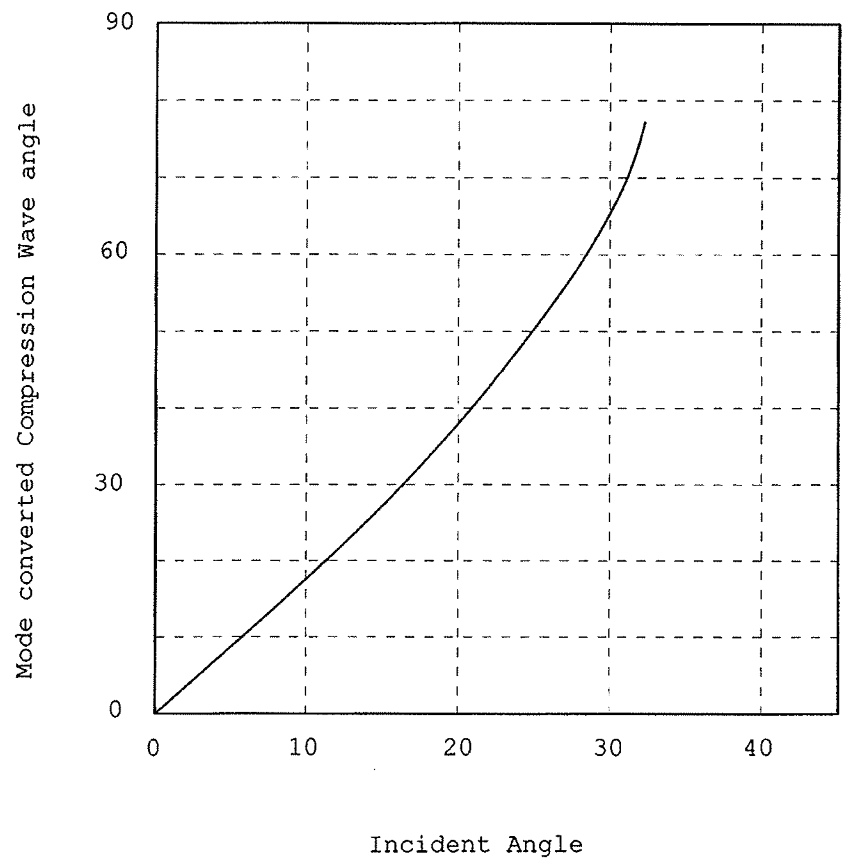


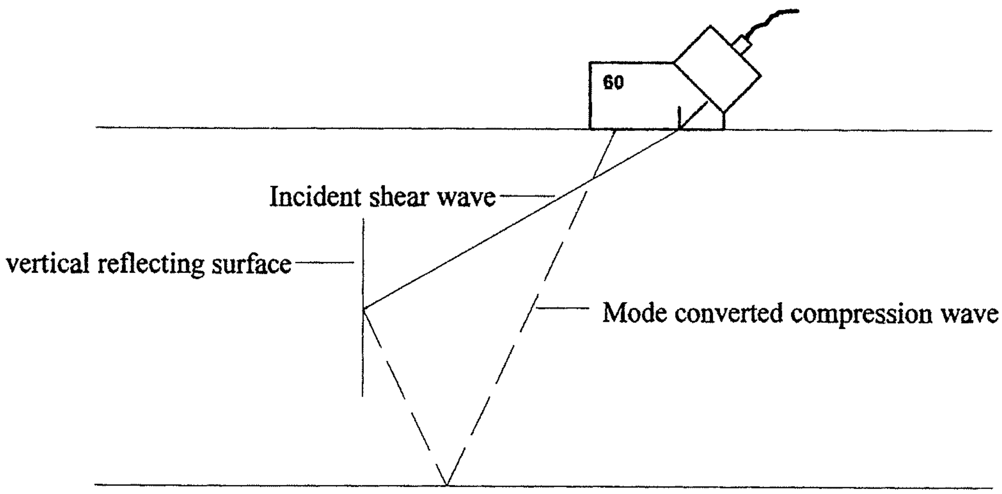

**FIGURE 7-14** Incident angle graph 3.

spaces between the circles represent the rarefaction part of each cycle of sound. If the pattern had been frozen a little later in time, the outer circle would have been bigger in diameter. The space between each circle represents one wavelength of the sound in the material. But an ultrasonic transducer is not a point source, it has a diameter and a surface area, all of which is active. This is called a "finite source" and Huygens said that this could be considered as being made up of an infinite number of point sources. Figure 7-17 shows a finite source with a few of these infinite point sources frozen a short time after the vibration has been started. It can be seen that the wave fronts from the point sources combine to make a united wave front as the "beam" propagates from the source. But notice how a little bit of the sound is lost around the edges of the source; it is said to "diffract" around the edges; this is one of the energy losses in the near field.

The next source of loss needs a little more explanation. Figure 7-18 shows a finite source again, but this time only the point sources in the center and at the edges are shown, for simplicity. In front of the source is a point "P," which is waiting for sound to arrive. In the diagram, the first pressure wave from the middle of the source has already arrived, but sound from the edges has some way to go yet. "P" is given a gentle nudge in the direction of propagation by the sound from the middle of the source. Figure 7-19 shows the situation a short time later, when the first pressure wave from the edges of the source arrive at "P."




(a)




(b)

**FIGURE 7-15** (a) Incident angle graph 4. (b) Mode converted beam.

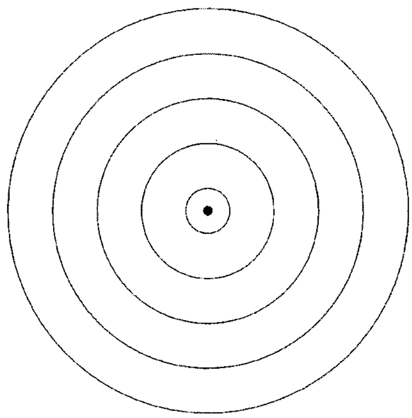


**FIGURE 7-16** Point source.

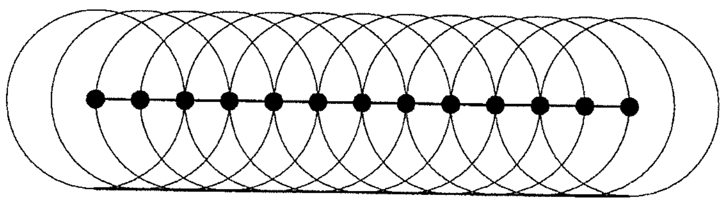


**FIGURE 7-17** Wavefront.

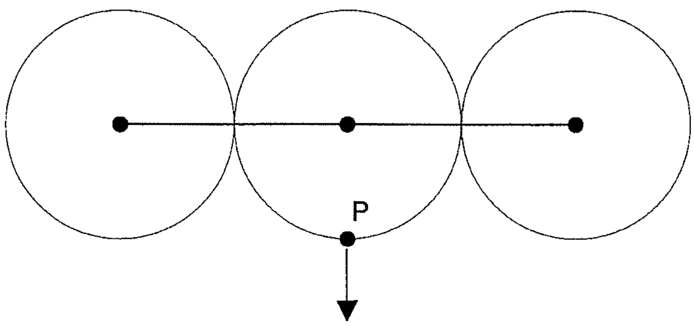



**FIGURE 7-18** Point source 2.

The result of these two nudges from either side is again to move "P" in the direction of propagation. But that is without considering what is arriving from the center of the source. In the illustration, the third pressure wave is arriving from the center, resulting in an extra-large nudge for "P," representing the combined nudges from the edges and middle. These three simultaneous nudges are called "constructive interference" because the end effect is a local increase in sound intensity. It has happened because 'P' is an exact number of wavelengths from both the center and the edges of the source for the frequency of the sound wave. A change in frequency or a shift in the position of "P" might result in the sound from the center and from the edges arriving at "P" out of phase, as shown in Figure 7-20.

