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CAMERA AND SOUND

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Subject: CAMERA AND SOUND **Credits: 4**

SYLLABUS

Camera

Lighting Equipments; Controlling Light Quality; Knowledge of Safety Precautions & Other Lighting Accessories; Light Meters & how to Achieve the Best Exposure; Latest Meters & the Views of Cameraman; Flash Meter V & VI; Elementary Sensitometry; Color Sensitivity & Spectral Sensitivity Area; Types of Printer; How to Shoot the Gray Card; LAD; Qualities of Natural Day light; View on Color, Light & the Magic Hour.

Sound -I

Sound Production Chain: Microphones, Connector, Mixing or Routing Device, Recording Device and the Monitoring Circuit of the Recording Device; Use of Different Kinds of Microphones; Studio Setup: Mixer, Midi, Harmonizer, Connecting a Sound Card, Cabling, Plugins, Monitoring Device, Subwoofers and the Signal Flow of a Studio Setup.

Sound -II

Recording Process and Post Processing for the Vocals; the Art of Sound Effects; Noise Reduction; Art and Technique of Post Production Sound; Digital Recording; Nonlinear Editing Systems and their Setups; Sound Editing Software; Sound Track and Dubbing; Sound Concepts; the Importance of Sound in a Film

Suggested Readings:

1. Camera, Jean-Philippe Toussaint, Matthew B. Smith, Dalkey Archive Press.
2. The Camera, Larry Hills, Capstone Press.
3. Sound, William C. Robertson, Brian Diskin, NSTA Press.
4. Sound Patricia Kruth, Henry Stobart Cambridge University Press.

LIGHTING EQUIPMENTS & TO CONTROL LIGHT QUALITY

Introduction

Every light source, whether it be the sun, sky, desk lamp, streetlight, fluorescent tube, candle, or professional lighting instrument, has its own character or quality. Color tempera-ture has much to do with the inherent quality of a given source. However, the most important indication of the qual-ity of light has to do with how it looks on a given subject. A picture of a subject taken outdoors under an overcast sky looks quite different from a picture of the same subject on a sunny day. An open window casts light of a distinctly dif-ferent quality than that of an overhead fixture.

Light Quality

This difference in quality affects the nature of shadows cast on the subject and background; they will appear hard or soft, depending on the quality of the light source. The hardness or softness of shadows cast by any light source is dependent on two factors-distances between the subject and the source and the size of the light source.

All other things being equal, when the distance between the subject and the background is increased, the shadow softens. In most shooting situations, the relationships between subjects cannot always be altered (e.g., the relation-ship between a subject's nose and face). Therefore, the sec-ond factor becomes very important-the size of the light source.

Light sources with large areas from which the light emanates will cast soft shadows, while those with small areas will cast hard shadows the light from large sources, such as an overcast sky, tends to wrap around the subjects and fill in any cast shadows.

But because apparent size of source is more important than actual size when it comes to determining light quality, the distance from the light source to the subject is also a fac-tor. When a lighting unit is very close to a given subject, the Source is also relatively large. However, when the light is moved further away from the subject, the source becomes relative]y small. A good example of a hard light source is the sun. Although it is actually the largest source available, its great distance makes it appear to be a small size source and hence very secular.

The larger the source,the more diffuse the light and the less dense the shadow.

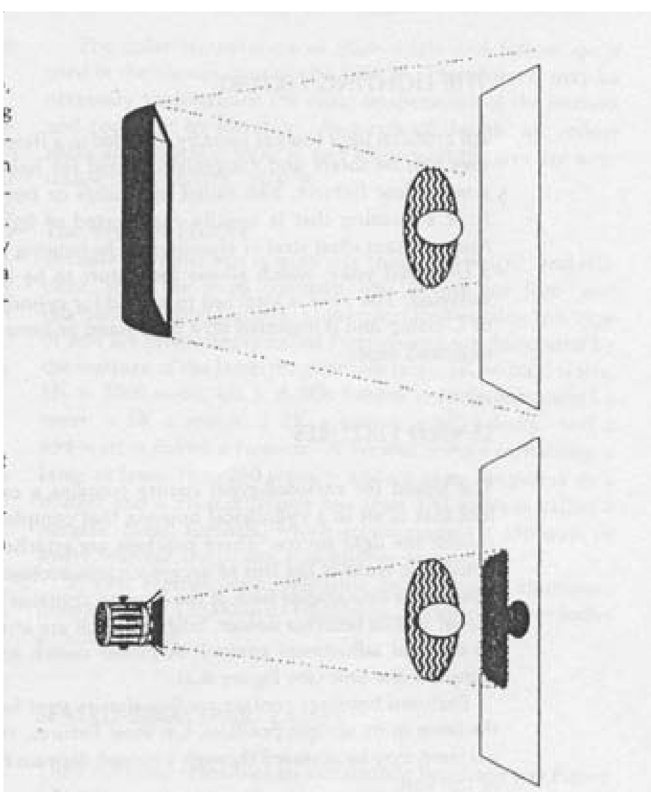


FIGURE 6.1 The larger the source, the more diffuse the light and the less dense the shadow.

Controlling Light Quality

Lighting units are generally selected for their ability to create or eliminate shadows, depending on their intended use. Some-times it is preferable to illuminate a subject with com-pletely shadowless lighting. This may be the case when shadows may interfere with an already intricate pattern, when shadows detract from the primary subject, or when soft, shadowless illumina-tion suits the mood of the script. In these cases, it is wise to use the largest lighting units available.

. In many lighting setups, hard sources of light are used to give shape, form, and character to the subjects. If a ball is illumi-nated with a completely shadowless light, it will photograph as a flat disk. When a hard spotlight is used, its round shape is immediately apparent. The shape of faces, bodies, and all other three-dimensional objects are revealed with clarity and vitality by directional luminaries.

Hard lights provide the directionality and characteristic look of the shot; soft lights raise the overall illumination levels and fill in dark shadows. The lighting quality of real settings can be duplicated by selecting' instruments that cast the same type of shadows as the sources that illuminate the real settings.

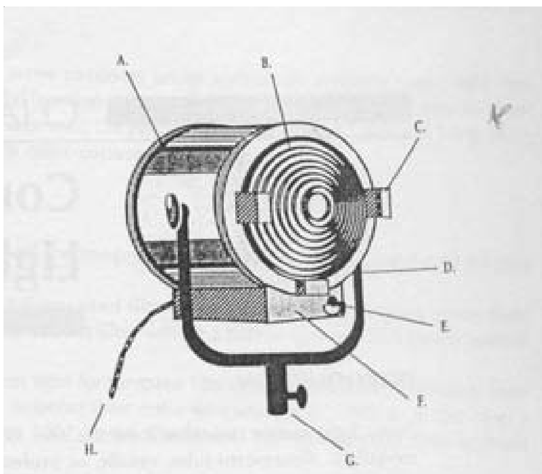
The Lighting Fixture

All artificial light sources must be installed in a fixture before they can be safely and conveniently used for lighting pur-poses. Most fixtures, also called luminaries or instruments, have a housing that is usually constructed of lightweight, heat-resistant sheet steel or aluminum. The housing pivots on a u-shaped yoke, which allows the fixture to be tilted and Swiveled. The yoke is attached to a spud (or cylindrical pin). Or c-clamp and is mounted on a light stand or hung from an overhead pipe

Lensed Fixtures

The lensed (or enclosed-type) fixture contains a condenser lens that is set in a cylindrical housing that completely sur-rounds the light source. Three brackets are attached to the housing in front of the lens to accept various accessories and filters. The box-shaped base of the housing contains the light socket and its reflector holder, both of which are attached to an external adjustment control. A power switch and cable complete the unit .enclosed housings contain cooling slots to vent heat from the lamp in its upright position. On most fixtures, the internal lamp may be accessed through a hinged door on the front of the unit.

Reflector Design



Except for some carbon arc lamps, all enclosed fixtures use a reflector, which is usually a curved plate with a polished metallic surface. A reflector primarily redirects light, but some designs actually
Absorb" infrared or uv radiation. Depending on the luminaries, the reflector may be spherical, ellipsoidal. Parabolic or a combi-nation of these shapes. The shape of the reflector largely determines the throw (the effec-tive length of its projected beam) and the quality of the light beam.
The spherical reflector is used in enclosed-type fixtures and scoops, the ellipsoidal reflector is used in both open- faced and enclosed fixture types, and the parabolic design is found in sealed-beam lamps, such as the parabolic aluminized reflector (par). The combination reflector is used in the soft light.
The light socket and reflector are mounted together on rails and are linked mechanically to a focus adjustment knob. The knob (located on the front or rear base) is either of a semi rotating paddle or rudder design, a rotating screw knob, or a sliding handle. The adjustment knob moves the socket and reflector toward the lens for floodlight position or away from the lens for spotlight applications.
Reflector design is crucial in indirect lighting fixtures, such as soft lights, where all light emanates from the reflector, rather than from the globes themselves. Soft light reflectors may be large, trough like units that emit scattered light over a wide area. Metallized umbrellas are often attached to various open-faced fixtures, essentially converting them into soft lights.

Lens design

.enclosed-type instruments used in motion picture and televi-sion produc'tiori employ at least one piano-convex or fresnel lens. Sealed-beam lamps use a self-contained, fluted lens design. Both designs refract the divergent rays of the radiat-ing source and focus them in parallel or convergent beams. The piano-convex lens, which has one flat surface and one convex surface, is the simplest design. Because of its great mass, it is quite heavy and does not dissipate heat. This lens is most often found in the ellipsoidal fixture.
The fresnel lens was originally designed to replace the heavy piano-convex lens for use in lighthouses. The french physicist augustine jean fresnel took the heavy, plano convex lens, cut away most of its convex surface, and duplicated the curved contour with a recessed series of sloped concentric rings, which collect and direct radiant light into parallel .the flat surface of the lens is also textured, which slightly diffuses the transmitted light. Light transmitted through the fresnel falls off gradually enough to allow for two lamps to blend softly in adjacent areas. The fluted lens, an integral part of par and other similar lamps, is discussed in "sealed-beam lamps."

The Ellipsoidal

Fixture

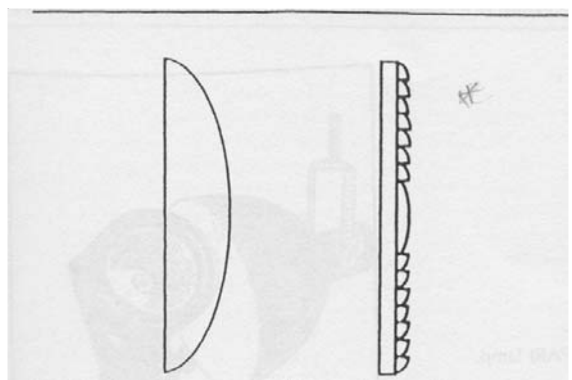


FIGURE 6.3 Plano-convex (left) and Fresnel lens (right) designs.

The ellipsoidal fixture, so called because of the shape of its reflector (see figure 6.4), contains one or two piano-convex lenses, thereby giving the ellipsoidal fixture its characteristic long throw and ability to project a pattern on a given area. The ellipsoidal is sometimes called a leko light. The ellipsoidal housing contains four internal framing shutters, which can be adjusted to project a pool of light with hard, defined lines, such as a square or rectangle. The instru-ment's beam is focused by sliding the telescoping lens tube at the front of the fixture. Ellipsoidal fixtures, used primarily in theaters and sound stages, are somewhat heavy and un-wieldy for location work. They do not have on/off switches and are usually mounted from overhead pipes or grids.
The follow spot, which may house several piano-convex or fresnel lenses, is used when a concentrated, hard circle of light is needed to follow a moving performer. The larger models use a xenon, hmi, or carbon arc light source to achieve a 300-foot or

4oo-foot throw. The follow spot is usually mounted on a heavy stand.

All follow spots have a shutter, a reflector, a lens to focus the light, an on/off switch, a quick-focus handle, an iris, and a front housing (called a boomerang) that contains a set of filter holders. There is also a douser for complete beam blackout.

The color temperature of ellipsoidal and follow spots used in the theater may not be 3200ok. Therefore, it may be necessary to determine the color temperature of the fixtures and correct it by filtration. As a rule of thumb, arc follow spots are generally 55000k and ellipsoidal fixtures are any-where from 2800 °k-34oo °k

The Fresnel Fixture

Because a fresnel lens is relatively thin, lightweight, and effi-cient, it is the most common lens in use for film- and television-lighting fixtures. Luminaries that employ this type of lens are often simply called fresnels and are designated by the wattage of the lamp they contain (e.g., 2k = 2000 watts, 5k = 5000 watts, etc.). A 10k fresnel is sometimes called a tener, a 5k a senior, a 2k a junior, a l k a baby, and a 650-watt is called a tweenie. A fresnel fixture containing a lamp of fewer than 250 watts is known as an inky-dink or a midget and a fresnel fixture less than 100 watts is called a peewee. Small fresnels, which accommodate a 650-watt or lower wattage lamp, are also known as peppers. A new fresnel lamp, featuring unpainted aluminum housing and a 12, ooo-watt hmi lamp, is called a silver bullet or silver streak.

Sealed-beam (Par) Lamps

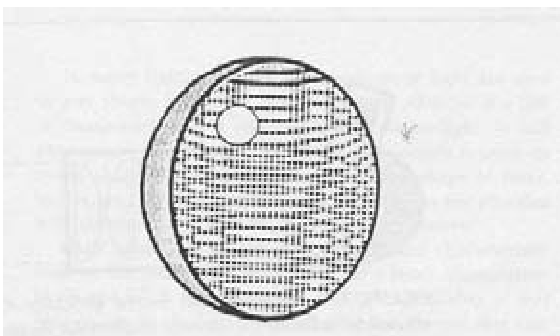


FIGURE 6.5 A parabolic aluminized reflector (PAR) lamp.

The par lamp resembles an automobile headlight (see figure);it has a tungsten filament, a silvered bowl reflector, and a fluted lens that is self-contained within the unit. The par angle of light is constant and may not be focused. To change the angle of light,' the lamp must be changed for another lamp of a different beam shape. The par may be mounted inside a simple cylindrical housing called a par can or used in a module in clusters of up to 12 similar fixtures.

A module is a flat housing designed to accept a single par lamp. Modules are often combined in rows or clusters of two or three lamps and can be bracketed together to make up two-, four-, six-, nine-, or 12-lamp groupings. Clusters of this type are often used for outdoor applications as a substi-tute for carbon arc fixtures.

A rectangular housing fixture that incorporates two or three pars and is mounted on top of a camera is called an obie light. Advantages of the par can are its relatively low cost, light weight, and durability. The most widely used par can is designed to house the par 64-a 1000.watt, 32000k lamp that is about 8 inches in diameter. Used extensively at rock concerts, its powerful punching beam with a soft edge has made the par 64 a favorite for lighting nighttime street scenes.

There are a variety of lamps available under the par designation. Each lamp is described by a three-letter code that specifies the beam width, wattage, and color tempera-ture of each. One of the most common, the fay lamp, is a 5000ok, 650-watt par globe with a dichroic filter built into the lens of the envelope.

Open Faced Fixtures

Enclosed fixtures, though versatile and controllable, are too heavy and bulky for some location work. Therefore, they are often augmented by fixtures that feature a lightweight, open-faced design.

Open-faced fixtures are those that have no lens; the light source is visible and easily accessible. Light rays are directed solely by the shape of the reflector. Open-faced fixtures are capable of much greater luminous intensity per watt than comparable enclosed fixtures, thereby making them ideal for location work. Some commonly used open-faced luminaries are the prime, compact, scoop, broad, soft light, and background fixtures.

Prime and Compact Fixtures

A prime fixture has a round housing, no base, and is fluted or perforated at the top to vent heat. Much of the intense heat generated by the lamp is dissipated at the open face of the instrument. Some prime fixtures have an on/off switch on the housing; most have an in-line switch on an ac cable that is connected directly to the housing. A focusing control at the back of the fixture controls the degree of flood or spot. Unlike the enclosed housing fixture, in which reflector and lamp move in conjunction, the light source is moved in rela-tion to the reflector, which remains stationary.

The compact fixture is a small, portable prime fixture that draws fewer than 1000 watts. The lowel pro-light, for exam-ple (see figure), offers a variety of interchangeable reflec-tors and numerous accessories for the location filmmaker.

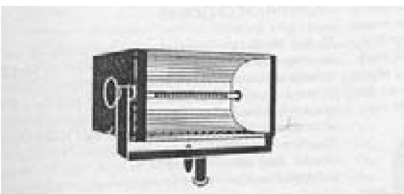
FIGURE 6.6 The Lowel Pro-light. (Courtesy Lowel-Light Mfg., Inc.)

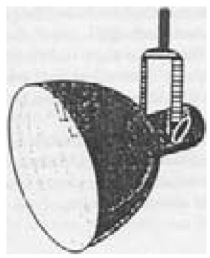
A 2k, open-faced fixture is often called a mighty, a lk a mickey, a 650w a teeny, and a 600w a teeny-weeny. Several manufacturers offer prime and compact fixtures as part of a location kit, which is an easily transportable case that con-tains three or four instruments, stands, and accessories.

Scoop and Broad Fixtures

Scoops and broads (see figure 6. 7) are used for soft fill light. The scoop, which is primarily a studio fixture, is so named for its large, bowl-shaped reflector, which also serves as the housing. The scoop produces an excellent, soft light over a widespread area. However, the scoop fixture is used infre-quently on location, because it is heavy and cumbersome, and its light is difficult to control.

The broad consists of a tubular quartz bulb and a shallow, metal reflector set in a small, rectangular housing.





A Scoops Fixture (top)

A Broad Fixture (Bottom)

Single broad fixtures are not usually focusable, but multibulb units are focused often. The illumination produced by a broad fixture is spread over a wide area, but the light is not very diffused. Shadows can be as harsh as any as those pro-duced by a spotlight. The broad fixture is particularly good for lighting an entire wall or other interior background. Because they are often tucked away in corners and other tight places on a shoot, small broad fixtures are frequently called nook lights. The broad tends to have a shorter throw than the scoop.

A popular and versatile variation on the broad is the lowel tota-light (see figure), which can function as an indirect light with an attachable umbrella reflector.

Soft Light Fixtures

The softlight (see figure) is designed to direct illumina-tion from the light source, which is shielded, into a large reflector that directs a soft spread of indirect light to the sub-ject.

The softlight has a rectangular, open face; a large reflec-tor of either aluminum baked white enamel. Or metallized black cloth (as the lowel softlite); and, often, a double light source (usually two 1000-watt bulbs).

The illumination of a soft light is considerably more dif-fuse than that of it scoop and is so scattered that it appears to wrap around a subject, thus giving the impression of a shadowless illumination. The softlight, however, is a rela-tively inefficient instrument in terms of luminous intensity.

There are also softlights that incorporate a bank of four to eight fluorescent tubes, rather than tungsten lamps. These fixtures are useful as supplemental illumination on locations that are

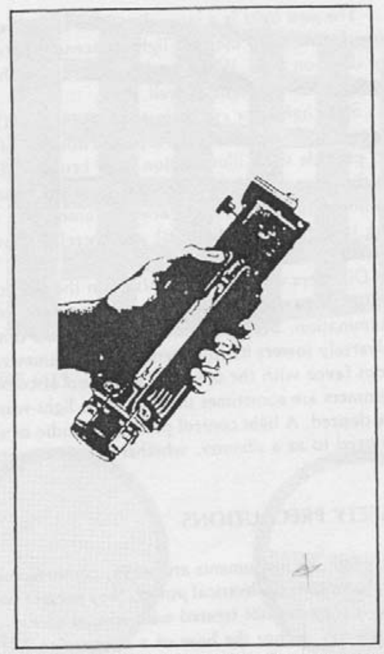


FIGURE 6.8 The Lowel Tota-light. (Courtesy Lowel-Light Mfg., Inc.)

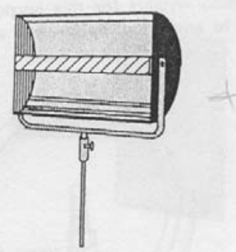


FIGURE 6.9 Softlight.

illuminated predominantly by overhead fluorescent panels. These soft lights do not emit a great deal of light, but they balance well with the existing fluorescent light without needing filtration.

Background Fixtures

Background fixtures are used to evenly illuminate large back-ground areas and include the pan and the strip light.

The pan light is a large-diameter, round-reflector fixture (see figure) used for lighting scenery outside a window or door on a set. When used with diffusion sheets, it makes an excellent fill light as well. Strip lights (or eye strips) (see figure) are multiple source units used along the tops and bottoms of backgrounds to provide wide illumination over broad surfaces, such as cycloramas and sky backdrops. They are usually run from dimmers (a rheostat that varies the amount of voltage flow-ing to a fixture or fixtures) and therefore have no on/off switch. Dimmers were used extensively in the days of black-and white photography as a means of decreasing and increasing illumina-tion. Because reducing voltage to a tungsten source adversely lowers its color temperature, dimmers have fallen from favor with the universal adoption of color film stock. Dimmers are sometimes used when its light-warming effects are desired. A light control panel in a studio or stage is often referred to as a dimmer, whether it uses rheostats or not.

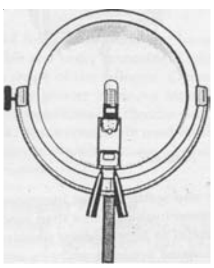
KNOWLEDGE OF SAFETY PRECAUTIONS & OTHER LIGHTING ACCESSORIES

Safety Precautions

Since lighting instruments are heavy, cumbersome, and draw great amounts of electrical power, they present potential haz-ards if they are not treated with special care.

Always anchor the base of a light stand, century stand (c-stand), or reflector stand with a suitable-size sandbag in order to keep the fixture from toppling. Special red canvas sandbags with built-in carrying straps are available in 15-and 35-pound weights for this purpose. It is particularly important to anchor all lightweight aluminum stands and

Any stand extended to its full height. A good rule of thumb is to use one sandbag for every rise of the stand you extend. Many gaffers route the fixture's ac cable through a round, keyring-type clip attached to the base of the stand. Finally, all cables should be taped down securely to the floor or walls with gaffer tape. Make sure that all fixtures are mounted right side up, with the cooling louvers on top. A fixture should never be allowed to operate in an inverted position, which defeats the venting design and shortens the life of the lamp.



a pan fixture.

A striplight

Lighting Accessories

Next to the fixtures themselves, lighting accessories are the most important tools used to control illumination. They pro-vide the means to change and manipulate light beam shape, intensity, quality, and color temperature. Lighting accesso-ries may be classified according to the way each is employed, whether they affix to the instrument or light stand, whether they are separately mounted go-betweens called gobos, whether they are peripheral accessories, such as sandbags

And apple boxes, and whether they are durable or expendable items.

Fixture-Mounted Accessories

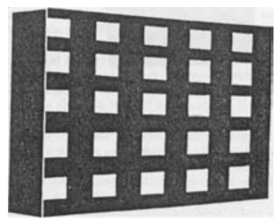
Barndoors

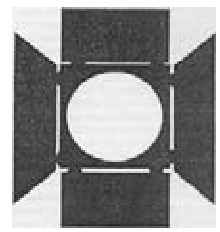
A barndoor (see figure) resembles a kind of bonnet that attaches to the front of a fixture; it has two or four adjust-able, hinged, black metal blades or "doors." the barndoor has a flange or bracket that enables it to slide into a holder on the front of a fixture. Some barndoors may be rotated in the holder; other barndoors are fixed.

A barndoor is used to keep extraneous light from falling on given areas of a set and it controls light by creating a soft-edged transition between lighted and shaded areas. The blades may be adjusted in any position between fully closed and fully open. The doors are often brought together to form a slit of light so that a sign or other detail can be highlighted. Barndoors, while effective on directional light fixtures, do lit-tle to control the highly diffused illumination of softlights. For this situation, a gridlike, bladed accessory called an egg-crate may be placed on the indirect lighting fixture to keep the beam tighter. An eggcrate looks like a christmas orna-ment box with the bottom removed

(see figure).

An Eggcrate Barndoors

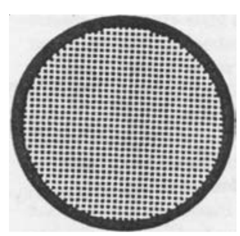


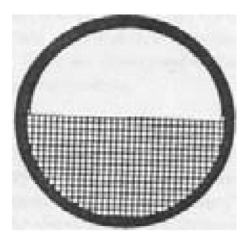


Scrims

Scrims are the primary tools for dimming a light beam with-out adversely affecting its color temperature. Scrims are cir-cular metal screens that fit into the same front brackets of a lighting fixture that hold the barndoor. The screen, which comes in various thicknesses, covers any portion or all of the scrim; it is usually made of stainless steel to withstand the intense heat given off by the fixture. The most common are one-quarter, one-half, and full scrims (see figure). Scrims also come in single- or double-mesh densities. The scrim cuts the intensity of a beam without greatly diffusing it. Fabric scrims, which mount on separate stands, are called nets.

Scrims

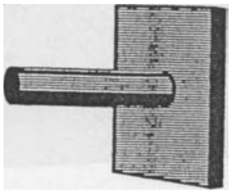




Snoots

A snoot is a conical or cylindrical piece that attaches to the front of a fixture (see figure) to concentrate the beam in-to a small, round area and eliminate spill light. Snoots are used to high-light and isolate selected areas in a scene, such as a single table in a nightclub. When a snoot is not available, one may be impro vised with a heavy flat black foil known as black wrap.

A snoot



Goosenecks

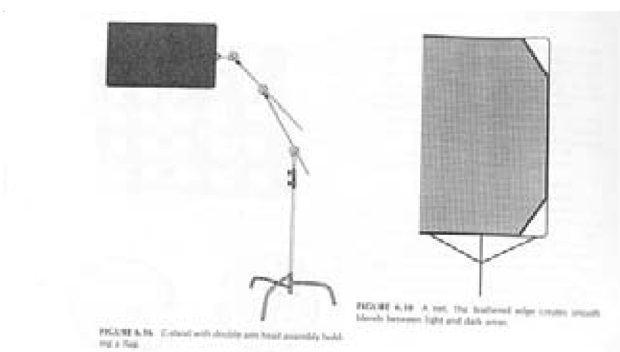
A gooseneck is a flexible tube that attaches with a clamp to the yoke of an instrument and holds a small gobo in the light beam to take down a flare or to eliminate a highlight.

Gobos

A gobo is a generic term for any opaque unit used to elimi-nate light from a particular area in a scene. Gobos are usually mounted on c-stands or gobo stands (see figure). A gobo head is fitted to a c-stand and accepts gobo rod exten-sion arms of various diameters that can be rotated in any position. Many c-stands have a sliding leg that adjusts to level the stand on an uneven surface; this is called a rocky mountain leg.

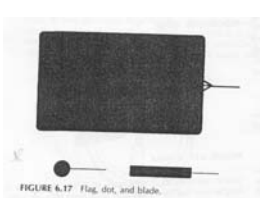
Flags

The most common gobo is the flag-a rectangular wire frame covered with duvetyne, which is a black fabric. A flag is used to block an area from light emanating from a fixture. Flags vary in size from 12 x 18in up to 30 x36in. A cutter is a long narrow flag used when light from more than one side-by-side fixture is to be controlled or when the shadow of a microphone boom must be eliminated. A blade is a small narrow flag, frequently made of translucent acrylic. Targets and dots are circular flags that can be introduced into a light beam to shade small areas, particularly useful for toning down harsh highlights and other hot spots (see figure).



C – stand with double arm headassembly holding a flag

flag, dot and blade



Nets and Silks

The terms scrim and net are often used interchangeably to describe woven screen materials that reduce the amount of light falling on a subject. In general, a scrim is made of metal and a net is made of cloth.

A net is made of cloth-mesh material stretched on a frame (see figure) and mounted on a c-stand

A cloth net may be used close to a fresnel instrument when it is in the full flood position, but should not be placed too close when the lamp is spotted down, as the heat will damage the fabric. A net should never be placed close to an open-faced fixture. A hexagonal weave (a honeycomb pattern) net fabric is preferred for professional lighting, as, unlike the traditional screendoor-type weaves,

A hexagonal weave breaks up light most evenly. Of course, the more threads (and fewer open-ings) in a weave, the less light will pass through.

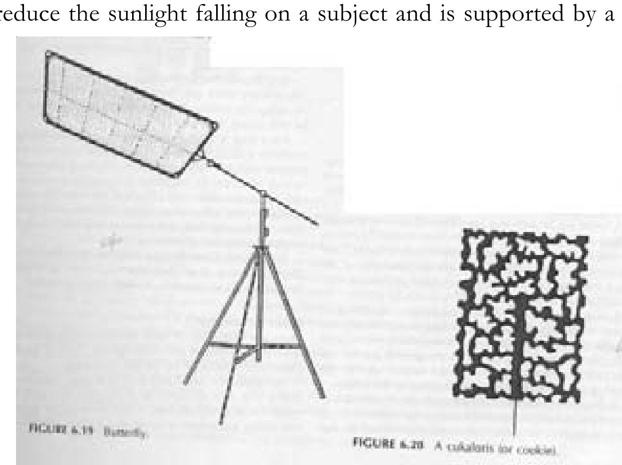
Black nets tend to cut light intensity with a minimum dif-fusion effect, lavender nets add diffusion, and white nets soften the light considerably. Nets come in several "strengths," depending on how many layers of weave are used. A net with a single layer that is color-coded green cuts about 30% of the passing light, while a double red net cut transmitted light by 50%. A single lavender net cuts light by 15%.

A silk is a white, fine-weaved net used primarily for add-ing diffusion. The amount of light the silk passes depends on the proximity of the silk to the light source. When sandwich-ing two or more nets, a moire pattern (a plaid effect) can be eliminated by rotating one of the nets in relation to the other. Miniature nets are available in various shapes and sizes, and are often used in flexible goosenecks to reduce light that is falling in specific areas within an otherwise evenly lighted area. In practice, the key light (or lights) is usually set first, each area of the set is given the proper illumination, and any obvious double shadows are eliminated. Large softlights are then placed near the camera, if possible, so that any shadows they create will fall behind the subject. The amount of fill light is adjusted to bring the shadows to the desirable expo-sure range. This lighting ratio is expressed as 2:1, 3:1, 4:1,

And so forth. A 2:1 ratio means that the fill light is one half as bright as the key light-a difference of one f-stop.

Butterflies

A butterfly is a large net (measuring 4ft x 4ft or larger) used to reduce the sunlight falling on a subject and is supported by a



single stand. The butterfly is big enough to cover a medium shot or two-shot and reduces the harsh look of hard sunlight (see figure).

When a translucent material is used on a butterfly frame. It is called a silk. When a butterfly frame is stretched with black duvetyne, it becomes a solid (really a giant flag). In order to maintain shade in changing sunlight, using a solid is often a wise alternative to re-setting up a shot. Butterflies are often set at an angle, rather than being held in a horizontal position.

Butterfly

Butterflies and overheads often come as kits, including stands, sectional frame, single and double nets, silks, and a solid.

Overheads

An overhead is a very large frame that measures up to 20-foot square and supports a stretched net, silk, or solid (a large flag). An overhead is used to filter or block light from a two-shot or three-shot and also to shade any large prop, such as an automobile. The overhead is supported by two heavy roller stands with oversized gobo heads. Overhead stands, used to hold anything that has to extend higher than the reach of a c-stand are often called high rollers. Wind can wreak havoc with a large overhead; hence they are often secured to the ground with ropes.

Cookies and frames

A cookie (or cukaloris) is a sheet of plywood, painted screen, or other material perforated with irregular holes (see figure).
when placed in front of a light source, the cukaloris breaks up the beam and projects a mottled pattern of light and shadow. The effect is useful for eliminating glare from reflective flat surfaces and adding visual interest to walls and backgrounds. The cast pattern becomes more distinct as the cukaloris is moved away from the light source and closer to the subject. A frame is a plywood cukaloris cut to project patterns of venetian blinds, paned windows, prison bars, or any other silhouette across a set.

Reflectors

Reflectors (which include heavy, stand-mounted shiny boards and lightweight reflectors such as flexfiils, foam core, and reflective polyester sheets) are essential for daytime lighting in most exterior locations. They are often used exclu-sively in place of exterior lighting instruments.

Shiny Boards

A shiny board (or reflector board) is a large, rigid panel that pivots on a yoke that is mounted on a heavy stand. The shiny board has two metallic sides-one “hard” and one “soft.” the hard side of the shiny board is a smooth reflective surface that reflects a highly specular light. The soft side bears a gridwork of delicate foil leafing that breaks up the lightto provide a softer light. These silvered surfaces are used to bounce the sun’s rays into shaded areas to reduce the contrast between sunlight and shadow.

Shiny boards are usually surfaced with silver-colored foil, but other colors are also available. A gold surface reflects amber light and can imitate sunset or sunrise effects. Gold reflectors give pleasing warmth to flesh tones and particu-larly enhance the appearance of dark-completed actors.

They also impart a lush, saturated look to plants and foliage in general. Blue- or cool-surfaced reflectors are sometimes used to boost the color temperature of a reflected tungsten lamp or the setting sun to 5500ok.

A board can be used as a key light, fill light, or a kicker (to back light a subject). If the light is too intense, a slip-on net may be used to cut down the intensity of the light. These slip-on nets should only be used on the hard side of the board, for they will war away the foil leaf grid if used over
The soft side. .

A shiny board must be positioned properly in relation to the sun to catch its rays. The easiest way to aim the reflector is to point the reflector at the sun, tilt the reflector down until the reflected beam can be seen on the ground, swivel the board until the beam is just under the subject, and tilt the board up until the reflected beam illuminates the subject.

Under gusty conditions, shiny boards may be difficult to control and should be attended by a grip (an assistant) at all times. In general. Be careful not to aim the reflector so that the actors are forced to squint; instead, move it off to one side of the camera to keep glare from the actor’s line of sight. The reflector should also reflect light on the subject from above eye level if a natural effect is desired. Don’t let the board wobble in the wind so that the light shimmers on the actor’s face. Lock the reflector board down and secure it firmly with sandbags. If the reflector light is too harsh, move the unit back or place a net over the board. When the shot is completed, the board should be tipped so that it is parallel to the ground with minimum surface to the wind.

Shiny boards have some disadvantages. Changing weather can create costly delays as clouds pass over the sun. In any case, the board will need to be constantly adjusted as the earth’s rotation changes the relative angle of the sun to the reflector. The flat, broad reflector boards catch the wind as well as the sun, and may tip over if they are not secured with sandbags or individu-ally manned. Stands can often telescope to a height of 10 feet or more, but extended high rise stands can be particularly hard to control. Shiny boards are heavy and rigid, and may require a pickup truck to transport them.

Lightweight Reflectors

The general unwieldiness of conven-tional reflector boards has encouraged the use of lighter, less cumbersome units on small-scale shoots.

Durable, moralized polyester (mylar) sheets, available in many different colors and textures, can be shaped and con-figured in various ways. They can be used in automobile in-teriors to reflect light onto faces and are also very handy for shooting in very small rooms, booths, passageways, and any number of other tight places.

Flexfills, which resemble large drum heads with one shiny side and one white side, are made of fabric that is stretched on collapsible frames. They are light and compact, but must be held steady by a grip.

Foam core, a stiff, styrofoam-center construction board available in 32 x 40-inch and 40 x 60-inch sheets, is a very popular reflector. Foam core has two brilliant white surfaces (one of which can be metallized by applying aluminum foil) and is also available with silver and gold metallic surfaces. All lightweight reflectors are susceptible to wobbling in the slightest air turbulence, so be very careful when using them outdoors or in drafty interiors.

Bead board is another popular reflector material. It is a fibrous, porous sheet that can also serve as a sound dampening panel as well as a bounce lighting board.

Other Accessories

There is a wide array of other standard and improvised ac-cessories used in film and video lighting. Some of the more common are discussed on the following pages.

An apple box is a closed wooden box with a grip hole on each side of the box. The apple box, which comes in several sizes, is used for propping up stands and raising equipment or actors to a higher elevation.

A cup block looks like a wooden ashtray mold with a shallow, bowlike recess and is used to stabilize wheeled stands and prevent them from rolling away.

A flex arm is an articulated extension arm with a number of ball joints designed to hold small gobos, such as dots. With a clamp at one end and a l/t-inch diameter locking receiver at the other, the flex arm is affixed to a c-stand (unlike the gooseneck, which mounts to the fixture itself). The gobo at the opposite end is placed across the light beam to cut down a flare or eliminate a highlight.

The gaffer grip (or gator grip) is a giant alligator clip with one or two s/.-inch pins on the handle and or jaws. The gaffer grip is used to mount lightweight fix-tures to doors, chairs, tables, pipes, and other places that may offer a stable grip. It is also handy for securing foam core sheets. Grip clips look like big metal clothespins, but are not as heavy as gaffer grips, and are available in four sizes. Grip clips have a variety of uses-to fasten gels, foam core boards, and other lightweight articles to stands and fixtures.

Pole cats are adjustable poles that extend vertically or horizon-tally between floor and ceiling or between two walls to hold lightweight fixtures.

Expendables

Any accessories that are used up or expended over a period of time are called expendables. Expendables include diffusion material, colored gels, and tape.

Diffusion is a term to describe any material that breaks up and scatters the light beam so that it appears to be larger and softer (le., shadows are less distinct or practically nonexis-tent) than its original source.

A diffused light source is difficult to control with barn-doors and flags, due to the scatter of its light rays. The degree to which diffusion material reduces and softens light depends upon the density of the material used.

Silk is a term for any cloth material that has a bright translu-cence. Tafetta, an artificial fabric, is finding greater use in “sils”, as it is more durable than traditional china silk. Heavier diffusion cloth of cotton or nylon is called gauze. Photogra-phers have often been known to use white bedshit as different gauze. Any diffusion material made of cloth must be placed away from light source for obivious reasons

Frosted gelatin, a highly fragile material once used widely for diffusion purposes, has been phased out in favor of the more durable plastics. Acetate is a stiff, sheet material that should be mounted on a c-stand in front of a fixture. If acetate is placed on the fixture, the heat will cause it to warp and melt. Polyester sheeting has become very popular because of its high durability. Spun glass is also popular as a diffusion material. Although it will not burn, spun glass may darken when placed too close to a fixture; another disadvantage is its skin-irritating fibers.

Filter gels are widely available in many colors and include color-balancing filters, color-compensating filters, and spe-cial color effects filters.

Diffusion and colored acrylic sheets, more durable than gels, are also readily available.

It is advisable to test the light transmission factor of any gel before using it in a sh09t. This may be accomplished simply by taking an incident light meter reading (see chapter seven) of a given source and comparing it to a reading of the same light through the material in question.

Other commonly used expendables include gaffer tape, a strong, fiber-backed, all-purpose adhesive tape used to fasten down cables and provide temporary joints. While similar to the often substituted and cheaper duct tape, gaffer tape offers greater bonding properties, is easily removable, and tends to leave less residue than the cheaper tape. Be aware that gaffer tape will pull up paint from any finished surface to which it is applied.

Wooden, spring-type clothespins, colloquially called c-47s, are often used to affix diffusion and colored media to bamdoors and frames. Other expendables include dulling spray, which is an aerosol sprayed on glossy surfaces to take down glare, and smoke liquid for fog machines.

A widely used expendable material is foam core, a stiff, styrofoam-center construction board often used as a reflec-tor.

A show card is another stiff art board used for bounce lighting; it comes in sizes as large as 4b-in by 60-in and has a matte black surface on one side, in addition to one white surface. The black side makes the show card useful as a flag.

Another popular, expendable reflector material is grif-folyn the trade name for extremely durable, three-ply rubber sheeting. Griffolyn sheets come in 6 x 6-foot, 12 x 12-foot, and 20 x 20-foot sizes, and are usually stretched on frames and mounted on heavy stands. Foam core, show cards, and griffolyn (all light-weight materials) are suscepti-ble to fluttering in the slightest air turbulence, so be very careful when using them in breezy or drafty conditions.

Notes

LIGHT METERS & TO ACHIEVE THE BEST EXPOSURE

Multiple Purpose Lights

Occasionally, you can make lights serve dual purposes and still maintain the three-point lighting effect. Here, a one-on-one interview is lit with only three lights. Note that each of the (very carefully placed) lights serves two purposes.

If distances are carefully controlled, the lights will be 50 percent brighter as back lights than as keys.

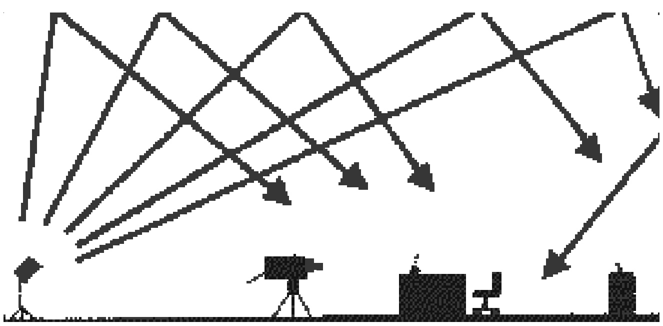
This can work well under carefully controlled situations where you know in advance the color of each person's hair (or, possibly, lack of hair) and the color of clothes that will be worn by each person. Obviously, you won't have much latitude in accommodating special needs. In this situation the chairs can't be moved without upsetting the lighting balance

Bounced Light

For short ENG segments bounced light can be used. The drawings below show approaches for large and small rooms. Although the soft lighting effect leaves a bit to be desired, this approach may be adequate for short segments.

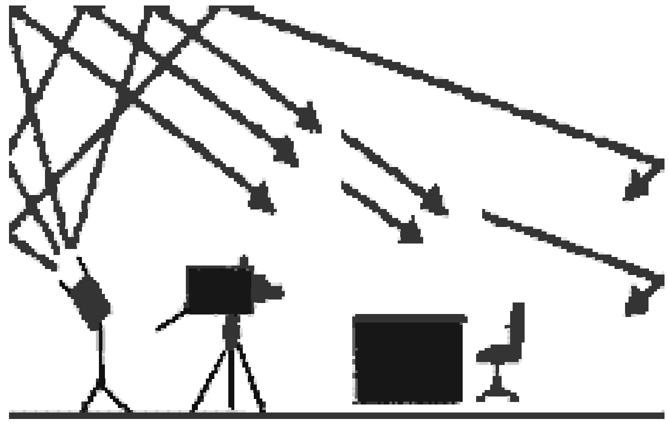
Note that this design uses a single light bounced off the ceiling. Obviously, you must be working in a room with a relatively low white or light-gray ceiling. The white, acoustic tile commonly found in offices works well.

Bounced light creates a soft, even light throughout the entire room, an effect that is similar to what we are used to seeing with overhead fluorescent lights.



If the camera is back far enough, a light mounted on top of the camcorder can be aimed at the ceiling for a bounced light effect. The camera and attached light should be far enough back from the subject so that the light will come down at an acceptable angle. If the light is too close to the subject, dark eye shadows will result. If the walls of the room are a light, neutral color, they will reflect part of the bounced light and more fully fill in shadow areas.

The second drawing assumes a smaller room. To keep the light from coming down onthe subject at too steep an angle, it's aimed at the back wall. Again, this approach creates an extremely soft effect, which may or may not be desirable.



Although much light will be lost in the double-bounce process, with today's highly sensitive cameras this should not be a problem.

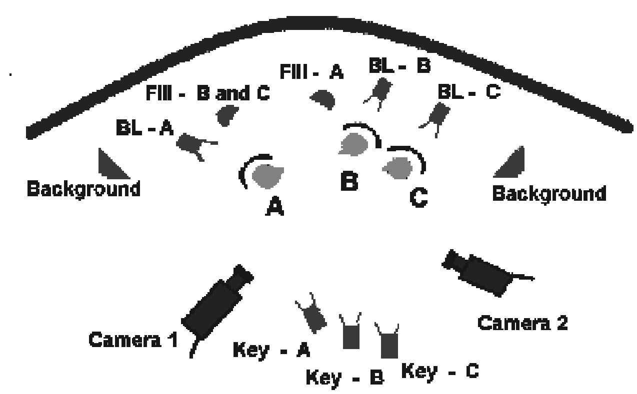
To help compensate for the color that the ceiling and walls will add to the light, be sure to color balance the camera under the bounced rather than the direct light.

More traditional lighting for a typical office interview is explained

Lighting Multiple Subjects

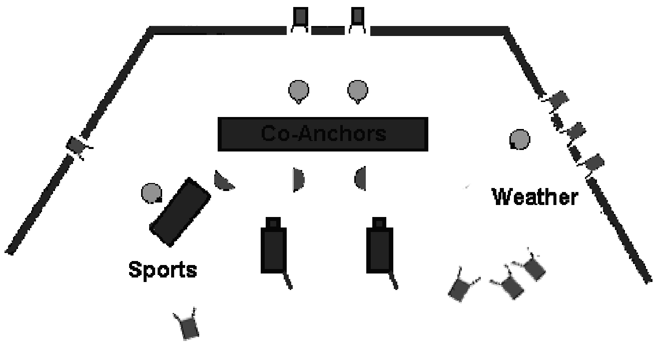
Thus far we have covered the lighting of one subject only. Life isn't always that simple.

First, we'll take a look at a typical three-person interview setup. Note below that even though things appear to be much more complex, we've only just repeated the basic three-point lighting setup for each of the three people.



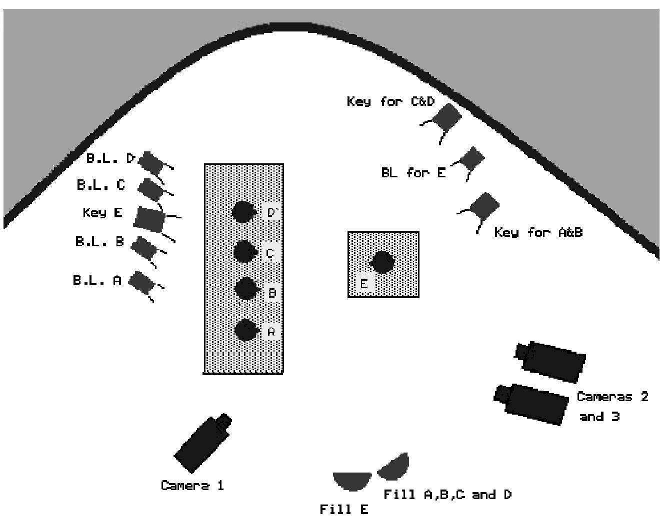
A simple news, weather and sports set is shown below. By panning to one side or the other, the two cameras can get one of the co-anchors, or the sports or weather person.

Also note that the positions of the key and fill lights provide three-point lighting coverage for each of these camera positions. Background lights are not shown—and may not be needed—because of the ambient light that all of the lights would produce.



As we previously noted, some lighting directors prefer a flat, high-key approach to lighting news anchors. This is done either by either making the key-fill ratios even (1:1), or by placing key lights directly over the cameras. Other lighting directors feel that a 1:1 ratio flattens facial features and robs scenes of dimension. And by putting key lights directly over cameras it can make teleprompters more difficult to read.

Now let's take a look at an even more complex example.



Note that two large Fresnels are used to key all of the subjects on the left, and five smaller Fresnels are used as back lights on these subjects. One of these backlights also keys the person on the right. Two scoops provide the necessary fill light for the entire set. Barn doors keep lights intended for one person from spilling over onto another person or area.

Sometimes you may need to develop non-formula lighting designs for special dramatic effects. The author once did a cabaret style music series where the following lighting design was used.

Area Lighting

So far, we've covered subjects conveniently confined to one place. But what if one or more subjects must be free to roam around a set while on camera? There are four ways this can be handled.

First, the entire area can be flooded with a base light, which is an overall, even light. Scoops or color-balanced fluorescents will work here, assuming the area isn't too large.

Important closeup camera talent positions are then keyed with lights at twice the intensity of the base light.

Small pieces of tape placed on the floor can provide marks for the talent to "hit" as they move from one major camera position to another.

With this approach you will probably not want to barn off the lights any more than necessary, because illuminated areas should be kept large enough to give the talent a margin of error in missing their marks.

The second approach involves keying, filing, and backing the entire area (generally, a dramatic setting). Here the whole working range—assuming it's not too large—is treated as a single subject. This will require a powerful (high-wattage) key light positioned at a great enough distance to cover the entire area.

If the key is placed in the center of the set, 90-degrees to the back wall, the angle will be appropriate for cameras positioned at each side of the set. One or more Fresnels with diffusers placed at either side of the set can serve as fills. (Scoops or banks of color-balanced fluorescent lights will not throw light far enough to reach the back of a large area.)

If multiple keys are needed to achieve a high enough level of illumination over the set, they should be positioned as close together as possible to reduce the problem of multiple shadows and multiple catchlights in eyes.

Over a large area multiple back lights will have to be used. They should be aimed to create slightly overlapping pools of light over the whole talent area. The talent should be able to walk from one area to another without obvious variations in back light.

The third approach to lighting a large area is to divide the set into individual areas and key, fill, and back each area. Often, large interior settings are divided into four or more parts for keying, filling, and backing.

Typically, the lights at the edge of each of these areas will just begin to merge. With this approach it's important to make sure that closeups on the talent will not be in the transition points between lighted areas.

Keep in mind the sources of light that may be suggested by the setting—visible table lamps, windows, etc. Place the key lights so they will be consistent with these suggested sources of illumination. This is called following source.

The last approach to lighting a large area would be appropriate to simulate an interior at night. This technique would use a low key lighting ratio from 3:1 to 6:1, and the talent would move in and out of specifically defined set areas. Only important, closeup areas would be lit, leaving the rest of scene relatively dark.

With this approach it's especially important to follow source; i.e., place keys so that they are consistent with the visible or suggested sources of light within the setting. If a person were sitting next to a reading lamp, the key would have to be angled so that the light would appear to be coming from the table light. In some cases you may want to use a low-level base light over the entire set to keep in-between areas from going too dark.

Using A Stand-In

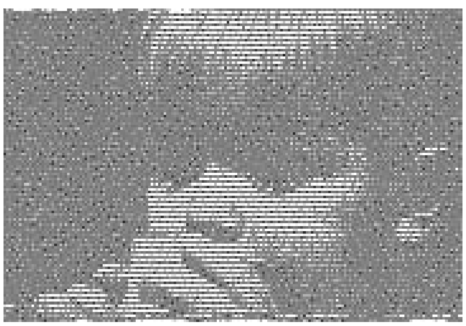
Whatever lighting approach you use, the lighting can be checked on camera by having a stand-in (a person of similar height, skin color, and clothing as the talent involved). This person should slowly walk through the positions on camera as the lighting is carefully observed on a good color monitor.

During the show's dress rehearsal with the actual talent any remaining problems can be spotted and then fixed during the break between the dress rehearsal and the actual production.

Existing (Natural) Light

In news and documentary work the most "honest" approach to lighting is to make use of the existing (natural) light present at the location. This shows things as they really are (within the limitations of the video process), rather than after they have been altered or embellished by "artificial" lighting.

The problem is that existing light is often unsuitable. The contrast ratio can be too extreme; there can be mixed sources of light (daylight, incandescent light and fluorescent light all at the same location); or, the light level can be too low for a quality video signal.



Note that the existing light photo here suffers from both underexposure and a high contrast ratio.

There's also another consideration: people are used to seeing interviews, etc., enhanced by good lighting. Without it, it appears to many viewers that "the picture's dark," or "the picture isn't too good."

This is not unlike the situation photojournalism faced a few decades ago when existing light still photography was first used in publications such as Life magazine. Since people were used to seeing flash-on-the-camera photos, natural light photography seemed unnatural—even though it accurately showed the actual conditions being photographed. (Flash was necessary in the early days of photojournalism because of the relatively slow speed of film and lenses.)

Occasionally in dramatic productions you have to fake lighting to make it look real.

Here, the light from a computer screen is insufficient to illuminate the child's face. So to simulate this setting, a light with a blue filter is positioned on the other side of the computer monitor.

As videographers strive to add an artistic dimension in their work, they start to rely more and more on shadows to convey meaning.

Altering

Appearances

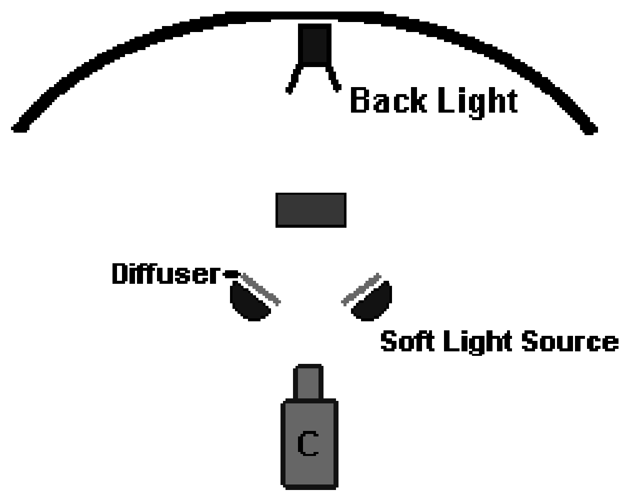
There are situations when you will want to consider special lighting applications to either alter the look of a subject or accommodate difficult subject matter.

Minimizing Surface Detail

First, let's look at how you can combine what we've covered to completely minimize surface detail. Although we've previously seen how the quality of light can be used to do this, now we're going to combine three steps to further enhance this effect.

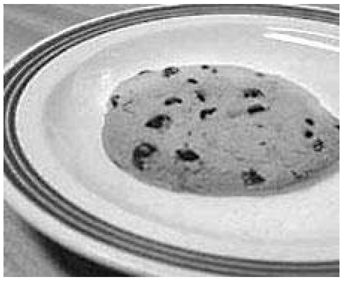
- decrease the key and fill angles
- use soft light sources
- reduce the lighting ratio

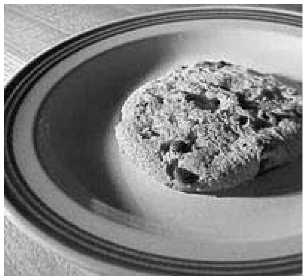
This lighting plot shows these modifications.



1. Instead of the normal 90-degree angle between the key and the fill, note that here the front lights have been moved as close to the cameras as possible. In the process, detail revealing shadows have been virtually eliminated.
2. Next, note that the soft light sources are equipped with spun-glass diffusers. The resulting ultra-soft illumination further minimizes detail-revealing shadows.
3. Finally, the lighting ratio between the two lights has been reduced from the normal 2:1 to 1:1, which means that the two front lights are of equal intensity. Keep in mind that you only need to change front lighting; the position or intensity of the backlight and background light will not change.

In most cases you will not want to go to the extreme of making all of these changes. (Note below the chocolate chip cookie on the left looks pretty dull and lifeless.) For example, you may decide to stay with a 2:1 lighting ratio and just move the lights closer to the camera and use diffusers; or, you might want to check the effect of just using diffusers.



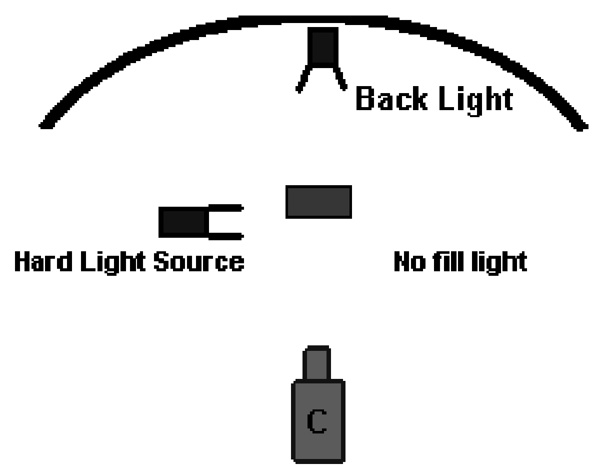


Now let's go in the opposite direction and maximize surface detail, as shown on the right above. This can be done by reversing all three previous modifications.

- increase the key-to-fill angles.
- use a hard light source for a key
- increase the lighting ratio

1 Note in the drawing below that the key has been moved to about an 85-degree angle to maximize shadows and surface detail.

2 Next, you will want to use a hard source of light for a key. A focusing light such as the ellipsoidal spot we talked about earlier will create the desired effect.



3 Finally, we would want to increase the lighting ratio to 4:1 or greater. By eliminating the fill light altogether we would go considerably beyond 4:1 and maximize the lighting ratio.

Again, it's generally not desirable to alter the back light position or intensity, or the intensity or position of a background light.

The photo on the left below shows how effective side lighting is in bringing out detail in this ancient carving. Without the effect of this type of light, much of the detail would be lost.

The photo on the left below shows how effective side lighting is in bringing out detail in this ancient carving. Without the effect of this type of light, much of the detail would be lost.

Two hard key lights, one on either side of the coin, lit the closeup of a 50-cent piece above. The angles of these lights are so oblique (and precisely aimed) that only the raised portions of the coin are illuminated. This keeps the background areas dark.

We are used to seeing primary light sources coming from above subjects—typically, sunlight, or the 45-degree angle of a key light.

When the primary (key) light is placed at a low angle, a dramatic or mysterious effect can be achieved.

High Key and Low Key

Two terms that are often misunderstood in lighting are high key and low key.

These terms have nothing to do with overall light intensity. Instead, they refer to the angle of front lighting (generally the key light) and the resulting presence or lack of shadow areas. The photos below might help in seeing this distinction.

A high key scene would be an evenly lit scene, one that has no intrusive shadow areas. (Note the surface of the strawberries.)

In the studio, sitcoms, variety shows, etc., are normally lit high key. The first cookie shown at the beginning of this module was lit in a very high key fashion.

On the other hand, a scene lit in a low-key fashion would have pronounced shadow areas. (Note photo on the right below.)

It's important to remember that in all these examples the intensity of the key light could be exactly the same. We are not talking about "bright lights" in high key setups and "dim lights" in low-key setups.

In actual fact, under "dim light" we would simply have to open the camera's iris in order to achieve adequate exposure, and then our so-called "low key" effect would instantly disappear. It's only the angle of the key and the lighting ratio that make the difference in these photos.

We've covered some special lighting situations in this module. But, remember, unless there is a compelling reason to use these techniques, you will find that the standard three-point formula approach covered earlier will yield the best results for standard video work.

The Exposure Meter

We have seen that there are two ways to measure quantities of light-the amount of light falling on a subject (incident light) and the amount of light reflected from a subject (reflecled light). The cinematographer measures light accord-ing to one of these two systems using an exposure meter designed to measure incident light, reflected light, or both.

Most modern exposure meters may be classified as either photovoltaic or photoconductive. The photovoltaic meter uses a light-sensitive cell consisting of selenium that is bonded to a metal base plate and covered with a thin film of gold or platinum. As light falls on the cell, current is gener-ated and measured by a microammeter, which measures minute amounts of voltage.

As a greater amount of light falls on the cell, more current is generated and a higher reading is displayed on the meter, and vice versa. Meters of this type are simple in construction and require very little maintenance. The cells have a spectral sensitiv-ity similar to film emulsions, so no special color com-pensation is necessary.

The one drawback to the selenium, photovoltaic meter is its relatively poor sensitivity to low light levels. Neverthe-less, the rugged, selenium meter design is well suited for incident-type meters used in the motion picture industry. If there is not enough light to move the needle on the light meter, there usually isn't enough light to shoot the film. However, for situations involving high-speed film and low light levels, and for reflected, spot, and through-the-lens (TTL) applications, a different meter design is often used.

The photoconductive meter uses a cell made of cadmium sulfide (cds). Unlike the selenium cell. Which produces a minute voltage when struck by light, the cds cell acts as a semiconductor that varies its electrical resistance according to the amount of light that strikes it. A battery provides the current and the. Cds regulates the amount of that currenthy its variable resistance. The changes in current are then dis-played by the microammeter as a light reading.

The advantage of the cds cell is its high sensitivity-about ten times that of an equivalent selenium meter. This feature makes the cds cell well suited for very low-light situations. For this reason, cds cells are used almost exclu-sively in reflected light meters, especially spot meters and TTL metering systems.

Light meters incorporating cds cells are very versatile and are often sensitive enough to give readings in moonlight.

There are, however, definite disadvantages. For one, they use batteries and, like so many battery-powered instruments, frequently need re-calibration due to voltage inconsistencies when batteries inevitably run down. Batteries and the voltage-regulating circuits that often accompany them add extra weight and bulk to the meter as well. The cds cell also suffers from sensitivity lag-a problem similar to light lag exhibited by certain video pickup tubes in low-light situa-tions. In dim light, the meter does not respond very quickly to small changes in illumination.

Light Meter Designs

Modern light meter designs fall into four categories, with some models able to convert from one type to another (when accompanied by accessories):

Hand-held, reflected light meters reflected light spot meters through-the-Iens reflected light meters incident light meters

Reflected Light Meters

Reflected light meters are designed with the light-sensitive cell located behind some sort of shield to control the light accep-tance angle. This shield may be a perforated grid, a simple narrow tube, or a lenticular magnifier. The purpose of the design is to produce a meter that is more or less direc-tional, so that the meter can be pointed toward a given object and not read light from all directions at once.

The reflected meter, which reads the brightness of the light reflected by objects in the scene, is calibrated for 18% reflectance (medium gray). Cinematographers must decide whether they want the subject to be reproduced as medium gray. For instance, a light-colored face reflects about 36% light, while a dark-colored face may have less than 18% reflectance. The meter, however, reads each face as having 18% reflectance and will not give ideal readings, thereby underexposing light flesh tones and overex-posing the darker complexions. The way to avoid guesswork is to use an 18% reflectance gray card and measure the light reflecting off of it, ii\stead of off of the subject.

There are several ways to use a reflected meter-taking integrated readings of a complete scene, measuring specific tonal areas, and reading an 18% reflective gray card.

An integrated reading of a complete scene is the most common method of 'using the reflected light meter, particu-larly among

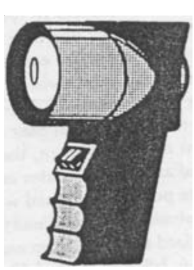
amateurs. This is also the least accurate method. Depending on how the meter is aimed at the scene, it may take in too much bright sky or dark earth and thus give false readings.
Measuring specific tonal areas of the scene will lead to more accurate exposure control. Reflected light readings of the darkest and lightest areas are taken and an exposure value is chosen from these two extremes. Such an exposure should, as closely as possible, capture the entire tonal range.
The third technique is to use the reflected light meter to read a subject of known and constant reflectance, usually a gray card. The gray card is specially designed to reflect 18% of the light that strikes it, which is an average of the bright-est, the darkest, and all the intermediate tones of any given scene. These medium-gray cards correspond to the calibra-tion of incident meters and enable the reflected meter to give the same reading (when used properly) as an incident meter. The card should be angled halfway between the light source and the camera for the most accurate reading.

Spot Meters

Spot meters are reflected light meters fitted with a lens (see Figure) and allow only a very narrow angle of accep-tance, usually from .5°-5°. Because the measuring angle is so small,. Spot meters also have a viewfinder. Spot meters are capable of giving accurate readings of small portions of a scene. They will also measure flesh tones from great dis-tances, making them ideal to use with long camera lenses.
Spot meters are ideal in situations where taking incident light readings would be inconvenient or impossible, such as at a concert or similar event. Unfortunately, like most spe-cialized precision instruments, they are very expensive.

Through-the-lens Meters

TTL meters are, essentially, spot meters built into camera bodies. With the advent of reflex cameras and zoom lenses with reflex viewing systems, it was a natural step to place a tiny, light-sensitive cell in the reflex light path and thus con-stantly monitor the brightness of the scene as viewed by the camera. The TIL meter has evolved into the fully automatic exposure system wherein the cds cell controls a current powering a tiny motor that instantly sets the aperture (or shutter speed).
There are two important drawbacks to the TIL metering system, due to its uncritical reliance on the basic reflected light meter. First, because this system essentially gives an integrated reflected light reading, it does not produce consis-tent flesh tones due to changes in backgrounds and object surrounding the subject. Second, when a camera pans from a dark scene to a lighter one, the result is often a distracting and abrupt aperture change.



A spot meter

Incident Meters

Since the incident light meter is the most valuable and versa-tile meter for motion picture and video lighting, we will devote the greatest part of our discussion to it.
The incident light meter is always recognizable by its round, white plastic dome (/umisphere or photosphere), which acts as a three-dimensional. Light-gathering surface. The light-sensitive cell is located behind the dome and senses the total intensity of the light striking the dome.
The incident meter is used to measure light falling on a scene and is usually held at the subject position with the dome aimed at the camera lens. The dome is designed to sim-ulate the shape of a person's face or other three-dimensional object within a scene. The advantage of this design is that the dome will automatically give a balanced reading of one or more light sources striking the subject. Thus, readings are integrated in a way that corresponds to relative light and dark portions of the three-dimensional subjects within' a scene. The incident meter is often used for determining expo-sure; however its most important feature is its usefulness in determining lighting ratios.

Using the Incident Meter

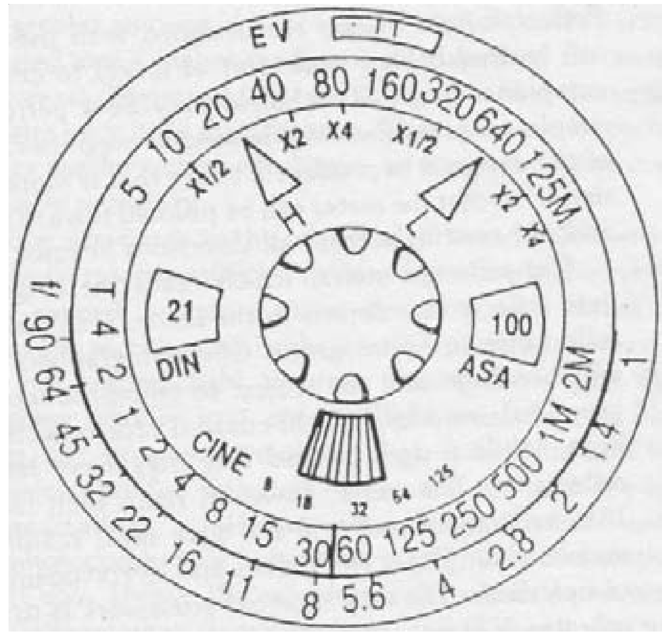
Popular incident meters include the Spectra Pro and the Sekonic Studio Deluxe. The American-made Spectra has its calculator dial on the back of the meter, while the Japanese-made Sekonic has a dial on its front side (see Figure). Despite their superficial differences, they work in the same basic ways. A meter that has recently gained widespreild popularity is the Minolta Flash Meter IV, which features a microcomputer and liquid crystal display (LCD) dig.jtal readout. Originally designed for photog-raphers who use electronic flash, this meter is extremely useful for cinematographers, due to its direct fps and motion picture shutter speed readings.
The readings given by most meters usually need to be in-terpreted in order to determine proper exposure; this is the purpose of the calculator dial mounted on the meter (see Figure 7.5). These calculators vary in design, but all have at least four dials that correspond to four variables-film sen-sitivity, expo-sure time, lens aperture, and scene brightness.
An exposure index dial is on each calculator. The IE dial has an indicator (or window) where the film sensitivity or “”SA rating may be set. There is also a scale with numbers that correspond to those that appear on the micr'oammeter dial. When a light reading is taken, the number indicated by the needle is noted and the calculator is rotated manually un-til the appropriate pointer is aligned with the corresponding number on the calculator scale. Finally, there is a scale that contains the standard series of f-stop numbers (i.e., f-1, f-1.4, f-2, f-2.8, f-4, f-5.6, f-8, f-11, f-16, f-22, f-32, f-45, f-64, f-90, etc.). This aperture scale rotates in direct proximity around' a scale that is inscribed with shutter speed numbers. Begin-ning with long exposure times of perhaps 60 seconds, these numerals proceed in increments equal to half the preceding time (30 seconds, 15 seconds, 8 seconds, 4 seconds, 2 seconds), move to one second in the middle of the scale, and then continue in fractions of a second. Fractions are usually indicated by a small slanted line. The series proceeds with reduced exposure times that halve the preceding exposure time (e.g., '/2 second, '/, second, 8, 15, 30, 60, 125,

250, 500, 1000). When the calculator dial is set at the point correspond-ing to the needle reading, the f-stop number is located directly adjacent to the shutter speed at which the camera is set. The aperture on the camera lens is then set to this f-stop.
Many incident meters have one or a series of perforated filter slides for adjusting the sensitivity of the light-sensitive cell. A filter is inserted in a slot on the top of the meter to cut down on the light striking the cell. The Sekonic comes with one "high" slide only, for use in exterior and other bright light situations. The Spectra comes equipped with a series of slides, including a X 10, a X 100 (cutting light by those amounts, respectively, for bright light applications), and various ASA-rated slides (see Figure).
The ASA slides may be used when shooting sync sound motion pictures (when a 1/48 or 1/50 shutter speed is con-stant). Just insert the appropriate ASA slide for convenient, calculation-free readings directly from the meter.
The advantage of using the incident meter is that a dark object in the scene will be rendered dark on the film, a white object rendered white, and a 36% flesh tone will reproduce on film as a 36% flesh tone, without the need of using a gray card. Consis-tent and accurate flesh tones are one important way of recognizing correct exposure; for that purpose, no other type of exposure meter will give more accurate readings than the incident meter. The i"cident meter does have some disadvan-tages. It cannot distinguish the lumi-nance of objects; it would not be a good choice for making readings of the sky, for instance. Also, it is not particularly sensitive in low-light conditions, but it is usually sufficient in all motion picture applications.

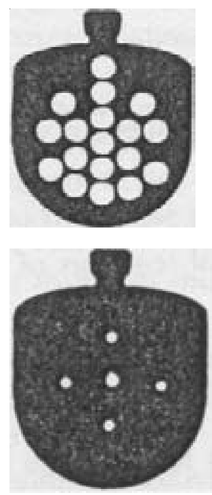
Other Incident Meter Uses

The incident meter may also be fitted with a flat, translucent, white disc (a photodisk or lumidis,k) in place of the white dome. The disk, being two-dimensional. Provides accurate readings of flat subjects-artwork, signs, and table tops. The disk aids in giving a reading that corresponds accurately to the effective surface illumination based on the angle of in-cidence to the light source. For readings of this nature, the meter is held with its back against the flat surface, with the disk parallel to it.
In addition to the dome and disk attachments, incident meters may also be fitted with a round filter (lumigrid or photomulti-plier), which changes the incident meter to a
Reflective meter. However, it is best to use a particular meter/:" for the purpose for which it was expressivly designed.

Sekonic l-398 incident light meter



Incident meter calculator dial
Spectra Pro Incident Light Meter with two light filter slides.



Care and Feeding of Light Meters

A light meter is a very sensitive instrument; it should nev~r be handled roughly or dropped. If it is dropped or damaged; the meter should be sent out for re-calibration before it is used again. Manufacturers allow themselves :t1/3 f-stop tolerance in calibration, so be sure to check the repaired meter against others; they may not necessarily be in agreement.

The perforation dimensions in filter slides are calibrated to each individual light meter. Store the light meter with its own slides and don't mix the slides up with another meter's slides. Meters are very possessive in this regard; they don't like their family split up in such ways and may give spurious readings as a result. If slides are lost or mixed up, the meter must be sent in for re-calibration. Also, be sure to keep the photosphere and disk free of dust and dirt, as the translucent plastic may discolor or darken with age. If this happens, the meter must be re-calibrated.

Notes

LATEST METERS & THE VIEWS OF CAMERAMAN

The Latest In Light Meters Sharing the views

Sekonics L 608 And L 508

Bob Shell, July, 2001

I've used Sekonic meters in my personal photography since the late 1970s. My reason for sticking with one brand, even though there are other good brands on the market, is that I have personal experience with their durability and their design philosophy. I know I can trust them to give me accurate meter readings time after time. My oldest Sekonic digital meter, the L-328 Digilite F, which is now 15 years old, is still just as accurate as the day I bought it and has survived more encounters with my studio floor than I care to admit. I gave up on meters using moving needles after dropping them and spending lots of money getting them repaired.

If my oldest Sekonic meter is still accurate, you may wonder why I have upgraded to newer models over the years. There is more to a meter than just accuracy, although that is the most important consideration. New features have been added over the years to make the meters more versatile and easier to use. When Sekonic came out with the L-408, which combines a reflection spot meter with an incident meter, my thought was that this meter had everything and there would not be much they could add. The L-408 was even waterproof, although Sekonic only claimed water-resistant. Mine has actually been submerged without harm. After using the L-408 for some time I have only one complaint—the 5° angle of the spot meter was too wide and you had to get in really close to isolate small areas. It was as if they had been reading my mind when the L-508 (L-508 Zoom Master, to give its proper name) came along, but I'm sure other photographers had the same thoughts. The L-508 has a much narrower angle for the spot meter, and it is variable from 4° down to 1° to allow fine-tuning of the area to be metered. Like the L-408, it is very well sealed against dirt and moisture and can be used in wet weather. The large incident hemisphere, which Sekonic calls Lumisphere, is mounted on a swivel head that can be rotated through 270°. This makes it easy to keep the meter display visible no matter where the Lumisphere is pointed.

Because the Lumisphere retracts, it is protected during travel. This also emulates a flat diffuser in this position and eliminates the need for a separate part. Like the L-408, the L-508 will meter both ambient light or flash. The L-508 can store up to three readings in memory. This makes it easy to take both an ambient and a flash reading and compare the two so that easy adjustments can be made to achieve the desired balance between the two. When desired, you can take three readings from highlights, shadows, and mid tones, and by pressing the Ave/EV button, determine an average of the readings. The meter will also allow you to measure the difference in brightness between two readings in 1/10 step increments.

Cords Be Gone

When working in the studio I have a policy—the fewer the number of cords on the floor the better. So I appreciate that the L-508 allows cordless flash operation. When used in this mode, you press the metering button to activate the meter and then pop the flash to get the reading. The 1/10 step display makes fine-tuning flash setups quick and easy. The L-508 will also give correct readings when using multiple pops of the flash system. You can measure up to nine pops in this way and get a reading which is more accurate than just assuming that flash pops are cumulative. In theory they should be, but in practice some flash units do not deliver constant power when used for multiple pop lighting. If you are using two different films during a photo shoot, the L-508 can be set for two different ISO speeds and you can quickly switch between them at the touch of a button. You do not have to take a new reading to get the meter to display information for the second ISO. You need only press the ISO2 button and it is recomputed automatically and displayed as long as you press the button.

Custom Functions

As with many modern cameras, the L-508 meter has custom functions. Inside the battery compartment there is a row of DIP switches. Using these, you can set the meter to match your personal requirements. You can set the shutter speed display to change in increments of one full speed, or in half step increments. You can select shutter or aperture priority for the read-out. You can customize to pre-select f/stop or EV read-out for easy comparison of brightness range. You can pre-select multiple flash measurement. By choosing the proper settings for your shooting style, the meter will always be ready to work the way you want when you pick it up

Getting Better All the Time

Once again, after using the L-508 for a while I just didn't see how they could possibly improve on it. And once again I was wrong. At photokina last year, Sekonic showed their newest light meter, the L-608 Super Zoom Master, and I was pleased to see that their engineers had once again improved their meter in important ways. One of the major changes has a lot to do with their US distributor. In the U.S.A. the Sekonic line of meters is distributed by Mamiya America. Besides the obvious of distributing Mamiya professional cameras, lenses, and accessories, Mamiya America has broadened their base and also distributes other professional photographic products through their various divisions. Sekonic is one division, and Profoto and PocketWizard are two others.

Unlike some distributors, the people at Mamiya America are proactive, working with the companies whose products they distribute and trying to encourage synergies among them. In this case they realized, as I mentioned earlier, the fewer cords on the studio floor the better. Even when using the L-508 and other Sekonic meters in their cordless mode you still have to fire the flash. Working with Sekonic, Mamiya, and FlashWizard, they have come up with some remarkably useful solutions to this problem.

Inside the battery compartment of the L-608 meter is a slot. As an option you can get a card which slips into this slot, and on the card is an ultra-miniature radio transmitter which can be used with a FlashWizard as a receiving unit to allow firing the flash from the meter. Neat idea, and very practical.

Additionally, as a service, Mamiya America will be able to modify some models of Mamiya cameras and "graft" this same ultra-miniature radio transmitter into the camera. Once equipped in this way the photographer can test fire the flash system from the meter while setting up the lighting, and can then synchronize the flash with the camera shutter, all fully wireless. If you've ever tripped over a cable in a studio and pulled a light stand over or knocked a camera to the floor you will appreciate the protection this can offer to equipment, plus the lowered liability of possible injury to people in your studio. Wireless communication is supposed to be the wave of the future, and now you can have it in your photo studio.

If that was the only new trick Sekonic had taught the L-608 meter it would be plenty, but the new meter offers a lot more on top of wireless flash triggering. With previous Sekonic meters you could measure ambient light or you could measure flash, but you could not do both in one meter reading. You had to change modes. The L-608 now offers the ability to measure flash and ambient light in one reading. It gives you the information you need to know in three different forms on the LCD panel; combined reading of both ambient light and flash, percentage of flash in total exposure, and simultaneous display of flash, ambient light, and combined readings.

No matter how you want to measure it, this meter will do it for you. Since I do a fair amount of mixed light photography I am seriously considering upgrading from my L-508 for this greater convenience.

While the L-508 can memorize three readings as mentioned earlier, the L-608 ups the ante again with nine. I can't imagine a setting in which nine wouldn't be enough. Also, if you use exposure compensation, with the L-608 you can now enter different values for the reflected and incident systems.

Although the L-508 has the same variable measuring angle for its spot meter you must take a reading with it and then look at the display on the side of the meter to find out the reading. The L-608 has the increased convenience of a digital display inside the viewfinder, making it quicker to take and mentally evaluate multiple

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Although the L-508 has the same variable measuring angle for its spot meter you must take a reading with it and then look at the display on the side of the meter to find out the reading.

All in all, I love my L-508 and have become really familiar with it. It lives in my camera bag and goes with me on all important shoots. Now that I have had a chance to work with the L-608 for a while for this article, I find that I like it even more, particularly for work in the studio. I may just have to upgrade meters and convert my studio into a modern wireless wonder. I think I would like that very much.

These meters are distributed by the Sekonic Division of Mamiya America, Inc. (914) 347-3300, fax: (914) 347-3309.

18%, Or 13% Gray?

One last bit about metering basics. You often hear and read about 18 percent gray being the standard to which meters are calibrated. I'm not sure just where this misconception came from, but the fact is that it is just plain wrong. Meters today are factory calibrated using an ANSI standard which is about 12.5 percent to 13 percent reflectance. They are not calibrated using an ISO standard because the ISO has adopted no standard for light meter calibration, apparently preferring to simply leave the old ANSI standard in place. This 12.5 percent to 13 percent is about 1/2 stop less reflectance than 18 percent, and for many films this is not much of a problem as their latitude covers up exposure errors this small. However, when working with narrow-latitude transparency films and some digital sensors, this can be a serious enough error to ruin images. If 18 percent is not the standard, why then does Kodak make their Gray Card to have precisely 18 percent reflectance? No one seems to be 100 percent sure at Kodak today, but some of the few who were around in earlier days point their fingers at Ansel Adams, who lobbied hard for 18 percent. I'll leave the details of this to the historians. If you read the instructions that come with Kodak's Gray Card manufactured since 1999 you will find that the instructions tell you how to hold the card properly at an angle and to make the 1/2 stop correction. The Kodak Professional Data Guide tells you the same. Those who want to know more about all this can find the full story in The Hand Exposure Meter Book which I co-authored with Martin Silverman and Jim Zuckerman.

In-Camera Meter Conundrums

We hear a lot today about multi-point or matrix metering systems used in advanced cameras, and I must admit that they work for many difficult situations and produce excellent exposures. The problem with such systems is twofold. First when dealing with certain subjects they just don't work, and second, and most important, you don't know when they won't work. For an amateur shooting snapshots this may be acceptable, but for the working pro who must deliver a properly exposed shot every single time those times when the system fails cannot be accepted. This is why professional photographers and serious amateurs always carry a hand meter no matter how sophisticated their camera.

A Case For Incident Meters

In these days, when just about every SLR and rangefinder camera has a sophisticated light meter built-in, I often hear people wondering why anyone would need a handheld light meter. There is a simple reason, really. All meters built into cameras read the light reflecting from the subject. Proper exposure, on the other hand, is determined by the light falling on the subject. This is true because every subject has a different reflectance, both in terms of the percentage of the light that it reflects and also in terms of the colors of the light it reflects. Any meter which measures light reflecting from the subject must be built based on assumptions of how much light is reflected from the subject must be built based on assumptions of how much light is reflected from the subject. If your subject is average, that is, it reflects the amount of light assumed by the designers of the meter, then your exposure determined with a reflectance meter will be correct. If your subject reflects more than average your reflectance meter will still be working on the same assumption and will give you a proper exposure to render your subject as average in reflectance, and this will result in underexposure.

A good example is snow. If you take a reflectance meter reading from snow the meter will give you the proper exposure to render the snow as average in reflectance and it will turn out gray in your photos and everything else will also be rendered far too dark. Conversely, photographing the proverbial black cat in a coal bin and using a reflectance meter will result in an exposure that renders the black cat and coal as average in reflectance, so will render both as medium gray and dramatically overexpose the image.

Incident Meters

If cameras could just meter the light falling on the subject this problem would be solved, but that is just not practical. A light meter used to measure the light falling on the subject is referred to as an incident meter because it measures the light incident to the subject. Such a meter is easily recognized by its hemispherical white dome over the metering cell. This white dome represents a three-dimensional subject, and simultaneously measures all light falling on it from a 180Â° angle. The only 35mm camera system ever to offer such a metering system was Topcon, who offered a white hemisphere in a filter ring that screwed onto the front of a lens. This converted the camera meter into an incident meter, and worked well. In medium format, Hasselblad offers a metering prism with a small incident dome on it, allowing it to be used as an incident meter.

When you want to measure incident light as it would fall on a flat surface, such as when doing copy work of flat pages, some incident meters come with an optional flat white diffuser, which replaces the hemispherical one. The approach Sekonic takes on some of their meters is to accomplish the same thing by retracting the hemispherical dome into the meter. This also eliminates the need to keep track of a separate piece.

Technical Specifications

Sekonic L-508 Zoom Master

Incident: 270Â° Swivel Head with Dual Function Retractable Incident Lumisphere
Reflected: 1Â° to 4Â° Parallax-free Zoom Spot Viewfinder
Photo Cells: Two Silicon Photo Diodes
Exposure Compensation: +/- 9.9 EV in 1/10 EV increments
Auto Shutoff: After 20 minutes
PC Terminal: Standard
Tripod Socket: 1/4-20 standard U.S.A. type
Power Source: 1 AA cell (alkaline-manganese or lithium)
Weight: 7.4 oz (210 g)
Dimensions: 84x156x40mm (3.3x6.1x1.6")
Measuring Range At ISO 100
Ambient Incident: EV -2 to EV 19.9
Ambient Reflected: EV 3 to EV 19.9
Flash Incident: f/1.0 to f/128 + 0.9
Flash Reflected: f/5.6 to f/128 + 0.9
Number of Multiple Flashes: Up to nine
Display Range
Film Speed (ISO): ISO 3 to 8000
Increments: 1/3 steps
f/Stop Range: f/1.0 to f/128 + 0.9

Increments: 1/10 step
Shutter Speed Range: Dip Switch pre-selectable full speeds or 1/2 step range
Ambient Speeds: 30 minutes to 1/8000 sec
Special Ambient Speeds: 1/200, 1/400
Flash: 1 sec to 1/1000 sec
Special Flash Speeds: 1/75, 1/80, 1/90, 1/100, 1/200, 1/400
Exposure Value Range (EV): EV -9.9 to 36.1

Sekonic L-608 Super Zoom Master

Incident: Convertible to flat diffuser (Lumisphere in down position)
Reflected: 1Â° to 4Â° nine element zoom
Light Receptors: Two Silicon Photo Diodes
Metering Modes
Ambient Light: Aperture priority metering; shutter speed priority metering; and EV metering value
Flash: With sync cord (cumulative, non-cumulative); without sync cord (cumulative, non-cumulative); and with radio triggering (cumulative, non-cumulative) Radio Triggering (Optional)
Radio Technology: Complex 16/24 bit digitally coded
Range: Up to 100 ft from transmitter to receiver
Channels: 1 to 16 channels standard, 17 to 32 “Quad Triggering”
Measuring Range At ISO 100
Ambient: Incident light: EV - 2 to EV 22.9; Reflected light: EV 3 to EV 24.4
Flash: Incident light: f/0.5 to 128.9; Reflected light: f/5.6 to 128.9
Display Ranges
Film Speeds: ISO 3 to 8000 (1/3 steps)
Shutter Speeds: 30 minutes to 1/8000 sec (full, 1/2, or 1/3 steps)
f/Stop: f/0.5 to f/128.9 (full, 1/2, or 1/3 steps)
EV: EV - 9.9 to EV 40.6 (1/10 steps)
Analog Display: f/1.0 to f/128 (in 1/2 steps)
Digital Display: f/0.5 to f/128.9
All-Weather Design: JIS (Japanese Industry Standard) standard water-resistance class 4, splash-proof type
Memory Function: Nine readings on analog scale (f/stop and shutter speed) with memory recall and clear feature
Multiple Flash Function: Unlimited flash readings (only last digit is displayed when more than 10 flash exposures are measured)
Average Function: Flash and ambient up to nine measurements
Flash Analyzing Function: Percentage of flash in total exposure from 0 to 100 percent in 10 percent increments
Brightness Difference: Flash or ambient light evaluation (original value vs. new measurement)
Exposure Out Of Range: E.u (underexposure) or E.o (overexposure) indication
Auto Shutoff: After 20 minutes
Exposure Compensation: Â± 9.9 EV for incident and reflected independently
Calibration Compensation: Â± 1.0 EV for incident and reflected independently
Filter Compensation: Â±5.0 EV on ISO2

Custom Programs: 11 custom program settings for frequently used features (12 on L-608 Cine)
Power Source: 3.0vx1 (CR123A lithium battery)
Dimensions: 3.5x6.7x1.9" (90x170x48mm)
Weight: 9.5 oz (268 g)

Sekonic L-358 Flash Master

The Sekonic L-358 Flash Master is an advanced, yet easy to operate, exposure analyzing light meter, that incorporates the latest technology in handheld meters

Sekonic L-358 Flash Master

Description
The Sekonic L-358 Flash Master is an advanced, yet easy to operate, exposure analyzing light meter, that incorporates the latest technology in handheld meters

Minolta

Minolta exposure meters have long been world-renowned for accuracy, repeatability, and reliability, and are used by more professionals than any other meter.
Professionals around the world rely on Minolta's quality and precision – and you can too. When you truly care about the quality and precision of your photographic images, Minolta meters give you a creative advantage.

Auto Meter IVF

Equipped with memory, averaging, and monitoring functions, the lightweight and compact Auto Meter IVF measures both flash and ambient light, and includes the important functions to help simplify your most critical applications in photography or cinematography, in the field or in the studio. This easy-to-use meter's many accessories make it ideal for a wide range of uses.

Precise and Versatile Operation

A wide range of optional accessories is available for use with the Auto Meter IV F to further expand its applications and versatility. With ambient light or flash, and with or without a sync cord, the Auto Meter IV F is designed to provide unparalleled repeatability and accuracy. In all measuring modes, with reflected or incident readings, it is a meter you can rely on completely, leaving you free to concentrate on the creative challenges at hand.

Large, Easy to Read LCD Display

The LCD display shows all the necessary information at a glance. Shutter speed appears on the left, and aperture on the right. The aperture display is oversized, for easy identification.
Extended Operating Range
Shutter speed range is 1/8000 sec. –30 min., in ½-stop increments. Film speed range is ISO 3-8000, and cine speeds range from 8 to 128 fps. The X-sync range is 1/500 – 1 sec., in ½-stop increments. Both f/numbers and EV numbers are displayed in analog scale, and digitally to 1/10th-stop accuracy.
User-Friendly Controls and Design
The compact and lightweight Auto Meter IV F fits the hand comfortably. Rubber buttons are positioned in a straightforward, easy to use configuration and help keep the meter dust and weather resistant. A rubber-lined frame assures a sure grip while providing additional protection.

2-Channel Memory

The Auto Meter IV F's memory stores two separate readings and displays them simultaneously, making it easy to determine subject contrast range or lighting ratio.

Averaging of Memorized Data

Calculate and displays the average of two stored measurements. This is useful when basing exposure on highlight and shadow readings.

Brightness Difference Readout

Displays the difference in +/-EV, and in 1/10th stops, between two readings. This helps you make quick and precise lighting adjustments.

Flash Reading With or Without a Sync Cord

With a sync cord connected, a press of the measuring button fires the flash and takes an exposure reading. Without a sync cord, pressing the measuring button puts the meter into standby mode until the flash is fired manually.

Ambient Light Readings

Simply set the film speed and shutter speed, position the meter, and press the measuring button to take a reading. Hold the measuring button down to take continuous readings throughout the subject area.

Cine Measurements

For cinematographers, the Auto Meter IV F has eight framing rate settings. Plus, the extended ISO range can be used to factor in filtration correction.

Accessories
A wide range of optional accessories is available for use with the Auto Meter IV F to further expand its applications and versatility.

Minolta's Spotmeter F

Minolta's Spotmeter F is designed for professionals and advanced amateurs alike. While looking through the meter's viewfinder, you can take precise, 1-degree spot measurements of flash or ambient light, to check the lighting on a small part of your subject. Three exposure-calculation functions (highlight biasing, shadow biasing, and averaging) can be used to determine the exposure of virtually any subject or scene.

Precise 1° Spot Readings with Versatility
Minolta's Spot Meter F lets you take precise 1° spot measurements of ambient or flash light. Liquid-crystal displays on the meter's side panel and in the viewfinder show a full range of exposure information in both digital and analog form. Exposure calculations for highlights, shadows, or midtone areas can be made at the touch of a button. Plus, the lighting contrast can be quickly checked while looking through the viewfinder.