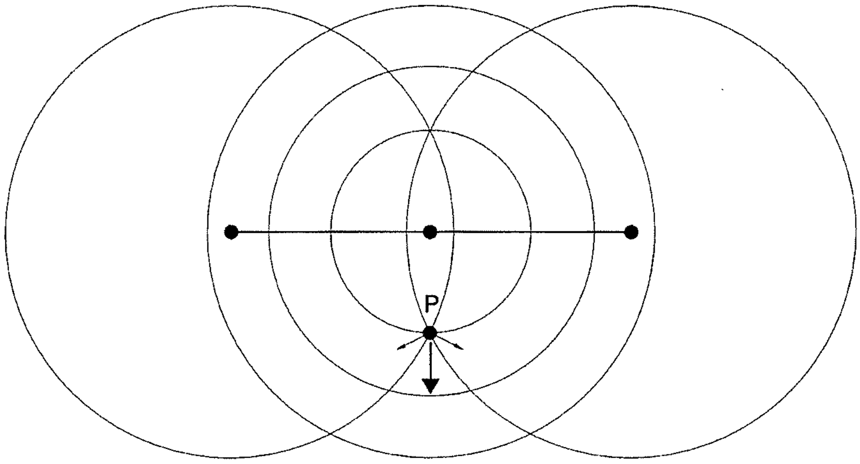



**FIGURE 7-19** Point source 3.

In this figure, the first pressure waves from the edges have arrived at 'P' as a rarefaction arrives from the center. Two forces are pushing and pulling at "P." This is called "destructive interference" and leads to a local reduction in sound energy. For an absolutely pure frequency continuous wave sound, the destruction could be total; in other words, there would be no sound at all at "P." The pulsed, broadband transducers used in ultrasonic flaw detection never quite cancel out.

The reason for this destructive interference is the difference in path length from 'P' to the center of the source and to the edges compared to the wavelength. This being so, eventually there will be a distance for "P" in front of the source where the path difference becomes significantly less than a wavelength (see Figure 7-21) and the interference effects cease. This distance is the end of the "near field." The near field distance, *NF*, can be calculated from

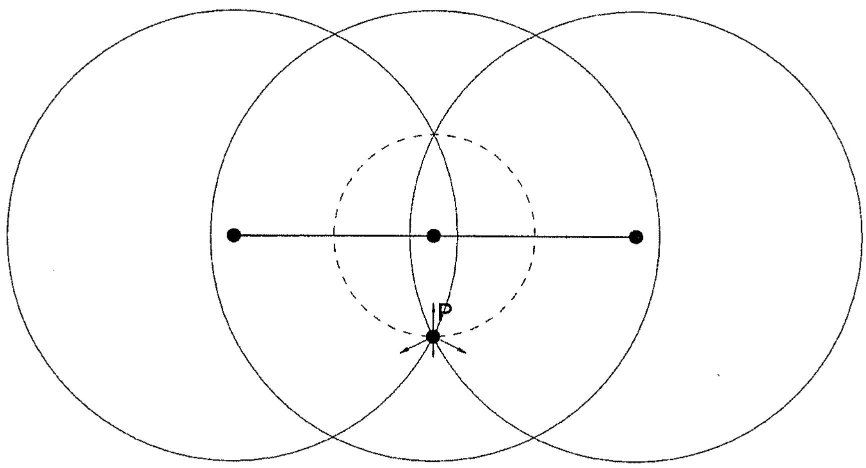



**FIGURE 7-20** Point source 4.

$$NF = \frac{D^2}{4 \times \lambda}$$

Alternatively, where the wavelength is unknown

$$NF = \frac{D^2 \times f}{4 \times V}$$

where:
$D$ = Transducer diameter
$F$ = Frequency
$V$ = Velocity

*Example 7.* Calculate the near field distance in steel for a 5 MHz compression wave when using a transducer that is 10 mm in diameter.

$$NF = \frac{D^2 \times f}{4 \times V}$$

$$NF = \frac{100 \times 5{,}000{,}000}{4 \times 5{,}960{,}000}$$

$$NF = \frac{5{,}000{,}000}{23{,}840}$$

$$NF = 20.97 \text{ mm}$$

***Absorption***
Sound propagates through the vibration of particles of a solid, liquid, or gas and the movement of those particles causes friction and absorbs some of the energy. The rate at which energy is absorbed depends on the material through which the sound is passing and the frequency of the sound. In general, the higher the frequency, the greater the absorption; or put another way, the lower the sound frequency, the further it penetrates into the material.

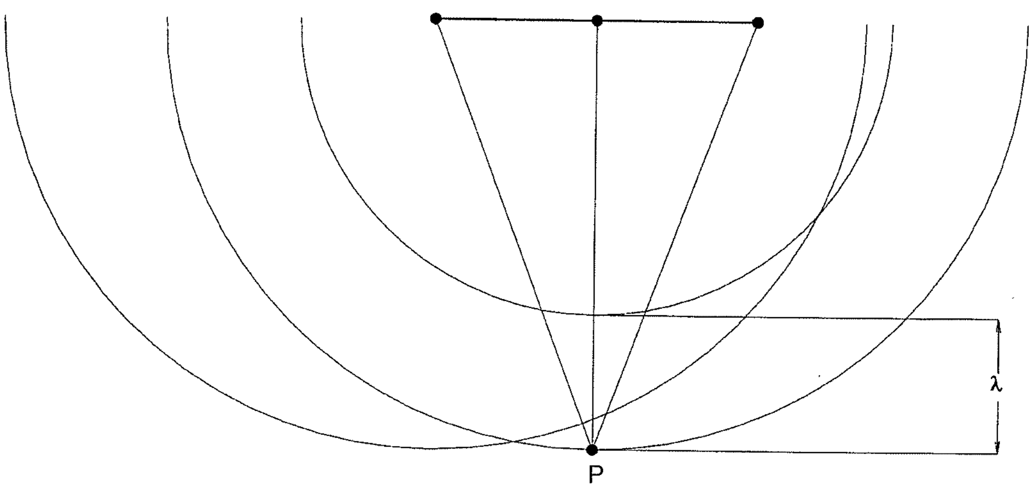



**FIGURE 7-21** Point source 5.

***Scatter***

Sound waves will reflect from interfaces within the material being tested, and grain boundaries in solids are interfaces that may be randomly orientated to the beam. This causes some of the sound to reflect in random directions or "scatter." Very fine-grained material causes very little scatter but coarse-grained material causes considerable scatter. Scattered energy that does not reach the receiver transducer is "lost" energy. Scattered energy that does reach the receiver is worse! It creates small signals across the timebase. This condition is called "noise," "grass," or "hash" and it tends to mask signals from discontinuities.