Accurate Flash Measurements
Flash measurements are no problem with the Spot Meter F. Simply select flash mode, connect the flash sync cord, and set the X-sync speed (1/1000 to 1 second). To take a reading, center the finder's 1° spot circle on the subject and press the measuring button to fire the flash. The required f/number (with tenth-stop accuracy) is digitally displayed in both the finder and LCD data panel. Ambient readings are just as easy – just set the shutter speed, then press the measuring button.

Brightness Difference Measurements
After determining the proper exposure settings, the user can quickly check the lighting contrast between the metered area and other areas. When the highlight, shadow, or average key is pressed, a "star" mark appears along with the digital exposure readout. By holding the measuring button, the meter displays the difference in brightness between the metered area and the area currently being measured. When the measuring button is released, the digital exposure readout is displayed again.

Three Exposure Calculation Choices
The Spot Meter F can automatically calculate the exposure for highlight or shadow areas. And, by using the memory function, the user can average two readings for a midtone exposure.

Digital and Analog Displays
Exposure readings are shown digitally in the meter's data panel and viewfinder displays. The data panel also has an analog f/number scale that can indicate up to four exposure readings. This full range of information makes it easy to determine the optimum exposure in nearly any situation.

Data Storage Function
When the Spot Meter F is switched off, the ISO and shutter speed settings are stored in memory. When the meter is switched back on, the most recent digital exposure readout and any memorized data is displayed.

Specification:
- Type: Spot-reading reflex-viewing exposure meter for ambient or flash light.
- Measuring Method: Reflected light by silicon photocell detector masked for 1° angle of acceptance.
- Optical System: Through-the-lens reflex type utilizing semi-silvered mirror and pentaprism; focus-fixed for readings 4-1/4 ft. to infinity; with optional close-up lens, 23.6 in. to 4-1/2 ft. Viewfield: 12 x 17° with 1° area marked by circle in finder. Magnification: 1.4X. Eyepiece adjustment: -2.5 to +1.2 diopters.
- Measuring Range at ISO 100: Ambient: EV 1.0 to 22.5; Flash: f/2 to 90 + 0.9 stop.
- Accuracy: +/-0.1 stop repeatability
- Electronic Components: Hermetically sealed microprocessor chip and two custom designed liquid crystal displays; display on side of unit has separate 3-digit readout and 4-digit input sections (each with unit identifications) and analog array; LCD in finder shows EV, f-number, or brightness difference.
- Controls: Measuring button (operates only when "TIME" is displayed); key to alternate film speed/shutter speed display; increase and decrease keys for changing film speed and shutter speed; f-number/EV display selection key; memory, recall, and memory-clear keys; highlight, shadow, and averaging calculation keys: ambient/flash mode switch; power switch; viewfinder display illumination button
- Digital Readouts/Displays: F-numbers: f/0.7 to 90 + 0.9 stop in 1-stop increments. EV numbers: -4.3 to +28.5 in 0.1-stop increments. Brightness difference: -9.9 to +9.9 stops in 0.1-stop increments. ISO range: 12 to 6400 in 1/3-stop increments. Ambient exposure times: 30 min. to 1/8000 sec. in 1-stop increments (cine: 1/50 sec.). Flash exposure times: 1 to 1/1000 sec. in 1-stop increments. NOTE: F-number, EV number, and brightness difference shown in both external and finder displays.
- Analog Readouts/Displays: F-numbers: f/1.4 to 45 in ½-stop increments (up to 4 indications possible when using memory/calculation functions).
- Other Indications/Displays: Analog and digital display readouts change automatically to reflect ISO/time input changes; "S", "A", or "H" on external display indicates exposure is calculated for shadows, average (midtone), or highlight areas respectively; "star" mark appears when reading on digital display is fixed for taking brightness difference measurements; flash mark appears when using flash mode.

- Brightness Difference Function: When "star" mark appears in external/finder digital displays, difference in brightness between calculated measurement and subsequent readings is shown in 0.1-stop increments; calculated reading displayed again when measuring button is released.
- Memory: 2-measurement capacity, both indicated by pointers on analog display; digital recall possible.
- Exposure-Zone Calculation: Analog/digital readout and recall of highlight, shadow, or averaged (midtone) exposures automatically calculated for optimum correspondence of brightness range of subject with film latitude.
- Power Source: One 1.5V AA-size alkaline manganese (Eveready E91 or equivalent), carbon-zinc, or 1.2V NiCd cell.
- Other: Threaded PC-type terminal for connecting flash sync cord, tripod socket, and strap eyelet, ISO table, "cine" table, luminance conversion table.
- Accessories: Neck strap, lens cap and belt case supplied with meter; close-up lens and Sync Cord II available separately.
- Size: 1 7/8 x 5 7/8 x 3 9/16 in.
- Weight: 8 ½ oz. without battery.

Notes

The Flash Meter V

The Flash Meter V is the world's first intelligent exposure meter. With an innovative intelligent AUTO mode, it distinguishes flash light from ambient light, and cord/non-cord operation automatically, while providing unparalleled precision and ease of use for the most demanding photographer or cinematographer. This meter raises the standard for high performance while creating a new standard for user-friendliness.

Minolta Meters - The Professional Standard
Minolta sets the industry standards for exposure meters. Minolta exposure meters have long been world-renowned for accuracy, repeatability, and reliability, and are used by more professionals than any other meter. The Minolta Flash Meter V continues this standard of excellence, while creating a new standard for ease-of-use.

Intelligent AUTO Mode
The Flash Meter V is the world's first intelligent exposure meter. Simply press the measuring button. The intelligent AUTO mode distinguishes flash light from ambient light, and cord/non-cord operation automatically, without changing measuring modes. If both flash and ambient light are present, the percentage of flash light in the total light is automatically displayed in the meter's analyze scale.

Easy Operation
The Minolta Flash Meter V is the world's easiest to use exposure meter. For most applications, AUTO mode is all that is required. Shutter speed and aperture values are changed by a positive-action dial. Additionally, pressing the unique instant-film ISO button displays the exposure requirements when using films of different sensitivity (for example, Polaroid film for test shots). Plus, the data panel lights automatically in low light or darkness.

Simple Design
The Flash Meter V features a simplified design in which switches that are used less often are located under a sliding cover. In addition, switches for setting basic meter functions (which will be changed rarely if at all) are hidden away inside the battery chamber.

ISO Button for Polaroid Test Films
This convenient feature displays the exposure requirements for instant films used for test shots, which are often a different sensitivity than the regular film.

Analyze Function
If both flash light and ambient light are present, the percentage of flash light in the total light is automatically displayed. Each line on the analyze scale (located below the f/no. readout) indicates 10%. For example, when 6 lines are shown on the scale, flash light makes up 60% of the exposure and ambient light makes up the remaining 40%.

Brightness Difference Function
The Flash Meter V's brightness difference function helps the photographer balance lighting or check for uneven lighting when taking product shots, portraits, etc. Simply measure the reference point, press the * (brightness difference) button, and then measure other areas of the scene. The meter will display the brightness difference in terms of EV in 0.1EV increments, or the Illuminance difference in terms of lux or fcd.

AMBI Mode
For shutter-priority measurements of ambient light. This mode should be used when taking measurements under flickering light sources (such as fluorescent lights), which may be mistaken for a flash light source in AUTO mode.

AMBI F No. Mode
For aperture-priority measurements of ambient light.

Flash Mode
For single-burst flash measurements. This mode should be used when the meter must be left in position and cannot be operated while the flash is fired. In this mode, pressing the measuring button sets the meter to flash standby for 1 minute; in AUTO mode, the meter is in flash standby only while the measuring button is held pressed.

Flash Multi Mode
For cumulative measurement of multiple flash bursts.

Memory Function
Up to 8 measurement values can be stored in memory by pressing the M (memory) button. Values stored in memory are displayed on the analog scale.

Exposure Calculation Functions
The Minolta Flash Meter V is equipped with three exposure calculation functions: Shadow, Average, or Highlight. Average can be used for both incident and reflected light measurements; Highlight and Shadow can be used for reflected light measurements only. To select the desired mode, simply press and hold the S/A/H button and turn the up/down dial to display either S, A, or H in the data panel.

Average

To display the average of up to eight readings stored in memory, simply press the S/A/H button. The average value is shown only while the S/A/H button is held pressed; releasing the S/A/H button returns the meter to normal measurement mode.

Highlight-biased measurements

If several points have been measured and stored in memory, the maximum value (the value for the brightest measured point) will be selected. In highlight mode, the exposure is increased by 2.3 stops (default value) to produce a reading that will render the measured area as a highlight in the photo. The highlight-biased value is shown only while the S/A/H button is held pressed; releasing the S/A/H button returns the meter to normal measurement mode.

Shadow-biased measurements

If several points have been measured and stored in memory, the minimum value (the value for the darkest measured point) will be selected. In shadow mode, the exposure is decreased by 2.7 stops (default value) to produce a reading that will render the measured area as a shadow in the photo. The shadow-biased value is shown only while the S/A/H button is held pressed; releasing the S/A/H button returns the meter to normal measurement mode.
NOTE: The meter's default values for highlight and shadow exposure calculation can be adjusted between 0 and +4 stops for highlight and 0 and -4 stops for shadow by the user, to fine-tune the calculation for the film type or the final medium (photographic reproduction, TV, or print).

Measuring Illuminance

To set the Flash Meter for cinematography use, set DIP switch 1 down. Use the up/down dial to set the desired framing rate (8, 12, 16, 18, 24, 25, 30, 32, 64, or 128 fps). To set the meter for normal photographic readings, position DIP switch 1 up. For Illuminance-only readings, set DIP switches 1 and 2 down.
To switch Illuminance units between lux (lx) and foot-candles (fcd), hold the MODE and M buttons pressed while switching on the meter.

Measuring Level Adjustment

The Flash Meter V is calibrated to Minolta standards. However, the user can adjust meter readings between +0.7 and -0.8 (in 0.1 step) with the measuring level adjustment screw. Turning the screw to the right increases the indication (less exposure), and turning it to the left decreases indication (more exposure).

Flash Meter VI

Minolta have announced the Flash Meter VI high performance exposure meter. The Flash Meter VI newly incorporates a spot meter function into the top end model of the MINOLTA range which has earned a reputation for excellence.
The MINOLTA FLASH METER series has earned the support of professional and amateur photographers as multifunctional tools for professional use. The Flash Meter VI features a compact and stylish design using a dioptric system and newly incorporates a high performance spot meter function in addition to the conventional analyse function, brightness difference function, exposure calculation function and other functions. Furthermore, the large size LCD can display a film latitude together with a measured value, enabling users to simulate how the subject is reproduced on the film instantaneously, while displaying the incident light measurement result and reflected light measurement result (spot meter reading) simultaneously.
In the field of photographic exposure meters, which is the base of our measuring instrument business, Minolta has provided high precision and highly reliable products continuously for about 40 years as a leading company in this industry.

Main Features

Integration of high performance spot meter

The Flash Meter VI incorporates a high performance spot meter function with an acceptance angle of 1°, in addition to the conventional functions of the Flash Meter series. The Flash Meter VI not only functions as two different meters in a single unit, but also displays the incident light and reflected light measurement results simultaneously and compares them by using the latitude display function. (See Item 3)

Compact and stylish design

The built-in spot meter uses a high precision dioptric unit based on MINOLTA's advanced optical technology, providing a compact and stylish body.
Exposure decision process navigation using latitude display function
The latitude range based on the standard exposure measured by the flash meter (incident light measurement) is displayed on the dot indicator of the analogue scale. Simultaneously, the spot meter reading (reflected light measurement) is displayed on the dot indicator of another analogue scale. The user can visually check how each part of the subject is reproduced on the film by confirming the difference between the measured values for highlight and shadow areas of a subject and the standard exposure. Conventionally, the exposure decision process significantly depends on the photographer's experience and know-how. With the Flash Meter VI, however, the user can easily determine the most suitable exposure for the intended photographic image, because the exposure decision process can be instantaneously confirmed on the LCD.
Multifunctional meter providing analyse function, brightness difference function, etc.
Analyse function: With a single flash light measurement operation, the Flash Meter VI can display the ratios of the flash light and ambient light on the LCD. The user can control the ratio of the flash light by changing the shutter speed setting, to

produce the desired lighting effect. The analyse function uses a new quadrant scale for improved legibility.
Memory function: With a push of the MEMORY key, the Flash Meter VI can store measurement data on up to 10 points.
Brightness difference function: The Flash Meter VI displays the exposure difference between a predetermined standard value and the measured value on target points on the EV value display and the dot indicator of the analogue scale.
Exposure calculation (S/A/H) functions: According to the intended photographic image, the Flash Meter VI provides three types of exposure calculations (Shadow based exposure, Average exposure and Highlight based exposure).
Custom settings mode: The display mode can be customised according to the user's preference. The selectable shutter speed increment setting is "1-stop", "1/2-stop" or "1/3-stop" increments. The selectable FNo. display mode is the conventional "intermediate stop display (1/10-stop increments)" or the "f-number direct-reading"** mode.
** The f-number direct-reading mode is useful for digital cameras providing intermediate f-number settings (e.g. F3.5, F6.5) that cannot be selected with conventional film cameras.

Technical Specifications

- Type: Digital exposure meter for measuring flash light and ambient light
- Reception methods: Incident light and reflected light (spot)
- Receptors: Incident light: Spherical Diffuser/Flat Diffuser*, 270° rotating receptor head
- (* Optional accessory)
- Reflected light (spot): Acceptance angle of 1°
- Receptor element: Silicon photocell
- Measuring modes: AMBI mode: Ambient light measurement
- CORD mode: Flash light measurement with sync cord
- NON.C mode: Flash light measurement without sync cord (for incident light only)
- Measuring range
- (ISO 100)
- Ambient light: Incident light: Ev -2.0 to 19.9
- Reflected light (spot): Ev 2.0 to 24.4
- Flash light: Incident light: FNo.1.0 to 128 + 0.9 stops
- Reflected light (spot): FNo.2.8 to 128 + 0.9 stops
- Measuring distance: 1.3 m to infinity (¥) (for spot measurement)
- Viewfinder: Single lens reflective type with fixed focal point
- Magnification: 1.2x
- Viewing angle: 12° (vertical) x 17° (horizontal)
- Dioptric adjustment range: -3.0 to +1.0
- Repeatability: ±0.1 Ev
- Calibration coefficients: Incident light: C = 330 (Spherical Diffuser),
- C = 250 (Flat Diffuser)
- Reflected light (spot): K = 14
- Display range: f-number (FNo.): F1.0 to F128 + 0.9 stops
- (0.1-stop increments)
- Exposure value (Ev): -17 to 40.9 (0.1-stop increments)
- Shutter speed: Ambient light: 30 min. to 1/16000 sec., Flash light: 30 min. to 1/1000 sec. (1-, 1/2- or 1/3-stop increments)
- Frame rate (Opening angle of 180°) 8, 12, 16, 18, 24, 25, 30, 32, 64, 128
- ISO: 3 to 8000 (1/3-stop increments)
- Exposure difference: -10.0 to +10.0 (0.1-stop
- increments)
- Analog scale: FNo.1.0 to F90 (1/2-stop increments)
- Analyse scale: Flash light proportion 0 to 100% (25% increments)
- Other functions: Latitude display function, Analyse function, Memory function (10 measured values), Exposure calculation function (S/A/H), Brightness difference function,
- Exposure correcting function: -10.0 to +10.0 stops
- (0.1-stop increments)
- Power source: Single AA alkaline dry cell
- Battery service life: Approx. 30 hours (for ambient light/ incident light
- continuous measurement using alkaline battery)
- Operating temperature/ Temperature: -10 to 50°C (14 to 122°F)
- humidity range: Relative humidity: 85% max. [at 35°C (95°F)]/no
- condensation
- Storage temperature Temperature: -20 to 55°C (-4 to 131°F)
- & humidity: Relative humidity: 85% max. [at 35°C (95°F)]/ no
- condensation
- Other: Sync terminal
- Dimensions: 63 (W) x 175 (H) x 30 (D) mm
- Weight: 170 g (excluding battery)
- Standard accessories: Spherical Receptor, Neck strap, Case,
- Single AA dry cell
- Flat Diffuser, Sync Cord III

Gossen Spot Attachment for Luna-star

Features

Transforms Luna-Star F2 into a 5 degree spot meter. Incredibly convenient for analyzing the subject contrast or metering the most important areas of the subject. With Parallax correction in the viewfinder for close ups of less than 3 feet.
The world's smallest meter for measuring flash and ambient light has been announced by German maker Gossen.
The new Gossen Digiflash which measures only 75x50mm and weighs a mere 40g, combines digital and analogue technology to produce highly accurate readings on the LCD display. The EV range is from 0 to 18. For ambient light, the Digiflash can give incident or reflected light readings, showing all combinations of shutter speed and aperture on teh scale of the dial at a glance.
The main functions of the Digiflash are easily controlled by two buttons, one to take a reading and one to change the function.
In addition to usual meter features, the Digiflash also incorporates a clock and timer together with a highly accurate thermometer.
The digiflash comes complete with a neat soft pouch to slip into bag or pocket and is priced at £139.99.

Gossen Digisix Digital Exposure Meter

New from Gossen and available exclusively from UK distributors Sangers Consumer Products the Digisix is a digital exposure meter for ambient light packed with features.
Weighing 40g and measuring 75x50x23 the Digisix is portable enough to accompany any photographer on the move. According to the press release, the metering system is one of the most advanced ever seen in such a small meter: a sbc diode with a measuring range of EV 0 to 18. Incident and reflected light can be recorded, with the last values retained in memory. Adjustments to the ISO ratings are automatically recalibrated into the exposure reading.
The operation of the Digsix is particularly easy, and will be familiar to users of Gossen meters, the measurement is taken and then applied to the internal setting window. Once this has been done, all the available shutter speed/f-stop combinations can be seen at a glance. With the press of a button, the metering mode can be changed to contrast measurement for total control of the shot. In all modes, fine adjustments to 1/3 stop can be made and recorded.
The Digisix also comes with a number of features for the field photographer - An inbuilt countdown timer for long exposures, a thermometer for ensuring that film is used in correct tolerance ranges and a clock (with alarm) for recording the time of the shot.
The Digisix has a SRP of £79.99.

Notes

Sensitometry is the science of measuring the response of photographic emulsions to light. “Image-structure” refers to the properties that determine how well the film can faithfully record detail. The appearance and utility of a photographic record are closely associated with the sensitometric and image-structure characteristics of the film used to make that record. The ways in which a film is exposed, processed, and viewed affect the degree to which the film’s sensitometric and image-structure potential is realized. The age of unexposed film and the conditions under which it was stored also affect the sensitivity of the emulsion. Indeed, measurements of film characteristics made by particular processors using particular equipment and those reported on data sheets may differ slightly. Still, the information on the data sheet provides a useful basis for comparing films. When cinematographers need a high degree of control over the outcome, they should have the laboratory test the film they have chosen under conditions that match as nearly as possible those expected in practice.
So, Sensitometry as a photographic science has been credited to two Americans, Hurter and Driffield, who first proposed a form of evaluation of the performance of photographic material in the late 19th century. Sensitometry requires that the photographic emulsion be exposed to a specified light source for specified times and then the emulsion is processed under closely controlled conditions. The resultant densities produced on the film are then measured and plotted against a logarithmic scale of the exposure. The standard illuminant is an incandescent electrical light source of a specified brightness and spectral emission.
The main application in radiography is how the film will react to X-radiation, and to the light produced by the intensifying screens. The exposure of the photographic emulsion
to controlled exposures and controlled processing conditions is a fundamental of all hotographic procedures. It is imperative that the student appreciates the factors which are involved and understands simple logarithmic relationships.

- Dense - describes a negative or an area of a negative in which a large amount of silver has been deposited.
- Densitometer - instrument for measuring the density of silver deposits on a developed image by transmitted or reflected light.
- Density - amount of silver deposit produced by exposure and development. It is measured in terms of the logarithm of opacity, where opacity is the light stopping power of a medium.

How to Reduce Density

Density can be defined as: The amount of blackening on the film.As a statement of fact this is correct. Mathematically, however, density is a logarithmic value. It can easily be obtained by eriving two simple ratios from Figure 8.1. No light-transmitting material is completely transparent, and therefore some light is always absorbed in its passage through the material. InFigure 8.1, the light travelling from the light source (the incident light, I) has been reduced in intensity as it asses through the X-ray film to the eye of the viewer (the transmitted light, T).

Opacity

$I/T = 100/10 = 10$ The value 10 is known as the opacity. Opacity will always be greater than unity and an reach quite high values. It is, again, a term which is not used (correctly) in radiography.
It is important to realise that in the example above, simple integers are used. Opacity could be expressed to two or three significant figures. Additionally, the figure obtained from the ratio would prove difficult to illustrate graphically. It is easier to make a graphical representation of numerical values when the numbers are expressed as logarithms.

Density

Consider the value of 10, obtained for opacity. The log value of 10 is 1.0. This log value, 1.0, is
the density of a film when the opacity is 10, therefore: Density is the log of the opacity. There are some important relationships about density; these are shown in Figure 8.2 and are as follows:

- That the silver weight of density on the processed film is related linearly to the blackeningon the film, a factor which assumes great importance if films are being sold for silver recovery. A film ith high densities (e.g. an extremity film) will contain significantly largeramounts of silver per unit area than, for example, a chest film.
- There is an interrelationships between the values, and opacity is going to reach very highnumbers when the value of density is 4 (i.e.opacity will be 10 000).
- Only 1% of the incident light reaches the viewer’s eye at density. In other words, 99% of the incident light is being absorbed by the film. In addition:
- Density always increases with exposure (with one exception, solarisation, see p. 165).
- The eye responds to tonal changes logarithmically. (This relationship can only be accidental, but it helps.)
- Density, because it is logarithmic, is very easy to manipulate mathematically. Density 1 + density 1 = density 2.

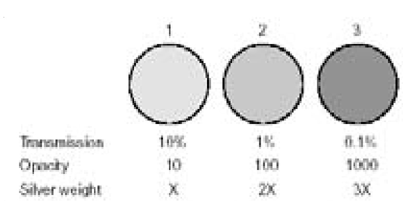


Fig. 8.2 Density: interrelation of silver weight, opacity and transmission.

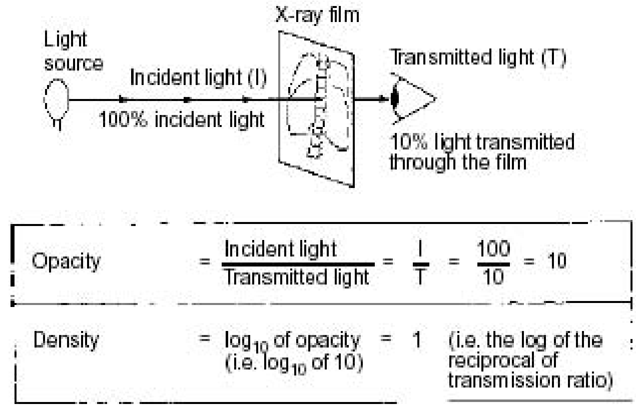


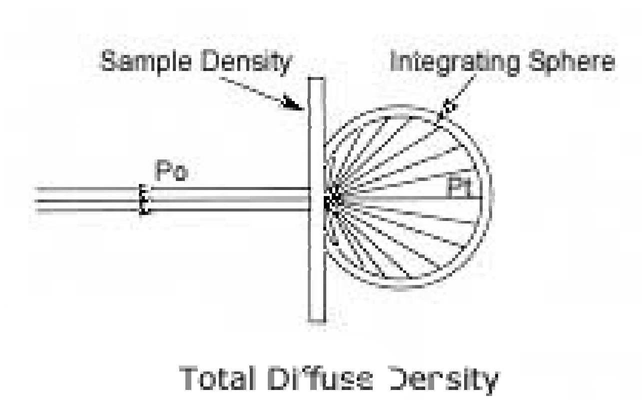
Fig. 8.1 Determining density.

Understanding Sensitometric Information

Transmission density (D) is a rneasure of the light-controlling power of the silver or dye deposit in a film emulsion. In color films, the density of she cyan dye represents its controlling power to red light, that of magenta dye to green light, and that of yellow dye to blue light. Transmission density may be mathematically defined as the common logarithm (Log base 10) of the ratio of the light incident on processed film (P_o) to the light transmitted by the film (P_t).

$D = \log_{10} \quad P_o P_t$

The measured value of the density depends on the spectral distribution of the exposing light, the spectral absorption of the film image, and the special sensitivity of the receptor. When the spectral sensitivity of the receptor approximates that of the human eye, the density is called visual density. When it approximates that of a duplicating or print stock, the condition is called printing density.



For practical purposes, transmission density is measured in two ways:

Totally diffuse density (figure) is determined by comparing all of the transmitted light with the incident light perpendicular to the film plane ("normal": incidence). The receptor is placed so that all of the transmitted light is collected and evaluated equally. This setup is analogous to the contact printer except that the receptor in the printer is film

- Specular density (fig.)is determined by comparing only the transmitted light that is perpendicular ("normal") to the film plane with the "normal" incident light, analogous to optical printing or projection.

To simulate actual conditions of film use, totally diffuse density readings are routinely used when motion-picture films are to be contact printed onto positive print stock. Specular density readings are appropriate when a film is to be optically printed or directly projected. However, totally diffuse density measurements are accepted in the trade for routine control in both contact and optical printing of color films. Totally diffuse density and specular density are almost equivalent for color films because the scattering effect of the dyes is slight, unlike the effect of silver in black-and-white emulsions.

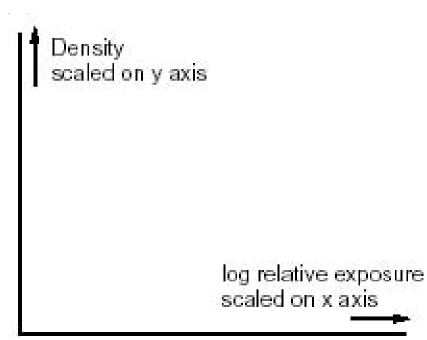
Production to the Characterstic Curve

In the literature the characteristic curve is variously called:

- A D log E curve
- An H and D curve (after Hurter and Driffield)
- A log It curve.
- All of these terms refer to the same curve.To draw this curve, the axes are labelled as follows:
- Density is represented on the vertical (y) axis of the graph._ The horizontal (x) axis carries the (relative or absolute) logarithmic value of the exposure, sometimes called the relative log exposure axis, or the log It (i.e. intensity multiplied by time) axis (Fig.). Because it is a ogarithmic value, each increase of log 0.30 represents a doubling of exposure, and each decrease of log 0.30 is equal to halving the exposure.There are a number of methods of producing the characteristic curve, each with its own advantages and disadvantages. Each will be considered in turn. It is assumed that the initial curves will be produced using the simple relationship of doubling the exposure, usually for 11 steps.

Time Scale Sensitometry

The kV, mA and distance are kept constant and the time of the exposure is varied, always by a factor of 2. This can be done either by simply doubling the time of exposure on the X-ray set, or by covering the cassette with lead rubber and exposing it section by section. In this case the
first part exposed will have received the most exposure, the last part, the least. Eleven exposures
are sufficient to provide enough points on the characteristic curve to plot a reasonable graph, although 21 would be ideal (this point is considered later in the chapter).



Establishing axes for characteristic curve.

Advantages

- It is possible to process films at a known time interval after the test and therefore prevent varying latent image problems.
- It is possible to process the film with the lower densities entering the processor first of all, thereby reducing 'bromide drag' which can cause streaking over the lower densities if the higher densities are processed first.

Disadvantages

- It has already been established that 11 exposures, doubling exposure in between exposures, will be adequate to produce the curve. If we start with 0.1 s as the first exposure we end up with the ollowing series:

  0.1, 0.2, 0.4, 0.8, 1.6, 3.2, 6.4, 12.8, 25.6, 51.2, 102.4
- The experimenter has to have either a unique timer on the X-ray set, or be an extremely accurate worker with a stopwatch.
- Using this range of exposures, or even one beginning with 0.01 s, eciprocity failure would almost certainly occur. After a time-consuming and difficult est the result could well produce a curve which, in the higher densities, does not resemble a haracteristic curve at all.
- The test is time consuming to perform.

Intensity Scale Sensitometry

This could be carried out using the same procedure as time scale sensitometry except that the kV and distance would be constant but the Ma values would be altered (i.e. constant time, varying mA).

Establishing Axes for Characteristic Curve

It is more usually performed by varying the height of the tube in relationship to the film, using the inverse square law to perform the calculations to vary the intensity of the X-ray beam reaching the ilm. This technique requires great accuracy in the X-ray set, in the calculations and Advantages and disadvantages are as for time scale sensitometry (above).

Calibrated Step Wedge

This method involves the use of an aluminium step wedge which has been calibrated in a specific way. The wedge should have a layer of copper on the base to help create a more homogeneous beam. It is the calibration of this wedge which is most important; it also requires considerable xpertise in the calculations for its construction. It has already been established that to produce a haracteristic curve we have to keep to a doubling of exposures between steps. Many people think that making a wedge with regular (i.e. 1, 2, 4, 8cm increments) steps would give the same posure increase, i.e. halving the exposure. Unfortunately, this relationship does not hold because of the differential absorption of the aluminium. The wedge has to be very precisely calibrated so that each step on the wedge produces an exact and regular increase or decrease in exposure. If the wedge is made with these precise calibrations, which demand very high engineering tolerances, it can be a very useful device.

Advantages

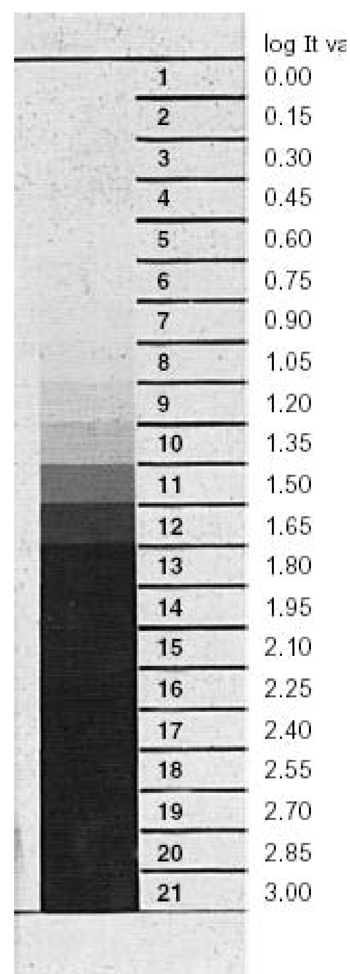
- The step wedge can be made with any number of steps and as long as accuracy has been observed in the specification and manufacture, an accurate characteristic curve will be produced.
- The step wedge can be reused.
- It can be used with different screen film combinations.
- It is possible to process films at a known time interval after the test and therefore prevent varying tent image problems.
- It is possible to process the film with the lower densities entering the processor first of all, and thus reducing 'bromide drag' which can cause streaking over the lower densities if the higher densities are processed first.

Disadvantages

- Because of the stringent specifications, a large calibrated step wedge can initially be expensive.

Sensitometer

A sensitometer is simply an exposure device which prints a pre-exposed negative directly onto the film. A sensitometer merely simulates,as nearly as possible, the spectral emission of the intensifying screens in use in the department.Sensitometers exist in many forms, from simple three patch sensitometers to large (and expensive) 21 step negatives for printing. Whichever one is chosen, care must be taken that the light output of the sensitometer matches the spectral sensitivity of the film. It would be pointless to use a blue-emitting sensitometer with an orhochromatic (green-sensitive) film.The result produced would be invalid. The sensitometer allows the users to prepare their own step wedges



easily, but it must be checked regularly to make sure that the unit is performing correctly. Figure shows the 21 step wedge produced by a typical sensitometer.

Advantages

- Quick and easy to use.
- It can be used with different screen film combinations.
- It is possible to process films at a known time interval after the test and therefore prevent varying latent image problems.
- It is possible to process the film with the lower densities entering the processor first of all, thus reducing 'bromide drag' which can cause streaking over the lower densities if the higher
- densities are processed first.

Disadvantages

- The initial cost of the equipment is high.

Typical sensitometer-produced 21 step wedge.

Pre-exposed Step Wedges

These are produced by a number of manufacturers.They are films which have been exposed by the manufacturer in a very accurate sensitometer.Films used for this type of work should have stabilised latent image storage and will be usable (after the film has been stored for a while)

for a short period of time. This is why this type of film has a very short shelf life, usually of the order of 6 months. The reason that the films are stored before release is that immediately after exposure there can be a number of ways in which the latent image can react. It can begin to fade, or remain the same, or even, over a short period, intensify.To overcome these variations, pre-exposed film is stored for a period before it is released. This factor must be taken into account when films are exposed in the department as well. It should be noted that some films cannot store a latent image for a long period of time.

Advantages

- Quick and easy to use.
- The films receive a constant, pre-determined exposure.

Disadvantages

- The cost of the films
- The films have a short shelf life.
- Can only be used to test processor consistency.

Reciprocity Failure

This is sometimes referred to as the Swarzschild effect. As long ago as 1876 two researchers, Bunsen and Roscoe, suggested that a relationship could be placed on E (effective exposure) and I (the light intensity) and t (the time of the exposure.) They expressed the relationship as: E . It Swarzschild then modified this formula in the late 1890s to: E . Itp where p was a constant for each emulsion, but varied from one emulsion to another. These experimenters discovered that a very long exposure to light does not produce the predicted lackening when related to a very short exposure time. The predicted reciprocity failure for a medical X-ray film would be a film which was stable at short exposure times, with an apparent loss of speed (less blackening) at very long exposure times. The only time a screen film encounters long exposure times to light is in the darkroom. Reciprocity failure at long exposures reduces the effect of long handling times in the darkroom safelight. As Swarzschild had discovered, this effect can vary enormously between different makes of film, but all manufacturers try to ensure little or no reciprocity failure at normal exposure times in radiography. Modern theory of latent image formation can account for this effect far more readily. It should be noted that this effect is only apparent where the exposure is due to light from the intensifying screens. With direct exposure to radiation, the X-ray quanta have such high energy, when compared to light, that they produce a latent image immediately in the silver halide grain.

Number of steps

There are three common types of grey scale (or photographic step wedge) produced. They are:

1 The three patch wedge

2 An 11 step wedge

3 A21 step wedge (usually referred to as a root 2 wedge).

All have a role to play, but the 21 step wedge gives the best results as it gives the user a far better curve. The reason it is referred to as a root 2 wedge is that the difference between each exposure is the previous exposure multiplied by root 2, or 1.414. This fits onto the log It scale very well, as the log value of root 2 is 0.15 (see Fig.).

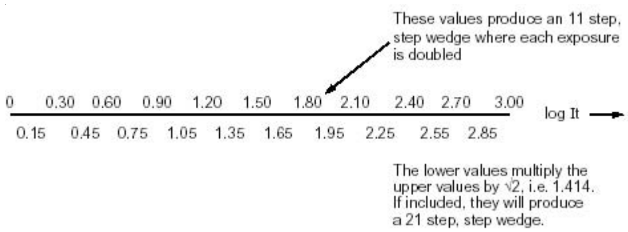
Calibration of a Step Wedge

Aluminium step wedges can be used as a test tool even though they produce totally arbitrary density steps. This step wedge can be calibrated quite easily as long as there is access to either a densitometer, an already calibrated step wedge or a pre-exposed step wedge film of known log It exposures. With the calibrated step wedge in position and a piece of lead rubber covering an area on the film equal to the size of the uncalibrated step wedge, an exposure is made.The uncalibrated step wedge is then placed on what was the covered area and lead rubber is used to cover all other portions of the same film. Another exposure is now made. The film is then processed. A characteristic curve is then drawn from the results obtained from the calibrated step wedge. The density values produced by the uncalibrated step wedge are now measured. Intercepts are drawn from each of these densityvalues to the curve and then to the log It axis.

These log It values represent the actual values for this step wedge at the kV selected (Fig.).

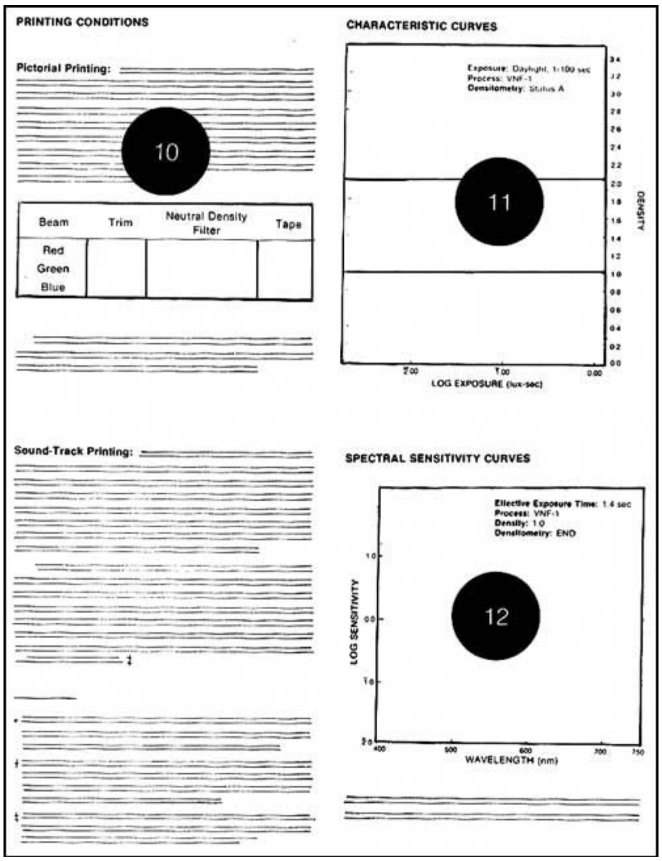
Densitometers

Densitometers read the relative density of the various steps on the film by measuring the quantity of light which passes through an area. The more light that passes through, the lower the density. This information can either be directly read off the LED display for each step and then the characteristic curve plotted by hand; or, in the more sophisticated models, the densitometer can be linked to a computer. All the steps are automatically read, and the quality control information can be automatically printed along with the characteristic curve for the film. In addition, information can be stored and weekly or monthlyprintouts produced if required to show the trends in processor activity and to allow comparisons between various processors that a department may have.



Log It scale, calibrated for both 11 and 21 step wedge.

Characteristic Curves Area



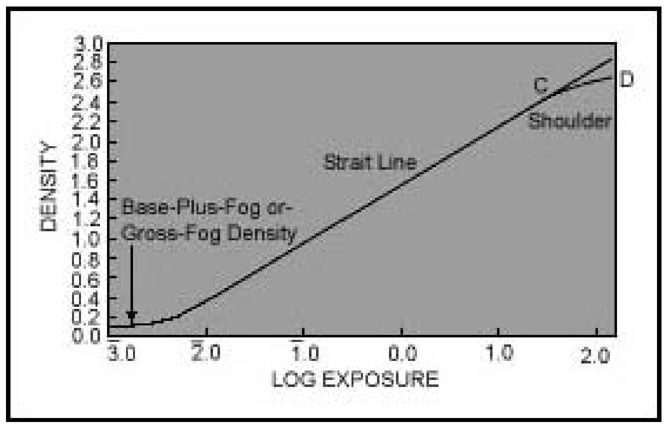
characteristic curve is a graph of the relationship between the amount of exposure given a film and its corresponding density after processing. The density values that produce the curve are measured on a film test strip that is exposed in a sensitometer under carefully controlled conditions and processed under equally controlled conditions. When a particular application requires precise information about the reactions of an emulsion to unusual light-filming action in a parking lot illuminated by sodium vapor lights, for example, you can filter the exposing light in the sensitometer can be filtered to simulate that to which the film will actually be exposed. A specially constructed step tablet, consisting of a strip of film or glass containing a graduated series of neutral densities differing by a conslant factor, is placed on the surface of the test strip to control the amount of exposure, the exposure time being held constant.

The resulting range of densities in the test strip simulates most picture-taking situations, in which an object modulates the light over a wide range of illuminance, causing a range of exposures (different densities) on the film.

After processing, the graduated densities on die processed test strip are measured with a densitometer. The amount of exposure (measured in lux [1]) received by each step on the test strip is multiplied by the exposure time (measured in seconds) to produce exposure values in units of lux-seconds. T"he logarithms (base 10) of the exposure values (log H) are plotted on the horizontal scale of the graph and the corresponding densities are plotted on the vertical scale to produce the characteristic curve. This curve is also known as the sensitometric curve, the D Log H (or E) curve, or the H&D (Hurter and Driffield) curve ..

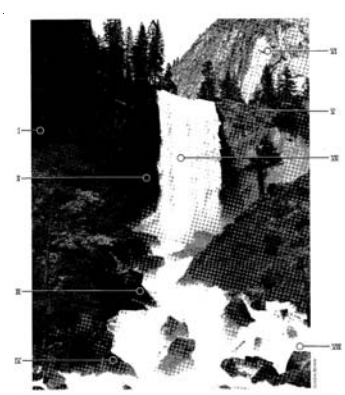
In the following table, the lux-sec values are shown below the log exposure values. The equivalent transmittance and opacity values are shown to the left of the density values

Typical Characteristic Curve



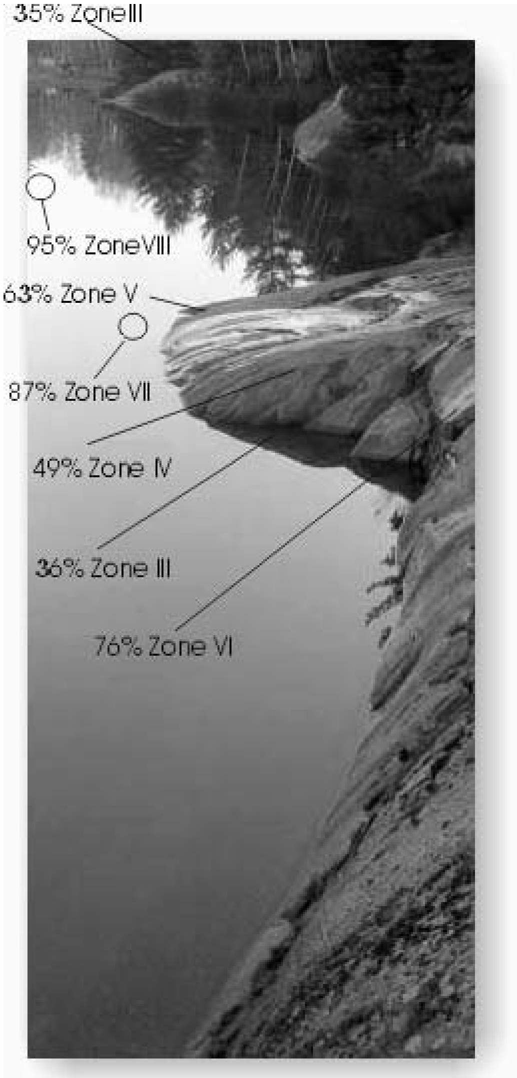
The characteristic curve for a test film exposed and processed as described in the table is an absolute or real characteristic curve of a particular film processed in a particular manner.

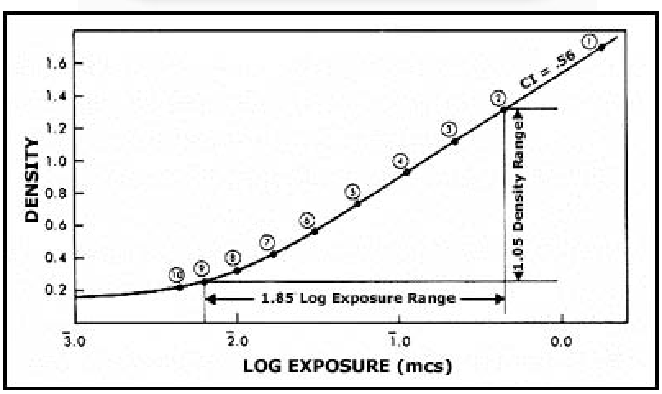
Sometimes it is necessary to establish that the values produced by one densitometer are comparable to those produced by another one. Status densitometry is used for this. Status densitometry refers to measurements made on a densitometer that conforms to a specified



Positive Image

Unfiltered spectral response (Dawson and Voglesong, Response Functions for Color Densitometry, PS&E Journal, Volume 17, No. 5 Sept/Oct 1973). When a set of carefully matched filters is used with such a densitometer, the term Status A densitometry is used.





The densities of color positive materials (reversal, duplicating, and print) are measured by Status A densitometry. When a different set of carefully matched filters is incorporated in the densitometer, the term Status M densitometry is used. The densities of color preprint films (color negative, intemegative, intermediate, low-contrast reversal original, and reversal intermediate) are measured by Status M densitometry. (DAK Densitometer Filter Sets are purchased directly from the manufacturers of densitometers. For further information, contact the densitometer manufacturer.)

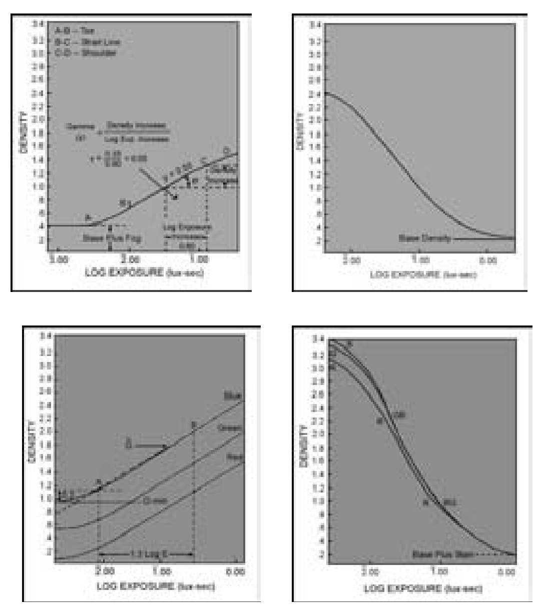
These illustrations show the relationship between subject luminance, negative density, and the characteristic curve. There is one stop difference in luminance between each of the points 2 to 10. Point 1 is a specular highlight which photographs as if it were about 2 stops brighter than point 2, which is a diffuse highlight. Point 9 is the tone to be reproduced just lighter than black. There are 7 stops difference between points 2 and 9, which is the typical range for normal luminance range subjects. Point 10 is about one stop darker than point 9, and reproduces as black. The graph shows where points of these brightness differences generally fall on a characteristic curve. Point 9 is exposed on the speed point of the film, which develops to a density of about 0.10 above the base plus fog density (the density of the clear film base after developing). The density range from point 9 to point 2 is about 1.05.

Representative characteristic curves are those that are typical of a product and are made by averaging the results from a number of tests made on a number of production batches of film. The curves shown in the data sheets are representative curves.

Relative characteristic curves are formed by plotting the densities of the test film against the densities of a specific uncalibrated sensitometric-step scale used to produce the test film. These are commonly used in laboratories as process control tools.

Black-and-white films usually have one characteristic curve (see Figures 5 and 6). A color film, on the other hand, has three characteristic curves, one each for the red-modulating (cyan-colored) dye layer, the green- modulating (magenta-colored) dye layer, and the blue-modulating (yellow- colored) dye layer (see Figures 7 and 8). Because reversal films yield a positive image after processing, their characteristic curves are inverse to those of negative films (compare Figures 5 and 6).

Typical Characteristic Curves



General Curve Regions

Regardless of film type, all characteristic curves are composed of five regions: D-min, the toe, the straight-line portion, the shoulder and D-max.

Exposures less than at A on negative film or greater than at A on reversal film will not be recorded as changes in density. This constant density area of a black-and-white film curve is called base plus fog. In a color film, it is termed minimum density or D-min.

The toe (A to B), as shown in Figure 9, is the portion of the characteristic curve where the slope (or gradient) increases gradually with constant changes in exposure (log H).

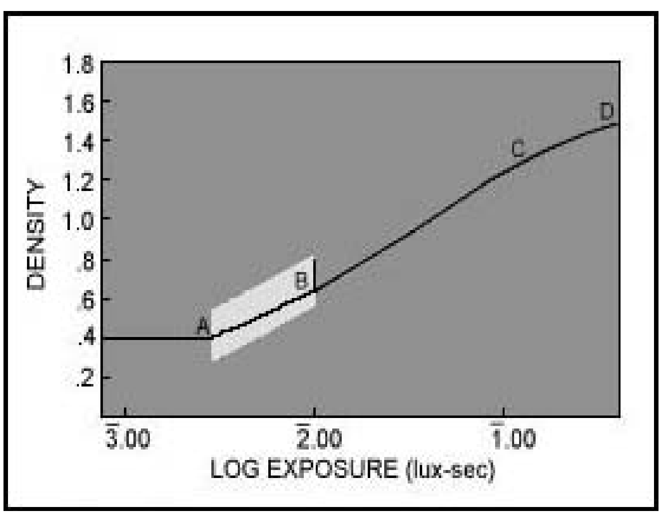
The straight-line (B to C), Figure 10, is the portion of the curve where the slope does not change; the density change for a given log-exposure change remains constant or linear. For optimum results, all significant picture information is placed on the straight-line portion.

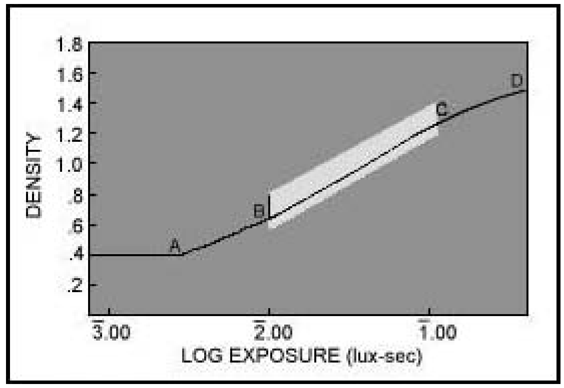
The shoulder (C to D), Figure 11, is the portion of the curve where the slope decreases. Further changes in exposure (log H) will produce no increase in density because the maximum density (D-max) of the film has been reached.

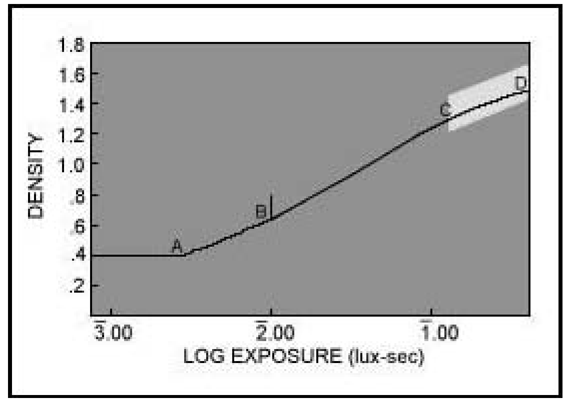
Base density is the density of fixed-out (all silver removed) negative- positive film that is unexposed and undeveloped. Net densities produced by exposure and development are measured from the base density. For reversal films, the analogous term of D-min describes the area receiving total exposure and complete processing. The resulting density is that of the film base with any residual dyes.

Fog refers to the net density produced during development of negative- positive films in areas that have had no exposure. Fog caused by development may be increased with extended development time or increased developer temperatures. The type of developing agent and the pH value of the developer can also affect the degree of fog. The net fog value for a given development time is obtained by subtracting the base density from the density of the unexposed but processed film. When such values are determined for a series of development times, a time-fog curve (Figure 12) showing the rate of fog growth with development can be plotted.

Curve Values







You can derive additional values from the characteristic curve that not only illustrate properties of the film but also aid in predicting results and solving problems that may occur during picture-taking or during the developing and printing processes.Speed describes the inherent sensitivity of an emulsion to light under specified conditions of exposure and development. The speed of a film is represented by a number derived from the film's characteristic curve.

Contrast refers to the separation of lightness and darkness (called "tones") in a film or print and is broadly represented by the slope of the characteristic curve. Adjectives such as flat or soft and contrasty or hard are often used to describe contrast. In general, the steeper the slope of the characteristic curve, the higher the contrast. The terms gamma and average gradient refer to numerical means for indicating the contrast of the photographic image.

Gamma is the slope of the straight-line portion of the characteristic curve or the tangent of the angle (a) formed by the straight line with the horizontal. In Figure 5, the tangent of the angle (a) is obtained by dividing the density increase by the log exposure change. The resulting numerical value is referred to as gamma.

Gamma does not describe contrast characteristics of the toe or the shoulder. Camera negative films record some parts of scenes, such as shadow areas, on the top portion of the characteristic curve. Gamma does not account for this aspect of contrast.

Average gradient is the slope of the line connecting two points bordering a specified log-exposure interval on the characteristic curve. The location of the two points includes portions of the curve beyond the straight-line portion. Thus, the average gradient can describe contrast characteristics in areas of the scene not rendered on the straight-line portion of the curve. Measurement of an average gradient extending beyond the straight-line portion is shown in Figure 13.

Curves for a Development-Time Series on a Typical Black and White Negative Film

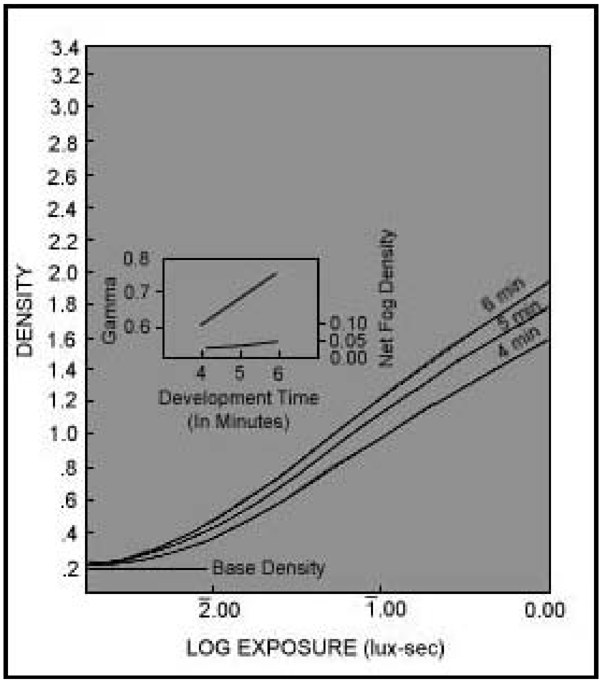


Figure 13

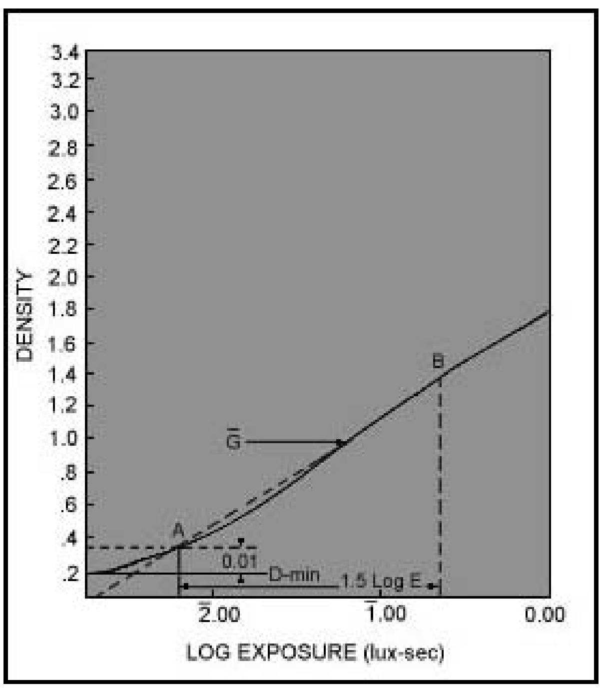
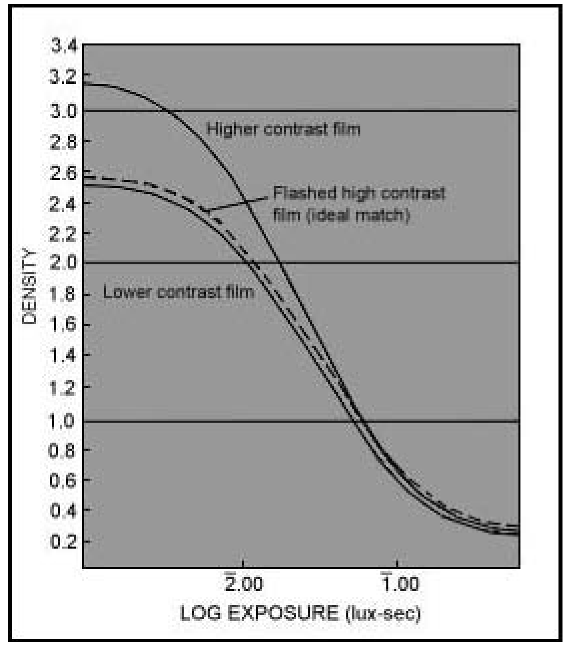


Figure 14

The particular gamma or average gradient value to which a specific black-and-white film is developed differs according to the properties and uses of the film. Suggested control gamma values are given on the data sheets for black-and-white negative and positive films.



If characteristic curves for a black-and-white negative or positive film are determined for a series of development times and the gamma or average gradient of each curve is plotted against the time of development, a curve showing the change of gamma or average gradient with increase development is obtained. You can use the time-gamma curve (Figure 14) to find the optimum developing time to produce the control gamma values recommended in the data sheet (or any other gamma desired). Black-and-white reversal and all color film processes are not controlled by using gamma values.

Flashing camera films to lower contrast is a technique[3] that involves uniformly exposing film before processing to lower its overall contrast. It's used with some color films. It is actually an intentional light fogging of the film. You can make the flashing exposure before or after the subject exposure, either in a camera or in a printer. The required amount of exposure and the color of the exposing light depends on the effect desired, the point at which the flashing exposure is applied, the subject of the main exposure, and the film processing. Because of potential latent image changes, a flashing exposure just prior to processing is the preferred method.

Figure 15

This fairly common practice is often used to create a closer match of two films' contrast characteristics when they are intercut. The hypothetical characteristic curves in Figure 15 show what occurs when one film is flashed to approximately match another film's characteristic curve. The illustration has been simplified to show an ideal matching of the two films. In practice, results will depend on the tests run using the specific films intended for a production.

Some film productions use flashing (called "creative flashing") to alter the contrast of the original camera negative of a particular scene to create a specific effect-making pastels from more saturated colors, enhancing shadow detail, and the like. Further discussion of this type of flashing is presented in "Creative Post-Flashing Technique for the The Long Goodbye," American Cinematographer Magazine, March 1973.

COLOR SENSITIVITY & SPECTRAL SENSITIVITY AREA

Color Sensitivity and Spectral Sensitivity area

The term color sensitivity is used on data sheets for some black-and-white films to describe the portion of the visual spectrum to which the film is sensitive. All black-and-white camera films are panchromatic (sensitive to the entire visible spectrum). Some laboratory films are also panchromatic: Eastman Fine Grain Duplicating Panchromatic Negative Film, Eastman Panchromatic Separation Film, and Eastman High Contrast Panchromatic Film.

Some films, called orthochromatic, are sensitive mainly to the blue-and- green portions of Lhe visible spectrum. Eastman Direct MP, Eastman Reversal BW Print, and Eastman Sound Recording II Films are all orthochromatic laboratory or print films.

Films used exclusively to receive images from black-and-white materials are blue-sensitive: Eastman Fine Grain Release Positive Film, Eastman High Contrast Positive Film, and Eastman Fine Grain Duplicating Positive Film.

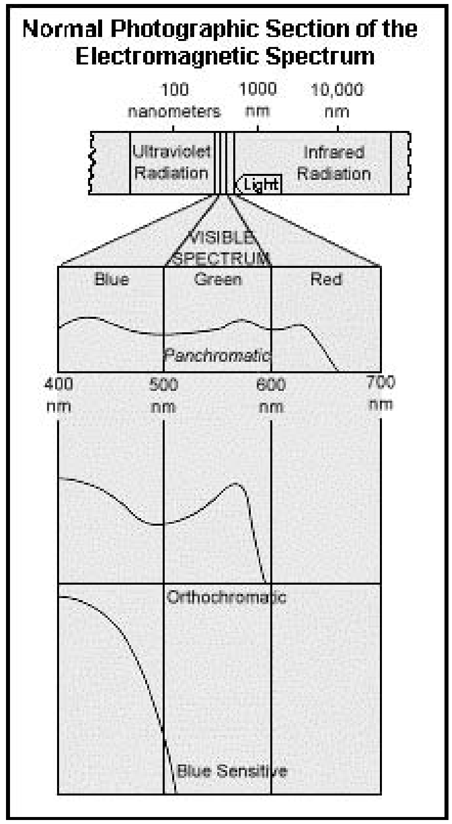
One film is sensitive to blue light and ultraviolet radiation: Eastman Television Recording Film. The extended sensitivity in the ultraviolet region of the spectrum permits the film to respond to the output of cathode- ray tubes.

While color films and panchromatic black-and-white films are sensitive to all wavelengths of visible light, rarely are two films equally sensitive to all wavelengths. Spectral sensitivity describes the relative sensitivity of the emulsion to the spectrum within the film's sensitivity range. The photographic emulsion has inherently the sensitivity of photosensitive silver halide crystals. Itese crystals are sensitive to high-energy radiation, such as X-rays, gamma rays, ultraviolet radiation and blue-light wavelengths (blue- sensitive black-and-white films). In conventional photographic emulsions, sensitivity is limited at the short (ultraviolet) wavelength end to about 250 nanometers (nm) because the gelatin used in the photographic emulsion absorbs much ultraviolet radiation. The sensitivity of an emulsion to the longer wavelengths can be extended by the addition of suitably chosen dyes.

By this means, the emulsion can be made sensitive through the green region (orthochromatic black-and-white films), through the green and red regions (color and panchromatic black-and-white films), and into the near-infrared region of the spectrum (infrared-sensitive film). See Figure 16.

Three spectral sensitivity curves are shown for color films-one each for the red-sensitive (cyan-dye forming), the green-sensitive (magenta-dye forming), and the blue-sensitive (yellow-dye forming) emulsion layers. One curve is shown for black-and-white films. The data are derived by exposing the film to calibrated bands of radiation 10 nanometers wide throughout the spectrum, and the sensitivity is expressed as the reciprocal of the exposure (ergs/cm^2) required to produce a specified density. The radiation expressed in nanometers is plotted on the horizontal axis, and the logarithm of sensitivity is plotted on the vertical axis to produce a spectral-sensitivity curve, as shown in Figure 17.

Figure 16



Equivalent neutral density (END)-When the amounts of the components of an image are expressed in this unit, each of the density figures tells how dense a gray that component can form. Because each emulsion layer of a color film has its own speed and contrast characteristics, equivalent neutral density (END) is derived as a standard basis for comparison of densities represented by the spectral- sensitivity curve. For color films, the standard density used to specify spectral sensitivity is as follows:

For reversal films, END = 1.0

For negative films, direct duplicating, and print films, END= 1.0 above D-min.

Spectral-Dye-Density Curves area 13

Proessing exposed color film produces cyan, magenta, and yellow dye images in the three separate layers of the film. The spectral-dye-density curves (illustrated in Figure 18) indicate the total absorption by each color dye measured at a particular wavelength of light and the visual neutral density (at 1.0) of the combined layers measured at the same wavelengths.

Spectral-dye-density curves for reversal and print films represent dyes normalized to form a visual neutral density of 1.0 for a specified viewing and measuring illuminant. Films which are generally viewed by projection are measured with light having a color temperature of 5400 K. Color-masked films have a curve that represents typical dye densities for a mid-scale neutral subject.

The wavelengths of light, expressed in nanometers (nm), are plotted on the horizontal axis, and the corresponding diffuse spectral densities are plotted on the vertical axis. Ideally, a color dye should absorb only in its own region of the spectrum. All color dyes in use absorb some wavelengths in other regions of the spectrum. This unwanted absorption, which could prevent satisfactory color reproduction when the dyes are printed, is corrected in the film's manufacture.

In color negative films, some of the dye-forming couplers incorporated in the emulsion layers at the time of manufacture are colored and are evident in the D-min of the film after development. These residual couplers provide automatic masking to compensate for the effects of unwanted dye absorption when the negative is printed. This explains why negative color films look orange.

Since color reversal films and print films are usually designed for direct projection, the dye-forming couplers must be colorless. In this case, the couplers are selected to produce dyes that will, as closely as possible, absorb in only their respective regions in the spectrum. If these films are printed, they require no printing mask.

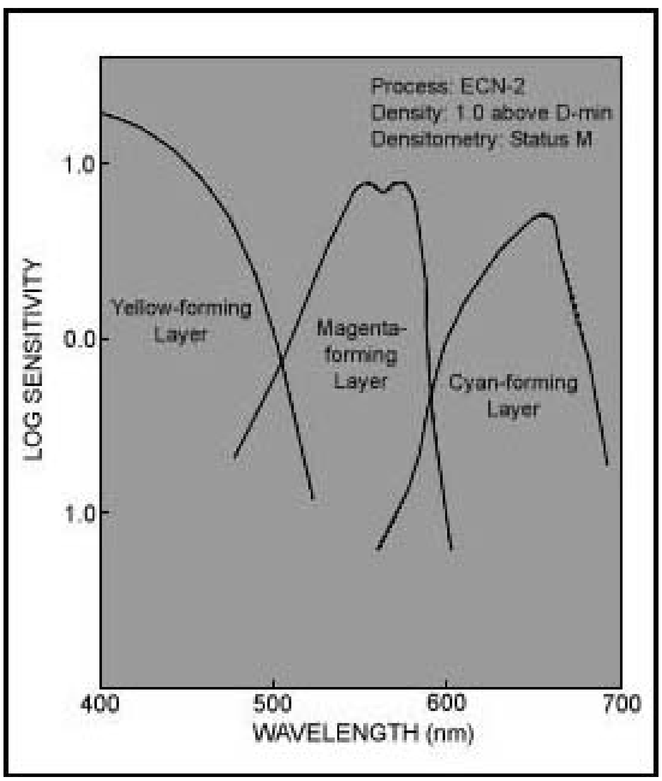


FIG.17

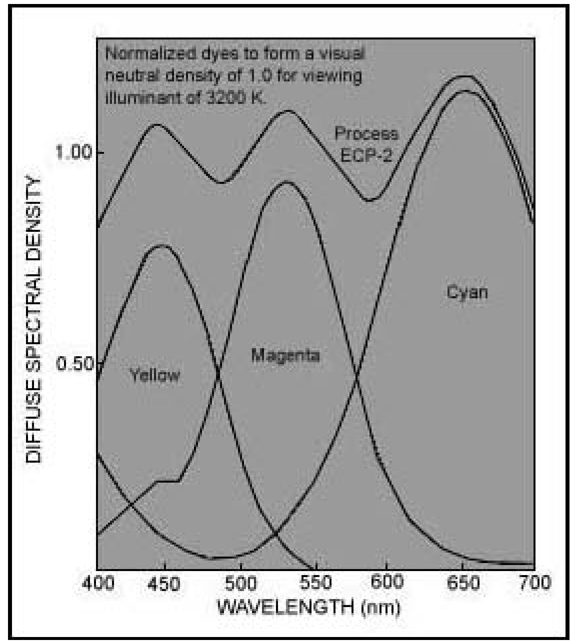


FIG 18

Notes

Motion Picture Printing

- Printers
- Wet-Gate Printing
- Printing Operations
- Additive and Substactive Printing
- Color Timing
- Motion Picture Laboratory Control of Color Duplication
- Digital Special Effects
- Sound-Track Printing

Printers

Continuous-Contact Printer. In its simplest form, printing consists of exposing the raw stock from an "original" or "printing master" to form the image using a light source to produce the exposure. When the image size of the print is the same as that of the original (i.e., 35 mm to 35 mm, 16 mm to 16 mm), the printing is usually done in a continuous-contact printer.

The large printing sprocket advances both the original and the print film at a constant rate past the light source. The original and print films are usually positioned emulsion-to-emulsion with the light passing through the original and exposing the stock to be printed. Depending on the application, these contact printers may operate up to thousands of feet per minute.

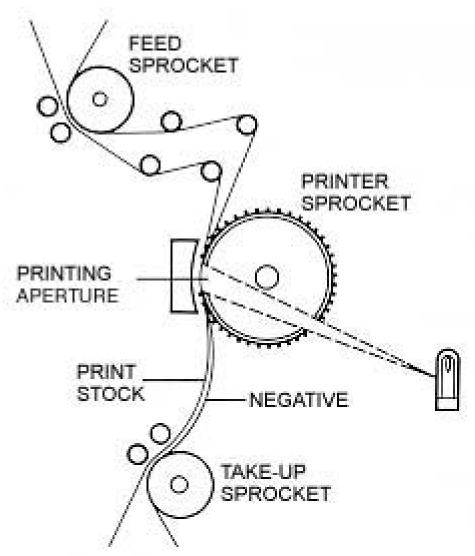


Figure 67
Schematic of a subtractive contact printer

Step-Contact Printer

Step-contact printers advance both negative and print films through the printer gate with an intermittent motion and shutter similar to that of a camera. Close-fitting register pins position the two films with extreme accuracy during exposure, and a pressure plate at the printing gate assures film flatness. Because of the complexity of the machine and the precision of film registration achieved, the speed of a step-contact printer is relatively low (2 1/2 to 40 feet per minute). Stepcontact printers are precision instruments used for making color separations and special-effects printing that may require several passes of the raw stock through the printer (for traveling mattes, master positives, and color intermediates, etc.). Generally, they are designed for roomlight operation to make the necessary operator control easier.

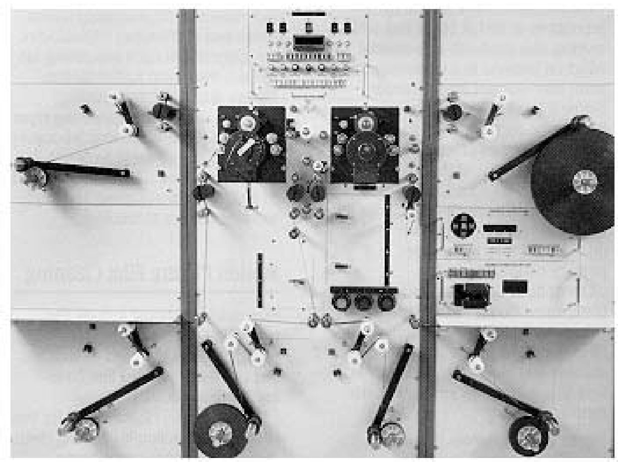


Figure 68
A continuous contact adative panel printer

Step-Optical Printer. The step-optical printer combines the precision of a step-contact printer with optical flexibility Like the step-contact, the step-optical printer finds its main use in the production of intermediates and special effects.

Whenever the image size of the print is different from that of the original or certain special effects are desired, an optical printer is used. The optical printer can be thought of as a projector on one side and a camera on the other. The image produced by the projector is focused at the plane of the film in the camera gate. A schematic of an optical printer used for reducing 35mm to 16mm is shown below. Optical printers can be quite complex, providing such effects as blowups, reductions, skip frames, anamorphic compression, zooms, mattes, etc.

Continuous-Optical Printer. These printers are used for high-volume reduction printing. Like a continuous-contact printer, the exposure is made through a slit, thus necessitating exactly matched relative film speeds. This is obtained by mounting both the sprocket driving the original Film and the one for the print film on the same shaft. The different diameters of the two sprockets provide the proper filmspeed ratio. The light path from original to print is U-shaped as a result of using the same shaft to drive both films. The addition of image-dividing lenses or prisms permits multirank printing.

Wet-Gate Printing

one of the most troublesome problems encountered by motion picture laboratory personnel are scratches (digs, abrasions, cinch marks, etc.) sometimes encountered on film from which prints must be made. These scratches print through to the release print and degrade the quality of the projected picture by introducing image elements that have no relationship to the originally photographed scene.

Figure 69
An optical printer

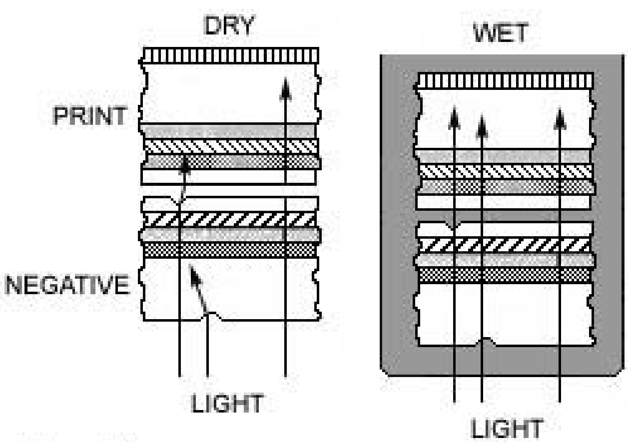


Figure 70
Wet-gate printing

A scratch on the support of a negative film acts as a diffuser that scatters light. Light from the printer passes essentially in straight lines through the undamaged portion of the support and emulsion of the original. When light strikes the scratch, it is scattered and displaced from the straight-line path, reducing the light on the receiving emulsion.

Scratches on the support of a negative film printed onto positive film usually produce more objectionable effects on the screen than scratches on reversal originals printed onto reversal print films. This is because scratches on the support of negative films appear white on the positive film and are generally of lower density than any other white in the picture. In reversal printing, scratches on the support of the original appear black on the screen print and generally tend to blend in better with the picture.

Scratches on the emulsion side of negative films present another situation. Shallow emulsion scratches on a black-and-white negative will appear white on the positive film. Emulsion scratches that penetrate to the support on a black-and-white negative will print black. Scratches on the emulsion side of color negative films may appear colored on the print, depending upon how deep the scratch is and whether image-bearing layers have been disturbed.

When base scratches exist, a “wet” or “liquid” gate is used to minimize or eliminate their effect, depending on severity In a wet gate, liquid having a refractive index close to that of the film base is applied to the original. The liquid fills in the scratches and reduces the light scatter. Wet-gate printing is applicable to any of the printing configurations, step or continuous, contact or optical. Wet printing is of little or no benefit to emulsion-side scratches.

Printing Operations

Image Orientation: Choosing a Duplicating Method.

The orientation of the image on the final print is an important consideration in choosing a duplicating method. Camera original film is normally exposed with the emulsion side facing the lens of the camera. When the Film is processed, the image reads correctly through the base side of the film. if a typical emulsion- to -emulsion contact print is made, the resulting print will read correctly through the emulsion side of the film. When several stages of emulsion-to-emulsion contact printing are involved, the image orientation changes with each successive stage.

In the case of 35"mm prints, the image orientation has been standardized. SMPTE Standard SMPTE 194-1997 specifies, “The photographic emulsion shall be on the side of the film which faces away from the projector lens,” (i.e., the image reads correctly through the emulsion side of the film). This is because 35 mm productions utilize a negative camera original contact printed to produce prints.

In 35mm production, the proper orientation is obtained when prints are made by contact printing the original, or in going through a master positive-to-duplicate negative-to-print duplicating system. When a duplicate negative is made directly from a print, the image orientation must be changed. This may best be done by optical printing through the base of the print. Some laboratories change the orientation by contact printing through the base, which results in a noticeable loss of sharpness.

Sixteen millimetre film started as amateur medium, using reversal camera original film that was projected after processing. Therefore, the emulsion had to be toward the projection lens for the image to read properly on the screen. SMPTE Standard SMPTE 233-1998 states, “For original reversal film, the emulsion side shall be toward the projection lens. For prints, the emulsion position is dependent upon the process of preparation; however, the preferred position for most uses, including telecine, is also emulsion side toward the projection lens." This permits intercutting of prints and originals without requiring change of focus during projection.

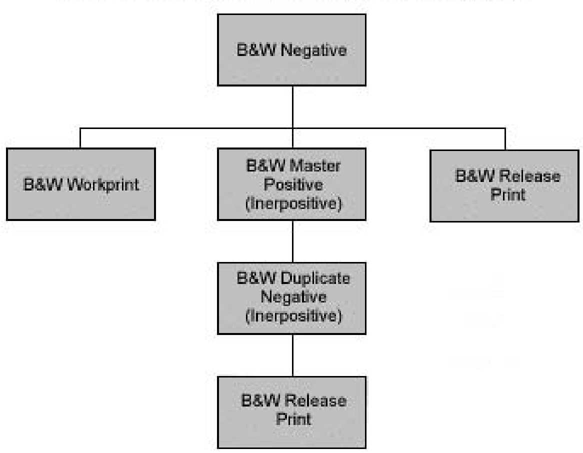
Image orientation is important for intercut materials because of the need to refocus either the printer or the projector (both picture and sound optics) each time the image orientation changes. Failure to refocus will result in an unsharp picture and loss of frequency response in the sound.

In 16 mm, the preferred orientation results when the camera original is projected, or contact release prints are made using an internegative or duplicate negative. Contact prints made directly from the camera original, or using the master positive -to - duplicate negative-to-print duplicating system will have to be shown with the emulsion away from the lens for the image to read correctly on the screen. Contact printing through the base to change orientation in 16 mm usually results in unacceptable loss of sharpness.

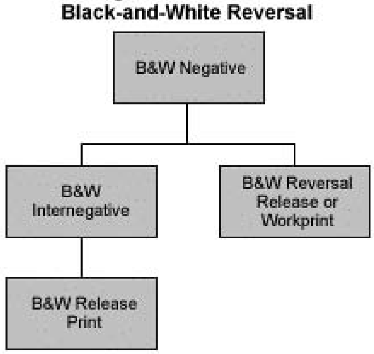
Black-and-White Printing

Black-and-white printing practices are essentially the same as color printing practices. However, the lack of such considerations as color balance, saturation, etc., make black-and-white printing a less complex operation than color printing. The printing flowcharts show some common methods employed by laboratories in producing black-and-white motion picture prints.

Printing Flowchart for Black-and-White Negative



Printing Flowchart for 16mm Black-and-White Reversal



Note: A change in image size requires optical printing. Where reduction stages are called for, it is best - in order to obtain the highest definition in the final print - to postpone reduction until the latest practicable stage.

Notes

HOW TO SHOOT THE GRAY CARD

Gray Card Plus

A standard reference for exposure evaluation and grading. The card is comprised of a large 18% neutral grey area bordered by 3% and 90% black and white patches to provide further reference for grading. The surface is specially treated to minimize glare.

The Kodak Gray Card Plus with black and white patches is used to:

- Determine telecine transfer points
- Provide exposure information
- Help the Colorist and Film Timer preserve what the Cinematographer created on film

Features:

- In Neutral gray center field, 18% reflectance
- Black side patches, 3% reflectance
- White side patches, 90% reflectance
- Available in two sizes: 9 x 12 inches, 18 x 24 inches
- Durable matte finish

User's Guide

In addition to the specific instructions for your application, reading the entire guide will give you a better understanding of exposure evaluation and transfer control using the Kodak gray card plus.

For the Cinematographer

Shoot the Kodak grey card plus in each new lighting setup. The Kodak grey card plus should occupy at least 15% of the frame in all cases, and if possible, a larger portion of the frame. The larger the card area, the easier it is for the Colorist to obtain the readings needed for exposure and grading reference.

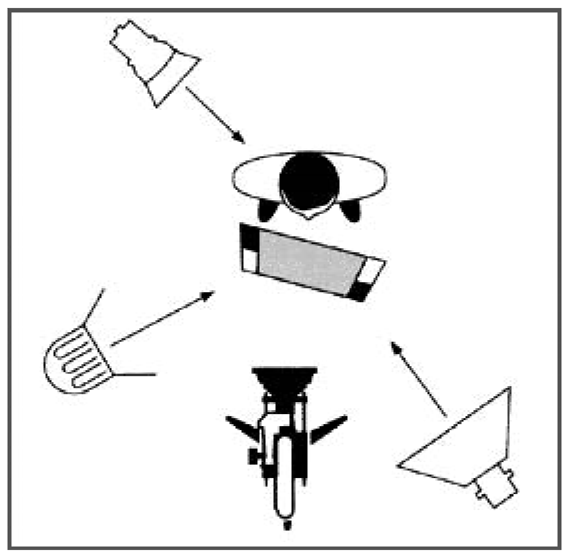
For Transfer Grading and Film Timing

- provides a transfer and printing reference for normal lighting and exposure.
- serves as a guide to preserve special lighting and exposures.
- indicates corrections needed to improve color balance and exposure.
- establishes a starting point for scenes to be transferred or printed for special effects.

For Standard Exposure Evaluation

- provides a standard reference when used with Kodak telecine exposure calibration (TEC) film.
- enables the transfer facility to determine transfer points similar to printer points from a film lab.

Shooting the Kodak Gray Card Plus



- When the card is shot as part of the scene it should...
- be placed near the main subject to represent the general lighting.
- occupy at least 15% of the frame.
- provide an accurate reference for color and exposure when graded to 18% neutral grey.

1 Position the card so it receives the same light (intensity and color balance) as the main subject and occupies at least 15% of the frame. Zoom or move in, if necessary.

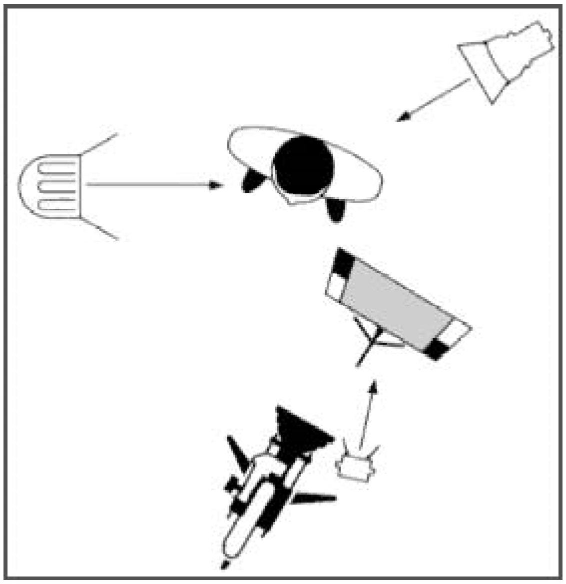
2 Turn or tilt the card so it is evenly illuminated without shadows or flare. The color temperature of the light reflected from the card should match the scene.

3 Determine normal exposure for the scene using an incident light meter or the method you prefer.

4 Take a reflected spotmeter reading on the grey portion of the card from the camera position. If necessary, reposition the card or add supplemental light for a reading that agrees with the aperture you have chosen for normal exposure.

5 Shoot the card including some of the scene for practical reference.

6 Repeat this procedure at the head of each roll and every time there is a major lighting change.



Note: Shooting the card in the scene will not preserve special lighting or exposure. The colorist or timer is instructed to always grade the card to 18% neutral, then use that setup as a strong starting point for final grading. To maintain special or altered lighting in the transfer or print, always shoot the card in the alternate position.

Shoot the Card in this Alternate Position When...

- placing the card in the scene will not provide an accurate grading or exposure reference.
- the card cannot reflect the main light of the scene and occupy at least 15% of the frame.

Examples: strong cross lighting or backlit scenes; wide shots or long exteriors where the card would be lost in the scene.

1 Place the card on an easel or other support near the camera. Make sure the light on the card is even and flat. Generally, one wide-beam lighting unit placed near the camera lens is sufficient.

2 Tilt the card, if necessary, for even reflectance. The color balance and exposure on the card should match the color exposure of the scene.

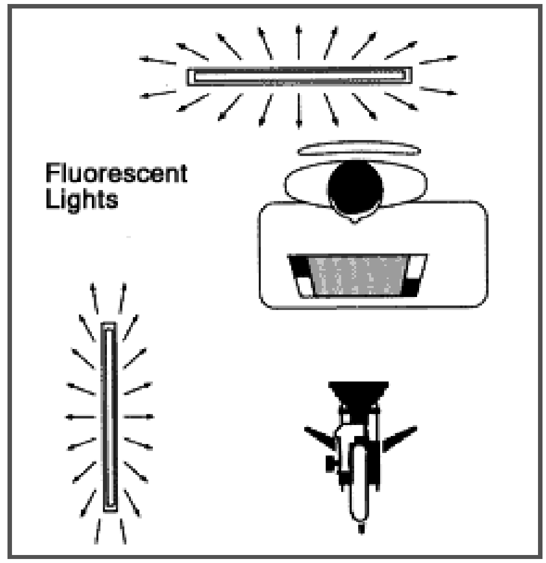
Exception: when the card is to serve as a grading reference for special lighting. (Darker or Lighter Grading and Special Lighting and Exposure Control.)

3 Take a reflected light reading on the grey portion of the card from the camera position. Select a lens aperture that will make the card a valid reference for grading and/or exposure evaluation. (See the instructions on using the card below.)

4 Shoot a few feet of the card (close-up) immediately preceding the scene it references and for every major lighting change. If possible, pull or zoom back and shoot the card a second time including some of the scene for practical reference.

Using the Kodak Gray Card Plus for Transfer Grading and Film Timing

Normal Exposure and Color Balance



Normal color balance is 3200°K tungsten; 5600°K daylight.

1 Follow the instructions for shooting the card as part of the scene or in the alternate position.

2 Shoot the card at the same aperture and color temperature as the scene.

Color Correction

When the lighting does not match the color balance of the film. Example: shooting under fluorescent or warm tungsten light without correction filters.

1 Make certain the card receives the same light (color and intensity) as the scene to be shot. If necessary, light the card separately as described under alternate position, maintaining the same dominant color balance of the main scene.

2 Shoot the card in the scene or immediately preceding the scene it references. When the shot with the card is graded to a neutral grey, the scene(s) following will be corrected for a more normal color balance.

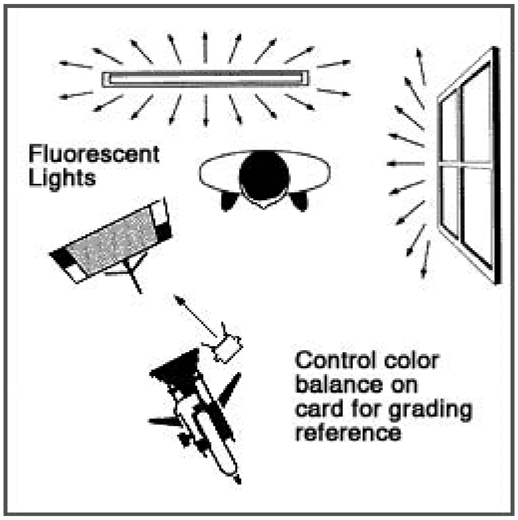
Note: If you shoot with color correction filters on the camera or the lights to help balance the scene, make sure you shoot the card under the same filtration. If you are using color filters on the camera or lights for special effects, do not shoot the card using these filters. Your special lighting will be lost when the scene is graded for a neutral grey card.

Mixed Lighting

When lighting consists of different color temperatures. Example: a combination of daylight, fluorescent or tungsten light.

1 Read the color temperature of the various areas in the scene.

2 Determine the color temperature of the area which is visually most dominant.



3 Shoot the card in this area as a reference for the color correction needed in the transfer or print.

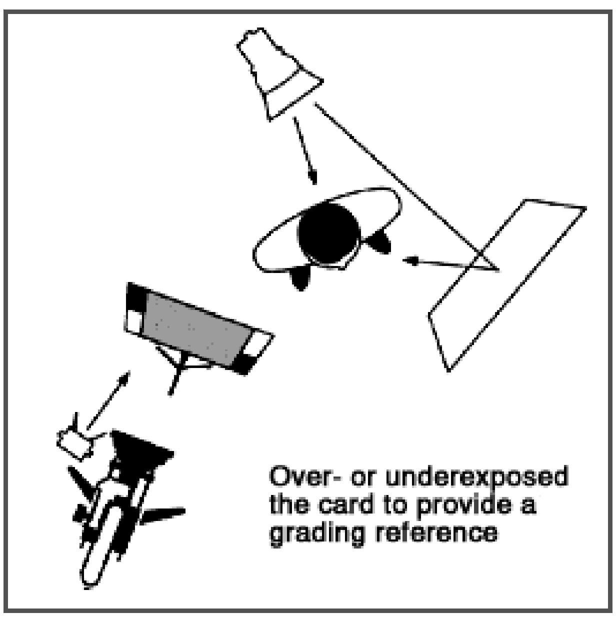
OR

1 Determine the average color temperature in the scene.

2 Using the alternate position, light the card for this color temperature. (If necessary, use separate light[s] with voltage control to achieve the desired color balance on the card.)

When the shot of the card is graded to a neutral grey, the scene(s) following will be corrected for a warmer or cooler color balance as determined by the light on the card.

Darker or Lighter Grading Normal Exposure



When the scene is to be graded darker or lighter.

Examples: day-for-night scenes shot at normal exposure; a scene to be transferred or printed two stops darker.

- Shoot the scene at a normal exposure to provide a full-range negative.
- Instead of shooting the card at the normal aperture, overexpose the card if you want the scene darkened. Underexpose the card if you want the scene lightened. The amount the card is over or underexposed will provide a grading reference for the way the scene is to be transferred or printed.
- To maintain colored lighting, make sure you shoot the card under "white light" (light balanced for the film). This will preserve your special lighting when the card is timed to a neutral grey.

Special Lighting and Exposure Control

When non-standard lighting and exposures are to be maintained in the print or transfer.

Examples: scenes intentionally over or underexposed; colored lighting and other special effects.

1 Light the scene and determine your exposure for the desired effect. Example: light for T-2.8 with colored gels; shoot at T-4 for one-stop underexposure.

2 Using the alternate position, light the card with "white light" balanced for the film.

3 Take a reflected reading from the camera position and shoot the card at an aperture that will provide a normal grey card exposure, in this case T-2.8.

4 Shoot the scene at the aperture you've chosen for under or overexposure. When the card is graded to 18% neutral grey, and this grading is maintained for the scene that follows, your special lighting and exposures should be maintained.

Note: If you light the scene with a wide mix of colors in which there is no one dominant source - fire, smoke, heavy gels - shoot the card in the alternate position under "white light" (balanced for the film) to maintain your special; lighting. If possible, include some of the scene with the card for practical reference.

Using the Kodak Gray Card Plus for Exposure Evaluation With the Kodak telecine exposure calibration (TEC) film to determine transfer points.

1 Shoot the card as part of the scene or use the alternate position. For exposure evaluation, the card and the scene must be shot at the same exposure. Generally, this is also how the card is shot for transfer grading and film timing. Exception: as a guide for darker or lighter grading or special lighting and exposure control, the card and scene are usually shot at different exposures. The grey card no longer serves as an accurate reference for exposure evaluation.

2 Shoot the card twice if different exposures are required for grading and exposure evaluation. Identify each reference.

The colorist reads the red, green and blue values from the grey card and compares these to the standard TEC film reference. Exposures are evaluated in terms of transfer points, similar to printer points in the film lab.

Using Filters and the Kodak Gray Card Plus

- If you are using filters on the camera or on a light source for color compensation (e.g., Kodak Wratten 85 filter on the lens or gel on a window), shoot the card with the filter(s) in place.

- If you are using filters on the lights to create a special effect, shoot the card with "white light" (balanced for the film) to preserve the effect.
- Remember to take the filter factor (FF) into account when the filter is on the camera, or you may underexpose the scene.

Here's a simple way to determine the filter factor...

1 Measure the light from a single narrow source (spotlight) with an incident light meter.

2 Without moving the meter, hold the filter directly in front of the sensor so that all the light reaching the meter must pass through the filter.

3 Compare these two readings. The difference in t-stops will give you the filter factor. One stop = FF 2; two stops = FF 4; three stops = FF 8, etc.

4 Dividing the normal exposure index (EI) of the film by the filter factor will give you the effective EI for the film with that filter on the lens.

Example: normal EI 100 FF 2 (one t-stop) = EI 50.

Based on your own experience and preference in determining proper exposure, you may sometimes choose to alter these procedures. The main thing is to be consistent, so that the colorist or timer will have a uniform reference. That way he/she can set up the transfer or select printer lights to maintain the look you worked to achieve.

For the Colorist

The Kodak Gray Card Plus helps you establish grading which best maintains the lighting in the original film. That can save considerable time and take the guesswork out of wondering, "How is this scene supposed to look?"

The card is also the standard reference for the Kodak telecine exposure calibration (TEC) film which gives you transfer points objective exposure information for the Cinematographer.

Using the Kodak Gray Card Plus as a Grading Reference

Always grade the center portion of the card to 18% neutral grey. That will give you a good grading reference to transfer the scene(s) that follow.

1 Center the controls using Kodak telecine analysis film or other baseline reference.

2 Locate the first shot of the Kodak Gray Card Plus. Roll into the shot to make sure the lighting does not change.

3 Set the voltages or IRE values for the white, grey and black areas of the card to those indicated below.

4 Use these settings for grading the scenes referenced by the grey card. Make corrections, if needed, to improve overall color and contrast.

5 Store the settings so that they may be recalled for future use.

6 Proceed to the next card and set up accordingly.

Grading Values for Kodak Gray Card Plus - normal exposure (component values)

Note: The values specified in the voltage and IRE tables are all COMPONENT RGB values with no setup. The IRE values indicated are percentages of component RGB voltages.

Values are based on 0-700 mV equal to 0-100% of voltage.

	Component Voltage	Component % Voltage
White	560 mv	80
Gray	320 mv	45
Black	140 mv	20

Note: If the Cinematographer has designed special lighting e.g., colored gels, fire, low-key, etc, or used special lens filters, the lighting on the card will not match the scene. However, grading the grey portion of the card to 18% neutral will help to maintain the desired effect in the transfer.

Using the Kodak Gray Card Plus as a Reference to Determine Transfer Points

The following are basic procedures for using the grey card with the Kodak telecine exposure calibration (TEC) film. Consult the TEC film user's guide for complete instructions.

1 Locate the first shot of the grey card on the roll of negative to be transferred.

2 Recall the TEC film settings previously stored in memory.

3 Read the red, green and blue values on the waveform monitor from the grey portion of the card.

Note: Do not zoom in or change the size of the grey card.

4 Convert these voltages or IRE values to transfer points using the TEC film conversion table or automatic reading system.

A normally exposed 18% grey card has RGB transfer points of 25-25-25. You can now assess exposures in much the same way a timer uses printer points. One t-stop is equal to about 7 printer points or 7 transfer points. Commonly understood exposure relationships are, therefore, maintained.

Note: If the card has been shot as a reference for darker or lighter grading or special lighting and exposure control, the card will not be an accurate reference for determining transfer points. In these cases the cinematographer is instructed to shoot the card a second time and identify it as an exposure reference.

For the Film Timer

Reading the Gray densities of the Kodak Gray Card Plus provides an objective evaluation of film exposure. Reading the black and white patches on the card will indicate the range of brightness and where the scene falls on the sensitometric curve for any given film stock.

Generally, you will time your dailies using a color analyzer. However, for those scenes where the Cinematographer used special lighting or intentionally altered exposure, the Kodak Gray card provides a good reference to establish printer lights which best preserve the Cinematographer's intent.

Using the Kodak Gray Card Plus to time special scenes

1 Read the red, green, blue densities from the Gray portion of each card on the film.

2 Use these values to establish your initial timing for the scenes following the Gray card reference.

3 Verify your timing on a color analyzer based on the lights you have determined best for the Gray card exposures.

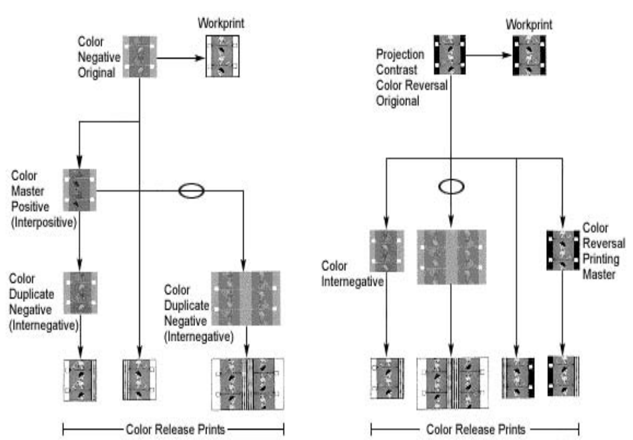
Color Printing

A contact printer, with provisions for scene-to-scene density and color-balance changes, is required for color printing. An optical printer is needed to make reductions or enlargements, where appropriate. If it is necessary to create separation negatives or positives for extended keeping purposes, a step-contact printer is required to provide precision in positioning each successive frame of film. Certain kinds of special effects may also require a step-optical printer.

The desire for high-volume production in laboratories has led to the use of multirow perforation formats to minimize handling. These systems for producing two or four rows of pictures on 16 mm or 35 mm raw stock require specially designed equipment. With the advent of video techniques, the demand for these formats is minimal.

The printing systems shown in Figures 71, 72, 73, and 74 represent those in general use at laboratories; however, they do not include all procedures currently used. Because they are only photomechanical reproductions, these charts are meant to serve as guides to the printing systems and are not intended for use in evaluating picture quality with respect to color balance, saturation, contrast, sharpness, or graininess. For loose-leaf charts and detailed descriptions of the printing systems, see KODAK Publication No. H-25, Motion Picture Prints from Color Originals.

16 mm Color Prints from 16 mm Camera Originals



NOTES
1. When multiple release prints are required, use a color negative origional.
2. The choice of printing procedure depends on a number of factors, including the types of printing and processing facilities available and cost. Final print quality will be dependant on those factors.
3. All film clips are shown with the *base* side up.
4. When using a Projection Contrast Color Reversal Origional, you may have to use process adjustment, flashing or other means to reduce the contrast.
5. The symbol ⬭ indicates optical printing, used here to maintain the prefered image orientation and multiple printing.

Figure 71

35 mm and 16 mm Color Prints from 35 mm Negatives

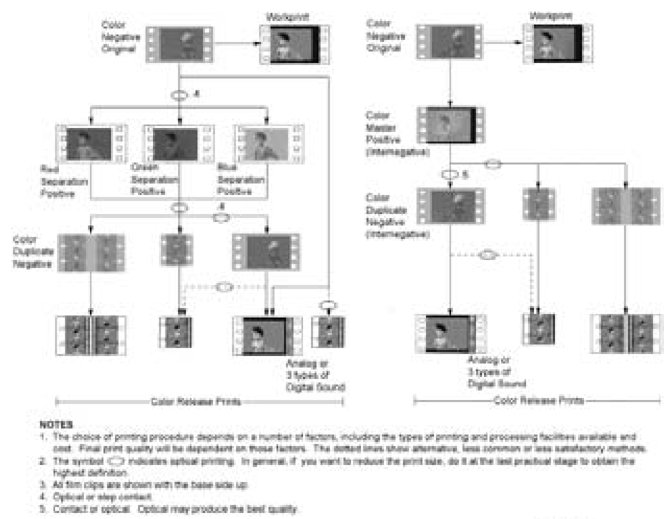
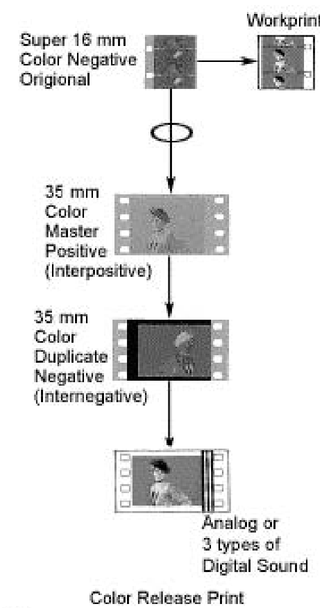


Figure 72

35mm Color Prints from Super 16mm Negatives



NOTE
1. Super 16mm in a camera system intended specifically for photography which is to be enlarged to 35mm. Normal 16mm single perforated rawstock is used but the camera aperature extends into the area used for the sound track on convential 16mm film, providing more picture area. This format requires caution in the use of special enlarged aperatures and equipment that will not abraid the extended picture area.
2. The choice of printing procedure depends on a number of factors, including the types of printing and processing facilities available.
3. The symbol ⬭ indicates optical printing to increase picture size.
4. All film clips are shown in *base* side up.
5. Not all systems necessarily follow these sequences. Howver, these represent commonly used systems today and will produce desired results.

Figure 73

70 mm Color Prints from 35 mm and 65 mm Negatives

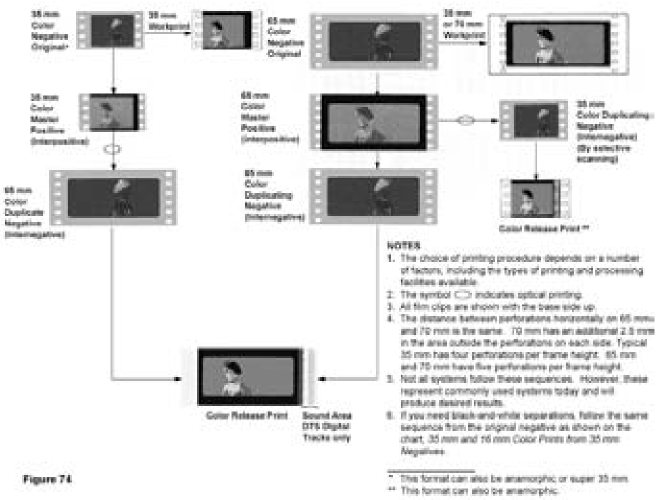


Figure 74

Additive and Subtractive Printing

Whenever color printing is involved, the printer or lamphouse must be able to control the red, green, and blue components of the white-light source. Two methods of control are commonly used: additive and subtractive printing.

In a subtractive printer, color correction (changing the relative amounts of red, green, and blue light) is achieved by inserting color correcting filters in the light path between the tight source and the printing aperture. Overall light changes (intensity changes) are made either by a variable aperture or a neutral density filter. Subtractive printing is sometimes used for "release" printing (making multiple one-light prints after an answer print has been approved) because there are no scene-to-scene color changes. Printing requiring any scene-to-scene color corrections is not practical on a subtractive printer.

The particular filter packs you use for subtractive printing will depend upon the characteristics of the optical system of the printer, the lamp voltage, etc. The filter pack is usually composed of color compensating (CC) filters.

Unwanted absorption in the dyes of such filters may modulate exposure of other layers to a lesser, but significant, degree. This makes precise exposure control a more cumbersome operation than it is in a well-designed additive printer.

The most popular printing method is additive printing. instead of a single light source with color- correcting filters, three separate colored sources - red, green, and blue - are combined to form the light source that exposes the film. Modern additive printers separate white light from a tungsten-halogen bulb into its red, green, and blue components by using a set of dichroic mirrors.These mirrors can be made to give sharp cutoffs at specified wavelengths and high efficiency in regions of the spectrum they are intended to reflect.

You can also combine them with certain KODAK WRATTEN Filters to give the required spectral bands. This allows independent (and often automatic) control of each of the primary colors using neutral density filters and electromechanical light valves. The red, green, and blue beams are then recombined and focused at the printing aperture. Usually, provision is made for the insertion of a filter (such as an ultraviolet-absorbing KODAK WRATTEN Filter No. 2B) in the recombined beam.

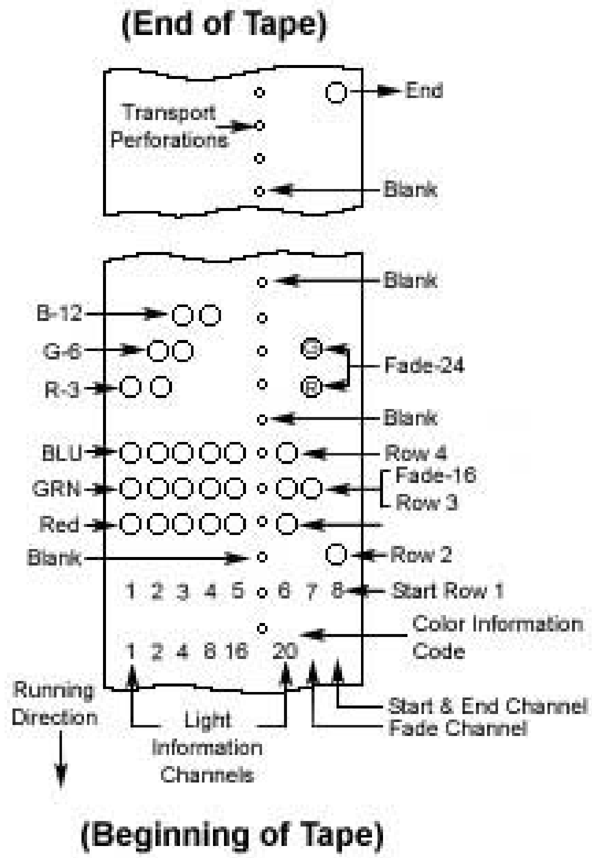


Figure 75
Perforated paper tape

You can control the electromechanical light valves manually or automatically. The manual control used to set the tight valve is usually called the TRIM setting and is used for overall color correction, for example, when changing print emulsions. A perforated paper tape (Figure 75) is used for automatic control of the light valves, called the TAPE setting.

The paper tape, which can be read through a high-speed reader, handles scene-to-scene color correction quickly and effectively. Consequently, almost all intermediates and answer prints are printed on additive printers, while one-light release prints may be printed on either an additive or a subtractive printer.

Color Timing

in printing color originals onto color print films, a difference in overall printing exposure as small as 1 printer light (0.025) can be detected in print comparisons. The variations, both in overall printing exposure and color balance that can be tolerated for a particular scene, however, depend on the scene composition, the subject matter, the brightness range of the scene, and whether a deliberate departure from neutral balance is desired.

Color timing demands considerable experience in printing a wide variety of scenes and careful observation of scenes along with careful observation of the results obtained in the final picture. in order to "calibrate the eyeball," it is helpful to make a series of picture tests on the equipment used for production printing. These tests, which are kept for reference, show the effects on the print of small changes in overall exposure and color balance.

It is possible to estimate roughly the photographic effect of a given color-balance change in the printer by observing a test print through selected combinations of color balances.

Even though each scene may be acceptable in itself, further modification in the scene-to-scene timing may be required when a given scene is assembled with others. Such changes are often necessary to overcome adaptation effects resulting from observation of the scene immediately preceding the scene in question when the print is projected. Often, you can decide these changes only after looking at the first trial print.

The most effective way to color-time any material is to use an electronic color analyzer. This instrument displays a positive video image of the original color negative, color reversal, intermediate, or print, and allows the operator to select color printing information.

Additive Timing. As described before, the red, green, and blue light valves in the additive printer can be adjusted automatically using the perforated paper tape. The TAPE values usually run 1, 2, 3 . . . up to 50 for each primary color and are called printer "points" or printer "lights." The addition of a printer point adds 0.025 Log Exposure, so adding 12 printer points adds a stop (0.30 Log Exposure) of exposure. The standard printer setup for a laboratory is usually 25-25-25 for the red, green, and blue TAPE settings. if the original to be printed was a stop heavier in density than the laboratory's "standard" original, the TAPE setting might be 37-37-37, allowing a one-stop exposure to compensate for the dense original.

Differences in the types of films being printed can be accounted for by changing the TRIM, or manual red, green, and blue settings. The TRIM settings can also be changed to adjust for emulsion crossovers and to make minor day-to-day printer control adjustments.

(The manual control used to set the tight valve is usually called the TRIM setting and is used for overall color correction, for example, when changing print emulsions.)

The TAPE settings tell the printer what red, green, and blue valve settings to use for a scene, and the cueing system tells the printer when to make the change. The cueing system to trigger the TAPE can use a variety of methods such as a microprocessor and a frame-count cueing (FCC) system.

Subtractive Timing. Scene-to-scene timing of color originals is seldom done on continuous subtractive printers because of the difficulty in making filter changes.

On most continuous subtractive printers, one printer light (diaphragm) is equal to 0.05 Log Exposure, and the light is used to make an equal exposure change in all three emulsion layers. The color-compensating filters are used to make an exposure change in each layer.

Motion Picture Laboratory Control of Color Duplication

Motion picture laboratories balance several sources of variability in producing consistent, high-quality prints through the two-stage master positive and duplicate -negative duplicating system. A paper published in the October 1976 issue of the SMPTE Journal (Vol. 85, No. 10), entitled "A Simplified Motion Picture Laboratory Control Method for improved Color Duplication" by John P. Pytlak and Alfred W. Fleischer, outlines a method for achieving high-quality prints based upon the concept of LAD-Laboratory Aim Density See Kodak Publication No. H-61, LAD -Laboratory Aim Density For more information.

In the past, direct printing information has been of little value to a cinematographer since it has usually been reported in terms of numbers on an arbitrary printer scale. In the LAD control method of reporting camera exposure, the printer is adjusted so that the LAD standard negative patch with its specified densities prints to a density of 1.0 END on the print near the center of the printer scale (e.g., 25-25-25). This printer exposure is considered standard. The difference in printer lights (1 printer light = 0.025 Log H) from this standard, necessary to produce a good print, is a reliable and reproducible measure of the printing characteristics of the negative. The printing characteristic of a master positive or duplicate negative can also be uniquely specified by the timing difference in printer lights for the LAD standard balance.

The LAD control method provides a simple and repeatable method of setting the calibration controls on an electronic color analyzer to correlate with the results obtained in printing. The printing exposure required for any printing original can be easily determined by viewing the film on an electronic color analyzer setup using the LAD method.

The specific Laboratory Aim Density (LAD) values for different film products are available from Kodak. Contact your Entertainment Imaging representative for details.

Digital Special Effects

Special effects play an important role in the storytelling process of motion picture films. The use of special effects has increased rapidly with the advances made by the leading companies and individuals in the field. Historically, these effects are produced with the use of film and optical printing equipment.

Eastman Kodak Company and others have developed electronic systems to handle all the steps necessary to yield a finished optical effect. These systems have a much shorter finishing time and provide overall improved quality.

The CINEON Digital Film System is a Kodak system which transfers images originated on film to a digital format for electronic compositing, manipulation and enhancement, and outputs back to film with no loss of image quality.

Cinesite is a Kodak subsidiary with facilities in the U.S. and Europe. Cinesite is a full-service digital studio that offers state-of-the-art visual effects and digital imaging services for feature films, large-format films, and television and music videos.

To create a standard blue-screen composite shot using film products and optical methods may take days or weeks. An electronic system can do the same task in one day, and the quality, with the use of a very high-quality intermediate film product, is better because it does not suffer optical generation losses.

Applications

The digital special effects system, with the use of film, has found application in the following areas:

Feature Films: There have been many new creative special effects that were impractical using conventional optical techniques.

Scene Salvage: Scenes that might be considered unsuitable because of unwanted artifacts, such as wires or microphones, or which have been accidentally damaged by scratching or misprocessing can now be salvaged. Color correction is simple.
Restoration: Scratches and other mechanical damage artifacts can be removed. Color splotches and dye fading can be compensated for. Unsteady images can be stabilized.
Stock Shots: A digital stock shot library can supply background images that do not suffer the generational losses of optical printing.

Sound-Track Printing

An "optical recorder" is the instrument that transfers the audio information from an electrical signal to an optical image. There are two main types of photographic sound recorders: those that use a combination of a mask and a moving-mirror galvanometer, and those that use a light valve.
The recorder's function is to place a uniform exposure over the appropriate area of the film. Using a galvanometer, the exposure is made with a narrow beam of light whose width is made to vary in accordance with the audio signal. in Figure 76, the light path is from the lamp to the film. The lamp, through the condenser, uniformly illuminates the mask. The mirror on the moving-mirror galvanometer reflects the light transmitted by the illuminated mask. This light is imaged by the lens onto a narrow rectangular slit. Finally, the light beam passing through the slit is imaged by the lens onto the film. The system is adjusted so that half of the aperture is illuminated through the mask when no signal is present. An input audio signal causes the galvanometer to oscillate. The oscillation causes the reflected image of the mask on the aperture to be raised or lowered,

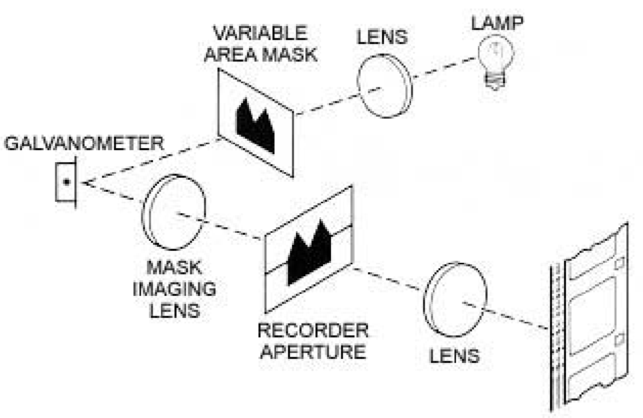


Figure 76
Schematic of a galvanometer-type sound-track recorder

thereby varying the width of the illuminated portion of the aperture.
A light-valve recorder (Figure 77) operates on a similar principle but replaces the mask and galvanometer with two or more metallic ribbons. The metallic ribbons are positioned in the field of a strong magnet, and the audio current is passed through them. A force is always exerted on a current-carrying conductor located in a magnetic field. This force is proportioned to the current and alters the separation of the ribbons in accordance with the audio signal.

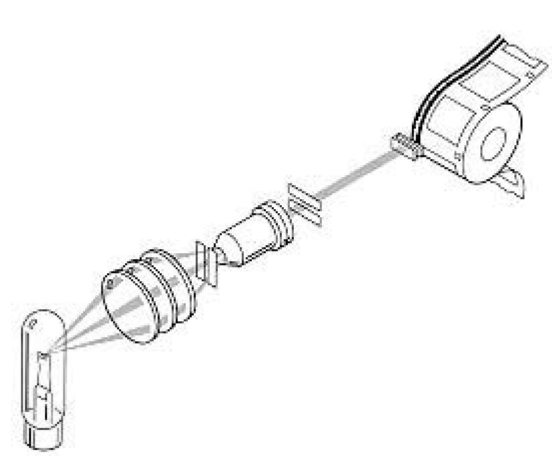


Figure 77
Schematic of a light-valve recorder

The variable-area sound tracks produced by these recorders are made up of dense and clear areas. in an ideal track, the dense parts are completely opaque and the clear parts completely clear. if the dense part is not opaque, there is a slight loss in the signal-to noise ratio. However, the clarity of the minimum density (D-min) portions of the track is much more important; the volume output is reduced rapidly as the D-min rises, and if the D-min is grainy, additional noise is produced.
An ideal variable-area track has perfectly sharp edges between the dense and clear areas. in reality, if an exposure is made through a narrow slit composed of two knife edges, the light will spread under the edges, causing some exposure (Figure 78). This exposure produces density in accordance with the film's characteristic curve. Thus, the image recorded is larger than the surface over which the light was incident. When the sound negative is printed onto the print stock, the print exposure is proportional to the negative density. if the negative and print densities are properly balanced, the final print transmittance is proportional to the original exposure (Figure 79). Thus, a two-step system is self- compensating for the effects of the image spread. Aside from production considerations, this self-compensation or "image-spread cancellation," is the major photographic reason for using a two-step system for printing photographic sound tracks.

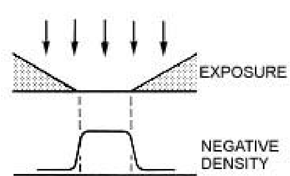


Figure 78
Image spread

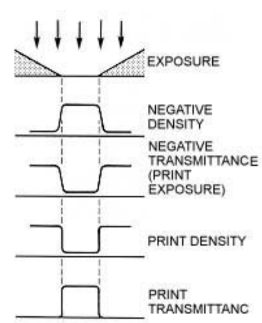


Figure 79
Image spread compensation

MORE ABOUT LAD

Laboratory Aim Density

In the Laboratory Aim Density (LAD) printing control method, a standard control patch specifies densities midway between the minimum and maximum of those typically obtained for a normal camera exposure. These specific densities have suggested tolerances for each film type in the duplicating and print system, and are listed in this publication, along with helpful hints for their use and evaluation. All color films in a production, regardless of film type or origin, are timed with respect to the LAD control film. Each frame of LAD control film, having the standard patch, with proven accurate exposures plus other information, is available from Eastman Kodak Company.

- Set Up an Electronic Color Analyzer and Printing Control
- LAD for EASTMAN Color Negative Film
- LAD for EASTMAN Color Print Film
- LAD for Master Positives. Using EASTMAN Color Intermediate Film 5243 / 7243
- LAD for Duplicate Negatives. Using EASTMAN Color Intermediate Film 5243 / 7243
- LAD for Master Positives. Using KODAK VISION Color Intermediate Film 2242 / 3242 / 5242 / 7242. Using EASTMAN EXR Color Intermediate Film 2244 / 3244 / 5244 / 7244
- LAD for Duplicate Negatives. Using KODAK VISION Color Intermediate Film 2242 / 3242 / 5242 / 7242. Using EASTMAN EXR Color Intermediate Film 2244 / 3244 / 5244 / 7244

Eastman, EXR, 2244, 3244, 5243, 5244, 7243, 7244, 2242, 3242, 5242, 7242 are trademarks.

Laboratory Aim Density (LAD) Control Method

To maintain optimum quality and consistency in the final prints, the laboratory must carefully control the color timing, printing, and duplicating procedures. Laboratory Aim Density (LAD) Control Film provides a simple, effective, and easily implemented control method for the production of master positives and duplicate negatives from negative originals.

All film in the printing original should be color timed relative to LAD Control Film. The LAD Control Film is printed at the center of the printer range, usually TAPE 25-25-25. Printer setup (speed, bulb voltage, TRIM, filtration, etc) is determined by printing the large gray patch in the LAD Control Film to the specified Laboratory Aim Density values on the duplicating film, chosen to be at the center of the usable straight-line portion of the duplicating film's characteristic curves. The Status M Laboratory Aim Density values for EASTMAN EXR Color Intermediate Film are as follows.

For the Master Positive LAD Aim:

Red	Green	Blue	Tolerance
1.15	1.60	1.70	+0.10 density

For the Duplicate Negative LAD Aim:

Red	Green	Blue	Tolerance
1.00	1.45	1.55	+0.10 density

The LAD Control Method assumes that the film and process sensitometry are within specification.

Film to Video Transfer

When you transfer the film directly to video, you can set up the telecine with a Telecine Analysis Film produced on EASTMAN EXR Color Intermediate Film 5244. The Telecine Analysis Film (TAF) consists of a neutral density scale and an eight-bar color test pattern with an LAD surround.

The TAF gray scale provides the scanner operator (colorist) with an effective way to adjust subcarrier balance and to center the telecine controls before timing and transferring a film. The TAF color bars provide the utility of electronic color bars, even though they do not precisely match the electronically generated color bars. Using the TAF will help obtain optimum quality and consistency in the film-to-video transfer.

Printing Conditions

In all printer setups for printing EASTMAN EXR Color Intermediate Film 5244/7244, include a heat absorbing (infrared) filter such as a KODAK Heat Absorbing Glass, No. 2043, and a KODAK WRATTEN Gelatin Filter No. 'E to absorb ultraviolet (U\0 light. For high light output with very long bulb life, operate the printer bulb at approximately 80 percent of rated voltage. Use a well-regulated constant current dc power supply.

Print the LAD Control Film at the center of the printer balance range, usually TAPE 25-25-25 on an additive printer. Print other scenes in the original as determined by color timing relative to the LAD Control Film. Choose the printer speed and filtration to normalize the additive TRIM settings near the center of their range to allow for slight variations in film and printer.

On subtractive printers, choose the filter pack and light control for both the removal and addition of filters for color correction. You can use EASTMAN Lamphouse Modification Filters in subtractive printers to more closely balance the spectral characteristics of subtractive lamphouses with additive lamphouses so that prints made on a subtractive printer more closely match those made on additive printers. On optical printers, set the lens aperture considering sharpness, depth of focus, and light transmittance characteristics. Use ground glass or other diffusers to improve uniformity of illumination. Clean and align optics for optimum light output and uniformity.

Printing Conditions

You can make satisfactory black-and white duplicates of black-and-white negatives or prints, or color prints using a continuous additive printer, such as the Bell & Howell, Model C, equipped with a 1000-watt lamp at 95 volts dc, ground-glass diffuser, printer speed of 180 feet per minute (35 mm film). For duplicating color prints, set the trim settings (red, green, and blue) at 24 and the vane settings (red, green, and blue) at 36. For duplicating black-and-white negatives or prints, set all trim settings at 24 and all vane settings at 33.

You can make duplicates of black-and-white negatives using a subtractive printer, such as the Acme optical printer, equipped with a 750-watt lamp at 86 volts dc, ground-glass diffuser, and at a printer speed of 20 feet per minute (35 mm), a 170-degree shutter opening, and aperture of f/5.6. You must insert a KODAK Heat Absorbing Glass, No. 2043, and a KODAK WRATTEN Neutral Density Filter No. 96 in the main beam.

Considerations in the Illumination of Photographic Darkrooms

Published in the March 1982 SMPTE Journal is a good introduction to the use of safelights in a motion-picture lab environment. However some of the recommendations are outdated by new technology.

The Process Lighting (Safelight) Group of Testing Technology Division at Kodak has experts to design and specify safelighting for film production areas.

They have demonstrated new safelight concepts that greatly reduce a films exposure, while maintaining operator comfort and safety. Light Emitting Diodes (LED's) are available in a variety of spectral outputs, and because of their narrow spectral output, long life, and visual efficiency, are ideal for safelight applications. LED's provide marker lights to outline halls, doors and obstructions. Clusters of LED's can provide general illumination. LED's can even replace the misused filtered flashlights. The luminescent "green fluorescent" markers are being phased out in favor of LED's.

Kodak has worked on some proprietary technology which may someday be licensed or sold outside Kodak.

General illumination (low pressure sodium vapor or Series 8) are generally not used in film manufacturing, because of the wide mix of products used, and the need to provide maximum "safe time". However, for a MP lab, low pressure sodium vapor provides excellent visual efficiency, with a very narrow spectral output (589nm) where ECP is least sensitive. Excellent units are made by Osram (Duka 10 Studio Safelight) and Thomas (Duplex Super Safelight). "Blocking Filters" should be used to filter out all but the 589nm line for maximum safe time. The Kodak Series 8 or Series 13 safelight filters can be used as blocking filters with a low pressure sodium vapor lamp. One drawback of a low pressure sodium vapor lamp is the time required to "warm up".

A Series 8 or Series 13 filter used with a tungsten bulb is good, but has less safe time for a given illumination level because it has a broader spectral output.

General safelight illumination (sodium vapor or Kodak Safelight filters) should never shine directly on the film. Filters must be checked periodically for fading or physical damage.

Safelight exposure tests should be run with the film periodically to test actual safe time. Safelight fog results most obviously in an increase in D-Min, but an undesirable softening of toe contrast may occur well before there is visible fog. So safelight tests must be run before and after the film has an image (sensitometric) exposure, as well as simple fogging tests.

Safelight test exposure times should be representative of the actual time expected in practice: the time at a presplicing operation is less than a minute, the time in the feed-on elevator of a high speed printer may be only two minutes, but the time a wound roll may sit in a safe-lit hallway and be edge-fogged may be several days. We have learned that reciprocity law failure is an important part of the equation in evaluating safelight sensitivity. (5386 is more sensitive to safelights than 5384 mostly because of its greatly improved reciprocity behavior).

A useful tool exists to help customers evaluate their safelights . . . the Eastman spectroradiometer. It can be used to measure spectral outputs of each safelight to determine if there is unwanted illumination. With most safelights there should be enough light to make a measurement if the probe is put right up to the safelight.

What is Additive Printing?

Instead of a single white light source with color correcting filters, three separate colored sources, red, green, and blue, are combined to form the light source that exposes the film. Modern additive printers separate white light from a tungsten-halogen bulb into its red, green and blue components by using a set of dichroic mirrors.

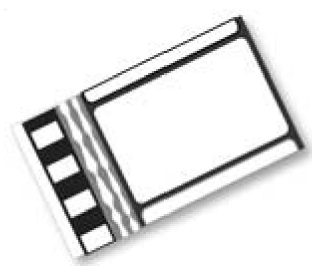
What is Subtractive Printing?

Whenever color printing is involved, the printer, or more exactly the printer lamphouse, must be able to control the red, green and blue components of the white light source.

A subtractive printer is a white light printer in which color correction (changing the relative amount of red, green, and blue light) is effected by inserting color-correcting filters in the light path between the light source and the printing aperture. Overall, light changes (intensity changes) are made either by means of a variable aperture or a neutral density filter. Subtractive printing is often used for "release" printing (making multiple prints after an answer print has been approved) because of the minimal number of scene-to-scene color changes required. Printing requiring a lot of scene-to-scene color corrections is very difficult on a subtractive printer because a new filter pack must somehow be physically inserted in the light beam for every correction.

Silverless Soundtracks

For many years the technical community of the motion picture industry has been searching for and evaluating methods of implementing "dye-only" analog soundtracks into release prints. The use of "dye-only" analog soundtracks would eliminate the need for soundtrack redevelopment, which is the historical means of reintroducing a variable silver image into analog soundtracks printed on color positive stock. The development and deployment of red L.E.D. (light emitting diode) sound track readers has now made it possible to chart an implementation path which can make this long awaited advancement a reality.



The Dye Track Committee, a group of motion picture executives dedicated to replacing silver-applicated analog 35 mm soundtracks with pure cyan dye tracks. The DTC, formed in 1998, includes motion picture distributors, exhibitors, film stock manufacturers, and film laboratories, as well as members of the motion picture audio technical community.

Converting to a silverless cyan track offers major benefits to both the motion picture industry and to the environment. The cyan track is not only safer and simpler to produce, it is less damaging to the environment, reducing the use of water and chemicals in the developing process.

The cyan dye track requires a red light reader in the projector's analog soundhead: without one, the new soundtracks will play badly or not at all. A red light reader provides several practical benefits to the theatre, beyond the ability to play the new soundtracks, and the Dye Track Committee encourages the conversion of all theatres to red readers.

Approximately 85 percent of US screens have already been converted to red light readers, with most major studios intending to evolve to the cyan dye track format. If you own or operate a theatre that is not yet equipped with a red light reader in your projector's analog soundheads, we recommend that you install one as soon as possible.

First Film Release with Pure-Dye Analog

Soundtracks on All Prints Due in September

Hollywood, August 11, 2003 - DreamWorks SKG and the Dye Track Committee have announced that all prints of the upcoming Jason Biggs, Christina Ricci romantic comedy Anything Else will feature pure-dye cyan analog soundtracks. Slated for release on September 19, 2003, the new feature is the first to use environmentally friendly cyan tracks in lieu of traditional silver-applicated tracks on 100 percent of its initial US print run.

Environmental benefits of the cyan soundtrack include the elimination of caustic chemicals and silver itself from the print manufacturing process. The switch also enables a significant reduction in water usage: were all US print manufacturing to convert to cyan tracks, the ongoing savings would be equivalent to the drinking water needs of a town of 75,000. Silver-applicated tracks have the further disadvantage of a comparatively high manufacturing reject rate.

"We at Dolby are proud to have been part of the Dye Track Committee since its inception and to have developed the technology to make this transition possible," said Ioan Allen, Vice President, Dolby Laboratories, and Dye Track Committee member. "As the world becomes more environmentally aware, it is good that the film industry is making its own contribution to the quality of the future."

In exhibition, cyan soundtracks require red-light soundtrack readers, which have been standard in new projector models for some time. It is estimated that approximately 85 percent of the projectors in use in the US are now equipped with red-light readers, and the members of the National Association of Theatre Owners (NATO) have announced a target of all their members' screens being equipped by this summer. This wide usage has made a 100 percent cyan release practical for the first time.

John Fithian, NATO's president, announced that "NATO and its theatre company members are pleased to support the conversion to environmentally friendly cyan tracks. We are grateful that DreamWorks has taken this important step, and we look for other studios to follow suit."

The move to pure-dye soundtracks has been spearheaded by the five-year-old Dye Track Committee, which consists of representatives of Eastman Kodak, Dolby Laboratories, and print laboratories that include Deluxe and Technicolor. The technology has been thoroughly and successfully tested by labs, film distributors, and exhibitors over the past three years. Digital soundtracks are not silver-applicated, so are not affected by the decision to use cyan analog tracks.

Frequently Asked Questions

What is the schedule for conversion to dye tracks?

The conversion to the interim High Magenta format is already under way, with all new theaters built in the last 5-6 years equipped with red readers. Retrofitting of red readers to projectors in existing theaters is proceeding steadily. Some laboratories are now making release prints in the High Magenta format, including all releases from Warner Bros and all prints made by Fotokem laboratory. When the proportion of red readers reaches about 85% (estimated in 2002 in the US) it is likely that the laboratories will announce their intention to switch their production to Cyan Dye tracks.

Have Cyan Dye sound tracks been tested in theatres?

Yes, a limited release of prints of Get Over It (Miramax) with cyan dye sound tracks was made in March 2001 and distributed to theatres known to be playing analog sound tracks using red readers. These prints ran completely successfully for 3 - 4 weeks. It is planned to make larger-scale releases of prints with cyan dye sound tracks during the remainder of 2001 to further demonstrate the technical feasibility of this sound track format and to encourage other theatres to convert their projectors to red sound readers in preparation for the switch to cyan dye sound tracks.

Can I use the same soundtrack negative to print standard silver, high magenta, and cyan dye print soundtracks?
No specific recommendations have been offered with regard to soundtrack negative density, although the optimum negative density for both the high magenta and cyan dye formats will typically be higher than that used for traditional silver plus dye prints. As always, optimum negative density should be determined for any print format using the cross-modulation test.

Will the same print soundtrack conversion(s) occur for 16-mm prints?
We've had no discussions with manufacturers of 16-mm projection equipment. We believe, however, that 16-mm projectors may be more difficult to convert to red readers than 35-mm projectors. As a result, we are not recommending a conversion to cyan dye tracks for 16-mm for the foreseeable future. While theoretically high magenta prints are very likely to be satisfactory for 16-mm, no tests have been run.

What are the implications for archival film elements stored in preservation vaults?
Black and white prints (with silver tracks of various formats including variable-area and variable-density types) have been reproduced in tests using red LED sound readers with excellent quality, indistinguishable from that produced by white-light readers. Color prints with redeveloped silver-plus-dye sound tracks may experience a little distortion when reproduced with red LED readers, as is the case with current release prints with reveloped silver-plus-dye sound tracks (hence the reason for introducing the compatible High Magenta sound track format). The printing of archived sound negatives, intended for making redeveloped silver-plus-dye sound tracks, to produce cyan dye sound tracks, is currently being evaluated and will be reported on to the Dye Track Committee in due course. It is anticipated that some film processing laboratories will retain the facility to print redeveloped silver-plus-dye sound tracks for some time, to provide this service for film preservation clients.

Notes

UNDERSTAND THE QUALITIES OF NATURAL DAY LIGHT

Sunrise In pre-dawn hours, the cobalt and purple hues of the night sky predominate. But as the sun inches over the horizon, the landscape begins to reflect the warm gold and reds hues of the low-angled sunlight. Within minutes, the light shifts to a rich blue. During this time of day, the green color of grass, tree leaves, and other foliage is enhanced while earth tones take on a cool hue. Landscape, fashion, and portrait photographers often use the light available during and immediately after sunrise.
Midday During midday hours, the warm and cool colors of light equalize to create a light the human eye sees as white or neutral. On a cloudless day, midday light is often considered too harsh and contrasty for many types of photography, such as portraiture. However, midday light is effective for photographing images of graphic shadow patterns, flower petals and plant leaves made translucent against the sun, and for images of natural and manmade structures.
Sunset During the time just before, during, and just following sunset, the warmest and most intense color of natural light occurs. The predominantly red, yellow, and gold light creates vibrant colors, while the low angle of the sun creates soft contrasts that define and enhance textures and shapes. Sunset colors create rich landscape, cityscape, and wildlife photographs.
Diffused light On overcast or foggy days, the light is diffused and tends toward the cool side of the color temperature scale. Diffusion spreads light over a larger area making it softer. Light may be diffused by clouds or an overcast sky; atmospheric conditions including fog, mist, dust, pollution, and haze; or objects such as awnings or shade from trees or vegetation. You can intentionally diffuse strong light by using a "scrim," a panel of cloth such as thin muslin, stretched tightly across a frame. The scrim is held between the light source (the sun or a studio light) and the subject to diffuse the light. Diffused light creates saturated color, softer and more "open" shadows, and highlights that are more subdued than in open light. Shadows are softer and provide less separation of foreground and background objects. Why do my pictures have gray snow and gray graduation robes?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This image shows a good range of grays from white to nearly black.
Cameras, specifically reflective camera meters, see only gray. Objects that you see as "neutral" gray, an even mix of black and white, reflect 18 percent of the light falling on them and absorb the rest of the light. In a black-and-white world, objects that you see as white reflect 90 percent of the light and absorb the other 10 percent. Objects you see as black absorb virtually all of the light. Everything else maps to a percentage, or shade, of gray. Each percentage, or swatch on a grayscale chart from black to white, reflects and absorbs different amounts of light.
In color scenes, the light and dark values of color correspond to the swatches of gray on the grayscale. A shade of red, for example, has a corresponding shade of gray on a grayscale. The lighter the color's shade, the more light it reflects.
A camera's reflective light meter (which measures light that is reflected back to the camera from the subject) assumes that everything you focus on is 18 percent gray. The meter also expects to see an average scene, one that contains a balance of dark and light tones. In average scenes, the camera's meter produces accurate rendition of what the human eye sees. Unlike the human eye, however, a reflective meter does not automatically adjust for non-average scenes, such as large expanses of snow, white sand, and black objects. Instead, the meter assumes these scenes also reflect 18 percent of the light. When a camera adjusts the exposures in these types of scenes to 18 percent reflectance, you get pictures with gray snow or gray graduation gowns that should have been black.
Gray cards You've probably heard of or may have used photographic gray cards, a simple card that consistently reflects 18 percent of the incident light (light that has not yet reached the subject). When the gray card is placed in the same light as the subject and angled slightly toward the main light, you can point the camera at the gray card and use the resulting meter reading (the f/stop and/or the shutter speed, depending on the shooting mode you use), to take the picture. Using a gray card tells the camera what objects in the scene are 18 percent gray. With the gray card as a reference point, the camera then produces good colors with full tonal ranges.
While it may be impractical to use a gray card to meter every scene, it's handy to know that you can also meter from green grass, or even from a pair of blue-jeans and get reasonably accurate readings.


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There's no reason to put your camera away in harsh light if you choose your subjects with this type of light in mind.

Reflectance and Color Light has color and reflectance that inevitably affects photographs. For example, if you photograph a person sitting on the grass, the closer the subject is to the grass, the more green will be reflected in the subject's face. Similarly, photographing a subject under an awning or near a colored wall also results in that color reflecting onto the subject. The amount of reflectance depends on the proximity of the subject to the color and the intensity of the color.The point is to be aware not only of the color of the existing light, but also of surrounding structures that reflect onto the subject. In some cases you can use a colored reflector (silver, white, or gold) to help offset colors reflected onto the subject by nearby grass, walls, and other objects.

What is the best light? Photographers often describe light as harsh or soft. Harsh light creates shadows with well-defined edges. Soft light creates shadows with soft edges. There are traditional uses for each type of light. Understanding the effect of each type of light before you begin shooting is the key to using both types of light, and variations in between, effectively.

Hard/harsh light Hard light is created when a light source, such as the sun in a cloudless sky at midday, an electronic flash, or a bare light bulb, produces a concentrated spotlight effect. This directional light results in dark, sharp-edged shadows as well as a loss of detail in highlights and shadows. Portraits taken in harsh overhead light create dark, unattractive shadows under the eyes, nose, and chin. This type light is also called contrasty light. Contrast is measured by the difference in exposure readings (f/stops) between highlight and shadow areas. The greater the difference, the higher the contrast. Although the light is contrasty, it also produces well-defined textures and bright colors. Hard light is best suited for subjects with simple shapes and bold color.

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I moved slightly into the shadows on a sunny day to capture this dogwood blossom.

Working with hard light To soften hard light, you can add or modify light on the subject by using a fill flash or a reflector to bounce more light into shadow areas. In addition, you can move the subject to a shady area, or place a diffusion panel (scrim) between the light and the subject. For landscape photos, you can use a graduated neutral density filter to help compensate for the difference in contrast between the darker foreground and brighter sky.

Soft light Soft light is created when a light source, such as the sun, is diffused by clouds or other atmospheric conditions. Diffusion not only reduces the intensity (quantity) of light, but it also spreads the light over a larger area (quality). In soft light, shadows edges soften and transition gradually, texture definition is less distinct, colors are less vibrant than in harsh light, detail is apparent in both highlights and shadow areas of the picture, and overall contrast is reduced.

Working with soft light In soft light, consider using a telephoto lens and/or a flash to help create separation between the subject and the background. While soft light is usually well-suited for portraits, it is less than ideal for travel and landscape photography. In these cases, look for strong, details and bold colors, and avoid including the overcast sky in the photo. Macro photographs are also suited to diffuse lighting.

© Charlotte Lowrie

This photo was taken in soft, mid-morning light on a cloudy day.

Why do people in my pictures look like they have raccoon eyes?

In part, the answer is the direction (or position) of light on the subject. Whether natural or artificial, the direction of light can determine the shadows in the scene. Raccoon eyes result from hard, top light. You can use both the type and direction of light to reveal or hide detail, add or reduce texture and volume, and help create the mood of the image.

Front lighting Front lighting is light that strikes the subject straight-on. This lighting approach produces a flat, one-dimensional effect with little texture detail, and with shadows behind the subject, as seen in many on-camera flash snapshots. Harsh top lighting creates dark shadows under the eyes and nose as shown here.

Side lighting Side lighting places the light to the side of and at the same height as the subject, essentially dividing the subject in half: one side of the subject is brightly lit, and the other side in deep shadow. While this technique can be effective for portraits of men, it is usually considered unflattering for portraits of women. However, a variation of side lighting is high-side

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lighting, a classic portrait lighting technique where a light is placed to the side and higher than the subject.

Top lighting Top lighting, as the term implies, is light illuminating the subject from the top, such as you'd find at midday on a sunny, cloudless day. This lighting produces strong, deep shadows. While this lighting direction is suitable for some subjects, it is usually not appropriate for portraits unless fill light is added using a flash or reflector.

However, a variation on top lighting is "Butterfly" lighting, a technique popularized by classic Hollywood starlet portraits. Butterfly lighting uses high, front, top light to create a symmetrical, butterfly-like shadow under the nose.

© Charlotte Lowrie

This photo was taken in soft, late-afternoon side lighting.

Backlighting As the name implies, backlighting is light that is positioned behind the subject. This technique creates a classic silhouette, and depending on the angle, can also create a thin halo of light that outlines the subject's form. While a silhouette can be dramatic, the contrast obliterates details in both the background and subject unless a fill flash is used.

In addition, backlighting often produces lens flare displayed as bright, repeating spots, or shapes in the image. Flare can also show up in the image as a dull haze or unwanted rainbow-like colors. To avoid lens flare, use a lens hood to help prevent stray light from striking the lens, or change your shooting position.

Midday light casts the dominant shadows in this photo

© Charlotte Lowrie

What Else Should I Know?

In addition to the terms and concepts mentioned in this article, here are more terms and definitions associated with photographic lighting.

Specular highlight typically referred to as the "hot spot" in an image, a specular highlight is a reflection of the main light source the brightest spot in the image.

Ambient light is the pervading light in a scene.

Key light is the main light in the scene. The key light casts the dominant shadows and reveals both texture and volume.

The inverse square law This law states that the intensity of illumination is inversely proportional to the square of the distance between the light and the subject. In simple terms, this means that if you double the distance between the light and the subject, the light falling on the subject is one-fourth of the original intensity because the light covers a larger area. This, of course, affects exposure. In theory, if the light-to-subject distance is doubled, exposure (f-stop) increases by two stops.

Light ratios are a way to express the differences between the exposure for the key light and shadow areas of a scene. The higher the ratio between the key light and the shadow areas, the greater the contrast. For example, if the key area (18 percent gray) meters at an exposure of f/11, you would assign it a value of 1. If the shadow area exposure meters f/8, a value of 2 is assigned to indicate that it is one stop less than the key value. The ratio between the key and shadow areas is 1:2. A third value can be assigned to the specular highlight. If the specular highlight exposure meters at f/16, one stop less than key, it is assigned a value of 2. That makes the ratio between the specular highlight, the key, and the shadow 2:1:2.

Putting it all Together

While you may not be able to control the light, especially natural outdoor light, you may be able to:

- Move the subject.
- Change your position.
- Use a reflector or scrim.
- Use a filter or changing white balance settings to balance color for or enhance the light.
- Wait for better light or a light color that enhances the subject or message of the picture.

© Charlotte Lowrie

Using a simple light setup and inexpensive tabletop studio, you can experiment with different types of light intensity and direction

While the concepts of lighting seem complex at first, they will become second nature to you as you discover the characteristics of light and how they affect your pictures. With this overview, you're in a better position to use light to create and control the shape, form, mood, and contrast of your photos and to get more accurate exposures, tones, contrast, and color.

If you consistently train yourself to "look for the light," you'll begin to see pictures, often spectacular pictures, that you may have overlooked before.

This photo takes advantage of midday sun and backlighting.

© Charlotte Lowrie

Views By Robert Winkler

Photography's overstated beliefs is that it is best to shoot either very early in the morning or very late in the afternoon. The light at those times, when the sun is low, has been called magical. Colors are warm, shadows are long, subjects glow with piercing clarity. But the use of this light has become so pervasive, and its images are so typical, that from the hours just after dawn and just before dusk, much of the magic has gone. If photography is writing with light, why use a dictionary with so few words? Photographers on the move are hard-pressed enough to make the most of existing conditions, whatever they are—we cannot always wait for the ideal light.

A correlative assertion is that portraits in the noonday sun should be avoided because shadowy eye sockets do not flatter the subject. But in Rose, a portrait by Edward Weston taken in Mexico in 1926, prominent shadows under a woman's nose and chin point to a sun directly overhead. We cannot see her eyes, not because of the direction of the light but because they are closed, and her long lashes cast shadows like tiny spokes on her upper cheeks. There is nothing unflattering about the portrait, but more important, it is unforgettable.

Dos and don'ts are of questionable value in photography, where an error of judgment will mean, at worst, a few wasted frames. At best, it will teach you what to do differently next time. Once a style becomes generally accepted, it is time to depart from the norm. When breaking the rule becomes the fashion, it is time to rediscover convention.

Light is the essence of photography, but there is no secret to understanding it. Light comes naturally to a photograph—it catches your eye, and all you have to do to possess it is trigger the camera. Although much has been said about controlling light and treating it as a tool, the way to know light is to let it happen.

Every change of light holds something for the photographer. Its intensity can be strong or weak. It can hit the subject from the front, back or side. It will bounce off surfaces differently, depending on their reflectance. It can come from a high or a low angle. It can be hard, with dark and distinct shadows, or soft and almost shadowless, or diffused, with shadows that are definite but faint.

Individual style in photography, the stamp of which is often a characteristic lighting, can still exist. But style in photography must be allowed to emerge. Its development is a process of discovery; by exploring all that light offers, you arrive at style naturally.

Watch how the light changes with each passing hour, and record distinct lighting effect on film. Shadows shift with time; the light begins warm, then gains in intensity and whiteness, and finally returns to warmth as the sun sets. Shadow pattern and light intensity have an emotional effect on the way a subject is perceived. As the light changes, so does the mood it evokes.

When a subject is of special interest, study it from various positions. If you shoot with the sun at your back, the light is with you, photographically speaking, because frontal lighting is flat (direct and even, with shadows not apparent) and bright. Exposures are relatively easy to measure, and the relative brightness of the subject means you can use a smaller aperture (for greater depth of field) and a faster shutter speed (to ensure camera steadiness and to freeze movement).

Photographing the same subject against the light, you may need to increase exposure, since the side facing the camera will be in shadow. To prevent flare in backlighting, use a lens hood. A lens hood also is advisable if the light hits the subject from the side. Exposure is trickier in sidelighting because contrast is high, with light and shadow coexisting equally. As a rule, if you are using negative (print) film, which is more tolerant of overexposure, take your exposure reading from the shadows. With slide (transparency) film, which is more tolerant of underexposure, read the highlights. If you are more concerned with a small area of the picture than with the scene as a whole, abandon the rule and expose for the critical section. Lighting contrast is of least concern when the sky is overcast. Shadows are practically nonexistent; the light is soft and even.

Sometimes you will want to wait for the proper light, or you might decide to return hours later. You can also come across a good subject only to find you've missed the right light for the day. Perhaps the subject now is entirely in shadow, while earlier it would have been bathed in light. Look up as well as around you when you study the light. Note the arc drawn by the sun, and build your shooting schedule around it.

Wherever we go, it is the same sun we see in the sky, but photographers occasionally attest to a place possessing a special kind of light. Two points on the globe, widely separate at different latitudes, will receive the sun from different angles, but will a photograph reveal the difference? The sun describes a certain arc in summer, a different one in winter, but does summer and winter light show up as such on film? A place's lighting character depends less on where the sun sits in the sky than on the particular landscape—natural and architectural—from which the light reflects. Special locations, therefore, may exhibit a light all their own: the American Southwest has its fiery desert reds, the waters of the Caribbean have their translucent blue, Antarctica casts a brilliant specular whiteness.

Interiors and Exteriors

There are two great lighting divisions in photography: daylight and artificial light. Color film cannot capture with accuracy the divergent color quality of both, so professional large-format and motion-picture films are manufactured in two types, one balanced for daylight, the other tungsten-balanced for interiors. Each type of film can be used over a range of color temperature. Color temperature, measured in degrees Kelvin, indicates the color of a light source. The color temperature of sunlight varies with the time of year and time of day; color temperature indoors depends on the type of lamps in use.

Sunlight has a high component of blue light; a light bulb, a high component of red. This is why daylight film used indoors without a filter will give pictures a reddish cast, while tungsten film used unfiltered outdoors produces bluish pictures. Black-and-white film is virtually unaffected by color temperature; the same film can be used indoors and out. In the 35-mm format, tungsten film is available only for making slides. Photo-processing labs can now do a passable job of adjusting the color balance of daylight negative film that is exposed indoors, so major manufacturers have stopped producing tungsten-balanced 35-mm print film.

Since electronic flash approximates the color temperature of daylight, it presents no problem when used with daylight film. The typically bright light and strong shadows of an undisguised electronic flash are sometimes desirable, and flash is useful in poorly lighted interiors. But flash has major drawbacks: it can destroy the existing light in a scene, it is difficult to visualize, its power and range are limited, and it is hard to control. To someone who wants to photograph unobtrusively, a flash unit draws attention and adds weight and bulk. However, color film has become so fast that no photographer need put up with the drawbacks of flash. Today, with color film rated at a speed of 800 or higher, you can shoot in deep shade or in fairly dim interiors without resorting to flash.

One of the best ways to learn about interior and exterior lighting is by studying the photographs you admire. Try to read them for hints about the direction, intensity, and hard or soft quality of the light. Then, try to duplicate the

photographer's lighting techniques with your own camera. Don't forget that light can create images of great power and beauty even when it is present in the least degree. Brassai's photographs of Paris at night are proof that darkness can shed light on photography, too.

The Varieties of Light

Certain qualities have been ascribed to photographic lighting that your own experience can confirm or contradict:

- Flat frontal lighting is thought to show surface features most accurately, while backlighting better describes a subject's form, and sidelighting enhances its beauty.
- Direct sunlight is usually too harsh for portraits, but sometimes the supposed defects it shows can be seen as features on a facial landscape, in which is written a story of experience.
- Hard lighting is described as cruel; soft lighting is said to beautify.
- High-key lighting may be ethereal; low-key lighting may be brooding.
- A light in the eye can give a portrait more life, and hard-angled lighting best conveys the impression of sharpness. Angled lighting also is best for revealing textures.
- Long shadows may add depth and beauty, or obscure the main subject and draw our attention away from it.
- Bright colors on a dull day can look all the more vibrant, or they can only underscore the dreariness.

Notes

VIEWS ON COLOR, LIGHT & THE MAGIC HOUR

Every landscape photographer worth his salt knows to foresake the noonday sun for the ever-changing light of dawn and dusk. Few, however, have a working knowledge of what makes these magical hours so attractive to the eye. Fewer still fathom why film transforms this natural magic into yet another dimension that can appear exquisitely mystical or dumfoundingly mundane.

My own learning curve is a case in point. Back in the sixties, I'd head off during magic hour with an arsenal of camera gear in my pack, sense that the light was beginning to turn warm, then only at the last minute pull out the heavy artillery to isolate an orange orb on the horizon in the center of a red glaze. My results were predictably trite. The visual power of magic hours involves so much more than selecting out a great sunrise or sunset by itself.

Early on, I realized that there was no simple trick that would turn one of the stark, raving beautiful light shows I was regularly witnessing into a fine photograph. That began to happen only after I gained a greater understanding of why I was so attracted to red light, why I tended to ignore what was happening to other parts of the atmosphere at the same time, and how my film was coding the colors of the scene before me in an entirely different way than my visual system.

Consider insects disturbing a party in your backyard by buzzing around a bare light bulb over the dinner table. They only start doing it during magic hour when they begin to mistake the bulb for the sun. You can't stop them, because they're on autopilot with guidance systems that keep them circling until you turn off the light or shoot them out of the sky with bug spray.

Humans also respond to a built-in visual bias toward brightness and hues at the warm end of the spectrum that can make them behave rather mindlessly with a camera in hand. But there's a difference. The bugs have no options. For them, the visual representation of an apparent sun triggers a predictable physical reaction. We have an infinity of choices. Creative photography begins where we stop blindly responding to the strongest visual representation before us. We can choose whether to mindlessly shoot or to use the stored visual memories of a lifetime to guide us toward a unique personal vision.

The ancients worshipped the sun, but couldn't look directly at it during the day. Thus the hours when the sun's intensity came to be attenuated near the horizon so that it could be clearly observed came to be considered spiritual. Legends were passed down about the otherworldly sources of the mystical colors. Not until the scientific revolution did logical explanations emerge.

In the mid-nineteenth century, a British scientist who climbed and explored mountains in his spare time came up with a plausible theory. John Tyndall, a close confidant of Darwin and Huxley, was a broad-spectrum kind of guy, so to speak. His habit of looking beyond the obvious in all the operations of the natural world paid off well in biology, where he helped verify Pasteur's germ theory of disease with experiments that negated the prevailing idea of spontaneous generation of life forms. In geology, his expertise on Alpine glaciers led him to Yosemite, where he validated John Muir's controversial theory that glaciation had sculpted its features. As a pioneer in atmospheric physics, he explained what became known as the "Tyndall effect," which accounts for light beams becoming visible from the side as they pour down from clouds, up from searchlights, or, in today's world, from lasers. He is less remembered for his counter-intuitive theory that red sunsets are caused by the same scattering of light which creates blue skies. Tyndall hypothesized that the same dust particles in the air that make light beams visible must scatter more blue light than red light at the higher-energy end of the spectrum. It made perfect sense: the scattered light of the sky and shadows is blue, while the transmitted light of the sun that has passed through lots of dust particles is red. Case closed.

Enter John William Strutt, later Lord Rayleigh, Nobel laureate for studies of atmospheric gases. After doing the math on light scattering by particles, he concluded that most dust lacked the precise size relationship to wave length to account for the profound blue sky effect. Tiny air molecules are the cause of what is now known as "Rayleigh's scattering." Blue light with a wave length around 450 nanometers is 3.2 times more likely to be scattered by air molecules than red light of 600 nanometers. When dramatic color sorting occurs as sunlight travels through the thickest air near the horizon, all white light vanishes and magic hour arrives.

Because our visual system has the greatest response to yellows and reds, we tend to pay the most attention to them until a certain moment in twilight when their intensity has fallen well below that of the blues. Suddenly, we're aware of a magical mixture of warm and cool tones, but it's too late for prime photographs, which should have been made before the saturated pinks and reds faded.

Whenever you're responding to warm light coming over the horizon, you're also unconsciously taking in the blues and violets in the majority of the scene. A photograph showing both warm and cool tones will appear more realistic and dramatic, but only if a graduated neutral-density filter is used to hold back the brighter warm areas of the sky so that it's possible to see detail in the blue shadows. The accompanying image of a Himalayan blue pine silhouetted at magic hour in the Khumbu Valley of Nepal was made with a soft-edged, two-stop SinghRay grad filter.

Now let's return to that backyard light bulb and consider what happens if we try to take a photograph of people sitting beneath it on daylight film. An amber wash that we didn't see permeates the entire scene. This happens because film has a fixed color response to the wave length of light, but our eyes do not. Even though the bulb seems white when you're close to it, it is giving off the color temperature of natural light about 20 minutes before sunset. So if those warm colors that you can't see will appear on your film under artificial light, they'll also appear without being seen 20 minutes before sunset in natural light, adding a cast that is far more pleasing on landscapes than faces.

We begin to see an overall orange or red cast only during the peak minutes of magic hour, as the wave length of light goes out of gamut for our biological white-balance system. Of course we can also see yellows and reds outdoors at noon, but only from surfaces that reflect the proper wave length, not from the natural light itself, which appears boringly white.

Next time you're out during magic hour, don't let yourself go buggy over warm light. Make your eyes adjust to the blue end of the spectrum by looking into the shadows. Then when you glance toward the warm tones, they'll appear as rich as on Velvia or E100VS for half a minute or so, until your visual system adjusts. When you look at the blues again, they'll also have the heightened appearance of film for a bit. Taking control over your visual experience not only helps you previsualize magic-hour photography, but also gives you far richer memories, and that's really what life's all about.

Notes

LOCATION SOUND - I

Let’s Discuss Sound for Picture

The concepts of location sound recording that we will discuss in this lesson are basically the same, whether you are shooting your tenth independent film or your first institute project with your first camcorder. Audio seems to be one of the most challenging areas for beginners and even experienced filmmakers alike. Video professionals typically find sound one of the most challenging aspects of production. Ten years ago, producing professional quality film or video was a much more cut and dried process. If you wanted decent sound for your picture, you either had the knowledge and equipment to record it yourself or you hired a location sound team. This is still true today but the differences are that there are a lot more people producing video today who may not have experience and skill in recording location sound than there were ten years ago.

DV users with many different experience levels and widely diverse backgrounds are producing their own projects. The fact is that almost all of the tools needed to produce broadcast quality video and DV “films” have become relatively inexpensive and widely accessible. Final Cut Pro and AVID Express DV both have about 90% of the capability of a AVID Media Composer system at a minute fraction of the cost. Camcorders like the Sony PD-150, Panasonic AG-DVX100 and the Canon XL-1S are capable of producing extremely high quality images.

PD-150

AG-DVX100

XL-1

The immense popularity of digital video means that a large majority of users today have access to the most advanced communications medium society has ever seen. We have small, relatively affordable, high quality camcorders that can make amazing images with far less light than ever before. We have very sophisticated and capable video editing tools available on both platforms. Assuming we all want to produce work of quality, what’s missing from this equation? You guessed it, the sound. The fact is that most independent, low/no budget projects that are produced today seem doomed to suffer with sound that ranges from merely average to barely usable. Whether you are producing video for your friends and family, to view, events, corporate audiences, or for broadcast or theatrical release, no matter which category your audience falls into, they expect “transparent” sound from your project’s soundtrack. Let’s define what “transparent” sound is.

Audio conveys almost all of the emotional impact in the visual medium. It’s a fact. If you watch your favorite scene from any film or TV show with the sound off, you soon discover that moving images on their own are typically not very emotionally involving. Don’t forget, silent films could be scary, sad, happy, dramatic or interesting

BECAUSE they were conceived without sound. To be totally fair, most silent films were viewed with either live or pre-recorded music. Obviously, most of us want to produce projects that will emotionally involve our audience. For most of us, video has become the single most common collective vocabulary in our lives. It is also a given that video is great for communicating almost any message to almost any audience, if done well.

The Experience Economy

What may be less obvious to you if you are new to film and video making, is that audiences of all kinds now expect to be entertained while you are conveying your message. If you are in the entertainment end of this business, this is understood, but also for those of you who want to create events or video for business, your content must also be entertaining and compelling. Emotional involvement from your audience is what defines good entertainment. You may not feel that the Shop Safety training video you are producing can or should be very entertaining, but if the production values and concept are not very high quality, your training video will bore your audience. If it’s done well, even a Shop Safety training video can be entertaining. Your sound is largely what will determine if your project is entertaining to your audience. Unless you want to conceive your project as a “silent film”, you have to be concerned (‘obsessed’ might be a better term) with your project’s sound.

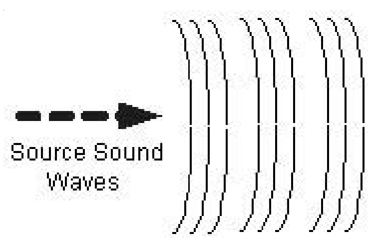
One of the toughest concepts for many newer DV users to grasp is that the better job you do with your project’s sound, the less it will be noticed. I feel that this concept is one of the reasons why most projects don’t end up with very high quality soundtracks. We are very used to spending time, effort and money on a better camera, lens, bigger and better lighting, crew and visual effects and seeing an immediate “payoff” when our images are viewed. It’s instantly recognizable if a scene is lit effectively or if a visual effect is done well. We feel “justified” in shooting on a higher quality, more expensive format or with a bigger crew because the end result is usually easily identifiable on-screen. Most of us can immediately recognize if a project was shot on 35mm film versus DV or if a project's motion graphics or visual effects were well executed. If we notice a sound mix though, it is usually because the sound was done incompetently. This is the central concept of "transparent" sound. If your location sound is recorded correctly, the easier it will be to work with the basic audio during the post-production process. The better job you do with the sound during video and audio editing, the less the audience will notice it. The only sound that is noticed in a visual medium is usually poorly executed. Great sound works on a subconscious level with the viewer by drawing them into what they are viewing. Great sound supports and enhances the stories you are trying to tell. Now that we have a basic understanding of the goal for your project's soundtrack, let's review what we have covered before we dive headlong into equipment and technique.

Four Points to Remember about Sound for Picture

- The principles of location sound are the same for almost everyone shooting anything.
- No matter who the audience is, at the very least, they expect "transparent" sound
- Sound conveys emotion - picture conveys information
- The better your soundtrack, the less it is consciously noticed

It's all Just a Chain

The easiest way to discuss location sound is to think of the entire audio path as a chain. In the case of location sound, the "links" are:



- The sound itself
- The microphone(s) that capture the sound

- The cables and connectors that carry the signal from the microphone to the mixing or routing device and from the

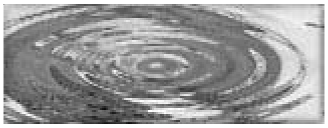
mixing or routing device to the recording device

- The mixing or routing device that carries the signal from the microphone to the recording device
- The recording device itself (typically a camcorder but could also be a VTR, Hard Disc Recorder, MD or DAT recorder)
- The monitoring circuit of the recording device

Just as in an actual chain, the audio path is only as strong as it's weakest link. This means that a high-quality, accurate recording device paired with a low-quality microphone will not be able to record anything better than what the microphone is capable of picking up. It means that a great microphone and audio mixer paired with a substandard recording device will only be able to record to the limitations of the device's recording circuit. While it is not practical for most DV users to acquire and use the best quality location sound equipment made, it should be the goal of every user to acquire and use components that match each other in features and quality. Don't hire the best mixer you can afford and then skimp on cables and connectors. Don't buy the best shotgun microphone on the market and then skip using a mixer because you spent your entire budget on the microphone. You get the idea.

The First Link - Sound itself

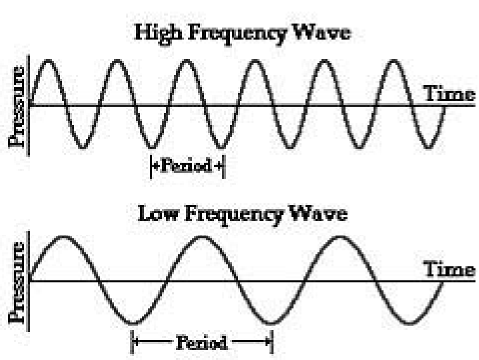
You have already learnt about the basics principles of sound in the previous semester let's revise some basic concepts of what sound is and why sound behaves the way it does. At the most basic level, sound can be described as waves moving through air. The farther apart these sound waves are, the lower the frequencies. The closer the sound waves are to each other, the



higher the frequency. In the most basic terms, sound is "messy". It bounces, reflects and behaves in ways that seem mysterious to most us. It cannot be seen and all of us perceive sound differently. There is an entire branch of study and academia called "Psychoacoustics" which is the study of how

humans perceive, process and react to sound. Sound is definitely a case of "perception being reality."

Sound waves cannot be seen in most cases, but effects of sound waves are evident if you know where to look. Although not actual sound waves, the ripples produced when a rock is dropped into water produce a nice visual approximation of what sound waves would look like if they were visible to us.



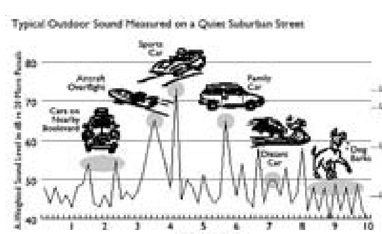
If you place something lightweight, like a piece of paper in front of a typical transducer, like a two way audio speaker (a two way speaker has only a woofer for generating low frequency sounds and a tweeter for reproducing high frequency sounds), you will probably see the paper physically move if placed in front of the woofer while the speaker outputs sound at a decent volume level. However, if you place the paper in front of the tweeter only, you will probably see either very little or no perceptible movement. This is because the high frequencies generated by the tweeter are much closer together and occur at much more rapid intervals. Too rapidly to perceptibly affect the mass of even something as lightweight as the paper unless the amplitude of the sound is increased to very high levels. Understanding this concept is central to understanding how sound waves behave.

Low frequency sound waves (bass sounds), because of their larger physical size, tend to interact with their surrounding environment in much more perceptible ways than high frequency sound waves (treble sounds) seem to. All sound waves reflect off of physical objects but because of their larger size, lower frequency sound waves reflect and penetrate objects more than higher frequency sound waves do. If you can grasp this idea, you will begin to have an understanding of how to help modify or eliminate undesirable sounds when you shoot in any location. If your apartment or hotel neighbor is cranking his stereo loudly in the room next door, which frequencies are you hearing through the walls? Obviously, the higher frequency sound waves do not have the strength to penetrate the wall. Few of the mid range sound waves will penetrate the wall. What you will mostly hear through the wall are mostly the low frequency sound waves.

"Preventative Maintenance"

What we deem as "poor quality" sound can be manipulated and adjusted all along the audio path in different ways but in most cases, it is best to adjust and compensate for the "poor quality" sound before it ever enters the audio path. Although this sounds like a simple concept, the reality is that it is much more common for people to hear what will obviously be a sound problem (undesired noise, excess ambient sound, technical sound problems like hum or buzzes, etc.) and to just go ahead and shoot, rather than determining the source of the undesirable sound and fixing it. In some cases, it's just sheer laziness. It's a pain to hunt down an audio problem and fix it before rolling. When shooting with a crew or in limited access or time-constrained situations, this is somewhat understandable but you should know that despite all of the great postproduction technology available, basically, what you hear is what you get. If you record poor quality sound, it will always be poor quality, no matter what types of filters, plug-ins or processes you run it through. It pays to spend a few extra minutes to fix audio problems before you shoot.

Below are the top five most common causes of location sound audio problems that most people will run into when shooting. You will notice that several of the categories kind of overlap each other in definition. Such is the nature of sound. Included



are some suggestions for mitigating, reducing or correcting the problems before you shoot:

Excessive Ambient Noise.

Too much background noise. Too much traffic noise. People speaking or making noise in the background or in the room next door. Dogs barking in the background. Sirens in the background. Aircraft flying overhead.

Possible Solutions.

Excessive ambient noise is one of the toughest issues to work with because often, the excess ambient sound is beyond your control. The solution can be as simple as repositioning your microphones or waiting until the excess ambient sound dies down or goes away to more elaborate steps such as sound-proofing a location. Obviously shutting doors and windows can help when shooting interiors. Exteriors are tougher. You must be creative in your thinking and do whatever is within your power to limit ambient noise when shooting. Microphone choice is extremely important here as well.

Building Noise.

HVAC (heating, vacuum and air conditioning) noise. Building creaking or settling. Elevator noise. Clocks ticking. Doors opening and slamming. Noisy lighting fixtures.

Possible Solutions.

HVAC can be tricky to deal with. If you have done your homework, when scouting a location to shoot, you should always determine where the access is for the HVAC controls for the areas you'll be shooting in. Unfortunately, many newer buildings have "zone climate" HVAC systems where there is one control for an entire area or even a floor of a building. So if you turn off the HVAC system to shoot, you may be fine in the room or area you are shooting in but people in other areas may be dying of heat or freezing to death. If you cannot obtain access to HVAC controls or personnel to control the HVAC system, you can also reduce the "whoosh" of vents by bringing sound blankets or foam and temporarily blocking a single HVAC vent. Blocking one or two vents at a time is rarely an issue for an HVAC system. We typically chill down a room to just above "teeth chattering" point before beginning an interview, and then turn the air conditioning off to get rid of the low HVAC system rumble. Video lights and shutting all of the doors and windows will typically heat the room right back

up fairly rapidly. The added benefit is that shooting in a cool room will also keep everyone alert and awake.

As far as creaking and settling of a building, there is not much you can do other than listen carefully as you shoot and shoot alternate takes. You can do the same with elevator noise. Carefully worded signs (quiet please - video crew shooting) on the doors can be of help or stationing a crew person at the offending adjacent door to act as a "doorman", carefully and quietly opening and closing the door can help.
Typically, the noisiest lighting fixtures are fluorescent light banks. As the ballasts age, they typically become more and more noisy. The obvious solution, if you are lighting the scene is to just turn the offending lights off. Many newer buildings have motion sensor light switches though and taping black gaffer tape over the light's sensor can take forever for the light to turn off. Always bring a stepladder and if necessary, you can just remove one tube and the bank or at least part of it will go out. As in HVAC systems, in many newer buildings, you may encounter a single set of lighting control switches that control all lighting in a large area or even an entire floor of a building. Learn how to quickly and safely remove fluorescent tubes from fixtures. Many times, it's the quickest, easiest option.

Machinery.

Fan/hard drive noise from computers. Carrier tone from computer and video monitors. Refrigerator/freezer noise.

Possible Solutions.

Beware of computers. The obvious solution is to turn the CPU off, if possible. When it is not possible, another solution is to isolate the CPU within it's own nest of sound blankets (a.k.a. known as furniture pads). Make sure to leave room around the CPU so that its fan system still has access to adequate airflow. We typically use two or three C stands with sound blankets grip clipped to the extension arms.

Beware of the "carrier" tone, the high-pitched whine that all CRT's emit when turned on. It varies from monitor to monitor

but if it is loud enough and at the correct frequency, it can be recorded and can be difficult to EQ out.

Refrigerators are commonly encountered in many locations. They can be tricky to hear because as the unit's compressor cycles on and off, you may hear the sound and not know what it is, then as you look around, the unit's compressor, may cycle off. It's not usually a good idea to unplug a refrigerator, unless you obtain permission from its owner. We usually will close doors and or isolate the refrigerator using sound blankets ands C stands. If you do unplug a refrigerator, it's a good idea to place your car keys inside so that it's impossible to leave without remembering to plug it back in.

Talent and Crew Noise.

Talent/crew fidgeting or moving. Creaking floorboards. Squeaks from shoes. Clothing rustling/rubbing. Microphone (usually lavalieres) rubbing on skin or clothing. Stomach rumbles. Heavy breathing. Mouth sounds (clicking or sticky mouth)

Possible Solutions.

Make sure that your talent and crew holds still and breathes softly during takes. This can be tough during long takes and having an assistant director on the crew can really help enforce this. Most crews are fairly savvy about this but at times, observers, crowds and even clients can cause a lot of off-set noise that can be a nightmare to deal with in post so deal with it before you roll.

Creaking floorboards can be challenging. If the talent's feet are not seen in shot, carpet or "dance floor", a portable type of flooring system can be laid down for crew and talent. Squeaking shoes can be remedied by removing them if they are out of frame, using baby powder or even a lubricant like WD-40 or silicon spray although these can stain shoes. Shoes squeaking on floors can be tougher to deal with than squeaking shoes themselves. A bit of sawdust or even once again, removing the offending shoes can help although it's a little strange to see a crew and talent doing scenes in their socks. Although it's probably not the safest way to shoot a scene. Grip equipment can be heavy and dangerous.

Clothing rustling and rubbing are also one of the most common challenges. A lot of this can be mitigated if the correct types of clothing are used. The "sound mixer's nightmares" are silk, rayon and corduroy. Trying to mic a female talent wearing a silk blouse will test the patience of even the most experienced sound person. The other half of this equation is just learning basic mic techniques. There are many different "tricks of the trade" when it comes to learning how to put a mic on talent and no having it rub and pickup extraneous noise.

Mouth sounds are more of a problem when doing voiceovers but can be a problem on set as well. Water helps. Always make sure that all talent has easy access to water as they shoot scenes. Depending on the person, water alone may not remedy "sticky" mouth. One of the best solutions for sticky mouth is, believe it or not, green (Granny Smith) apple slices. The acidity and sugar content balance of this particular variety tends to stabilize the saliva and make it so that mouth clicking and smacking is reduced or eliminated.

Sound/Video Gear.

Ground loop hum/buzz. Loose or defective connector/cable crackle. "Hits" when using wireless microphones. Camera motor noise. Video monitor carrier tone. Substandard microphone mounting/handling. Wind buffeting.

Possible Solutions.

Ground loops are perhaps the most common equipment problem as far as location sound. Many of these issues occur because professional balanced gear is interfaced with consumer-unbalanced gear. Using the proper gear will remedy most ground loop issues. In general, using Radio Shack style adaptor plugs and jury-rigged cable adaptor plug combinations are a recipe for hum or buzzes. Always use professional balanced XLR gear as much as possible. More on this in the equipment segments later in this article.

Cables are easy. If you experience bad or loose connectors or cables, mark them with a piece of tape and either repair them, if you are talented with soldering cables and connections or replace them.

Wireless microphones are susceptible to noise. From the very cheapest consumer models to the top of the line pro units, all wireless systems can experience interference. The more expensive the unit, generally, the less susceptible to extraneous noise, plus the higher the sound quality. Only use wireless when you must. Do not shoot sit down interviews with wireless systems because you are too lazy to run a cable on the floor. With the advent of new high-end digital wireless systems, noise is becoming less of an issue than with analog units. Eventually, all wireless systems will probably be digital and we'll be free to use them anywhere we want for the most part.

Camera motor noise is almost always a by-product of using an on-camera microphone. It's really simple to fix this one. Don't use an on camera microphone unless it's ONLY for ambient sound. NEVER try to record talent with an on-camera microphone; it's the perfect recipe for bad sound. The on-camera microphone will pickup very little of the talent's sound and lots of background ambient sound. If you must use an on-camera mic, buy a high quality microphone mounting system and you should eliminate most camera motor noise.

Video monitor carrier tone can be picked up if it occurs near the microphone. Move the monitor or turn it off.

Microphone mounting or handling. Booming technique is an acquired skill. You don't just buy a boom and start using it successfully. It takes training and practice to become a good boom operator. As far as the microphone mount itself, beware of using cheap, rubber band mounted microphone mounts. These are simple metallic ring systems where the microphone is suspended in the ring using rubber bands. These types of mounts are okay for stationary use but for hand booming, many of these mounts are inadequate. A high quality micro-

phone mount can take a lot of handling and not transmit it to the mic element.

Wind buffeting can be very tough to prevent when shooting outdoors. The easiest solution is to use a quality microphone mounting system, zeppelin and windsock.

Four Points to Remember about the "First Link" in the Audio Chain and "Preventative Maintenance"

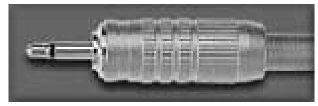
- The audio chain is only as strong as it's weakest link
- Low frequency sounds, because of their larger physical size, tend to interact with their surrounding environment in more perceptible ways than high frequency sounds seem to.
- Fix as many sound "issues" as possible BEFORE shooting
- If you record poor quality sound, it will ALWAYS be poor quality

The Second Link - Microphones

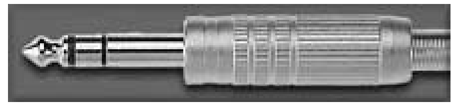
In our discussion of sound, we talked about some basic concepts of what sound is (and sound waves) and a few basic characteristics of how sound waves behave, (they are 'messy', the differences between how low frequencies behave versus high frequencies).

A little revision of the types of microphones. Microphones are, at their most basic, transducers that convert input energy of one form into output energy of another. Microphones are broken down into two separate categories; Dynamic and Condenser. Most Dynamic microphones are the moving coil type. Moving coil microphones use a magnet, a coil wrapped with wire and a diaphragm that sits over the top of both. Sound pressure hits the diaphragm and moves the coil across the magnet. This creates the voltage that travels out and along the mic cable on the way to the mic preamp. With Condenser microphones, Phantom power from a battery inside the mic, or from mic cable from the mixer or recording device, or a separate power device is used to power the microphone. The phantom power charges a capacitor. This capacitor holds a charge in the microphone's fixed backplate. In front of the backplate, a thin diaphragm is located. When the diaphragm moves in relation to the fixed backplate, a charge is created in proportion to how much movement the diaphragm makes. Unlike the signal created by the dynamic mic, a condenser's signal is very weak and must be amplified before it gets to the mixer or recording device. Condenser microphones contain a small amplifier that boosts the signal before it leaves the mic

It may help you to visualize what is happening inside the microphone by imagining how a speaker works. A microphone works the exact same way that a speaker works, only in reverse.



When a speaker receives an electronic signal, it moves its



transducer in response to the amplitude and modulation of the signal that it is receiving from the amplifier. When a microphone element is moved by the sound waves, it generates the same kind of electronic signal. The main difference is that the microphone SENDS the signal out to a monitoring or recording device, whereas the speaker RECEIVES a signal from an amplifier.

Balanced Versus Unbalanced

What do the terms "balanced" and "unbalanced" mean? Without going into a lengthy explanation of impedance and electronic signal measurement, balanced line-level equipment operates at a much higher nominal level (+4dB). Besides a positive and negative lead, it also uses a third grounded lead (or 'earthed' lead as we say in India), and usually, but not always, uses three-pin, locking XLR connections. Unbalanced line-level equipment operates at a lower nominal level (-10dB) and has only a positive and a ground lead (no negative) and mostly uses non-locking RCA connections, although unbalanced connections are also occasionally 3.5 mm mini plugs, 1/4" (tip-ring TR) plugs and spade lugs.

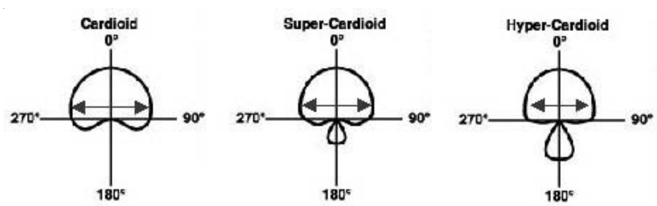
Balanced XLR
Unbalanced 3.5 mm mini-plug
Balanced 1/4" or unbalanced stereo1/4"
Unbalanced RCA

Professional audio and video equipment predominantly uses balanced cables and connections. Consumer equipment almost exclusively uses unbalanced connections. Why should you care? Improperly wiring to unbalanced equipment is probably the single most common cause for ground loop problems, interference, hum and buzz. I suggest you use balanced professional equipment whenever and wherever in your audio chain that you can. Balanced equipment is much less susceptible to interference, but can still pick up ground loop hums especially if there are grounding problems in the electrical system the equipment is plugged into. However, you will still have fewer problems with interference in your audio if you use balanced equipment. Almost all professional microphones and mixers operate using balanced connections for this reason.

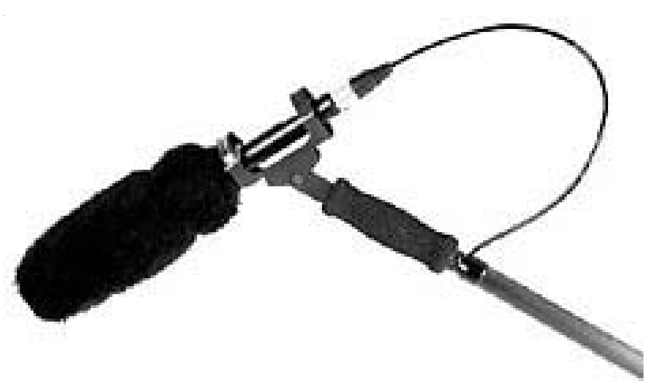
Microphone Types and Purposes

Although there are probably more than a dozen different microphone types, it probably makes sense to discuss the most common types of microphones used in film and video production. Just as a golfer uses a specific type of club to "tame" specific areas of the golf course, a discerning location sound mixer might use several types of microphones during the typically varied shooting situations encountered in a typical project.

You have learnt about the different kind of microphones and their functioning in the previous semester. In this lesson we see the makes and uses of these different microphones in the



industry. Let's discuss the various microphone types that are used in 90% of shooting situations, Shotgun/hyper cardioid, cardioid microphones and lavaliere microphones. Besides physical construction and size, the main factors that categorize microphones are their pickup patterns and characteristics. Do you recall our discussion of what sound waves look like and how they behave? Certain pickup patterns are better than others for picking up different subjects and specific portions of the audio frequency spectrum. Certain pickup patterns have specific desired characteristics. Add up these factors and you will see that there is not always a "right" or "perfect" microphone for any given situation. You could give three different cinematographers the same lighting and grip package and each of them will use a different combination of lights and support gear to realize each of their individual interpretations of how the scene should look and feel. You could give three different location sound mixers a sound kit and each of them might possibly use a different combination of microphones and techniques to shoot a scene. It should be apparent to you by now that there is no such thing as a "general use" microphone. This should also



prepare you for the fact that you will need to eventually rent multiple microphones.

Shotgun/Hyper Cardioid & Cardioid Microphones

These are pickup patterns for three types of cardioid microphones.Notice how even the narrowest pickup pattern (hyper cardioid) still picks up sounds from the rear axis. Some of the terminology in classifying microphones can become confusing. The term "shotgun" describes a type of microphone that is a long narrow tube, not unlike the barrel of a shotgun, hence the term. You will hear terms described below like "long shotgun" and "short shotgun". In simple terms, a longer shotgun usually has a narrower angle of acceptance of sound and rejects more sound from the sides, also referred to as "off-axis". Shorter shotguns will usually pickup more sound from the sides and will not isolate a single element as much as a longer shotgun will. The term "shotgun" is sort of slang for any long, narrow tubed microphone. The official terms are variants of the term "cardioid" If you refer to the above illustration, you can see the differences between cardioids, hyper cardioids and super cardioids. These types of microphones are used in about 90% of sound for picture recording. What this means for us is that these kinds of microphones can be aimed at talent as they record dialogue and somewhat isolate the talents sound from most of the extraneous background sound. A shotgun or cardioid mic is almost exclusively operated mounted to a microphone boom so that it can be usually suspended about two to three feet above the talent, depending on framing.