Both absorption and scatter exist as sources of lost energy in both the near field and the far field. Beam spread is the remaining cause of energy loss affecting the far field.

***Beam Spread***

In the near field, the beam is taken to be roughly cylindrical and the same diameter as the transducer crystal. Beyond the near field, in what is called the "far field," the beam spreads out like a cone. The angle of the cone, as shown in Figure 7-22, can be calculated from

$$\sin\frac{\theta}{2} = \frac{1.22\,\lambda}{D}$$

Where:
$\theta/2$ = Half angle of beam spread
$\lambda$ = Wavelength
$D$ = Transducer crystal diameter

The above equation includes the constant 1.22. This calculates the beam spread to the absolute limit of the beam where sound ceases to exist. This is not a practical limit for the ultrasonic practitioner because if sound doesn't exist, it can't be detected or measured. In practice, it is more usual to replace the constant 1.22 with either 0.56 or 1.08. The 0.56 value predicts the limits of the beam where the sound has dropped to one half of the intensity at the beam center. The 1.08 value defines the limits where the sound is one tenth of that at the beam center.

*Note:* The constants used above (e.g., 0.56, 1.08, and 1.22) are commonly used for calculation of theoretical beam shapes. If the shape of the beam is required for discontinuity sizing purposes, it is more practical to plot the beam shape using a special calibration block, rather than calculate the beam spread (see Intensity Drop Technique on page 7.92).

*Example 8.* Calculate the beam spread angle for a 5 MHz compression wave in steel when using a 10 mm diameter transducer ($\lambda$ = 1.192 mm).

$$\sin\frac{\theta}{2} = \frac{1.22\,\lambda}{D}$$

$$\sin\frac{\theta}{2} = \frac{1.22 \times 1.192}{10}$$

$$\sin\frac{\theta}{2} = \frac{1.45424}{10}$$

$$\sin\frac{\theta}{2} = 0.145424$$

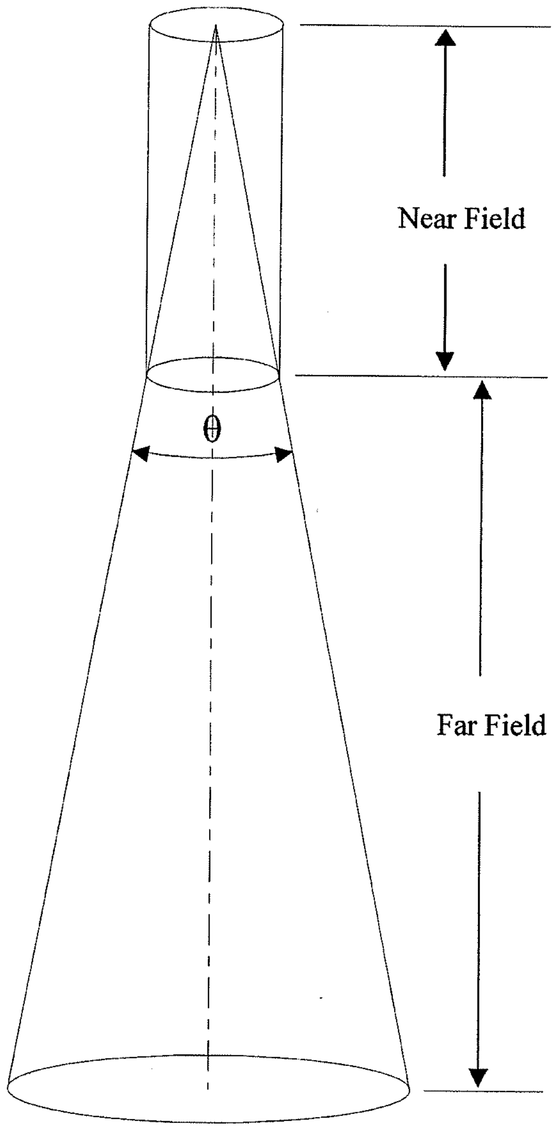



**FIGURE 7-22** Near and far fields.

$$\sin\frac{\theta}{2} = 8{\cdot}36°$$

$$\therefore \theta = 16{\cdot}72°$$

Figure 7-22 shows the overall beam shape, including the near-field portion.

Figure 7-23 shows the way in which amplitude changes along the beam center. In the near field, there are fluctuations in amplitude because of the interference effects. The last maximum amplitude marks the end of the near field and the beginning of the far field. This is called the $Y_0$ point. In the far field, the intensity can be seen to decay exponentially. From a practical point of view, Figure 7-23 implies that it is unreliable to use amplitude as an acceptance criterion for flaws detected in the near field. In some ap-

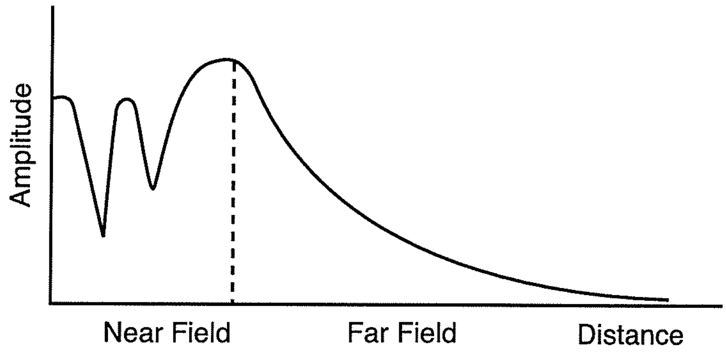



**FIGURE 7-23** Near and far field plots.

plications, this problem is overcome by using a delay line (a Plexiglas column) between the transducer and the work piece so that the near field is contained within the delay line.

***The Decibel System***

In order to compare data, some form of measurement is necessary; e.g., for length, the standard used is either inches or millimeters. Because electric power is proportional to the square of the voltage produced, it can be said that the voltages produced at the transducer are relative to the sound intensity. When measuring sound intensities, the unit of measurement is the bel. The bel is named for Alexander Graham Bell (1847–1922), inventor of the telephone.

The bel being a large unit of measurement, it needs to be broken down into smaller units. These units are called decibels or dB. "Deci" is a prefix that is borrowed from the Latin. It means "one tenth," so a decibel is one tenth of a Bel. The decibel is a unit used to express the intensity of sound energy. It is equal to twenty times the common logarithm of the ratio of the pressure produced by the sound energy, to a reference pressure. In other words, it is used to express the ratio of the magnitudes of two reflections (signals), each having different magnitudes, equal to twenty times the common logarithm of the voltage or current ratio. In practical terms, if there are two signals on the screen and the difference between these signals needs be known, it can be calculated. Alternatively, if a signal of known amplitude were to be reduced by a certain percentage, this reduction in gain can be calculated. For example, if a signal height of 100% is to be reduced to 10%, the reduction in gain can be calculated.

Of course, this can be measured by using the gain control or attenuator on an ultrasonic instrument, assuming that the instrument is linear in the vertical axis (see subsections on linearity in Section IV).