A common misconception is that a shotgun microphone has a longer "reach" than other microphones, as if it can "magically" record sounds that originate a considerable distance from the microphone element. This is the wrong way to think about how microphones and sound behave. Longer microphones don't have any "reach"; they have a narrower angle of acceptance than other microphones. Even the longest shotgun microphone will pick up lots of extraneous ambient sound in the right situation. Remember that sound is "messy"? This means that the undesirable sound is reflecting and bouncing into the area where you are trying to record clean sound mixed with just a slight amount of ambient sound. In order to understand the reason that overhead booming is used whenever possible, think about what lies beyond the talent. If the mic is pointed at the talent at mouth level from near the camera, not only will the microphone pickup the talent, the microphone will also pickup whatever is behind the talent. By pointing the microphone element down toward the talent from above, it picks up the voices of the talent and mainly, the floor or ground as the case may be, rather than all of the sound of the activity behind the talent. Shotgun microphones are more directional than other types of microphones. Hyper cardioid and cardioid microphones have a slightly wider angle of acceptance than shotguns but narrower than most other microphone types. Some seem to prefer the more open sound of a cardioid or hyper cardioid instead a shotgun, but it can be difficult to get enough isolation on talent when using a cardioid, especially when shooting exteriors. In my experience, the single most typical problem

encountered when shooting on location is excessive ambient

sound creeping into the recording.

Above all other factors, the distance between the mic and the sound source will have the largest influence on the overall sound in your recordings. Regardless of pickup pattern, cost or quality of your mic, if it is placed too far away or aimed improperly, then your sound quality will be diminished.

Which Shotgun/Cardioid Microphone should i buy?

A low cost shotgun microphone cost around Rs 10,000 to 15,000. Probably the most common shotgun microphones

used in sound for picture production are the Sennheiser MKH series. The Sennheiser MKH-416 is an industry standard short shotgun and for good reason. It is very flexible, reliable, simple and efficient. This MKH-416 was slated to be replaced totally by the newer MKH-60, years ago but the continuing demand for the MKH-416 has been such that Sennhesier has continued manufacturing and selling both models for years. The MKH-60 is a more modern, sophisticated take on the MKH-416 and both are excellent short shotgun microphones. Two other favorites are the Neumann KMR-81i and the Sanken CS-3. The Neumann line of microphones are world renowned for their superb German construction quality and sound and KMR-81i is a beautiful sounding example. The Sanken CS-3e was developed in conjunction with NHK, Japan's largest broad-caster and has fast become very common on film and television sets all over the world. It is an excellent sounding microphone.

Sennheiser MKH-416

MKH-60

Neumann KMR-81i

Sanken CS-3

As far as long shotguns, The Sennheiser MKH-70 is a great workhorse long shotgun that is very useful for exterior interviews and dialogue as it's narrow angle of acceptance reduces exterior ambient to a low level. It is also quite rugged and reliable. The Neumann KMR-82i is basically a longer version of the KMR-81i and also features that smooth sound quality that all Neumann microphones are known for.

Sennheiser MKH-70

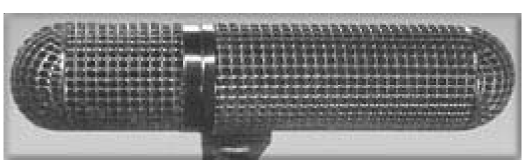
Neumann KMR-82i

There are also numerous other shotgun, hyper cardioid and cardioid microphones available from Schoeps, AKG, AudioTechnica, Beyer Dynamic, ElectroVoice and others that are great microphones. As we discussed earlier, plan on using, renting or owning several different microphones. There are no absolutes when it comes to microphones; it's just that certain brands may offer a sound quality that is more appealing or more effective in various situations. Certain brands and models may be more rugged or have a more delicate sound but also a delicate construction. Each is a unique tool that is better suited to certain situations, budgets and uses. The bottom line here is to rent, borrow, demo and listen before you buy. Certain microphones will be better for your situation.

What else do i need to use Shotguns/ Hyper Cardioids and Cardioids?

If you are serious about recording high quality sound, you soon realize how futile mounting a microphone on a camera is. The time will come when you will finally decide to get serious and go with a microphone mount, zeppelin, windsock and

boom pole. You should remember that all of these tools that are necessary to work in conjunction with shotguns, hyper cardioids and cardioids, in total, may cost as much or more than the microphone itself. Allocating your budget intelligently here comes into play once again. It would make much less sense to buy a top of the line Neumann shotgun, then skimp by purchasing a rubber band mount only and not purchasing a zeppelin, windsock and real boom pole. Buy a more reasonably priced microphone and all of the support gear needed to use it correctly. And if you are hiring your sound equipment make



sure of what all the studio has and what all they'll give you with the equipment.

Microphone Mounts

As we discussed earlier in this lesson, there are many different models of "rubber band" mounts. Even if you are on a tight budget, this is not a place to try to skimp. If all of your sync sound shooting is stationary interviews, you may be able to get away with using a rubber band mount, but if you are planning on doing any "walk & talks" you must go for a more sophisticated mounting system. Both Lightwave Systems and Rycote make excellent microphone mounting systems. An extremely effective, very versatile and well engineered mount is the Universal Mini Mount. The beauty of the Universal is that with the proper adaptors, the same mount can be used on a boom pole, on a camera or mounted on a handgrip for sound gathering.

"rubber band"
Universal Mini Mount Lightwave Systems

Zeppelins and Windsocks

A Zeppelin is a protective enclosure that totally encapsulates a microphone. The function of the zeppelin is to not only protect the somewhat delicate microphone element but also to filter out extraneous wind and HVAC air movement. The ideal situation is to have separate sizes of Zeppelin for both long shotguns and shorter shotguns and cardioids although in practice, you can be satisfied in using shorter shotguns in our longer size Zeppelin.

Zeppelin

Windsocks (dead kitties or road kill due to their usual gray or black fur appearance) are synthetic fur-covered sleeves that are designed to slip on over a Zeppelin. Most offer Velcro tabs and

or zippers to snugly fit a Zeppelin. Windsocks offer a much higher degree of wind resistance than just using a Zeppelin alone. Some of the manufacturers even offer two different length of fur on their windsocks, a shorter hair for moderate wind and a longer hair for strong wind. Besides diminishing wind noise and buffeting, using a furry wind sock will cut out some of your high frequency response so you should not use one all of the time, you need to listen to what the microphone is picking up and choose to use or not use a windsock in addition to the Zeppelin accordingly. There are also socks that

are "velvety" instead of furry and work very well in winds up to 15 mph and give minimal high-frequency attenuation.

Windsock

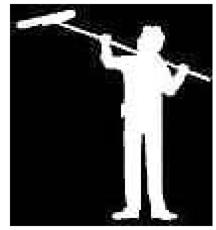
Another alternative to a fully encapsulated Zeppelin are the slip-on windscreen systems. Lightwave refers to their system as "The Equalizer" and Rycote refers to their system as a "Softie Windjammer". The advantage to these systems over using a full Zeppelin is that they are both less expensive and are smaller and quicker to use. This can be handy for more "run & gun" ENG (electronic news gathering) style shooting. The disadvantage of the slip on windscreens is that they do not provide the same amount of isolation and wind buffeting protection since by design; they only cover the front portion of the microphone element. When shooting outdoors though, you will obtain better sound using a full Zeppelin and windsock. You should also know that you might sometimes need to use windsocks indoors, particularly when you cannot turn off HVAC systems or in when shooting large auditoriums.

"The Equalizer"

"Softie Windjammer"

Boom Poles

Now that you have at least one good shotgun, hyper cardioid or cardioid microphone and a microphone mount, zeppelin, windsock or a slip on windscreen, how do you use it? The first step is to obtain a boom pole. Boom poles are available in a variety of lengths, styles and materials with the most popular poles being made of aluminum and carbon fiber. Generally, it would be better to own two lengths of boom pole, a short one



for travel and small setups and a longer one for larger wide shots.

Aluminum

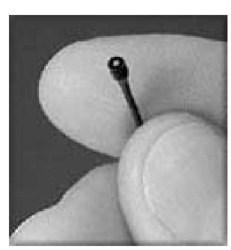
Carbon fiber

You will notice that as you begin to look at boom poles, there are a lot of different models on the market. The other decision for you to make besides material construction and length is whether or not you want to go for an internally cabled model or not. Internal cabling is much handier and a LOT more expensive. The K-Tek version of Gitzo with a coiled internal cable system runs almost three times as expensive in price than as Gitzo. When you are doing a lot of different setups, hand booming a lot or when you travel, a cabled pole is more

convenient. You just have to decide if the extra money is worth it to you.

Booming

The rule of thumb when booming is to get as close as possible to the subject, which is just outside the camera frame line. The further the mic is away from subject, the greater the background noise and echo, so every inch closer improves sound quality. The mic is normally positioned several inches to a foot over the actors head. Up to two feet may be acceptable depending on the situation. The camera operator will help you determine how close you can position the mic without passing the frame line.



Booming can be done from overhead or underneath the subject:

Booming from Overhead - In overhead booming the mic is suspended above and slightly ahead of the subject. Overhead booming is the most favored technique since, in most situations, it provides the most natural sounding dialogue with the least amount of mixing.

Overhead booming allows multiple actors to be recorded with a single mic, yielding a pleasant blend. Sufficient sound effects are picked up to give the soundtrack a full texture. Perspective is easier to maintain, too. Since faces are closes to the mic, dialogue dominate sound effects. On wider shots, the mic tends to be higher resulting in thinner, more distant sounding dialogue. On closer shots, the mic is closer, resulting in greater presence.

With all of these benefits, it easy to see why overhead booming is the first preference in the hierarchy of microphone placement. Because of physical obstructions, this technique is not always possible to apply. The next favored option is booming from underneath.

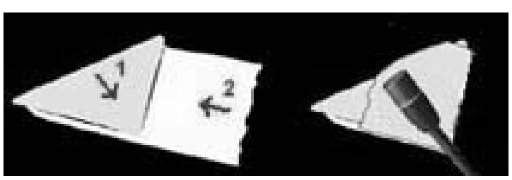
Booming from Underneath - When booming from underneath, the mic is suspended below and slightly ahead of the subject.

Booming from underneath has several drawbacks when compared to overhead booming. First, dialogue tends to be bassy because the mic picks up more sound emanating from the chest cavity. Second, the mic is closer to the actor's hands so sound effects may appear louder than dialogue, throwing perspective off. Third, booming from underneath can be tricky on sets tight with furniture and actors.

Despite these drawbacks, booming from underneath can be an alternative to overhead booming and provide perfectly acceptable sound under the right conditions.

Lavaliere Microphones

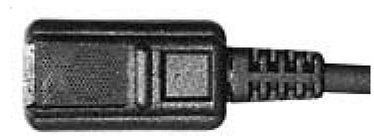
A lavaliere is defined as "a pendant worn on a chain around the neck" which should give you a good idea about what a lavaliere microphone ('lav' for short) is. In the early days of broadcasting, the smaller microphones were actually worn around the



neck in the same manner as a necklace. You may not realize that today's lavaliere microphone is an incredible feat of engineering. The most popular lavaliere microphones today are incredibly small. Smaller than a match head. So small that they can easily be hidden in the talent's hair of even behind a tiny button on a shirt. Lavaliere microphones come in usually one of two flavors, omni-directional and uni-directional, although unidirectional lavalieres are rare and are limited mostly to newscaster style applications. An omni directional unit has a pickup pattern that picks up fairly well from all sides. A unidirectional lavaliere has a pickup pattern much more like that of a shotgun or hyper cardioid microphone; it must be "aimed" at the talent's mouth. Because a unidirectional microphone must be aimed at the talent's mouth, the opportunities for hiding the microphone element are mostly eliminated, limiting the use to mostly planting the microphone in the center of the talent's chest.

Lavs are given less preference than booming because they are prone to problems with perspective and contact noise:

Perspective - Since lavs are attached directly to actors' bodies, their voices dominate the soundtrack. Consequently, actors always sounding close to the camera, even in long shots. In



addition, the track is devoid of sound effects and ambience. Overall, lavs result in sterile, less natural audio.

There are two ways to improve lav perspective. First, the lav can be moved further down the actor's chest. This opens up the air space and allows the mic to pickup more ambience. Second, a supplemental boom mic can be used to capture ambience and sound effects that the lav might ignore, such as footsteps.

Contact Noise - When rigging lavs on actors, contact noise occurs when clothing rubs over the mic head or cable. To avoid contact noise, clothing must be secured on all sides of the mic head. This best way to do this is to sandwich the lav head

between two triangular wads of cloth tape, such as camera or gaffer tape. The wads are created by folding them like a flag, sticky side out.

Making a "Sticking Wad"

In addition, tape or pin down any loose flaps and soften stiff areas of clothing with some water. Another way to avoid clothing noise is to attach lavs to non-traditional areas, such as hat brims and props.

Noise from the lav wire can be prevented by taping a few lengths of cable to the clothing. Double sided tape or sticky wads prevent the clothing from rubbing on the actor's body. A loop should be formed near the mic to provide strain relief.

Besides being used on talent, certain lavaliere microphones are also handy for using as a "plant" microphone. A "plant" microphone is typically placed in a hidden spot on the set to pickup talent as a supplement to or instead of a boom microphone. So if you see a "roundtable" or "dinner table" setup, the sound mixer may be covering the scene with one or more "plant" microphones hidden in the centerpiece on the table. Plant microphones can also come into play when scenes are shot with a Steadicam or large dolly moves where it may be impractical to have the boom operator try to follow the camera.

Another instance could be that wardrobe restrictions make using body mounted lavalieres tough or impractical.

These are a Few Lavs

The Tram TR-50B has been a workhorse industry standard for many years. It was the first lavaliere microphone to use a "vampire" clip mount (a small plastic clip that holds the microphone and has two small 'needle-like' barbs) and it sounds very neutral. The Tram is available in several different configurations and a frequency-modified version is sold as a Sonotrim. Another similar mic with a slightly different sound but a similar look and construction is the PSC MilliMic. All three of these microphones share the same basic rectangular form factor and all Sonotrim and PSC seem to be variants of the basic TR-50B design.

For a completely different approach, the Countryman B6 is the smallest lav in the world. You must hold one in your hand to comprehend exactly how small this microphone is. The really cool part is that besides being the smallest, it might also be considered by some as one of the nicest sounding as well as

one of the most flexible designs. This mic is so miniscule, that it can be easily used as a hair or wardrobe microphone. The microphone is also available with small plastic caps that can alter the basic frequency characteristics in case a different response is desired or if the microphone's response from being buried under wardrobe is too muffled. Very cool! The microphone is also sweat-proof and waterproof so if the talent will be in scenes with water or rain (the mic element itself is waterproof, not the power supply although it could be made waterproof with the addition of gaffer tape and silicon sealant) or if the talent will be sweating a lot. The Countryman B6 is also available in different flesh tones as well as black and white.

The Sanken COS-11 is another popular lavaliere that is slightly larger than the Countryman B6. The Sanken offers what some people feel is a slightly smoother response than the Countryman B6 at the sacrifice of a slightly larger physical size. The Sanken is also waterproof and available in flesh tone as well as black and gray. The Sanken is another marvel of engineering and is considered a premium, top of the line lavaliere.

Wireless Microphone Systems

There are also two methods that lavalieres are typically used with in production, wired or wireless. While wireless transmitters can also be used with a handheld microphone, they are most commonly teamed with lavaliere microphone elements. Some sound mixers also use wireless systems on their boom mic (this is very typical in reality type 'run & gun' style shows like 'COPS')

Also, some location sound mixers use wireless systems to "cut the tether" from the sound mixer to the camera although this strategy can be risky unless the camera operator is always monitoring the camera's audio. In general, use cables whenever you can although for certain shooting styles and locations, wireless must be used. A wireless system consists of a single transmitter and a single receiver. For some strange reason (costs?), many new DV users seem to think that a single receiver can simultaneously receive the output signals of more than one transmitter. The reverse is actually true, more than one receiver

CAN receive the output of a single transmitter simultaneously. So in multi-camera shoots, it is possible to have a single transmitter's output go to multiple receivers on multiple cameras.

Wireless systems are improving but the bandwidth that wireless systems operate on is becoming more and more crowded. Diversity (dual) antenna systems are considered far superior to single antenna systems because in a diversity system, when the reception between transmitter and receiver is weak or encounters interference on one antenna, the system can switch over to the other antenna. The UHF (ultra high frequency) band is generally considered more effective and less crowded than the VHF (very high frequency band). There are brand new digital systems hitting the market that have great potential to make wireless usage more reliable than it ever has been. Unfortunately, the first digital systems available are on the pricey side for DV users, costing considerably more than most high-end camcorders. That's the bad news. The good news is that the digital technology will trickle down the price scale just as it did with digital camcorders. In few years, probably almost all wireless systems will be digital. When that occurs, wireless microphone system

usage will undoubtedly skyrocket, as wireless systems are so much more convenient to use than dealing with long runs of cable. But until we are all using digital wireless systems, advice is to only use wireless when you must.

Another term to look for is "frequency agile". If your wireless system is "frequency agile", it means that when your system encounters other users, static or interference, it can be switched over to another operating frequency. I am amazed that all wireless systems DON'T have this feature, but they don't.

Some of the good Wireless systems for DV users?

Lectrosonics, are one of the best in wireless systems. There are systems available that perform as well as the Lectrosonics UHF systems, but the problem is that they cost a lot of money. The Lectrosonics 210 systems are another industry standard that deserves their excellent reputation. This system is probably used on lot of feature films and television shows made in the US. Lectrosonics' product is built incredibly well and performs reliably.

If you cannot afford a Lectrosonics 210 UHF system, you can check out some of their lower cost systems, but not all of them are frequency agile.

The Sennheiser Evolution 500 series is another viable, low-cost option for a wireless system. This system is available with several different options and performs reasonably well, especially considering it's cost.

When using wireless systems, consider renting over purchasing. This is another situation where you probably will not need

wireless systems all of the time so it probably makes more sense for the average DV user to rent the best rather than buy something "middle of the road". When using wireless systems, it is essential that you feed the system brand new batteries quite

often. Wireless systems eat a lot of batteries and begin to perform poorly as the batteries get weaker. If using wireless all day during a typical 10-hour day, plan on at least one and possibly two battery changes.

Other Microphone Types

Although a shotgun and a lavaliere are the standard production microphones, there are certain instances when you will want to consider using or supplementing your audio kit with other types of microphones.

Dynamic

Shotguns, hyper cardioids, cardioids and lavalieres are usually condenser microphones. This means that they each require internal or external power sources to amplify the signal that they

output. They must have an internal battery or require phantom power. Phantom power is supplied to the microphone via the XLR connector from the XLR outputs of recording devices or mixers. Phantom power is typically rated at 48 volts, abbreviated as 48V, although some microphones require less voltage. Why all of this discussion about condensers and power? Because a dynamic mic DOESN'T require any power, hence it is more reliable. Another characteristic of dynamic microphones is that dynamic microphones can generally record at much higher sound levels than condensers without distorting. When we discuss a dynamic mic, we are talking about such venerable "institutions" like the Shure SM-57 (the standard rock singer stage microphone) and the Electro Voice RE-50 (the standard newscaster location microphone). The downside is that dynamics are not very sensitive so they really only pickup sounds that are very close to the diaphragm. Unless you want to have the microphone in your shots, the dynamic microphone is not very useful in most production situations. When you are shooting something that is supposed to look "newsy", the hand held dynamic microphone is the "weapon of choice". Other than this application or recording live, high volume music or sound effects, its use is severely limited in normal video or film production.

Shure SM-57
ElectroVoice RE-50

Boundary

Boundary microphones (PZM - Pressure Zone Microphones) are usually flat or plate shaped microphones that are set on a table or fastened to a wall. These microphones are useful for recording roundtables or audience reactions with certain caveats. These types of microphones use the surface that they are fastened to pickup sounds. When used on a table with talent, these microphones will pickup finger tapping, pencil rolling, cups being picked up and set down and any other physical sounds that reflect off of the surface they are fastened to. Boundary microphones are also effectively used to mic large areas of audience and work well for picking up laughter and applause. They are not effective at picking questions from the audience.

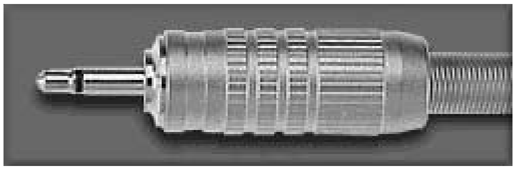
Stereo

Stereo microphones are more commonly used in recording musical or dramatic performances than in typical narrative dialogue situations. Dialogue almost always ends up as a mono element, even in 5.1 surround mixes so there does not seem to be much reason for recording dialogue in stereo since voice is not a stereo element. Another factor to consider is phase cancellation. Without a decent knowledge of stereo recording and postproduction processing and mixing, you may encounter phase cancellation where certain sounds within your mix will be cancelled out. On the other hand, stereo microphones can work very nicely for recording ambient environments. Any environment sounds more realistic in stereo since stereo recording mimics how our ears perceive our actual environments. There are stereo microphones on the market, ranging from inexpensive consumer units that sell for Rs 5000 to units like the Sanken CS-5, a high-end stereo shotgun. In summary, stereo recording is the exception rather than the rule in sound for picture recording. Rule of thumb for recording in stereo is to determine if stereo recording would enhance the subject matter, live musical or dramatic performance, for instance. In most cases, dialogue would not be enhanced by stereo.

Choir

Choir microphones are designed to be suspended above live choirs. They are good for picking up a large crowd of voices. Although they are good for recording an ensemble of voices, they are not nearly directional enough to pickup individual voices out of a crowd. If you shoot a live event, choir microphones are great for picking up the audience reaction, although once again, choir microphones cannot pickup individual questions or statements from an audience very well. There is no microphone that will pick up an individual sound well unless the microphone element is fairly close to the subject.

Notes :

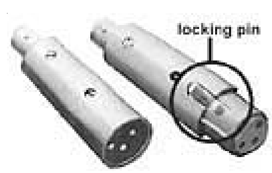


We have already discussed the differences between unbalanced and balanced connections and cables. Cables and adaptors are one of the most underrated yet most important parts of the

audio path. It pays to go for the highest quality cables and connectors. Canare, Mogami and Neutrik all make excellent connectors and cables. While there are literally dozens of different connections and adaptors used in audio and video production, let’s go over the most common ones:

XLR

XLR connections and cables use three leads, positive, negative and ground. The XLR connection has two components, a male connector and a female receptacle. XLR connectors usually have a locking pin so that they cannot be accidentally pulled out. XLR cables and connectors are the standard audio connections in most professional sound for picture equipment.



1/4"

Quarter-inch connections are more common in music gear than in sound for picture gear although there are popular devices like

the Mackie line of audio mixers that are commonly used in both music and video applications that use 1/4" connections. 1/4" connections can be mono or stereo and can be either balanced or unbalanced. These connections are also common as headphone output jacks on larger gear.

1/8" mini plug (3.5mm mini plug)

These connections (usually unbalanced) are very common on consumer camcorders as headphone output jacks and as microphone input jacks. This connection is fairly fragile since it is so small and hanging any kind of weight with cables, etc. off of the jack is not recommended. Plug stress is the most common cause of shorts, grounding problems or failure.

RCA (cinch connections, phono)

RCA connections are the same as the connections used to hook up your home stereo components. Because RCA connectors are always unbalanced, they are almost exclusively used on consumer gear although a lot of pro gear features RCA connectors as secondary connectors, to be used in addition to or instead of XLR connectors.

Impedance Matching

As we discussed above, there are many more types of connections and adaptors to go from any of the above connectors to any other type of connection, but the four listed above will be the most common ones that you will encounter. Special care should be taken when interfacing balanced with unbalanced equipment. There are special devices known as “impedance matching" boxes that are very useful for interfacing pro to consumer gear. You will generally have a better result when using a powered impedance matching device over a passive one. One such device is the Henry Engineering Matchbox. This is a box that can simultaneously convert balanced XLR connections to unbalanced RCA and unbalanced RCA to balanced XLR. These are very useful for dubbing setups from professional DVCAM and Beta SP decks to consumer VHS decks, for example.

Distance

Another factor to consider is cable distance. Unbalanced connections generally are not very good for distances over about twelve feet because of the lack of a balanced negative connection; noise also has a tendency to creep into the signal. The relative nominal signal strength of -10dB doesn't help either when it comes to distance. Balanced XLR connections generally are safe to use for runs of up to around 200 feet. At distances farther than 200 feet, even balanced lines can use a boost from a line amplifier to restore the signal's strength. Generally, in a location sound kit, you end up using mostly 25-foot cables for the lines from the microphones to the mixer. It does pay to build up a small kit of various adaptor plugs. You never know when you may have to take a sound feed from a telephone to the camera, a CD player to an MD recorder, etc. It pays to be prepared.

The Fourth Link - Mixing and Routing Devices

Most professional productions use an audio mixer. The exceptions are single camera shoots where the audio is just being used for reference or background ambient. But if you are shooting dialogue, you should be using an audio mixer. It's really that simple. If you are shooting with a stationary camera and AC power is accessible, you can get a great result with small desktop mixers. Just remember, with a location audio mixer you get:

- Tone generator - You must have a tone generator in order to record reference tone so that the editor has a constant point of reference as to the levels that the audio was recorded at the head of each reel.
- Roll offs and cuts - Location audio mixers often have high and low roll offs and cuts. These are useful for eliminating rumble, hiss and other undesirable sounds before the sound is sent to camera
- Superior Quality - Almost all location audio mixers have considerably higher quality microphone pre-amps than almost any camcorder, even professional high-end camcorders.
- Slate Microphone - This is a microphone located on the mixer that lets the operator insert their voice onto the mixer's output. Very useful for audio slates, for example; "room tone :30" when you go to record room tone.
- Better Monitoring - Location audio mixers usually have a tape return circuit that lets the operator compare the output of the mixer to the output of the recording device by flipping a switch.
- Panning and mixing - A mixer lets you determine the panning of as many channels of microphones as you have capacity for. Some of the better mixers also have multiple outputs. The PSC M4 MKII mixer that lets you route a set of outputs to the camcorder and another set of outputs that we route to a Mini Disc recorder for backup. With a four-channel mixer, you could route three lavaliere microphones to the right channel on the camcorder and the boom to the left channel.

- Microphone power - Certain mixers can also actually power certain wireless receivers.
- Phantom power - Almost all location audio mixers can supply phantom power to at least one of their outputs and some to more than one. This saves camcorder battery power, if the camcorder has phantom power. If it doesn't have phantom power, the mixer is able to provide the phantom power instead of having to buy a separate microphone power supply.
- Better metering - Most camcorders do not have very accurate, high quality audiometers. Most location audio mixers have very high quality audiometers.
- The ability to "ride" gain - Generally, most location sound mixers "ride" gain, smoothly setting the gain levels being sent to the recording device as the talent's levels rise and fall. This does not mean raising and lowering the volume levels as the talent speaks, it means making sure that the signal

being recorded does not clip. Try riding gain as someone is shooting with a typical consumer camcorder. In many models, you cannot even adjust the gain unless you go into a menu.

Favorite Location Audio Mixers

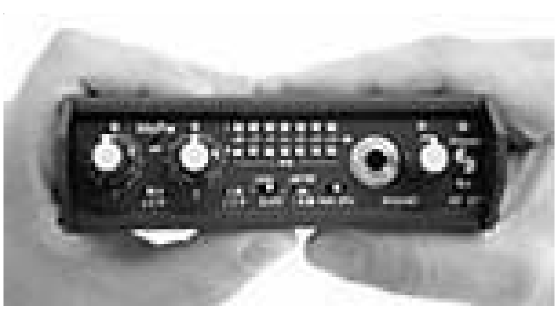
There are actually quite a few high quality audio mixers available but let us talk about just three here that would most likely appeal to DV users. All three of the mixers described are only two channel models but most are available in three, four and five channel versions. Any of the three mixers listed below are an excellent value.

PSC Mjr

The PSC Mjr is a two-channel version of the PSC M4 MKII, a very popular four- channel mixer. Valencia, Ca.-based PSC (Professional Sound Corporation) makes excellent products and this mixer has gained considerable popularity as a high quality, basic simple mixer.

Wendt X2

The Wendt X2 is a two-channel version of the Wendt X4, also a very popular four- channel mixer. Built by Westlake Village, Ca. based Wendt Audio with military grade parts, rugged construction and high quality assembly; this is an impressive mixer for the money. It is simple to use and provides outstanding sound quality. This mixer has been used to mix MTV's "Road Rules" and on "C.O.P.S." and is a good choice for high-stress, "run & gun" reality television shooting.



Sound Devices Mix-Pre

Ex-Shure audio employees founded Sound Devices, the makers of the Mix-Pre. This mixer is also sold as the Shure FP-24, but is actually built to Sound Device's specifications and is the same unit. The form factor on this unit is a little different than most of its competitors and its cosmetics and design are sleeker and less boxy than the competition. The Mix-Pre is acknowledged to be a high-quality audio mixer and is often seen on ENG news crews.

Impedance Conversion Devices

Beachtek DX-A4
Studio One XLR PRO

The devices we will discuss here are not really true audio mixers, they are impedance conversion devices. Both the Beachtek DX-A-4 and the Studio One XLR PRO adaptors are small boxes that take balanced XLR microphone level inputs and output unbalanced two-channel audio through a 3.5mm mini plug terminated 6" cable that most consumer camcorders can accept. Because the unbalanced portion of the output is only 6" cable, then extraneous noise does not enter the signal path. Both boxes also have internal capacitors that defeat the small voltage that most consumer camcorders output from their microphone input jacks for powering unbalanced consumer microphones. Either box attaches to the camcorder via a screw that connects the box to the camcorder via the camcorders tripod screw-mounting hole. If you own a consumer camcorder that doesn't have XLR inputs, you will need to buy one of these boxes. There are cables and adaptors that can also be used in place of the boxes but only some of these cables/adaptors have the capacitors needed to defeat the powered camcorder microphone jack output. The other advantage of these boxes is that the

inputs are microphone/line level switch able and both have

separate volume potentiometers for both left and right channel. 6v charge tripod screw-mount

Since the possibly heavy XLR cables and plugs from the mixer's output also plug into a sturdy box that is mounted to the bottom of the camcorder, there is no strain on your camcorder's fragile 3.5mm mini plug microphone input jack. You might experience grounding "issues" when using our Sony TRV-900s with the Beachtek when switching over to AC power. The unit sounds perfect when using batteries but a nasty buzz creeps in when using AC power with this particular setup. A grounding wire from the box to the camera's battery connectors solves the ground loop but you may not want to tape a wire to your camcorder. This problem may also be only isolated to the particular combination of camcorder and adaptor I was using. The latest BeachTek DXA-6 model also has a groundlift switch as well as phantom power on one mic input.

The Fifth Link - The Recording Device

The most commonly used recording device for most DV users would be your camcorder. For the sake of covering this section completely, we will also discuss some alternative recording devices including Mini Disc recorders, DAT recorders and hard drive recorders and the dual system sound recording workflow.

DV Camcorder Audio

The unfortunate truth is that if you use a camcorder, you are already beginning your quest for high quality sound with somewhat of a handicap. A few common consumer camcorders have decent sound quality but many of the most popular consumer camcorders have major sound issues straight out of the box. Keep in mind, these limitations are exclusive to each model and have nothing to do with the quality of the audio that is fed into each unit from your mixer. These are strictly the limitations of each device from an audio perspective and are opinions of many in the field:

Canon XL-1/XL-1S

Two major audio liabilities. The first liability is the use of either of Canon's XLR/shoulder pad mount combinations, the MA-100 or MA-200. Using either adaptor is not advised if you are trying to record high quality audio as both of them add considerable noise, hum and possible buzzing since they ineffectively transport and transcode the rear balanced XLR inputs to unbalanced input for the camcorder itself. The second issue with both of the Canon XL-1 models is that they both feature one of the worst headphone output stages ever seen in a consumer camcorder. They both lack sufficient output level and both seem to add a pseudo "limiter/ compressor" effect that makes it seem as if the audio levels the camcorder is recording are varying more than they are in actuality. It is difficult to record high quality audio if you cannot hear what the camcorder is actually recording. Other than these two limitations, the XL-1/XL-1S seem capable of recording high-quality audio with a Beachtek or Studio One impedance adaptor box in place of Canon's lousy adaptors. It can be tough "flying blind" though. If you are a Final Cut Pro user, there is also the Canon "sync adjust movie" issue and Canons "interesting" audio sample rates to deal with as well. Canon GL-1/GL-2

The lower priced Canons seemed to always be capable of recording decent quality audio. The only problem with the GL-1 seemed to be that it had no manual audio input control. Using ALC (automatic level control), it would occasionally react with incorrect levels when the input was overdriven and "pumping" when levels were not hot enough. The GL-2 seems to have remedied this issue by including manual audio input levels. Unfortunately, the flimsy optional cost Canon MA-300 audio adaptor XLR adaptor is guaranteed to break the first time any load or stress is put onto it so if you buy a GL-2, still budget for a Beachtek or Studio One adaptor. Oh, and both still suffer from the Canon sample rate and FCP issues.

Sony PD-150

The first run of the Sony PD-150s had horrendous hiss problems. That problem was subsequently "fixed" during later model runs. All of the PD-150s still have a considerable amount of hiss, although granted, they did reduce the amount of hiss when they "fixed" the issue. The problem is that Sony

didn't also "fix" this model's lousy overall audio quality. The PD-150 was also has a lower S/N (signal to noise) ratios. The PD-150 is one of the few consumer camcorders that also will accept line level input.

Sony VX-2000

Same exact story as the PD-150 except all of the poor suckers who were stuck with these are stuck with permanent hiss problems.

Sony VX-1000

This old warhorse was the first three-chip consumer DV camcorder. Released before the DSR-300/500 existed. Interestingly enough, even though it was saddled with a 32KHz only audio recording capability, the audio that it

recorded was actually decent, much better than any of the current or new Sony models.

Dual System Sound

Some DV users utilize separate audio recording devices like these to record their project's soundtrack.

From left to right, a Sharp MD-MT770 Mini Disc Recorder, Tascam DAP-1 DAT Recorder, Creative Labs Nomad hard disk recorder.

For the past few years, there has been much discussion about using Mini Disc, DAT or hard disk recorders for dual system sound recording when shooting DV. In theory, it's really easy. Sure, your VX-2000 may record horrible, unusable audio with a lot of hiss. Just bring along an MD recorder, pop in a disk and life is good, right? Well, it's too bad that in reality, it's just not that simple.

First of all, from a mechanical standpoint, even though both of the devices (camcorder and MD, DAT or hard disc recorder) are both digital, they are also both completely independently synchronized to their own internal master timing device. You would think that if both devices are digital and both are using the same sample rate that it would be a simple matter of lining up the picture and the MD, DAT or hard disc recorded audio on the same timeline and let them go. The problem is that neither device is synced to each other's master "clock" so eventually, the two sources will lose sync with each other. The two may lose sync at 5 minutes; they may lose sync at 45 minutes. There are no hard and fast rules about how long each set of devices can hold sync. There is no way to determine exactly how long it will take unless you test the devices and notate how long it takes for the two to lose sync. Still sounding easy and straightforward? Let's say that your particular combination of DV camcorder and sound recording device will stay in sync for 19 minutes at a time before they lose more than one or two frames sync. You did your homework and determined the number. The next challenges become ones of routing, organization and media management. Let's go over each of these:

Routing

You have decided to do the dual system sound thing. That's cool. Your first issue to determine is routing. Realistically, you need the mixer's output to go to your camcorder as well as your digital audio recording device. You DON'T want to use the camcorder's microphone to record the audio for the scene because if you are any significant distance from the action with your camera, there is a distance-induced delay that can make it challenging to match your "guide" audio in postproduction. You need the audio from your mixer on your camcorder as a guide for lining up the audio from your digital audio recording device. In this way, both devices will receive an identical signal. It is recommended not to use the audio from digital audio recording device audio during the actual post process. If you keep comprehensive sound reports and are organized, it is more efficient to edit your show, approve the cut, and then add the "quality" audio at the end. "Post conform" the audio, as it's called. Does your mixer have dual audio outputs, one set for the camcorder and one for the digital audio recording device? If so, great. If not, now you are looking at an "interesting" combination of either an audio distribution amplifier or splitter or some other way of splitting the mixer's output without destroying the quality of the signal. Is using double system sound still sounding simple?

Organization

Organization is the key to making double system shooting work well. The problem is that most DV users don't have a dedicated audio crew, so who is going to keep sound reports? Sound reports are small sheets of paper that are commonly used to keep track of which sound "reel" goes with which film or video "reel". The sound reports are also used for notation about any individual quirks or anomalies. Who is going to slate each take? It's not absolutely essential to slate each take but it sure does make it easier to organize your material when you enter postproduction. What if you are shooting a scene called "Jose enters room"? You do 17 takes of it, have to change videotape and MD during take 8. How are you or the editor going to be able to keep track of which take is which? A slate at the head of each take at least gives you a point of reference about which take is which.

Media Management

Media management is the simple act of making sure that the sound and picture elements are cataloged and kept organized. This is the act of making sure that each time a videotape or film reel is changed on the camera that a new Mini Disc or DAT is inserted and begun. Having a separate sound element for each reel is not the most economical way to shoot but it is the best way to keep the elements organized. If you shoot with a hard disc recorder, this becomes impractical but it works well when shooting with MD or DAT.

You're Really sure about this?

So are you getting the picture here? The bottom line is that shooting with double system sound and not having it turn into a nightmare means three things. It takes more time (time is money, right?), it takes more crew positions (who would keep the sound reports, slate and roll the MD, DAT or hard disk recorder each take?) and it takes more work in postproduction. I make the conclusion that most DV users are ideal candidates NOT to use the double system sound workflow. Most DV users do not have enough budget and enough crew. Of course, all of these factors are proportional to how long and involved your project is. If you are shooting a :30 commercial and just a few locations and shots, the double system sound may work fine for you. On the other hand, if you are shooting 60-100 hours of footage for a documentary, think long and hard before you commit to the workflow. But as film makers you should know about all the options possible when you are budgeting for your next project.

Mini Disc

Probably the most popular double system sound format, Mini Disc is a good value for the money. Mini Disc uses a compression scheme called ATRAC that is fairly good, especially for dialogue. Mini Disc does have a somewhat limited dynamic range so it would not be my first format choice to record highly dynamic sound. There are many Mini Disc recorders on the market and there are several that can even take the optical output of a digital microphone preamp. The media is cheap and easily available. One of the downsides of MD is that most of the non-pro units only have digital input but not output so you end up having to purchase a home MD player with optical output as well as an audio card or Firewire audio interface if you want to transfer the MD's digital signal into your computer and keep it digital.

DAT (Digital Audio Tape)

DAT is probably the choice you would use for higher quality than MD if you were shooting with double system sound. The upside is that DAT is basically uncompressed audio. It sounds better than MD. The downside is that it is linear and the media costs four to five times as much as MD media. DAT recorders, even the cheapest DAT "walkman" cost more than MD recorders with comparable features. DAT does give you the option of using time code although time code capable portable DAT recorders are quite expensive and are not currently being manufactured. They are still easy to rent from most audio houses though. DAT is still used on most Hollywood features although more and more of them are using an advanced and very expensive time-code-capable hard disk recording system called Deva by Zaxcom.

Other Hard Disk Recorders

Creative Labs Nomad, an inexpensive hard disk, is a portable CD player-sized hard disk recording system. The advantages are pristine quality and relatively low cost. The Nomads are capable of recording uncompressed but of course, uncompressed recording uses up lot of disk space. There is also the issue of input and output, interface, reliability, verification and the fact that all material must transferred to another format to archive.

The Sixth and Final Link - The Monitoring Circuit of the Recording Device

The monitoring system on your recording device is the last part of the equation. As we have already discussed, there are several recording devices that have substandard headphone monitoring circuits, including the two most popular consumer DV camcorders, the Canon XL-1 and the Sony PD-150. It can be difficult to judge your audio efforts if you cannot hear what is being recorded properly. The good news is that if you have invested in a decent audio mixer with a tape return monitor circuit, you can at least compare the audio signal you are sending the recording device with what the monitoring circuit in the recording device is reporting back to you. By doing this comparison numerous times over a long period of time, combined with listening to what you recorded on accurate audio monitors that you hopefully have hooked up to your edit system, hopefully you can begin to hear past the deficiencies in the monitoring circuit.

Headphones

The last component in the audio chain are your headphones. The headphones you use to monitor what you are recording serve several very specific purposes:

- You must be able to judge the characteristics of the room, your talent's voice, sound effects and ambient levels.
- You will need to be able to determine if your microphones are picking up the traffic outside, the helicopter in the distance, etc.
- They must be portable enough that you will always bring them and use them on everything that you shoot that has audio
- They must be efficient enough to output high enough sound levels from weak or inefficient headphone circuits.
- They must cover your entire ear. You cannot judge what your recording device is recording with headphones that only partially cover your ears. You will be hearing too much of the live sound and not enough of what the recording device is recording to make informed judgments about what you are recording.

Some of the headphones available are as follows.Some location sound mixers favor the more accurate sound of the Sennheiser HD-25s but they are twice as expensive and are not as small as

the Sonys. It's up to you as to which features and sound will suit your needs.

Sony MDRV-7506s
Sony MDRV-600s
Sennheiser HD-25s

Renting Versus Owning

Renting Pros

1 To rent the best, state of the art gear is still fairly inexpensive. You can get you a basic location audio kit at quite cheap price. You just need to figure out the best place to get your gear. Do not compromise on sound.

2 You can try out different types and brands of gear before owning to see what works best for you.

3 The rental company takes care of the maintenance and some expendables

4 If it breaks, the rental company has to deal with the repairs, warranties, manufacturer, and dealer

Renting Cons

1 It can get expensive to rent the gear on long-term projects

2 Sometimes, certain rental houses will still try to rent thrashed equipment. Build a good relationship with a reliable, trusted rental house and this shouldn't be a factor.

3 It's not as convenient to have to go pickup the gear and return it

Owning Pros

1 If you own it and you use it, you will know who thrashed it or took care of it

2 Convenience. When a shoot comes up, you are ready to go.

3 Cheaper to own on long term projects (if you are shooting enough of them or if they are long term enough)

4 Audio gear doesn't outdate itself very often. Changes in gear are often incremental and not dramatic. Unlike computers and camcorders, rarely will you buy a piece of audio gear and regret it because the "latest thing" just replaced it.

Owning Cons

1 Gear breaks. Gear needs to be maintained. It takes time money and effort to keep everything in top condition and working perfectly.

2 Unless you are willing to spend the big bucks, you will not have the best gear, you will probably have to compromise in order to keep your kit affordable.

3 Eats up storage space in your office or business

4 If you have one location sound kit and end up where you are doing multiple projects at once, you will have to rent anyway for a second or third kit.

Putting it all to Work

Location sound is a craft, not just a crew position. There is no way that you will learn how to record location sound by just reading articles on the Internet. Remember, location sound is somewhat of a paradox in that the better you are at doing it, the easier it will seem. When you sit down to edit the footage you have recorded quality sound on, you will probably not think to yourself, "Wow, I did an awesome job with this sound." You will hopefully hear it, realize that it is fine and proceed with the editing. In a way, doing a really great job with location sound recording is a somewhat thankless pursuit, even if you are editing your own footage. If you do the job well, nobody will complain about the sound. That's about the best you will probably experience with it.

Preparation

One of the most important things you can do to improve your location sound is to be prepared. You cannot record high quality sound without the proper tools to do the job. If you sit down with the director and producer, even if you happen to be both of them, and you determine that you will need eight wireless microphones and two boom operators to successfully record that three camera, live, one-take scene, then so be it, those are the tools that will be needed to successfully record the sound. You may have to be prepared to justify to the producer/client/financier why you will need the tools that you will to get the job done. In order to learn which tools you will need to do a particular job, you must gain experience. The only way to gain experience is to get out there in the field and begin to record.
I am a big believer in location scouts for sound as well as for picture. Go to the location you will shoot at, close your eyes and listen. Listen to the room characteristics or the environmental characteristics, if it's an exterior. Listen to the ambient noises you will have to work around and work with. Which sounds can you eliminate? Which sounds can you minimize and how?

Prepping on the Set

When you arrive to the shoot location, after setting up the sound equipment, DO A SOUND CHECK! If there are actors or just regular non-professionals that will serve as your talent,

setup your audio equipment and have the talent rehearse the scene or if it's an interview, setup your microphones and have the talent chat a while hear how their voice projects and what levels you should set. If you are setting up before talent is on the set, have a crewmember stand in and get some levels.

Action!

The location sound mixer's job is to ensure that the best possible sound quality gets onto the tape. Anything else that is expected of the sound mixer is a distraction to that goal. This is the tough part of resolving recording high quality location sound with the entire DV "do it all yourself cheap or free" mentality that is falsely propagated in advertising for DV products and all over the Internet. The big secret that nobody seems to want to admit is that it still takes talent and people to fill certain crew positions to effectively perform all of the functions necessary for producing high quality work. Cinematography is cinematography; it doesn't really matter if it's lighting for DV or 35mm film. There are differences in how to light for each of the two mediums but actually any DP will tell you that DV is harder than 35mm film to make look good. Audio is the same way. It's easier to work with top of the line audio equipment with film cameras or HD/Digital Betacam and obtain a good recording than it is to work with the convoluted signal path of professional front end audio equipment (microphones and mixers) feeding recording devices with unbalanced inputs, inadequate or no audio metering and questionable sound quality. The main point to all of this is to make it clear that the "DV Revolution" really only has to do with equipment price and performance parameters. The DV Revolution has nothing to do with the talent, skills and education required to be effective at the art of film and video making.

Fifteen Tips and Tricks

There are some fundamentals and tips that should prove helpful in at least getting you started. Here are a few:

Channel Redundancy?

When you are running a single microphone source into the two channels on a camcorder, split the output with a mixer or "Y" cable and feed both audio channels of the camcorder. Set one channel to peak a few dB lower than the other channel. In this way, you will have a buffer of a few extra dB in case the talent or subject gets too dynamic. Also, this redundancy is helpful in a more practical way. The audio tracks on a DV tape are smaller than a human hair so having two of them is always better. A chunk of debris can cause a dropout on the tape that may be just during an important part of the scene. Having that extra track of audio, even when using a mono microphone only is cheap insurance.

Auto Versus Manual input Level Control

Should you use auto audio levels if you have manual audio input levels? This is dependent on the specific camcorder and situation you are shooting under. On most consumer camcorders, setting the camera's audio input level to under 50% and controlling the levels with the mixer will get you the best possible sound quality. Each situation is different though.

Audio Meters

Generally, when shooting with consumer DV, most of the camcorders used do not have segmented meters. The only way to determine which levels on the ambiguous camcorder unmarked meters means what actual dB level is to run tests with a tone generator and FCP. In that way, you can learn what the "red" segment on your un-marked meters is in dBs.

Panning

When recording two channels through an audio mixer, it is generally a good idea to hard pan each channel all of the way to the left and right so that the two signals are recorded separately and independently of each other.

Room Tone

What is room tone? Room tone is the ambient sound of the shooting environment that is recorded separately for use in patching, smoothing and matching ambient background levels when editing. It is imperative to record at least :30 of room tone in every sound setup.

Phantom or Battery Power

Double powering. Beware of double powering a phantom microphone. If your camcorder has phantom power switched on and your mixer has phantom power switched on, this is not a good situation. Only one phantom power source at a time please.

Checking Playback

After the audio is setup, camera is setup, roll tape on a sound check. Then rewind tape and listen to the playback. This verifies that picture, as well as sound is recording okay. You can get burned by tape head clogs or other problems that won't be apparent unless you check playback.

Lavaliere Batteries

Most professional lavaliere microphones can be powered off of phantom power from the mixer but if you are using batteries to power your lavalieres, make sure you have a good supply of them on hand. The Tram TR-50B, for instance, takes an unusual size of mercury battery that can be difficult to locate.

Connections

When using non-XLR connections, many common audio issues can be traced to plugs not being inserted all of the way into jacks. Make sure that all of your connections are tightly connected when investigating noise issues.

Boom Pole Holder

Try to keep a good boom pole holder with you.

Lavaliere Placement

Hairpiece tape, which can be bought at most beauty supply stores, is excellent for taping lavaliere microphones to skin when setting up lavs under wardrobe.

Sound Reports

Using sound reports, especially on long form or involved shoots is a great habit to build and will help you be more efficient in postproduction.

Clothing

There are professional sound vests available with compartments for lavs, batteries, cables, tape, etc. Very handy. Although wearing one makes you look a bit geeky, when time is of the essence on set, it can be very helpful to have some of your tools at your fingertips as you setup the microphones for each shot.

Tools

Most of the grips on set will have a small nylon belt pouch that will contain a Mag Light and a Leatherman or Gerber all-in-one tool. You should have one of these as well. The Mag Light will come in handy when set lighting is not on, for tracing cable runs, checking connections, etc. The Leatherman-type tools are handy for a huge number of tasks.

Microphone Level Versus Line Level

Whenever possible run line level from the mixer to the camcorder (some camcorders accept line level, others, only microphone level). Line level is much higher than microphone level, giving your audio a better S/N ratio and less chances of noise creeping into a weak signal.

Location Sound - Beyond the Basics

In this article, we've covered a lot of ground about location sound and the part that it plays in making your project appeal to your audience. We've discussed preparation, planning for the shoot and we've covered the specifications and use for a lot of location sound equipment. Although we have covered a lot here, just realize that we have barely scratched the surface and that producing great audio for your projects really begins with the simple desire to communicate with your audience on a more sophisticated and effective level.

There are many great resources available for learning more about the details of recording location sound but really the most effective way to learn about location sound is to get out there and do it. Use your ears and listen to what works and why. The skills and a trained ear are so much more important than the equipment. Anyone can buy or rent this equipment, but not just anyone can use it effectively. You have to invest the time and effort to learn how to use it correctly.

Believe it or not, one of the most enjoyable aspects of location sound is that it is never perfected. Once you reach a certain level of competence with sound, then the nuances of fine-tuning the sound come into play. It is much more pleasant to think of ways of enhancing the sound on your projects than trying to repair major audio disasters. There are always new challenges to face and new problems to solve. As we move forward into the age of High Definition, digital filmmaking, DVD, multimedia and surround sound, the requirements and appetite for high quality sound will increase.

Notes :

These terms will let you know about mixers. They will help you compare the features of the boards you see.

What does a Mixer actually do?

A mixer allows you to balance, position, effect and equalize its different audio channels into a good sounding sonic image that we call a mix. You can add effects to some channels but not others, position instruments to a location in the stereo field, route channels to outboard gear that produces an interesting effect and "sculpt" the sound of each channel with a dedicated equalizer where you can vary the bass, treble and mid range in such a way that the whole song/voice/music "gels" into a work of beauty.

What's a Bus?

A bus is a major pathway to a single fader. You can take everything going to that fader out of the mixer where you can send it to another piece (or rack) of gear. You can also bring the signal back in to the mixer on spare channels. On mixers with busses, there is a switch on each channel that lets you route the whole signal to one of the busses. The Main bus is often called the L/R bus. Other busses are sometimes grouped in pairs, like the 1-2 bus, 3-4 bus, etc. There is also a switch, usually, that lets you route these bus faders to the Master fader. Typical uses of busses are to send a track or groups of tracks to a digital multitrack, or to a soundcard or audio interface. That's the critical point. Make sure you understand it. Yet you can also be very creative with them, such as sending them to samplers, grooveboxes with analog inputs, surround encoders, separate compressors and more.

What's a Send and Return?

A send is a major audio path that goes out of the mixer. On each channel, ideally, there is a knob for each send so you can direct variable amounts of the channel to the pathway. This can function as a separate monitor mix if you are doing a live show, where every player has a monitor in front of them. Or in the recording situation, the send typically goes to an effects unit. The signal is brought back to the mixer by the returns, then is added to the main signal. Creatively speaking, a send is a submix, do with it what you want. You don't have to bring back the sends to their returns. You can bring them back to an empty channel where you can continue to process with EQ, or to a bus fader if you want. You can use the returns like any other line input, patching in synths, other mixers, computer soundcards even your cd player, turntables, TV whatever.

What is a Channel insert?

An insert is a pathway out and then back into a single fader. You use it to patch in an external piece of gear that only affects that one channel. Typical uses of inserts are patching compressors, outboard EQs, exciters, pedals, multi-track recorder i/o, and FX boxes. Lots of people route channel inserts to a patchbay where they can plug in various devices conveniently. We are talking Hardware plugins here. On a well-featured mixer, there are inserts on individual channels, buses and the master fader.

You should be getting a sense that there is no one way to set up a mixer. Its really just a matrix of audio pathways that can be used to build up a sculpture of sound that is called a mix.

What is "In-Line Monitoring"

In line monitoring is a feature on some recording consoles, like the Mackie 24-8 or Behringer MX900 and Alesis Studio 32 that allows each channel strip to have 2 separate inputs, with a switch that sends either input to the fader on that channel. Usually, the input that is not selected goes to a knob on that channel where it can be turned up and be "monitored" on one of the busses. This is how 16 channel boards can have 32 inputs at mixdown. This is sometimes called a "Mix B" feature.

"Balanced" Ins and Outs.

A mixer with balanced inputs and outputs allows you to connect gear that is balanced with XLR or 1/4 inch TRS cables. This essentially allows you to run cables in greater lengths, even to the next room, and not pick up hum or noise. Mixers with unbalanced ins and outs require shorter cable runs, and use 1/4 inch TS cables or RCA jacks. If your audio interface has balanced inputs and outputs you may want a mixer with balanced inputs and outputs. Most mixers with balanced inputs will accept unbalanced cables too. So if you are using a soundcard with unbalanced RCAs you can still use a mixer with balanced inputs most of the time.

Who needs a Digital Mixer?

- People with standalone digital multi-track recorders
- People that want to keep their signal totally in the digital domain
- People that want to use the digital mixer instead of a software mixer or in conjunction with a software mixer

Going to digital mixing is not a decision you want to make lightly. The question: is the perceived audio result worth the trouble of learning yet another gear language, dealing with menus and submenus? It's not for everyone. Many of the functions of a digital mixer can be had without one. MIDI sequencers can do volume fades, effects fades, and can automate virtual effects sends and returns, pans, even eq sweeps. If you are planning to do automation at the sequencer level, do you really need another layer of automation after that?. However, if you are interfacing a stand alone multi track recorder that does not have an onboard mixer (or only has a simple one) such as an ADAT, Roland, Mackie, Korg or Tascam 8, 16 or 24 track recorder, then you bet, a digital mixer will let you automate your rough tracks and polish them down to sweetness. And for the true die-hard tweaks who want every nuance of their mix recallable, including bus destinations, internal effects settings, onboard eq and compression settings, a digital mixer will reward them with consistent, repeatable performance.

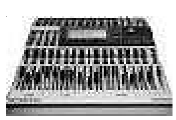
Perhaps the main advantage of going digital is that you can keep your signal path totally in the digital domain all the way to the computer. Is this True? Well yes. If. That is, if most of your studio is digital. If you like to use global inserts on the mix bus, that is, route the master out through a compressor, exciter, eq, you better make sure it's digital too, or you will be doing extra da/ad conversions. Read up on the quality of analog to digital converters, this is a picky point with the pros. Also double check on the number of analog inputs you get. Its very common for a piece to tout 24 channels but only have 8 analog inputs. When you add in the price of 2 extra analog expander modules to take you to 24 you find yourself at a premium price point over and above a classy 48 input analog dream board. Don't think that because the board is "analog" that it is "old" and not as good. People love the dedicated faders knobs, the warm sound, and the immediate response of the sound to the twist of a tweak.

Examples of Digital Mixers:

Tascam DM24 24 Channel Digital Mixer

24-bit, 96kHz compatible. 24-channel/8bus configuration. The DM-24 is the first affordable digital mixer that combines 24-bit, 96kHz audio quality with highly flexible routing, extremely powerful built-in automation, built-in professional-quality effects, dynamics processing and parametric EQ. Built to interact flawlessly with the MX-2424 hard disk recorder or any other high-quality multi-track system, the DM-24 is destined to become the centerpiece of great studios around the world.

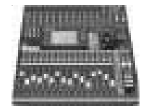
Behringer DDX3216 Digital Mixer



32 channels. 16 busses. Analog feel.
The DDX3216 represents an affordable, fully automated 32-channel, 16-bus digital console with 8 aux sends and unusually flexible routing options. Incorporating the latest technology developments in digital signal processing, the DDX3216 is destined to challenge the best analog and digital consoles on the market. 100-mm motorized faders, built in effects, plus Compressor/limiter plus gate, 4-band parametric EQ, sweepable high-pass and phase reverse on all 32 channels. Getting great reviews.

Yamaha 01V96 Digital Mixer

Yamaha's flagship digital mixing consoles are the accepted standards throughout the world: the awesome PM1D for sound reinforcement, broadcast, and theater; the DM2000, DM1000, and 02R96 for sound and music production. Now the 01V96 brings you the same performance and reliability in a smaller, more affordable format that's perfect for the smaller



professional production studio. Still, it has a maximum 40-channel input capacity and can be cascaded for applications that require more.

Roland VMBasic 71 VM7100 Vmixing Package

The VM-7100 is a processor with 10 analog inputs plus room for one 24 channel R-BUS expansion.

Roland M1000 10-Channel Rack Mixer

The M-1000 10-channel Digital Line Mixer is a 1U rackmount mixer designed to mix digital signals of varying sample rates. The M-1000 comes with four stereo S/PDIF digital inputs and a stereo analog input all with 24-bit/96kHz sound quality making it perfect for mixing electronic musical instruments and recording gear equipped with digital outputs. The M-1000 can even mix audio from a PC via its USB port. Word Clock is also provided, and multiple units can be linked for greater mixing power.

So, are you using a modular multi-track like an Alesis ADAT, or a hard disk recorder like an HR 24? If the multi track has no mixer of its own, then you will need either a large analog mixer or a digital mixer with the right number of channels. Count up the number of tracks. 24? You will need that many channels to play them all at once. Now add more channels for inputs to the multi track recorder. This is where boards with "in line monitoring" come in useful. You actually use the same channel strip for input and output of the recorder.

Mixing is like sculpting. Something the hands do on a board. This is why analog boards remain popular. While these craftsmen may be the last to think they need a digital mixer, they are probably the most suited to realize their benefits, the biggest of which is to be able save and collect these beautiful, hand carved mixdown configurations, with all the send and returns set just right, recallable when the need arises. Professionals doing film scores already know this. They don't want to waste time remaking the scenes that come up again and again in their work. It allows them to have templates for the may types of music they are called upon to make.

If you are using a digital multi track that already has a built in mixer, you might only need a rackmount mixer for your arsenal of synths. You can route different sources to different tracks easily. Just make sure you have enough preamps to cover your mics.

Digital mixers make a lot of sense with 24 track machines because you can use the digital piping schemes like ADAT lightpipe, TDIF and R Bus to send 8 channels down a single digital cable.

Who Needs an Analog Mixer?

- Those who simply want to add mics and guitars to their stereo soundcard

- Those who want to mix down tracks coming from an audio interface
- Those with lots of synths and outboard gear
- Those who want to monitor their midi synths without recording to audio

Analog Mixers can be Classed Meaningfully by 2 Variables.

1 The number of channels and

2 The number of busses.

That's why you see most mixer have numbers in the model name, like a Mackie 32-8. That means 32 channels, 8 busses. Sometimes they add another 3rd number. Lets take a Behringer 2442 for example. That means 24 channels, 4 busses, 2 master outs. Unfortunately manufacturers don't always follow rigid definitions here and the numbers can be deceiving. Some "4 bus" mixers count the master outs as a bus, so its really a 2 bus+2 master outs. We could call those 2+2 bus mixers. If you are interested in one of these, check the specs and make sure the mixer is a "true" 4 bus if that is what you want. For the number of channels, some manufacturers count the effects returns as a channel and some don't. So a 16 channel mixer might only have 12 channels and two stereo effects returns. Confusing? Yes. Lets try and get you through this. In all cases, read specs and find out how many channels the board really has not counting the aux returns. And how many busses the board has independent of the Master outs.

Types of Analog Mixers for a Computer Recording Setup

The Main question: How many tracks need to be recorded simultaneously (i.e., at the same time)? You need as many inputs on your soundcard or audio interface to cover that first off. Then you need that many independent outputs (sub outs) on your mixer to mate with these inputs. These independent outputs are called "sub outs" or the "alt 3-4 bus". These are different terms for the same thing. You can also use direct outs if the mixer has them, if you don't mind re-patching the gear you want to record into it's path.

Types of Analog Mixers

- 8 Bus (ideally for 8x8 audio interfaces, standalone multitrack recorders)
- 4 Bus (ideally for 4x4 audio interfaces)
- 2+2 Bus (Ideally for 2x2 stereo soundcards)
- Stereo Only (Ideal for those only connecting mics and guitars for tracking though workarounds are possible)
- Preamp only (for those going mixerless on a budget)
  We'll talk about each one with examples to help you get a better idea.

"8-bus" Mixers (up to 8 mics/Players)

If you intend to record a full band in your studio, you need a board that can handle lots of mics and instruments simultaneously and let you listen to tracks already recorded. Remember the drum kit might take 4-5 mics itself. You'll probably need to send a monitor mix out one of the busses, have inserts on every channel for patching stuff in during mixdown, generous sends and returns. 8 bus mixers are a great way to go. Take a

look at the Behringer MX9000, MX3282a, Mackie 24-8 and 32-8 mixers. If you are using a computer to record, pair that to an 8x8 audio interface, like a delta 1010. You are also able to mix all 8 outputs coming from the audio interface on mixer faders.

Examples of 8-bus Mixers

Behringer MX9000 Eurodesk Mixer

The MX9000 mixer features an expander port for linking consoles, comprehensive monitoring facilities, and integrated meter bridge. The MX9000 has been around for years at much higher prices. The built in meter bridge is cool; it has TWO inputs for gooseneck little lights, 8 busses and the MIX B tape returns gives it the flexibility of 48 audio channels. That's enough for all you synths, your multichannel audio interface, FX, tape machines, mics whatever. This is a BIG board. 37" wide, 29.5 " deep. It's heavy too-70 lbs. It's solid steel. Nice layout. The knobs are well spaced where you can actually turn one without touching others.

Behringer MX3282A Eurodesk Mixer

32-channel (24 mic/4 stereo line) 8 bus console, 8 aux sends per channel (on 6 pots, 4-8 switching pre/post). No inline monitoring on this one, but with 32 inputs, you might not need it Mackie B328 Recording PA Console (32X8X2) 8-Bus consoles have recorded more platinum albums and major motion picture soundtracks than any other mixer in their class -they set the standard for affordable 8-bus consoles. Excellent for project studios and digital multi-track recording, when combined with digital multi-track or hard disk recording systems, a Mackie 8ÝBus console can create not just major label CDs, but major motion picture soundtracks and network quality commercials. Used by pros and projects studios. Note that the meter bridge you see is an option.

"4-Bus" Mixers (3-4 players)

If you are planning to record 3-4 musicians into the computer simultaneously, make sure you have at least that many audio interface inputs and preamps going to your computer.
Mixers that do have 4 sub outs are the Behringer UB2442, MX2642a, Alesis Studio32, Mackie 1604 VLZ Pro. Of course there are more. You have to read the specs carefully. Again, don't go by the digits of the model number. These extra outputs can be switched on for channels you are recording, and can be removed from the main mix while doing so, which prevents feedback. A mixer with sub outs lets you record to and play from your soundcard at the same time and have the mixer control all of it. If you have a 4x4 audio interface, like a Delta 44 or Delta 66, a mixer with 4 sub outs is fine.

In the case of 4 musicians, each musician gets their own mono channel, assigned to a separate bus out to a separate channel on the audio interface. That way, in the sequencer, each will have their own isolated track.

"2+2 Bus" Mixers (1-2 players)

If you do all the recording yourself, one or two tracks at a time, or with one other person, you each having a separate mono track, a mixer with only 2 sub outs is fine. The UB1832FX, MX1604a, the UB1204FX all are in this class, as is the Behringer 1604 and 2004. So are the Mackie 1202, 1402, and many others. A suitable 2x2 soundcard for these is all you need, like an m-audio audiophile 2496. Of course you could use a 4x4 or 8x8 interface with these too. In the case of a 4x4, you could route all 4 outs back to the mixer and play with your mix there.

Stereo Mixers

These are for people that just want to get their stuff recorded on the computer and are monitoring from the output of the soundcard or to a stereo receiver connected to the soundcard. That's fine. However, a little stereo mixer that only sums to one stereo out is NOT going to cut it here if you want to control the input and output of the soundcard. Many people buy stereo mixers and later find they really needed a 2+2. Don't confuse the "main outs". "control room outs" "tape outs" with sub outs. Rephrased: You cannot use these to record to a computer and have the mixer play back the recorded output. You'll get feedback, or you will be unable to separate the track your are recording from the tracks playing back. The really inexpensive mixers (like the Behringer 802a and even some mid priced mixers) do not have sub outs. Stereo mixers are good for those only recording their mics and guitars and not running the soundcard out back to the mixer. Lets recap, stereo mixers are not ideal for computer recording.

There are workarounds, if you already have one. If the stereo mixer has direct outs (most do not) you can use those to route to the soundcard. You can also use the aux sends to get a signal to the soundcard. That means forget about adding effect processors. If your mixer has a switchable tape in out path that removes it from the master outs, you might be able to monitor from those. However, many inexpensive boards don't have that facility. If you use the master outs, control room outs or tape outs you may have feedback problems. So my advice is, if you want to both record to and playback from the computer on your mixer, get a 2+2 or a 4 bus. It's so much easier and you'll be happier with the mixer and will be able to keep it longer as the studio expands.

Inexpensive Preamps for 2x2 Soundcards

This is a solution for the person that want to monitor from their speakers connected to the soundcard but want to connect higher quality mics with XLR connectors. Using this mixerless" approach, you would do all your mixing in your computer application's software mixer.

Summing Up

To sum up for now, mixing music is an art form. It is the control surface for your musical ear. Consider that first. Be careful not to take on more technology than your vision inspires, yet remember with mixers you want to cover your future needs as well as your current needs. Yet if you envision a music that takes the most sophisticated of tools to get where you want it to be, follow your flow. After all, its just a flow of electrons disturbing the air in ways we have associated with pleasure.

Mixer setup

Basic Layout

The power switch for the mixer is located behind the board on the right side next to the power cord. Make sure all faders are down before turning the power on.

Device Assignments

Each fader on the mixer should be labeled with the device it controls. Say, if you want to play a DAT, bring up faders 1 and 2. The Cassette player could be on faders 3 and 4. A CD player could be on faders 5 and 6. Other devices possible include digital effects device etc.

Bring up the faders for the devices you are using. You must also bring up the main fader on the far right(master faders).

If you don't hear anything, Check the Following:

- Is the keyboard turned on?
- Is the volume knob turned up on the synthesizer?
- Is the amplifier on?
- Is the mixer on?
- Make sure the MUTE buttons are not pressed above the faders you are using.
- Make sure the L-R button is depressed for the channels you are auditioning.

Buttons and Knobs

- The MUTE buttons above each fader mute that particular channel.
- The PAN knobs on the mixer allow you to adust the stereo location for a particular mixer channel. Generally, you should pan a stereo output left and right.
- Above the Pan knobs are knobs for the EQ (equalizer). You can modify these as you like, but generally you should keep these pointing to 12 o' clock.
- Above the EQ knobs are the knobs for the Auxillary sends. These are normally used to send the signal from a particular channel to an effects device. Auxillary sends are used in conjunction with the audio patchbay above the mixer.

Recording a Cassette of your Work

1 Press the MUTE/ALT buttons for the faders you are not using.

2 Bring up the faders for the cassette deck on the mixer. Make sure the MUTE/ALT buttons are NOT pressed above these channels.

3 Record enable the cassette deck and bring up the recording level.

4 Play your sequence. You should see the meters on the cassette deck responding. If faders are up, you should also hear the sequence as it comes from the cassette deck. If you don't, make sure the cassette deck is monitoring the SOURCE, not the TAPE.

Using the Patchbay

The patchbay allows you to change the organization of the mixer without pulling cables out. The top row has the output from each mixer channel as well as Auxillary sends 1-4. The second row has inputs to each of the mixer channels. Plugging an audio cable into the input of a particular channel will over ride the device that fader normally controls. For example, if you bring in your electric guitar and plug it into the patchbay input for channel 3. Fader 3 will now control the output level of the guitar.

Using the Aux Sends on the Mixer

Let's say you would like to send the output of the channel(voice) to the input of the effect unit. The best way to do this is as follows:

1 Make sure all the Aux send knobs are set to the lowest level. This is especially important for the effect unit faders!

2 On the mixer channels for the voice, turn up the knobs for Aux sends 1 and 2.

3 Using two audio cables hanging on the rack on the wall, connect the outputs of Aux sends 1 and 2 on the patchbay (top row) to the inputs of the effect unit. These are located in of the patchbay. (Patchbay should be labeled)

4 The Aux send knobs determine the strength of the signal being sent to the effect unit. This is also dependent on the faders, because the Aux sends are "post-fader."

5 Now bring up the faders for the effect unit on the mixer.

This setup allows you to mix the "dry," unprocessed sound of the vioce with the "wet," processed sound from the effect unit. Note that you can send as many devices to the effect unit as you like by simply bringing up the Aux send knobs for those channels.

Playing Digital Audio Files

Audio 1, Audio 2, and Audio 3 are equipped to play and edit digital audio soundfiles, though each computer requires a different set up. The discussion below describes how to play

soundfiles on each computer, either through the studio monitors or through headphones.

Yamaha 01V96 Digital Mixer

Lets take an example of Digital mixer along with its specs. 24-bit/96-kHz Performance. 24 Analog & Digital Channel Inputs.

Product Description

Yamaha's flagship digital mixing consoles are the accepted standards throughout the world: the awesome PM1D for sound reinforcement, broadcast, and theater; the DM2000, DM1000, and 02R96 for sound and music production. Now the 01V96 brings you the same performance and reliability in a smaller, more affordable format that's perfect for smaller professional production studio. Still, it has a maximum 40-channel input capacity and can be cascaded for applications that require more. And, of course, 24-bit/96-kHz operation is standard. Mixer functions and effects are all inherited from the top-of-the-line DM2000, so you know you're getting the best. Prepare to be amazed at how far Yamaha digital evolution has come. If you thought that cutting-edge digital mixing and processing performance was still beyond reach, here is a very good reason to smile.

Cutting-Edge Performance, Capacity, Control & Compatibility

You simply won't find another digital console this compact and affordable that offers this much performance and flexibility. The 01V96 fits comfortably in the small-studio space - and budget while delivering sound, capacity, control, and compatibility on a par with much larger consoles.

24-bit/96-kHz Performance Takes Digital Sound to the Next Level

Digital audio technology has come a long way since the early days of 16 bits at 44.1 kHz - a format that many considered to be the reason for "harsh", "cold" sound. The entire industry is now settling on 24-bit/96-kHz digital operation for significantly superior sonic quality. The 01V96 does give you a choice - you can work at 44.1 kHz, 48 kHz, 88.2 kHz, or 96 kHz, depending on the needs of each individual project. But when you want the truly transparent, incredibly dynamic sound of 24-bits at 96 kHz, the 01V96 is ready to deliver. It even includes a comprehensive range of superb 96-kHz compatible stereo effects with 32-bit internal processing.

24 Analog & Digital Channel Inputs ... Expandable to 40

Right out of the box the 01V96 gives you 16 analog channel inputs-12 with high-performance microphone head amplifiers - and eight digital channel inputs via a built-in ADAT optical interface. The first 12 analog channels will accept microphone signals or balanced/unbalanced line-level signals, while the remaining four channels can be used either as individual balanced/unbalanced line inputs or two stereo pairs. Without going any further you're ready to handle a comprehensive mix of analog and digital inputs. When you need more, Yamaha offers a range of Mini-YGDAI expansion cards that can simply be plugged into the 01V96 expansion slot to provide additional I/O in a variety of formats: ADAT, AES/EBU, TDIF or analog.

20-Bus Configuration

The 01V96 offers a main stereo program bus, eight individual mixing buses, two solo buses, and eight auxiliary buses - a total of 20 in all. This gives you plenty of signal-routing options to adapt to just about any mixing requirements.

Built-in ADAT Optical Interface

The 01V96 comes with an industry-standard ADAT optical digital I/O interface built right in - no options necessary. ADAT "Lightpipe" optical I/O is standard on a wide range of current digital sound gear, so you can simply plug in via optical cables for 8 digital inputs and 8 digital outputs that will handle your digital signals without compromise. Additional optical I/O capacity can be added via the 01V96 expansion slot, as necessary.

Fast, Flexible Digital Patching

All available inputs, outputs, effects, and channel inserts can be assigned to any of the console's channels or outputs via the 01V96's remarkably versatile, easy-to-use digital patching system. For example, any of the effect processors can be assigned to an auxiliary bus for send-type operation, or inserted directly into any input channel as required. A direct out function also allows the signal from any of the input channels to be routed directly to any digital or analog output. The eight auxiliary buses can also be patched to anywhere in the system. Centralized control means you'll never have to run around to physically re-patch cables whenever you need to reconfigure the system, and patch setups you might want to use again can be stored in the 01V96 "patch library" for instant recall at any time.

99-Scene Memory

Complete console setups can be memorized and instantly recalled via the 01V96 SCENE MEMORY controls. Memory is provided for up to 99 scenes. In addition to recalling scenes from the panel controls you can recall them remotely via MIDI program change messages, providing a handy degree of automation capability.

Integrated DAW Control

The 01V96 has been designed to integrate tightly with leading digital audio workstations to create a complete production and mixing environment. Extensive support is provided for Digidesign's Pro Tools¢ç system as well as Steinberg's Nuendo¢ç DAW - full control of mixing and processing parameters, as well as transport/track-arming control and access to editing functions - directly from the 01V96 control surface. There's also a "General DAW" mode that provides compatibility with other workstations.

Internal Effects Fully Support 96-kHz Processing

You could use external digital effect processors with the 01V96, but what's the point when it features top-performance 24-bit/96kHz effect processors built-in? Also, you're going to compromise audio quality if you have to convert down to a lower sampling rate for effect processing - which is exactly what's going to happen if you use hardware or software processors that don't offer 24-bit/96kHz performance anywhere in your signal chain. That's why Yamaha included a comprehensive range of 96-kHz compatible stereo effects in the 01V96. You can use two effects simultaneously at 88.2/96- kHz, and up to four effects at lower sampling frequencies.

Top-quality Compression, Gating, EQ and Delay

All input channels on the 01V96 feature flexible, independent compression and gating/ducking processors for dynamics control. All bands on the 4-band parametric channel equalizers are fully sweepable from 20 Hz to 20 kHz, with bandwidth variable from 0.1 to 10 and a +/-18dB gain range for extraordinary equalization flexibility. The channel delays also go well beyond the norm, with a maximum delay of 452 milliseconds (96 kHz mode). Even the stereo bus, eight mix buses, and eight aux buses have individual compression and EQ!

Expandable Data Libraries

Setting up EQ, compression, and other parameters for a mix from scratch can be a daunting task, so Yamaha has provided an extensive selection of presets in a range of "libraries" that can simply be selected and used unmodified, or edited to suit specific requirements. Libraries are provided for effects, compression, gating, EQ, I/O patching, and channel setups. Of course, your own setups can be added to the libraries for instant recall whenever they are needed.

01V96 Cascade Link

When you really need high capacity - particularly for sound reinforcement applications - the 01V96 offers "01V96 Cascade Link" capability that allows two 01V96 units to be cascaded to create up to an 80-channel mixing system at an unbelievably affordable price!

Surround Panning

Surround is becoming an important part of modern sound production. The 01V96 features 6.1, 5.1 and 3-1 surround panning modes so you can create surround mixes without having to sacrifice features or performance in other areas.

Studio Manager Software Supplied

The 01V96 comes supplied with Yamaha's Studio Manager software application for both Macintosh and Windows platforms. Studio Manager gives you complete access to all parameters for either on-line or off-line control, and the program's visual interface makes it easy to relate on-screen controls to the corresponding console functions. The Studio Manager can also be used to manage an extensive archive of mix data.

Refined user Interface

Even the 01V96 user interface has evolved from the original 01V in many ways, now more closely resembling its big brothers - the 02R96 and DM2000 - in terms of layout and ease of operation. Overall, you'll find that the 01V96 allows analog-style hands-on operation that will contribute to smooth, efficient workflow. There are even eight user-defined keys that can be assigned to functions of your choice.

Large LCD Panel

The new 01V96 display is a high-resolution 320 x 240 dot LCD panel that provides easy visual access to all of the consoles functions and parameters. Many parameters are displayed graphically so you can see what's happening at a glance - EQ curves and compression parameters are especially "readable" in this format.

Channel Strips with Precision 100-mm Motor Faders

The 16 channel strips on the 01V96 panel provide access to the most essential operations for the corresponding channels. Depending on the currently selected layer, the channel strips will control channels 1 through 16, channels 17 through 32, or the eight AUX sends and eight buses (the "Master Layer"). Also the channel faders will function according to the settings in the FADER MODE section. In addition to a 100-millimeter motor fader, each channel strip includes a channel ON/OFF key, a SOLO key, and a SEL key that assigns the channel as the console's "selected channel". Detailed panning and EQ control for the currently selected channel is available via the SELECTED CHANNEL controls. The master STEREO fader is also a 100-mm motor type, with its own ON and SEL keys.

Layer-switching for Fast 32-channel + Aux/Bus Fader Access

One of the advantages of digital control is that it allows extraordinary power and flexibility to be packed into minimum space. The 01V96 has 17 physical 100-millimeter motor faders. The first 16 can be instantly switched to handle input channels 1 through 16, 17 through 32, or auxiliary sends 1 through 8 and buses 1 through 8, via the console's LAYER switches. There's also a ST IN layer switch that switches between the stereo 1/2 or 3/4 inputs for the stereo layer controls. Having all controls right in front of you at all times not only save space, but it also means that all operations can be carried out without having to move away from the monitoring "sweet spot".

Fader Mode

The FADER MODE keys allow the 01V96 faders to be instantaneously switched between fader and auxiliary level control. And because the faders feature fast, precise motor-drive mechanisms they immediately respond by flying to the appropriate settings for the selected mode.

Display Access

The DISPLAY ACCESS keys determine which type of data will be shown on the LCD panel - a total of 12 selectable categories. This approach minimizes the need to scroll through on-screen lists when you need access to a particular type of data.

Selected Channel Controls

The SELECTED CHANNEL controls include the hands-on panning and EQ controls for the currently selected channel, with analog-style buttons and knobs for direct, easy access to the parameters. Need to adjust the high-mid frequency a little? Just tap the HIGH MID key and turn the FREQUENCY knob until you get the sound you want.

Scene Memory

Here's where you can store all console parameters as a new scene, or instantly recall previously-stored scenes. The current scene number - 01 through 99 - is shown on the LCD panel. Additional scene memories can be managed via a computer running the supplied Studio Manager software.

User Defined Keys

These 8 keys can be assigned to control any functions you choose. You could, for example, use them to recall input patch setups, to arm MTR tracks for recording, or to handle locator functions. When the REMOTE layer is selected, the USER DEFINED KEYS are automatically assigned to Pro Tools¢ç control functions by default.

Data Entry

Large cursor, INC/DEC, and enter keys are complemented by a data entry dial that lets you spin in values quickly and easily. The data entry dial also doubles as a shuttle/scrub dial for recorder or DAW control.

Analog Input Section

Most of the 01V96 input connectors are top-mounted for easy access in any application. Inputs 1 through 12 feature high-performance head amplifiers for microphone or line input that deliver a pristine signal to the console's precision 24-bit/96-kHz A/D converters. 48- volt phantom power for condenser microphones is switchable in 4-channel groups, trim controls and pad switches facilitate optimum level matching with the source, and channel inserts make it easy to insert external analog processing gear into the pre-A/D signal path. Inputs 13 through 16 accept line-level signals singly (each input has an independent trim control) or in pairs for stereo input.

Rear Panel

The rear panel is home to balanced analog stereo and monitor outputs as well as four balanced "omni" outputs. The optical IN and OUT connectors for the 01V96's built-in ADAT interface are also located on the rear panel. There are also digital 2-track inputs and outputs featuring coaxial connectors. On-board sample rate conversion allows CD players and other digital sources connected to the digital input to be monitored or routed to an input channel without having to be synchronized to the system clock. A range of synchronization and control options are available via word clock inputs and outputs, MIDI connectors, and a USB "TO HOST" connector which can be used for computer control via the supplied Studio Monitor software. The rear panel also has an expansion slot which will accept a wide range of Yamaha mini-YGDAI expansion cards that can add up to 16 additional channels in a variety of formats.

Technical Description

Number of scene memories 99

Sampling Frequency Internal 44.1 kHz, 48 kHz, 88.2 kHz, 96 kHz

External Normal rate 44.1 kHz -10% ¢¯ 48 kHz + 6%

Double rate 88.2 kHz - 10% - 96 kHz + 6%

Signal Delay ¡Â1.6 ms CH INPUT to STEREO OUT (@Sampling frequency = 48 kHz)

¡Â0.8 ms CH INPUT to STEREO OUT (@Sampling frequency = 96 kHz)

Fader 100mm motorized x 17

Total Harmonic Distortion*

Input Gain=Min. CH INPUT to STEREO OUT

¡Â 0.005% 20 Hz to 20kHz @+14 dB into 600¥Ø

¡Â 0.01% 1kHz @+24 dB into 600¥Ø

(@Sampling frequency = 48 kHz)

¡Â 0.005% 20 Hz to 40kHz @+14 dB into 600¥Ø

¡Â 0.01% 1kHz @+24 dB into 600¥Ø

(@Sampling frequency = 96 kHz)

Frequency Response CH INPUT to STEREO OUT

0.5-1.5dB 20Hz - 20khz @+4 dB into 600¥Ø(@Sampling frequency = 48 kHz)

0.5-1.5dB 20 Hz - 40khz @+4 dB into 600¥Ø(@Sampling frequency = 96 kHz)

Dynamic Range

(maximum level to noise level) 110 dB typ. DA Converter (STEREO OUT)

105 dB typ. AD+DA (to STEREO OUT) @FS = 48 kHz

105 dB typ. AD+DA (to STEREO OUT) @FS = 96 kHz

Hum & Noise**

(20Hz-20kHz

Rs=150¥Ø

Input Gain = Max.

Input Pad = 0dB

Input Pad = 0dB

Input Sensitivity = - 60dB -128dB equivalent input noise

-86dB residual output noise STEREO OUT
STEREO OUT off
-86dB (90dB S/N) STEREO OUT

STEREO fader at nominal level and all CH INPUT faders at minimum level

-64dB (68dB S/N) STEREO OUT

STEREO fader at nominal level and all CH INPUT faders at minimum level

Maximum Voltage Gain 74dB CH INPUT (CH1-12) to STEREO OUT/OMNI (BUS) OUT

40dB CH INPUT (CH13-16) to STEREO

74dB CH INPUT (CH1-12) to OMNI (AUX) OUT (via pre input fader)

74dB CH INPUT (CH1-12) to MONITOR OUT (via STEREO BUS)

Crosstalk (@1kHz)

Input Gain=Min. 80dB adjacent input channels (CH1-12)

80dB adjacent input channels (CH13-16)

80dB input to output

Power Requirements USA 120V 90W 60 Hz

Dimensions 16.92"(W) x 21.25"(D) x 5.90"(H)

Weight 33.6 lbs

Operating free-air temperature range 10-35oC

Storage temperature range -20-60oC

* Total Harmonic Distortion is measured with a 6 dB/octave filter @80 kHz.

** Hum & Noise are measured with a 6 dB/octave filter @12.7 kHz; equivalent to a 20 kHz filter with infinite dB/octave attenuation.

LIBRARIES

Number of factory presets Number of user libraries

Effect libraries (EFFECT1-4) 44 76

Compressor libraries 36 92

Gate libraries 4 124

EQ libraries 40 160

Channel libraries 2 127

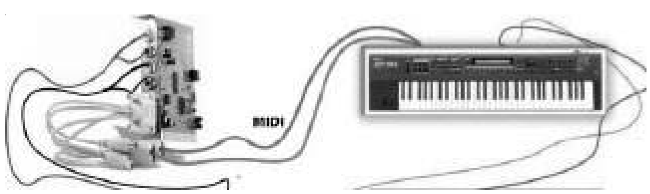
Input patch libraies 1 32

Output patch libraires 1 32

Notes :

The MIDI Sector

1. MIDI, the Musical Instrument Digital Interface is a protocol developed in 1983 by major synthesizer manufacturers to allow one synth to play another remotely. They quickly found out a computer could record and playback this data and it revolutionized the way music is produced. MIDI works by sending NOTE ON and NOTE OFF events down a midi cable as well as timing information and controller (knobs, wheels, and sliders) information. IMPORTANT: Understand that MIDI does not, cannot, and will not send an audio signal. The sound of the keyboard or module does not go down the MIDI cable, only these computer encoded "events" do. This is NOT digital audio data, its MIDI data. There's more to it, but that is the basic point you have to fully understand. We'll get to digital audio later on the page. Now lets move on with understanding MIDI.



2. MIDI Sequencers (i.e., Sonar, Logic, Cubase) work by recording these NOTE ON/OFF events along a timing grid that can be played back at any tempo. When you press Play on a sequencer, it simply sends the data back to the synth, which plays it just like a player piano. Get it? Because the Synth receives the exact same events that it sent when you played the track, it sounds exactly the same. Makes sense so far?
3. MIDI instruments (i.e., keyboards and hardware sound modules) typically have 16 channels they can send and receive on. Much like your TV, you can have a different program on each channel. The only difference is that MIDI can play all 16 channels at the same time. These channels correspond to tracks in your sequencer. On each track, you can assign a different instrument. You record each track while the previous tracks play back. Because MIDI has 16 channels, you can have 16 instruments playing together, like a 16 piece band, if you want, that is totally of your own design.
4. There are 16 channels for every MIDI PORT on your computer. Your computer's soundcard are where the ports are, and when you need more you can buy a MIDI interface to add more, but to get started we'll just look at the typical soundcard so we can get you running.

There are INTERNAL MIDI ports and EXTERNAL MIDI ports. The internal port(s) go to a MIDI synthesizer that is a physical sound chip or processor "on" your soundcard. These are sometimes called "wavetables" "soundfont devices". You can see if your computer has them already installed-it probably does.In win98se, look in the Windows control Panel. Double click Multimedia, go to the MIDI Tab. There they are, all your ports. The EXTERNAL MIDI port is there too. The external port goes out your soundcards JOYSTICK port. You need a cable that connects to it on one end and gives you 2 MIDI connectors on the other end, a MIDI IN and OUT.

This is what the cable looks like.

You need it because that's how you connect a keyboard. Any keyboard with a MIDI out will work, even if it sounds like crap. Remember, you only need it to trigger the NOTE ON NOTE OFF event to the sequencer, which will send them to the soundcard's synth(s), which will send the sound to the speakers. Get it? Though the playing happened at the keyboard, the sound was triggered at the soundcard. You can even use an old CASIO with the speakers ripped out as long as it has a MIDI out.

But if you have a nice keyboard, The external MIDI out port can connect back to your keyboard's MIDI IN and play 16 channels of voices if the keyboard is "multi-timbral" (meaning it can sound many voices at once). Some are not. Old keyboards, like a DX7, only respond to 1 channel unless you hot rod it. Most modern professional keyboards are multi timbral. You can usually set the keyboard to respond to only one channel if you want or to only channels 1,2,3,7,8, for example, or to all 16. Turning off channels allows you to daisy chain more keyboards or modules by cabling up the 2nd machine from the MIDI THRU on the 1st machine (which is an exact copy of the data coming in the MIDI IN) to the MIDI IN of the second machine. It is possible to have one MIDI port to control 16 different keyboards if you want it too! Usually, if your rig gets this large you will buy a MIDI interface with 4-8 additional ports so you can play 7 channels of your Triton with 10 channels from your Proteus 2000, a couple for your Electribe, one for each effects box, another for a drum machine, then 16 for the sampler, 16 more for your digital mixer...ooops sorry, I forgot we are just getting started.

Basic MIDI setup using only the Soundcard Port

Keyboard MIDI OUT————>Computer MIDI IN(essential)

Computer MIDI OUT————>Keyboard MIDI IN(if you are only using your soundcard's "onboard" synth this connection is not required. If you want the computer to play your keyboard, or if you want to connect a 2nd module it is required)

Keyboard MIDI THRU ———>Keyboard or Module #2 MIDI IN (optional)

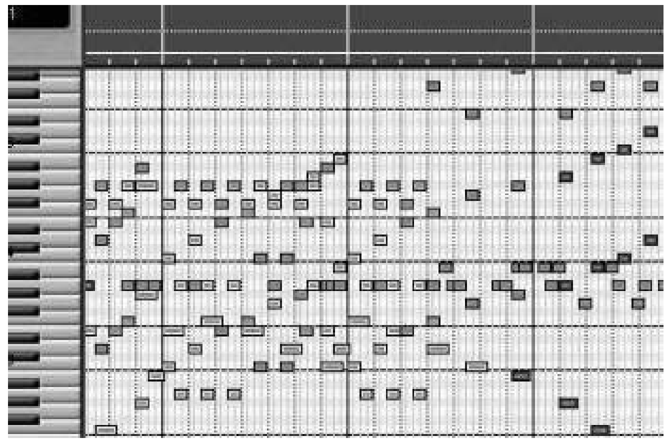
Keyboard or Module #2 MIDI THRU ——> Keyboard or Module #3 MIDI IN (optional)

MIDI In/Out/Thru Brain-work. Study the diagram above. Lets say you have 2 machines connected on the same port. If, for example, you are sending a track down channel 4 from the computer and have the 1st machine is turned OFF on channel 4, it will ignore all that data. Because all of the data coming in to the 1st 'board is also going out the MIDI THRU, the second machine will play the track, if you have channel 4 turned ON. So you ask, what happens if channel 4 is set to ON for both devices? Bingo dude! You got it, they both play. Now if each synth was on a different PORT, would both synths make a sound. Nope! Each port is totally discrete. The data on Port A does not go down Port B. Remember, each port has 16 channels. The Port is the parent; the channel is the child. So if you get a midi interface that has 8 ports, how many channels do you have? Yes, you math genius, 128. Does that mean you can connect 128 keyboards to ONE 8 port midi interface. YES! But don't call me to wire it ok? Most people put each synth on it's own dedicated port so they don't have to turn channels on and off all the time.