To calculate the difference between two signal amplitudes, the following formula is used:

$$dB = 20 \times \log\left(\frac{A_1}{A_2}\right)$$

where
$A_1$ = the first percent signal height
$A_2$ = the second percent signal height

**TABLE 7-1** Amplitude Ratios

| dB | Ratio |
|---|---|
| 3 | 1.41:1 |
| 6 | 2.00:1 |
| 9 | 2.82:1 |
| 12 | 3.98:1 |
| 14 | 5.01:1 |
| 20 | 10.00:1 |

*Example.* Two signals are noted on the screen. The first has an amplitude of 80% full screen height (FSH) and the second is noted to be 40% FSH. Calculate the difference in dB between the two.

$$dB = 20 \times \log\left(\frac{80}{40}\right)$$

$$\log 2 = 0.3010$$

$$20 \times 0.3010 = 6.02 \text{ dB}$$

The 80% signal is 6 dB greater than the 40% signal. It can be seen from this calculation that +6 dB is twice the amplitude.

Conversely, if the 80 and the 40 were inverted, that is, if the 40 were placed over the 80 and divided, the result would be 0.5. The log of 0.5 is the same as the log of 2; however, it calculates to negative or –3010. If this is multiplied by 20, it can be noted that –6 dB is one half the amplitude

This means that if the signal amplitude were to increase by 6 dB, the signal height would double. If the signal amplitude were decreased by 6 dB, the signal height would be half of its original amplitude.

The above formula applied to a first signal amplitude of 100% and second signal amplitude of 10% will result in a difference of 20 dB (see Table 7-1).

Having noted the significance of the pressure (amplitude) differences of reflections, it follows that the same rationale applies to the cross section of the ultrasonic energy "beam." If the beam intensity were shown as a slice across the beam section, it would appear as in Figure 7-24. Information regarding the actual pressure differences across the beam section is very important, particularly when using transducer movement to investigate or evaluate the characteristics of a discontinuity (see Section IV).

## *III. EQUIPMENT FOR ULTRASONIC APPLICATIONS*

As with computers, the technology concerning ultrasonic equipment and systems is becoming somewhat transitory. Ultrasonic systems are either battery operated portable units, multicomponent laboratory ultrasonic systems, or something in between. Whether they are based on modern digital technology or the fast disappearing analog original, "systems" (often defined as instrument plus transducer and cable) basically comprise the following components. The appropriate controls are shown in the ellipses in Figure 7-25.

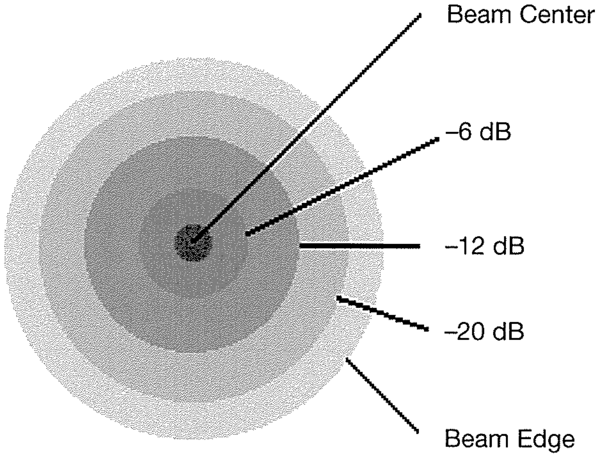



**FIGURE 7-24** Beam slice.

1. Transducer
2. Pulser (clock)
3. Receiver/amplifier
4. Display (screen)

To understand how a typical ultrasonic system operates, it is necessary to view one cycle of events, or one pulse. The sequence is as follows.

1. The *clock* signals the *pulser* to provide a short, high-voltage pulse to the *transducer* while simultaneously supplying a voltage to the *time-base trigger* module.
2. The *time-base trigger* starts the "spot" in the CRT on its journey across the screen.
3. The voltage pulse reaches the transducer and is converted into mechanical vibrations (see "piezoelectricity"), which enter the test piece. These vibrations (energy) now travel along their "sound path" through the test piece. All this time, the spot is moving horizontally across the CRT.
4. The energy in the test piece now reflects off the interface (back wall) back toward the transducer, where it is reconverted into a voltage. (The reconverted voltage is a fraction of its original value.)
5. This voltage is now received and amplified by the *receiver/amplifier*.
6. The amplified voltage is sent to the "vertical (*Y* axis) plates" (top and bottom) in the CRT. At this time, the upper *Y* axis plate attracts the spot upward. This motion produces the "signal" on the screen that signifies the time that the energy has taken to make the round trip through the test piece, from the moment the energy leaves the transducer until it is received by the transducer. The spot is set to start its trip at the time the energy enters the test piece. This is manually adjusted by using the *delay* or *zero* control. This step is particularly necessary when using a Plexiglas delay line (see Glossary of Terms).
7. The same "packet" of returning energy has by this time reflected down off the test piece's top interface and now makes a second trip down through the test piece. (The spot continues its horizontal journey across the screen.) The energy reflects once more off the back wall interface and returns again to be received and amplified. The amplifier once again sends the voltage to the *Y* axis plates. The spot is again drawn up toward the upper *Y* axis plate, this time at a "later" position on the time base. This is a "repeat signal" (multi-

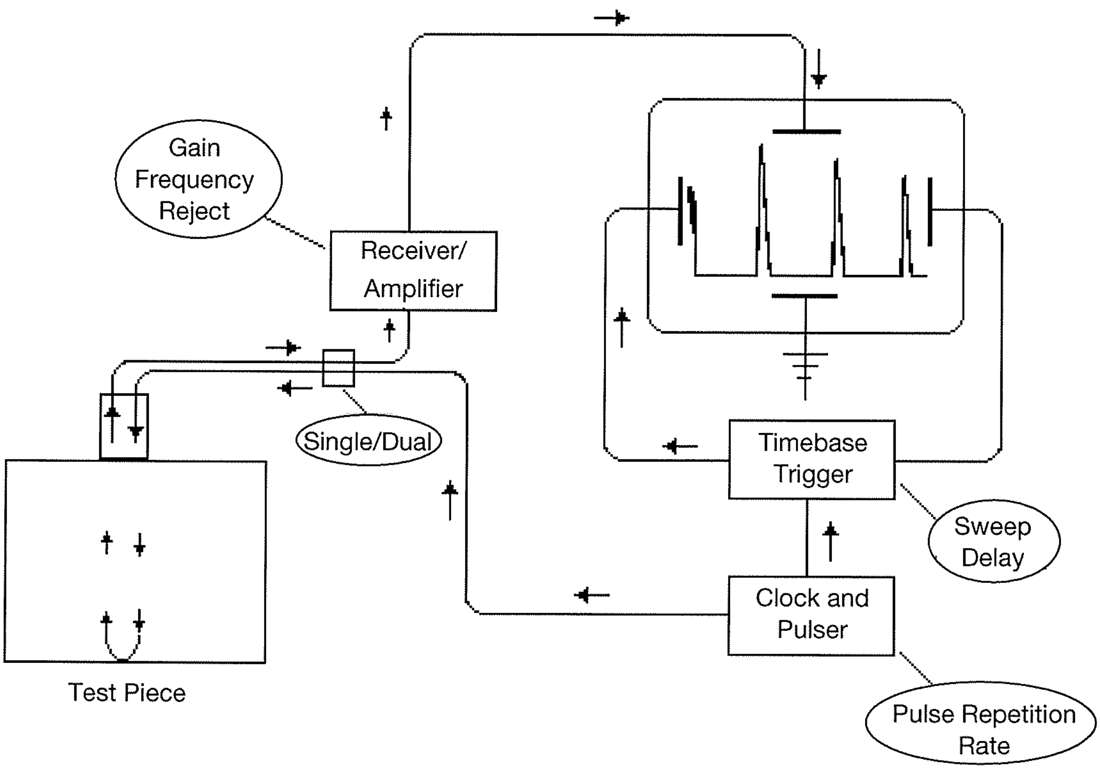



**FIGURE 7-25** Block diagram, analog.

ple) that is lower in amplitude because of factors such as attenuation and other losses. The spot is then released to continue its journey across the screen, and the above sequence repeats again and again until the energy in the test piece has been attenuated. The display will show multiple repeat signals, as many as are available in the calibrated time base and according to the amount of amplification (gain) selected. For example, if the screen is calibrated for 5 inches of steel and the test piece is 1 inch thick, there will be five signals on the screen, representing five "round trips" of 1 inch thickness. (The energy has, in fact, traveled two inches each trip, one forward and one back, but it is displayed as a series of 1 inch trips on the screen.)