General MIDI. Most soundcards follow a specification called General MIDI or GM. This specifies which programs are available for the 16 channels. Drums are always on channel 10. The rest you define how you like. The basic GM set consists of 128 programs which always have the same program number. Piano is always program 1. 34 is always Picked Bass, 67 is always Baritone Sax. Because there is this common reference, any file you make with GM can be played on any computer or synth that follows the GM spec. That's what a MIDIFILE is. Its a sequence made up of tracks filled with timing information and NOTE ON/OFF information. A "MIDI" or Midifile has no audio data again. It only has note and controller and time data. But because manufacturers follow this standard spec, it sounds the same, or similar, on different soundcards. Its possible to do a great composition with GM, but the limits are in the limited palette of 128 sounds. But its an excellent way to get started. After a while you might want better, more varied, or more specific sounds-that's when you know its time to move on to standalone modules that give you often thousands of sounds, or focus on specific genre's like dance sounds, orchestral, hip hop, world fusion, R&B, etc.

The Computer and MIDI, a great match. Because MIDI data is compact in size its easy for a computer to manipulate it. You can easily have 50 tracks of midi instruments running in a song if you have enough ports and channels. In the sequencer, midi tracks can be copied, looped, transposed, doubled, stretched and edited to fine detail. You hit a wrong note? No problem, edit it to the proper value. You have a great idea but your timing is off? No problem. Simply apply a quantize template and your notes will lock to a grid or a groove. You don't have to be a good keyboard player to make excellent sounding compositions. Simply correct the tracks till they sing for you. From the sequencer, every element of the MIDI stream can be modified. You can call up a track and edit it on a musical stave like the "old" composers used to or on hi-tech grids, or in a list of events.

Why Mess with MIDI? : OK, all you homeboys and dudes who think that MIDI is "old" and not as cool as hot audio loops and beats take note: Most commercial loops are built with MIDI data controlling digital samplers. The true and authentic tweak-head and groovemaster either uses MIDI to make their loops or has such impeccable timing they just record it cold. Of course you could buy and use store-bought loops, but you'd be restricted to using other people's sounds and stuff. Get a sampler, makes some noises, edit to perfection with MIDI, grooving, quantizing, transposing and you are 100% originally you! Record the output and edit the sample and



you just made your own audio loop. That's how the true tweaks do it.

A view of Logic's Matrix Editor. All sequencer's have a similar editor that allows for exacting placement of notes along a timing grid. Extremely elaborate sequences are possible because the sequencer only sends note triggers to your synth, not audio data.

Don't think for a minute that MIDI is not a serious musical tool. Nearly all TV/film scores today make use of MIDI. Many top 40 songs are made with sequencers in home studios and converted to professional audio. MIDI is the core building block on which to build a totally original musical masterpiece on a computer.

Harmonizer

What is a Harmonizer?

A Harmonizer is a device that samples incoming audio material and automatically creates several time stretched versions at different pitches and combines them all to an audio output. All of this happens on the fly in real time. You hear this device all the time on the radio. Whenever you hear ultra tight vocal harmonies that sound "too perfect" you are listening to a harmonizer at work. There's also a lot of times where it is at work where it's not obvious. Harmonizers can be used to thicken, detune and add effects to instruments and can, for the creative, make very strange effects out of any audio material.

Why get a Harmonizer?

People tend to think that a harmonizer is just a vocal processor. The truth is, and remember this, any piece of gear can be used for unintended functions and the harmonizer is one of those devices that can do lots of things. In addition to vocals, harmonizers can be applied to guitar, keyboards, even drums (you can make massive hits with detuning), and incredibly thick ambient pads (in fact many a fat R&B string pad was made with a harmonizer). The process here is simple. Play or Sing in one note; 1-5 notes come out, depending on which model you have and how you have set it up. Many harmonizers also come with effects, such as delay and reverb, and some even have Mic preamps. So think for a minute. Instead of pouring out the cashola for a hardware reverb and delay box (which software plugins can do pretty well now), why not invest in a box plugins can't do, a box that harmonizes, and has as a bonus reverb and delays and a pre. Your browser does not support inline frames or is currently configured not to display inline frames.

Using a Harmonizer

The good thing about many of today's Harmonizers are, yep, presets. Even if you know little about music you can just connect your mic and spin the dial till you find something really cool. Ideally, though, you need to know the key (root note and scale) your music is in, like the key of C#major, A minor, or Beethoven's favorite, E flat minor...but I digress. Once you set the key, the harmonizer will then figure out what note you are singing, or playing and create harmonies that are consonant with the key. So If I punch in the key of A minor and sing an "A", it could produce harmonies at C and E.

You can set up your presets in many ways. You can have all voices at unison, which makes for a really thick vocal sound, or you can specify intervals, such as voice one will be up 2 semitones, voice two will be up 7 semitones, voice 3 will be down 5 semitones. Or you can specify the chord of the moment and the harmonizer will create voices based on the chord. C maj7 any of you cool breeze jazz dudes? In no time you'll be on the beach sipping from a coconut shell.

Using MIDI with your harmonizer

You can also set up some harmonizers so that MIDI tells it what the current chord is. Here the harmonies that are created will follow the chords you play on your MIDI keyboard. This is not as hard as it sounds. It's the same principal that a vocoder uses. Just make a midi track that contains nothing but simple block chords every bar or two and route that to the harmonizer. Then step up to the mic and be amazed as your harmonies track the song.

Types of Effects

The main effect, naturally, is the harmonies. The typical use is to record your chorus tracks through the harmonizer and lay this track under the main vocal.

One ability of my unit, the TC VoiceWorks, is "gender shifting" as well as pitch shifting. I can make my voice really low or really high. Of course it can do Darth Vader and munchkins effects, but that's all old hat. With the gender features you can add more maleness or femaleness to your voice. Ever wanted to sound like a chick, you macho dudes? Sure you have. Ok forget that then, turn yourself into a 12 foot 600 lb blues singer with a really deep voice.

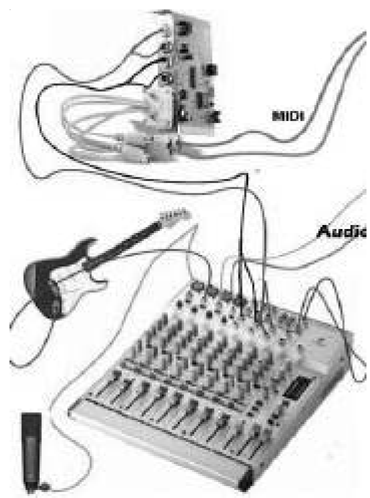
Another effect easily achieved is thickening, doubling (or tripling and quadrupling), detuning. This not only sounds good on vocals, but one other instruments too.

Many harmonizers have effects like delay and reverb. If you wanted to you could just use the box for these and bypass all the rest, so if you are working live there won't be any reverb hiccup as a preset changes and you can hit a footswitch and go from harmonies to straight vocals.

Setting up a Harmonizer

There are many kinds of harmonizers out there and you have to pay attention to the different ins and out so you get one that will fit in with your existing gear. For example, if you get one without a Mic preamp or want to use a different preamp you need to make sure the unit has line ins and outs at the with the connectors you need. Keep in mind some may have only one set of XLRs, TRS, TS or RCA connections.

For use with an analog mixer, you would put the harmonizer on inserts, or in a typical sends/return effects configuration. Using sends and returns may be superior if your sends can be set to "prefader" where you can remove the original source and just hear the effected signal. With inserts, there may be a need to devote an extra channel. Harmonizers may only have a single mono input and the output is usually stereo, so you need an extra channel.



Setting up your Audio

How to Hookup a Soundcard

The Audio Sector

The Audio section of our budget studio is found right on the Soundcard with its Mic IN, Line IN and Line Out jacks. On your soundcard is a microprocessor called the digital audio converter (DAC) which contains 2 parts. 1) Analog to digital conversion (a/d) and 2) Digital to analog (d/a) conversion. Some call the DAC an ad/da converter. The Audio Line in goes to the a/d then to the CPU, memory, and storage. The stored digital audio (called a .WAV file) goes back to the memory, CPU, then out the d/a to the Line Outs. Simple enough.

Connecting Microphones and guitars and other sources. A good soundcard will let you connect these to the soundcard inputs. A cheap sound card that only has MIC and LINE ins might pose a small problem. You'll have to use a cheap mic to connect to the little 1/8 inch phone jack, and you'll need a little adapter to connect the guitar to the MIC in. (be careful to keep you guitar at very low volume). If you have a direct box or a pedal, connect that to the LINE IN. A better solution here is to get a small mixer that lets you connect the mixer output to the LINE IN jacks on the soundcard. The Mixer will let you use better microphones with XLR jacks and will have better clearer sounding preamps. Also a Mixer will be able to boost the guitar signal just right, making it easy to use pedals and all the cool guitar gadgets you might have.

How MIDI and AUDIO work together. The digital audio/ MIDI sequencer allows you to record the analog output of your synths as .wav files. You simply connect the audio outputs of the synth(s) to your soundcard's audio line inputs. The audio goes through the DAC to computer memory and hard disk. This type of data is correctly called digital audio data, where each second of sound is divided into 44,100 slices! So you should be digging that audio data is thousands of times larger than midi data, right? It is.

Once your synth tracks are recorded as .wav files, then you can process these perfectly synced wave files with "plugins" or effects. Or you can keep your synth tracks in the MIDI domain (where they are always freely editable) and add your voice, or guitars, or whatever else you want as audio tracks. (or you can record everything as audio then "mixdown" all the audio tracks to a single stereo .wav file. Getting the idea? In the sequencer, you can have MIDI and Audio tracks side by side. A good MIDI/Audio sequencer gives you the software you need to make a complete piece of music. You can sing or speak over massive orchestras, hip-hop beats, trance tapestries, audio loops, sampled snippets ripped from your music collection, whatever you can get in there. When it all sounds the way you like it you MIXDOWN to your format of choice. You can do this internally in the sequencer (if you recorded your MIDI tracks as Audio tracks). This process is called "bouncing", where several sequencer audio tracks are combined to a single stereo track. Or if you are using external keyboards and synths that you are not recording as audio tracks, you can blend them with the soundcard output in your mixer and route that to a tape deck, DAT, cd recorder or even to the inputs of a second soundcard. You should be getting an idea of how flexible a MIDI/Audio sequencer is, and how there is not a necessarily "best" way to go about making a final master. If it sounds good and you can record it-that's the bottom line.
An alternative to the computer midi and audio sequencer is a dedicated standalone multi-track recorder. You can read about those here on this site and learn how they can integrate with computers, if you want them to.

All About Plugins

The audio goes "through" the plugin depends on where they are placed in your mixer. There are basically 3 types of plugins.

1. Track based
2. Bus based.
3. Mix based

A track based plugin is an insert that affects only that track. Plugins that should be used on every track individually are compressors (and on tracks that must be as loud as possible, a limiter). If necessary, you may want to add a noise gate, eq, and delay, chorusing or flanging to a track. Vocal tracks may benefit from exciters, de-essers and tuning plugins. Guitars should have their processing at the track level.

A bus based plugin is one that you want to apply to more than one track, but not the entire mix. The typical candidates at this level are reverbs. You may need more than one. Having different tracks go through the same reverb is tried and true studio technique to give the impression the performers are actually playing in the same room. It's also critical for setting the front/back depth perspective of your soundstage. For creative tweaking, there's no end to the things you can do with sends and returns in the virtual environment of the sequencer.

A Mix or Master based plugin goes as an insert to your master out-its the last step in the chain before the final print and all of the audio goes through it. These plugins are the ones that are most critical to the "overall sound" of the final product. And it is here where we can solve many of the problems listed in our above exercise.

Plugin formats. There are several plugin formats which include VST, DirectX, RTAS, TDM, AudioSuite and MAS. We are only going to talk about the 1st two types, which are the most popular and found on many PC sequencers. First, you should check to see which format plugins your sequencer supports. Cubase, for example, supports VST plugins. Cakewalk, with its reliance on the Windows MME, relies on Direct X. Logic supports both VST and Direct X. Sometimes you can, with the addition of a "wrapper" plugin, use DX plugins in a VST based application and vice versa. But you should always check that out first. The cool thing about plugins is that they work in all compatible audio applications you have on your system. For example, if you get a quality DX plugin for Logic, you can also use it in ACID, SoundForge, Vegas. You can use the VST plugins that come with Cubase in other VST applications like Wavelab and even in Logic. You simply copy the plugin folder called "VstPlugins" from your Cubase directory to your Logic directory. Logic's plugins, however, cannot be moved to Cubase as they are part of logic itself which uses a proprietary format. Yet regardless of the format of the plugin, they are all used in the same ways, on tracks, busses, and master outs.

Free plugins A very cool thing going on is the availability of free, good sounding plugins on the internet. This is particularly the case with VST plugins. At the end of the article I'll post some links to the places where the good ones are. Yet free plugins should be used with caution. They may crash your sequencer or make it unstable. I personally will not use them. It's just not worth it to add a questionable device to an application that has to work in real time.

The best plugins The best plugins are not cheap. They don't sound cheap either. One secret the people in large expensive studios have over you is they too use plugins, sometimes the same ones you already have in your sequencer. Yet you will also find the top plugins in their arsenal. The quality of a high end plugin can indeed match the quality of its hardware counterpart.

Studio Monitors

So lets talk about studio monitors. A thorny subject to be sure, with as many opinions as is believable. As with microphones and sequencers, the debate often turns into flame-fests. Yet there has been some progress in our understanding of monitors. Nearly everyone seems to agree that one needs a "near field" monitor to accurately assess the music they are creating. A near field monitor is one that you keep pretty close to you, typically between 3-5 feet. The idea here is that the music we hear is dramatically altered by room acoustics. Go try a little experiment. Take any speaker and put it in the corner of a room. You will hear bass resonating through the room that does not exist when the speaker is in the middle of the room. The near field monitor is the solution to minimize as much as possible the interaction of sound with the room. Because they are positioned close to you, you hear the sound directly. Because they are so close you need less volume, so less of the sound travels to walls and corners and bounces back at you. Unlike a hi-fi speaker, which is designed, hopefully, to make all audio material sound pleasing to the ear, the studio monitor has as it's main objective to paint an accurate audio image of the material, with no unnatural emphasis or de-emphasis of particular frequencies. This is what it means when a monitor is said to be "flat". and "uncolored" or "transparent". That's the theory at least. Reality sometimes paints a different picture. And this is where the arguing typically begins.

Should a Monitor be Flat? The Story of Studio A vs. Studio B

Should a monitor really be flat? Some say no. They say buy a monitor that you like to listen to, that your favorite music sounds great on, then match your efforts in the mix to produce that sound you love. Others, however, say if certain frequencies are over represented, you will under-represent them in your mix. Huh? Ok here we go. Studio Jock A has a bass-shy set of nearfields. S/he like the bass strong and thumpin' So they make those little 6 inch woofs pop to the beat. When Jock A goes to Jock B's studio, equipped with subwoofers, Jock A realizes his or her mix is nothing but one giant wash of low frequency mud. Ah, you lost that one.

Now Jock B in his premium subwoofer enhanced bass studio smiles condescendingly at Jock A, tweaks a mix to pure sonic bass perfection. The Bass is just blowin' everyone's chest away like a heart attack, but the rest of the mix appears strong and balanced in all frequency regions. Jock B smugly walks a CD to Jock A's studio and loads his latest mix. To their astonishment, the bass is gone, not even present except a little wimpy tap-tap-tap of a cardboard sounding kik drum. Jock B sputters, having lost his composure, "W-w-w.here did my bass go!??" Ahem, the "truth" is that because Jock B heard the bass so well in his studio, he did not put enough into the mix, and the little "flat" monitors at studio A didn't give a flying cone excursion.

So whatever your monitors are good at is exactly what your mix will be bad at. So then, what is the "Truth" in monitoring? Who has the Truth? The "truth" really has less to do with the monitor itself and more to do with the experience of the producer. The truth is that if your music is successful, people will listen to it in the car, on boom boxes, on their walkman, on mom's kitchen radio with the 3 inch speaker, in living rooms with surround systems, in listening booths a cd stores, on TV, and every once in a while, maybe 1 out of every 100 listens, someone will hear it on good speakers, and maybe, if you are lucky 1 out of 1000 on studio monitors. The real truth is in the understanding of how your mix on your monitors translates to other listening conditions. That is, you have to really "know" your monitors. The main thing is to get a set you are comfortable with, that you can listen to all day. Your ears will actually "learn" the monitor. As you check your mix on other systems, you will learn about your systems deficiencies and compensate. The truth is not a quality of the object, but a quality of ear, mind and experience. "It's in your head".

So what Makes a Good Monitor?

So, then, what makes a good monitor, other than the sound it produces? A good monitor is rugged, and can handle peaks, overloads, feedback mistakes and come back ready for more. It's funny. Someone started his sound development business on hi-end 3-way audiophile speakers, which worked great for a year. But with the constant tweaking of sound experiments in sub bass regions, the woofers just popped right out of their cones with a 360 degree rip around the cone. Hi-Fi speakers are not made to take abuse above typical programmed material. Sure you can use them, just don't use them as the main monitors.

A good monitor does not artificially exaggerate frequencies. You do not want a speaker that sounds like a "disco smile". That's where the bass and the treble are boosted and the mids are cut. They call it a "smile" because that's how it looks on a graphic equalizer. Lots of people really like that sound. If you like that sound, mix up a nice smile for your monitors. Then it might actually translate on other systems. But if you speakers make that automatically, you mix will be shown lacking in bass and high transients. Using that principle was the secret behind the Yamaha NS-10s, the most horrible sounding speaker ever made. On an NS10 there was no bass, no high end, just screeching awful sounding peaky mids. People who referenced on them had to boost the bass massively and cut the mids. The resultant mix? Yep, the disco smile. It made hit after hit and people thought they were magic. If you could make the shrill box sound passable, it would sound great everywhere else.

So which monitors are the best? All the monitors you see are workable solutions. So how do you know if your ears like them? You have to listen of course, but listen "comparatively". A suggestion is taking along your absolute most known CD-a mix you know intimately and have head a few million times, like "Stairway to Heaven". You "know" how it should sound. It's in your bones. If you hear stuff you have never heard before, give that monitor a point. Now listen for deficiencies in the bass. Is the voice coil working really hard to produce very little bass? Is there a certain bass resonance, a sound of air rushing out the ports? Is the high end smooth? Or is it harsh and grainy. Is the box squawking at mid range frequencies? These are all “box deficiencies”. The more speakers you listen to, the more you will hear the problems. Then it dawns on you, you can find a problem with every enclosure and know there is no perfect monitor. So you look at the higher end and you find that the better priced monitors typically exhibit fewer objectionable problems. The end is always a compromise with what you like and what you want to spend.

Summing Up

Set the monitors, placing them exactly where your ears want them, the learning process begins. As you gain experience attending to your audio, you will learn to improve your mixes. You’ll know your monitor’s “blind spots” and will learn to check on other systems. The more you do that, the more your monitors become a true reference. I hope I have shown that reference is not a quality of a box, but a quality of your attention and experience.

What is the Difference between Active Monitors and Regular Speakers?

Active monitors have have a power amp built right into them. Usually, these are mounted on the back. “Regular speakers” which are also called passive monitors, require an amplifier to drive them. The idea behind active monitors is to match them to an amp which is ideal to drive them. This takes some of the guesswork out of buying a monitoring system. Also, since active monitors are connected directly to your mixing board or audio interface, they eliminate speaker wires and hence another source of potential trouble.

I like my Bass! Do I need a subwoofer?

If you really like bass you need a subwoofer so you can actually hear what bass you are creating. With small speakers you are NOT going to hear the full bass tone and you won’t know if you hit a low C or a C#. But the people with subs will know. However, be advised to keep your subwoofer at a very low level when mixing, because, as I point out in the article, you run a risk of creating a bass-shy mix.

Subwoofer Types

Subwoofers are becoming more and more crucial to the home theater experience. When you go to the movie theater, you marvel not only at the images projected on the screen, but the sounds emanating around you. What really grabs you, though, is the sound you actually feel; the deep bass that shakes you up and gets you right in the gut.

A specialized speaker, known as a subwoofer, is responsible for this experience. The subwoofer is designed only to reproduce the lowest of audible frequencies.

With the popularity of sound systems resulting in specialized speakers for center channel dialogue, main soundtracks, and surround effects, the need for a speaker to reproduce just the deep bass portion of a movie soundtrack is all the more important.

Although these subwoofers are not quite as “thunderous” as the subwoofers employed at the local movie theater, these unique loudspeakers can still shake the studio down Subwoofers come in two basic types, Passive and Powered.

Passive Subwoofers

Passive subwoofers are powered by an external amplifier, in the same fashion as other speakers in your system. The important consideration here is that since extreme bass needs more power to reproduce low frequency sounds, your amplifier or receiver needs to be able to output enough power to sustain bass effects in the subwoofer without draining the amp. How much power depends on the requirements of the speaker and the size of the room (and how much bass you can stomach!).

Powered Subwoofers

To solve the problem of inadequate power or other characteristics that may be lacking in a receiver or amplifier, powered subwoofers are self-contained speaker/amplifier configurations in which the characteristics of the amplifier and sub woofer are optimally matched.

As a side benefit, all a powered subwoofer needs is a line output from an amplifier. This arrangement takes a lot of the power load away from the amp/receiver and allows the amp/ receiver to power the mid-range and tweeters more easily.

Additional Subwoofer Characteristics

Additional subwoofer design variations include Front-firing , and Down-firing, and the use of Ports or Passive Radiators. Front-firing subwoofers employ a speaker mounted so that it radiates the sound from the side or front of the subwoofer enclosure. Down-firing subwoofers employ a speaker that is mounted so that it radiates downward, towards the floor. In addition, some enclosures employ an additional port, which forces out more air, increasing bass response in a more efficient manner than sealed enclosures. Another type of enclosure utilizes a Passive Radiator in addition to the speaker, instead of a port, to increase efficiency and preciseness. Passive radiators can either be speakers with the voice coil removed, or a flat diaphram.

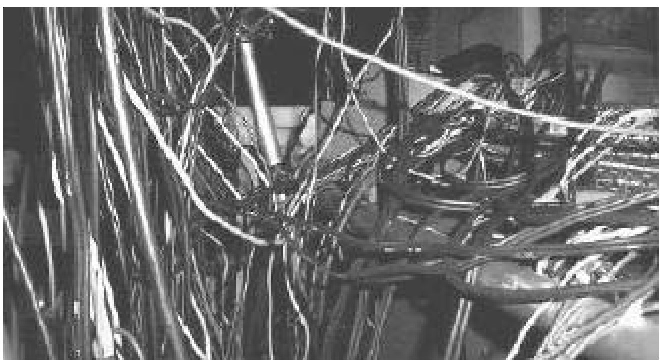
Subwoofers - Crossovers

Typically, a good subwoofer has a “crossover” frequency of about 100hz. The crossover is an electronic circuit that routes all frequencies below that point to the subwoofer; all frequencies above that point are reproduced the main, center, and surround speakers. Gone is the need for those large 3-Way speaker systems with 12" or 15" woofers. Smaller satellite speakers, optimized for mid-and-high frequencies, take up much less space.

In addition, since the deep-bass frequencies reproduced by the subwoofers are non-directional (as frequencies that are at or below the threshold of hearing). It is very difficult for our ears to actually pin-point the direction in which the sound is coming. That is why we can only sense that an earthquake seems to be all around us, rather from coming from a particular direction.
As a result, the subwoofer can be placed anywhere in the room, however, optimum results depend on room size, floor type, furnishings, and wall construction. Typically, best placement for a subwoofer is in the front of the room, just to the left or right of the main speakers.

Cables

Putting together a requires a mind that can visualize, in a second, the signal flow of the entire studio from every remote piece though interfaces, patch bays, mixers, you computer and

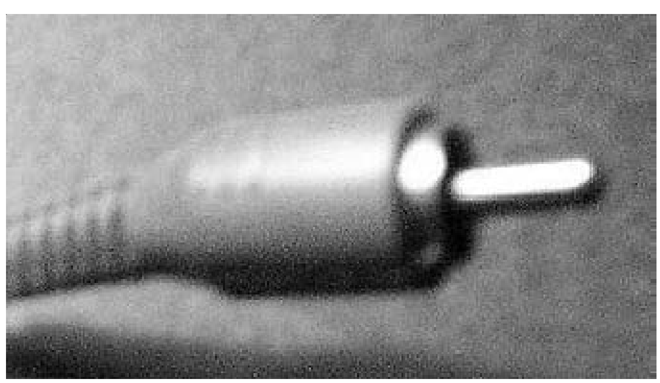


then the seemingly thousands of virtual cables and switches inside each computer application.

The glossy ads in the trade magazines always show gear in it's ideal pristine state, shiny, new, and never with cables connected. Perhaps this is a psychological point, to make the prospective buyer get a sense of how nice it would be to have this box sitting there. The real world, however, is a tangle of issues for installing almost anything. Most of these entanglements are cable related. With each piece, the living sea of cables behind tables and racks get denser and more knotty clusters develop, making the act of even tracing a cable from origin to destination a matter of great intricacy.

Large Rig Blues

You buy a midi module to add some sounds. As soon as it's out of the box, you go digging for cables and a place to put it. Lets see, oh shoot, my MIDI interface is maxed out, now I have to go through a MIDI thru. You look and the closest usable thru is 12 feet away and you have one MIDI cable left and its 8 feet long. But! You have a 15 footer somewhere else where it only need 6 feet and you decide, let's swap them out. But as you peer behind your midi interfaces the cables are so tightly packed, flowing over and under each other... you can't read the numbers on the back. You pull out 3 cables only to realize you disconnected the ins when you were looking for an out. You find the 12 footer at last, but can't pull it out unless you remove 6 adjacent cables. Trouble. Now you have 9 cables out and you know you this is going to take a while and yep, at 5am there you are troubleshooting the MIDI interface, trying to get your sequencer working again. Moral of the story: Use different colored cables, never unplug more than one cable at a time. If you have a choice between buying short or long cables, go longer. Of course, when you purchased your new MIDI module you were not thinking whether you had any mixer channels open. Who has mixer channels going unused? No one. But that's another problem. Patch bay city here we come. The more cables you have the harder they are to control. You think, oh, I need a patchbay to control them all. But the



patchbay itself will generate even more cables. Then you think, I got it, I'll use cable ties. But these to have problems of making sure all the cables in the bunch have the right length,

which again leads to more cables to get all the lengths even.

You might refrain from getting more gear so as not to upset the fine cable job you did.

Common Analog Cables

The XLR cable (3 prong) This is the common balanced microphone cable. May also be used as an AES/EBU cable or a balanced line level cable.

The TRS "Tip-ring-sleeve" cable (3-conductor "stereo" cable) This is a balanced cable just like the XLR above, it just has different connectors.

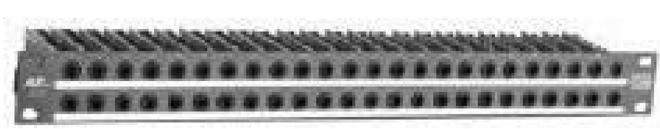
The TS "Tip-sleeve" cable (2 conductor "phone jack") This is an unbalanced cable

The RCA Cable ("home stereo" cable) Because the RCA only has 2 conductors, it is also unbalanced.

A variation on this is the Soundcard 1/8" stereo plug to dual RCA

The MIDI Cable (5 pin DIN cable)

The Soundcard Joystick to MIDI cable looks like this



The Insert Cable (stereo to mono "Y" cable). Insert jacks on mixers are not balanced. Basically, on the "stereo" end, it carries both the input and the output of the mixer channel with a common ground. These cables allow you to insert a device in the channel's path, like a compressor or EQ or to carry the signal to a patchbay where devices can be conveniently patched in.

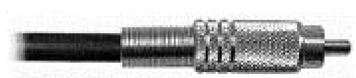
The Elco Cable (56 pin connector to 16 TRS) For professional multi-track recorders

What is a Snake? A snake is a bunch of cables bundled together in one casing. This helps keep the cable jungle a little more under control by only having one thick cable to step on or trip over, rather than 8 or 16. High quality snakes for XLR cables are expensive and include a "stage box" where the mics and other instruments are connected. Multi track snakes can be found in many types from TRS to TRS, to TS to TS, RCA to

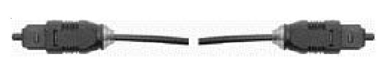
TS, and TRS to TS (an insert snake). These come in varying qualities. One problem with inexpensive snakes is that one of the 8 cables may go bad and you are stuck with loose ends hanging out.

How Do I Connect a PatchBay?

Here's the standard normal procedure. Connect an insert cable in the insert jack of the mixer. The output of the Mixer goes in the bottom rear of the bay and goes out the top rear of the bay back to the Mixer. If nothing is connected to the front jacks of the patchbay, this signal will just pass through and back. To take the output of the channel from the front of the bay, you



insert a cable in the bottom front jack. To patch another signal into the mixer to replace the existing signal, you patch a cable into the upper front jack.



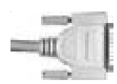
Common Digital Cables

Q: What does all this digital mumbo-jumbo like S/PDIF, AES/EBU, Coaxial and Optical really mean? What's TOSLINK?

Stereo (2 channel) digital cables

A: These are all different methods of sending 2-channel (stereo) digital audio data down a cable to another device. Note, this is DATA, not an analog signal. The data bypasses all analog circuitry to make the perfect clone of the original data. AES/EBU requires a cable with microphone (XLR) connectors. This method is used a lot with pro gear. Common lengths for these pathways is from 3-15 feet. Can you use a Mic cable as an AES/EBU cable? Well, they say it needs special shielding, but go ahead and try. It works.

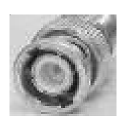
S/PDIF: (Stands for the Sony/Phillips Digital Interface) There are two types: Coaxial and Optical. So when some says they are connecting via S/PDIF, you should then ask, Coax or Optical? Coaxial requires a cable with RCA connectors. They look just like common stereo line plugs. (in fact, sometimes you can get a high quality stereo cable to work if the run is short). The limit on length is 6 meters or 19.8 feet which is fine for the home studio.



Optical is another flavor of S/PDIF and has the square connectors called TOSLINK connectors. This format allows for very long cable runs of 15 meters (50 feet) When you buy gear you want to make sure your digital devices follow the same standard or you will have to buy a converter box. You cannot run a coax into an optical jack with an adapter, the signal needs to be converted.

Multi-Channel Digital Cables

The three main methods are Roland's RBUS, Tascam's TDIF or Alesis ADAT Optical. These are all multichannel digital data piping schemes. They differ from S/PDIF and AES/EBU, which are other digital data transfer protocols, because they send up to 8 channels while the latter only send a stereo channel. R-BUS and TDIF look the same as a 25 pin connector, like a large serial port cable . ADAT light pipe cables look just like TOSLINK (optical) cables, but they are not. These carry 8 channels of digital audio.



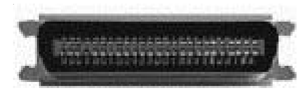
TDIF Cable

Sync Cables

ADAT Sync. This is a 9 pin D-sub cable that look like a small serial cable connector. It sends MTC (MIDI Time Code) and other synchronization signals. Note: Many devices can send and receive MTC on MIDI cables.

Word Clock Cables. These are 75 OHM cables (like TV cable) but have the "press and twist" BNC connector which you may have seen on the back of your computer monitor.

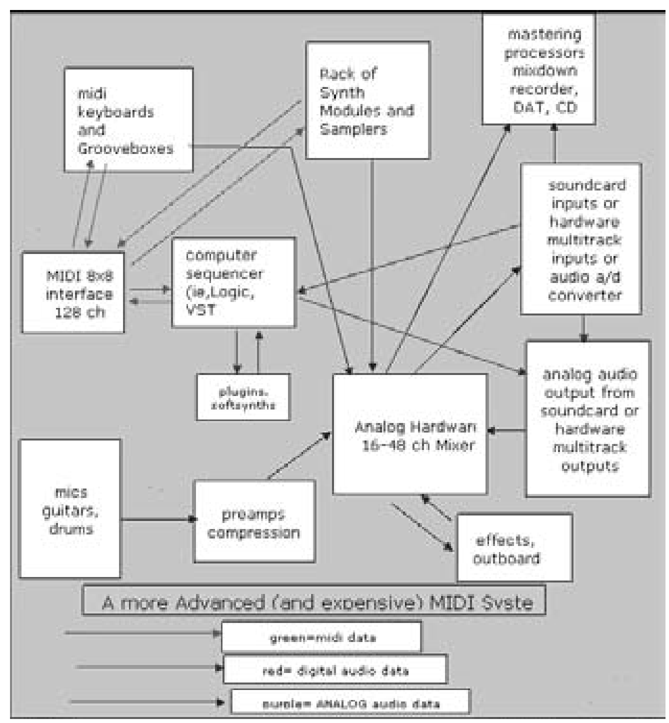
Computer Interface Cables



Firewire cables are increasingly being used to connect audio interfaces as well as Camcorders and digital video equipment.

USB (Universal Serial Bus) cables connect MIDI interfaces and some audio interfaces as well.

SCSI (Small Computer Systems Interface) Cables come in many types. Sampler by E-Mu use the "old" 50 pin Centronic connectors. Don't confuse these with the 50 pin "half pitch"



SCSI III connectors which are smaller. Inside the computer or sampler you will typically find 50 pin internal SCSI cables.

Other scsi devices may use the DB25 "Mac" connector which looks identical from the outside to a 25 pin serial cable. These are also called SCSI II connectors.

There are also DB68 pin SCSI III connectors. These have 68 pins and are common on SCSI hard drives and SCSI cards. So what if you have a DB50 SCSI connector on your card and need to connect a 50 pin Centronics connector? You got it, SCSI adpater City, here you come.

The Signal Flow of a Recording Studio

ADding Midi and Digital Audio Software an Hardware to Professionalize your Sound and Capabilities

A More Advanced Recording Studio Setup

Don't let this scare you. This is a more no-compromise system. As you see we have dedicated external synths (where you can add any midi module like the old 80's analogs, proteus, Roland JVs, Triton racks, hardware samplers like the emu e6400, akai s5000, etc) and a dedicated hardware mixer which takes all the audio outs and mixes them. Note that the mixer goes back and forth to the sequencer's audio tracks through the soundcard or (for purists) outboard analog/digital converters. We have dedicated outboard effects units and compressors and preamps. Note the pic has everything mixed to DAT, but you don't have to do it that way. Route it back to the computer and make a wave file, or send the wires to a cd recorder.

The Software Revolution

What you see in the pic below is a typical hardware/software setup. With today's fast computers and with all the available music software, it is now possible to run a software only studio. Take out the rack of synth modules and hardware samplers, take out the analog mixer and take out the mixdown recorder. The computer software revolution has replaced these items with software equivalents. Does that mean that the hardware devices are obsolete? No! Far from it, actually. As you get more serious with your recording studio you may feel the need to have hardware synths and samplers and other dedicated hardware devices. The people who work with music gear everyday know that hardware is at a high state of the art, it's easier as it does not stress the computer, has more up-time and finally, hardware devices sound excellent. It's almost funny that many of the top studios jocks want nothing to do with software at all, particularly Windows software, though that has changed a bit in the last few years.

So where does that leave you? If you are just starting out, hook up your soundcard, or get an audio interface with a breakout box that has a mic preamp on it, add a keyboard, a mic, plug in your guitar. You usually can't use the little mic preamp on most soundcards unless it is noted to be particularly good. (The new Audigy series look promising). Then get a software sequencer and hook it all up. Mess around with general midi for a month and then see if you are still serious. If you are, you might want to expand your soundmaking aresenal with soft synths and sampler or maybe a hardware synth.

If you like the hardware approach it's time to start thinking of getting a mixer. If you are wanting to invite friends over to record, or even record bands, then you probably will want to go down the hardware path further to a digital 8 track or 16 track. Why is that? Can't a computer function as an 8 track? Sure can, you can easily get a fast computer to do 16, even 32 tracks if you want. Heh, notes from my lab.

But if you are a one-person-performer/producer/engineer, and you like doing it all yourself, then the software studio has lots of promise. There's a vast selection of software synths, samplers, effects processors that will fit right into your sequencer. You can get effects with software that would take a 20 foot rack of gear to do in the hardware realm. The sound of these devices gets better all the time, and there are some modern producers that actually prefer soft samplers to hardware for their sound. If you are after new and otherworldly sounds you need to go down the software path. Perhaps the main advantage is the awesome thing you can do with sample editors and soft samplers with plugin effects in your sequencer track. The power is awesome and its right at the control of your mouse. Perhaps the main disadvantage to going a software only route is that you must, of necessity, be extremely computer literate. To keep your studio running you need to maintain your PC, deal with driver conflicts, be able to upgrade components, and most important, learn how to troubleshoot. Heh, some computer tweaks are so cautious they won't even let their music computer touch the internet or network. I think that's extreme, as often you need to connect to get drivers, download updates,etc., just don't let the internet hucksters take over your desktop or start up folder.

Of course you can do a mixture of hardware and software, depending on your goals. That's what I do. Synths and samplers both hard and soft, grooveboxes and sequencer, outboard and plugins. The power of the combination is sometimes overwhelming. As you start out, I only give one piece of advice. Keep it simple.

Notes :

THE ART OF RECORDING AND PROCESSING VOCALS

It seems easy; you just get some lines of words into an audio track and tweak it down. But it is not. The truth is that recording and processing a vocal is one of the more challenging things for a studio to do. Basically, there are two phases to getting a good vocal. First is the recording process where the vocals are recorded. It's important to get your recording process as perfect as possible. Why? Because there are some things you just can't "fix in the mix". It is very hard, no, impossible, to fix damaged, sustained overloads which result from poor recording technique. You have to get the level right the first time. Not even the best restoration software will help. The second process of getting a good vocal sound is "post-processing". This is after your vocal tracks are recorded. Here's where you can fix things like off key notes (with pitch processors), surgically remove short clicks and pops (with a pencil tool in an audio editor), replace words or phrases if necessary to make a great sounding "composite" vocal track. This is also where you apply effects and other processors to make the vocal tracks sound more professional. Well get to as much of post processing as we can.

Part I: The Recording Process

The most common mistake is recording vocals too loud or too soft. The main goal to recording a solid vocal is to get all of the performance. It's not easy to set levels with a good, dynamic vocalist. As soon as you think you have the level pegged, they do something like move a few inches and you find out they are louder than you thought and meters are in the red. So you lower the level and find out that the meters are barely moving at all. If the vocalist is nervous and moving around, you might spend hours and never find an optimum level. The human voice is extremely dynamic, from soft whispers to piercing screams. If the level is too low, you will be bringing in noise and hum if you amplify it later. However, if you record too loud, there will be times when the file goes "over" which will likely result in damage that cannot be corrected later. The solution to this madness is to use a compressor in the chain after the preamp. The compressor, essentially, automatically lowers the volume when the input exceeds a certain threshold. It's like an invisible hand on a volume control. This allows a vocalist to get louder without going into the red. One of the favorite settings is to have the input to the compressor boosted so that all the "soft" words come through with a strong level. As soon as the vocalist gets louder, the clamping down begins and if they scream, it clamps down hard. The ideal is to have more consistent loudness no matter what they are doing. Wouldn't it be easy if you just new 'the best settings for your gear' so that you could just set it up that way and get the perfect vocal. But there's no absolute anything. But there are a few things to keep in mind in your quest for the ideal settings. The ideal settings depend on a few things: the mic's sensitivity, the vocalists dynamics and use of proper mic techniques, the trim level on your pre-amp and finally your compression settings. We will go over each one of these in enough detail to get you going.

Microphone Sensitivity

The more dynamic (louder) the vocalist, the less sensitive the mic needs to be. Some condenser mics will distort like madness if the vocalist is too close when they scream and it is an awful sound, especially if you are wearing headphones. There is nothing you can do to fix that audio either. Because the distortion happened before the signal hits the compressor, all the compression in the world cannot help. If there is a -10 or -20 pad on the mic, use it with untrained wild vocalists. Otherwise, use a dynamic mic, which is less susceptible to break up under high sound pressure levels (SPL). Or you can have them take a step back before they commit their bellow from their personal living hell. That's what comes in the next section.

Proper Mictechnique.

This depends on the volume of the vocalist. A soft sensitive voice requires that the vocalist nearly devour the mic. Don't really eat the mic. You just have to be about 4-6 inches away. Otherwise, the rule of thumb is about 1 foot away. The vocalist should back away a few inches when they get loud and come in a few inches closer for quiet intimate parts. The vocalist should not sing directly into the mic, or bassy wind noise will get in the way. Just a few degrees to the side is better. A pop filter should always be used. This is not only a good device for getting rid of plosives and spitty sounds, but can be used to keep the vocalist from getting too close and out of the range where a proximity effect might engage excessively.

Proximity effect is the tendency of some microphones to exaggerate the bass frequencies of a vocal when the vocalist gets within 1 inch the mic. Comedians, radio announcers and performers often use this to great effect, but in a pop song, you typically don't want this sudden bass enhancement.

Pre-amp Trim level

This is the amount of gain (volume) applied to the mic signal, and it is calibrated in db (decibels) from 0 to typically 60db All mics differ a bit on how much juice they need. If you have a condenser mic, phantom power needs to be engaged to power the preamp. Dynamic mics don't need phantom power. Most mics will fall between 15-40db of boost. Have your vocalist practice singing and try to get the loud peaks to peg close to 0db. This will give the compressor a healthy level to work with. If you are not using a compressor you will have to lower the trim to ensure the signal never reaches 0db. That is a much lower signal than you might think.

Compressor Settings

Setting Gates: Compressors do add noise to a signal, and they do destroy dynamic range. Noise is taken care of by gating the signal. When it dips below a certain threshold, the audio signal is muted. This is effective for getting rid of low level noise you do not want in the file, such as bleed from headphones, or the vocalist moving, turning pages on lyric sheets, etc. Gates have two parameters: 1) The noise floor threshold, and the Rate. The Noise floor threshold eliminates all of the signal when it dips below the threshold, which is set from -50db to -10db. I keep mine set to -30db. Yet one has to be careful. If the gate is set too high, then the attack of the vocalists words may be cut off or come in too abruptly. The Rate parameter "fades out" the audio signal as the gate come on. This is effective to prevent the gate from chopping off the tails of the words. Usually a rate of 1-1.5 sec is enough.

Setting Threshold: The Threshold is the all-important level at which the compressor kicks in. If you set the threshold to -10, it will leave all of the signal under -10 alone. When the signal exceeds -10 then it starts compressing at the ratio. -10 is an excellent place to start. Don't confuse this with the fact that your gear is outputting -10 or +4 impedance wise. Though the threshold seems like it is a volume control, it is not. It is merely telling the compressor at what level compression takes over the signal.

Setting the Ratio 2:1 is probably the most common setting for a compressor recording or playing back nearly anything. A great starting point. What this means, simply, is that it takes 2 decibels of sound energy to raise the output meter by 1db. You can read the 1st number as the db IN and the second as the db OUT. Again, 2db IN equals 1 db OUT. Easy, huh? Yeah, with 2:1 you simply divide by two. So lets test yourself, do the math, then you will grasp this fully.
Answer this: If your vocalist was singing at -10db and suddenly got 20 db louder, without compression, where would the meters post?
That's easy -10+20=+10. The meters would post at +10

Correct! Which, as you should know is way to loud and would ruin the track. Now, if you had 2:1 compression applied, where the output is half of the input, where would the output meters post?

-10+(20/2)= zero db

Yes! Why is that? The vocalists 20db burst was compressed to an actual 10 db difference in gain. (the ratio 2:1 is the same as 20:10, or half). Makes sense?

Lets go one step further, make sure you got this in your head. If you had the compressor set at a 10:1 ratio what would that mean? It would mean for every 10 decibels of gain the meters would only go up one db. So in our example, then, the 20 db burst would only let the meters go up by 2db (10:1 is the same as 20:2, or 1/10th of the original sound), Since they started at -10, the overall level would be only at -8 during the sudden 20db boost. Hardly any change in the output level at all. Would that sound "squashed"? You bet.

Ok, you just got through the hard part. Congrats!

Setting Attack and Release: These settings can be tricky as they can "delay" the effect of compression on the attack and make it hold on a bit too long on release if set improperly. We suggest till you get these tricky settings figured out (which takes quite a bit of experimentation) you simply use the fastest attack and enough of a release so the vocal is not boosted as the word trails off. Otherwise a word may pump on you unnaturally.

Setting the output: This is the final adjustment as the signal leaves the compressor. It's sometimes called the "make-up gain". They call it that because compression often lowers the overall signal and you may need to boost it back up. Basically you want to optimize this so it does not ever go over 0db in the recorder. With luck you should see a consistent healthy level on the recorder's input meters regardless of how loud the vocalist is singing.

Just a final note, you can compress again after the vocal is recorded as you prepare your tracks for the mix. So, don't get too wild with settings at the input (recording) stage. You want the recorded vocal to sound natural, where the compressor just makes it an overall more useful signal to tweak later with exciters, harmonizers, pitch intonation correctors, and effects like reverb, delay. etc.

Preparing for a Vocal Session

Most of the problems that will occur later are a result of not taking the time to properly set up the session. Here's the mental checklist you can use when going into a vocal session.

1 Does the vocalist have the lyrics? It is best if they were given these a few days in advance with basic instrumental track. If not, the vocalist will have wing it on the fly and do their experimenting with the clock running. If you did not get them advance copy, do you have them printed out for them in an easy to read font?

2 The Mic should be set up prior to the session and the preamp level should be set. If you are using a compressor going in, have that setup too. You might have to tweak that a bit once they arrive but if you have typical generic setting already set up this will be easier.

3 You have patched a reverb into the monitoring chain. You do not have the signal of the reverb going to the recording input, but only to the monitors and headphones so the vocalist hears their performance with reverb. Most vocalists will give a better performance if they hear some depth on their voices.

4 Songs are loaded and it plays back as it should, tracks are created for the incoming audio.

5 Take great care to make your vocalist comfortable. It's their performance that is going to make the song and you want them to feel relaxed and confident.

6 Don't make them practice too long before recording. Vocalist's typically deliver their best in the 1st hour of the session, so don't waste their voice on superfluous stuff.

7 Never give negative feedback to a vocalist. Don't say, "You sounded a little off key there, lets try again". Instead, say "What did you think of that take? They will probably say, "Oh I thought I was a little off, I want to try it again." Let the vocalist judge their mistakes. It is their voice on the line. You should, however, make sure they know when you hear something you like. "That was great the way you held that note!"

Q) Should I Use The Limiter on my Compressor for Vocals? How do I do this?

You typically engage the limiting function by setting the ratio to infinity: 1. This means no program material will be output over the level set in the threshold. You can use it for vocals, but it often causes extreme gain reduction and can make for a lifeless vocal. If the threshold is set high enough it will only flatten the peaks, but will not offer much improvement over dynamics of the original signal.

Q) Does a "soft knee" help with vocals? What is a "soft knee"?

A) A soft knee is a feature that affects the "slope" of the ratio as it goes over the threshold. The soft knee is a little more gentle on audio material as the compression kicks in gradually rather than abruptly as the signal crosses the threshold. It's very subtle and you might not hear any difference at many settings. Leave it on for vocals.

Q) Why use a compressor at all at the input stage if you can compress later on? Why not just "normalize" the signal after is recorded?

Valid point. Compression cannot be undone. This is yet another reason to only use gentle settings like 2:1, which are unlikely to wreck the dynamics of the vocal. However, without compression, as stated before, the recording will have to be at a much lower level and will come in "spiky". By spiky, I mean the average level will be too low and the peak level too high. After normalizing it will still be spiky. Without compression one might have to do a vocal take over and over due to the vocalist ripping through the roof.

Q) How important is having a great mic?

Great mics impart subtle, sweet characteristics, i.e., "flavors" to the sound. One of the harder characteristics to emulate is "smoothness" using a cheap mic. A great mic sounds sweet by itself with little tweaking. Microphone purists want to preserve the great sonics of their mics and do little tweaking. However, you can dramatically alter a mics response with EQ and excitement and improve it's overall performance quite a bit. Almost any professional condenser mic with XLR cables can be made to sound perfectly acceptable with careful placement and judicious processing.

Part II Post Processing Vocal Tracks

Now that we are past the recording process, the real fun begins, the tweaking! Here's where you apply the magic to make the vocal sound full, lush, and ring with great clarity.

You can experiment with these processes, but I typically work my vocal tracks this way. Note, you don't have to use all these processes. These are simply the ones that are used a lot. Your song, your mileage, varies.

Define your Tracks

In your audio sequencer, create several audio tracks for the vocal. You can work with different settings on each. Try different combinations of effects and enhancers till you find one you like. This is as easy as moving the vocal track from audio channel to audio channel. When you settle on the channel you like best, rename that as "Main Vocal". By the way, you might want to use mono tracks for the main vocals. No, don't protest, I know they are harder to set up. Just do it. You will find vocals more consistently stay in the center of the mix where they need to be.

Now you can make alternate vocal channels-perhaps one for choruses, a few for effects, one for doubling and perhaps one for sampling. You can cut up the vocal in the sequencer and put different parts on different tracks with different effects. The most obvious here is to put the choruses on a track with more processing to make them stand out a little more. I also develop a set of tracks for doubling as well where the vocals are hard panned left and right and playback two different takes of the same chorus for that classic thick sound.

Time Correction

It happens all the time, the vocalist gives a stellar performance on a chorus but came in late. With any vocal track in a sequencer you can slice it up by words, even syllables to make it fit in time if you need to. It's a good idea not to trash bad vocal takes as later on you may find all you need is one word to replace a botched word in the choice track. The joys of processing vocals in a sequencer is that you can mix and match segments from many takes down to one. This is called a composite vocal track. It's true, some of the stuff you hear on the radio might actually be a composite of 3-30 takes. The performance may have never existed in reality. Ever wonder why some unnamed Divas can't sing their song live very well? Of course a truly great vocalist will nail the track in one take.

Pitch Correction

You might not think your vocalist needs it. If they are rock on pitch, they might not. However, if you listen to most popular songs you will find that most vocal tracks are pitched perfectly, dead on. How many slightly off keynotes do you hear on the radio? None! How many slightly off pitch notes are in your vocal track. 10? 30? 50? Uh huh, Case in Point! Even the best singers will have days when certain notes are consistently a little sharp or flat. Even the best vocalists benefit from some pitch correction, and a bad vocalist might actually get by with correction. A good pitch correction processor will gently (or abruptly, if you want) bend the note to the exact perfect center of the pitch, and you can also add vibrato and even wilder yodel like effects if you want. After building the composite track, correcting timing and pitch errors, you should mixdown the vocal track to a new file. This way you can remove any plugin processors used so far and clear automation to start fresh as you go into the next round. You also can go back to your source tracks at any time if you screw something up.

Destructive Enhancements

Here's some things to do in an audio editor which may enhance the track before you add plugins. Track cleaning. Open your newly mixed main vocal in an audio editor. We are going to clean the track. Here you zero out (silence) all the dead space between phrasings. Breath control. A long debated question is: Do you keep the vocalist's breaths in the recording or zero them out? If you take out all the breaths, the vocal will sound strange. Certain breaths are very musical, like the ones leading up to a loud part. However, there are usually some that are excessive or out of sync, or just get in the way. Those are the ones to remove. Remember you still have your source files in case you botch this. Gain Optimization. Look for words that do not ring out as clearly or may get buried in the music. If you built a composite track you might have different takes at different levels. You want them all to sit up in the audio editor in the same way if possible. Here you can use normalization to good use. But don't normalize the whole track, normalize whole phrases. This brings the soft ones up to the same level as the loud ones.

Setting up Insert Effects

In the main vocal track, start with compression to smooth out the levels a little more. Since you compressed going in, you may not need much. However, I find it to be real important to keep the vocal consistently above the music. If you are hearing too many "SSS" sounds in the vocal, it is time to apply a de-esser. After compression, it gets exciting. No not like that, but with an exciter. An exciter essentially gives you "sheen" on the high frequencies by adding harmonics to the source signal. This is more than the boost that EQ gives. An exciter actually adds harmonics that were not present in the original signal while an eq just raises the volume of those that were buried. With a combination of eq and excitement, you can get the vocal as bright and crispy as you want it. Most popular records have vocals processed with great brightness. It increases intelligibility and makes the vocal sound "clear" even on inferior car and boom box speakers.

Setting up send and Return Effects

Now that we have our main vocal channel set, we move to the sends and returns. Here we put the "common" effects that may be used for all the vocal tracks and even for some instrument tracks as well. Of course I am talking about reverb here. On our software or hardware mixer, route a send to an effect. In the software mixer, you create a bus and put a reverb on it and send the signal to this destination from the send knob on the vocal track. On a hardware mixer the "aux send" goes out the back and goes to an effects box. The output of the effects box comes back to the mixer via the returns. Its a common mistake to use too much reverb so don't overdo it. Other excellent effects that can be applied here are delays. Just a little bit goes a long way, especially when you also have reverb running.

Spot Effects

If you listen to my stuff, you know I am a big fan of "spot effects" which is done simply by putting part of the main vocal track on a different track with strong effects. Some effects that can be used on different tracks are harmony processors, radical EQs for lo fi effects, vocoders, extreme delays and reverbs, distortion, and whatever else you feel helps make the artistic statement.

Because your main vocal tracks are centered, for effects you may want to move them off center. This adds a little more dimension. Remember a good effect is one that defines it's difference relative to a norm. So your main tracks should be dead center, loud and clear, full and rich. Your effects tracks can be of great contrast, i.e., all the lows removed, all the high's removed, totally gnarled, nothing but reversed echoes, whatever.

Sampler Effects

Don't forget, you can use your soft or hard sampler for vocal effects too. Toss the whole vocal file in recycle, slice it, then port it over to the EXS, Kontakt, your E-mu for some dangerous keyboard controlled effects, like stutters, filter swept voices, LFO Mods.

Volume Automation

Your sequencer has automation, use it. As the Mix plays, not any sections where the vocal needs a boost or a cut. Draw them in. Grouping If you have a multi output audio interface and enough busses on your mixing board you can consider making a "group" for just the vocals. This can also be called a vocal "submix". Rather than having each vocal track set to different mixer channels, route them all, post insert effects, to a single group channel. This gives you one fader that raises lowers all the vocal tracks. It is important when getting the overall level of the vocal set against the context of the music. You may use automation on this bus too.

The all Important Final Level

So we are almost done. We worked hard to get here, but all of the work is in vain if the final level is not set correctly. The whole point of all this compression, excitement, eq, and post processing was to get a vocal that sits up in the mix properly, where every word is intelligible, where the track never gets drowned out by instrumental tracks but does not drown them out either. Be real careful with the final fader tweaks. Try to get the vocal where it "floats" on top of the mix in a nice way. Pan other instruments to the sides that might compete with the vocalist's sweet spot and avoid putting so much reverb on the voice so it sound like it is behind the instruments. You might try doing 3 or 4 mixes at different setting for the overall vocal just so you can listen elsewhere to critically evaluate the level.

Mastering Considerations

After you mix your final wave file, you still have one more chance to touch up the vocal during the final mastering process, which will be burned to CD. A good quality parametric EQ can touch up the frequency response of the vocal's sheen (as well as the entire mix's overall frequency balance.) You shouldn't have to do much here, since you were so careful during the recording and processing of your mix. But a little bit of eq or multi band compression can raise or lower the "temperature" of the mix quite dramatically.

Q) I have a harmony processor. Uh, where does it go in the recording chain? It has a mic input.

A) Of course you can record through it up front, and sometimes you may want to do that. However, you are stuck with the result later and it can't be changed. I would put it as an insert on a vocal channel of your mixer. That way you can tweak the settings later on and get them to fit perfectly with your song.

Q) How do i get vocal effects like on Cher's "Believe"? I heard it was done with a Vocoder.

A) Nope. It was done with a pitch/intonation processor, like Antares AutoTune. You get that effect by "abusing" the settings. You tell autotune you are singing only Ab and Eb notes and you sing F and C notes. Auto tune will bend your vocal notes to the closest selected pitch giving that "yodel-like" sound

Q) I want my vocal to stutter at points as an effect. How do I do this? Should I slice it up on an audio track and use copy and paste?

A) That works. A sampler works better though as you can control it with MIDI. This allows very fast 32nd note stutters, which would be very tedious copying and pasting. If you use a sampler you can also modulate it with a square wave LFO so it stutters through the whole waveform, not just repeat the attack.

Q) How do I get "formant" processing effects and what is that.

A) Formant processing is a "modeling" technique where a digital model of a vocal cavity is applied to audio material. You can turn a male voice into a female voice for example and many more wild things. You need hardware or software that does this, like Kontakt or a Roland Variphrase

Recording Vocals

Lets try and look into the vocal recording process and the stages in details now. We might be repeating a few things already mentioned above but they are essentials in the vocal recording processes and important for you to be very clear about them. 'Vocals: the all-important lead instrument in many genres of music. Not only do vocals outline the melody of the song, they also provide meaning for the song. So obviously getting the best possible vocal recording is an extremely important part of creating a killer song.

This segment outline some techniques you can put to use in your studio immediately to improve your Pro Tools music productions.

Preparing For a Vocal Recording Session: The Shootout Before the vocalist even sets foot in the studio, there are a bunch of things you can do to prepare for the recording session. Choosing your signal path -microphones, mic preamps, and effects - is the first step. Often, this process is done in a demo session before the "real" recording session, where the vocalist sings through several different mics, possibly through several mic pres, and even through a few different compressors, to find the best sound for their particular voice. This is called a "shootout." If you've only got one mic, one mic preamp (on your Mbox or 002), and one compressor plug-in, you don't need to do a shootout -but you should optimize the gear and signal path that you do have. Finding the best position for the mic and the best compression setting for the vocalist can save time and frustration, and will probably lead to a better performance.

Using a Compressor Plug-in

After choosing the best signal path for your vocal input, set up Pro Tools for your recording session. Instead of using an outboard compressor, I often record through a compressor plug-in inside Pro Tools. To record though a plug-in, you can't simply insert a plug-in on the audio track that you're recording to. Why not? Because inserts on audio tracks are post fader — that is, the plug-ins on audio tracks process the audio after it's already been recorded on your hard drive. So you've got to route the input vocal signal through the compressor on an aux track first. Notice the signal routing here: The vocal signal comes into the aux track, where it's processed by the compressor plug-in. Then it travels out of the aux track on a mono bus to the audio track input, where the compressed vocal track is recorded.
I often use the Universal Audio 1176 or LA-2A plug-ins on vocal tracks. I recommend using a light compression setting (with small ratios like 2:1) for most vocal recording applications, because you don't want to over-compress the vocal right away. You can always add more compression in the mix session. Sometimes I also use the Antares Microphone Modeler plug-in as a post-fader insert on the vocal audio track to simulate the sound from a different mic.

EQ and Effects

Sometimes it's beneficial to make use of the high-pass filter (a.k.a. "roll off") on your microphone or Digi 002 Rack to eliminate unwanted low-end frequencies or proximity effect on the vocal track. (Proximity effect is an increase in bass frequencies resulting from the vocalist being close to a directional microphone.) However, I often use EQ in the mix for this purpose instead, to make sure I don't unnecessarily roll off some frequencies that are important to the vocalist's sound. Reducing 200 Hz on the vocal track will often take care of a vocal track with proximity effect. Although EQing a vocal track right off the bat is not uncommon, I find that EQing the vocal is best saved for the mix session, so you can hear how it sits with the rest of the instruments in the mix and then adjust accordingly.

Once you've got your input signal path set up, add some flattering effects to the vocal track, like reverb. Reverb often makes vocalists feel more comfortable with their voices. Making the vocalist comfortable pays off big, because they'll have greater confidence in themselves and will "let loose" more in the recording session. Set up a send/bus/aux track to add reverb to your vocal track.

Headphone Mix

Another key to capturing an incredible performance from a vocalist is a great headphone mix. I recommend boosting the guitar, keyboard, or other harmonic instrument in the headphones so that the singer has a solid pitch reference. To further enhance this, keep the effects on those instruments to a minimum. Keep at least one rhythm instrument relatively high in the headphones too, as a rhythmic reference. Pay close attention to the level of the vocal in the headphone mix, too - vocalists tend to go flat if their voice is too high in the cans and sharp if their voice is too low, because they try to push out volume to overcompensate.

Lyric Sheets & Take Sheets

Knowing and being able to sing the melody and lyrics to the song that you're recording can be a huge time saver, because that familiarity with the song will help you work with the artist faster. At the least, have the lyrics written down in front of you while tracking. To be more professional and organized about it, create a "take sheet" that has the lyrics broken down by phrases, with room to write notes about each overall take and specific phrases, words, and even syllables. This enables you to keep track of the best performances for each part of the vocal track, and makes editing and comping the vocal track much faster.

In the Vocal Recording Session

The vocalist will usually need more than one take to nail a performance. Except on the rarest of occasions, editing is necessary to create a comp of the best parts from each recorded vocal take. It's common to work with four to seven vocal takes at once during a session, keeping only the good to excellent parts. However, I sometimes simply record everything, keeping track of the best material on a take sheet, or even by marking or renaming regions within Pro Tools itself.

Recording to Playlists vs. Multiple Tracks

There are two ways in Pro Tools to record multiple takes of a track, using either multiple playlists on one audio track or multiple audio tracks. Both have their advantages. Recording different vocal takes to different playlists on one audio track means you only have to set up one vocal track, and it's really easy to edit the best parts from each playlist to create the master take (comp). This is a great technique if you have a limited number of tracks available in your session, and if you want to simply record everything without being limited to the available tracks (or voices).

Recording vocal takes on multiple tracks instead of playlists is easy too. After you make one vocal track (with all of its plug-in settings, volume levels, etc.) you can duplicate it using the Duplicate Selected Tracks command from the File menu. This command creates a new track with all of the same settings as the original track. The main advantage of using multiple tracks for vocal recordings is being able to see and compare each track as you record it -but it can make a session heavy on track count. It can become overwhelming if you're using more than four to seven vocal tracks, yet it's just as easy to create a comp using this technique.

Pre-Roll

When punching in to capture better performances on certain words or phrases, use the right amount of pre-roll for the situation. Pre-roll is the amount of the song that's played before a track is record-enabled. When starting off, vocalists often want more pre-roll as they get used to the groove of the song. However, as vocalists get deeper into the recording session, they'll often only need a measure or two of pre-roll. Short amounts of pre-roll keep the vocalist focused and keep the session moving along quickly.

To set the pre-roll amount, go to the Pre-roll area in the Transport window, type in the amount you want, and press the forward slash key (/) to enter the value. For example, if you want to enter two bars as the pre-roll amount, first make sure "Bars:Beats" is selected as the Time Scale (from the Display menu), then type "2" in the pre-roll section of the Transport window as in Figure 3, and press forward slash.

Summing up

Getting your signal path and Pro Tools session set up correctly, as well as running a smooth and organized recording session, will help you capture much higher quality vocal tracks.

Part 2: Editing

Now we're going to move on to the next step: editing vocal tracks. If you produced the vocal recording session well, captured excellent performances from the singer, and took good notes about which vocal takes (phrases, words, syllables, etc.) were the best, the editing process should be relatively straightforward. But even with great source material, using good editing techniques is the step that turns your collection of raw performances into one killer track, ultimately taking your song to the next level.

What is Comping?

The first step in editing your vocal track is called "comping." Comping is the process of compiling all the best parts from multiple takes into one master track. For example, say you have three vocal takes of the first verse of a song. If you edit together the first line from take #2, the second line from take #3 and the third and fourth lines from take #1 into one "keeper" take, that's comping.

Often, the goal in creating a vocal comp is to get the most emotive performance possible. Always keep this in mind while editing -think of the song first, and choose performances that are best for expressing the emotional content of the song. Often, the best vocal performances are not technically perfect. There might be slight pitch issues, slight mispronunciations, slight mic level issues, or even noise and distortion on the track. Yet, as a whole, the performance is perfect in delivering the emotion of the song. So don't get caught up in finding the best technical performance by sacrificing the emotion in the editing process.

If you used a take sheet as recommended in last month's column, follow the notes you made to steer your vocal edits. Without take sheets or other written comments on your recorded tracks, you'll probably need to listen to each take and make notes now, during the editing process. This makes the editing process much more time consuming.

For vocal tracks, I recommend analyzing the delivery of every phrase, every word, and even every syllable. Although you may not have gotten that detailed when recording the vocal tracks, it's good to do it now. Write down additional notes on the take sheet or lyric sheet about edit points. Pick and choose each part of the vocal performance carefully when creating your master comp. Professional producers go to this level of detail. Will you?

NOTE: Don't be afraid to edit between syllables. In singing (and speech) there is sometimes less amplitude in the signal in the middle of a word than there is between words. Bringing out the best syllables and creating words with them can really improve the impact of a performance.

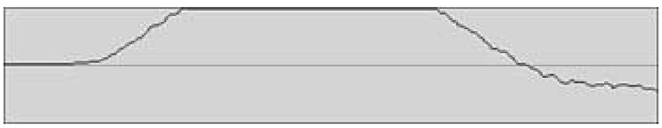
How to Comp

Multiple Playlists on One Track: I often perform comps from multiple playlists on the same track in Pro Tools. It's easy to copy and paste parts from each take onto one master comp take. This technique reduces the track count in your Pro Tools session, and you also only need to adjust one volume fader and one set of insert effects on the track. Each playlist makes use of the same volume level and effects on the track.

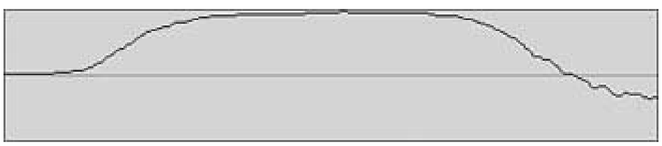
One Playlist on Multiple Tracks: Whether you record your vocal takes in multiple playlists on one track, or onto separate tracks, the comping technique is essentially the same. By working with different takes on multiple separate tracks, you always have access to each track, you can see each track all the time, and you have a dedicated fader for each take. However, you need to make sure the levels and the effects on each track are the same. Alternately, you can bus them all to one separate aux track, so you only have to control one fader and one set of effects.

NOTE: Sometimes performances aren't always recorded at a consistent volume. Quieter passages may not sound as "good" as louder passages simply because they're not as loud -but you should still consider boosting their levels with volume automation and using them in your comp.

Regardless of which recording and editing process you prefer, it's best to limit the number of takes of the track that you're comping to a relatively low number, like five or six. Otherwise, you can easily be overwhelmed by the sheer number of options, and lose track of the good takes. If you were smart enough to use a take sheet while recording multiple takes of a track, then you should have notes about what parts of each take are "keepers."



Repairing Waveforms



Pro Tools allows you to redraw waveforms by altering audio files on the sample level. This is particularly useful for repairing vocal plosives, pops, or clicks in the track. Be careful when using this tool, though; redrawing a waveform permanently alters the audio file on your hard drive once it's saved (in other words, it's destructive editing), so make a backup of the original file before editing.

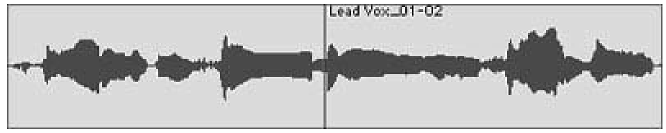
When repairing a waveform by redrawing, try to keep the same basic form of the sound wave or you might create an even nastier sound than the pop or click you were trying to fix. On the other hand, you can create some wacky effects while editing on the sample level. Experiment with your waveform drawing technique…and remember, you can always undo your edits before saving.

Before: Using a pop filter reduces the occurrence of pops

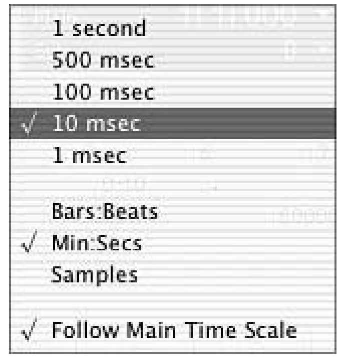
After:...but you can use the Pencil tool to redraw them if necessary

Nudging

Once you've created a comp (or while you're creating it), you may need to nudge certain words or phrases in time so that they're rhythmically tight with the song. What does it mean to nudge something? Nudging can be defined as "pushing slightly or gently." When editing vocals, you may want to slightly push



a word or phrase so that it lines up with the beat, or with a specific rhythmic hit.



To nudge a word or phrase, first find the word(s) in the waveform. Zoom in close enough to see where the word begins and ends. Separate the word into its own audio region, then move it slightly earlier or later in time. I recommend trying 10ms as a starting nudge value, and adjusting from there - sometimes even a few milliseconds can make a big difference in feel. You can select the nudge settings (including nudge amount and minutes:seconds versus other units of measurement) using the pull-down list in the black bar at the top of the Pro Tools Edit window.

Separate the word or phrase with the Control + E (PC) or Command + E (Mac) shortcut

Choose the nudge value in the nudge pop-up window

Select the region with the Selector or Grabber tool, then press the Plus (+) key on the numeric keypad to move the selection forward by the nudge value or the Minus (-) key on the numeric keypad to move the selection backward by the same amount. The Nudge command works exactly the same in all Edit modes (Slip, Shuffle, Grid, and Spot).

Be careful using the nudge editing technique, though. You can go crazy trying to align every word, killing way too much time and taking away from the real performance - the "vibe" of the part. If the part needs that much fixing, re-record it!

Summing up

Now you've got the tools to put the pieces together for a great lead vocal comp track. Join me in the following months for more Pro Tools vocal production techniques, including vocal tuning, rhythmic alignment, automation, and mix techniques.

Part 3: Tuning and Aligning

Groundwork. A great name for this column (thanks Dusty!). That's what I try to cover here: production techniques that provide a solid foundation for working with Pro Tools. Well, what's more fundamental and essential when producing a song than making the vocals sound great? In many genres of music, and to many listeners, the lead vocal is the song. Period.

Two months ago, in Part 2 of this column, I covered techniques for vocal comping -putting the best parts of a vocalist's performance together into one killer lead track. In this third installment, I'll cover techniques for tuning vocals and aligning lead and harmony vocal parts.

Tuning a Vocal Track: Tools of the Trade

After you've comped your lead vocal track, it's common to fix minor tuning problems using an auto-tuning program such as AutoTune by Antares, or PitchDoctor by SoundToys. These programs can automatically tune the track for you in real time, as the track plays back. However, using automatic settings on an entire track can often result in a noticeably artificial-sounding performance (unless you're going for the "Cher" effect). But if you take control of the tuning process and tune the vocal track manually, you can achieve excellent results. To perfect the tuning of your vocal track, you should analyze and tune each vocal phrase separately, bouncing each newly tuned vocal phrase onto a new track when completed, and eventually creating a completely tuned vocal track.

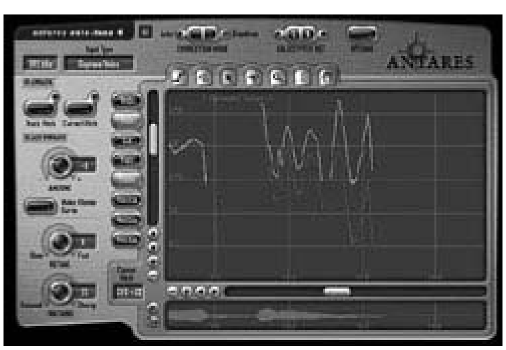
Note: Some artists make a point not to use auto-tuning. If the singer on your project can sing totally in tune (or in tune enough), don't bother with auto-tuning and leave the raw performances. Often, the "realness" of these raw performances creates a better overall feel on a recording than "perfect" processed vocals.

TIY -Tune It Yourself

Both AutoTune and PitchDoctor enable you to manually adjust the pitch of any note in a vocal performance. This does take some time to do, unless you become very proficient at it. Some of the most talented Pro Tools experts even pitch-correct on the fly during recording sessions. However, if you're working as producer and recording engineer in your own studio, I highly recommend concentrating on getting the best performance during the recording session, and waiting until the session is over to make your comp and tune your vocal track.

Let's talk about the tuning process using the AutoTune plug-in. Here's what to do: Instantiate AutoTune on your comped vocal track. Then create a new audio track, and bus the comped vocal track out to the new track.

Manual pitch correction in AutoTune starts with selecting the Input Type, Scale (I almost always use the Chromatic scale), and Graphical instead of Automatic as the Correction Mode. AutoTune then automatically switches to the Graphical mode window, as shown in Figure. **Figure 2** After adjusting the Retune and



Tracking rotary dials (I usually select single digit numbers for both), highlight a phrase on the comped vocal track that you'd like to tune. Press "Track Pitch" in AutoTune and play the phrase. A red line will appear in the AutoTune window that mimics the pitch.

At this point you can use several tools to adjust the pitch of the phrase. I usually press the "Auto" button to start with. AutoTune creates what it thinks the pitch correction should be, based on the chosen Retune and Tracking parameters. This yellow correction line is what the plug-in would do if it were in Auto mode, and I find it's often a good foundation to start tweaking from. You can fix any discrepancies by altering the yellow line using a variety of drawing tools.

Click the "Correct Pitch" button and AutoTune will alter the pitch of the track when you press Play in Pro Tools. Listen for any continuing pitch problems and unwanted AutoTuning artifacts. In most cases, the idea is to make the track sound like it was not intentionally tuned. Now that some listeners know what AutoTune can do, they can pick out AutoTuned notes — so take your time and get it right. It's the lead vocal. It's got to sound great!

Follow the above procedure for each note or phrase in the track that needs pitch correction. Note: You may experience a slight delay between the comped vocal track and the tuned track, due to both the AutoTune plug-in delay and the internal bus routing delay. Nudge the tuned track back in time to compensate if you hear a delay.

Also consider this: If you tune the lead vocal before recording the backup vocals, the backing vocals will probably require much less tuning, since the backup singers will be harmonizing with the correct lead vocal pitch.

Special Effect Tuning

With PitchDoctor, you can use a guitar track (or any other monophonic instrument track) as a guide to correct the pitch of the lead vocal track using automation within Pro Tools. For example, you can record a lead guitar track, analyze its pitch to generate a pitch "score," and apply that to your lead vocal track. On top of that, you can manually automate the pitch "score" like you would any other automatable Pro Tools parameter for even more pitch control, as in Figure 3.

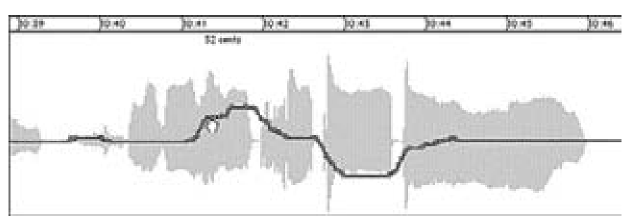
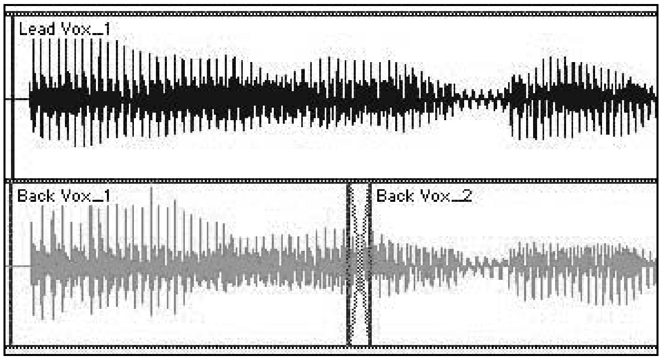


Figure 3. PitchDoctor enables you to easily draw automation to adjust the pitch of your lead vocal track.

Similarly, you can use a MIDI keyboard to trigger the pitch of a vocal track using the Target Note Via MIDI in AutoTune. This



is a production technique that's becoming more common in popular music, specifically in the electronic genres. You can get really creative with your vocals using either of these plug-ins. Try 'em out!

Aligning Vocal Melodies and Harmonies: Cut It Up

After comping and tuning the lead vocal track, it's a good idea to tighten up the rhythms of the harmony vocal tracks to match the lead vocals. The point is to get the start and end point of each harmony note tightly aligned with the lead vocal, as well as any obvious hard consonant sounds (b, d, g, k, t, etc.) that might distract from the rhythm of the track if not aligned. This means if a harmony note is longer than the lead note, you need to shorten the harmony note. Cut some of the middle of the note out and create crossfades between the remaining regions, as in Figure 4. You may need to use equal power crossfades here, and the crossfades may have to be a little longer than usual to make this sound natural.

Figure 4. Cut down the size of a harmony note to match the length of the lead line.

If the harmony note is too short, then you should copy a section of the harmony note and paste it into the middle of the note to make it longer. Then apply crossfades to smooth it out. In this case too, you may need to use equal power crossfades, and the crossfades may have to be a little longer than usual to make this sound natural. Also, trim the starts and ends of the notes so they are rhythmically tight.

VocALign

Another cool tool for aligning vocals is VocALign Project from Synchro Arts. This plug-in helps you match the timing of one audio signal to another, such as for doubling lead vocals and aligning harmony tracks.

Time Compression/Expansion

If all else fails, another technique you can use to align vocals is time compression and/or expansion. With Pro Tools, you can expand or compress an audio region without affecting its pitch using a variety of available tools, including Digidesign's own AudioSuite TC/E plug-in, Serato's Pitch 'n Time, Wave Mechanics' Speed, Synchro Arts' TimeMod, or Waves' Time Shifter. For example, instead of performing the somewhat complicated editing on the harmony vocal lines described earlier, you can apply time compression or expansion to the harmony notes and align them with the lead part in seconds. This editing technique has become even easier to do now that Pro Tools has a Time Trimmer tool. This tool automatically uses your default audio compression/expansion plug-in to shorten or lengthen any audio region.

Summing up

After comping, tuning, and aligning the timing of all the vocal tracks in your session, consolidate each track on a new playlist so you have one "perfect" region for each vocal track.

Part 4: EQing Vocal Tracks

We've discussed recording and editing vocal tracks in Pro Tools, from capturing the best sound to getting a great performance to editing together the best parts into one solid, emotive performance. But even with all that work, there's still more to making a vocal track shine. Let's jump right into mixing techniques for vocal tracks.

Chopping the Timbre

In almost every song with vocals, the instrumental tracks in the mix are subordinate to the lead vocal track. The lead vocal must cut through the mix -its clarity is the primary focus.

There's a problem, though. Many instruments in a mix share common frequencies. Almost every instrument in your mix (including the human voice) consists of multiple frequencies, and those frequencies overlap. Even if an instrument is only playing one note (the fundamental pitch), it also generates additional pitches with lower amplitudes (known as harmonics and overtones) that color the overall sound and define the instrument's timbre. These frequencies can interfere with the clarity of the vocal track in your mix. Enter equalization, also known as EQ.

There are two main reasons to apply EQ: First, to adjust the frequency content of a sound (its timbre) so that the sound is more subjectively appealing in the overall mix, and second, to make room for certain instruments to cut through the mix. This is sometimes called "carving EQ holes."

Carving the Holes

It's a common practice to carve EQ holes to make room for the primary frequencies in specific tracks. For example, if you want to enhance the sound of both the kick drum and bass tracks without having them compete, you can boost and/or cut different frequencies within each of their ranges. In this case, you might boost 60 Hz on the kick and 100 Hz on the bass, then cut 100 Hz on the kick track and 60 Hz on the bass. Doing this will create EQ holes for each instrument and add clarity. (Note: I am not saying that each instrument should have its own dedicated frequency range in a mix. Instruments will continue to share frequencies, but clearing a path for the predominant frequencies of certain instruments can "open up" a mix.)

This technique also applies to vocal tracks. Vocal intelligibility is mainly found in the 1 kHz to 4 kHz frequency range. Carving out some of the overlapping middle frequencies on your guitar and keyboard tracks can make room for the lead vocal to shine. You can go one step further: Boost the vocals in this range while attenuating the same frequencies in the rest of the instrumental tracks, to make the vocals stand out while maintaining a natural overall blend.

Tweaking the Freqs: Basic Techniques

Not all vocal tracks need to be EQ'd. However, it is common practice to bring out certain frequencies to improve a vocalist's overall sound, as well as to create a more emotive performance. When searching for the right frequency to adjust, try this technique: Insert an EQ plug-in on a track, increase the gain significantly (like +12 dB) on one of the parametric bands, make the Q value high (for a narrow bandwidth), and sweep across the frequency range until you find the frequency that you want to boost or cut. Then adjust the gain and Q parameters accordingly to sculpt the modified sound. Listen to the sound both soloed and with the rest of the mix. Some call this the "boost and twist" technique. Try to work quickly when EQing. Once you think you've found the right frequency, tweak the sound to your liking, then toggle the bypass button on the EQ plug-in and compare the original sound to the EQ'd sound. Pro Tools comes with two DigiRack EQ plug-ins, a one-band and a four-band parametric EQ (in both mono and stereo varieties). You also get IK Multimedia's T-Racks EQ plug-in. Try using these plug-ins to sculpt your vocal sound.

Using Your IQ to EQ

Because everyone has a different-sounding voice, vocals are one of the most challenging "instruments" to EQ. Vocal range and gender affect the recorded track most, but EQing at the frequencies in the following table will can really improve the sound of a vocal performance.

To change this sound:	Adjust this frequency range:
Fullness	140 Hz to 440 Hz
Intelligibility	1 kHz to 2.5 kHz
Presence	4 kHz to 5 kHz
Sibilance*	6 kHz to 10 kHz

Also try these additional tweaks:

- To increase brightness and/or open up the vocal sound, apply a small boost above 6 kHz (as long as it doesn't affect the sibilance of the track).
- Treat harsh vocals by cutting some frequencies either in the 1 kHz to 2 kHz range or the 2.5 kHz to 4 kHz range to smooth out the performance.
- Fatten up the sound by accentuating the bass frequencies between 200 Hz and 600 Hz.
- Roll off the frequencies below 60 Hz on a vocal track using a high-pass filter. This range rarely contains any useful vocal information, and can increase the track's noise if not eliminated.
- Create an AM radio or telephone vocal effect by cutting both the high frequencies and those below 700 Hz, while dramatically boosting frequencies around 1.5 kHz.
- Add airiness to a vocal track by gently boosting 13 kHz to 14 kHz.
- Adjust the formant frequencies.

Formant? What's a Formant?

You can use a singer's formant frequencies to your advantage to help a vocal part really stand out. A formant is an area of special resonance within a vibrating body - in this case, the singer's vocal tract. Because every singer's vocal tract (vocal cords, mouth, nose, tongue, lips, etc.) has a slightly different structure, the formant of each person's voice is unique. Even when the pitch changes, the formant stays the same, giving that person's voice its own characteristic sound. (Instruments also have formants, by the way -it's not just voices.)

For men, the main formant range is around 2.5 kHz to 3 kHz, while in women, the range is roughly 3 kHz to 3.5 kHz. There are also low formant ranges: 500 Hz for men and 1 kHz for women. With these formant values in mind, try adjusting the EQ of the vocal track around these frequencies and see what results you achieve.

De-ess al Fine

The final step in EQ manipulation on a vocal track is de-essing. De-essing is a kind of fast-acting compression on specific frequencies. De-essers are useful for controlling sibilance on vocals, wind instruments, hi-hats, cymbals, and other instruments that sometimes produce annoying frequency boosts between 2 kHz and 10 kHz. Most often, sibilance refers to the hissing effect produced when a vocalist speaks or sings an "ess" sound. With some vocalists this "ess" sound is very prominent (and irritating), and needs to be reduced to improve the overall vocal performance.To reduce sibilance, insert a de-esser plug-in on the vocal track, like the DigiRack DeEsser plug-in that comes with Pro Tools. Hone in on the sibilant frequency by pressing the Key Listen button. Then locate the most offensive frequencies by adjusting the Frequency parameter.

The Threshold parameter sets the level at which the frequency-specific compression is activated. Set this level so that only sibilants trigger the de-esser -otherwise you may create unwanted compression and gain reduction on the track.

Some Final Thoughts on EQ

When you're working with EQ on vocal tracks, it helps to keep a few things in mind:

- Not many people can hear a boost or cut of 1 dB or less when altering the EQ on a sound. In fact, most people won't even notice a 2 dB to 3 dB change (except people like us). However, subtle changes like these are often very effective. Large changes in EQ (boosts/cuts of 9 dB or more) should be avoided in most cases, unless an extreme effect is desired.
- Instead of automatically boosting a frequency to make it stand out, always consider attenuating a different frequency to help the intended frequency stand out. That is, try "cutting" a frequency on another track rather than boosting the track you want to emphasize.
- Be aware that any EQ settings you change on a particular instrument or voice will affect not only its sound, but also how the sound of that instrument interacts with all the other tracks in the mix. When altering EQ, don't listen to the track in solo for too long (if at all). You may make the

track sound great on its own, but it might not work with the rest of the instruments.

Summing up

De-essing and tweaking the frequencies of a vocal track can really improve the vocal's sound, as well as help it rise above the other instruments and enhance the emotion of the performance.

Notes :

Tip #1:

Condenser vs. Dynamic Microphones

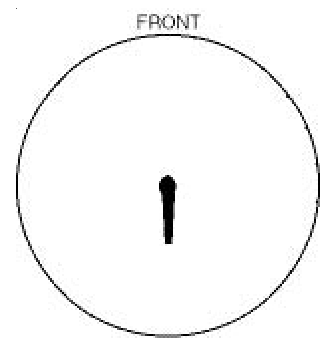
- "Condenser" and "dynamic" refer to two ways that a microphone can convert sound into an electrical signal.
- In a condenser microphone, the diaphragm is a very thin plastic film, coated on one side with gold or nickel, and mounted very close to a conductive stationary back plate. A polarizing voltage is applied to the diaphragm by an external power supply (battery or phantom power) or by the charge on an electret material in the diaphragm or on the backplate - charging it with a fixed static voltage. All Crown mics are the electret condenser type. The diaphragm and back plate, separated by a small volume of air, form an electrical component called a capacitor (or condenser). The capacitance between these two plates varies as the freely suspended diaphragm is displaced by the sound wave. When the diaphragm vibrates in response to a sound, it moves closer to and farther away from the back plate. As it does so, the electrical charge that it induces in the back plate changes proportionally. The fluctuating voltage on the back plate is therefore an electrical representation of the diaphragm motion.
- The dynamic (moving-coil) microphone is like a miniature loudspeaker working in reverse. The diaphragm is attached to a coil of fine wire. The coil is mounted in the air gap of the magnet and is free to move back and forth within the gap. When the sound wave strikes the diaphragm, the diaphragm vibrates in response. The coil attached to the diaphragm moves back and forth in the field of the magnet. As the coil moves through the lines of magnetic force in the gap, a small electrical current is induced in the wire. The magnitude and direction of that current is directly related to the motion of the coil, and the current then is an electrical representation of the sound wave.
- Condenser microphones typically have a wide-range frequency response and excellent transient response, while dynamic microphones typically do not. There are exceptions.
- Condenser microphones' frequency response tends to be uniform, while dynamic microphones' typically is not. There are exceptions.
- Condenser microphones require an external power source (phantom power or battery) while dynamic microphones do not.
- Condenser microphones are easy to miniaturize, while dynamic microphones cannot be miniaturized.
- Condenser microphones are typically used on acoustic instruments and studio vocals. Dynamic microphones are typically used on guitar amps and drums, and for vocal sound reinforcement. However, Crown makes rugged condenser mics for vocal sound reinforcement.

Tip #2:

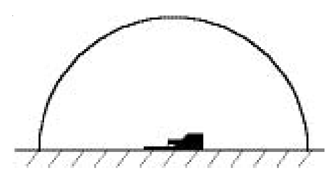
Microphone Polar Patterns

(also called pickup patterns or directional patterns)

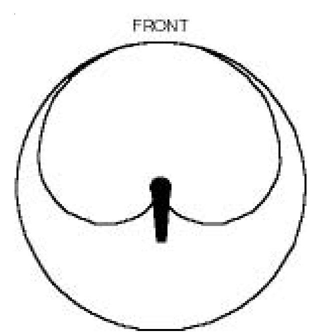
A polar pattern is a graph of a microphone's sensitivity vs. the angle of the incoming sound wave.



The farther from center a point on the graph is, the stronger is the mic signal at that angle.

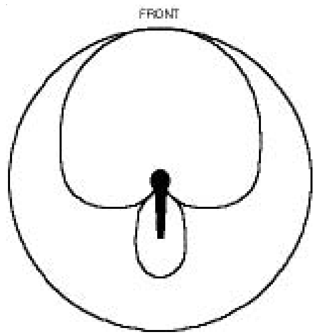


- Omnidirectional: Picks up equally in all directions.
- Half-omnidirectional or hemispherical: Picks up equally over a 180° spherical angle. This is the pickup pattern of PZMs. All of the following patterns are considered unidirectional because they pick up mainly in one direction.

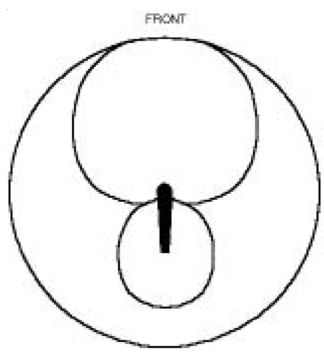


- Cardioid: "Heart-shaped" pattern that offers maximum rejection (null) at the rear of the microphone.

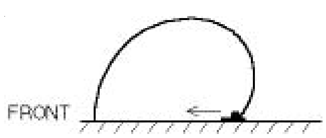
- Supercardioid: Has a narrower pickup pattern than cardioid, but also has some rear pickup. Note that there are two nulls of maximum sound rejection.



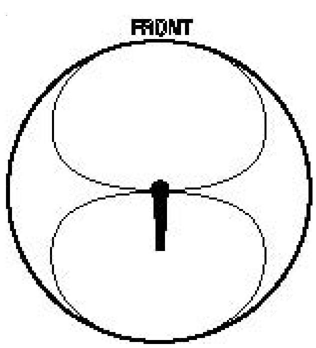
- Hypercardioid: Has a narrower pickup pattern than supercardioid, but also has more rear pickup than supercardioid. Note that there are two nulls.