8. The clock now sets off the pulser a second time and the next pulse is produced. The complete scenario is repeated over again, *n* number of times per second. The *n* number of pulses per second is referred to as the *pulse repetition frequency* (PRF) or the *pulse repetition rate* (PRR).

## Digital Instrumentation

In principle, the above also applies to digital flaw detection instrument (see Figure 7-26).

### *The Controls and Their Functions*

Instrumentation varies by manufacturer; however, there are three controls that are common to most ultrasonic flaw detection equipment. These controls are sweep (range), delay, and gain.

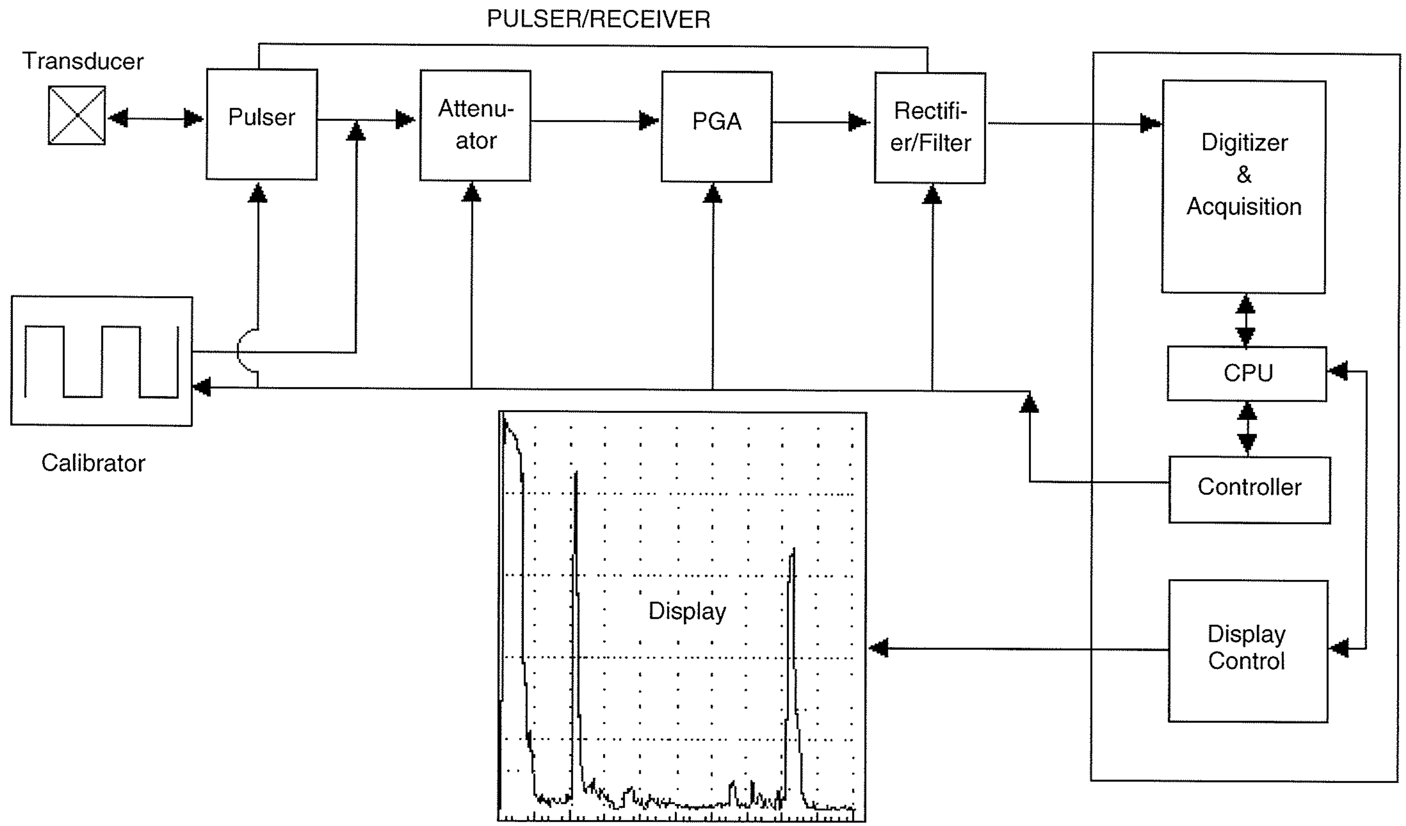



**FIGURE 7-26** Block diagram, digital.

*Sweep (Range—Coarse and Fine).* These controls usually exist on analog instruments. On digital units, the controls are usually combined into one, designated as "range." The function of this control is to adjust the speed of the screen spot in order to accommodate displays of varying sound paths. The spot will move slower when the screen is to represent a long sound path, e.g., displaying a time base for a long shaft of steel (slow time base). Conversely, the spot will move very fast across the screen when displaying a few millimeters of steel across the full screen (fast time base).

*Delay (Zero).* This control delays the spot from beginning its journey across the screen. For example, when using a dual transducer or a transducer with a Plexiglas wedge (standoff or delay line), the start time of the spot has to be delayed for an instant to allow the sound to travel through the Plexiglas wedge before it enters the test piece. The display need only show the area from the top of the test piece and onward. Displaying the wedge on the screen is unnecessary and also confusing. When calibrating the system, test piece material zero should appear at screen zero. The delay (zero) control will be used to accomplish this.

*Gain (Attenuator).* Because everything is relative to something else, and to facilitate the ability to make decisions about the nature of the signals that are observed on the CRT when conducting an ultrasonic examination, certain comparisons need to be made. The meaning of a signal on the screen is rather limited unless it is related to something tangible such as another signal from a known reflector.

If the signal from a reflector in a test piece is compared with a signal from a known reference reflector, theoretically (all things being equal), the size of the reflectors can be compared. [Note: signal amplitude in itself does not necessarily indicate reflector (discontinuity) size.]

If accept or reject decisions are made based on signal amplitude, consideration of a signal that saturates the screen may be impossible. That is because it exceeds 100% full screen height (FSH). Since the signal is above the viewable screen, it is impossible to make any comparisons unless the signal height is adjusted so that the top can be seen. If the instrument gain were to be adjusted so as to lower this signal until it becomes the same height as our reference signal, the actual reference signal height may be reduced so much that it cannot be quantified; or, worst case, it may be so low in amplitude as not to be visible at all, so signal amplitudes can not be compared.