- Half-unidirectional: The pickup pattern of PCC microphones.



- Bidirectional (figure-eight or cosine): Picks up mainly in two directions (in front of and behind the mic) and rejects sound from the sides.



Traits of Different Polar Patterns

Omnidirectional

- All-around pickup
- Most pickup of room reverberation
- Not much isolation unless you mike close
- Low sensitivity to pops (explosive breath sounds)
- No up-close bass boost (proximity effect)
- Extended low-frequency response in condenser mics. Great for pipe organ or bass drum in an orchestra or symphonic band.
- Lower cost in general

Unidirectional (cardioid, supercardioid, hypercardioid, hemispherical, half-cardioid, half-supercardioid)

- Selective pickup
- Rejection of room acoustics, background noise, and leakage
- Good isolation—good separation between recorded tracks
- Up-close bass boost (except in mics that have holes in the handle)
- Better gain-before-feedback in a sound-reinforcement system
- Coincident or near-coincident stereo miking
- Broad-angle pickup of sources in front of the mic
- Maximum rejection of sound approaching the rear of the mic
- Supercardioid
- Maximum difference between front hemisphere and rear hemisphere pickup (good for stage-floor miking)
- More isolation than a cardioid
- Less reverb pickup than a cardioid

Hypercardioid

- Maximum side rejection in a unidirectional mic
- Maximum isolation—maximum rejection of reverberation, leakage, feedback, and background noise

Bidirectional

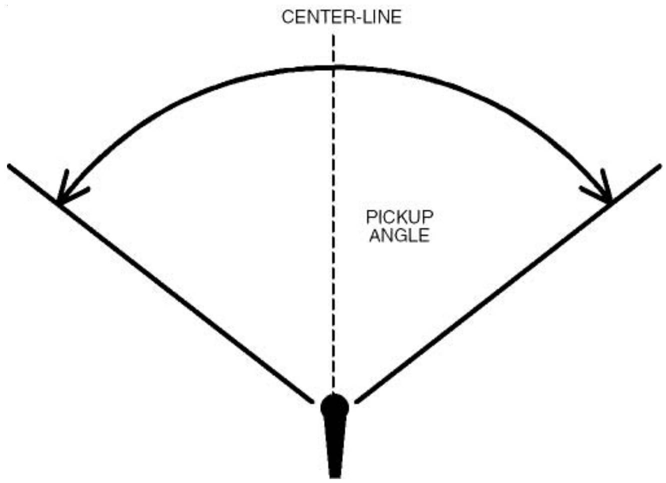
- Front and rear pickup, with side sounds rejected (for across-table interviews or two-part vocal groups, for example)
- Maximum isolation of an orchestral section when miked overhead
- Blumlein stereo miking (two bidirectional mics crossed at 90 degrees)

Tip #3:

Microphone Pickup Angle

Microphone pickup angle: The angle in front of a microphone where a sound source can be located without a noticeable change in loudness.

For unidirectional microphones (cardioid, supercardioid, etc.), the angle from the center-line to the point where the output of the mic is noticeably lower (3 dB down) is one half of the pickup angle. The typical pickup angle of a cardioid microphone



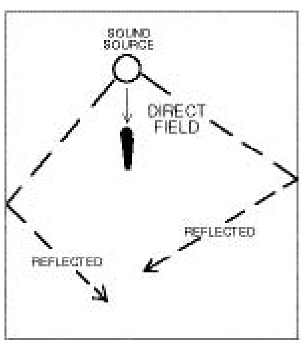
is 131° (65.5° to either side of the center-line of the microphone).

Microphone Pickup Angles

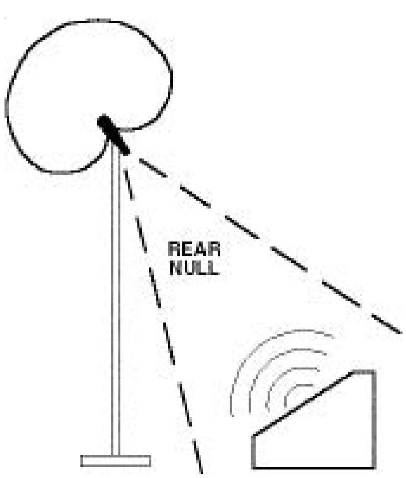
Omnidirectional	360°
Cardioid	131°
Supercardioid	115°
Hypercardioid	105°

Tip #4:

Microphone Performance in the Direct Sound Field



The direct sound field is defined as sound reaching the microphone directly from the sound source without having been reflected off walls, ceilings, floors or other reflective surfaces.



In the direct sound field, the null of the pickup pattern can be directed at an unwanted sound source, thus substantially reducing feedback or sound leakage.

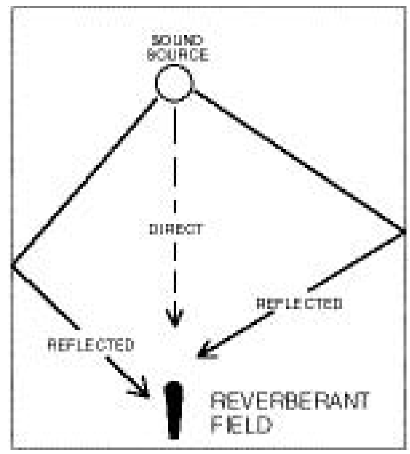
Question: What is the best pickup pattern to use in conjunction with a floor monitor? Why?

Answer: Usually cardioid, because it's easy to aim the rear null of the cardioid pattern at the monitor.

Tip #5:

Microphone

Performance in the Reverberant Sound Field



The reverberant sound field is defined as all sound reaching the microphone that is reflected off walls, ceilings, floors, and other reflective surfaces.

In the reverberant sound field, the null of the pickup pattern cannot be used to control offending sound sources. However, a unidirectional pickup pattern, compared to an omnidirectional pattern, will provide improved gain-before-feedback and lower ambient sound pickup. In this situation, the narrower the pickup pattern, the better the performance, with the hypercardioid being the best, followed by the supercardioid, followed by the cardioid.

As a gauge of performance, a microphone pickup pattern has an associated directivity index (D.I.). The greater the index number, the narrower the pickup pattern.

Pattern	Directivity Index	dB of Reverb Rejection
Omnidirectional	1.0	0 dB
Cardioid	1.7	4.8 dB
Supercardioid	1.9	5.7 dB
Hypercardioid	2.0	6.0 dB

Example: The reverberant field efficiency (D.I.) of a supercardioid is 1.9 times better than that of an omni – a 5.7 dB improvement. This means that the supercardioid picks up 5.7 dB less reverb than an omni when both mics are in a reverberant sound field.

Question: Compared to an omni in the reverberant sound field, how much improvement in gain-before-feedback can be realized by replacing the omni with a cardioid?

Answer: 4.8 dB.

Question: If an omni in the reverberant sound field is placed 1 foot from a person speaking, how far away could a hypercardioid be placed from the person speaking and yield the same result?

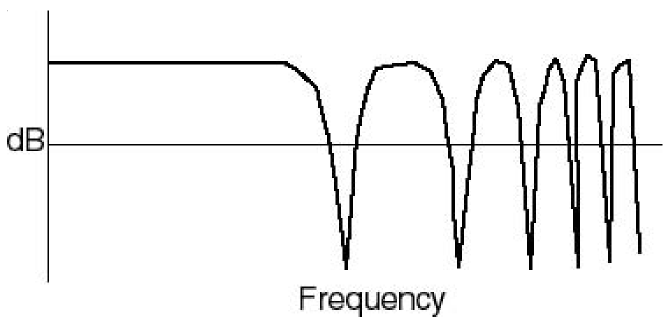
Answer: 2 feet.

Tip #6:

The 3 to 1 Rule

Suppose you have set up several microphones to pick up sound sources. Each sound source has its own close-placed mic. You are mixing the mic signals through a mixer.

Sound from a single source arrives at each microphone at a different time. So, a mic that is distant from the source is picking up the source with a delay, which causes variable phase



shift vs. frequency. When you combine the close and distant mic signals in your mixer, certain frequencies cancel out due to phase interference, creating a "comb-filter" effect. The frequency response of a comb filter has a series of peaks and dips (see figure below.) This response often gives a thin, hollow, filtered tone quality.

Audible comb filtering can occur whenever two or more mics pick up the same sound source at about the same level but at different distances, and are mixed to the same channel. This problem can be minimized or eliminated by following

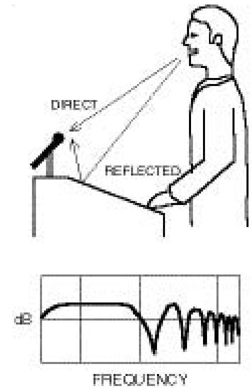
the 3:1 rule: Separate the mics by at least 3 times the mic-to-source distance. This creates a level difference of at least 9 dB between microphones, which reduces the comb-filter dips to an inaudible 1 dB or less.

In general, place mics close to their sources and keep the mics far apart to prevent audible comb filtering.

This figure shows how to mike two sound sources with two mics while following the 3:1 rule. If the mic-to-source distance were 2 feet, the mics should be at least 2x3 or 6 feet apart to prevent audible comb filtering.

The left-side frequency response results when two mics are mixed to the same channel at equal levels, and you followed the 3:1 rule.

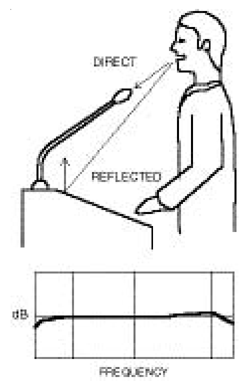
The right-side response results when you don't follow the 3:1 rule.



Tip #7:

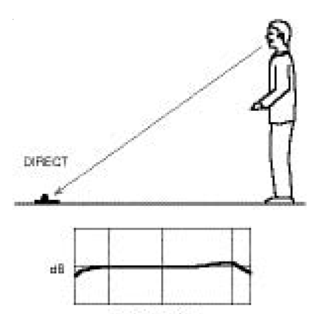
Microphone Techniques for Lectern and Stage

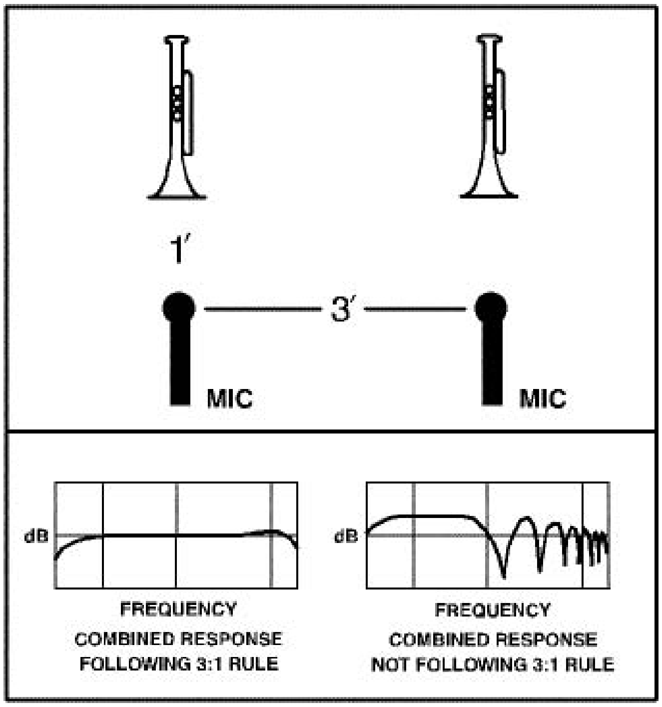
A microphone positioned above or near a reflective surface receives a direct signal from the sound source, and a reflected signal from the surface. Because sound takes time to travel, the reflected signal arrives at the microphone later than the direct signal. The direct and delayed signals combine at the mic diaphragm. This causes an uneven frequency response (see below) called a "comb filter effect," which results in an unnatural sound quality.



The figure above shows a poor way to mike a person at a lectern. The microphone is too far away from the mouth, resulting in pickup of reflected sound from the lectern's surface. This will result in an audible comb-filter effect, which sounds hollow or tonally colored.

The figure below shows a better way to mike a person at a lectern. The microphone is close to the mouth (about 8 inches). The sound reflected from the lectern arrives toward the rear of





the mic, where sound is rejected. This will greatly reduce the audible comb-filter effect.

The figure below shows an effective way to mike an actor on a stage. Use a boundary mic, which is designed with the mic capsule on or very near the reflected surface. In a boundary mic, the reflected path is nearly equal to the direct-sound path length, so the direct and reflected sounds are in phase. This will greatly reduce the audible comb-filter effect. To reduce feedback, use a Crown PCC-160 boundary mic, which is directional. For maximum clarity and gain-before-feedback, use a wireless lavalier mic on the actor.

Microphones

This is a list of the microphones that are available for use in the studio.

Shure SM-57 -Classic dynamic microphone and a perennial workhorse of the recording industry. Many top engineers use it for electric guitar, snare drum, gritty, rock and blues vocals, and many other things.

Shure SM-58 -A dynamic microphone, just like SM-57 -the same mic, but a bit more of a mid to low end boost -use this one for vocals.

Shure KSM32 -A side-address, cardiod pattern, large diaphragm condenser microphone. This microphone requires 48V of phantom power. The microphone is addressed from the side with the Shure logo. The KSM32 offers an extended frequency response for an open, natural sounding reproduction of the original sound source. We have 2 KSM32s so they are an excellent choice for stereo applications. The KSM32 has a switchable low-frequency filter to provide greater flexibility to reduce background noise or to counteract proximity effect.

Flat Response -use this setting for any application where you desire the most natural reproduction of the source. The microphone will reproduce ultra-low frequencies, and the suspension shock mount should be used to reduce mechanical vibrations transmitted through the microphone stand.

Low Frequency cutoff-provides 18dB-per-octave cutoff at 80Hz. Helps eliminate stage rumble or other low-frequency room noise such as that coming from heating, ventilation, cooling (HVAC) systems and computer noise. It may also be used to compensate for proximity effect or to reduce the low frequencies that can make an instrument sound dull or muddy.

Low Frequency roll off -provides 6 dB-per-octave roll off filter at 155 Hz. Use this setting with vocals or instruments to compensate for proximity effect or to reduce the low frequencies that can make an instrument sound dull or muddy.

The mic has a built in three-stage pop protection grille to reduce “pop” and other breath noise making it an excellent choice for vocals. I used this for most of my vocal recordings before we got the Neumann TL-103.

Setting Attenuation on the Shure KSM32 -the attenuation switch on the back of the KSM32 reduces the signal level from the cartridge by 15dB without altering the frequency response. This can prevent extremely high sound pressure levels from overloading the microphone. To activate the attenuation, move the switch to the “-15dB” position.
In situations where the high output capability of the KSM32 might overload the microphone preamplifier of a console or mixer, use an attenuation switch or pad in the mixer, rather than on the microphone.

Recommended Applications for the Shure KSM32:

Voice -solo, background, voice over, broadcast.

Acoustic instruments -such as piano, guitar, drums, percussion, strings.

Wind Instruments -brass and woodwind.

Low frequency instruments -such as double bass, electric bass, kick drum.

Overhead drum mic-ing -drums or percussion.

Ensembles -choral or orchestral.

Room ambience pick-up -guitar amplifier or drums.

Sennheiser MD 421 II -A very popular dynamic microphone. It has a defined, rich tone that works well on electric guitar and drums, and it features a useful onboard low-cut filter. The 421 is also a highly regarded voiceover microphone that offers an instantly identifiable, radio ready sound.

MicroTech Gefell M300 -A compact, small diaphragm condenser microphone with a cardiod polar pattern. The frequency response is practically linear for a wide range of sound incidence and has a smooth treble boost rising to about 3 dB between 6 and 10 kHz. Small diaphragm condenser microphones are well suited for vocal and instrumental soloists. This microphone requires phantom power.

We have a pair of M300s so they’re a good choice for stereo mic-ing applications such as acoustic guitar, cellos, percussion, drum overheads and really any where you want to capture the sound realistically, “as is.”

Sony ECM-999 - Electret Condenser Stereo Microphone. This microphone employs the Mid-Side system to give excellent sound image and faithful stereo sound reproduction with less "hole in the middle." The directive angle between the left and right channels can be changed progressively from 0 degrees (monaural) to 150 degrees according to the sound source. The sum of signals of the mid microphone unit (uni-directional) and side microphone unit (bi-directional) and the difference between them are used for the right and left channels respectively. The directive angle controller should be set as follows depending on the use. 0 degrees for a monaural recording. To pick up a narrower sound source such as a soloist, locate the microphone close to the sound source, set between 90 and 120 degrees. To pick up a wider sound source such as a chorus, orchestra, natural sounds, urba n sounds, speeches at conferences, etc. place the microphone at a greater distance and set between 120 and 150 degrees.

Royer R-121 - A side address, Ribbon-Velocity studio microphone. The R-121 is a thoroughly modern ribbon microphone designed to meet the demands of today's studio environment, exhibiting a flat frequency response and well-balanced panoramic soundfield.

The R-121 is an ideal choice for digital recording. The R-121 delivers a consistent, natural acoustical performance with stunning realism. Its figure-8 pattern delivers superb ambience when used for room mic-ing applications, orchestral and choral recordings

The figure-8 pickup pattern allows the R-121 to be addressed from either side with equal sensitivity. The in-phase signal is achieved when the microphone is addressed from the front, indicated by the Royer logo.

Ribbon microphones are somewhat more sensitive to direct blasts of air. Discretionary use of a windscreen or a pop-filter is highly recommended for situations such as close mic-ing, especially with vocalists or certain types of percussion and wind instruments.

The Royer's smooth frequency response characteristics and ability to capture detail make it a fine choice for many instruments, as well as broadcast applications. Its gentle low-frequency proximity effect makes it especially useful for announcers and vocalists. Female vocalists often benefit from the R-121's ability to capture high frequencies without distortion or edginess. Orchestral instruments are captured in a very natural sounding way and free from microphone-induced "hype." The R-121 has exceptionally smooth high frequency characteristics. Phase-related distortion and irregular frequency peaks are conspicuously absent. Theater organs and electric guitar amplifiers sound big and fat, without unnatural coloration, when reproduced with the R-121.

Recommended applications include: close mic-ing, electric guitar, overhead drum mic-ing, percussion instruments, brass instruments and horn sections, and woodwind instruments. Acoustic pianos can be captured accurately without the comb-filtering effects associated with condenser microphones. It's also good choice for acoustic guitar and harp.

The Royer works well on electric guitar coupled with a Sennheiser 421. The Royer works well on horns and percussion, and is great on violin and any other potentially "scratchy" sound source. The R-121 is also excellent for acoustic bass, cello, brass instruments, and hand drums.

Neumann TLM 103 -A studio microphone of the FET 100 series with a cardioid polar pattern. The letters TLM stand for Transformerless Microphone. This microphone requires 48V of phantom power. The TLM is addressed from the front, marked with the Neumann logo. The grill houses the large diaphragm K 103 capsule. It has a linear frequency response up to some 5 kHz with a wide flat presence boost of 4 dB at the top end. The capsule is based on that of the U 87 microphone and uses their back electrode and diaphragm.

No resonance effects are used to obtain the characteristics mentioned above. As a consequence, the microphone features excellent transient behavior and transmits all transient phenomena of music or voice without distortion.

As the TLM 103's amplifier is linear below 20 Hz, extremely low frequency signals can be transmitted without distortion as well.

On the other hand the microphone is therefore more sensitive to low-frequency noises like structural-borne or wind and pop disturbances. For specific applications it is therefore recommended to use protective accessories such as pop screens and wind screens.

The TLM 103 is a great large diaphragm condenser microphone. It makes a great ambient drum-room mic, and it is a solid choice for vocals, acoustic instruments, percussion, electric guitar (as an ambient room mic -do NOT use to close mic a guitar amp!), and other sources.

AKG D224E -A small condenser microphone. This microphone requires 48V of phantom power.

When to use the Different Microphones.

Drums -use a large diaphragm condenser on the kick drum and dynamic microphones on the snare and toms. Use a pair of small diaphragm condenser microphones as overheads.

Bass guitar amp -large diaphragm dynamic mic used in conjunction with DI.

Electric guitar -use two dynamic microphones -or the Royer ribbon as the room mic and the Sennheiser 421 up close.

Acoustic guitar -small matched pair of condenser microphones to stereo mic guitar, piano, or a percussion setup.

Vocals -use a large diaphragm condenser microphone.

Tougher vocal sound -use the Shure Sm-58 or 57.

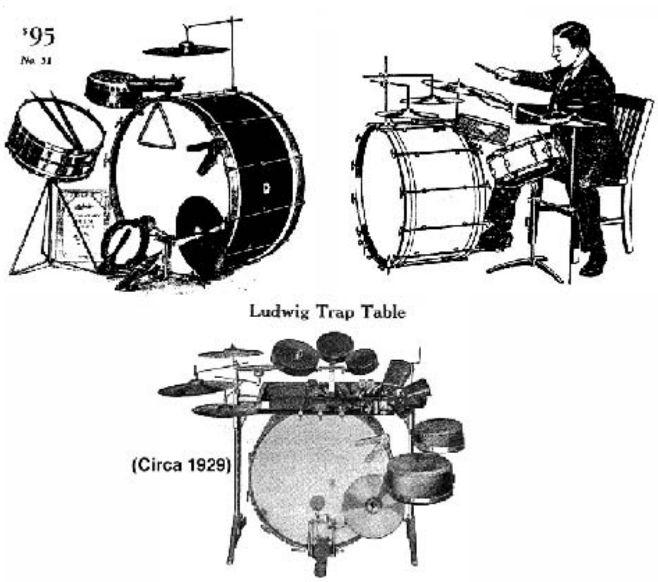
Room Ambience -large diaphragm condenser or the Royer R-121 ribbon microphone.

A Brief History of Sound Effects

Drum sets are peculiar things. The more you think about it the more you realize that they might be the only traditional (indigenous?) musical instruments that are actually made up of other instruments. This probably evolved out of myriad of things -part circumstance, part chance and part necessity.
In the early 1900s, bands and orchestras had two or three drummers each. One played the bass drum, one played the snare drum, and a third (the"utility man") played cymbals, wood blocks and sound effects (or " traps " and " trappings", as they were commonly known - a term which was taken from the word "contraption").

The need for all those people changed with the inventions of things like the bass drum pedal and snare drum stand. With them, one person could sit behind the bass drum and play it with his or her feet, which freed the hands to play the snare drum, cymbals and other effects. Snare drum players prior to this either hung the drum from a sling around their necks or placed the drum on a chair or table. And so was born the drum set, "trap set" or "kit."

Typical early kits consisted of a snare drum and a bass drum from which hung cowbells, wood blocks, triangles, Chinese Tom Toms and cymbals held or suspended by various clamps and rods.



In the years since this has been further expanded upon and customized by the tastes and needs of the performer or situation, with various devices and components being added, adapted, invented and re-invented still to this very day.
Back in the 60s lot of kids tried out things: Improvised and appropriated various objects such as pot lids or hubcaps for cymbals, pans or large round cardboard tubes with lids for extra tom toms, etc.

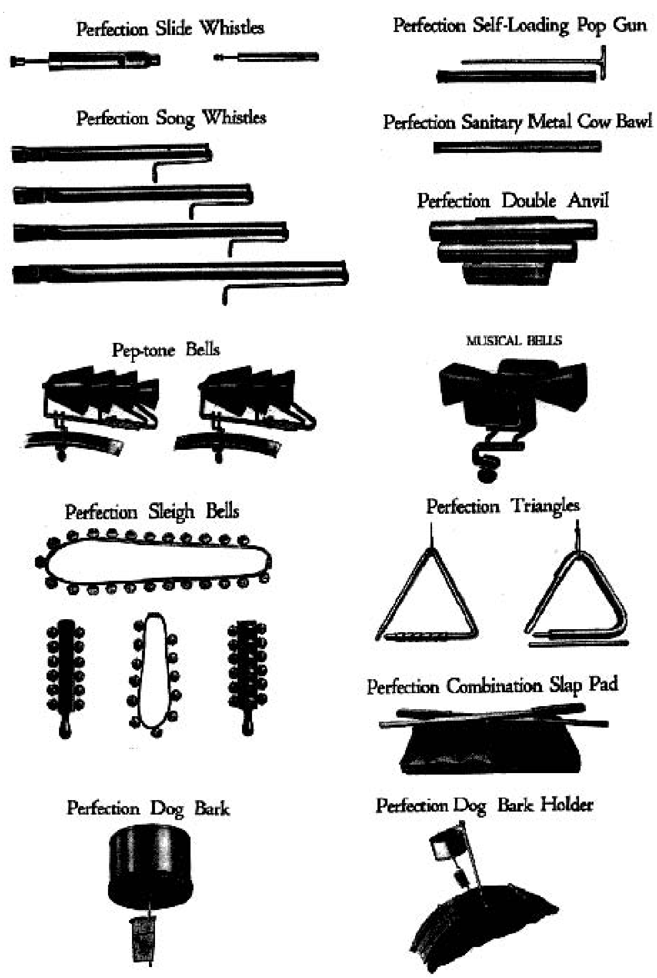
At the time there were various sounds effects that the catalogs offered such as the ACME siren, different bird calls, wood and temple blocks, ratchets, triangles, pop guns, slap sticks, slide whistles, sleigh bells, etc.

[Above from Yerkes Catalog]

It was strange that you found all these items included in a drum catalog. But if one was exposed to music, old movies and shorts, cartoons, catalogs and articles, one would realize the importance of these effects to those early drummers.

And that's when one realized how effective, expressive and how much fun sound effects could be not just in music, but in everything ranging from early dramatic theatrical productions, to Burlesque and Vaudeville, Radio & TV, cartoons, film and so on. You would see that the older a catalog was, the more sound effects it seemed to offer.

Indeed, there were companies that specialized in sound fx. In the 1900s, sound effects were just as important and vital an item as any instrument. The Walberg & Auge Perfection catalog of 1915 lists about 40 different types of bird calls and whistles alone. These included song or slide whistles, ocean liner whistles, train whistles, ferry boat and fog homs, etc., not to mention various wood blocks, castanets, rattles, slap or shot pads (for making gun shot sounds), fourteen different types of



bells, railroad & locomotive imitations, pop guns, dog barks and more.

Sound Effects (aka Sound FX & SFX) have been providing illusory sounds for audiences as far back as the theater of Aeschylus, Euripedes and Sophocles. For audiences in ancient Greece, the rumble of thunder was provided by “leaden balls bounced on stretched leather.” In Shakespearean times the popular method was “rolling a cannon ball down a wooden trough which then fell onto a huge drumhead.” Large single-headed frame drums, often used in pairs, were also used to produce thunder sounds. In his book Magic, Stage Illusions, Albert A. Hopkins describes how in large opera houses a more complicated system was used. A cabinet with five or six slanting shelves in it was placed up high against a wall backstage. Each shelf held about six cannon balls that were held in place by a hinged door. When the signal was given, a stage hand would open one or more of the compartments and the balls would drop down into a wooden trough about 20 feet long which had irregular surfaces in it lined with zinc plates, adding to the rumbling effect. At the end of the trough the balls would drop through the floor to the gallery by means of special slants, and provisions were made so that the balls could be stopped anywhere along their path. By regulating the number of balls

dropped, almost any type and length of thunder effect could be produced.

Another device Hopkins mentions was called the “Rumble Cart”. This was a large box filled with heavy objects or material and mounted upon irregularly shaped wheels. When pulled or pushed along the floor backstage it produced a thundering effect.

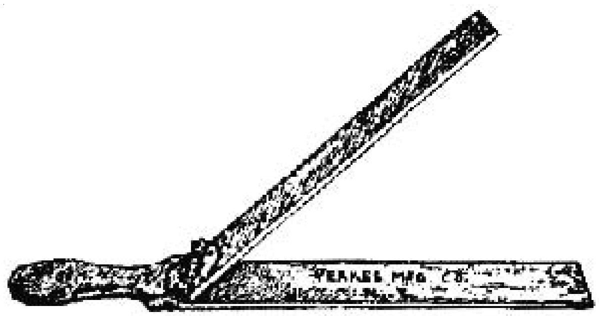
In 1708 John Dennis invented a new method for providing thunder for one of his plays by vertically suspending a large copper sheet by wires with a handle on one end and giving it a shake whenever the effect was needed. His play wasn’t a success, but his thunder effect was, and other producers started borrowing this idea. Every time he would encounter someone

using his effect he would accuse them of "stealing his thunder," and that's how that expression came into the English language.

Another method for use in Radio, invented by NBC sound FX artist Stuart McQuade, was called the "Thunder Screen".

[Thunder Screen photo from Robert Mott, Radio Sound Effects: Who Did It, and How, in the Era of Live Broadcasting]

This was a copper screen stretched onto a frame onto which a contact microphone was attached. Striking the screen with a small mallet produced the thunder as well as other explosive



WHIP CRACK
(Spring-hinge) can be operated with one hand. $1.20

sounds. This was probably similar to (though more primitive than) the sounds that a spring reverb tank in an electric guitar amp makes when bumped against or shaken.

Slapstick comedy actually takes it's name from a device called a "Slapstick" (aka "whip crack").

Those old comedies and variety shows were interesting to watch. Laurel & Hardy, Chaplin, Keaton, the Marx Brothers, the Little Rascals, Keystone comedy shorts, Red Skelton, or Jackie Gleason, not to mention any Fleischer Brothers cartoons, Walter Lantz's Woody Woodpecker, Jay Ward's Rocky & Bullwinkle, and any Warner Brothers cartoons featuring Mel Blanc, who did most of the voice characterizations and vocal imitations, and Treg Brown who did all the wonderful SFX. (Incidentally, it was radio actor Arthur Q. Bryan that provided the voice of Elmer Fudd.)

Three Stooges seemed to have more sound effects per square inch of film than anything except for cartoons. Joe Henrie (and later, Morrie Opper) were responsible for dubbing the sound effects for the Stooge shorts. Well you could call them the Fourth Stooge (and half of the fun). Next time you see a Three Stooges short, check out how prominent and up-front the FX are mixed into the sound track. Not only are they funny FX, but they are larger than life.

Some of the techniques and methods used for the Stooges' antics were: cracking walnuts for knuckle crunches, ukulele or violin plinks for eye pokes, a muffled kettle or bass drum hit for a bump in the stomach, hitting a rolled up carpet with a fist for body blows, various ratchets or twisting stalks of celery for when ears or limbs were twisted. The glugging/drinking effect was done by pouring water out of a one gallon glass bottle into cotton batting (which would muffle the splashing).

Many of the sound FX used in the Stooges shorts were first recorded for use in a Walter Catlett short called "Get Along Little Hubby" in 1934. On the re-recording stage, the studio's production chief was infuriated when he saw producer Jules White mixing these outrageous Sound FX in a reel, and stopped the session. Jules White insisted that the comedies needed the "cartoon quality" of sound effects in order to accent the visual gags. The two men argued, and Jules said he would abide by whatever audience reaction the film short got at the preview. The audience response proved Jules was right, and from then on sound FX were a key ingredient in the Stooge shorts.

Cartoon sounds not only emphasize the action in an absurd and amusing way, but can also convey emotions. In early cartoons (such as the Fleischer Betty Boop cartoons), drummers' traps were used extensively for sound FX. Not only did they furnish funny sounds for the action, but they were often used to compensate for lack of dialog, where an irate character was often accompanied by angrily played duck call.

Naturally, a brick on the head would hurt, but when it's coupled with the hollow sound of a temple block, it suggests that the person's head is empty. This magically transforms a violent act into a humorous one and makes it less painful. I can only imagine how this may have been expanded upon each time by the sound effects artists, editors, producers and directors who might have given it their own signature, extending the idea for the gag as it evolved into a hysterical (historical?!) sequence of events - adding a slide whistle as a character slid down the wall, then dropping a sack of potatoes or foot locker on the floor as he hit the ground, then adding canaries singing to suggest being unconscious. (Incidentally, it was producer Jules White's whistling talents that provided a lot of the bird chirping FX in the Stooge shorts.)

Ed Bernds, who directed some of the Stooge shorts, thought that Joe Henrie "was damn close to being a genius as a sound FX editor," and said "he was never satisfied with a sound effect until he had gotten it as startling and as funny as he could make it." It was in the early 40s that Joe Henrie was replaced by Morrie Opper (Henrie left to join the Armed Services in W.W.II). Morrie added the violin plinks whenever someone got poked in the eye. Stooge shorts prior to this did not have any sound effect associated with this gag. (You did hear a ratchet or crunching effect during this interim period.)

Ed (who was also a former sound mixer himself) recalls having to add sound to a previously released silent picture called "Submarine" when he first came to Columbia back in 1929. The picture was being remade as a "sound effects picture" and he remembers a "dreary all night session on the sound dubbing stage trying to sync noises live while watching the screen.

One interesting note is the use of sound FX in the currently popular Simpsons cartoons. Most of the FX used in the show are not of the cartoon variety. They consciously use real sounds for most of the action on the screen to make it more believable, and the only time you hear cartoon-type sounds used are in the "Itchy & Scratchy" segments.

It was the genius of SFX artist Monty Fraser, working in the popular Fibber McGee & Molly shows on live radio, who created the sound of a tremendous pile of junk falling out of an overstuffed closet every time Fibber opened the door. This became one of the most listened-for sounds in Radio and was a highlight of every show.

Incidentally, Spike Jones was the drummer in the band for some of the Fibber McGee & Molly radio broadcasts. Before the inception of the City Slickers, a fellow musician recalls that "Spike approached comedy through the back door. He had some funny contraptions on his drums that didn't come out of a drum accessory catalog. Things that he put together himself." Spike Jr. says, "It was often that little extra something that got Spike the gig for whatever he was auditioning for. His trap set was very embellished and he was sought after because of this. Most drummers just had a xylophone and a basic set of drums." Among Spike's other credits were the Al Jolson Show, The Burns & Allen Show, and The Eddie Cantor Show as well as becoming the staff drummer for CBS and performing on many other record dates.

With the City Slickers, Spike used tuned cowbells, bulb and klaxon ("ah-oo-gah") homs, blank pistols, small cannons, bird calls, washboards, wheel rims, temple blocks, jaw harps, sirens, vocal imitations & FX, razzer whistles (a short flat thin tube of

flexible rubber with a wooden mouthpiece on one end which when blown gives you "the raspberry" or "Bronx Cheer").

Some of the musical contraptions Spike and his members created were amazing and humorous works of art. One of the instruments was a toilet seat strung with catgut called the 'Guitarlet" (aka "Latrinophone"). Another one was made from old bug sprayers (known as "Flit guns"). These had two accordion reeds in each gun, which would produce two distinct notes depending on which way it was being pumped. When you got four or five people playing these things together you could play a song.

In a 1988 PBS/Storyville Films documentary on him, Spike is seen explaining how he got the idea of combining funny sound FX with music while attending a concert once by Igor Stravinsky conducting The Firebird Suite. Apparently Stravinsky had on a brand new pair of patent leather shoes on, and every time he would rise to give a downbeat his shoes would squeak. Spike had a lot of trouble suppressing his laughter at this (as well as infecting others around him) and went home thinking that if he could plan ,"musical mistakes" or substitute sound FX in the place of notes in the score he might be able to get some laughs out of people.

By the mid 30s he acquired a penchant for collecting any kind of junk that made a funny sound or noise and indulged himself whenever possible. His collection began to fill his apartment and take over his car as well. In 1934 he finally formed a novelty band (originally called "Spike Jones and his Five Tacks"), got some of his friends over and made some home recordings, and that's how he got started.

But that's one story. (there are many ...) Another story goes that he got the idea from when he missed hitting the opening chime on the Bing Crosby show and out came a loud thud, causing the band and audience to go into gales of laughter. And yet another story he told was that he was simply inspired by other comedy bands of the 30s such as The Korn Kobblers, Hoosier Hotshots, and the Schnicklefritzers, and decided to form his own comedy band.

As if Spike's zaniness wasn't enough, when he got his own TV show he even had a sound effects person from the studio, named Parker Cornel, working behind the scenes.

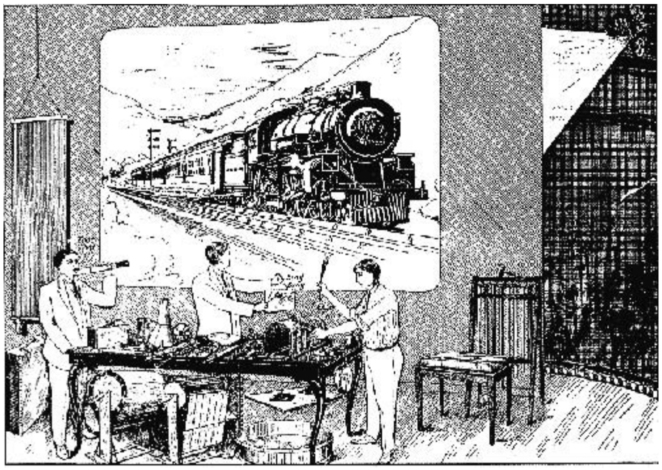
Joe Siracusa (Spike's drummer from 1946-52) also went on to become a very successful Sound FX man whose credits included the sounds for Alvin and the Chipmunks, Rocky & Bullwinkle, Mr. Magoo, Gerald McBoing Boing, (a cartoon about a boy who spoke in sound effects instead of words), and a number of Warner Brothers cartoons.

Mel Blanc was also known to grace the ranks of Spike and his crew from time to time as well.

Spike Jones was not the first to use funny sound FX in music, but he was probably the first to make it popular. Verdi's comic opera Falstaff has a sequence involving a "deflating chromatic scale of brass" which has been referred to as "Spike Jones before his time," and Haydn and Beethoven were also known to be very fond of musical jokes.

Jordan Young, author of Spike Jones - Off The Record (highly recommended!) says the explosive possibilities of combining music and humor were not fully explored until the dawn of the record business. French musical comedy star Maurice Farkoa recorded his famous "Laughing Song" back in 1896, and Ed Meeker made his Edison cylinder recording of "I'm a Yiddish Cowboy" back in 1909. Jordan credits him as "perhaps the first to use sound effects to accompany each line of a song, instead of throwing them in haphazardly."

The Six Jumping Jacks were distinct predecessors to Spike Jones and the City Slickers. They were among the earliest bands to play novelty songs and include sound fx in a precise manner in the tunes in appropriate places. Irving Aaronson and his Commanders made several humorous recordings in the late 20s with sounds such as barking dogs, birds chirping and cute childlike vocals (all of which Spike Jones would later claim as his trademark), and The Kidoodlers were a male vocal quartet which toured Vaudeville in the 30s and augmented their voices with over 100 toy instruments and props.



We could say the golden age of sound FX was the 1900s through the Vaudeville & Burlesque era and into the days of early Radio & TV. Comedians and performers in Vaudeville and Burlesque often called upon the drummer to accentuate highlights in their routines or execute a sound effect for a gag or pratfall (which literally means "a fall on the buttocks"). It's such a classic mode of inherited expression that it's still used today. Just watch David Letterman or any other talk show. Inevitably there always seems to be a point in the dialog, which elicits a "Badda, Dum"..., or a cowbell clunk from the drummer.

As stage productions and acts grew more elaborate these early show drummers were called upon to come up with more inventive and outrageous sound FX. During the silent film era not only did theaters have a pianist or organist to accompany a film but they often had SFX artists behind the screen creating sounds to accompany the picture.

Caption reads:"A Practical Demonstration of the simplicity and effectiveness of YERKES' SOUND EFFECTS. Audience delighted with the absolute realism of the Picture. If you use YERKES' SOUND EFFECTS, you need not worry about box office receipts."

Johnny Cyr (another drummer Spike used on occasion) recalls playing for silent pictures as a boy of 9 or 10 years old with his violin teacher, a piano player and himself on traps and sound effects.

In Japan during the silent film era there were men called "benshis." A benshi was a man who stood on stage next to the screen and not only did the dialog for both the men and women on the screen but the film's sound effects as well. These men were so popular that many people just came to see the benshis no matter what film was playing. Watching them do the ranting of an outraged emperor, the protestations of his suffering wife and the growls of the royal Pekinese all in the same scene was well worth the price of admission.

The Yerkes Mfg. Co. provided many devices for the sound effects field. This illustration was taken from a rare 1910 Yerkes catalog, courtesy of William Ludwig Jr.

As Vaudeville began to die out and Radio came to be the more popular form of entertainment, many of these show drummers and their traps were recruited to do sound FX for radio (and later TV), adding a sense of realism that suggestive dialog alone could not create. Prior to the use of sound effects in early radio broadcasts, some directors and producers resorted to confusing dialog like "I wonder why that car is stopping in front of my house?" Of course the audience didn't actually hear a car drive up so it was decided that using dialog to describe sounds couldn't continue, and what was needed was for the audience to actually hear the sounds (or what they imagined were the actual sounds). Radio producers then turned to the sound effects that had been used in theatrical productions for countless years, and adapted and improved on them for broadcast use.

SFX helped relieve radio's incessant talking and allowed the audience to use its imagination.

Whenever a producer needed an effect to help get an idea or mood across or to add more realism to the story, it was the sound effects men and women who had to create that illusion, whether by using the natural characteristic sound of an object, or an imitative, suggestive, or implied sound.

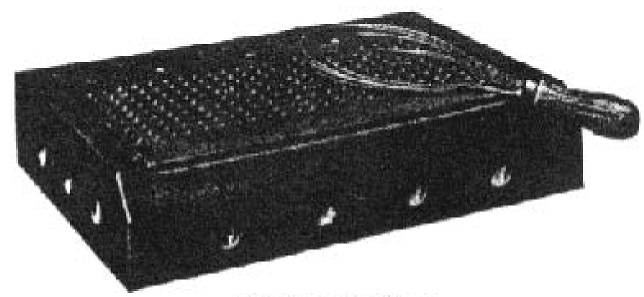
Ora and Arthur Nichols are regarded as the two people most responsible for bringing sound FX to Radio, bringing several sound effects props from their many years of theatrical and silent film experience. Arthur once spent nine months working fourteen hours a day to build one multiple-effects device. It had nine-1/8 horsepower motors, one 1-horsepower motor, and tanks of compressed air for operating all the bird and boat whistles. The machine was five feet high and two feet deep and could reproduce sounds ranging from a small bird chirping to 500 gunshots a minute.

There were, however, producers that insisted on using the real thing instead of something that sounded like and easily passed for the original sound. On one occasion, Orson Welles was doing a radio play for the Mercury Theater On The Air. The play took place in the desert and he had the floor of an entire studio filled with sand because he wanted the actors to do their own footsteps instead of a sound effects artist mimicking their steps

in a sand box. He later realized this was a bad idea because he had to turn the mics up so loud that they picked up every little noise in the studio and the actors dialog overloaded the mics.

On another occasion there was a scene which called for a man to be cutting his lawn, so he had half the studio filled with sods of grass with the actor pushing a real lawn mower instead of the "usual" way this effect was done, which was feeding strips of newspaper into a more compact, stationary version of a lawnmower. The only problem was - Orson couldn't make the grass grow between the rehearsals and the live broadcast!

The advent of real sound effects recorded at the source and put onto 78 rpm records greatly aided the sound effects artist in his work, along with other electronic devices. These were invented to replace manual or imitative effects, some of which either took up enormous amounts of space, or were unreliable or awkward to use. Some of the sound effects departments in



Heavy service type
For use in Bands and large theatres

these radio stations were massive. At one point CBS had 40 people working in its sound effects department, and the prop rooms were overflowing with equipment and devices to make sounds.

[Photo on the top is from Sound Effects: Radio, TV and Film (Focal Press); the next two are from Radio Sound Effects (MacFarland), both by Robert Mott]

Some of the more common early manual effects were different size wooden doors and frames (even metal car doors), and large single-headed drums containing buck shot which provided the sound of ocean waves when slowly rolled around. A device that dropped birdseed onto a piece of stretched wax paper created the sound of rain. Filling a large shallow box with cornstarch and walking in it created footsteps in the snow. There were Splash Tanks for creating watery FX, Wind Machines (which produced blustery wind-like noises), "Nail Pullers," Barking Dog FX, Lion Roarers, and innumerable other devices adapted and invented.

FX for creating locomotive sounds were made with a "scratch box," which was a shallow wooden box and a piece of tin with holes punched in it. (There were many variations.)

By rhythmically sweeping a stiff wire brush over the holes a steam engine chugging effect could be produced. Directors expected this chugging sound to accelerate rapidly as the train pulled out of the station and continue for as long as needed, or until the artist's arm fell off, which is why sound effect departments relied so heavily on hiring drummers.

Sadly and unfortunately I've learned from Mr. Mott that most of these effects and special one-of-a-kind props have either been demolished and thrown away by the studios, or stolen. There

has been some talk about creating a museum for these priceless artifacts which have played such an enormous part in the history of theater, Radio and TV, but nothing has been established so far.

The need for more sophisticated SFX that couldn't be produced by the old manual FX used in the theater became a problem in the early 30s, and many artists began recording sounds on 78 rpm records.

Unfortunately, sound recorders in those days were about the size and weight of a large washing machine, and the sounds were limited to those that could be produced in a studio. Soon, more portable equipment was developed that enabled the SFX artists to go in the field to record realistic SFX, and Radio was given another element of realism.

In order to use these new sound FX records, modified phonograph players with two tone arms were made. They had potentiometers in them so that if you needed the sound of wind or waterfalls behind a long scene, you played the sound with one pickup arm, and as the needle was coming to the end of that cut you cross-faded, slowly turning down the sound from one pickup while turning up the sound from the other pickup arm which was placed on the beginning of the same cut.

[The above photo and the next are from Radio Sound Effects by Robert Mott (MacFarland).]

In addition to this double-arming capability, the motor speed could be increased or decreased with a potentiometer control, which added a great deal of potential variety to a sound. A recording such as the Mogambi Waterfalls could produce such varied sounds as gunshots, surf, traffic (by adding horns), or a jet airplane. It was even used as the sound of an atom bomb exploding in a news broadcast by SFX artist Robert Mott.

Sometimes you practically had to be an octopus if you were using several records for a show. If there was only one effect needed from each record, you ran the risk of not picking up the needle in time and playing a fog horn in the middle of a jungle skit, as each record usually had a few different, unrelated sounds on it.

Another problem was that as this method caught on, the same car crash and skid could be heard on consecutive different shows. It was realized that more varieties of a particular sound were needed. This was eventually solved by advancements in making recording equipment more portable so that SFX artists could go just about anywhere on location and record more varied sounds.

Another advent was the development of re-recording technology. In 1933, on the soundtrack to King Kong, Murry Spivak (head of the SFX department at RKO) used this technique to create and manipulate SFX for King Kong. He recorded the sound of a lion roaring, then took that back to the studio and re-recorded it at half speed (dropping it down an octave) and then mixed in the sound of a tiger roaring backwards. The result was stunning, as no one had ever heard an effect like that before.

Purely synthetic sound originated with the work of Reuben Marnoulian in the 1932 film Dr. Jeckle and Mr. Hyde. His effects included exaggerated heartbeats, reverberating gongs played backwards, and expressionistic bells. In addition, he experimented with direct optical recording, using the Fischinger system of painting the track edge of the film with light photographed to create frequency patterns that produced sounds upon playback.

Another company that specialized in manual effects and special fx instruments was the Carroll Sound company.

They sold such exotic things as: "Marching Men Effect,' hand-crank air raid sirens, NBC chimes, "boing boxes," musical saws, boo bams (a percussion instrument made out of tuned bongo heads with square wooden resonators and laid out like a marimba), and many other wonderful devices. We can only assume it was the advent and advances of electronic technology like Makenzie Carts, synthesizers and samplers that caused their demise.

Makenzie Carts were an endless tape loop/cartridge system, which could be cued up instantly with whatever pre-recorded sound& you wanted and mixed simultaneously with other carts. Each Makenzie machine had five tape mechanisms and other Makenzies could be stacked and linked together.

One unusual electronic device from the early 50s that was used in radio and TV was called the "Foster Gun", invented by SFX artist Don Foster.

The sounds from this were so realistic that the Marine Corps was interested in using it to accustom new recruits to the sounds they might encounter in battle. It could produce everything from a single shot to a machine gun, or a ricochet to an explosion, all at the flick of a switch to find the proper frequency. Prior to this, blank or "starter" pistols were used, but were proven to be dangerous and unreliable. Not only could the flame from the barrel cause burns, but "firey spitballs" from the wadding shot out of the barrel, and they often misfired on air, too. One story goes that a blank pistol jammed during a live broadcast. The stunned sound effects artist was rescued by a quick thinking actor who ad-libbed "Shooting's too good for you. I'll stab you instead, like this!"

(Of course at this point the gun suddenly went off...)

The Foster Gun could also be triggered by sound. If the sound was intended to be heard off camera the SFX artist triggered it manually. If a gun was intended to be seen on camera, the sound of the blank pistol (with less powerful charges) could trigger it. Unfortunately, it couldn't discriminate between a blank pistol or any other sounds of the same volume, so the Foster Gun was eventually retired.

Raymond Scott licensed a lot of his compositions to Warner Brothers, where they were re-orchestrated by Carl Stalling and used in their "Merry Melodies," "Looney Tunes," and other cartoons. Not only was Scott a prolific composer, but an inventor as well. In 1948 he invented a $10,000.00 device called Karloff, which could imitate chest coughs and wheezes, kitchen clatter, the sizzle of a steak frying and jungle drums. Another was the Circle Machine that was used in an Autolite commercial to simulate the sound of a storage battery dying, ending in a short circuit. It was reported that it sounded like "a 23 second breakdown of a Lilliputian merry-go-round followed by a soft string of firecrackers." The Circle Machine consisted of a motor-driven disc with a series of switches positioned around it, each of which triggered a specific sound effect.

Another device was an analog waveform generator. It had a ring of incandescent lamps, each with its own rheostat, and a photoelectric cell on a spindle that whirled in a circle above the lights. As you turned the rheostats down to dim or brighten the individual lamps, the output of the spinning photoelectric cell would change, altering the shape of the generated waveform.

Other devices he invented were the Serial Doorbell ('66), a synthesized Chinese gong ('68), and the Clavivox, which was a keyboard device designed to simulate a Theremin (it was actually made of two Theremins). It could accurately do portamento slides smoothly from one pitch to another on the keyboard without a break in the sound.

[Clavivox]

In the 1900s, early electronic musical instruments were being built which also had sound FX capabilities. In 1906 Thaddeus Cahill debuted his Telharmonium, which was an early type of tone organ. The early Telharmonium concerts included several classical pieces, imitations of instruments, and sound effects, such as the passing of a drum and fife corps.

Sound FX have also been used as complete musical compositions. A group of Italian composers called Futurists wrote music for machine guns, steam whistles, sirens, and other noisemakers as early as 1912. The Futurists were among the first composers to include noise as an inherent part of the music, not merely as a side effect with traditional instruments.

Today Sound FX are a big business and more important than ever. Motion pictures often devote several tracks for Sound FX and Foley work. The term "Foley" was coined in the 50s from the work of sound editor Jack Foley. Jack didn't actually invent the techniques (as they were commonly used), but was credited with them because some directors had seen Jack adding live sound effects to a film (as opposed to laying them down in post production later). They liked it and wanted to implement sound FX "the way Jack Foley did it," and that's how the term caught on. According to Mott, if you were making a film in Mexico and needed an effect synced live, the term was known as "Garcia-ing", after the name of the post production sound person who was known for that technique there. In Italy, it would be whatever the last name of that person was, etc. It just so happened that the term "Foleying" caught on and is universally accepted now.

Sound FX have been greatly enhanced by the advent of electronic sound-processing equipment, with the evolution of sound effects devices from early electronic musical instruments through to present-day digital effects. Some of these sound-processing effects include reverb, the Echoplex (an analog tape

delay), flangers, fuzz boxes, phase shifters, and octave dividers (I think Jimi Hendrix was credited for spawning some of these), not to mention synthesizers, etc. Today it's happening more in the digital domain with computers and powerful sound processing software, such as Arboretum's Hyperprism, HyperEngine, and Harmony for the Macintosh (named after an Edgar Varese composition), which has many different reality-bending FX. This amazing software processes sound in real time by drawing a "path" in a "Process Window." Depending on which point you are in a window, it yields a different dimension and/or intensity of the effect.

There are also many other dedicated FX processing devices and musical instruments. These include such as Kurzweil's K2500 series sampler workstations, which give you the ability to record, edit and process sounds and save them as a completely new waveform, and Emu's Vortex, which can "morph" from one effect to another.

A whole industry has sprung up around sound FX. Today there are several manufacturers offering libraries on Audio CD, CD ROM for Macs & PCs, and other dedicated (and inter-changeable) file formats, containing thousands and thousands of sounds from normal everyday FX to alien atmospheres. And there doesn't seem to be any end in sight.

They say "a picture is worth a thousand words" ... but the people who first married sound effects with actions to help tell a story created an art form and spawned an entire industry the end of which we will probably never see, thanks to creative sound FX designers, inventors, editors, Foley artists, software developers, instrument manufacturers and many others-in the industry in search of that perfect sound.

Back cover from a 1910 catalog showing George Way & George Stone with various animal sound fx.

The Purpose of Sound effects

The function of sound effects is three fold...[1]

- Simulating reality
- Creating illusion
- Mood

Simulating Reality

In a western barroom fight our hero is hit over the head with a whiskey bottle.
The bottle is fake. It becomes real with the addition of an actual glass bottle crash from the sound editors library. In gun battles the weapon actually is actually loaded with blanks and what is called quarter loads which means one-fourth of normal amount of gunpowder contained in a real bullet. The actual sound is just slightly louder than a cap pistol until the sound editor has completed work.

These are but two of the more obvious examples of the sound effect taking a fake bit of theatrics and making it real by adding a real sound.
You see it - you hear it - you must believe it!

Creating illusion

Creating illusion was one of the biggest additions to the art of film by sound.

A man and a woman walk into a café. Several other people are sitting at various table in deep conversation. The main couple sits at a table and a series of close ups for their conversation are presented.

By adding the sound of the off-scene diners the audience is convinced that they are still in the café. Obviously, the producer does not want to pay a group of extras to sit off camera. The sound editor places them there with his crowd walla for the sound (Walla is an industry term for the sound of people talking without hearing specific words)

A woman is sitting in her living room. The door opens and her husband walks into the room.

With the addition of a few sound effects, it is possible to inform the audience that he has driven up to the house, parked his car, walked to the door, and used his key to unlock the door. None of this was shot. It was an illusion created with effects.

A safari makes it through the jungle.
The sound editor cuts a lion roar. Not has he placed a lion in the film where none exists but he has also placed the safari in danger.

Mood

A cowboy sits around a small campfire. The mood of a campfire is warm.

- Add an off-scene owl and it becomes lonesome.
- Add a wolf howling in the distance and it perhaps harken danger.
- Cut a gunshot and you are telling the audience that another human is nearby. Is he friend or foe?

Anticipation, possible danger, fear, joy - all are being evoked. Which should you be feeling? Move forward in your seat because surely the film will soon tell.

A pair of lovers is caught in the midst of an argument. Suddenly, the woman turns, walks to the phone, and lifts the receiver.

- You can cut the first phone ring just ahead of the reaction and there is nothing unusual.
- Cut two or three rings before she reacts and you are telling the audience that she was so involved in the argument she did not even react to the phone until the third ring. That is revealing about her mood and depth of involvement.

A leading lady awakens in the morning snuggled in her bed.

- The sound of a distant train whistle make is a lonesome scene.
- Replace the train whistle with the sound of kids playing outside, and the audience perceives an entirely different emotion

The leading man drives up to a house.

- As he parks, we hear the sound of a small dog yapping. No particular danger is perceived. Inside is probably a child or an old lady.
- Change the small dog yapping to the sound of a vicious Doberman, and the mood is again changed.

The sound effect is a major part of the magic of Hollywood. The plywood wall becomes real when the sound of a big wood crash is added. The avalanche of large boulders (actually foam rubber or styrofoam is) real and menacing because of the sound effect.

The sound effect is not only a mechanical tool (e. g., fire a gun, cut a shot), it can also be employed creatively and artistically.

In the Museum of Television & Radio in Los Angeles, they use a lot of manual sound effects, partly because they don't have an electronic "sampler" keyboard to trigger sound effect recordings from and partly to make for a fun workshop that preserves the golden age of radio. However, even in the old days, manual effects were not the only ones. Some sounds were cheaper and better as records, especially cars, planes, and weather. Here's a list of the sound effects we use. In most cases they are easy to make from common materials. A lot of the sound is how you manipulate objects. You'll need to experiment to get just the right technique to produce the right sound. Also, test to see what your effects sound like over a microphone, something that sounds fine to your ears will come off as weak over a mike. Useful tip: We use a single omni-directional microphone for sound effects—it picks up everything and adds a nice ambience to the sound, making it appear more real than if you held the sound effect up to a regular uni-directional mike. It makes life much simpler.

Sound Effects in Radio Drama

To Sound Effect or Not to Sound Effect?

Please keep in mind that radio drama is not merely a play with "lots of sound effects". A good radio play is good drama, full of conflict and even action, evoked through dialogue, music and sound effects, in that order. Sound effects are an important ingredient, but by far, the least important. In radio, the dialogue and narration contribute roughly 75% to the drama, with music another 15% and sound effects a paltry 10%. Sound effects merely sketch in the action or punctuate dialogue and don't generally provide background in the way sets or locations do in plays and films. Sound effects suggest action, but they can easily be confusing or misinterpreted if relied upon too heavily. Intrusive sound effects will make the dialogue harder to follow. Just the same, sound effects can turn a mere script into a real-time drama as opposed to just a story being told. Sound effects add realism, but should be used sparingly. In the production of radio drama, if a sound effect cue is missed, few listeners will notice. Use that as a guide and employ a minimum of sound effects. Only significant sounds are necessary. Sketch the scene, don't perfect it. This isn't the same as the "Foley" sounds used in film. They deal primarily with human sounds (footsteps, keys, brushing lapels, fist fights, etc.) to re-create the real world in detail. Radio sound effects artists do both the Foley type sounds as well as car crashes, gunshots, bubbling vats, thunder, machines, battles, even the destruction of a planet.

Suggestions for a Good Sound Effects Kit

These are just examples for how to have your own kit to produce different sounds. You could try out some of these or just work on some of your own ideas. You will have to try out different things sometimes to get the sound that you want. It can sometimes be just an amazing experience.
CRASH BOX -made from a metal Christmas popcorn container. This is could be a useful device.

OLD DIAL TELEPHONE - and ringer bell box -the kind companies use for delivery doorbells. Don't bother trying to make the old phone ring-it'll cost hundreds for a step down transformer to deliver the 84 volts necessary to engage the phone ring circuit. Just build a ringer box.

THUNDER SHEET - 2x4 foot 16th inch high-impact polystyrene.

Look in your local yellow pages for "Plastics" and call around.
WALKBOARD - 2x3 foot doubled 3/4 inch plywood for footsteps. Put tile on one side to get a different sound.

SMALL DOOR - Ideally, cut a regular door just after the latch and make a door frame to fit.

GRAVEL BOX - Wooden drawer-type box with gravel-for horses/dirt. Also, coconuts and short 2x4s as "boots"-with spurs!

CAVEMAN CLUBS -Large hollow caveman style toy clubs-great for fights and bodies dropping to the floor.

STIFF PLASTIC BAGS - For fire, static, even marching feet.

CLIP BOARDS - For gunshots-but they need a resonator box under them to sound "bigger". I think a metal trash can might be the ticket.

TOY RATCHET - Plastic New Year's noisemaker. Good for handcuffs, winches, drawbridges... ratchets.

VIBRO PEN - A "Dizzy Doodler" pen that writes wobbly-for planes/jackhammers. Turn it on and apply the top (not the pen) to a cardboard box for a convincing airplane. Do the same on a stack of saucers and you'll have a great jack-hammer.

TOY CELL-PHONE - For radars, space beeps, even cell-phones! Lately, these have been really hard to find.

SLIDE WHISTLE - Besides eeeeeYOOOP, it can also be quickly slid back and forth for radars and space sounds.

PLASTIC EGG MARACAS - for jungles, rattlesnakes, weirdness. You could make some with plastic easter eggs and rice > you might get them in some stores with finer gravel that sounds very good.

WIND MACHINE - also useful for other mechanical sounds-machine guns, cars, rotors.

TEACUP SAUCERS - Good "dishes" sound. Apply vibro-pen to two stacked saucers for a great jack-hammer.

METAL SPOONS/SPATULAS-Get a really big pancake flipping spatula and some large metal cooking spoons for great sword fights.

PLASTIC TUMBLER - For pouring water. Drop AA batteries in empty tumblers for ice cubes.

CIRCUIT BREAKER - A home circuit breaker (the kind that go in electrical panels) gives you that old fashioned, LOUD click of a light switch.

-ADVANCED SFX-

GLASS SCRATCH -A box with a 6x6 inch plate of 1/4 inch thick glass. It has a spring loaded X-brace with spikes in it. You rotate a crank and it scratches the glass. Sounds like a train brake or fingernails on a chalkboard.

DOORBELLS/BUZZER/BELL - A wooden box with the typical ding-dong (and single "ding"-for hospital pages) and an office door latch buzzer, and a ringing bell to simulate an old style telephone-4 seconds, two seconds off.

CREAKER BOX - Cotton clothesline wrapped around a dowell rubbed with violin rosin—sounds like a creaking door or thumbscrews. It's rigged in a wooden shoebox sized thing.

CREAKER DOWEL -A 1/2 inch dowel wrapped by a rubber hose with a hose clamp. It's a different kind of creak.

BOING STICK - A box with a plywood bow that has a banjo string on it. You pluck the string and bend the bow for the famous "Road Runner" cartoon "beeeeyoooooooo" boing sound.

SCREEN DOOR SLAMMER - a hinged door with a long screen door spring.

COMPILED SOUND EFFECT "HOW TO":

BIRDS CHIRPING:

Twist a little bird call. (Not too often, though). Available at camping outfitters.

BLACKJACK:

Plastic club hits once on wooden walk board. (Large "Flintstone" toy clubs)

BLOWGUN:

Blowing through a cardboard tube, then slapping the tube.

BODY DROPS TO GROUND:

Plastic clubs beating on cardboard box.

BODY DROPS:

Two plastic clubs fall on cardboard box.

BREAKING THINGS. (FURNITURE, BOXES):

Manipulate a crash box. (A giant potato chip/pretzel can filled with glass, wood, metal and stones)

BUBBLING VAT:

Blow bubbles into large drinking cup. Close to mic.

CABIN DOOR OPENING AND CLOSING:

WINDOW SHUTTERS CRASHING:

CAR CRASH:

Crash box.

LAMP CRASHES:

FURNITURE BREAKING:

Shake the crash box.

CAR DRIVES OFF:

CAR DRIVING:

Electric car driving toy.

CHAINS RATTLING:

Drop chains in wooden wagon. (a wooden box using old roller skate wheels)

COOKIE TIMER RINGS:

Triangle and striker. (one time) "Ding!"

COOKIE TRAY DROPPED:

Drop metal tray with coasters on walk-board.

COWBOY BOOTS WALKING ON DIRT:

FOOTSTEPS RUNNING ON A DIRT FLOOR:

FOOTSTEPS ON UNDERBRUSH:

MEN DISMOUNTING FROM HORSES. (JANGLING SPURS, FOOTFALLS):

Use wooden "feet" in the hoof box. (A wooden box 14 x 18 x 4 filled with gravel.)

CRAWLING IN UNDERBRUSH:

Crinkle large plastic bag.

CUTTING NOISE. ("A KNIFE CUTTING A SILVER BULLET"):

Scrape butter knife against some metal.

DOG BARKS:

Vocalize it. "Rarf Rarf Rarf!"

DOOR KNOB JIGGLED AND DOOR OPENS:

DOOR SLAMS:

DOOR OPENS:

Small 18" x 15" wooden door on hinges in a 2x4 frame with a door knob.

DOORBELL CHIME:

Doorbell SFX device. (a wooden box with a doorbell and buzzer installed)

FACE BEING SLAPPED:

Slapping hands together

FACE SLAP:

Slap stick. (two 15" x 3" boards connected by a spring loaded hinge)

FIGHT BELL RINGS:

Large bell SFX device. (An old school bell-about 15 inch diameter)

FIGHT:

BODY FALLS TO THE GROUND:

MEN STRUGGLING DURING A FIGHT:

PUNCH:

Whack two plastic clubs together or on a leg. Grunt.

FIRE BELLS:

Triangle and striker.

FIRE:

Repeatedly crinkle a stiff plastic bag.

FOOTSTEPS on UNDERBRUSH:

Twist straw (hay) or a broom's brush..

FOOTSTEPS:

Shoes on wooden walk-board. (walk-board is 2'x3' doubled 3/4 plywood)

GIFT WRAPPING PAPER CRINKLING:

Crinkle a stiff plastic bag.

GLASS DROPS:

Knock drinking glass on table. (So it does NOT break!)

GLASSES CLINKING:

Drinking glasses clinking.

GUNSHOTS:

Clip board snaps. The pros used guns firing blanks and later electronic sound devices. We've found the clip boards to be safe and reliable. Cap guns don't sound good over a microphone.

HAND CLAPS:

Clap hands three times.

HAND PICKS UP PHONE:

HANG UP PHONE:

Rattle phone receiver (handset) on phone.

HANDCUFFS LOCKING:

Handcuffs locking, first one then the other.

HORSES GALLOPING, TROTTING, WALKING:

Use coconut shells in gravel hoof box.

ICE CUBES IN A GLASS:

Drop Christmas tree light bulbs into a glass. (Carefully!)

INDIAN TOM TOM. (ONE-2-3-4, ONE-2-3-4 ETC.):

Use bongo drum with hand (or toy drum with beater).

INTERMISSION CHIME:

Doorbell SFX device. (one tone not the ding-dong of front doors)

JACKHAMMER:

Apply vibrating toy pen to 3 tiny plates on table.

MEN YELLING:

"Hey!" "Look out!" "Here they come!" "Get 'em!"

NAIL PULLER:

Pull lid on nail puller SFX device. (a 2"x12" wooden box with a flat wooden lid that is nailed on one end.

NIGHTCLUB WALLA WALLA:

Chatty mumbling, some laughter.

OFFICE BUZZER:

Electric buzzer SFX device. (like a door bell device but it produces a "buzzzzzz" sound)

Open and close the small door.

OPEN/CLOSE DOOR:

CAR DOOR SLAMS:

Wooden door. [and old metal ice chest would be better]

OPEN/CLOSE WINDOW:

Roll wooden cart or old fashioned roller skate back and forth on table.

OVEN DOOR OPENS:

Roll wooden wagon across table.

PHONE RINGS: (old fashioned Bell telephone style)

Phone bell SFX device. (a typical modern bell-similar to a door bell device. Ring it 4 seconds on 2 seconds off)

POURING WATER:

Pour water from pitcher into a glass. Close to mic.

PRIZEFIGHT WALLA WALLA:

Boisterous voices. "Huh", "Brother", excited mumbling.

RAY GUN BLAST :

Toy light saber.

SAWING EARTH:

Vibro-pen on cardboard box.

SCREECHING AUTO BRAKES:

"Glass scratch" SFX device. A 7"x7" wooden box containg a thick piece of glass that is scratched by several large nails operated by a crank. The sound is similar to fingernails on a chalkboard, but less unpleasant.

SIREN:

Blow on siren whistle.

SLIDE WHISTLE DOWN:

Pull plunger on slide whistle. "Eee-yoooooop!"

SMALL CRASH:

Shake the crash box SFX device a little bit.

SWORD UNSHEATHED:

Scrap a metal cooking spoon against a large pancake flipper.

TAPPING ON WINDOW:

Tapping on glass sheet.

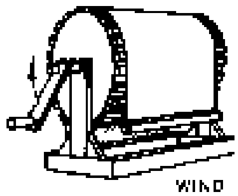
TELEPHONE RECEIVER PICKED UP:

TELEPHONE HANGS UP:

Rattle phone handset on cradle.

TELEPHONE RINGS:

Phone bell ringer SFX device. (Ring it 4 seconds on and 2 second off)



THUNDER:

Shake a 3'x4' 1/8 inch thick plexiglassc sheet. Use the crash box first for the thunder crack.

TRAIN BRAKES:

Glass scratch SFX device. A 7"x7" wooden box containg a thick piece of glass that is scratched by several large nails operated by a crank. The sound is similar to fingernails on a chalkboard, but less unpleasant.

TRAIN CHUGGING:

TRAIN SLOWING:

Drummer's brush on an old fashioned wash-board. A metal pie tin with holes punched it could substitute for the wash-board.

TRAIN CRASH:

Crash box SFX device. Shaken a lot.

TRAIN STATION WALLA WALLA:

Quiet voices. "Where", "OK", mumbling.

TRAIN WHISTLE:

Wooden train whistle. "Whoo-whoo!"

YOO-HOO WHISTLE:

Human whistles.

How to Build Sound Effect Devices:

Lets see how some of the devices could be built for sound effects. You will observe it's not the props themselves, but the way they are manipulated that make the difference. It's all in how you use the sound effects devices. Here's some of the simple SFX devices that can be used quite often.

Crash Box:

The crash box is one of the most useful SFX devices that can be used. You can use it for car crashes, planets being destroyed, ghostly clunking about and also as a contributing background noise under medieval wars and gun battles. It's also a fine first part for doing a thunder-crack (followed by the rumble of a "thunder sheet" being flexed-see below).

Some of the old time radio shows had crash boxes that resembled a small metal trash can on a crank. You could use a popcorn can, (11 inches high and 10 inches in diameter-a bit larger than a basketball) and fill it with junk.

Fill the can with broken ceramic coffee mugs, a crushed aluminum can, pennies, nails, pieces of wood (about the size of a fist), and two handfuls of gravel. Tape the lid shut with grey duct tape. You want to keep the lid on tight or the junk or its soon-to-be fine dust will leak out. Glasses or wine bottles powderize too much. The ceramic coffee mugs are sturdier and sound similar.

As in most SFX work, manipulation is everything. Use a two handed shake and roll motion to get a variety of crashes out of it. When shaking it for a sound effects cue, you have to remember to end the motion with the can upright or you'll create unwanted crashing/settling as you put the can down.

After a while of use, the mugs and rocks grind down and the crash may not be as loud, so you may have to put in another coffee mug. At some point the debris will turn to such fine dust that it begins to leak out the seams. Dump everything out and start over-or get another popcorn can and start from scratch. You may have to tape up the seam, but don't cover the whole can with duct tape or you'll deaden the crash too much.

Thunder Sheet:

Convincing thunder and other low rumbles as well as odd space sounds can be wrung from a 2 x 4 foot sheet of high impact styrene plastic-with a thickness of about 60 mil. You can manipulate it in various ways to get different sounds. To get thunder, grab it with two hands from the 2 foot end and move your hands in a punch-after-punch motion (like a boxer working a speed bag at a gym.)-you ripple it. To get a really convincing thunder-crack, have a second person quickly jerk a crash box and then follow it up immediately with the thunder sheet. You can get some outerspace "wup-wup, wup wup" sounds by grabbing each 2 foot end with a hand and flexing it in broad slow strokes.

Wind Machine

The old time radio shows used wind machines for Superman flying, storms and spooky backgrounds. The sound is produced by rotating a drum of wooden slats against a canvas sheet that is pulled lightly against the slats. It's not too tough to make your own, but will require some carpentry skills. You can make one out of plywood and a 1 1/2 inch closet pole dowel.

The drum could be arround 12 inches in diameter and 16 inches long. For the ends of the drum, you could use two 12-inch circles of 3/4 inch particle board. Drill two 1-and-7/16 holes in the center of the circles and file it so allow a tight fit for the closet pole-which serves as the axle. Then cut 18 slats - 1-inch wide by 16 inches long, from a piece of 1/4 inch plywood.

NOTE: The slats must be of a fairly hard wood or they won't be loud enough when rubbing against the canvas sheet. You could use tiny nails to attach the slats to the circles leaving about an inch of space between them. They don't have to be perfectly spaced-just nail one then it's polar opposite and continue by halves, quarters, eighths, etc., until the drum is covered with slats.

Build the drum platform out of a 20 inch by 16 inch rectangle of 3/4 plywood and use two triangles to serve as braces for the drum. The dimensions depend upon how much axle you use. These are just rough measurements you could experiment and come up with your own.

For the axle, use a 1 1/2 inch closet pole and cut it to about 19 inches. Use one of those plastic end-caps for hanging closet pole to hold the axle on one end of the dowel and just drilled a 1-and-9/16 hole through the other brace. The drum is attached to the axle just where it goes through the circles. The crank can be just a short piece of 2 x 5 inch plywood with a bit more dowel for a handle. Attach them with several screws.

You can use a cut up “butterfly chair” seat for the canvas sheet. It is wrapped around an extra slat at either end (like an old roll-up window shade) and secured against the platform on one end only.

The cranking motion really makes the device slip around, so cut out a piece of a thin rubber-backed welcome mat and attached it to the bottom of the platform-carpet side facing the bottom of the platform-rubber side facing whatever tabletop surface you put the device on.

To get the wind sound, you crank and, optionally, pull the canvas tighter against the rotating drum. You don’t have to crank too fast to get a convincing wind storm. You can also leave the canvas away from the slats and apply other things (playing cards, a drummer’s wire brush, etc.) against the rotating slats to get other mechanized sounds.

Such a wind machine isn’t too loud (maybe harder slats would increase the volume), but it really produces that classic dust storm or Superman flying sound. It’s an amazing little contraption.

Walk Board

The walk board is used for running, walking, dancing, and even dragging ghostly chains on. Use a piece of 2 foot by 3 foot plywood, doubled up (two 3/4 inch pieces attached on top of one another). You may want to cover one side with tile or carpet. We have people walk and stomp on the board, but noticed that many people wear athletic shoes, which don’t make much noise. You could get some leather-soled shoes and walk them on the board with your hands, or you could just have people stomp louder. If you prop up one end of the walk board with a two by four, you can simulate a stair step sound.

Gravel Box

The gravel box is generally used for horse hoofs and walking. You could make yours with an 18 inch by 30 inch wooden box (from 1 x 6’s and plywood) filled with a layer of garden gravel. Could use two coconut shells for horses and two two-by-four blocks (7 inches by 4 inches) as cowboy “boots” to walk on the gravel. Some old shows used a canvas bag filled with gravel, but we’ve found the box sufficient-also having the gravel exposed allows us to manipulate it for other sounds-such as Crusaders wading through an ocean of bones. Also a small plywood “deck” covering part of the box could be used for streets or bar room floors. A piece of ceramic tile might help for cobblestones.

Roller Skate Box

Take an old fashioned kid’s rollerskate and attach it to the bottom of a wood box (10 x 4 x 4). The old skate should be the kind that would strap onto a kid’s shoe and use four metal wheels-you want something noisy, not fancy. This can serve as a horse-drawn buggy, an elevator door opening, a double-sash window opening and even the creaking of a sailing ship at sea. You can put chains or gravel in the box to jostle around too.

Egg Maracas

You can buy these plastic egg maracas at musical instrument stores or make your own out of egg sized plastic easter eggs filled with seeds or rice. Get two and shake them very fast, then vocalize some jungle bird sounds and you’ve got an instant rainforest. This is very evocative.

Sword Fight

Bang together several large metal cooking spoons and pancake flippers. The metal spoons can be plain or have strainer holes. The crash box and some battle cries add extra mayhem too.

Five Gallon Bucket

Get one of those plastic 5 gallon paint buckets fill it one third full and swish water around inside. This can serve as rowboat oars or swimming or sharks. A plumber’s plunger can work as a swish stick, but what’s better is an “X” shaped cross of 1/4" plywood on the end of a stick. To get the sound of a splash you don’t plunge into the bucket-that’s splash water all over you-instead you put the X-stick into the bucket and pull it out. That’s how they did on the Lone Ranger. Plus you can use the bucket to carry around other small SFX gear.

Gun Shot Clip Boards

The typical office clipboards can be snapped to make decent gunshot noises. I find the wooden backed ones to be noisier. I think they really need a resonating box to amplify the sound. Maybe snapping the clipboard in a small metal trash would work. Another idea is to use a drumstick to hit a throw pillow or vinyl covered drummer’s throne-however this requires some skill to slam the body of the stick onto the pillow. If you’re not precise with your hits, you might end up shooting “blanks” when real bullets are called for.

Notes :

NOISE REDUCTION

Noise reduction techniques have been applied to analogue tape machines of all formats, radio microphones, radio transmission and reception, land lines, satellite relays, gramophone records, and even some digital tape machines. The general principles of operation will be outlined, followed by a discussion of particular well-known examples. Detailed descriptions of some individual systems are referred to in the Further reading list at the end of this chapter.

Why is Noise Reduction Required?

A noise reduction system, used correctly, reduces the level of unwanted signals introduced in a recording-replay or transmission-reception process (see Figure). Noise such as hiss, hum, and interference may be introduced, as well as, say, print-through in analogue recording, due to imperfections in the storage or transmission process. In communications, a signal sent along a land line may be prone to interference from various sources, and will therefore emerge with some of this interference signal mixed with it. A signal recorded on a cassette machine replays with high-frequency hiss. Unwanted noise already present in a signal before recording or transmitting, though, is very difficult to remove without also removing a part of the wanted signal one could roll off the treble of a cassette recording to reduce the hiss but one would also lose the high frequency information from the sound, causing it to sound muffled and 'woolly'

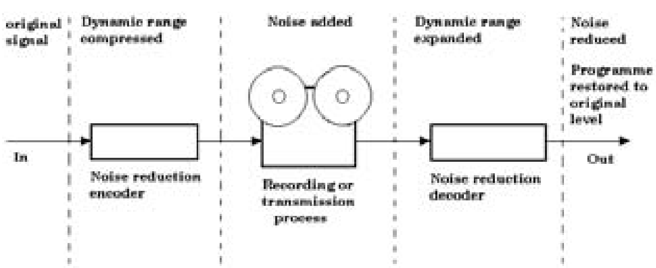


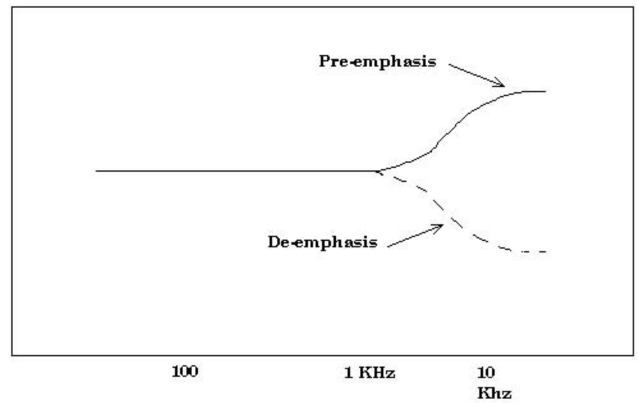
Figure:- Graphical representation of a companding noise reduction process

Methods of Reducing Noise

Variable Pre-Emphasis

Pre-emphasis is a very straightforward solution to the problem of noise reduction, but is not a panacea.

One approach to the problem of reducing the apparent level of noise could be to pre-condition the incoming signal in some way so as to raise it further above the noise. Hiss is most annoying at high frequencies, so one could boost HF on recording. On replay, signals would therefore be reproduced with unnatural emphasis, but if the same region is now attenuated to bring the signal down to its original level any hiss in the same band will also be attenuated by a corresponding amount, and so a degree of noise reduction can achieved without affecting the overall frequency balance of the signal. This is known as pre-emphasis (on record) and de-emphasis (on replay), as shown in the diagram.



Many sound sources, including music, have a falling energy content at high frequencies; so lower level HF signals can be boosted to an extent without too much of risk of saturating the tape. But tape tends to saturate more easily at HF than at LF so high levels of distortion and compression would result if too much pre-emphasis were applied at the recording stage. What is needed is a circuit, which senses the level of a signal on continuous basis, controlling the degree of pre-emphasis so as to be non-existent at high signal levels but considerable at low signal levels this can be achieved by incorporating a filter into a side-chain which passes only high frequency, low-level signals, adding this component into the un-pre-emphasised signal. On replay, a reciprocal de-emphasis circuit could then be used. The lack of noise reduction at high signal levels does not matter, since high-level signals have a masking effect on low-level noise.

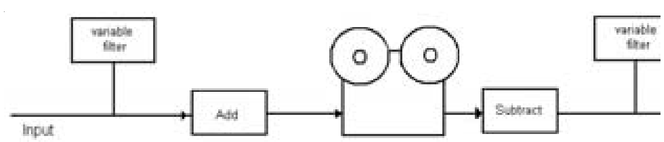
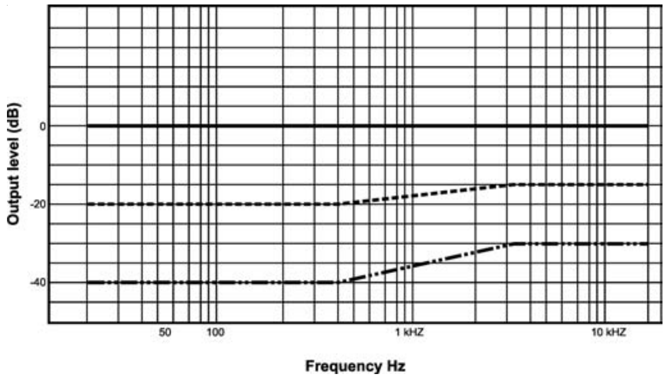


Figure: A simple complementary noise reduction system could boost high frequencies at low signals during encoding, and cut them on decoding **(One Figure on next page)**

Such a process may be called a compansion process, in other words a process that compresses the dynamic range of a signal during recording and expands it on replay. The variable HF emphasis described above is an example of selective com-pansion, acting only on a certain band of frequencies. It is most important to notice that the decoding stage is an exact



mirror image of the encoding process, and that it is not possible to use one without the other. Recordings not encoded by a noise reduction system cannot simply be passed through a decoder to reduce their noise. Similarly, encoded tapes sound unusual unless properly decoded, normally sound-ing over-bright and with fluctuations in HF level.

Dolby B

The above process is used as the basis for the Dolby B noise reduction system, found in most cassette decks. Specifically, the threshold below which noise reduc-tion comes into play is around 20 dB below a standard magnetic reference level known as 'Dolby level' (200 n Wb m^{-2}). The maximum HF boost of the Dolby B system is 10 db above 8 khz and therefore a maximum of 10 db of noise reduction is provided. A high quality cassette deck, without noise reduction, using a good ferric tape will yield a signal to noise ratio of about 50 dB ref. Dolby level, when Dolby B noise is switched in, 10 dB im-provement brings this up to 60 dB (which is more adequate foe good-quality music and speech recording). The quoted im-provement is seen when noise is measured according to the CCIR 468-2 weighted curve and will not be so great when measured unweighted.

Dolby B incorporates a sliding band over which pre-emphasis is applied, such that the frequency above which compansion occurs varies according to the nature of the signal. It may slide as low as 400 Hz. This aims to ensure that maximum masking of low-level noise always occurs, and that high-level signals at low fre-quencies do not result in 'noise pumping' (a phenom-enon which arises when a high level signal in one band causes less overall noise reduction, causing the noise in another band to rise temporarily, often not masked by the high-level signal due to the difference in frequency of the signal and the noise).

The Dolby process, being level dependent, requires that the reproduced signal level on decoding is exactly the same with respect to Dolby level as on encoding. This means that a particular cassette machine must be set up internally so that Dolby-encoded tapes recorded on it or other machines will replay into the decoder -at the correct electrical level for proper decoding. This is independent of the actual output level of the machine itself, which varies from model to model. If the replay level, for instance, is too low, the decoder applies too much treble cut because -20 dB threshold level will have moved downwards, causing recorded signal levels above this to be de-emphasised also. Frequency response error will therefore be the result. Similarly, if the frequency response of a cassette machine shows significant errors at HF, these will be exaggerated by the Dolby record/replay process.

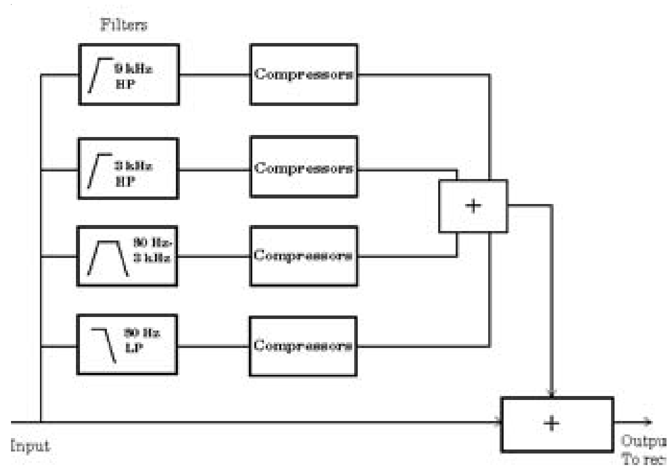
A so-called MPX (multiplex) filter is mandatory with Dolby B systems, and removes the 19 kHz pilot tone present in FM stereo radio broadcasts. This is needed because the pilot tone may still be present in the output of an FM tuner, artificially affecting the encoded level of HF signals on a recording from the radio. Since the frequency response of many cassette machines does not extend to 20 kHz the tone would not be reproduced on replay, and thus the decoder would not track the encoded signal correctly, leading to noise pumping and response errors. On some recorders the filter is switchable. On cheaper machines the filter simply rolls off everything above 15 kHz, but on better machines it is a notch at 19 kHz.

Dolby C

Dolby B became widely incorporated into cassette players in the early 1970s, but by the end of the 1970s competition from other companies offering greater levels of noise reduction prompted Dolby to introduce Dolby C, which gives 20 dB of noise reduction. The system acts down to a lower frequency than Dolby B (100 Hz), and incorporates additional circuitry (known as 'anti-saturation'), which reduces HF tape squashing when high levels of signal are present. Most of the noise reduction takes place between 1 kHz and 10 kHz, and less action is taken on frequen-cies above 10 kHz (where noise is less noticeable) in order to desensitise the system to HF response errors from such factors as azimuth misalignment, which would otherwise be exaggerated (this is known as 'spectral skewing'). Dolby C, with its greater compression/expansion ratio compared with Dolby B, will exaggerate tape machine response errors to a correspondingly greater degree, and undecoded Dolby tapes will sound extremely bright.

Dolby A

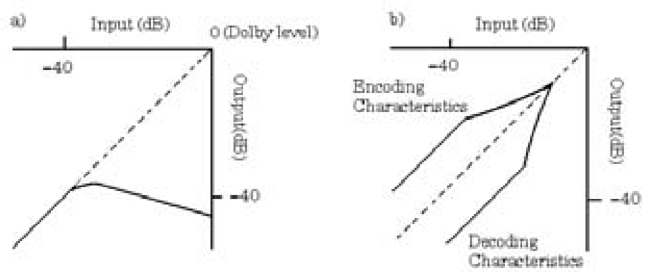
Dolby A was introduced in 1965, and is a professional noise reduction system. In essence there is a similarity to the processes described above, but in the Dolby A encoder the noise reduc-



tion process is divided into four separate frequency bands, as shown in Figure.

Figure In the Dolby A system a low-level 'differential' signal is added to the main signal during encoding. This differential signal is produced in a side-chain which operates independently on four frequency bands. The differential signal is later sub-tracted during decoding

A low-level 'differential' component is produced for each band, and the differential side-chain output is then recombined with the main signal. The 'differential' component's contribution to the total signal depends on the input level, having maximum



effect below -40 dB ref. Dolby level. as shown in the figure a} and b} below.

Figure a) Differential signal component produced in a Dolby A side-chain. (b) Input level plotted against output level of Dolby A unit after adding or subtracting differential component

The band splitting means that each band acts independently, such that a high -level signal in one band does not cause a lessening of noise reduction effort in another low-level band, thus maintaining maximum effectiveness with a wide range of programme material. The two upper bands are high pass and overlap, offering noise reduction of 10 dB up to around 5 kHz, rising to 15 dB at the upper end of the spectrum.

The decoder is the mirror image of the encoder, except that the differential signal produced by the sidechain is now subtracted from the main signal, restoring the signal to its original state and reducing the noise introduced between encoding and decoding.

Dolby SR

The late 1980s saw the introduction of Dolby SR - Spectral Recording - which gives greater noise reduction of around 25 dB. It has been successful in helping to prolong the useful life of analogue tape machines, both stereo mastering and multi-track, in the face of the corning of digital tape recorders. Dolby SR differs from Dolby A in that whereas the latter leaves the signal alone until it drops below a certain threshold, the former seeks to maintain full noise reduction (i.e.: maximum signal boost during recording) across the whole frequency spectrum until the incom-ing signal rises above the threshold level. The band of frequencies where this hap-pens is then subject to appropriately less boost. This is rather like looking at the same process from opposite directions, but the SR system attempts to place a comparably high recording level on the tape across the whole frequency spectrum in order that the dynamic range of the tape is always used optimally.

This is achieved by ten fixed and sliding-band filters with gentle slopes. The fixed-band filters can vary in gain. The sliding-band filters can be adjusted to cover different frequency ranges. It is therefore a fairly complex multiband system, requir-ing analysis of the incoming signal to determine its energy at various frequencies. Spectral skewing and anti-saturation are also incorporated

Dolby SR is a particularly inaudible noise reduction system, more tolerant of level mismatches and replay speed changes than previous systems. A simplified 'S'-type version has been introduced for the cassette medium, and is also used Oh some semi-professional multitrack recorders.

dbx

dbx is another commonly encountered system. It offers around 30 dB of noise reduction and differs from the various Dolby systems as follows. dbx globally compresses the incoming signal across the whole of the frequency spectrum, and in addition gives pre-emphasis at high frequencies (treble boost). It is not level dependent, and seeks to compress an incoming signal with, say, a 90 dB dynamic range into one with a 60 dB dynamic range which will now fit into the dynamic .:range capabilities of the analogue tape recorder. On replay, a reciprocal amount of expansion is applied together with treble de-emphasis.

Owing to the two factors of high compansion ratios and treble pre- and de -emphasis, frequency response errors can be considerably exaggerated. Therefore, dbx type 1 is offered which may be used with professional equipment and type 2 is to be used with domestic equipment such as cassette decks where the noise reduc-tion at high frequencies is relaxed somewhat so as not to exaggerate response errors unduly. The degree of compression/expansion is fixed, that is it does not depend on the level of the incoming signal. There is also no division of noise reduction between frequency bands. These factors sometimes produce audible modulation of back-ground hiss with critical programme material such as wide dynamic range classical music, and audible 'pumping' noises can sometimes be heard. The system does, however, offer impressive levels of noise reduction, particularly welcome with the cassette medium, and does not require accurate level alignment.

telcom c4

The ANT te1com c4 noise reduction system arrived somewhat later than did Dolby and dbx, in 1978. Capitalising on the experience gained by those two systems, the telecom c4 offers a maximum noise reduction of around 30 dB, is level dependent like Dolby, and also splits the frequency spectrum up into four bands which are then treated separately. The makers claim that the c4 system is less affected by record/ replay-level errors than is Dolby A. The system works well in operation, and side-effects are minimal.

There is another system offered by the company, called 'hi-com', which is a cheaper, simpler version intended for home studio setups and domestic cassette decks.

Line-up of Noise Reduction Systems

In order to ensure unity gain through the system on recording and replay, with correct tracking of a Dolby decoder, it is important to align the noise reduction signal chain. Many methods are recommended, some more rigorous than others, but in a normal studio operation for everyday alignment, the following process should be satisfactory. It should be done after the tape machine has been aligned (this having been done with the NR unit bypassed).

For a Dolby A encoder, a I kHz tone should be generated from the mixer at +4 dBu (usually PPM 5), and fed to the input of the NR unit. The unit should be in 'NR out' mode, and set to 'record'. The input level of the NR unit should normally be adjusted so that this tone reads on the 'NAB' level mark on the meter . The output of the unit should then be adjusted until its electrical level is also +4 dBu. (If the tape machine has meters then the level can be read here, provided that these meters are reliable and the line-up is known.)

It is customary to record a passage of 'Dolby tone' (in the case of DoJby A) or Dolby Noise (in the case of Dolby SR) at the beginning of a Dolby-encoded tape, along with the other line-up tones During record line-up, the Dolby tone is generated by the Dolby unit itself, and consists of a frequency-modu-lated 700 Hz tone at the Dolby's internal line-up reference level, which is easily recognized and distinguished from other line-up tones which may be present on a tape. Once the output level of the record Dolby has been set then the Dolby tone button on the relevant unit should be pressed, and the tone recorded at the start of the tape.

To align the replay Dolby (set to 'NR out', 'replay' mode), the recorded Dolby tone should be replayed and the input level adjusted so that the tone reads at the NAB mark on the internal meter. The output level should then be adjusted for +4 dBu, or so that the mixer's meter reads PPM 5 when switched to monitor the tape machine replay .

For operation, the record and replay units should be switched to 'NR jn'.

Dolby SR uses pink noise instead of Dolby tone, to distinguish tapes recorded with this system, and it is useful because it allows for line-up of the replay Dolby in cases where accurate level metering is not available. Since level misalignment will result in response errors the effects will be audible on a band of pink noise. A facility is provided for automatic switching

between internally generated pink noise and off tape noise, allowing the user to adjust replay level alignment until there appears to be no audible difference between the spectra of the two. In normal circumstances Dolby SR systems should be aligned in a similar way to Dolby A, except that a noise band is recorded on the tape instead of a tone. Most systems use LED meters to indicate the correct level, having four LEDs as shown in Figure

Figure Dolby level is indicated on Dolby units using either a mechanical meter (shown left), or using red and green LEDs (shown right). The meter is normally aligned to the '18.5 NAB' mark or set such that the two green LEDs are on together

Operational Considerations

A word may be said about varispeed . it is not uncommon for the replay speed of a tape to need to be adjusted slightly to alter pitch , or total playing time .in creative work massive amounts of speed will be inaccurate since the frequency bands will not now correspond to those during the recording process , and dolby mistracking will result.

Professional noise reduction systems are available as single-channel units, stereo packages and conveniently grouped multiples of 8, 16 and 24 for multitrack work..they generally fit into standard 19 inch (48 CM) racks. Certain models are designed to fit straight into multitrack recorders so that the complete recorder plus noise reduction combination is conve-niently housed in one unit.

Each noise reduction channel is manually switchable between encode for record and decode for replay, and in addition a special input is usually provided which accepts a remote DC signalling voltage, which will switch the unit into encode. Removal of the DC causes the unit to revert to decode ready for replay. Profes-sional tape machines can usually provide this DC requirement, linking it to record status of each track. Those tracks which are switched to record will now . automatically switch the appropri-ate noise reduction channels to encode ready for recording. The system enables the selection of correct noise reduction status to be left to the recorder itself which is a very convenient feature particularly when a large number of channels are in use.

Single-Ended Noise Reduction

General Systems

Several companies offer so-called 'single-ended' noise reduction systems, and these are intended to 'clean up' an existing noisy recording or signal. They operate by sensing the level of the incoming signal, and as the level falls below a certain thresh-old the circuit begins to roll off the treble progressively, thereby reducing the level hiss. The wanted signal, being low in level, in theory suffers less from this treble reduction than would a high-level signal due to the change in response of the ear level High-level signals are left unprocessed. The system is ill fact rather similar to the Dolby B decoding process, but of course the proper reciprocal Dolby B encoding is absent. The input level controls of such systems must be carefully adjusted so as to bring in the effect of the treble roll-off at the appropriate threshold for the particular signal being processed so that a suitable compromise can be achieved between degree of hiss reduction and degree of treble loss during quieter passages. Such single-ended systems should be judiciously used -they are not intended to be left permanently in circuit - and value judgements must always be made as to whether the processed signal is in fact an improvement over the unprocessed one.
If a single-ended system is to be used on a stereo programme, units which are capable of being electronically 'ganged' must be

employed so that exactly the same degree of treble cut is applied to each channel; otherwise varying frequency balance between channels will cause stereo images to wander.

Noise Gates

The noise gate can be looked upon as another single-ended noise reduction system. It operates as follows. A threshold control is provided which can be adjusted such that the output of the unit is muted (the gate is 'closed') when the signal level falls below the threshold. During periods when signal level is very low (possibly consist-ing of tape or guitar amplifier noise only) or absent the unit shuts down. A very fast attack time is employed so that the sudden appearance of signal opens up the output without audible clipping of the initial transient. The time lapse before the gate closes, after the signal has dropped below the chosen threshold level, can also be varied. The close threshold is engineered to be lower than the open threshold (known as hysteresis) so that a signal level which is on the borderline does no! confuse the unit as to whether it should be open or closed, which would cause 'gate flapping'.

Such units are useful when, for instance, a noisy electric guitar setup is being recorded. During passages when the guitarist is not playing the output shuts down so that the noise is removed from the mix. They are sometimes also used in a similar manner during multitrack mixdown where they mute outputs of the tape machine during the times when the tape is unmodulated, thus removing the noise contribu-tion from those tracks.

The noise gate is frequently heard in action during noisy satellite link broadcas3 and long-distance telephone-line operation. An impressive silence reigns when no one is talking, but when speech begins a burst of noise abruptly appears a accompanies the speaker until he or she stops talking. This can sometimes disconcerting for the speaker at the other end of the line because he or she gets the impression that the line has been cut off when the noise abruptly disappears

Noise gates can also be used as effects in themselves, and the 'gated snare drum is a common effect on pop records. The snare drum is given a heavy degree of gated reverb, and a high threshold level is set on the gate so that around half a second or so after the drum is hit the heavy 'foggy' reverb is abruptly cut off. Drum machines can mimic this effect, as can some effects processors.

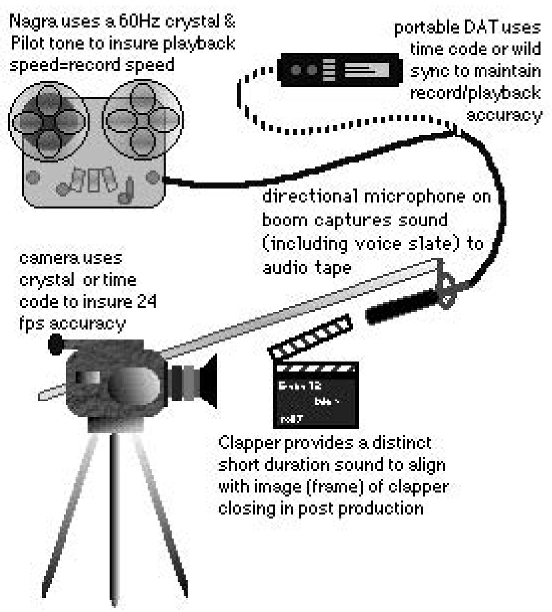
Digital Noise Extraction

Extremely sophisticated single-ended computer-based noise reduction systems been developed. They are currently expensive, and are offered as a service by -limited number of facilities. A given noisy recording will normally have a short -period somewhere in which only the noise is present without any programme , for instance the run-in groove of an old 78 rpm shellac disc recording provides a sample of that record's characteristic noise. This noise is analyzed by a computer and can subsequently be recognized as an unwanted constituent of the signal, and then extracted electronically from it. Sudden discontinuities in the programme caused by scratches and the like can be recognized as such and removed. The gap is filled by new material which is made to be similar to that which exists either side of the gap. .Not all of these processes are currently 'real time', and it may take several times longer than the programme's duration for the process to be carried out, but as the speed of digital signal processing increases more operations become possible in real time.

Notes :

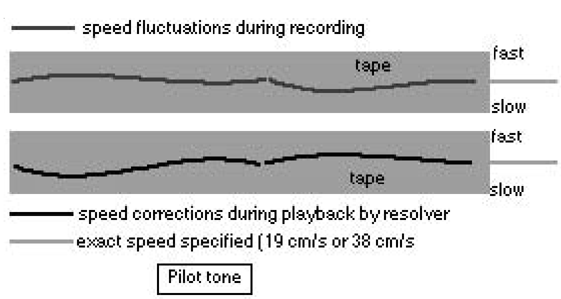
The Sync Path for Film

We have briefly discussed the use of Nagra to obtain sync sound in the previous semester. Let us look into the sync path for the film format from production to post. In this chapter we would also be looking into the other options in video.



On Location

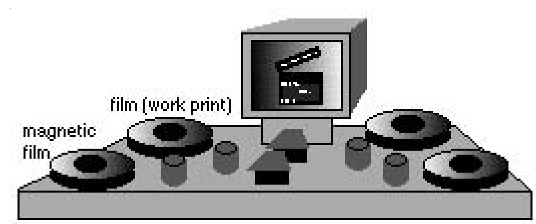
On location, audio is recorded onto analogue recorders like a Nagra (1/4" reel - reel tape recorder), a Walkman Pro (portable audio cassette) or a digital recorder or DAT. Synchronization on professional shoots usually involves the use of SMPTE time code and the use of a stereo Nagra or a "time code DAT".



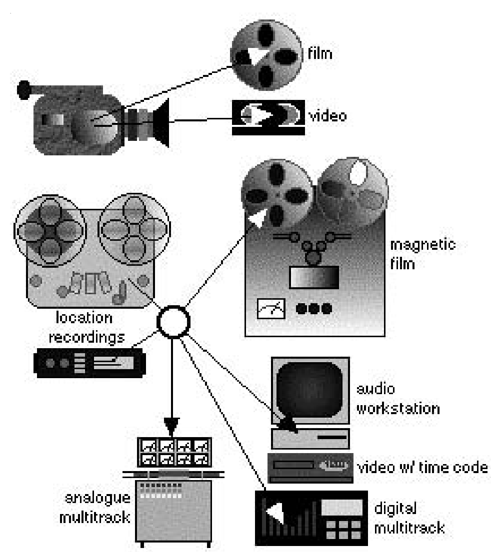
For semi-pro and student shoots, it is more common to find a mono Nagra using Pilot tone or a more simple, less expensive DAT. In any situation where lip sync is required, the camera must run at an exact 24 frames per second (generally 25 fps outside of North America). Also the camera must be relatively silent (or blimped) - especially when filming indoors.

For the student shoot, the Nagra maintains synchronization with the film through the Pilot tone recorded on the tape.
The slate or clapper plays a key role in this process by providing the sound and image for synchronization and by identifying pertinent sections of the project for screening "dailies" and for post production.

The procedure is for the recordist to start the recorder and to state ("rolling") when the device is ready to synchronize (for the Nagra this is when the white Pilot tone flag is visible on the recorder, for time code DATs this is when time code input is indicated) and for the clapper/loader to position the slate in front of the camera. The boom is positioned near the slate and when the two parts of the slate are snapped together after the camera has started filming, the scene, roll number, take, etc., are announced and the filming begins in earnest. The film is processed and either (usually) transferred to video for editing or a work print is made and that is cut on an editing machine.



Steinbeck film editing machine



Post Production

The Nagra tapes may be transferred via a resolver to a number of different systems:

1 Magnetic film, using a magnetic film recorder (Magnasync, Magnatech, etc.)

2 Digital Audio Workstation (DAW) such as a Macintosh running Deck, ProTools, etc.

3 Non-linear Video Editor such as an Avid, or DVision, etc.

Common to all Systems

Splitting Tracks

In the systems listed above there are many principles common to all. Primary is the splitting of sound into various tracks (real or virtual). The usual categories are:

- dialogue (minimum 2)
- sound effects
- ambiance
- foley
- music

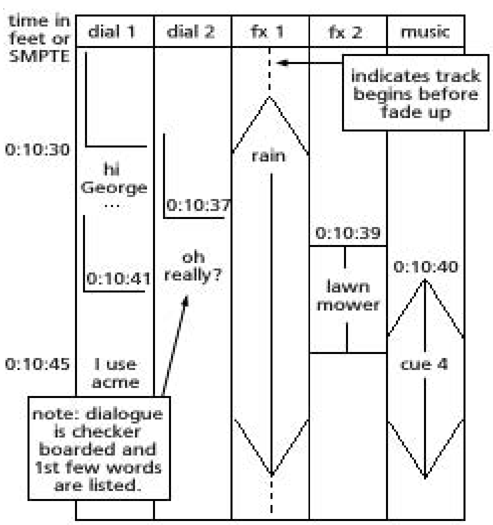
The MIX

Once the tracks have been prepared the next step is to prepare for “the mix”.

“The mix” refers to when the tracks are mixed to the final presentation format.

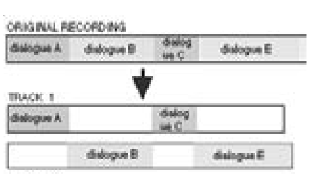
The usual venue is a studio configured for this purpose with a large projection screen and a mixing console customized for this purpose.

Preparation for this with magnetic film requires running the



audio tracks through a synchronizer.

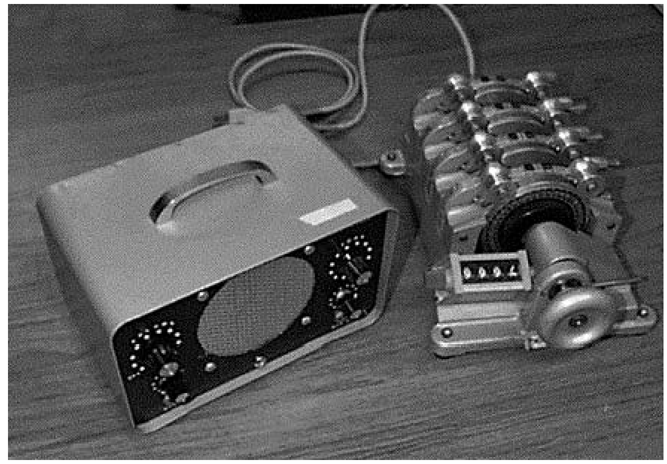
The second major principle is laying the sound onto these tracks so that sounds in the same category, which follow another closely, are placed on different tracks. (Checker boarding or splitting tracks). This is necessary for the mixing stage, which is



to follow, and insures flexibility as well as proper level and equalization settings.

Sync Block

This is a simple hand cranked system which insures that the tracks are in sync by providing interlocked sprocket wheels for all of the audio tracks and the work print. This process is necessary



because editing machines (like Steinbeck) cannot always be trusted to be frame accurate.

An accurate mixing sheet is then drawn (DAWs can create these automatically) for the mixers to follow. In the mixing studio a wide variety of audio playback and image systems may be found. Conventional studios use 16 or 35 mm dubbers or sound film playback machines synchronized to the work print or a black and white copy of the work print known as a slash print. The later may be marked with punched holes (cue punch) or lines (streamers) as visual cues for the mixers. A video projector may be used instead of a film projector: although the image lacks resolution, video rewinds/fast forwards much faster than film. Audio may also be stored on analogue or digital multitrack, time code DAT, sound film loops or be mixed directly from a professional DAW (such as a Synclavier or a ProTools system).

These various sources are mixed to a multitrack format - the minimum being three (dialogue, sfx, music) for mono releases. This is required to allow the dialogue to be dubbed into another language without affecting sfx or music and to facilitate the mixing process. (Usually dialogue is mixed first, then sfx and foley, followed by music.) Various mixes will be done for feature films; Dolby Surround, mono, International (w/o dialogue), etc.

The multitrack mix tape is then transferred to optical film, video or digital film format for release.

Audio Sync - A Short History

Any time an electronic signal must be coherent with a carrier, another signal or the real world, one practice of insuring such alignment is to use sync signal.

In video, we use a sync signal to insure that the horizontal scans, the vertical resets, the frequency and phase of the color subcarrier are all predictable and in phase with other signals with which they must be mixed. For example, if we are to mix two video signals, the sync reference on each signal can be processed and aligned so that both signals start each scan line at the same time.

In modern electronics, the idea of synchronisity is quite a bit different than in the past. Because of the accuracy of modern crystals it is possible to keep two independent signals “in sync” for long periods of time. Modern “syncing” has become more an issue of establishing and maintaining phase rather than simply frequency.

The double system sync system that we use in the motion picture business is quite typical of the various ways that many other electrical devices keep signals in sync in other applications. Originally, both camera and sound recorder were run off the same 3-phase electrical line. The motors used in these systems were designed to maintain their RPM based on the frequency of the AC power source. You can still see, on many stages in Hollywood, the old 3-Phase power connectors that were used to run the cameras and full coat recorders. Since both motors would run at exactly the same RPM (revolutions per minute), when the sound track and picture elements were mechanically locked together in projection, the actors voices could easily maintain the exact sync they had when they were recorded. By photographing and recording the “slapping” of the “sticks”, the visual indication of the sticks closing and the “pop” heard on the sound track, could be lined up as a reference. If both elements are then started exactly the same, were on a 3-phase motor system, and were in sync to begin with, the sound and image would have a perfect marriage.

Then, with the advent of super accurate crystals, it was possible to run both picture and sound drive motors at exactly the same speed from day to day, week to week and even month to month. This was only possible because both the picture stock and the sound recording media were perforated. If you ran the motor at exactly the same speed from one time to another, the track and pictures would follow easily because of the interlock of the drive shaft sprocket and the perforation in the recording or photographing media.

Full coat recorders were bulky, heavy and were terrible in the field. As capstan recording became possible, some method needed to be devised to compensate for the inherent slip in the recording media as it was fed through the system by the capstan and pinch roller.

The modern (non-timecode) Nagra has a system for effectively recording sprocket holes in the audio tape as the sound track is being recorded. This system is done by adding an additional track, usually between the two sound tracks, and recording a crystal based “pilot” on this additional track as the recorder is recording the production audio. Later, when the sound is being duplicated to a “full coat” master, a system called a “resolver” listens to the pilot tone, compares it with an internally generated master clock signal, then determines the exact speed that the capstan motor must run to compensate for any slip.

Typically, this pilot is a 50 hz signal. When the tape is resolved, if the signal is read at less then 50 hz, the electronics tells the motor to turn a little faster. If the signal is read at more than the internal reference of 50 hz, then the electronics tells the motor to turn a little slower.

It is easy to keep a sound system in sync for many minutes with a system that has an accuracy of at least two decimal places. For example, if the original pilot accuracy is 50.002345 hz and the reference in the resolver is 50.02543, then we will hear noticeable slip in the audio to image sync within a few minutes. We generally feel that a final frequency of 50.00XXXX (where XXXX can be almost anything) will be acceptable. With modern electronics this is a snap.

In certain situations, we can fool a system to run a little faster or slower by feeding an altered audio sync signal to the recorder when it is mastering the original sound.

Timecode

On a time code system, audio sync is maintained exactly the same way it is on a pilot system. Instead of a 50 hz tone, the additional track records an FSK (frequency shift keyed) record of a digital time code signal. This signal contains a running hours:minutes:seconds:frames record as well as some other information that is called User Bits and a few sync pulses.

The useful thing about time code is that each 1/2 frame (usually counted at 25 fps) is uniquely marked on the audio tape in a longitudinal fashion. In other words, each 1/8 th inch of the tape is uniquely encoded with a time coded digital word. In video mastered materials, this can be useful as the video tape can also contain this time code record. We hope that our film systems will someday be able to easily record time code on the film stock. Many experiments and prototype systems are being used currently, but wide spread use of a film recorded timecoded sync system is still in the future.

Other than the fact that the time code word contains “time of day and frames” time code information, the speed of a time code based audio tape is resolved exactly the same way that a pilot recorded tape is resolved.

Even in a timecode based system, different time based timecodes can be recorded on the tape. It is important to know what time base was used so that the tape can be properly resolved.

Digital audio is resolved the same way. Digital data on the tape contains sync bits that are unique and regular. If the player of a digital (DAT or Digital Audio Tape), signal detects the sync bits are being read slower than the sync reference in the player, the tape is made to go faster to compensate.

Audio Sync - A Description

In Film Situations for Film Finish

Pilot

A 50 hz tone is recorded on a dedicated section of the audio tape when it is mastering the audio track. In post production, this 50 hz tone is compared to another 50 hz tone. If there is a difference, this difference becomes an error signal that alters the capstan speed.

If the film camera is to be run at speed other an exactly 24.00 fps, we can feed a similarly altered pilot tone to the pilot track on the sound recorder. This is sometimes during video playback and process projection.

In Video Playback, if it becomes necessary to drive the camera with a television sync generator, the exact frequency of the camera speed will be 24.021777 fps. This is due to the fact that some video equipment will only operate when all of the sync signals are at proper divisions of the subcarrier. If this is the case, we can manufacture a new pilot with the same shift.

That is if the camera speed becomes a little more than 24.00(say 24.02) the Sync or Pilot frequency will all increase by a very small amount. Same is the case for when the camera speed becomes less than 24.00 fps.

We usually have devices on board our video systems to generate the standard pilot shifts that are necessary for creating the new altered pilot. If not, the postproduction department can be informed to resolve the material to an altered reference. If we do our job correctly, even if the camera is running off speed, the sound will run off speed at exactly the same rate and the sound and picture will stay in sync.

Timecode and DAT

There is no way to feed a modern timecode Nagra an altered sync signal during the recording session. If altered sync is necessary, the only wan to adjust is to adjust in post production. We simply indicate on the sound log that the camera was running at a different speed. This speed for our purposes will typically be 24.02 fps.

In Film Situations for Video Finish

When film is being shot for telecine, we have a new set of problems. Though many shows are photographed at 29.97 fps(in the US), many people prefer to photograph at 24.00 fps. This is usually an artistic decision, however a show can realize a 20 percent saving in their film budgets if they stay at 24 fps.

NTSC

When a film shot at 24.00 fps is transferred to video on a RANK telecine, the film does not really go though the RANK at 24.00 fps. Because a RANK is operating at 29.97 fps, in order to get all the information from each frame of film on a frame of video, the 3-2 pull down system is employed. Basically this means that the film runs though at an exact ratio of 24 to 30 that really translates to 23.95 to 29.97. Since no modern camera can run at other than 24.00 or maybe 29.97 or 30.00 and be at crystal speeds, when the film is transferred, it will be off speed. The practice of most video finish film shows is the same. We use a 60.00 hz word time code signal on the audio and run the camera at 24.00 fps. When the film is run in telecine, it will be a little slower by the difference between 23.95 and 24.00. Similarly, the sound, time code or pilot, that was recorded at 60.00 hz will be resolved to a television sync generator that runs at 59.94. The result is that both sound and picture are shifted in real time to slightly slower, but since they are shifted the same, they stay in sync.

PAL

Standard PAL/SECAM Telecine

PAL/SECAM video runs at exactly 25 fps, but film runs at 24 fps. Since the difference between 25 fps and 24 fps is so small, most PAL/SECAM films are telecined by simply speeding up the film by 4% to 25 fps. Since the video is sped up, they also have to speed up the audio by 4%. In addition, the audio is pitch-corrected to prevent voices and other sound effects from being too high pitched.

Do I Need to Inverse Telecine Standard PAL/SECAM Telecined Videos?

The answer to this question can be yes or no. Unlike NTSC telecining, PAL/SECAM telecining generally does not produce duplicated fields. However, even without the duplication of fields, interlacing artifacts can still arise. This is possible because there are two different ways in which the fields can be arranged in a PAL/SECAM telecined video:

TVFrame #1	**TVFrame #2**	**TVFrame #3**	**TVFrame #4**
1T	2T	3T	4T
1B	2B	3B	4B

"Good" PAL/SECAM Telecine (no interlacing artifacts)

TVFrame #1	**TVFrame #2**	**TVFrame #3**	**TVFrame #4**
2T	3T	4T	5T
1B	2B	3B	4B

"Perverse" PAL/SECAM Telecine (interlacing artifacts visible on a computer monitor)

In the "good" PAL/SECAM telecine technique, all of the fields are placed in the correct frames. This means that there will be no interlacing artifacts.

In the "perverse" PAL/SECAM telecine technique, half of the fields are shifted by one frame. This means that fields that do not belong together are stuck into the same frame. For example, the film field 2T will be combined with the film field 1B to produce TV Frame #1 (see the diagram directly above). This telecine method looks fine on a television, but produces interlacing artifacts on a computer monitor.

Fortunately, the artifacts of "perverse" telecine can be corrected without any loss of detail. If you look at the diagram directly above, you'll see that if we shift all of the top fields to the right by one frame, we'll have the correct output. 2T will then be combined with 2B; 3T will then are combined with 3B, and so on. There will no longer be any interlacing artifacts.

Once any "perverse" telecine is corrected in a PAL/SECAM telecined video, there are no interlacing artifacts, the file size is roughly the same, and the video plays as smoothly as the original 24 fps film. The only disadvantage to standard PAL/SECAM telecined videos is that they run 4% faster than the original film. For most purposes, the increased speed presents no problem.

If you did want to convert the movie back to its original 24 fps, you'd first have to use a program like VirtualDub to change the frame rate back to 24 fps. Then you'd have to use an audio editor like Cool Edit to slow down the audio by about 4% while pitch-correcting the audio to keep the pitch from being too low.

Alternate PAL/SECAM Telecine

In some cases, instead of speeding up the video, one frame is added to the video every second to change the frame rate from 24 fps to 25 fps. I'm not sure how this extra frame is introduced, but I believe it is added through two separate fields like

in NTSC telecining. If this is true, then interlacing artifacts will be present in the video.
To inverse telecine this kind of video, you should be able to use the Telecide filter for VirtualDub.

When to Use Audio Sync

It is necessary to use audio sync whenever the video system may be responsible for variation in the film camera speed from normal operation. It is necessary for the film camera to maintain .00 fps accuracy to insure that there will be no slip of audio sync to the camera speed when the audio is resolved to the film for viewing on a projector.

Use audio sync under the following circumstances.

1 When driving the film camera from a broadcast type sync generator, Grass Valley or Lenco at 24.02 fps. (When the OPTO-BUFFER is not in use).
2 When using a REAL-TIME Graphic System that runs at 24.02 fps.
3 When using the Video-Effects System.
4 When using a camera with unknown speed characteristics such as a visual effects plate camera when connected to an OPTO-BUFFER. See section on “Unusual Film Camera Synchronization”.
5 When performing video playback in a multi-camera show and it is necessary to drive all the cameras at other than 24.02 fps.

The vertical frequency (frame rate) of 24.021777 Hz causes a slight loss of sync in audio systems running at 24 fps. The error comes to approximately .02 fps, and may not be noticeable over small running times of one to two minutes.

Below is a list of running times and their corresponding sync loss:

Minutes	1	2	3	4	5	6
Frame Error	1.31	2.61	3.92	5.23	6.53	7.84

By adjusting the audio-sync frequency slightly, we compensate for this error.

Formatting Tapes for Sync Playback

As many of you are aware, when film shot at 24 fps or 30 fps is transferred to video for post-production (electronic editing), the actual running time of the film is reduced by one-tenth of one percent in order to compensate for the difference in frame rates

(frame lines) between cinematic projection and the continuous cathode ray scans of the video medium.

Other than rare instances of “recreational” chemical abuse, this accounts for the main reason that many rock videos are slightly out of sync.

In order to compensate for this speed reduction that occurs

during the film to video transfer, it is necessary to maintain a similar speed change in terms of our audio. That means that the playback version of the music that talent lip-syncs to must be played back on the shooting set one-tenth of one percent FASTER than the true speed of the song that will appear in the finished (video) product.

Using a Timecode Nagra for Playback:

If you will be using a Nagra IV-STC stereo timecode reel-to-reel recorder as your playback source, adhere to the following guidelines. Prepare an EDIT MASTER version of your song with an accompanying SMPTE timecode of25frames/sec. This is the version of the song that you will use in the edit bay to cut your video to.

From the EDIT MASTER, prepare your PLAYBACK DUPES. The dupes should be exact copies or mixdowns of the EDIT MASTER, with precisely identical 25 frames/sec time code recorded onto each copy.

At the head of the song, there should be a series of timing beeps (or countdown intro) so that the performers will be able to hit the first note of the song in unison. Record at least a couple of song passes onto each playback dupe to save rewind time.

Make at least two or three physical tapes, in case a segment of tape becomes damaged or recorded over during the shoot.

On the set, play back your dupe at the frame rate of 25frames/

sec. This will have the effect of speeding up your music by one-tenth of one percent in order to compensate for the fact that your film footage will eventually be slowed down by that same amount.

The timecode from the playback Nagra needs to be transmitted via a Comtek wireless system to the timecode slate, since the timecode numbers that we want to photograph need to be the timecode of the playback tape, and have no relationship to real clock time nor record start/stop time (as it would if we were doing live dialog recording for a motion picture).

Make sure that when you order your Nagra that you indicate that you will be doing sync playback, since the Nagra does not self-resolve without a special accessory box (resolver/ TCstripper) that must be added to your package, along with the Comtek transmitter system for the timecode.

Using DAT for Sync Playback:

The use of portable DAT recorders for sync playback has become increasingly popular. If you will be using a consumer or non-timecode DAT recorder, then it is necessary to prepare your PLAYBACK DUPES with a monaural audio mix on one track and SMPTE timecode on the other Consumer DAT's will only playback the tapes exactly as recorded, with no provision for speed-ups nor slow-downs. Therefore, DO NOT make an exact replica of your EDIT MASTER for sync playback. Instead, arrange for the recording engineer to record the tape at 25 frames/sec.

What you play back on the set must be the faster version of your song along with the faster time-code rate! Again, think in terms of multiple passes of the song on each tape, and multiple tapes in case of physical damage.
It will be necessary to transmit the timecode via Comtek to the slate, so make sure that you have all of the required adapter cables.

Using Timecode DAT for Playback:

When using a more sophisticated DAT recorder such as the Fostex or HHB machines that have timecode capability, it is essential to remember that tape speed and timecode speed can be independent of each other in the DAT domain.

Tape speed is determined by the sampling rate, period! A DAT tape initially recorded at 48K and played back at 48.048K will reflect a one tenth of one percent speed increase, or vice versa. However, the time code output will not change.

Similarly, a 48.048K tape that is played back at 48K might still output timecode at the 25 rate. So make sure that the machine operator is familiar with the peculiarities of his/her machine, and remembers to re-set both the sampling rate and the timecode output rate to the appropriate settings.

If the PLAYBACK DUPE is an exact replica of the EDIT MASTER, we can assume that it is recorded at 48K with a 25 time code. Play it back on the set at 48.048K, with the code reset to 25 frame.

If the DAT only does pull-downs (slow-downs), then transfer from the EDIT MASTER to the slower 47.96K sampling rate, so that when you play it back at 48K the song will be speeded up on the set by the proper percentage.

Determine whether or not the DAT machine in question automatically reconforms the output timecode to match the speeded up rate or does it keep the original timecode.

Very often, to eliminate confusion, the recording studio will format the PLAYBACK DUPE for proper playback at 48K/25 frame with instructions for the playback operator to not worry about pull-ups and pull-downs. But never assume. Ask. And ask again!

CD Playback:

A new process that we are developing is to prepare the PLAY-BACK DUPE onto a writ able audio CD, with audio on one track and correct time code on the other. Speed and time code changes are corrected during the transfer to CD, so that correct playback on the set is as simple as selecting a track on the CD and pressing play.

The PLAYBACK CD will contain a complete version of the song, along with a number of short excerpts for instant cueing.

Ask, but Never Assume

The bottom line is that the version of the song being played back on the set must be one tenth of one percent faster than the version of the song that the video editor is going to use in the finished video.

Timecode DAT demands attention to both speed and timecode changes during the playback and/or transfer process, depending on the capabilities of the make and model of the DAT recorder. Make sure that the operator is familiar with the machine and its programming routines! Make sure that you are not double-dipping by changing the playback speed of a PLAYBACK DUPE that has already been speed corrected by the studio.

Alternative to Timecode

There is a trend in our industry for clients and producers to clamor for the newest and latest technology, regardless of whether or not that technology will really improve the end product.

For example, with great amusement the enthusiasm a particular music video producer exhibited over using a DAT recorder for sync playback. "This is going to be stupendous! Imagine, we're going to shoot our video with digital playback! It's going to be hot!"

Well, does that make any difference? Since the playback track is only a guide track, and does not ever appear in the finished product, it makes absolutely no musical difference whether one plays back from a standard Nagra, a DAT, or any other sync device. The sound being played back is only so that talent has something to hear on the set.

But this producer was not looking at the technical process of making a music video. Instead, buzzwords and appearing trendy was at the forefront of his mind.

Case in point, recording SMPTE timecode on the audio track for shows that will be edited non-linear.

Having SMPTE timecode on the audio track that will match timecode on the picture is nice, but far from absolutely necessary. Considering the expense of purchasing or renting timecode recorders and slates compared to being able to use existing non-timecode equipment, one should definitely explore all of the post-production ramifications before blindly leaping into costly, albeit trendy, production sound decisions.

Did you know that up until only recently, TV shows such as "Beverly Hills 90210" did not use SMPTE timecode when recording production sound? All audio was done with the venerable Nagra 4.2, and then transferred to non-linear digital for post. Why? Because it was cheaper to do it that way, and gave them the same results!

Here is what happens when audio is recorded with SMPTE timecode. Timecode is recorded, along with production sound, on a Nagra IV-STC stereo recorder or a sophisticated DAT such as the HHB or Fostex PD4. Matching (jam sync'd) timecode may or may not be recorded on the film by means of in-the-camera keycode and an Aaton master clock module. A Denecke slate is filmed at the head of each scene, displaying a visual timecode as well as providing an old fashioned clapstick marker.

In post, the film is transferred to video in the telecine and then digitized into the non-linear editing system. Audio is resolved at the proper speed (slowed down slightly to match the picture slowdown created by telecine) and also digitized into the non-linear system. Using the timecode numbers as a "beginning of the scene" startmark or line-up reference, the editor performs a series of in-computer "audio insert edits" to sync up the dailies (matching up the picture and corresponding sync audio) for each take.

Now, examine what happens if no timecode is recorded on the audio during production. Just as before, the picture is loaded into the edit computer. Audio is resolved at the proper speed, and also digitized into the system. In order to sync the dailies, the editor goes to the picture start of the take (clapstick frame) and "parks". Audio is advanced to the audio "marker" (the clapstick impact); and then the mark-in edit points are punched in.

Finding the start mark of the audio without timecode is easy. If one watches the visual waveform of the audio (the "optical track"), it is rather easy to locate the clapstick because it sticks out like the Washington Monument! With very little practice, an editor can sync dailies almost just as fast as with timecode, and at considerable savings of the production budget.

But without timecode, how does the edit computer keep everything in sync? The same way it always does, by means of its own internal timecode. Since most production timecode is discontinuous, it is only used for negative matching; the actual editing is done with a form of continuous timecode within the system.

It is true that without timecode, we cannot go back to the original production audio tapes and conform them with the negative for post. But why would we want to or need to? The audio coming out of the non-linear system is digital CD quality or better, far higher quality than we ever got off of a Moviola. In the old days of tape splicing, we had to re-transfer and conform the audio in order to correct for the entire damaged sprocket holes, bad splices, and unintentional edits. But since out digital soundtrack is perfect, we do not need to return to the original tapes before moving on to advanced soundtrack building.

So you could go without the timecode as well. But if you can afford it and its in the budget and then why not go for it, right?

Notes :

THE ART AND TECHNIQUE OF POSTPRODUCTION SOUND

The credits for John Ford's My Darling Clementine (1946) include Wyatt Earp as technical consultant but only one person responsible for all of postproduction sound (the composer). The credits for Lawrence Kasdan's Wyatt Earp (1994) list the names of thirty-nine people who worked on postproduction sound. The difference is not simply a matter of expanding egos or credits.

"An older film like Casablanca has an empty soundtrack compared with what we do today. Tracks are fuller and more of a selling point," says Michael Kirchberger (What's Eating Gilbert Grape?, Sleepless in Seattle). "Of course a good track without good characters and storyline won't be heard by anyone."

With soundtracks much more dense than in the past, the present generation of moviemakers has seen an exponential growth in the number of people who work on the sound after the film has been shot. What do all those people add both technically and esthetically? "When I started out, there was one sound editor and an assistant," says picture editor Evan Lottman (The Exorcist, Sophie's Choice, Presumed Innocent). " As editor for a big studio picture in the early Seventies I usually cut the ADR [dialog replaced in postproduction-EW] and the music as well." Today an editor on a major feature would either hire a supervising sound editor who gathers a team of sound specialists, or go to a company like C5, Inc., Sound One, or Skywalker that can supply the staff and/or state-of-the-art facilities.

Sound is traditionally divided into three elements: dialog, music, and effects (any auditory information that isn't speech or music). Although much of the dialog can be recorded during principal photography, it needs fine tuning later. And almost all other sound is added during postproduction.

How does sound get on pictures? The following is a rough sketch of the procedure for a major Hollywood feature production. But it is not a blueprint; exact procedures vary tremendously with the budget and shooting schedule of the film. Blockbuster action films, for instance, often devote much more time and money to sound effects than is described below. The process certainly does not describe how the average film is made in other countries; few other cultures have such a fetish for perfect lip-synching as ours-so even dialog is recorded after the shoot in many countries. In India the trends keep changing. Most of the film are dubbed later, but films like Laggan have on location sync sound.

This lesson can only begin to suggest how digital technologies are affecting post-production sound. For one thing, there is wide variation in types of systems; for another, digital sound techniques are evolving faster than alien creatures in a science fiction movie.

Production

Even the sound recorded live during principal photography is not wedded physically to the image and has to be precisely relinked during postproduction. It is usually recorded on 1/4" magnetic tape (though there are alternatives) and marked so that it can be ultimately rejoined with the picture in perfect synchronization.

On the set the location recordist (AKA production mixer) tries to record dialog as cleanly and crisply as possible, with little background noise (a high signal-to-noise ratio). A boom operator, usually suspending the microphone above and in front of the person speaking, tries to get it as close as possible without letting the microphone or its shadow enter the frame.

As you already know an alternative to a mike suspended from an overhead boom is a hidden lavalier mike on the actor's chest, which is either connected to the tape recorder via cables or wired to a small radio transmitter (cordless mics) also hidden on the actor. But dialog recorded from below the mouth must be adjusted later to match the better sound quality of the boom mike. And radio mikes can pick up stray sounds like gypsy cabs.

While on the set, the sound recordist may also ask for a moment of silence to pick up some "room tone" (the sound of the location when no one is talking), which must be combined with any dialog that is added during postproduction (with reconstructed room reverberation) so that it matches what is shot on the set. (We don't usually notice the sound of the breeze or a motor hum, but their absence in a Hollywood product would be quite conspicuous.) The set recordist may also capture sounds distinctive to a particular location to give the postproduction crew some sense of local color.

Post Production

Theoretically, the first stage of sound editing is "spotting," where the editor(s) and possibly the director go through each second of the film with the supervising sound editor in order to generate a list of every sound that needs to be added, augmented, or replaced. This practice has fallen prey to demands for early previews, which have wreaked havoc on postproduction schedules.

Dialog

Dialog editing is mostly a matter of cleaning up production sound. The work can be as detailed as reusing a final consonant of one word to complete another where it had been obscured or removing an actor' s denture clicks.

Some of the dialog heard in the completed film was not recorded on location. Shooting silent (MOS) is much easier than having to achieve perfect quiet from the crew, the crowd watching the film, or airplanes and birds passing overhead. Even with the compliance of onlookers, nature, and ubiquitous car alarms, however, miked dialog may be unusable because it picked up extraneous noises such as a squeaky camera dolly or clothing rustle.

Despite these difficulties, directors almost always prefer production dialog, which is an integral part of the actors' performances, to looping (rerecording speech in post-production). Although there is a trend in looping sessions toward using booms and the original microphones to mimic the situation on the set, it is nearly impossible to duplicate all the conditions of the shoot. Orson Welles found that out after shooting the festive horseless carriage ride in The Magnificent Ambersons. Because the scene was photographed in an ice plant with tremendous reverberation (which wouldn't be heard outdoors), the dialog of all six characters had to be looped. When Welles heard the original looping, he rejected it because the voices were much too static; they didn' t sound as though they were spoken by people in an automobile. The soundman's low-tech solution was to redo all the lines with the performers and himself seated on a twelve-inch plank suspended between sawhorses. For a week, says James G. Stewart, "As we watched the picture I simulated the movement of the car by bouncing the performer and myself up and down on the plank."

It is tough, however, for actors to match later the emotional level they achieved on the set. Ron Bochar, who supervised the sound on Philadelphia, describes the powerful scene where Tom Hanks is responding to an opera recording as a case in point. Ideally the aria and the dialog would be on separately manipulable tracks so that the dialog could be kept intelligible. But Hanks wanted both the freedom to move around and the ability to hear and react to the singing of Maria Callas. As a result, both his dialog and her aria are recorded on the same track and the dialog is less than ideal. But everyone involved agreed that the live performance was preferable to looping the scene. "That' s one of those things about 'mistakes' that get put in because you are forced to or they just happen," says Bochar. "They turn out to be things that you could never re-create. You'd ruin the scene by making it cleaner."

Today, one of the first jobs of dialog editors is to split spoken lines (usually from different camera-hence microphone-angles) onto separate tracks. Doing so, says Kirchberger, "makes them as independently controllable as possible, so that we can later 'massage' them in such a way that they fit together seamlessly." This is not to say that filmmakers can't do creative things with dialog.

Robert Altman, most notably, developed with rerecording mixer Richard Portman atechnique for creating his unique multilayered dialog style. During the shoot Altman, who allows a lot of improvisation, mikes each of his simultaneous speakers on separate tracks (sixteen for Pret-a-Porter). Later the rerecording mixer can raise and lower the relative volume of each track to create a weaving effect among the various actors' lines.

Dialog can also be edited to affect characterization. Suppose the director wants to make an arch-villain more domineering. A mixer could raise the volume of his voice and adjust the tonal qualities to make him sound larger than life. It's the aural equivalent of someone invading our space by standing too close to us. The picture editor could enhance the villain's sense of menace by regularly cutting to his voice before we see him. Because he seems to lurk just beyond the edges of the frame, the viewer will feel uneasy about his potential reappearance whenever he is not present.

ADR

Dialog that cannot be salvaged from production tracks must be rerecorded in a process called looping or ADR (which is variously said to stand for "automated" or "automatic" dialog replacement). Looping originally involved recording an actor who spoke lines in sync to "loops" of the image, which were played over and over along with matching lengths of recording tape. ADR, though faster, is still painstaking work. An actor watches the image repeatedly while listening to the original production track on headphones as a guide. The actor then reperforms each line to match the wording and lip movements. Actors vary in their ability to achieve sync and to recapture the emotional tone of their performance. Some prefer it. Marlon Brando, for instance, likes to loop because he doesn't like to freeze a performance until he knows its final context. People have said that one reason he mumbles is to make the production sound unusable so that he can make adjustments in looping.

ADR is usually considered a necessary evil but Bochar has found there are moments when looping can be used not just for technical reasons but to add new character or interpretation to a shot. "Just by altering a few key words or phrases an actor can change the emotional bent on a scene."

Sound Effects

Dialog editors are usually considered problem solvers rather than creative contributors, but there's considerable room for artistic input in choosing and editing sound effects. For one thing, sound effects tracks are normally built from scratch. We would not want to hear everything that really could be heard in a given space, Even if it were possible to record only the appropriate noise on the set while the film is being shot, it wouldn't sound right psychologically. Sound is very subjective and dependent upon the visual context and the mood set up in the image. The soundtrack of real life is too dense for film. In the real world, our minds select certain noises and filter out others. For instance, we mentally foreground the person speaking to us even if the background is louder. On film, the sound effects editors and rerecording mixers have to focus for us.

Focusing on selected sounds can create tension, atmosphere, and emotion. It can also impart personality to film characters. Walter Murch (the doyen of sound designers) once described the character sounds (in a film he directed) as "coronas" which can magnify each character' s screen space. A figure who is associated with a particular sound (often suggested by his or her clothing), has "a real presence that is pervasive even when the scene is about something else or the character is off-screen."

Indeed, sound is a major means to lend solidity and depth to the two- dimensional screen image. Furthermore, new digital release formats allow filmmakers to literally "place" sounds at various locations throughout the theater. Thus sound can expand space, add depth, and locate us within the scene.

A crucial difference between visual and aural manipulation of the audience is that even sophisticated audiences rarely notice the soundtrack. Therefore it can speak to us emotionally and almost subconsciously put us in touch with a screen character. In a film like Hitchcock' s The Birds, for example, any time we see a bird we know we are being titillated. But by merely adding a single "caw" to the soundtrack on occasion, Hitch was able to increase the tension without our being aware of his manipulation.

To understand the manipulability of effects it is useful to know how effects tracks are created. A regular source of effects is a stock library, where sounds are stored on CD. The rest have to be recorded or combined from several sources. Foleying is the "looping" of sound effects by a specialized department in a studio designed for watching the picture and creating the sounds at the same time. The process is named after its developer, legendary sound man Jack Foley of Universal. Because virtually all footsteps are replaced, a foley stage usually includes several pits with different sounding surfaces on which the foley artist will walk in time to the one or more characters he or she is watching. Clothing rustle (another sound we never notice until it's missing) and the movement of props such as dishes are likely to be recorded here as well. Even kisses are foleyed. A steamy sex scene was probably created by a foley artist making dispassionate love to his or her own wrist. The foley crew will include the artist or "walker," who makes the sound, and a technician or two to record and mix it.

Foleying needn't be a slavish duplication of the original object. The sound crew can characterize actors by the quality of the sounds they attribute to them-say, what type of shoes they wear. To attribute some subtle sleaziness' to Nicolas Cage's lawyer in It Could Happen to You, Michael Kirchberger's foley crew sonically added a squeaky shoe and rattling pocket change as Red Buttons walks around the courtroom. It's the opposite shoe of the one that squeaked in Jerry Lewis movies, says Kirchberger.

Usually the more exotic-less literal-sounds are created by the effects staff. According to Murch, "That's part of the art of sound effects. You try to abstract the essential quality of a sound and figure out the best way to record that, which may not be to use the thing itself but something else." Thus, some sounds have nothing to do with the original source-the real thing would be unconvincing. Mimi Arsham, who worked on Ben-Hur, reports that the sound of a whip cracking was actually a hefty steak being slapped on a thigh.

Most sounds need processing (fiddling with). The most common strategy is to start with a sound made by a source that is the same as or similar to what was photographed and then to distort it. One simple method is to slow it down or speed it up. Two other common processing tricks are to choose just part of the frequency spectrum or to run a sound backwards. As far back as 1933 the original sound man at RKO created King Kong's voice by playing backwards the roar of a lion he recorded at the San Diego Zoo. Today digital editing techniques have vastly expanded the possibilities: a sound editor feeds a sample of a sound into a computer, which can then manipulate it and provide a whole range of sounds from the original. One powerful tool is the Synclavier, which combines a computer sampler and a keyboard that can play a sound (or sounds) assigned to any of seventy-three keys with the stroke of a finger.

New sounds can also be created by mixing disparate sources. In order to accentuate the idea that the pen is mightier than the sword, the final close-up of the typewriter keys pounding out the Watergate expose in All the President's Men combines gunfire with the sound of clacking typewriter keys.

Many of today's sound effects are "stacked"; they are layers of combined sounds from different sources that often begin organically but are processed digitally. Kirchberger reports that he created the roar of the Komodo Dragon in The Freshman by starting with tapes of vultures recorded for Ishtar. The sound was processed, added to other sounds including a pig, and then vocalized through digital sampling. "I knew we had something that was vaguely reptilian. What made it 'talk' was altering the pitch as we played back the stacked sample. That gave it the vocalization we needed, as opposed to its being just a screech or a caw."

Much of the freedom in sound design comes when making horror or science fiction films, where stylization is the norm. Most sonic sources are hard to identify unless we see them—and films of the fantastic have sources we have never heard in real life. So there is great latitude in deciding how something should sound.

However technically sophisticated the equipment that processes sound, the original source can be quite mundane. Gary Rydstrom, the lead sound designer at Skywalker, likes to challenge listeners to a game of "name that sound," that is, to guess the sources of his sounds- -exotic noises he created from prosaic origins. One favorite tool, he says, is air compressed in a can. The source of the "sliming" noise in Ghostbusters, for example, is Dust-Off sprayed into Silly Putty. He is also proud of the sound of the mercury-like character (T-1000) passing through steel bars in Terminator II. Seeking a sound that was part liquid, part solid, Rydstrom came up with the sound of dog food being extruded from a can.

The majority of the sound crew is not brought onto a picture until it is "locked" that is, the image is finalized. On films where sound is considered a major creative element, directors may hire a sound designer like Walter Murch (Apocalypse Now, The Conversation, The Godfather) or Skip Lievsay (who creates sound for the Coen brothers, Martin Scorsese, and David Lynch). "Sound designer" is an elusive term which can refer to a person brought on to create just one kind of effect (for example, Bochar was hired late in the postproduction of Wolf just to create the effects that accompanied Nicholson turning into a beast). In some cases, however, sound designers are thought of as artists who are brought on staff during the planning stages of a film, along with the set and costume designers, and who do their own mixing. In these instances, the sound designer works with the director to shape an overall, consistent soundtrack that exploits the expressive possibilities of the sound medium, is organically related to the narrative and thematic needs of the film, and has an integrity not possible if sound is divided among an entire bureaucracy. A case in point would be Jurassic Park, where Gary Rydstrom first designed the sounds of the dinosaurs and then models were built to match those roars.

On the average A-picture the first postproduction sound person brought onto the film is the supervising sound editor, who not only directs and coordinates the creative contributions of the postproduction sound staff but also must handle all the related administrative duties like scheduling mixes.

Although the supervising sound editors are usually not consulted during shooting, in the best of all possible worlds they are in touch with the location sound recordist during and after the shoot so that their work can be coordinated. Bochar feels strongly that his work should start early on: "To me the whole adage is that postproduction begins the first day of production."

Like most filmmakers, sound personnel work under extreme time constraints. One way for them to get a headstart is to work on a picture one reel at a time. Thus, if a director and editor are satisfied with reels two and three, they can send them on to the sound editors while they are still solving picture problems on other reels.

Scratch Mixes/Temp Tracks

Today the tendency is to bring the supervising editor on earlier and earlier. The main reason is the changing demands for sound in early screenings. According to Kirchberger, this practice has engendered the greatest changes in the logistics of postproduction sound in the last two decades.

As Lottman describes it, "On most A-pictures a sound editor will come on some time before the picture gets locked. You can't put them on too soon; that's too expensive. But you put them on, say, before the first screening. Now there's this big trend towards scratch mixes at screenings. Most directors don't want to screen a picture for anybody unless it has a complete and full soundtrack-a temp track with temporary sounds, temporary music and dialog to give the audience a preview of what the final, polished soundtrack will belike. They'll try to iron out a dialog scene where the sound shifts dramatically from cut to cut. They didn't use to do this at all. Now they do it on any mid- to high budget film. You try to keep it simple: you have just one sound editor and an assistant, perhaps."

Because of demands for scratch mixes the sound editors are under greater time constraints than ever. By the first scratch mix, the editors must have cleaned up noticeable sound-image problems and supplied the major effects. Yet this is also the best time to introduce their most inventive ideas, while directors and producers are still open to experimentation.

One result of scratch mixes is that they become weeding-out processes. During this stage sound editors, given the time, have a certain amount of latitude to present creative options to the director. One downside, says Kirchberger, is that if the director likes parts of the scratch mix, those sounds may never be refined even though they were just presented as a sketch.

Music

Like the foley crew, the music personnel are a discrete department. The composer may be brought in as early as the first cut to discuss with the director and editor the general character of the music and its placement in the film.

The person who spends the longest time on the scoring is the supervising music editor. It is the job of the music editor to spot every cue, that is, to make a precise list of timings-to the split second- for each appearance and "hit" (point of musical emphasis) of the music. In addition, the editor will log all the concomitant action and dialog during each cue. The composer then has about six weeks to come up with a score. The supervising music editor will set up recording sessions, which for, say, thirty minutes of music, take four to five days. Each set of instruments has its own microphone and track so that scoring mixers can balance them.

Apart from esthetic issues, film music composers must deal with particular technical requirements. For the sake of clarity, a film composer must orchestrate with instruments that do not overlap much with the frequency of the human voice or any dominant sound effects to be heard at the same time. In theory, composers keep in mind any anticipated noises for a sequence so that the music and effects aren't working at cross purposes. In practice, music editors often serve as master tacticians caught between the work of the sound editors and the composer who says: "Dump those goddamn sound effects!"

The scoring is also affected by the need for scratch mixes, for which the music editor has had to select temporary music. This may be a counter-productive trend. The editor will probably use music that was composed for an earlier film. As the producers and directors get used to their temporary track they often want something similar, so the composer is inadvertently rewarded for not straying far from what has already proved successful.

One of the more positive changes in scoring practices has been made possible through computer programs and synthesizers for musicians. Instead of presenting their ideas to the director at a piano, composers can now present them in a form "orchestrated" with simulations of different instruments.

Rerecording (The Mix)

The climactic moment of postproduction sound is called the "mix" in New York and the "dub" in L.A. On the screen the credit goes to a rerecording mixer, but that term is rarely heard in daily parlance, says Lottman; "If we said we were going to a rerecording mix, they' d laugh."

At the mix all the tracks-singly called elements-are adjusted in volume and tonal quality relative to each other and the image. (At some mixes the music editor and effects editors may be sitting at the "pots" controlling their subsets of tracks.) During the mix the director and/or picture editor will decide with the mixer which sounds should be emphasized. A composer can find that a particularly inspired fugue has been dropped in one scene in favor of sound effects or dialog. However much effort the composer and effects editors may have put into their creations, their efforts are sub-sentient to the ultimate dramatic impact of the overall sound plus picture. Asked what makes a good mixer, Bochar says, "The best mixers, like Richard Portman, Lee Dichter, and Tom Fleischman have the ability to leave their egos at the door. No one has to lay claim on the track. Mixing becomes an experience, as opposed to a job and drudgery. When those moments hit, it just soars."

The top mixers are orchestrators who create a sonic texture. You can' t have wall-to-wall noise, says Rydstrom; like music, the sound effects have pitch, rhythm, and pace which must be varied to create interest and may be manipulated to raise and lower dramatic tensions.

The mixer also has to equalize, blend, and balance the tracks for the seamless, invisible style that characterizes Hollywood style cutting. Thus, at a minimum, the mixer must match sounds created by dozens of technicians in different times and places. The engine roar of a 1954 Chevy may include sound obtained from a stock library, recorded on the set, and augmented with new recordings during postproduction. It may have been "sweetened" with synthesized sound. But it has to sound like one car.

Mixers have a number of tools. Equalizers and filters, for example, can boost or decrease the intensity of low, middle, or high frequencies in order to make dialog or sound effects match those that came from microphones and sources with different characteristics. Filters are also used to eliminate unwanted steady frequencies, such as the buzz of an air conditioner. In dealing with image size, the mixer adjusts perspective (determined mainly by the ratio of direct to indirect or reflected sound), which can be manipulated through the addition of artificial reverberation.

Great rerecording mixers are artists as much as technicians. The mixers' console is their palette: they have an infinite number of choices for blending. Their tools can be used in expressive ways. For example, an annoying voice can be adjusted to sound more screechy, or the roar of an approaching truck can be made more ominous. At the mix some of the many sound effects are heightened and others are lowered or eliminated. Sounds can be emotionally effective even when they are reduced to near inaudibility. (See, for example, the sidebars on Silence of the Lambs and Philadelphia.) And the most eloquent " sound" of all may be silence. In our age of dense soundtracks, the sudden absence of noise can have a stunning impact.

A mix on an average A-picture combines at least forty to sixty tracks, and perhaps hundreds. Therefore for manageability of dense soundtracks there may be any number of premixes, wherein groups of tracks are combined and equalized in relation to each other. For example, twenty- four tracks of foleys may be boiled down to one or two six-track elements. A typical final mix might begin with seven six-tracks: two six-tracks each for effects and foley, and one each for backgrounds, dialog, and ADR. Dialog is usually mixed first. In Murch's words, "Dialog becomes the backbone of the sound and everything else is fit into place around that."

Given that a mix costs from $400 to $800 or more an hour, sound editors do as much in advance as possible so that the mixer can worry about the bigger balance rather than hundreds of small adjustments. With track separation, the remixed tracks need not be permanently wed to one another. If at the final mix of a car crash, the director chooses to emphasize one sound of shattering glass, that specific element can still be manipulated if necessary. Often the director or editor is given a choice among several types of sound for a given effect.

Technology has inevitably affected the esthetics of the mix. A few decades ago, merely pausing to make a correction would create an audible click, so an entire reel had to be mixed in one pass or started over. Then, with the advent of "rock 'n' roll" systems, mixers were able to move back and forth inch by inch. Once consoles became computerized to "remember" all the mixer's adjustments, says Murch, he was able to think in larger units. "You take a sweep through the reel, knowing that there are certain things you're doing that are not perfect. You get the sense of the flow of a ten minute or longer section of film, rather than doing it bit by bit. So you go through in ten minute segments until you've got the basic groundwork for what you want, knowing that there are things wrong in there that you can fix later. It's like a live performance: sometimes there's something that happens spontaneously that way, that you can never get when you're trying to do it inch by inch. Thus automated mixing allows you to work in large sections but it also encourages you to be very finicky about small things and it doesn't penalize you for that."

The end product of the final mix is not just one printmaster from which domestic exhibition prints are struck; effects, dialog, and music are kept discrete to allow for release in different formats ranging from monaural optical 16mm tracks, to multichannel digital systems, to foreign versions minus the dialog.

Directors and Sound

The soundtrack is perhaps the most collaborative component of filmmaking. It is created by all the personnel mentioned above plus their assistants. Nevertheless, the editor and ultimately the director do call the shots. How do sound personnel communicate with directors?

There have always been a few directors particularly attuned to the expressive potential of sound; these include Robert Wise, Orson Welles, Robert Altman, and Alfred Hitchcock. Hitchcock, for one, usually prepared a detailed list of sounds and was actively involved in choosing them. (For the sound of the knife entering the body in Psycho's shower scene, Hitchcock did a blind sound test among different types of melon, finally settling on a casaba.) These sound-sensitive directors often incorporate sound as part of the basic conception of their films. For example, Hitch experimented with expressionistic sound (Blackmail), interior monologues (Murder), subliminal sound (Secret Agent), and electronic sound (in The Birds, which orchestrates computer-generated noises and has no underscoring).

Other directors do not think creatively about sound but choose personnel who do. These directors may have unerring instincts for the best sound when presented with several specific options. Most directors, however, do not use the expressive potential of the soundtrack and leave sonic decisions up to their staff.

In general, the younger generation of filmmakers are more savvy than their elders. For one thing, they were part of the revolution in music technologies. For another, they were probably exposed to sound courses in film school. According to Murch the very raison d'etre for Coppola' s team in creating Zoetrope was to have their own sound facility. And a few of today's directors consider sound an equal partner with image. (But even these directors still may have to figure out how to convey their sonic ideas-Jonathan Demme has been to known to ask his sound editors for "something blue.")

The best way to appreciate the expressive possibilities in an American soundtrack is to study in great detail virtually any movie by the sound-sensitive directors, such as Altman, the Coen brothers (try Barton Fink) or David Lynch, among independents. To find the most interesting soundtracks in other Hollywood productions, check the sound credits. The most respected sound designers and supervisors may be called technicians, but their artistry can be heard in all the films they touch.

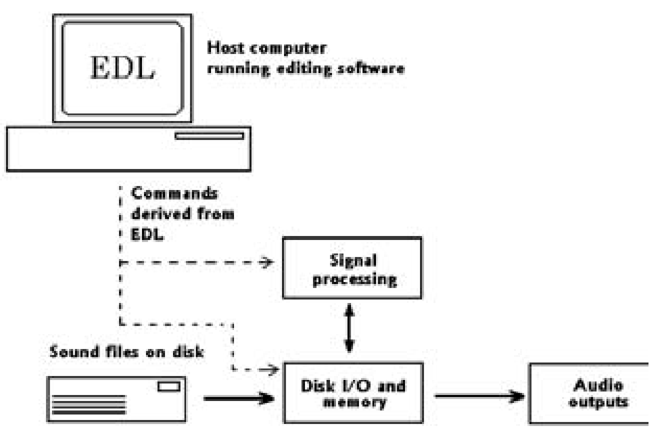
Notes :

Non-Linear Editing

Speed and flexibility of editing is probably one of the greatest benefits obtained from non-linear recording. With non-linear editing the editor may preview a num-ber of possible masters in their entirety before deciding which should be the final one. Even after this, it is a simple matter to modify the edit list to update the master. Edits may also be previewed and experi-mented with in order to determine the most appropriate location and processing operation which is less easy with other forms of editing.

The majority of music editing is done today using digital audio workstations, indeed these are now taking over from dedicated audio editing systems because of the speed with which takes may be compared, cross-fades modified, and adjust-ments made to equalisation and levels, all in the digital domain. Non-linear editing has also come to feature very widely in post-production for video and film, because it has a lot in common with film post-production techniques involving a number of independent mono sound reels.

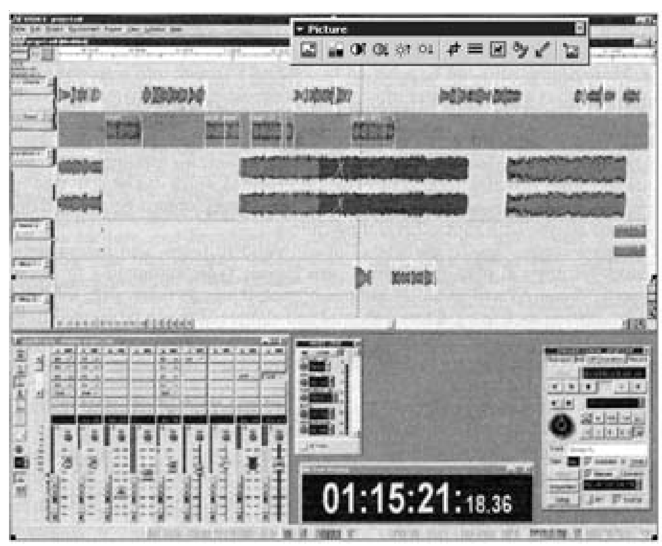
Non-linear editing is truly non-destructive in that the edited master only exists as a series of instructions to replay certain



parts of certain sound files at certain times, with certain signal processing overlaid, as shown in the Figure.

The original sound files remain intact at all times, and a single sound file can be used as many times as desired in different locations and on different tracks without the need for copying the actual audio data. Editing may involve the simple joining of sections, or it may involve more complex operations such as long cross-fades between one album track and the next, or gain offsets between one section and another. The beauty of non-linear editing is that all these things are possible without in any way affecting the original source material.

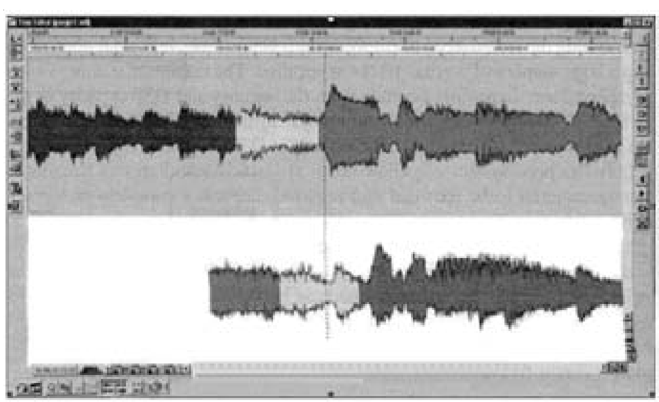
The manipulation of sound files on screen is normally under the control of a word processor-type of PC program, and it enables fast, intuitive editing. In the example illustrated here, the programme material is recorded onto the hard disk via a break-out box, which carries analogue and digital inputs and outputs. Traditional-style transport controls are provided on-screen, these being operated with the mouse. Elsewhere on the screen a section will show mouse-operated mixer-type controls such as fader, pan, mute and solo for each channel, plus output sections The main part of the screen is occupied by a horizontal display of empty recording tracks or ‘streams’, and these are analogous to the tracks of a multitrack tape recorder. A record icon associated with each stream is used to arm it ready for recording. As recording proceeds, the empty streams are filled from left to right across the screen in real time, led by a vertical moving cursor. These streams can be displayed either as solid continuous blocks or as waveforms, the latter being the usual mode when editing is undertaken. After recording, extra streams can be recorded if required simply by disarming the record icons of the streams already used and arming the record icons of empty streams below them, making it possible to build up a large number of ‘virtual’ tracks as required. The maximum number, which can be replayed simultaneously, depends upon the memory and DSP capacity of the system



used. A basic two-input/four-output system will usually allow up to eight streams to be replayed (depending on the amount of DSP being used for other tasks), and a fully equipped system can allow up to at least 32 simultaneous streams of programme material to be recorded and replayed, i.e. it is a complete multitrack recording machine.

Replay involves either using the transport control display or clicking the mouse at a desired position on a time-bar towards the top of the screen, this positioning the moving cursor (which is analogous to the tape head) where one wishes replay to begin. Editing is performed by means of a razor-blade icon, which will make the cut where the moving cursor is positioned. Alternatively, an edit icon can be loaded to the mouse's cursor for positioning anywhere on any individual stream to make a cut. Word processor-type operations such as delete, cut and paste, copy and paste and remove gap can then be performed on highlighted sections, streams or sections of streams can be dragged across to other positions and on to other streams, and there are many stages of undo and re-do so that nothing need be permanent at this stage. More precise edits can be achieved by loading a section into a special trim -editor which gives a greater degree of waveform magnification, and parameters such as cross-fade characteristics can be experimented with and auditioned. When a satisfactory edit is achieved, it can be written back to the main display where it will be incorporated. Scrub and jog actions for locating edit points are also possible. A useful 'lock to time' icon is provided which can be activated to prevent horizontal movement of the streams so that they cannot be accidentally moved out of sync with each other during editing.

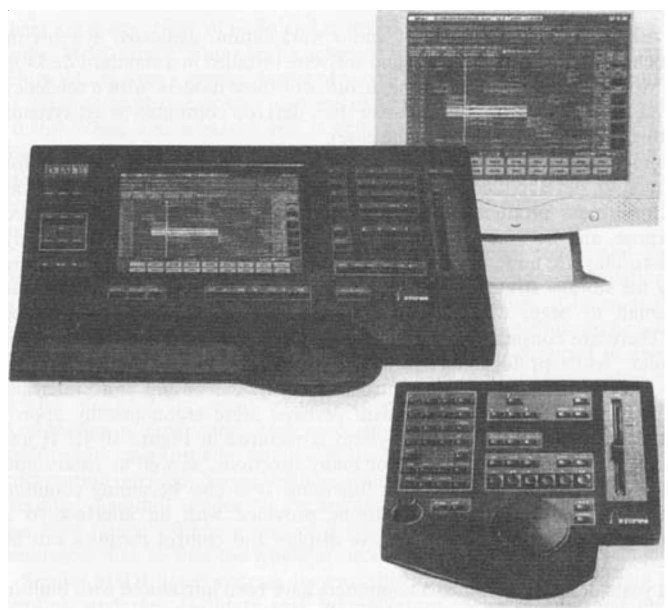
The mixer section can be thought of in conventional terms, and indeed some systems offer physical plug-in interfaces with moving fader automation for those who prefer them. As well as mouse control of such things as fader, pan, solo and mute, processing such as eq, filters, aux send and compression can be selected from an effects 'rack', and each can be dragged across and dropped in above a fader where it will become incorporated into that channel. Third party 'plug-in' software is also available for many systems to enhance the signal processing features. Automation of faders and other processing is also possible. The recorded material itself resides on a (usually) removable hard disk drive, and the edit decision list (the information created during editing which tells the computer how to play the recorded material) resides on the computer's internal disk drive once the project has been 'saved'. When a project is complete, the latter can be loaded onto the removable disk so that the whole project is contained therein.



Similar MIDI-based systems are available for controlling and editing MIDI data streams, and they facilitate easy manipulation of sequencing and other editing processes. MIDI sequencing can also be integrated with the audio-based systems.

There are fundamentally two types of audio workstation: dedicated systems or systems, which use additional hardware and software installed in a standard desktop computer. Most systems conform to one or other of these models, with a tendency for dedicated systems to be more expensive than desktop computer-based systems (although this is by no means always the case).

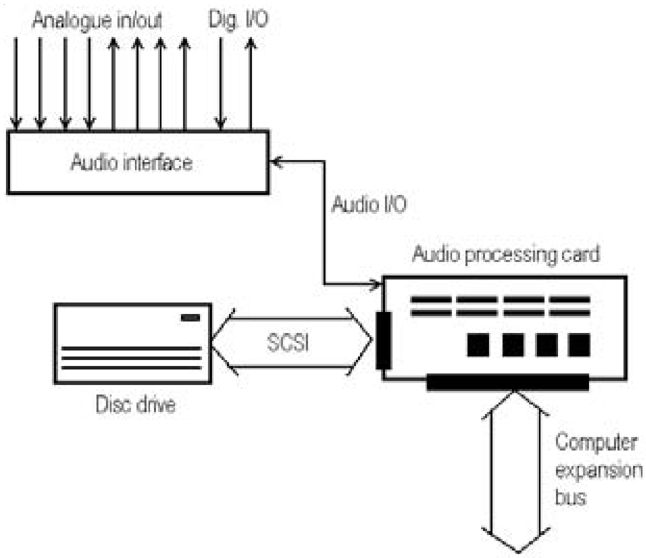
It was most common in the early days of hard disk audio systems for manufacturers to develop dedicated systems with fairly high prices. This was mainly because mass produced desktop computers were insufficiently equipped for the purpose, and because large capacity mass storage media were less widely available than they are now, having a variety of different interfaces and requiring proprietary file storage strategies. It was also because the size of the market was relatively small to begin with, and considerable R&D investment had to be recouped. There are considerable advantages to dedicated systems, and they are very popular with professional facilities. Rather than a mouse and a QWERTY keyboard the user controls the system using an interface designed specificaIly for the purpose, with perhaps more ergonomically appro-priate devices. An example



of such a system is pictured in the Figure below. It has a touch screen and dedicated controls for many functions, as well as rotary and slider controls for continuously variable functions. It is also becoming common for cheaper dedicated editing systems to be provided with an interface to a host computer so that more comprehensive display and control facilities can be provided.

Figure: Dedicated digital audio workstations.

In recent years desktop multimedia computers have been introduced with built-in basic A V (audio-video) facilities, providing limited capabilities for editing and sound manipulation. The quality of the built-in convertors in desktop computers is necessarily limited by price, but they are capable of 16 bit, 44.1 kHz audio operation in many cases. Better audio quality is achieved by using third party hardware.



Many desktop computers lack the processing power to handle digital audio and video directly, but by adding third-party hardware and software it is pos-sible to turn a desktop computer into an A V workstation, capable of storing audio for an almost unlimited number of tracks with digital video alongside. An audio signal-processing card is normally installed in an expansion slot of the computer, as shown in Figure below.

Figure: Typical system layout of a desktop compute- based audio station

The card would be used to handle all sound editing and post-processing operations, using one or more DSP chips, with the host computer acting mainly as a user interface. The audio card would normally be connected to an audio interface, perhaps containing a number of A/D and D/A convertors, digital audio interfaces, probably a SMPTE/EBU timecode interface, and in some cases a MIDI interface. A SCSI interface (the most common high-speed peripheral interface for mass storage media) to one or more disk drives is often provided on the audio expansion card in order to optimise audio file transfer operations, although some basic systems use the computer's own SCSI bus for this purpose.

Consumer Digital Formats

Minidisk

CD has been available for some years now as a 16-bit 44.1 kHz digital playback medium; it was joined by CD-ROM, CD-R (recordable) and CD-RW (record able and rewritable). The mini disc (MD) is now an established consumer recording and playback format, and it is of the M-O (magneto-optical) type. Sampling frequency is fixed at 44.1 kHz, and resolution is nominally l6-bit. A coding system similar to those originally developed for digital audio broadcasting (DAB) known as Adaptive Transform Acoustic Coding (ATRAC) is used whereby the incoming signal is first split into three bands: below 5.5 kHz, 5.5-11 kHz, and above 11 kHz, and each band is individually analyzed with respect to frequency content and level over successive short periods of time via Modified Discrete Cosine Transform (MDCT) filter blocks. Within the three blocks, non-uniform frequency splitting into 20, 16 and 16 further sub-bands takes place, and the circuit then discards material which it deems will be masked by other sounds which are present at higher signal levels and/or have a similar frequency content. A data rate of about one fifth that required for CD is adequate to encode the resulting signal (CD's data stream is 1.4 Mb/s, MD's is 292 Kb/s) and this allows usefully long playing times to be obtained from a disc which is somewhat smaller than a CD at 64 mm in diameter. Since the format involves considerable data compression (a slight misnomer for data reduction), it is not suitable for professional master recording or archiving, but is used quite widely in applications where the highest sound quality is not required such as broadcast journalism.

DVD-A and SACD

CD has now been with us for a couple of decades, and it was inevitable that new formats would be developed which offered larger storage capacities together with higher audio quality and the provision for multichannel surround sound. The two present contenders are DVD-Audio (DVD-A) and the Super Audio Compact Disc (SACD). There are a number of similarities between them, but also some differences, which make them potentially incompatible. The similarities will be looked at first. Physically, both are exactly the same size as the CD, and the greatly increased data storage capacity (up to 4.7 G Bytes as against CD's 650 M Bytes) has been achieved using a combination of reduced track pitch, reduced length of the pits on the surface of the disc, altered channel code and a higher linear reading speed. Loss less data compression has also been specified for both systems. Table below summarizes the physical differences between the new formats and CD.

Parameter	DVD/SACD	CD
Track pitch (microns)	0.74	1.6
Pit length (microns)	0.74	0.834
Speed (m/s)	4	1.2
Maximum data rate (Mbit/s)	11.08	1.536
Laser Wavelength (nm)	635-650	780-790
Data Capacity	4.7 Gbyte one layer 8.5 Gbyte two layer	650 Mbyte (approx.)

Both types of disc can be dual layer, the upper carrying the high-density informa-tion on a semi-Tran missive layer which is read by a short wavelength laser, and on Super Audio CD the lower carrying normal CD information which can be read by a longer wavelength laser which shines through the top layer to come to a focus upon it. (The lower layer can also be another high-density layer.) The latter is provided to maintain backwards compatibility with CD players, but its provision is not a mandatory requirement.

DVD-A is one of a DVD family which also includes DVD-Video, DVD-Rom, DVD-R and DVD-RAM, in contrast to SACD which was conceived from the outset as a super-quality audio-only format, although the latter has an area on the disc reserved for other data which can include, for example, video clips, text and graphics.

DVD-A supports sampling rates of 44.1, 48, 88.2, 96, 176.4 and 192 kHz, split into two groups, which are respectively multiples of 44.1 and 48 kHz. Resolution is16, 20 or 24 bits, and combinations of these can be used to obtain the required playing times for up to six channels of audio. If more than two channels are used these are split into two groups: group 1 contains the front channels, group 2 contains the others, and the latter could be coded to a lower standard in the interests of the conserving data. This is called scalable audio. It is desirable that multichannel mixers give good results on stereo-only replay, and to this end a System Managed Audio Resource Technique (SMART), mandatory in players but optional at the produc-tion stage, is incorporated which makes possible a stereo mixdown of multichannel information in the playback machine which is under content producer control.

A data compression system called Meridian Lossless Packing (MLP) can be incorporated to reduce the data rate of audio signals without apparently rejecting any information. Unlike other data compression schemes, which rely on psycho acoustic masking to reject unessential data, MLP is more like the software programs that reduce the size of computer files without altering information. Repetitive patterns are looked for, and more efficient ways of packing the data are sought to make use of the available space in the most economical way.

DVD-Audio V is a similar disc, which in addition holds video objects (video clips, Dolby AC-3 compressed audio and other information). Three types of player are envisaged: audio, video and universal, and there is a degree of cross-compatibility -between them. The audio-only player will play just the audio part of a DVD-Audio V disc. The video-only player will play just the video part of a DVD-Audio V 6;

SACD uses a different coding system from DVD-A. Instead of multi-bit linear PCM, a 1-bit delta-sigma coding system called Direct Stream Digital is used.

A lossless data packing method called Direct Stream Transfer (DST) achieves roughly 2: 1 data compression to allow room for high-quality multichannel as well as stereo coding to be present on separate areas of the disc's playing surface.

Copy protection measures have been developed for both systems. DVD-A can include a 'watermark' in the bit stream. SACD uses Pit Signal Processing (PSP), which contains an invisible physical watermark, which is read by the player. In addition, a visible watermark can be imprinted on the disc, this involv-ing the injection of a modulation signal during the mastering stage which alters the actual size of the pits to produce an image which should be impossible to replicate.

Digital Audio Interfaces

It is often necessary to interconnect audio equipment so that digital audio data can be transferred between them without converting back to analogue form. This preserves sound quality, and is normally achieved using one of the standard point-to-point digital interlaces described below. These are different from computer data networks in that they are designed purely for the purpose carrying audio data in real time, and cannot be used for general-purpose file transfer applications.

Computer Networks and Digital Audio Interfaces Compared

Real-time digital audio interfaces are the digital audio equivalent of signal cables, down which digital audio signals for one or more channels are carried in real time from one point to another, possibly with some auxiliary information attached. A real-time audio interface uses a data format dedicated specifically to audio pur-poses, unlike a computer data network, which is not really concerned with what is carried in its data packets. A recording transferred over a digital interface to a second machine may be copied 'perfectly' or cloned, and this process takes place in real time, requiring the operator to put the receiving device into record mode such that it simply stores the incoming stream of audio data. The auxiliary infor-mation may or may not be recorded (usually most of it is not).

In contrast, a computer network typically operates asynchro-nously and data is transferred in the form of packets. One would expect a number of devices to be interconnected using a single network, and for an addressing structure to be used such that data might be transferred from a certain source to a certain destination. Bus arbitration is used to determine the existence of network traffic, and to avoid conflicts in bus usage. The 'direction' of data flow is determined simply by the source and destination addresses. The format of data on the network is not neces-sarily audio specific (although protocols optimised for audio transfer may be used), and one network may carry text data, graphics data and E-mail, all in separate packets, for example.

Interface Standards

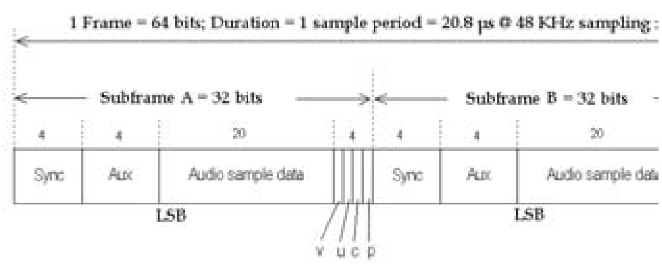
Professional digital audio systems, and some consumer systems, have digital inter-faces conforming to one of the standard protocols and allow for a number of channels of digital audio data to be transferred between devices with no loss of sound quality. Any number of generations of digital copies may be made without affecting the sound quality of the latest generation, provided that errors have been fully corrected. The digital outputs of a recording device are taken from a point in the signal chain after error correction, which results in the copy being error cor-rected. Thus the copy does not suffer from any errors which existed in the master, provided that those errors were correctable.

AES/EBU interlace

The AES/EBU interface allows for two channels of digital audio to be transferred serially over one balanced interface. The interface allows audio to be transferred over distances up to 100 m, but longer distances may be covered using combinations of appropriate cabling, equalisation and termination. Standard XLR-3 connectors are used, often labeled D1 (for digital in) and DO (for digital out).

One frame of data is made up of two sub frames (see the diagram), and each sub frame begins with one of three synchronizing patterns to identify the sample as either A or B channel, or to mark the start of a new channel status block. Additional data is carried within the sub frame in the form of 4 bits of auxiliary data (which may either be used for additional audio resolution or for other purposes such as low-quality speech), a validity bit (V), a user bit (U), a channel status bit (C) and a parity bit (P), making 32 bits per sub frame and 64 bits perframe.

One frame is transmitted in the time period of one audio sample, and thus the data rate varies with the sampling rate.



Channel status bits are aggregated at the receiver to form a 24 byte word every 192 frames, and each bit of this word has a specific function relating to interface operation. Examples of bit usage in this word are the signalling of sampling rate and pre-emphasis, as well as the carrying of a sample address 'timecode' and labelling of source and destination. Bit 1 of the first byte signifies whether the interface is operating according to the professional (set to 1) or consumer (set to 0) specification.

Bi-phase mark coding, the same channel code as used for SMPTE/EBU timecode, is used in order to ensure that the data is self clocking, of limited bandwidth, DC free, and polarity independent. The interface has to accommodate a wide range of cable types, and a nominal 110 ohms characteristic impedance is recommended

There are a number of types of digital interface, some of which are international standards and others of which are manufac-turer-specific. They all carry digital audio with at least 16-bit resolution, and will operate at the standard sampling rates of 44.1 and 48 kHz, as well as at 32 kHz if necessary, with a degree of latitude for varispeed.

Manufacturer-specific interfaces may also be found on equip-ment from other manufacturers if they have been considered necessary for the purposes of signal interchange, especially on devices manufactured prior to the standardization and prolifera-tion of the AES/EBU interface. The interfaces vary as to how many physical interconnections are required, since some require one link per channel plus a synchronization signal, whilst others carryall the audio information plus synchronization information over one cable.

The consumer interface (historically related to SPDIF - the Sony/Philips digital interface) is very similar to the professional AES/EBU interface, but uses unbal-anced electrical interconnec-tion over a coaxial cable having a characteristic impe-dance of 75 ohms. It can be found on many items of semi-professional or consumer digital audio equipment, such as CD players and DA T machines. It usually termi-nates in an RCA phono connector; although some hi-fi equipment makes use of optical fibre interconnects carrying the same data. Format convertors are available for converting consumer format signals to the professional format, and vice versa, and for converting between electrical and optical formats.

Two common manufacturer-specific multichannel interfaces are TDIF, devised by Tascam, and the ADA T 'light pipe', devised by Alesis. TDIF is most commonly found on eight-track tape machines conforming to the DTRS format and on some digital mixers. It is quite a cumbersome interface using 25-way D-type connectors. Owing to its small size and ease of use, the optical ADA T interface is increasingly I used between computer sound cards and other digital audio equipment to carry up to eight channels of audio data.

Notes :

The Main Windows in Nuendo

Like with any other software, you need to understand the various windows. Here we discuss the main windows in Nuendo.

The Project Window

The Project window is the main window in Nuendo. This provides you with a graphic overview of the Project, allowing you to navigate and perform large scale editing. The Project window is divided vertically into Tracks and has a time line going from left to right. Each Project has one Project window.

- Tracks of different types.
- The Project Cursor.
- The area with various settings to the left is called the Track List.
- The area to the right in the Project window is called the Event Display.
- This is where you view and edit Audio and MIDI Events, Automation curves, etc.

The Project Browser

The Project Browser window provides a list-based representation of the Project. This allows you to view and edit all Events on all Tracks by using regular value editing in a list.

The Transport Panel

The Transport Panel features transport controls, much like those found on a conventional tape recorder. It can also be used for locating Marker positions, setting Tempo and Time Signature, etc.

- Marker Locators
- The Tempo and Time Signature display Position Display
- Transport controls
- The Left and Right Locators are used for defining where to start and end recording and what section to Cycle.

The Pool

All Clips, Audio or Video, that belong to a Project are listed in the Pool. There is a separate Pool for every Project. In the Pool you can organize, convert and audition Clips, amongst many other things.

- Audio folder
- Trash folder
- Video folder
- Audio Clips
- Waveform image

The Sample Editor

In the Sample Editor you can view and manipulate audio, by cutting and pasting, removing or drawing audio data. By using the Offline Process History, you can undo changes or revert to the original versions at any point.

- Thumbnail overview.
- Waveform view.
- A selected range.

The MIDI Editor

Editing MIDI data is done in the MIDI Editor. The MIDI Editor window shows the contents of a single MIDI Part. The MIDI notes are represented by "boxes", whose vertical position corresponds to their pitch.

MIDI notes are represented by "boxes", with the vertical position corresponding to the pitch.

This section is called the Controller display. It shows "continuous" MIDI Events (such as Controllers), the velocity values of notes.

The Tempo Track Editor

For each audio and MIDI Track, you can specify whether it should be time based or tempo based. Tempo based Tracks follow a tempo, which can either be fixed through the whole Project or follow the Tempo Track. In the Tempo Track Editor you can draw curves that determine how the tempo will change over time.

- Time Signature Events
- The Tempo Curve

The VST Mixer

The VST Mixer is where you mix your audio channels, that is, adjust the levels (volume), stereo panning, effect sends, EQ, etc.

- The Common Panel contains settings that affect all mixer channels.
- Level fader
- Level meter
- Channel automation controls
- Pan control

Channel Settings

The Channel Settings window is used for adding effects and EQ to individual Audio or Group Channels. Each Channel has its own Channel Settings window.

VST Effects

The VST Effects "rack" is where you select and activate Send Effects. There is a similar window for selecting and activating effects in the master output path.

- Channel Inserts
- Channel Sends
- Equalizer section

VST Outputs and Master Gain

In the VST Outputs window you can set the output level of each Output Bus. The number of Buses depends on your audio hardware. The output level of the Master Bus is controlled with the Master Gain fader in the VST Mixer. The Master Bus may be in stereo or have several channels, depending on the chosen configuration. It is connected to the corresponding number of VST output Buses.

- Stereo Bus Faders
- Bus Active button
- Bus output routing pop-up. This is where each Output Bus is routed to a physical output on the audio hardware.
- The Master Bus and Master Gain fader in a stereo configuration.
- Clicking the "Inserts" button opens the Master Effects window.
- The Master Bus and Master Gain fader in a multi-channel (surround) configuration.
- Clicking the "Ins" button opens the Master Effects window.

Recording and Playing Back Audio

Here is a step-by-step description of how to make a simple audio recording and play it back. The purpose is for you to try out some of the most common recording and playback features.

Creating a New Project

Before you can start recording, you need a working environment - a Project:

1 Pull down the File menu and select "New Project". A dialog appears, listing a number of Project Templates for various purposes.

2 Click on the "Empty" item in the list and click OK. A file dialog appears, allowing you to specify a location for the Project folder. This will contain all files related to the Project.

3 Navigate to the desired location of the Project folder, and type the name of the folder in the "Directory" field. Click Create. The Project folder is created on disk, and an empty Project window appears. At this point, you can make various settings for the Project, such as sample rate, resolution, etc. However, to keep things simple we will use the default settings for now.The next step is to create an Audio Track to record on:

4 Pull down the Project menu and select "Add Track". A submenu appears, listing the various types of Tracks available in Nuendo.

5 Select "Audio". An empty Audio Track appears in the Project window.

Preparing to Record

Before you can start recording, there are some preparations to make:

Selecting Stereo or Mono

You need to decide whether you want the recording to be in stereo or mono. This is done by clicking the Stereo/Mono button in the area to the left of the Audio Track.

Activating and Routing Inputs

Pull down the Devices menu and select "VST Inputs".

The VST Inputs window appears. This lists all audio inputs on your audio hardware, allowing you to turn inputs on or off.

Locate the input pair to which you have connected your audio source, and make sure its

"On" button in the Active column is lit.

If not, click the button to turn the input on.

Close the VST Inputs window, and open the VST Mixer from the Devices menu.

This is Nuendo's mixer window, used for setting levels, etc. The VST Mixer contains an audio channel strip for each Audio Track in the Project window, so currently there will be a single stereo channel strip.