The solution to this problem is to employ a "volume" control, much the same as those found in stereo systems. This device is known as either a "calibrated gain control" or, in some cases, an "attenuator." The difference between the two devices is their functionality. This is discussed below. If a numeric value could be attributed to the amount that the signal amplitude (gain) is adjusted, numeric signal height comparisons could be conducted.

Gain controls and attenuator controls operate using similar circuitry. The difference to the user is that when using an attenuator, increasing the attenuation results in diminished signal height. When using the calibrated gain control, increasing the gain level increases the signal height. The differences in signal height and variations in gain need to be known in order to obtain accurate data.

Other may vary between instruments by respective manufacturers; however, the following features are generally common to most:

*Single and Dual Transducer Selection.* This switch isolates the transmitter side of the circuit from the amplifier. It allows the reception of voltage generated solely by the receiver side of a dual transducer or from a transducer used as a receiver in a "pitch catch" or through transmission mode. In the case of a single transducer, the first voltage to reach the amplifier is from the pulser and transducer (in combination). This produces the signal on the left-hand side of the screen that is often referred to as the "initial pulse" or "main bang." It is obvious that the initial pulse "blinds" the display for an area under the transducer (front surface) in the test piece. Reflectors close to the test surface will not be resolved because they occur within the time of the initial pulse. Using the "dual" setting with a dual transducer essentially eliminates the "initial pulse" signal from the left hand side of the screen, thus increasing the available test time close to the front interface (top) of the test piece.

*Frequency Selection.* Transducers operate at a predetermined nominal frequency based on their thickness. The transducer oscillates at its resonant frequency, but it also produces other frequencies, some higher and some lower than the nominal center frequency. It is sometimes necessary to filter out undesirable frequencies as they can produce low-level noise. This can adversely affect (reduce) the "signal-to-noise" ratio. It is important that the signal from a reflector be visible above the background noise caused by material grain and other factors such as instrument circuit noise. To this end, some ultrasonic instrumentation is designed so that individual frequencies are user-selectable. Other instrumentation is designed so as to accept a range of frequencies that are not user-selectable. These are classified as having either narrow band receivers or broad band receivers. The narrow band receiver is usually user-selectable. In this case, the user selects the frequency nearest to that of the transducer being used. The effect is that the receiver processes only the frequency selected, within a certain "bandwidth"; e.g., a 5 MHz selector may be receptive to energy from 4 MHz to 6 MHz, depending on the design specifications. This circuitry will filter out frequencies outside this bandwidth. Another name for this type of circuitry is called a "band-pass filter." There are other types of filters. Those that allow frequencies higher than a certain value to be processed are called "high-pass filters." Conversely, filters that blank out frequencies above a certain value are called "low-pass filters." When conducting a test on very grainy material, low-frequency energy is used to help overcome the grain noise. In this case, it is advantageous to use a low-pass filter to reject the scattered higher-frequency energy. This usually helps to increase the signal-to-noise ratio and provide superior data.

### Gates

Electronic gates are used to produce some action based on a signal being present in the gate. A gate is a device that is inserted into the time base at a user-selected location, as shown in Figure 7-27. It is usually seen as an extra electronic line on the time base. A "positive gate" will enable any signal interrupting this gate to cause a voltage to be sent to a selected apparatus, for example, an audible or visible alarm. Conversely, if used in a negative direction, the absence of a signal in the gate can cause a similar action to take place. For instance, the back-echo signal can be gated. If there is some diversion causing the back-echo to disappear, this occurrence could signal an alarm of some kind. The alarm may also be in the form of a paint gun used to mark a discontinuity area on the item being examined, or a pen recorder used to record the event on a strip chart or an X–Y recorder. Depending on the circuitry, the signal can cause the gate to deliver a voltage that is proportional to the signal amplitude. The gate can be used to produce a numeric display of the signal's horizontal position or percentage amplitude on the screen. The gate threshold can be selected so that it functions at a predetermined signal amplitude. The user, dependent upon the application, also determines the gate position and width. Modern instrumentation uses the gated signal to provide information that is used to calculate and numerically display flaw depth or, in the case of an angle beam transducer, the location distance of a reflector in front of the transducer by programmed trigonometry. Gated signals are also used to produce a "C scan" image (see Figure 7-31). Generally, during a manual ultrasonic examination, it is not necessary to use a flaw gate, although it can also be used to electronically mark a position or amplitude on the time base.

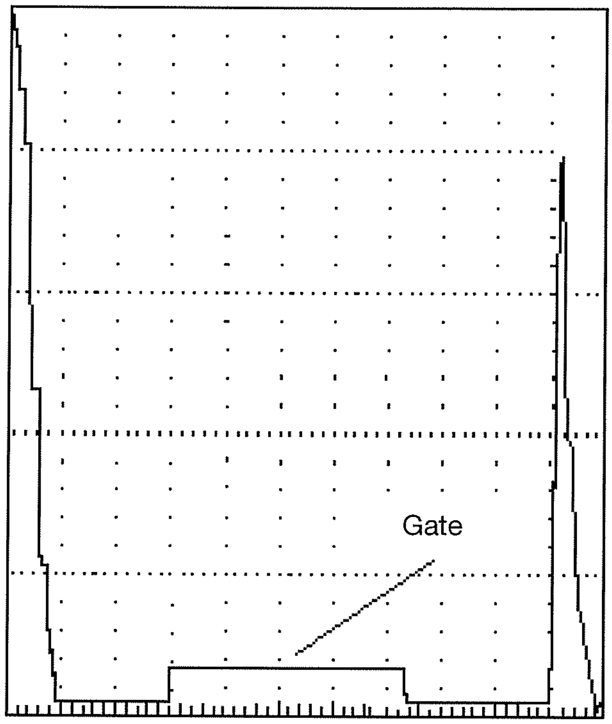



**FIGURE 7-27** Gate.

*Reject.* This control is used to dismiss low-level "noise" on the screen. The effect of reject is visual. The principle of reject can best be described by the exercise of holding a ruler horizontally along the bottom of the screen and raising it until the noise level has been masked by the ruler. (It can be imagined that this practice would affect the instrument's vertical linearity; however, modern circuitry overcomes this problem). Reject can on rare occasions be a useful tool when conducting thickness measurements. [Note. The effect of reject is to electronically suppress the low-level signals on the time base. As a result of this, low amplitude signals from discontinuities may not be observed.

***Storage Memory***

Digital instrumentation usually provides the facility to store calibrations or waveforms. This is a very useful function. A multitude of calibrations can be stored for retrieval at any time. Waveforms (screen dumps) are also stored and can usually be uploaded to a computer for inclusion in subsequent printed reports.

***Displays***

There are a few different ways that the ultrasonic information can be displayed. Typically, "A scan" presentations are viewed with conventional ultrasonic flaw detection equipment. There are other displays to discuss. Figure 7-28 shows a test piece containing two reflectors with an "A scan" presentation. (Note that the presentation has been rotated in the figure to show the reflectors on the screen relative to their actual positions in the test piece).

A scan displays can be "rectified" or "unrectified," as illustrated below Figure 7-29. The unrectified trace has both positive and negative deflections. Other display options can be seen in Figures 7-30 and 7-31.

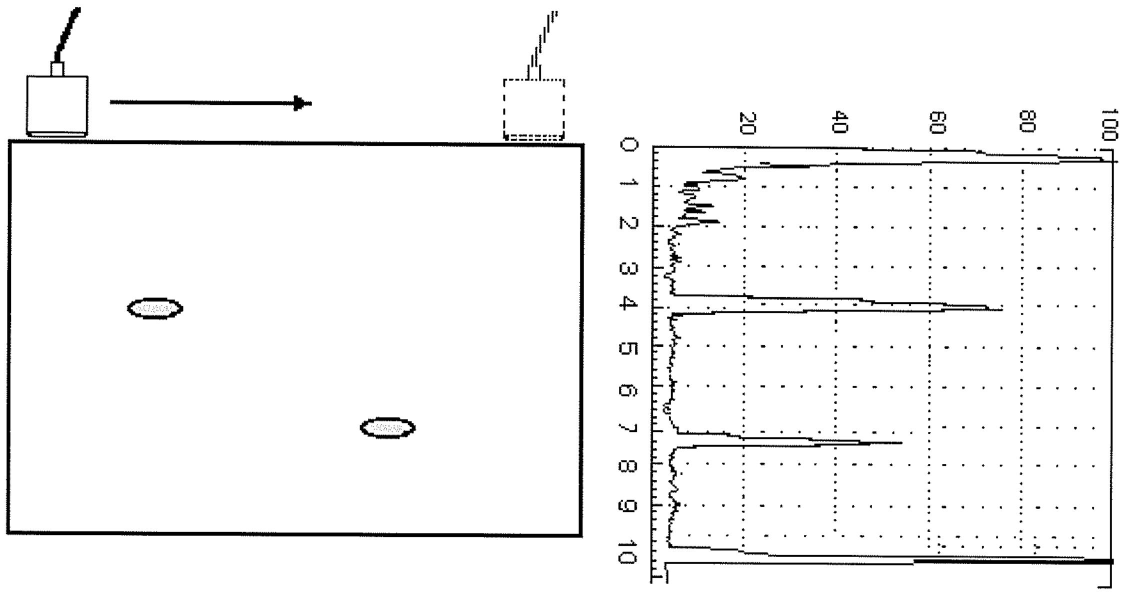



FIGURE 7-28 A-scan.

"B scan" displays show a "slice" through a section of the test piece. In other words, a cross-sectional view. The signals show as bright spots or lines on the screen. In the display shown in Figure 7-30b, the spot is synchronized with the search unit as it is moved across the test piece (at the same speed as the search unit). Note how the back wall signal drops out as the internal reflectors "shadow" the sound energy from the back wall as the transducer passes over them. Figure 7-30a illustrates an ultrasonic test B-scan display of a plastic specimen with various back surface variations.

A "C scan" is a "map" type of display. It is a "plan" view of the test piece (see Figure 7-31). The first C scan recordings were produced with external recorders that were activated by a signal entering an electronic gate set up on the time base, generally between the top and bottom surfaces of the test piece where reflectors were expected to occur. A pen device "wrote" on the recording paper in the areas that were activated by the signal in the gate. Technology today allows the recording of these images digitally and displays them in different colors on a monitor. The transducer is moved back and forth and indexed so as to scan the entire test piece as can be noted in Figure 7-31. (This pattern is known as a "raster" scan.)

The B scan and C scan techniques are well suited to applications where a permanent record is required. It is common to digitally store A, B, and C scan data.

***Transducers***

The transducer is the actual "front end" of the system. It is analogous to the microphone in a public address system. If the public address system is of the best quality and a poor microphone is used, the sound will be only as good as the microphone. The same principle can be applied to an ultrasonic system. The use of an inferior quality transducer on the best system can result in deficient data.

Other terms used to describe the transducer are "probe," "search unit," and "test head." The word "transducer" is from the Latin "transducere," which means to lead

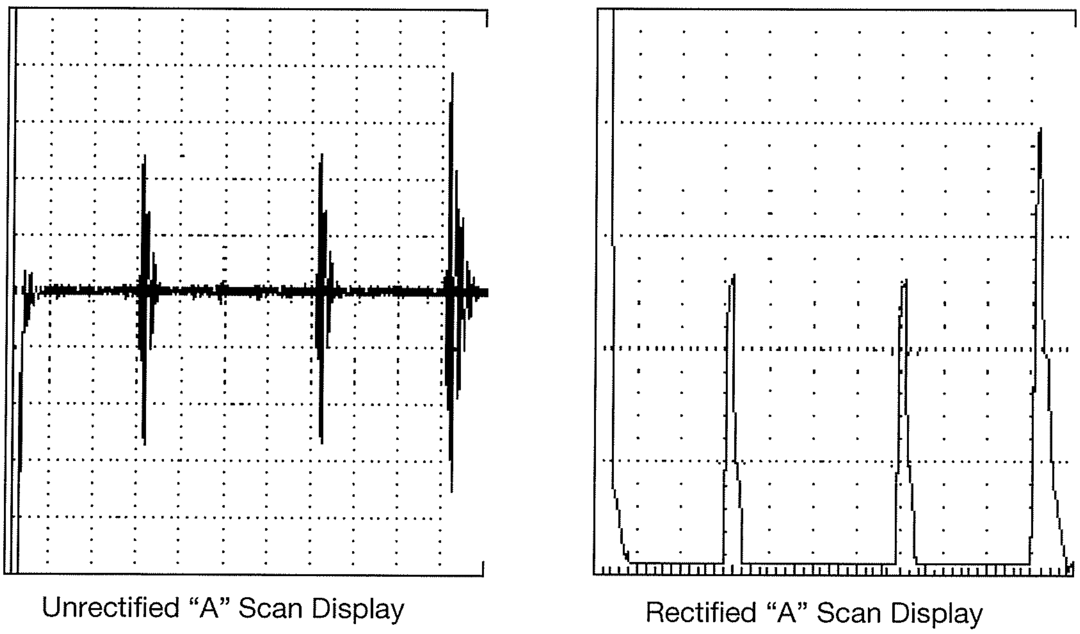
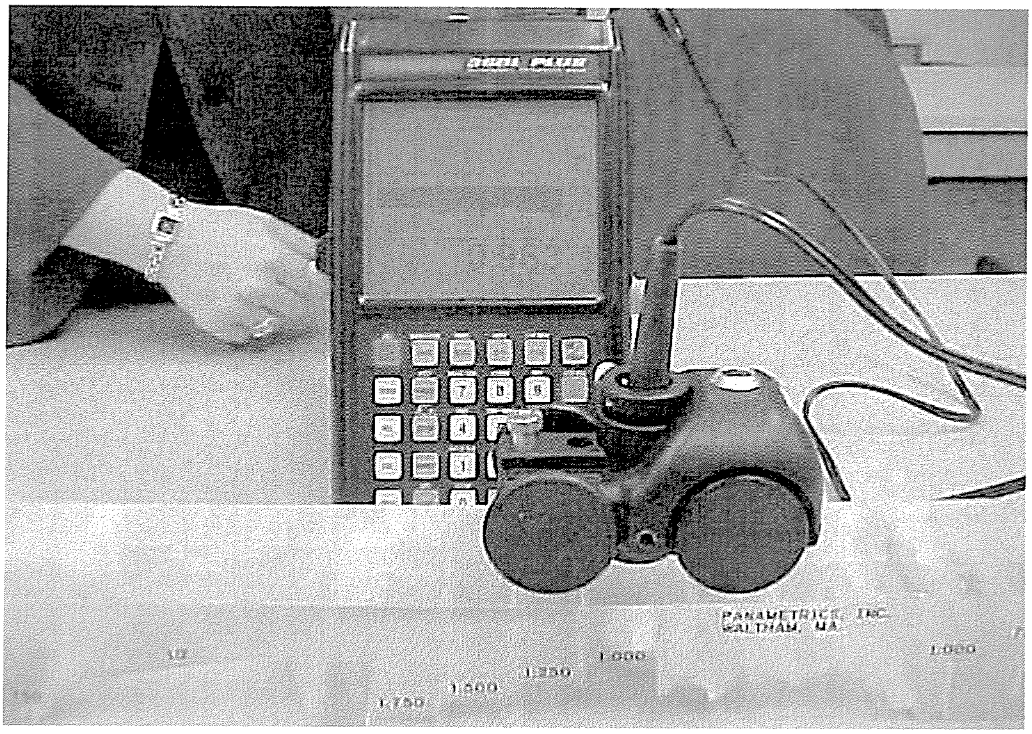
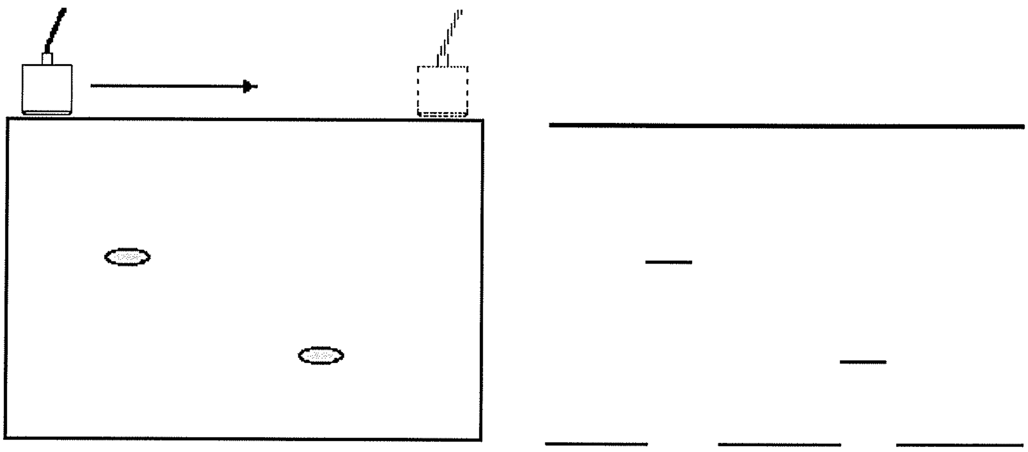



**FIGURE 7-29.**



(a)



(b)

**FIGURE 7-30** (a) B-scan instrument. (b) B-scan.

across or transfer. The function of the transducer is to transfer electrical energy to mechanical energy and vice versa.

As far back as the year 1880, the Curie brothers, Pierre and Jacques-Paul, discovered that when sectioned in specific planes certain crystal materials would generate a voltage when distorted. This is called "piezoelectricity"—electricity due to pressure. The opposite effect is also valid; i.e., if a voltage is applied to the crystal material, it will distort.

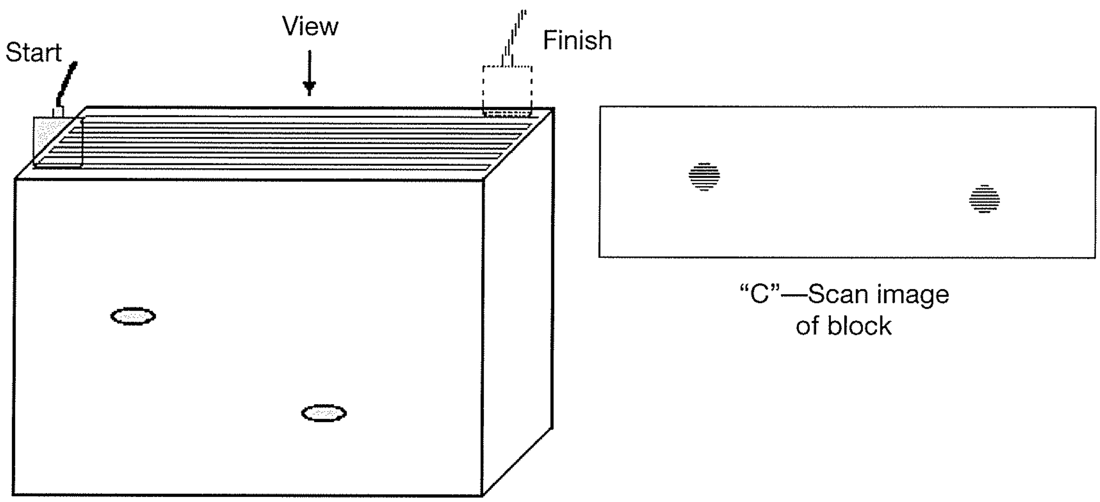



**FIGURE 7-31** C-scan.

Lippman documented this about a year later. Quartz crystal is a prime example of this type of crystal. Other naturally occurring piezoelectric materials exist, such as tourmaline and Rochelle salt. These crystals were used in the early days of ultrasonic testing until polycrystalline ceramic materials—materials that do not exhibit piezoelectric properties in their original state—were developed to perform this function. Some of the more common polycrystalline materials used in transducers are lead zirconate titanate (PZT) and lead metaniobate (PMN). The material is mixed in the form of a slurry, poured into a mold, then dried under pressure. It is then sliced to the required thickness. This is the thickness at which the element will resonates at its designed frequency. (Materials resonate according to their formulation and thickness. For example, a 5 MHz transducer element from PZT may be a different thickness than its counterpart made from PMN). The slices are then placed on a lapping table and precision lapped to the final thickness. The next step is to coat the element with a very thin layer of conductive material, usually silver. This is sometimes electrostatically applied. At this stage, the element is not yet active. It comprises many microscopic piezoelectric elements that are randomly oriented. These have to be aligned or polarized in order for the element to be useful for the purpose of generating and receiving ultrasonic energy. This is referred to as the "poling" process. Electrodes are attached to the faces and the element is immersed in a bath of oil. The oil is heated to a temperature that is around the "Curie" temperature of the specific material also referred to as the "critical" temperature). A high polarizing voltage is applied to the element. The element is then allowed to cool while the field is active. Once cool, the voltage is eliminated and the element is now polarized. This means that the element will have + (positive) and – (negative) polarity. For longitudinal wave generation, the elements are polarized so that the element deformation is as shown in Figure 7-32a. Shear waves can be generated when the elements are polarized to deform as shown in Figure 7-32b.

*Note 1.* Transducers used for angle beam shear wave applications are usually longitudinal. The shear wave component is generated upon the energy traversing two different material velocities at predetermined angles (see Refraction and Mode Conversion on page 7.13).

*Note 2.* Heating a transducer above its Curie temperature will allow the microscopic elements to depolarize. This will remove the piezoelectric properties. When conducting