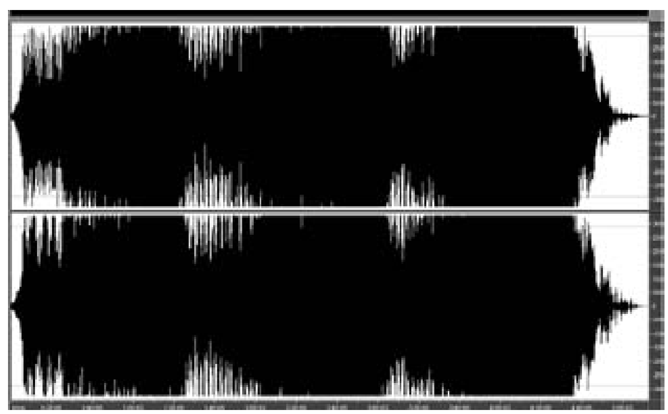
Click on the arrow button at the top of the channel strip, to pull down the Input pop-up menu.

This is where you select which audio input should be routed to the audio channel for recording.

Select the input pair to which you have connected your audio source.

Your audio source is now routed to the audio channel, so that it can be recorded on the Audio
Track. Let the VST Mixer window remain open for now.

Checking the Input Level



To avoid clipping, you need to check the input level before recording:

Lets first understand what **clipping** is.

Clipping occurs when a device is transmitting more signal than it was designed to handle. The distinct sound of audio clipping is created by the waveforms getting "chopped off" before they (the waveforms) reach the peaks of their excursion. Creating, esentially, a square-wave. The higher the amplitude, the closer the waveform resembles a square-wave, and thus, the more noticable the sound becomes. Clipping is found useful in some cases ie: overdriven guitar effects.
Now getting back to our discussion about checking the input level.

Click the "In" button above the level meters (next to the fader on the mixer channel strip).

When the "In" button is lit, the level meter will show the input level (as opposed to the level of the playback signal).

Activate your audio source.

You will see the level meters reacting.

Adjust the output level of your audio source so that the meters go as high as possible without activating the red Clip indicator above the meters.

If the Clip indicator lights up, the input level is too high. Reset the Clip indicator by clicking on it, lower the output level of the audio source and try again.

Making the Track ready for recording

Make sure the Transport Panel is visible.

If not, pull down the Transport menu and select the "Transport Panel" item at the top.

Make sure the buttons on the Transport Panel are set:

If any of these buttons are lit, click on them to deactivate them.

Click in the Ruler (the time scale area above the Track in the Project window), at the position where you want to start recording.

When you click, the Project Cursor (the black vertical line) is automatically moved to the click position. Setting the Project Cursor position in the Ruler.

Click on the "R" button in the area to the left of the Track, so that it starts blinking.

The Track is now Record Enabled.

You are ready to record!

Recording

Start recording by clicking the Record button on the Transport Panel.

The "R" button will light up (stop blinking), and the Project Cursor will start moving.

Play your instrument, etc.

During recording, a rectangle will appear, covering the recorded area. This is the recorded Audio Event.

When you are done, click the Stop button on the Transport Panel.

Recording stops. Nuendo will calculate a waveform image of your recording and display it in the Audio Event.

If you are done with recording, click the "R" button in the area to the left of the Track, so that it goes dark.

Playing back what you just recorded

Move the Project Cursor to the beginning of the recorded Audio Event.

This could either be done by clicking in the Ruler, or by using the Rewind button on the Transport Panel.

Click the Play button on the Transport Panel.

Your recording will be played back.

When you are done, stop playback by clicking the Stop button on the Transport Panel.

Recording more Events

At this point, you may want to continue recording audio, on the same Track or on a new Track.

Recording more on the same Track

To record more audio on the same Track, move the Project Cursor to a new start position and proceed as when you recorded the first time.

It is possible to record audio events that overlap each other, but only the visible events (the events at the top) will be heard when you play back.

Recording a new Audio Track

How to record a new Audio Track, while listening to the first recording.

Create a new Audio Track by using the "Add Track" submenu on the Project menu.

Decide whether you want the new Track to be stereo or mono by using the Stereo/Mono button in the area to the left of the Track.

Pull down the Devices menu and open the VST Mixer.

As you will see, a new channel strip has been added to the Mixer.

Use the Input pop-up menu at the top of the channel strip to make sure that the correct audio input is selected for the new Track.

If the audio source is another than the one you first recorded, you need to check the input level Again.

In the Project window, Record Enable the new Track by clicking its "R" button.

Make sure that the "R" button for the first Track is disabled - otherwise you will be recording on both Tracks at the same time.

Move the Project Cursor to the desired start position.

Activate recording by clicking the Record button on the Transport panel.

While you are recording, the first Audio Track is played back.

When you are done, click the Stop button on the Transport panel.

Playing back in a Cycle

You could continue starting and stopping playback this way, moving the Project Cursor manually each time. However, if you want to try out some mixing features, it is more practical to have Nuendo play back your recorded audio repeatedly, over and over again:

Click on the recorded Audio Event to make sure it is selected.

A selected Audio Event has red and blue handles at its beginning and end.

Pull down the Transport menu and select "Locators to Selection".

This moves the Left and Right Locator (two special Nuendo markers) to the beginning and end of the selected Audio Event, respectively. In the Ruler, the area between the Left and Right. Locator is indicated by a green line.

Click the Cycle button to the left on the Transport Panel so that it lights up.

Move the Project Cursor to the beginning of the recording and click Play.

Playback starts. When the Project Cursor reaches the end of the recording (the Right Locator), it will immediately jump back to the Left Locator and continue playback. Leave playback running for now.

Mixing

This segment contains basic descriptions of the VST Mixer and information on how to use the effects and automation in Nuendo. The purpose is to introduce the basic elements involved when mixing audio.

Before you Start

This segment assumes the following:

That you have opened a Project, containing at least one Audio Track and some Audio

Events that you can play back.

Opening the VST Mixer

To open the VST Mixer, select it from the Devices menu.

The VST Mixer window is similar in appearance to a conventional hardware mixer, with a level fader for each audio channel strip. The VST Mixer will contain the same number of channels as the number of Audio Tracks present in the current Project. Beside each channel's level fader, there is a level meter, which indicates the signal level of audio events on the corresponding Audio Track during playback.

Setting Level and Pan

1. With the VST Mixer window still open, activate playback. Make sure that you have a mixer channel playing back a signal in view.
2. Click on the level fader handle and drag down or up. You will hear the volume of the playback signal being changed. The meters will also reflect the change of level.
3. Click on the blue line in the Pan control box above the fader, and drag to the left or right. You will hear the stereo balance being changed. Adjusting Pan.

Mute and Solo

Each channel strip has a Mute and a Solo button, allowing you to silence one or several audio channels. The following applies:

- The Mute button silences the selected channel. Several Channels can be muted simultaneously. A muted channel is indicated by a lit Mute button.
- The Solo button mutes all other channels, so you only hear the selected channel.
- A soloed channel is indicated by a lit Solo button. Several channels at a time can be soloed. For "exclusive Solo" (only one Soloed channel at a time), press [Ctrl] and click the Solo button.

Adding EQ

For each channel in the VST Mixer, there is a Channel Settings window. This contains a duplicate channel strip, Send- and Insert Effect sections, and an Equalizer (EQ) section. Equalization shapes the tone of a signal by boosting and/or cutting selected frequencies. To add EQ to a Channel, proceed as follows:

1. Click the "EQ" button for the desired channel in the VST Mixer. The Channel Settings window opens.
2. Activate as many EQ modules as you need (up to four) by clicking the "On" buttons.
3. Set the parameters for the activated EQ module(s) by using the knobs.

The EQ curve display in the EQ section will reflect the parameter settings. You can also make settings directly in the display, by dragging curve points.

Adding Effects

Send Effects

When you use send effects, audio is routed through the effect processors via independent

Effect Sends for each channel, just like on a "real" physical mixer.

1. Pull down the Devices menu and select "VST Send Effects". An "effect rack" window appears, with eight slots, all empty.
2. Click in the black field for the effect slot at the top (in the area that says "No Effect"). A pop-up menu appears, listing all available effect plug-ins.
3. Select an effect from the list.

The effect is loaded into the first effect slot. The lit red "Power button" indicates that the effect is activated.

- You can make settings for the effect by clicking the "Edit" button, but for now we will just use the default settings.

4. In the VST Mixer click on the "FX" button at the top of the channel strip. The Channel Settings window appears. This is where you set up the effect sends.
5. In the Sends section, locate the first send (the one at the top of the left send column). Note that an abbreviation of the effect name is shown in the name field below the send level knob.
6. Click the "On" button for the first send, so that it lights up. The send is now activated.
7. Click on the send level knob for the first send and turn it up by dragging the mouse. You will hear the effect being added to the channel being played back.

Insert Effects

An insert effect is inserted into the signal chain of an audio channel, which means that the whole channel signal passes through the effect. To route an audio channel through Insert Effects, proceed as follows.

1. Locate the channel you want to add insert effects to, and click the "Ins" button. The Channel Settings window opens. To the left of the Sends section, you can find four Insert Effect slots.
2. Click in the Effect type field for one of the Insert slots, and select an effect from the pop-up menu that appears.
3. Activate the effect by clicking the "On" button. The Channel is now routed through the Insert effect.

Master Effects

Master effects are added to the signal on the Master bus, the final mix output. They operate like Insert effects, i.e. the whole Master bus signal passes through the effect.

1. Pull down the Devices menu and select "VST Master Effects".
2. An "effect rack" window appears, with eight slots, all empty.

3. Click in the “No Effect” slot and select an effect from the pop-up menu. When you select an effect, an effect panel opens in the slot.
4. Make sure the red “Power button” is lit. The Master bus is now routed through the Master effect.

Automating the VST Mixer

Virtually every VST Mixer and Effect parameter can be automated. Each Channel has an Automation Track, which is by default hidden. When you use Write Automation the automation Events that are generated are recorded on the corresponding channel’s Automation Track. These Events can be viewed and edited on Automation “subtracks”, one for each channel parameter that has been automated. Automation subtrack editing is described in the chapter “Automation” in the Operation Manual.

Using Write/Read Automation

There are separate Write (W) and Read (R) buttons for each channel in the VST Mixer, and for the Master fader.
The Write (W) and Read (R) buttons for a channel in the Mixer, and for an Automation subtrack in the Track List.

- If you activate Write for a channel, all mixer parameters you adjust during playback for that specific channel will be recorded.
- If you activate Read for a channel, all your recorded mixer actions for that channel will be performed during playback, just like you performed them in Write mode. There are also Global Read/Write buttons in the VST Mixer’s Common Panel.
- When Write All is activated, all VST Mixer actions you perform during playback (for all channels) will be recorded as Automation Events.
- When Read All is activated; all your recorded mixer actions for all channels will be performed during playback.

An Example

For a quick step-by-step description of Write/Read automation, proceed as follows:

1. Open the VST Mixer.
2. Activate Write automation for a channel by clicking the “W” (Write) button. The button lights up.
3. Start playback.
4. Move the channel fader up or down.
5. While still in playback, adjust the Pan by clicking the blue line in the Pan control box above the fader and dragging to the left or right.
6. Click Stop on the Transport Panel, and go back to the position where you activated playback.
7. Click on the Write button to deactivate Write mode.
8. Click on the Read button so that it lights up. Read mode is now activated.
9. Start playback.

The Volume fader and Pan control will now mirror the actions performed while in Write mode.

- To redo anything that was recorded, activate Write mode again, and start playback from the same position.
- You may have Write and Read activated simultaneously, if you want to watch and listen to your recorded mixer actions while you’re recording fader movements for another mixer channel, etc.

Editing in the Project Window

This segment describes some of the procedures for editing in the Project window. Typically, this is where you do the “large-scale” editing and rearranging of Events.

- Although the following examples describe editing Audio Events, many of the techniques can also be applied to Audio Parts and MIDI Parts.

Selecting Events

To select Events in the Project window, proceed as follows:

1. Make sure the Arrow tool is selected. If not, click on the Arrow icon in the Toolbar.
2. To select a single Event, click on it. To select several Events, use [Shift]-clicking or click and drag a selection rectangle.

Selected Events are displayed with red and blue handles at the edges. These are used for resizing, changing the gain and creating fades, respectively.

Moving and Copying Events

To move Events in the Project window, proceed as follows:

1. If you want to move more than one Event, select them as described above. If you want to move a single Event, you don’t need to select it.
2. Click on one of the Events with the arrow tool and drag it to the desired position. If Snap is activated on the Toolbar, this determines the exact position of the moved Events.

When you are moving several Events, they will keep their relative positions.

- To make copies of Events, hold down [Alt] and proceed as when moving.

There are also special Duplicate and Repeat functions, as described in the Operation Manual.

Resizing an Event

You can resize Events by dragging their start or end edges, thereby “hiding” material at the start or end:

1. Make sure the Arrow tool is selected.
2. Click on the Arrow tool icon so that a small pop-up menu appears.
3. Make sure “Normal Sizing” is selected.
4. Select the Event you want to resize. The red resize handles appear at the lower edges of the Event.
5. Click on a resize handle and drag it to the left or right. The Event is resized. If Snap is activated on the Toolbar, this determines the resulting length of the resized Event

About the Snap Setting

When you are rearranging and editing Events in the Project window, it is often useful to have a “guide”, helping you to find the correct positions. This is the purpose of the Snap setting on the Toolbar. Let’s say you are moving Events, and want them lined up so that one Event should start exactly where the previous Event ends:

1. Pull down the Snap pop-up menu on the Toolbar. This determines which positions should be used for snapping.
2. Select the “Events” option. In this mode, Events will snap to the end and start positions of other Events.
3. Click the Snap on/off button to activate Snap. Snap activated on the Toolbar.
4. Try moving an Event. You will find that the start of the moved Event is “magnetic” to the start and end of other Events.

Creating a Fade-in

There are several ways to create fades. This example shows how to create a fade-in using the fade handles in the Project Window:

1. Select the Event. The blue fade handles appear at the upper edges of the Event.
2. Click on the fade handle at the beginning of the Event, and drag it to the right. This creates a fade-in of the default shape (initially a linear fade).
3. If you now play back the Event, you will hear the volume gradually rising in the fade-in area. If you don’t want the fade to have the default shape, you can adjust this in the Fade dialog. Let’s say you want a smooth, more exponential fade-in:
4. With the Event selected, pull down the Audio menu and select “Open Fade Editor(s)”. The Fade In dialog appears. You can also open this by double clicking on the actual fade curve. The current fade shape is shown on top of the audio waveform in the graphic display.
5. Click the left “Curve Kind” button. With this selected, the fade curve will consist of spline-curve segments, rather than linear segments.
6. Click on one of the Curve Shape buttons below the display. These are preset curve shapes, but you can also modify the shape manually by clicking and dragging in the display.
7. Click “OK”. The dialog is closed and the new curve shape is applied to the fade-in.

Creating a Crossfade

Audio Events on the same Track can be crossfaded, to create a smooth transition. In this example, we have two Audio Events positioned so that they overlap slightly, and the crossfade will be created in the overlapping area:

1. If necessary, resize the Events so that the overlapping section is suitably long. You may want to zoom in to see what you’re doing - use the horizontal zoom slider in the lower right corner of the Project window.
2. Select one of the Events (or both).
3. Pull down the Audio menu and select “Crossfade”. A crossfade is created in the overlapping area, with the default shape (initially linear).
4. To modify the crossfade, make sure one or both Events are selected, and select “Crossfade” from the Audio menu again. The Crossfade dialog appears. This is similar to the Fade dialog, but has separate sections for the fade-in and fade-out parts of the crossfade.
5. Modify the crossfade shape by using the Curve Shape buttons or by clicking and dragging in the curve displays. You can use the “Play Fade” buttons to listen to the fade-in and fade-out separately, and the “Play” button to listen to the crossfade with the current settings.
6. Click “OK” to apply the settings and to close the dialog. Clicking “Apply” applies the changes without closing the dialog. The crossfade is adjusted.

Adjusting the Volume of an Event

If you want to adjust the volume of an Audio Event, you can do this by dragging the volume handle (the blue handle in the middle of the Event’s upper edge).

- You can also adjust the volume of a selected Event on the Info line (above the Event display, shown by clicking the “i” button on the Toolbar). This allows you to boost the volume by up to 6 dB if you like.

Undoing the Editing

You can always undo any action you have performed by selecting “Undo” from the Edit menu. The number of Undo levels can be specified in the Preferences, and you can also view and edit the entire Undo history in a separate dialog.

Editing Audio

About this Chapter

This chapter describes the basic procedures for editing audio in the Sample Editor and how to use the Process functions.

The Sample Editor

The Sample Editor allows you to edit audio, by cutting and pasting, removing, drawing or processing audio data. This editing can be called “non-destructive”, in the sense that you can undo changes or revert to the original versions at any point, using the Offline Process History.

What is Shown in the Sample Editor?

An audio event plays a section of an Audio Clip. When you open the Sample Editor for an audio event, it will display the waveform image of the corresponding Audio Clip. Above the Waveform display is the Thumbnail display, which provides an overview of the whole Clip, with a blue rectangle corresponding to the section currently shown in the Waveform display. You can navigate by dragging or resizing the blue rectangle in the Thumbnail display.

Editing Audio in the Sample Editor - An Example

This example describes how to remove a section of audio and insert it at another position, by using cut and paste in the Sample Editor:

1. Open the Sample Editor by double clicking an audio event in the Project window.
2. Select the Range Selection tool by clicking its icon on the Toolbar.
3. Select a section of the Clip by clicking and dragging in the Waveform display. The cursor changes to an arrow as you start dragging. Click at the position you wish to start the selection and drag to make a selection range.

4. Release the mouse when the selection is complete. You can adjust the selection by dragging its edges.
5. Select "Cut" from the Edit menu. The selection is removed from the Clip and moved to the Clipboard. The section to the right of the selection is moved to the left to fill out the gap. Selecting "Paste" from the Edit menu will copy the data on the Clipboard into the Clip according to the following rules:

- If there is a selection in the editor, the pasted data will replace it.
- If there is no selection (if the selection length is "0"), the pasted data will be inserted starting at the selection line.

The selection line can be placed at any position in the Event by clicking with the mouse. The section to the right of the line will be moved to make room for the pasted material.

6. For this example, make the end of the event visible, either by using the scrollbar or by moving the blue rectangle in the Thumbnail, and click to place the selection line at the event end position.
7. Now select "Paste" from the Edit menu. The selection that was cut from the event is inserted at the position of the selection line.

Processing Audio

The Process submenu on the Audio menu contains a number of audio processing functions. The functions can be applied to selected audio events or Clips, or to a selected range. In this example, we will apply Normalizing to a selected audio event. The Normalize function allows you to specify the desired maximum level of the audio. A common use for Normalizing is to raise the level of audio that was recorded at too low an input level.

Proceed as follows:

1. Select an audio event by clicking on it with the Arrow tool in the Project window.
2. Pull down the Audio menu and select Process. A submenu opens, containing all processing functions available.
3. Select "Normalize" from the submenu. The Normalize dialog opens.
4. For this example, use the default "Maximum" setting of 0.00 dB.

- You can use the "Preview" button to listen to the result of the processing if you wish.

The processing will not be applied, just auditioned.

5. Click "Process" to apply the processing.

The audio event is normalized.

Applying an Effect Plug-in

You can add "real-time" effects in the VST Mixer. However, sometimes it is useful to apply an effect directly to an audio event or Clip. This is done using the Plug-ins submenu on the Audio menu:

1. Select an audio event in the Project window. In this example, we select the event that was Normalized in the previous example.
2. Pull down the Audio menu and select Plugins.A submenu appears, listing all installed effect plug-ins.
3. Select the "JetFlange" effect. The JetFlange dialog appears.
4. Use the parameters in the upper part of the dialog to set up the effect as desired. Clicking the Preview button will let you audition the settings before actually applying the effect.
5. Click the "Process" button to apply the effect.

Using the Offline Process History Dialog

The Offline Process History dialog allows you to remove or modify audio processing at any time. It is even possible to modify or remove some processing "in the middle" of the Process History, while keeping later processing! Whether this is possible or not depends on the type of processing performed, as described in the chapter "Audio Processing and Functions" in the Operations Manual. In this example, we will remove the Normalize function, but keep the applied Jet Flange effect:

1. Select the audio event in the Project window.
2. Pull down the Audio menu and select "Offline Process History". The Offline Process History dialog appears. This contains a list of the processing functions applied to the Audio Clip, with the most recent operation at the bottom of the list.
3. Select the "Normalize" action by clicking in the list.
4. Click the "Remove" button. You are asked whether you really want to remove the processing.
5. Click "Remove". The Normalize processing is removed, but the JetFlange effect is kept.

Watching a Video

This segment describes the basic procedures for playing back a video file in Nuendo.

Select a Video Playback Engine

Nuendo can play back video using one of three different playback engines: Direct-Show, Quicktime or Video for Windows. This ensures compatibility with as wide a range of video hardware as possible. In this example, we use direct Show:

1. Pull down the Device menu and select "Device Setup". The Device Setup dialog appears.
2. In the list of devices to the left click on the "Video Player" device. The Video Player settings are displayed under the "Setup" tab to the right in the dialog.
3. Make sure "DirectShow Video" is selected on the Playback Method pop-up menu.
4. Click OK to close the dialog. Create a Video Track. To play back a video file in Nuendo, you first have to create a Video Track:

- From the Project menu, select "Add Track", and then select "Video" from the submenu that appears.

The Video Track is added to the Track list. You can only have one Video Track in each Project.

Importing a File

When the DirectShow playback engine is selected, Nuendo can import and play back video films in the formats AVI and MPEG. To import a video file, proceed as follows:

1. Select the Video Track, by clicking it in the Track list.
2. Pull down the File menu and select "Import", then select "Videofile…" from the submenu that appears. This opens a file dialog, allowing you to locate the file you wish to import.

- In the lower left side of the Import Video dialog, there is a tick box named "Extract Audio".

If ticked, this will add any audio contained in the video file to a new, automatically created audio track.
The new audio event will start at the same time as the video event, so that they are in sync with each other.

3. Locate and select the file, and click Open. When you import a file this way, a Clip is created for the video file and an Event that plays the whole Clip is added to the Video Track, at the position of the Project Cursor.

Playing it back

Playback is done together with all other material, using the Transport panel. Video files are displayed as Events/Clips on the video track, with thumbnails representing the frames in the film.

- Pull down the Devices menu and select "Video".

A video window appears. In stop mode, this displays the video frame at the Project Cursor position.

Basic Nuendo Concepts

This segment describes the basic "building blocks" and terminology in Nuendo. Please make sure you grasp all of this before you proceed further.

The Project

The native document format of Nuendo is called a Project. Before you can start recording, playing back or editing you always have to create a new Project, or open a saved Project file from disk. There can be several Projects open at the same time, but one is always the active Project.

The Project to the left is the active Project, as indicated by the lit red light indicator in the upper left corner of the window.

About the File and Folder Structure

A Project file has the extension ".npr" and is always associated with a Project folder on your hard disk. Several Projects can share the same Project folder (which is practical if you have several versions of your Project, for example).
Typically, a Project folder is structured like this:

- The Audio folder contains audio files referenced by the Project. It is also possible for the Project to refer to audio files elsewhere on your disk(s), but having all audio files in the Project's Audio folder makes the Project easy to move and archive, and is a good safety measure.
- The Edits folder contains audio files created automatically by editing and processing operations in Nuendo. As a rule, you shouldn't touch the files in this folder. To remove unused edit files, it is better to use the Cleanup function.
- The Fades folder contains audio files created by fade and crossfade operations in Nuendo.
- The Images folder contains waveform images for the audio files in the Project.
- The Project file itself contains all references to audio and video files, along with playback information, MIDI data and settings for the Project (such as sample rate, frame rate, etc).
- Video files are never automatically copied to the Project folder. This is because video files are often very large, and it doesn't make sense to copy them into different Project folders. However, nothing stops you from creating a Video folder inside the Project folder and storing your video files there.
- You may also find additional files in the Project folder.
- For example, Nuendo's Auto Save feature stores backup copies of the Project file in its Project folder.

Audio Terminology

When you record Audio in Nuendo, this is what happens:
An Audio File is created on the hard disk.

- In Nuendo, an Audio Clip is created. The Audio Clip refers to the audio file on disk.
- An Audio Event is also created in Nuendo. This plays back the Audio Clip. There are good reasons for this long chain of references:
- The Audio Event is the object that you place on a time position in Nuendo. If you make copies of an Audio Event and move them to different positions in the Project, they will still all refer to the same Audio Clip.

Furthermore, each Audio Event has an Offset value and a Length value. These determine at which positions in the Clip the Event will start and end, i.e. which section of the Audio Clip will be played back by the Audio Event. For example, if you resize the Audio Event, you will just change its start and/or end position in the Audio Clip - the Clip itself will not be affected.

- The Audio Clip does not necessarily refer to just one original recorded file! For example, if you apply some processing to a section of an Audio Clip, this will actually create a new audio file that contains only the section in question. The processing will then be applied to the new audio file only, leaving the original audio file unchanged. Finally, the Audio Clip is automatically adjusted, so that it refers both to the original file and to the new, processed file. During playback, the program will switch between the original file and the processed file at the correct positions. You will hear this as a single recording, with processing applied to one section only. This feature makes it possible to undo processing at a later stage, and to apply different processing to different Audio Clips that refer to the same original file.

Audio Tracks, Parts and Channels

For an Audio Event to be played back in Nuendo, it has to be placed on an Audio Track.

This is similar to a track on a multi-track tape recorder, and allows you to view the Event and move it along the timeline. You can place any number of Audio Events on an Audio Track, but only one at a time can be played back. You can have up to 500 Audio Tracks, although the number of Tracks you can play back at the same time depends on your computer performance.

Even though Audio Events can be placed directly on Audio Tracks, sometimes it is convenient to gather several Audio Events into an Audio Part. This is simply a “container”, allowing you to move and duplicate several Audio Events as one.

Each Audio Track has a corresponding Audio Channel in the VST Mixer. This is much like a channel on a hardware mixer, allowing you to set levels and panning, add EQ and effects, etc.

MIDI Terminology

When you are recording MIDI (or entering MIDI data manually in an editor), MIDI Events are created. For example, each note you record is a separate MIDI Event, and if you record the movement of a modulation wheel or other controller, a large number of densely spaced Events are created. MIDI Events are always placed in MIDI Parts. These are “containers”, allowing you to move or copy a number of MIDI Events (e.g. a recorded MIDI melody line) as one item. MIDI Parts are placed on MIDI Tracks. For each MIDI Track you can specify on which MIDI Output and MIDI Channel its MIDI Events should be played back. This allows you to have different Tracks play back different sounds, in the same or different MIDIinstruments.
A MIDI Part on a MIDI Track. The black lines in the Part indicate MIDI Events.

Video Terminology

- When you import a video file from disk into a Nuendo Project, a Video Clip is created that refers to the file.
- A Video Event is then created, referring to the Video Clip. Video Events can be moved, copied and resized without affecting their Video Clips.
- For a Video Event to be played back, it has to be placed on the Video Track. There can only be one Video Track in a Nuendo Project.

Basic Methods

This segment contains descriptions of the general methods and procedures used in Nuendo. As this information applies to all parts of the program and all ways of working, please take time to read this chapter before continuing with the Operation Manual.

Using Menus

Main Menus

The menus in the main Nuendo menu bar are always available, regardless of which window is active. However, menu items that are not relevant in the current window may be greyed out. You select items from the main menus following the standard procedure of the operating system.

Pop-up Menus

Pop-up menus can be found throughout the program and are often used for selecting options or values. A pop-up menu is indicated by an arrow button next to a field showing the currently selected option/value.

- To bring up the pop-up menu, click the arrow button. Selecting is done as with regular menus. Selecting from the Snap pop-up menu.

The Quick Menu

In Nuendo, clicking the right mouse button will bring up a contextual pop-up menu. Some areas have special context menus with functions or settings that only apply to the corresponding area (for example, right-clicking in a Ruler brings up a pop-up menu with display format options).
However, right-clicking in the main area of a window brings up the Quick menu. As a rule, the Quick menu contains:

- The tools (provided that the window has tools).
- The most relevant menu items from the main Nuendo menus.
- Settings that are specific for the window.

For example, in the Sample Editor the Quick menu contains settings for which Elements should be displayed in the Waveform display.

Using Tools

Editing in Nuendo is largely done with the various tools. Typical examples are selecting and moving Events with the Arrow (Object Selection) tool, drawing with the Pencil tool, deleting with the Eraser tool, etc. There are different tools for different windows. Tools can be selected in three ways:

- By clicking the corresponding Tool icon on the Toolbar. When you click a Tool icon, the pointer takes on the shape of the corresponding tool.
- By using the Quick menu. Clicking with the right mouse button in the main area of a window brings up the Quick menu. The tools will be listed (along with their corresponding icons) at the top of the menu - to select a tool, simply select it from the menu.
- By using key commands. By default, the keys [1] - [0] on the alphanumeric part of the keyboard are used, so that pressing [1] selects the leftmost tool and so on. You can also use key commands to step between the tools on the Toolbar. By default, pressing [F9] selects the previous tool and pressing [F10] selects the next tool.

About Tool Tips

If you position the pointer over a Tool icon (or any other icon or button in Nuendo), a label will appear after a moment, informing you of the function of the icon or button.

- This feature can be turned off by deactivating the option “Show Tips” on the User Interface page in the Preferences dialog (accessed from the File menu).

Changing Values

Throughout the program you will encounter various value fields that can be edited. These can be divided into three categories: position values, regular numeric values and names.
Editing Position Values

Depending on the selected display format, position values in Nuendo are usually divided into several “segments” (the exception being the “Samples” display format, in which values are edited as regular numeric values). Two examples: If the “Seconds” display format is selected, positions are shown as hours:minutes:seconds.milliseconds. If the “Bars+Beats” display format is selected, positions are shown as “bars.beats.1/

16th notes.ticks” (with 120 ticks per 1/16th note). Each value segment can be edited separately, in one of the following ways:

- Point at the upper or lower edge of the segment and click. Clicking at the upper edge will raise the value of the segment one step, clicking at the lower edge will lower the value.
- Point and click directly on the value segment, type a new value and press [Return].
- If you are using a wheel mouse, point at the value segment and use the wheel to raise or lower its value. You can also edit the whole value (all segments) by double clicking and typing in a new value.
- To separate the value segments, you can use spaces, dots, colons or any other character that isn’t a number.
- If the “Bars+Beats” display format is selected, and you enter a value with less than four segments, the largest position value segments will be affected and the program will set the lesser segments to their lowest values. For example, if you enter “5.3”, the position will be set to “5.3.1.0”.
- If one of the frame based display formats (all formats called “fps” or “dfps”) is selected, and you enter a value with less than four segments, the smallest position value segments will be affected and the program will set the larger segments to their lowest values. For example, if you enter “2:5”, the position will be set to “0:0:2:5”.
- If the “Seconds” display format is selected, value editing works as with the frame based formats, with one addition: The smallest value segment (milliseconds) is considered to be the decimal part of the seconds segment. This means that if you enter “2:50”, the position will be set to “0:0:2:500”, rather than “0:0:2:050”.

Editing Regular Numeric Values

Numeric values other than positions are edited by clicking the value and editing numerically from the computer keyboard.

- For some values, you can hold down [Alt], click on the value and keep the mouse button pressed to display a value slider. This allows you to scroll the value by dragging up or down with the mouse button pressed. When you release the mouse button, the value slider is hidden. Adjusting the Event Volume setting on the Info line.
- These values can also be edited using a wheel mouse: point at the value and use the wheel to raise or lower it.

Editing Names

To edit a name, click on it, type a new name and press [Return] (or click outside the name field).

Using Knobs and Faders

In the VST audio windows, most parameters are shown as knobs, sliders and buttons, emulating real-world hardware interfaces. For knobs and sliders, you can select the desired way of making adjustments in the Preferences dialog:

Option Description

Circular: To move a knob, you click on it and drag in a circular motion, much like turning a “real” knob. When you click anywhere along the knob’s edge, the setting is immediately changed.

Relative Circular Works like the “Circular” option, but clicking does not automatically change the setting. This means you can make adjustments to the current setting by clicking anywhere on a knob and dragging, without having to click on the exact current position. Linear To move a knob, you click on it and drag up or down (or left or right) with the mouse button pressed - as if the knob were a vertical (or horizontal) slider. **Jump:** In this mode, clicking anywhere on a slider will make the slider handle instantly move to that position. Touch In this mode, you have to click on the actual slider handle to adjust the parameter. This reduces the risk of accidentally moving sliders. Ramp In this mode, clicking anywhere on a slider (but not on the actual handle) and keeping the mouse button pressed, will cause the handle to move smoothly to the new position.

Selecting Objects

- Selecting Nuendo objects such as Audio and MIDI Events is generally done with the Arrow tool, according to standard selection procedures.
- Clicking on an object selects it (and deselects any previously selected objects).
- Holding down [Shift] and clicking on an object selects it without deselecting any other objects.
- You can also create a selection rectangle by clicking in an empty area and dragging with the mouse button pressed. All objects partially or totally enclosed by the rectangle will be selected.
- If an object is already selected, you can use the left and right arrow key on the computer keyboard to select the previous or next object, respectively.

Holding down [Shift] and using the arrow keys allows you to select the previous/next object without deselecting the current object. There are several additional ways to make selections in the different Nuendo windows.

Zoom and View Techniques

Scrolling the View

If the active window isn’t large enough to show all its contents, you can scroll the view by using the standard window scroll bars. However, if you are using a wheel mouse, there are two additional ways to scroll:

- Rolling the wheel will scroll the view horizontally. If you hold down [Shift] and use the wheel, the view will be scrolled vertically. Just make sure not to point at a value field, as this will edit the value instead.
- If you aim in the main area of a window, click the wheel and keep it pressed, the pointer takes on the shape of a hand. You can now scroll the view freely by dragging the mouse horizontally and/or vertically.

Zooming

All windows that contain graphical displays can be zoomed horizontally and vertically. While some windows have special zoom functions, a few methods are commonly available:

Using the Zoom Sliders

At the lower right corner of all zoomable displays, you will find two zoom sliders.

- To zoom in horizontally, drag the horizontal zoom slider handle to the right.
- To zoom in vertically, drag the vertical zoom slider upwards. There is one exception to this: in the Project window, dragging the vertical zoom slider upwards will decrease the height of Tracks (in effect, zooming out).
- Clicking on a zoom slider will move the handle to the click position, instantly changing the magnification. If the Project Cursor is visible when you zoom in or out horizontally, the magnification will be "centered on the Cursor". In other words: if possible, the Project Cursor will remain in the same position on screen.

Using the Magnifying Glass Tool

You can use the Magnifying Glass tool to zoom in and out horizontally, using the following methods:

- Click once to zoom in one step. Zooming will be centered on the click position.
- Double click (or press [Alt] and click) to zoom out one step.
- Draw a zoom rectangle by pressing the mouse button, dragging the pointer and releasing the mouse button. The view will zoom in horizontally, so that only the area enclosed in the zoom rectangle is visible.

Using the Zoom Menu

At the bottom of the Edit menu, you will find a Zoom submenu with various zoom functions. Exactly which items on the submenu are available depends on the currently active window.

- The Zoom submenu is also available as a separate menu item on the Quick menu.
- As with any menu item, you can specify key commands for the functions on the Zoom submenu, for quick access. Key commands are set up in the Key Commands dialog on the File menu.

Zooming in the Ruler

If the option "Zoom while Locating in Time Scale" is activated in the Preferences dialog (Transport page), you can use the Rulers for zooming. This allows you to quickly zoom in or out on a certain position, without having to select a special tool:

1. Click in the Ruler and keep the mouse button pressed. The Project Cursor is automatically moved to the click position. If you don't want to move the Cursor, press [Shift] and click in the Ruler instead.
2. Drag down to zoom in (horizontally) or drag up to zoom out. Zooming will be centered on the Project Cursor.

Window Handling

Generally, Nuendo windows are handled according to the standard procedures. However, the Window menu contains some functions that make-work quicker and easier:
Note that there will be a number after the Window menu in the menu bar. This indicates the currently selected Window Layout.

Menu item Description

Close Closes the currently active window. If this is a Project window, you will close the current Project.
Close All Closes all windows, including all open Projects.
Minimize All Minimizes all windows.
Restore All Restores all minimized Nuendo windows.

Tile Horizontally/Tile Vertically

Arranges the open windows next to each other on screen.
Window Layouts A configuration of windows for the active project is called a "Window Layout". By storing different window combinations as Window Layouts, you can quickly switch between different working modes.

Windows... This opens a dialog where you can manage and make settings for all open windows.

Cascade Arranges the open windows in a partially overlapping pattern.

The open windows list Selecting a window from the list at the bottom of the menu brings it to front.

The open windows list Selecting a window from the list at the bottom of the menu brings it to front

The Windows dialog

By selecting "Windows..." from the Window menu, you open the Windows dialog. This allows you to manage the open windows in various ways. The display to the left lists all open windows, hierarchically arranged (so that editors and other windows that belong to a certain Project are listed under the corresponding Project window). To the right are various window functions. To use one of the functions, proceed as follows:

1. Click in the field below the OK button to select one of the selection modes:
2. If you selected the "Selected" or "Cascaded" modes, select the desired windows by clicking in the list. As usual, you can select multiple items by holding [Shift] or [Ctrl] and clicking.

Mode Description

Selected Only the windows selected in the list will be affected.
Cascaded The selected windows will be affected, along with all their "under-windows". Typically, if a Project window is selected in the list, all open windows belonging to that Project will be affected. All windows will be affected, regardless of the selection.

3. Use the buttons to the right to activate (bring to front), minimize, restore or close the specified window(s). Closing a window will also remove it from the list.
4. When you are done, click OK to close the dialog.

The Device Panel

If you like, you can manage VST Audio windows and the Video window from a central

Device panel:

1. Pull down the Devices menu and select "Show Panels". The Devices Panel appears.
2. To display a closed or hidden window, click on its button in the Devices Panel.
3. Clicking the button again will close the window.

Working with Window Layouts

A configuration of windows for the active Project is called a "Window Layout". By storing different window combinations as Window Layouts, you can quickly switch between different working modes. You may for example want as large as possible a Project window when you are editing, whereas you may want the Mixer and Effect windows open during mixdown. Window Layouts are listed and managed on the Window Layouts submenu on the Windows menu.

Basic Rules

There will always be at least one Window Layout in a Project.

- One Window Layout is always Active (selected). The Active Layout is indicated by a tick mark on the Windows Layout submenu, and by the number after the Window menu on the menu bar (if the menu is titled "Window1", this means that Window Layout 1 is Active, etc).
- If you haven't specifically created or edited Window Layouts, there will be a single item on the Window Layouts submenu, called "Layout 1". This contains the current configuration of windows in the Project, and will be continuously adjusted when you open, close or move windows (since the Layout isn't locked - see next point).
- Window Layouts can be locked or unlocked. If the Active Layout is unlocked, any changes you make to the window configuration (e.g. by opening or moving windows) are automatically stored in the Layout. If the Active Layout is locked, you can still change the window configuration as usual, but the changes are not stored. Activating the same Layout again will bring back the original (stored) Window configuration.
- For each Layout, you can specify which window properties (zoom, window position and/ or Track height) should be affected when you make the Layout active.
- It is also possible to create Window Layouts which are global to all open Projects.

Editing the Active Window Layout

To make changes to the Active Window Layout, proceed as follows:

1. Make sure the item "Lock Active Layout" isn't ticked on the Window Layouts submenu.
2. Make the desired changes to the window configuration. This may include opening, closing, moving and sizing windows, and adjusting zoom and track height. The changes are automatically stored for the Active Layout.
3. If you like, lock the Layout by selecting "Lock Active Layout" from the Window Layouts submenu (or by using a key command, by default [Alt]-[0] on the numeric keypad.) This makes sure that you don't accidentally change the Window Layout.

Creating a New Window Layout

Before creating a new Window Layout, you may want to make sure the current Layout is locked ("Lock Active Layout" is ticked on the Window Layouts submenu). Otherwise, the changes you make in step 1 below will affect the current Window Layout as well.

1. Set up the windows you want to include in the Window Layout. This may include opening, moving and sizing windows, and adjusting zoom and track height.
2. Pull down the Window menu and open the Window Layouts submenu.
3. Select "New...".
4. In the dialog that appears, enter a name for the Window Layout.
5. Use the checkboxes to specify which settings should be included in the Window Layout. Actually, all settings are included, but only the specified properties will be applied when you activate the Layout. You can change this setting at any time in the Organize dialog (see below).
6. Click OK.

The Window Layout is stored and will appear on the Window Layouts submenu. It will now be the Active Layout.

- By default, new Window Layouts are not locked.

Activating a Window Layout

1. Pull down the Window menu and open the Window Layouts submenu.
2. Select the Window Layout from the list on the submenu. The windows are closed, opened, moved and/or resized according to the stored Window Layout.

- You can also activate any of the first nine Window Layouts using key commands. By default, this is done by pressing [Alt] and the corresponding key on the numeric keypad ([Alt]-[1] selects Layout 1, and so on).

Organizing Window Layouts

If you select "Organize..." from the Window Layouts submenu, a dialog opens, listing all available Window Layouts.

- To rename a Window Layout, double click its name in the list and type in a new name.
- To adjust the properties of a Window Layout, use the checkboxes.
- To create a new Window Layout based on the current window configuration, click the New button. The new Layout appears in the list allowing you to adjust its name and properties.
- To activate a Layout, either select it and click the Activate button, or double click in the number column to the left. The Layout is activated and the dialog is closed (unless the "Keep window open" checkbox is ticked).
- To remove a Window Layout, select it in the list and click the Remove button. The Layout is removed from the list.
- To close the dialog, click the OK button. Note that you can continue working in other windows with the Organize Layouts dialog open.

Global Window Layouts

While the regular Window Layouts belong to a specific Project, you can also create Global Window Layouts, that are available for all open Projects. These are useful for defining positions for windows that are not Project specific (such as the VST windows).

- To create Global Window Layouts, you need to close all Projects. When no Projects are open, you can use the regular Window Layout functions for creating Global Window

Layouts. When you later open Projects, the Global Window Layouts will be listed at the bottom of the Window Layout submenu.

- When you select a Global Window Layout from the submenu, the stored windows will be added to the current window configuration.

Note that the new window configuration will be stored in the current (Active) Project Window Layout, provided that it isn't locked.

Undo

There are two ways to undo operations in Nuendo: by using "regular Undo" and by using the Offline Process History.

• The regular Undo can be applied to virtually all actions you perform, in multiple levels.

It is accessed by using the "Undo", "Redo" and "History" items on the Edit menu, as described below.

• The Offline Process History allows you to remove and modify applied processing, and is different from the "regular" Undo in that you don't have to undo processing functions in the order they were performed.

The Offline Process History is described in the chapter "Audio Processing and Functions" in the Operation Manual.

The Undo and Redo Commands

Nuendo offers wide-ranging, multiple Undo, allowing you to undo virtually any action you perform.

- To undo the last performed action, select Undo from the Edit menu, or use the corresponding key command (by default [Ctrl]-[Z]). If you select Undo again, the previously performed action will be undone, and so on.
- To redo the last undone action, select Redo from the Edit menu or use the corresponding key command (by default [Ctrl]-[Shift]-[Z]).

Undone actions will be available for Redo until you perform another action (at which point the "Redo Stack" is cleared - see below).

You can specify how many levels of Undo should be available with the "Maximum Undo"setting on the User Interface page in the Preferences dialog on the File menu.

The Edit History window

Selecting "History" from the Edit menu opens the Edit History window. This contains a graphic representation of the "Undo Stack" (the performed actions, with the most recent action at the top of the Stack) and the "Redo Stack" (the undone actions, with the most recently undone action at the bottom of the Stack). The two stacks are separated by a divider line.

- The Redo Stack.
- The Undo Stack.
- The most recently undone action.
- The most recently performed action.

The Edit History dialog allows you to undo or redo several actions in one go, by moving the divider between the Undo Stack and the Redo Stack (in essence, moving actions from the Undo Stack to the Redo Stack, or vice versa):

1. Click on the divider line and drag it up or down. Drag up to Redo actions, drag down to Undo them. In this case, the Delete and Split actions will be Undone (and moved to the Redo stack).

- You can also click directly between two items in the list, instantly moving the divider line.
- When you move the divider line by dragging or clicking, the actions are immediately Undone or Redone. The changes are reflected in all open Nuendo windows.

2. When you're finished, click OK to close the dialog. The Undo and Redo Stacks are cleared when you close a Project.

By now you should be comfortable with the software and can try out different projects. Remember of course these software are just tools helping you out. Trying to get a good mix the most important thing is your ability to identify different sounds and this is where our ear comes into play.

Notes :

Non Sync Sound Track

Technical Processes

There are many ways in which you can technically start constructing your soundtrack:

1. Totally mix the soundtrack first (like a finished radio play or sound piece) and edit all your images and scenes to the soundtrack. With video you could lay the soundtrack down first (Insert Edit the sound only on channels 1 and/or 2). With Super 8 film you could edit the whole thing together while playing the soundtrack (to give you its feel and then mix it onto either a separate cassette or the film stripe (dubbing it onto the projector) which would run at the length which the edited film ended up being.
2. Same as 1 except divide the film into appropriate sections/ scenes/movements and work with a set of different finished soundtracks to match the visual segments. However because of the synchronism needed to signal those changes, this method wouldn't work for a separate cassette soundtrack.
3. Totally edit the visuals together and then - either as a whole or in segments - project it from a bio box and record dialogue and/or FX in 'foley' style (ie. post-dubbing). You could work on precise synchronization (realistic) or generalized synchronization (stylized). For either, though, you would need someone operating the projector, someone mixing the sounds (presumably a number of microphones would go into a mixer and then either onto the film stripe or onto the sound channels of the video deck) and a pile of people to do the dubbing.
4. Totally edit the visuals together and then chart all the actions and work on the soundtrack in a non-synchronous 'double-system' style, which could be produced using the 4-track or 8-track recorders in the Sound Bays. This processes is most involved, so following is a step-by-step guide:
   a. Firstly, a FINE CUT of you film/video is needed; if it is a film, do a telecine of the fine cut; insert a COUNTDOWN at the head of the video (one that contains a "10, 9, 8, 7, 6, 5, 4, 3, 2 countdown then going to black).
   b. If your film/video is over 2 minutes long, you will have to break the film/video down into sections of about 1 & 1/2 to 2 minutes long (maximum); insert a countdown at the head of each of these sections.
   c. Dub off these sections in sequence onto a VHS SHUTTLE copy of the video (this is the copy you will be playing backwards & forwards again & again while you do the sound post-production).
   d. Using a VHS edit controller, reset the control-track indicator to zero and line up with the first frame of the video; play the video and note the time-position of each action that requires either a definite start or finish of a sound to which you will have to synch a sound effect, music cue, atmosphere, etc.; do this for each section of the video if it has been broken down into sections, making sure each section starts at zero.
   e. Set up a VHS deck & monitor in the sound room; compile a fresh 4-track or 8-track tape ready for sound post-production by splicing WHITE LEADER TAPE at the head of each section required.
   f. Line up the first section on the 4-track or 8-track so that the leader tape is aligned to the RECORD HEAD; set the 4-track or 8-track COUNTER to ZERO; play the video - as soon as the countdown hits the 2 mark, hit the PLAY on the 4-trackor 8-track; play through the section, noting as you go the rough counter position of the 4-track or the LED time on the 8-track in relation to the time-positions of each action that requires either a definite start or finish of a sound to which you will have to synch a sound effect, music cue, atmosphere, etc. (as done in (d)). (For any crucial synch points you could also lightly tap a china graph pencil onto the tape as it passes the RECORD head - you can then later go back and make a definite mark at this point to aid later on when trying to synch a sound at these crucial synch points.)
   g. Work out a 4-track or 8-track score for the soundtrack by drafting 3 GUIDE COLUMNS: one with clock time positions (from (d)); one with the COUNTER positions ((f)); and one marking every CUT or SHOT in the film/video; then allow for 4 or 8 additional columns, one for each TRACK of the 4-track or 8-track; assign each track a CATEGORY (say, all dialogue on channels 1&2, all SFX on 3, and all music on 4, etc.); PENCIL in all sounds you will be requiring (you should now be able to see which tracks are free at any available point, and whether you are trying to put in too many sounds at once). Leave space in each TRACK column to mark in the VOLUME levels for the tracks when you come to doing a mix. (Note: on the 8-track chart you can also mark where sync dialogue-DIAL-occurs if you wish, as well as note the location-LOC-of a shot.)
   h. Record everything according to your score and in the right places (guided by your chart, counter and tape markings); it is best to do a track at a time; note that you must always start at the HEAD of the tape each time (where you have spliced the leader tape); when recording a sound onto a track further in from the head of the tape, play the video, hit PLAY on the 4-track when the video countdown hits 2, and in during

appropriate gap on that track (check this with your SOUND CHART) hold the RECORD down and the hit PLAY again - you will now be recording onto that track.

i. Mix down the finished soundtrack direct onto either the film stripe or video channels, or first onto a 1/2-track recorder and then dub that onto the film or video; this process can be utilized for the film/video either as a continuous whole or as a sequence of sections; if done in sections you can then edit the video sections together.

Dubbing

Dubbing has two meanings in the process of television production. It is used to describe the replacement of one sound track (music, sound effects, dialogue, natural sound, etc.) by another. The technique is used in the production of both audio and audiovisual media. It is a post-production activity, which allows considerable flexibility in "editing" the audio component of the visual. Dubbing includes activities such as the addition of music and sound effects to the original dialogue, the omission or replacement of unwanted or poorly recorded audio, or the re-recording of the entire dialogue, narration and music. Much like literary editing, dubbing allows considerable freedom to recreate the product. Synonymous terms include postsynchronizing, looping, re-recording, and electronic line replacement.

Dubbing is also one of the two major forms of "language transfer," i.e. translation of audiovisual works. Dubbing, in this sense, is the replacement of the dialogue and narration of the foreign or source language (SL) into the language of the viewing audience, the target language (TL).

Inherited from cinema, dubbing is extensively used for translating other-language television programs. Some countries and cultures prefer dubbing to subtitling and voice-over. In Europe, for example, the "dubbing countries" include Austria, France, Germany, Italy, Spain and Switzerland.

Dubbing, unlike subtitling, which involves a translation of speech into writing, is the oral translation of oral language. However, unlike "interpretation" in which the SL speaker and the TL interpreter are separate persons talking in their own distinct voices, dubbing requires the substitution of the voice of each character on the screen by the voice of one actor. It is, thus, a form of voice-over or revoicing. Dubbing is, however, distinguished from voice-over by its strict adherence to lip-synchronization. In order to seem "natural" or authentic, the performed translation must match, as closely as possible, the lip movements of the speaker on the screen. Moreover, there should be a strict, though easy to achieve, equivalence of extra-linguistic features of voice, especially gender and age. The matching of other markers of speech such as personality, class, and ethnicity is most difficult because these features are not universally available or comparable. Another requirement of successful dubbing is the compatibility of the dubber's voice with the facial and body expressions visible on the screen.

Lip synchronization is usually seen as the strongest constraint on accurate translation. The script editor modifies the "raw translation" of each utterance in order to match it with the lip movements of the person seen on the screen. Given the enormous differences between even closely related languages such as English and German, it is difficult to find TL words that match the SL lip movements; this is especially the case when speakers are shown in close-up. It has been argued, however, that a word by word or sentence by sentence translation is not needed, especially in entertainment genres such as soap operas. Lip synchronization can be better performed with a more pragmatic "plot-oriented translation." If translation aims at conveying the general tone of each scene rather than locating meaning in each sentence, there will be more freedom to find appropriate words for lip synchronization. Moreover, it is important to seek the equivalence of not only word and sentence meanings but also genres, text quality, character and cultural context. This approach is consistent with the claims of pragmatics, a new field of study which examines language use in social interaction (Luyken 1991:162-65). In either case, it would be more realistic to view dubbing, like other forms of language transfer, as an activity involving a recreation of the original text.

As the transnationalization of television and film increases the demand for language transfer, the controversy about the aesthetics, politics and economics of dubbing and subtitling continues in exporting and importing markets, and in multilingual countries where language transfer is a feature of indigenous audiovisual culture. The polarized views on dubbing/subtitling highlight the centrality and complexity of language in a medium, which privileges its visuality. Audience sensitivity to language can even be seen in the considerable volume of intralanguage dubbing. The miniseries Les filles de Caleb, for example, produced in the French language of Quebec, was dubbed into the French standard for audiences in France. And Latin American producers and exporters of telenovelas have generally adopted a Mexican form of Spanish as their standard, following the lead of the earliest successful programs. Thus, dialect also acts as a barrier in the transnationalization of television within the same language community, and highlights the complex issues surrounding this apparently simple industrial process

The Sound Crew in Post Production

Sound Transfer Operator

Responsible for transfer of production sound rushes from 1/4in tape to sprocketed magnetic film. Also responsible for transfer of sound from other media, such as disc, CD, R.Dat and from sprocketed magnetic film to sprocketed magnetic film. The transfer department is also responsible for the transfer of final magnetic master tracks, to make the optical sound negative, required for release printing. Sound Transfer Operators must have good knowledge of all the various sound media and good hearing in order to be able to detect flaws that may occur during transfer.

Sound Editing

This singularly creative process is usually divided into the categories of dialogue, sound effects and music, with a separate Sound Editor for each. Sound Editors do not necessarily specialize in one category only but are competent in all.

New digital technology in the form of hard disk editing is now being introduced into sound track laying, especially in television film drama series.

This system does away with the transfer to and-subsequent handling of the sound required for editing, on 16 or 35mm sprocketed film.

Once the dialogue, sound effects and music, along with the appropriate, time code, are recorded on the disks within the system, the Sound Editor watching the action on a visual display unit can by manipulation of the controls, cut, join, mix, fade in, fade out, lengthen, shorten, or alter pitch of any of the sounds.

Besides good knowledge and experience of film cutting room equipment Sound Editors should have a good understanding of sound recording for motion pictures.

They must be able to judge sound quality to enable them to assess the usability of all the sound elements.

Understanding the film script and the Director's ideas is important as besides collecting together all the sounds required and editing them into proper order he/she will be expected to make creative suggestions as to how music and sound effects can be used to enhance the dramatic effect of the story.

The Dialogue Editor

Responsible for ensuring that the dialogue recorded during production, whether in studio or location, is of usable quality for the final sound track. Organising automatic dialogue replacement (ADR) recording sessions and preparing the "guide track" for the artistes to post-synch to; adjusting the ADR tracks for accurate synchronisation; assembling all the dialogue tracks and laying them in a suitable form to allow efficient handling by the Re-recording Mixer; preparing charts -of all the footage and cues the Re-recording Mixer will require.

The Sound Effects Editor

Responsible for collecting together all the sound effects tracks, deemed necessary for the final mixed track and the music and effects only track, required for international foreign language versions. The sound effects required, if not recorded during production shooting, may be obtained from library, or recording sessions organised using special sound effects artistes. The sound effects tracks have to be edited in synchronisation with the picture action, the tracks laid in a manageable form for the Re-recording Mixer, along with suitable cue sheets indicating sound effects starts and stops.

The Music Editor

Responsible for recording the music. As well as good technical experience, should have a musical education, in order to be able Responsible for laying the music tracks and synchronising them with picture action. May have to make edits in tracks so therefore should have good knowledge of music.

Assistant Sound Editor

Each Sound Editor will have assistants to help them with the various operations. Not yet specialists in sound editing they must have good knowledge of cutting room equipment, techniques and procedures.

The ADR/Sound Effects Mixer

Responsible for recording of dialogue that is being replaced because of technical or performance inadequacies in the original recording. Also responsible for the recording of sound effects needed to make up the International Track. Works in conjunction with the Dialogue Editor, Sound Effects Editor and the Footsteps Artistes, who produce the necessary sound effects.

The Music Mixer

To converse with composer and musicians. Good ears are necessary for quality or recording judgement, as well as musical appreciation. Must understand film dubbing process and be able to advise composer when music may clash with other sound components. Music these days is usually recorded at studios not solely engaged in motion picture music recording.

The Re-Recording Mixer (Dubbing Mixer)

Responsible for the sound quality and balance of dialogue, sound effects and music in the final sound track.

Work includes pre-mixing dialogue, which means re-recording all dialogue tracks on to one track. During this operation the tracks have to be "cleaned up", equalised and levelled out. "Clean up" means getting rid of, by filtering, all or as much as possible, of extraneous background noise on the production sound tracks. "Equalise" means altering the frequency spectrum of post synch dialogue to match production recorded tracks. "Level Out" means adjusting the loudness of dialogue from shot to shot, so that all artistes sound to be engaged in the same conversation.

Sound effects also have to be pre-mixed, the quality and level of them adjusted to suit the action and mood of the picture. Much the same applies to music. Finally, all the tracks have to be mixed together and the levels of music, dialogue and effects, adjusted for the final balanced track. The Assistant Re-recording Mixer

Responsible for assisting Re-recording Mixer in pre-mixes, taking control of one or more of the tracks to be mixed. The very experienced Assistant may, in fact, do most of the sound effects pre-mixing. In larger dubbing theatres there may be two Assistant Re-recording Mixers. Re-recording Mixers and Assistant Re-recording Mixers must have good sense of hearing, good eyesight and fast reactions. With the new automated mixing consoles, now in use, a knowledge of computer operation is essential.

Dubbing Theatre Sound Camera Operator

Responsible for operating the multi track sprocketed recording machines that record the pre-mixes and final mix. Monitors the recording from the play-off head, to make sure there are no magnetic imperfections.

Sound Maintenance Engineer

Responsible for the correct running of all electronic and mechanical equipment used in sound recording processes. Must be able to diagnose and rectify faults in any of the equipment, whenever they occur. Faults developing during a recording session must be repaired with the minimum of hold up. The Sound Engineer should also be capable of developing and constructing equipment, and keeping pace with new technology.

Editing

Long before the dub (or possibly running alongside if it is a very long dubbing process, such as for a feature film), the dubbing editor(s) will have been assembling the required sound elements. In days of old, this would have meant transferring everything to an audio equivalent of the picture film -essentially 16 or 35mm recording tape with sprocket holes punched down one side, called 'sep-mag' (short for separate magnetic track). 'Everything' refers to the entire constituent sounds -the dialogue and sound effects recorded on location, ADR (dialogue replacement tracks recorded in the studio), foley effects (specific synchronous sound effects, such as footsteps, replicated by someone watching the on-screen action in the studio), library sound effects typically from CD collections, and music (your bit!).

Reverse EQ

Sep-mag machines really only ran at normal speed - either forwards or backwards - which meant that if you ran through a section to rehearse a premix pass, you had to run backwards through it again to get back to the top. Dubbing mixers quickly learnt to use this reverse play pass to their advantage and to minimise wasted time, by building the fader balance on the forward pass and, for example, making any equalisation adjustments as the tracks replayed backwards! With digital workstations, the ability to instantly recue has made this skill redundant, but it was a fascinating thing to watch and hear. Dozens (or even tens of dozens) of these sound reels would typically be created. Each reel would contain a logical group of sounds such as just location dialogues, various background effects, foreground 'spot' effects, music and so on. Each sound element would be physically edited to the right length and positioned within the reel using silent leader tape to ensure accurate synchronisation with the associated pictures (which would already have been edited).

Having created each reel, the dubbing editor would draw up a 'dubbing chart'. This consisted of a collection of large sheets of paper, each divided into vertical strips representing the various reels. Shaded areas on the chart represented sound elements on each reel to show the mixer what sounds were on which reels and at what times. The shape of the shading gave an indication as to how the editor envisaged the various tracks being combined, and there were usually written notes too.

These days, the role of dubbing editor remains, but the numerous sound elements are typically compiled and loaded into some form of hard-disk audio workstation (Pro Tools or Akai systems, for example). Once on the hard drive or magneto-optical (MO) disk, these elements can then be assembled in a playlist to create many tracks (performing the role of the old reels) with each audio element positioned to match the required synchronisation. Another advantage of workstations is that the playlist screen forms an automatically scrolling dubbing chart, allowing the mix engineer to see when various sound elements are about to happen, and on which tracks.

The Dub

For the dub, the various pre-edited audio components arranged on a collection of hard drives, often from a range of different systems, are then moved across to the dubbing theatre and configured to replay into the mixing console.

Lightpipes Explained

The dubbing chart is a major aid to the dubbing mixer, but another visual assistant, usually only found in the older, film-based theatres, is the lightpipe. This system looks a little like a collection of giant LED bar-graph meters under the projection screen, but in fact each represents the timeline of a specific replay track. A typical system might have 100 LEDs in each bar with a driver circuit which detects the presence of audio above a preset threshold. The audio signal would be derived from a pre-read head on an analogue replay system (or in conjunction with a delay unit in a digital system) operating typically four seconds in advance of the real output appearing at the sound desk.

If audio is present, the first LED would be illuminated, and this illumination then clocked down the line once per video frame (25 frames per second). The visual effect is of a block of illuminated LEDs travelling across the bar graph, the length of the block directly related to the length of the sound. When the block reaches the right-hand edge of the bar-graph, the sound is heard. In this way, the mixer can prepare to open and close the fader with greater accuracy than is possible from looking at the dubbing chart and comparing timecode readings.

It is not unusual to find that the dialogues have been compiled on one workstation system, the sound effects on another and the music on a third -dubbing editors, not unreasonably, like to use the editing tool best suited for each particular task. Until recently, it has not been possible to take a disk pack recorded on a Pro Tools system and install it in, say, an Akai system or an Audiofile, and still be able to edit and modify the files (although it has been possible to play native files across many machines for some time).

However, the recent AES31 specification for hard disk audio systems defines a common file format and protocol which should make audio workstations from different manufacturers entirely compatible... when they have all implemented it, anyway. At last, we might re-attain the ability to work with material recorded on any platform, on any other platform -just as a reel of quarter-inch tape recorded on a Revox could be replayed and edited on a Tascam, Studer or Otari machine!
Not all audio sources at a dub will necessarily be replayed from hard-disk formats. There may also be a range of material on tape-based systems, typically DTRS machines, but also timecode DAT. All of these disparate audio replay devices would be

synchronised to the picture source via timecode- the pictures being projected from a conventional film projector, a video master, or possibly from a professional random-access digital video recorder of some form. Film does not have an inherent timecode capability and film projectors produce something called a 'bi-phase' control signal. This relates the speed and direction of the film transport, but not the absolute position. However, bi-phase-to-timecode converters are available for synchronising more familiar transports.

Although digital workstations locate and enter play very fast, and modern tape transports synchronise far quicker than any previous generation of transport, it still takes several seconds for a complex timecode synchronisation system to achieve

frame-lock and stable speed. This, however, is a lot quicker than a machine room full of 'clockwork' sep-mag recorders!

The Dubbing Theatre

Dubbing theatres come in many different shapes and sizes. The smallest are one-man operations, typically centred around something like a Yamaha O2R console with an Akai or SADiE workstation. Moving up the scale are AMS Neve Logic consoles with Audiofile editors, and the Fairlight FAME systems, to name but two from the many alternatives. The top-end dubbing theatres, generally only used for mega-budget feature films, employ two or three audio engineers sitting behind an enormous console with a vast array of audio sources. Harrison, SSL, and AMS Neve are the major console players in this part of the market, and digital consoles are increasingly becoming de rigeur.

The idea of the multi-operator console is that each engineer can look after a specific subset of all the audio sources during each premix pass, thereby making it much faster and easier to handle the huge number of original sound components and to produce the finished result. This approach makes a lot of sense when you realise that a major feature film could easily have well over a hundred reels of audio tracks running at any one time, many of these sound elements being stereo or even surround encoded already, all needing sophisticated equalisation, panning and other types of signal processing!

The Process

The dubbing process usually starts with 'premixes'. This is the first stage of submixing to reduce the hundreds of separate audio elements into something rather more manageable! The various background effects might be mixed first to create a couple of different background premixes, then the spot effects, foleys and so on. The dialogues will also be premixed, combining the location and ADR voices to produce a seamless voice track.

Eventually, the premixes will, in turn, be mixed to produce the M&E and, finally, the finished soundtrack. Another term frequently used instead of premix is 'Stem'. A Stem is a complete premix, usually in a multitrack format arranged for surround sound. So you might have an effects stem, a music stem, a dialogue stem, and so on.

Again, the practice has changed somewhat with modern technology. This process used to involve recording the premix onto a new reel of sep-mag so that the dozens of separate original reels would gradually be replaced with a smaller number of premix reels. Now, the premixes are either recorded directly onto a digital workstation or, in the cases of the largest consoles with enough inputs to accommodate all the replay audio sources simultaneously, the audio is not recorded at all - the console automation data is recorded instead!

One of the major problems with physical premixes is that if some part of the balance doesn't work when all of the premixes are combined (if, for example, just one spot effect is a little too loud or quiet in the context of the complete mix), the offending premix reel has to be remixed again to correct the problem. In the past that meant taking off all the other premix reels, loading up the original effects tracks and remixing all (or part) of the relevant premix. Then the source reels were replaced by the other premixes and another trial of the final balance could be done - a very slow and laborious way of working!
The advantage of the huge modern digital consoles with advanced automation is that they can accommodate a vast number of input sources with negligible deterioration of the noise performance, and tweaking a premix balance is very easy and incredibly fast. Making a minor change to the automation data can usually be done on a computer screen in seconds, or the appropriate control can be accessed and adjusted on the fly, the necessary alteration being made live during a second pass.

Drop-ins

The chances of mixing several dozen effects reels to produce the perfect premix in a single pass are pretty slim, even for the most experienced dubbing mixers. Consequently, it is necessary to have some means of dropping in to a mix just before the point where it all went horribly wrong, and then continue mixing until the next time it falls apart! Drop-ins are nothing new, of course - musicians have been dropping into to separate tracks on a multitrack recorder to replace a guitar part, for example, for decades. However, dropping in on a complete mix is a little more challenging.

The problem is that at the point of the drop-in the old mix and the new mix have to be 100 percent identical, otherwise the mismatch will produce a click, a gap or some other clearly audible artefact. Also, if the synchronisation of the disparate replay sources is not stable, there might even be some audible phasing or flanging through the drop-in crossfade period! With console automation, matching the positions of the faders is relatively easy, but in days of old the positions of everything had to be remembered (or marked with wax pencils on the fader escutcheons!) and the balance matched by ear.

This is where the term 'PEC/Direct' switching raises its head. Look at a brochure for any dubbing console and you will find mention of this obscure-sounding facility. In fact, this is really nothing more than a set of multi-channel monitoring switches - multi-channel because of the need to record either stereo, matrix surround (four tracks), or discrete surround (six or eight tracks). PEC stands for 'photo-electric cell', which was the original method of decoding the optical sound track on a film. Although optical tracks are still available on every theatrical release film print, they are not generally used in the dubbing process any more. However, the term has continued to be used, these days referring to the output of the multi-channel recording device (tape, sep-mag or hard disk). The 'Direct' position refers to the mix output of the console.

The idea is that by operating the PEC/Direct switch rapidly back and forth as the film plays over the section immediately prior to the point where it all went wrong, it is possible to compare the previous recording pass with the current output of the console. The dubbing engineer can then adjust the faders and other controls until the sound is identical in both positions of the switch. At this point it is possible to drop into record, safe in the knowledge that the punch-in will be inaudible, and have another bash at mixing. If it all goes wrong again the next drop-in would normally be performed slightly earlier than the last one, otherwise you could end up with a sequence of drop-ins in close proximity which might become audible.

M & E

M&E stands for 'music and effects' and is a complete soundtrack mix of everything except the dialogue tracks. The M&E is used as the basis for foreign-language versions of a programme, where all the voices would be dubbed in the relevant language in a specialist studio. One of the reasons for performing premixes, aside from the need to reduce the number of sound components to something easier to handle when constructing the final mix, is that it naturally leads to the M&E mix as part of the process.

Any film or television programme with potential overseas sales should have an M&E track made up, although many production companies try to cut costs by not having this premix recorded separately. It is, of course, far easier (and considerably cheaper) to prepare an M&E at the time of the dub, rather than having to come back to it some time later and work through all the premixes again!

It is also possible that a number of different versions of the M&E mix might have to be made because of copyright clearances on the various music tracks. For example, it might be possible to acquire rights to use a commercial music track in a programme for public viewing in the UK, but not for use overseas. In this case, a separate 'world' M&E would have to be rebuilt using alternative music tracks.

Final Mix

The nature of the final mix depends, to some extent, on the intended audience. For example, the acoustic character of a large cinema is entirely different to that of a domestic living room, and the balance (both spectrally and dynamically) would be adjusted accordingly. Loudspeaker monitoring systems are aligned to well-documented standards for different end-listening conditions - the 'X-curve' is used for cinema mixes, for example, where a lot of HF energy is built in to the mix to compensate for the air losses in a large auditorium. That is one reason why THX-approved home-cinema systems roll off the high end a little to avoid feature films appearing excessively bright in a domestic environment.

The final mix, whether in stereo, matrix or discrete surround, would usually be archived onto a timecode DAT or DTRS tape and eventually laid onto the film or video master, again with timecode as the synchronisation reference. If it is anticipated that some further dubbing work might need to be done (such as re-editing a long programme into a series, or vice versa), then the various premixes might also be archived as stems onto tape or hard drives to minimise the amount of remixing (and therefore cost and time) required.

Notes :

A BASIC MUSIC MIXING METHOD

Mixing is one of the most difficult things to get right in music production and one of the most important. Every producer and engineer will approach a mix differently and like many elements of music production, there are no right or wrong ways to go. It's your music and only you know how it should really sound. A unique mixing and production style can go a long way towards creating your own personal style or sound.

Start by cleaning up your tracks. Solo each track and remove any unwanted noise or through the use of mutes or gates. You want to eliminate any extraneous sounds like coughs, squeaky chairs and lip smacking before you begin to mix. Also check for unwanted noise on continuously playing tracks like amp hiss on an electric guitar part. Eliminate it with a noise reduction unit.

Most people start with a solid foundation to build on. That usually means mixing the rhythm section first. Start by adjusting the master volume to a level that you might expect a home listener to be at when listening to your music. Most people won't be listening to your music at deafening levels, try to be sensible.

Start with the bass drum, and then add the snare and then the rest of the percussion. Avoid using any EQ at this point -when the rest of the instruments are added the rhythm section is going to sound different anyway.

Once the rhythm section sounds good, add in the bass line. We're still just working with volume for now to get the parts sitting where you think they should be in the mix.

Once the fundamental groove is established the vocals or lead instrument should be added. We work with these two elements first simply because these are the most important parts of the mix. The groove will get people listening and the vocal is what they'll hum along to and remember later.

Once the vocal is sitting about right in the mix you need to add in the other instruments. If you're using guitar add it in next, but keep the volume level slightly below the vocal. Next add in any pads, keyboards or sythesizers and balance these out to suit how you want the track to sound. Remember that for the moment you are looking for a gentle balance between all of the instruments in the mix. You're still just working with the volume levels of the track more than how it all fits together. Once you think you have it go to an adjoining room or stand in the doorway to try to gain some perspective on the mix so far. Adjust any levels, which sound a little off and leave for the evening. Do not over tax your ears and beware of ear fatigue, especially at the beginning stages of mixing. Each time you accomplish a part of the mix it is best to walk away - have dinner, listen to some CDs and come back the next day with a fresh set of ears and a new perspective. Listen carefully to your favorite CDs. It is often a good idea to bring them into the studio and A/B them with your own mix to see how your levels compare with what you consider to be a good mix. Also, this will help you become familiar with the speakers and the room that you're mixing in. Our studio isn't acoustically perfect by any means. If you have a well-mixed CD and try to replicate that in the studio by A/B-ing your mix and the CD, you should come close to your ultimate mix.

Processing Your Mix

Now that are the volume levels are pretty well set we can begin to process the mix. A likely first place to start is compression using the Waveshell plugins. Compressing every instrument is a matter of taste and opinion. The New York style of mixing generally uses a lot of compression and we'll compress each instrument. If that's not for you, skip ahead to EQ-ing.

It's best to start by compressing the bass drum and snare. Start with an attack time of around 5ms, a release time of 10ms, a threshold of about -15db and a ratio between 5:1 and 8:1. Keep adjusting the ratio until you get a good, solid, tight sounding kick and snare combination that really pierces the mix. Compress the kick to where it's punchy, but not ringy. Over compression on the kick will make it vibrate and ring and eat up the bottom end.

After compressing the volume will have decreased slightly, so use the compressor's make up gain to get the volume back to where it was originally. There's no need to compress the rest of the percussion parts since we'll be compressing the entire again mix in the final stages of our mix.

The bass line is next. If the source is a sample or synth it will already be heavily compressed at the source. If you're using "real" bass sounds they'll need compressing. Start with an attack of 5ms, a release of 10ms, threshold of -6db and a ratio of 8:1. Again use makeup gain to increase the volume level of the track to its original level.

Vocals will definitely require compression, mainly to reduce any peaks in volume and to keep them at a consistent level. The best settings are generally the fastest attack time, a threshold of -5db and a ratio you can set, a release time of around 0.5ms and a ratio between 4:1 and 12:1.

To make a rock vocal really stick out, compress the track really hot to get it as loud as possible. Next run it through a little tube distortion if possible. Finally, copy the track to another channel and boost the high end EQ. This will bring the vocal close to the brink of disaster for that old-fashioned Rolling Stones type of vocal sound.

Again use make up gain to bring the level back to where it should be and once again give a listen from the doorway or the next room. Are the elements distinct? Do certain parts of the mix fade away if you're not sitting right in front of the speakers? You've been working a long time now, so have another break or call it a day and come back with fresh ears.

EQ-ing Your Mix

With EQ, it's better to cut, rather than boost a signal as this tends to make an instrument sound unnatural. Boosting the EQ of a track will also tend to increase the amount of noise - especially if you are using an analog mixer.

You need to practice the art of subtractive mixing on midrange frequencies. Start by dialing in a fairly radical cut - say 15db. Set the bandwidth to a narrow Q setting. Now sweep the frequency spectrum slowly from hard left to hard right while listening to how the cut affects the signal at different frequencies. Next do the same thing with a 15db boost and compare. What you're listening for is how a cut on one side of the frequency spectrum results in a sound quite similar to a boost on the other side. For example, a low-mid cut can result in an apparent brightening of the signal - much like what you would expect from boosting the mid-highs.

It's best to EQ an instrument in the context of the entire mix. Soloing an instrument to EQ it can be helpful, but it only matters what an instrument sounds like in the mix.

The bass is usually the instrument most in need of EQ because its frequency range goes below 80Hz and can frequently clash with the kick drum. Try to keep the bass between 80 and 100Hz. IF the bass is still overpowering the kick drum you can try a few tricks to make either one or the other seem louder and more up-front.

In rock music you need to record the bass guitar so that it is bright enough to cut through the mix. There's often a frequency hole between the bass and the low end (or lack of low end) of the guitars. This is one case where you may need to boost the mid-range of the bass. Find just the right amount of boost so that the bass will peak through under the guitars. A trick is to re-mic the bass through an amp or to add a little distortion to get more grind out of the top end.

One idea is to use reverb, which will make an instrument move back in the mix. Unfortunately this will make the instruments seem washed out and possibly destroy the rhythmic integrity of the track. One way around this is to use a reverb pre-delay with a very short release time. The pre-delay fools the brain into thinking that the instrument is further away than it really is and it doesn't wash out the sound.

Another idea is based on the way our brain perceives sounds. If two sounds have the same relative volume level but one starts slightly before the other, then we perceive the first sound as louder. You can use this trick on bass line or drum track. If they both sound at the same time, try moving the bass line a tick or two forward to make it sound more prominent in the mix. If the bass drum and snare are still fighting with the bass after EQ-ing, try cutting the kick drum at around 40 to 70Hz and boost the snare drum at around 105Hz. In general, you don't want to cut any low end from the kick drum, but cutting a little around 400Hz can take out some of the boxiness. It's best to keep the kick and bass at good relative levels to avoid problems during the mastering stage. That way, if there's not enough or too much low end, it's easy for the mastering engineer to fix it with EQ.

Vocals shouldn't require much EQ, but if you have too much sibilance try cutting a little bit around 6kHz. If that still doesn't help, you'll need to use a de-esser -but these can make your vocals sound quite artificial.

Again, step back from the speakers and listen to the mix from the doorway and see how you mix sounds. If things are sounding good it's time to start adding effects and working with the stereo spectrum. If the guitars are overpowering the mix at this point don't drop their volume as it will affect the entire mix and you'll start raising and lowering volumes of other instruments to compensate as you try to acquire the correct balance again. This is where many beginners mess up. You need to use EQ, not volume to fix the guitars. In nature, sounds that are brighter, are perceived as closer and slightly louder. If guitars are too loud, cut the EQ slightly to drop then further back in the mix. Conversely, if they are too quiet, boost the EQ slightly to bring them forward. Go easy on the EQ and keep the entire track playing as you adjust specific instruments, then solo the EQ'd track to hear clearly what effect you're having on it.

Guitars, pianos, and synth pads are notorious for taking up too much of a mix's frequency spectrum. Sometimes no amount of EQ will help the situation and it's

Best to employ an effect to thin out the offending sound. A chorus/flanger effect is designed to add movement to a mix and make it louder, but with the correct settings you can also use them to thin out a sound. A setting with a very short delay (5 or 6ms.) and the delayed sound output set to -50 creates a comb-filtering which will result in a thinner sound.

Another trick is to EQ the effect on an instrument rather than the instrument itself. This is especially useful if you're happy with the quality of the signal (how well you recorded the instrument) and only want to alter the tonal balance subtly.

Effects

A common mixing mistake is the over or under use of effects which results in a muddy sounding mix or a hole in the area between the bass and guitars. If your mix has this hole, it's best to fill it in with another part. If you don't want to tinker with the arrangement then you'll have to use reverb. Guitars have a large amount of treble associated with them and not much bottom end to fill in the gap between the bass and guitars. A modulated reverb or a reverb with a low bandwidth setting is ideal to tackle this common problem. TC's Native reverb plug-in is ideal for this as it can be set with a low bandwidth setting. Be careful not to wash the guitar in excessive amounts of reverb - it will make the mix muddy. Using a small amount of pre-delay with an even shorter tail is often a good solution. Muddy mixes are almost always caused by excessive reverb. Use just enough reverb to make to make it noticeable if it isn't there. Another good reverb plug-in is the Waves TrueVerb.

One of the more common uses of too much reverb is on vocals. Reverb affects the overall tone of a sound and therefore interacts with the rest of the mix. When too much reverb is applied it affects the sibilance of the vocals. Cutting EQ around 12kHz can help if you need to use large amounts of reverb on vocals, but it's best not to use too much to start with.

An underused effect is the gate. If you're using a large amount of reverb with a long tail, then a gate can come in handy for cutting the tail end dead. If the tail goes on for too long it can swamp other instruments. Clever use of gating can prevent the tail from continuing when the next note is played. The same trick can be applied to any delay effects you may be using to cut off any repeats which don't need to be present. They can also prove useful for any sudden stops in a track - music that suddenly stops and goes dead silent has a much bigger impact than music that suddenly stops to reveal background hiss.

So far we've been mixing in mono- if it sounds good in mono then it'll sound much better in stereo. A gentle pan her and there can help separate parts of the mix which are occupying the same bandwidth as each other and can't be EQ'd. In general, you don't want to pan instruments hard left or right as it often un-balances a mix. Panning should be seen as part of the mixing process rather than as an effect. Bass, drums and vocals should remain dead center in the mix unless it's for creative purposes. Background vocals and percussion can be made more noticeable if panned either right or left a small amount. If two instruments are fighting each other for the same sonic space try panning them far apart in the stereo field. On the other hand, most older rock records recorded on old 4 track consoles had only three pan positions -hard left, center and hard right. Almost all of the Beatles records were panned in this manner and sound great.

Final Production

Hopefully your mix sounds pretty full at this point. The instruments should all sit well in the mix - each clearly heard and distinctive and the stereo spectrum should sound full. If there are still problems you need to single out the problem by soloing each track one at a time and listening to it. If you think one track is causing problems, mute that track and listen to the rest of the track without the offending track. If the track sounds better without the track you'll need to re-record the track, use a different instrument for that part or leave it out all together. A mix only sounds as good as the arrangement itself. If the mix seems too busy, try to simplify the arrangement. If you can't lose any of the instruments then try dropping them back in the mix so that you only notice them if you turn them off. Remember that with most songs it's only the vocal or melody and the fundamental groove that matters. Everything else is just decoration.

It's worth adding a little reverb to the entire track once it's finished to add a little more sparkle. A little pre-delay with an almost nonexistent tail set at 50/50 should work. It's also advisable to reduce the bandwidth of the track with a little compression to give the track more energy. It depend what you're expecting to do with the mix. Radio and Television broadcasters will compress your track a staggering amount. Waves RCL compressor is probably the best plug-in around for finishing up a mix. Set the threshold around -9db with a ratio of 2:1 and a quick release and attack time.

If the track sounds a little dull after compression applying a small amount of distortion can add some extra harmonics and warmth like a tube used to. If you're looking for radio or TV play, apply a small amount of EQ boost at around 4kHz to lift the treble of the track which will make it sound more polished over the air.

Finally listen to your track over as many different systems and speakers as possible.

The dbs and the Different Instruments

There are many ways to get your songs to final form. Lets assume, for this lesson, final form means a beautifully polished piece of music in 16 bit 44.1 khz digital audio (i.e., the "red book" cd audio standard) or a standard wave file. You need to start, of course, with a fully or almost finished song. This is the point where the writing ends begin. We are going to talk about some tips on Mixing and Mastering in the old analog style. Mixdown and Mastering, traditionally speaking, are two very separate processes. Mixdown is the art of leveling, equalizing and effecting all the various sources from many tracks down to a stereo Mix. Mastering is the process of taking the stereo mix and putting it in the final album-ready form. Recent software and hardware developments make these processes easier and less expensive than they ever have been in the history of making music. Given that much of the time we can stay in the digital domain we can add processing to our heart's content and maintain a high signal to noise ratio and achieve optimum dynamics for the piece at hand.

The Mix Process

Please consider these parameters not as rules but a starting point for you mixes for the standard pop song or ballad using an analog mixer. Of course the instruments change if you are doing techno or symphonies, or ambient stuff, but the reference may still be helpful.

Step one is always to calibrate the mixer. Use a test tone of 0db (that's LOUD, so turn down the monitors). Set the fader at 0db on the board. If you don't have a tone to use take the loudest sound that the channel does during the mix, Set the trims so at the loudest, the meter pegs at 0db. Do this for every channel in the mixer. This gives you a reference. a zero db signal will meter at zero db when the fader is at zero db. Now you know what those numbers are for that are silk-screened on your mixer! Do It!

Match the following instruments when soloed in place to the db markers on your mixing desk or your mixdown deck or software.

Kick drum- 0db +3 eq at 50 Hz +1 db at 3khz -3db 275 hz No FX except maybe subtle ambience You will tweak the kick again, this is just to get you going. In an instrumental piece, the kick is the first and last tweaked. It's got to be just right.

Snare- 2 db eq to taste in the frequencies above 4khz. Add reverb if the song calls for it. Do the best you can to keep it out of the way of the vocal, even if you have to pan it a few degrees. Near the end of the mix you need to come back here to perfect it.

Lead Vocal 0db use a low cut filter to eliminate rumble and plosive pops around 100-200 hz. Carefully enhance the delicate high end around 15khz to add air and sheen and don't overdo it! This is the trickiest adjustment and may often spell hit or dud. Perfectly center the vocal and pan it not with pan controls, but with very subtle left/right hi freq eq's. Put on the cans (headphones) and make sure its in the absolute center of your forehead.. Every word must be intelligible. Add reverb and delays but don't let it get smeared. Before you print to tape or DAT or whatever, check the vocal any make those tiny adjustments that are needed.

Cool trick: Split the main vocal track to two seperate faders. Compress the main vocal and send the secondary, uncompressed vocal to a reverb unit. This way the reverb stays out of the way until the vocalist gets loud. Hey that's they way it works in real life.

Cymbals -25 db Avoid letting these get in the way of the vocals. Pan them to 2 o'clock and remember their main function is to add the glue to a track to hold the music together—they do not have to be loud or present. Think about how horrible they will sound on your girlfriend's or boyfriend's car stereo if you let then get too loud.

Tip: Never let the drummer in the control room, except under extreme sedation, unless you want all your mixes to sound like Led Zepplin

Synth pads -20 db Do these in stereo and hard pan left and right with generous effects if needed. However, keep them in the back. Pads indeed are beautiful additions to a song but don't let them overshadow any of the main elements of the song. Yet for a sense of dimensionality, let these create a "landscape" the listener can walk on.

Cool trick—you want a really BIG Pad? Delay one side of the Left/Right by about 10-12 microseconds. You'll be hearing a landscape if you do it right. Don't let any engineer tell you these have to be mono. Make him earn his pay by fighting the phase issues. All you do is do a mono check on the mix and make sure the stereo pad didn't disappear.

Bass -10 db maybe hotter Always front and center. If you use FX restrict yourself to chorusing or a light flange—no reverb. Note that the quality we associate with "good" music is a tight syncopation of kick drum and bass. If you hear any duff notes make sure you fix them.

Cool trick: Bass does not have to hit exactly on the kick drum. But it a wee bit after so the listener hears the kick 1st. Do microseconds count? Yep. Ears are really good at detecting even tiny, tiny delays in what we hear. Are there more secrets in the micro-timing domain? Yer catchin' on dude—good work!

Rhythm guitar -15 db pan off center eq: use a low cut filter to get rid of any bass and add a mid range eq for a slight narrow boost, but make sure it is not competing with the vocalist's sweet spot.

Hot tip: Bass KILLS, remember that. Get rid of ANY bass frequencies you don't absolutely have to have. "B-b—b-ut" you sputter, "my guitar now thounds like thiiit" Want wine with that cheese?. When the solo is on it does, in the mix, listen to it blend so well with everything.

Percussion -20db- put these elements off center unless they are essential to to basic beat. EQ in a tasteful way if necessary. I shoot to get a little skin sound on the hand drums if possible. It's tricky, don't add too much.

The Mix itself

Now, watch the meters when you play the whole mix through the board. On an analog board you should have peaks at no more than +3db. If what you have is more notch down every fader in 1 db increments until you get there. Shoot for 0db. On a digital board (or software mixer) you never want to go over 0db, anywhere, ever.

Mono Check: Always check you mix in Mono and look for sudden drop outs or instruments that disappear. That's phase cancellation at work, and it happens with stereo tracks and effects.

No faders above 0db rule: When getting a mix started follow this religiously. If you find your vocal doesn't sound good unless its at +5db then move everything down 5 db. Conserve headroom. You don't want your mix compromised by that awful crackle at the peak of your song.

Now you fine tune to taste. Listen for the quality to "lock". There is a definite point where this happens. Suddenly it all falls into place, given you have good material. A great mix of a great song will fill you with absolute elation. You'll be blown away and in awe. You will feel in love with it. No kidding. Might sound corny to the less mature among us, but I assure you its true. A great artist friend of mine puts it this way. Greatness in art depends solely on how much love you put in to a work. You put it in, it pays you back, your friends back, and everyone who listens. Moral of this lesson. Never take mixing and mastering lightly. The tiniest fader movements make a difference.

The Mix is a Dynamic, Moving Process

Don't just sit there while your mix goes to tape, or disc, or DAT. If you are using a board, assign the faders to subgroups. For example, if you have 4 subgroups you might want to send your vocal tracks to groups 1 and 2 and everything else to 3 and 4. This way you can slightly alter the balance between the vocalists and the band as the piece goes to tape. This technique, while tricky, can yield outstanding results. You can give the vocalist a touch more edge just when they need that oomph and when the vocalist takes a break you can subtly boost the band a bit. If you have 8 busses you might dedicate 5 and 6 just to drums and 7 and 8 just to effects, nudging each as is appropriate. If you have a digital mixer, this is where you want to automate.

The Role of Compression at Mixdown

On it's way to the recording device, you can patch a compressor/limiter/gate. The Gate simply cuts out any audio below a certain threshold so that any hiss or noise coming from your synths or mixer is eliminated before the music starts. The limiter keeps your peaks under a certain fixed level and will not let them go higher. A Compressor is a volume slope applied to the audio material going through it. It can amplify the "valleys" and attenuate the "peaks". Essentially compression reduces the dynamic range we have just struggle to achieve in our mix. You might wonder why you would want that. In many circumstances, you don't want it. However, in the majority of cases you will find it useful, especially if you want your music to be "hot", "have punch" "be as loud as possible", or have the consistency of a radio mix. The stereo compressor also helps balance the song and give it a uniform character we are so used to hearing in commercial music. It essentially gives you the strongest and smoothest mix and calms down some of the 'jaggged edges' that might disturb the casual listener. However, it is also very easy to make a mix totally lifeless with a compressor and reduce its dynamic power. What started as a powerful orchestral arrangement can end up a wimpy piece of Mall Muzak so be careful and bypass it frequently to make sure you like what you are tweaking up. Compression works well to attenuate that occasional peak that rips through the roof of a digital audio recorder and ruins the track. Also if you have the cash for a fine analog tube compressor or even a high quality compressor plugin, there is lots of magic you can do at this stage.

The Role of the Mastering processor

Mastering processors are becoming more popular these days. The effective use of a mastering processor can transform a good mix into a great master recording. If you have one, you might consider using that in lieu of a compressor at mixdown as mastering processors usually have all the functions and additional functions such as mastering eq, multi-band compression as well as limiters and gates. These mastering tools can go a long way to giving your music a unique sonic imprint. There are many uses. In addition to adding the refining touch to your mix as it goes to the recorder, it can be used to give all your songs on an album a consistent uniform character and balance the volume between widely different songs giving your project a professional touch.

Using narrow band mid range eqs can give you a very contemporary sounding presence and make your dance tracks come alive with freshness. Pumping the compressor a little at 50-60hz can give you the "kick in the chest" kick drum without wrecking the delicate dynamics of the high end vocals. There are many more applications such as using them to send midi tracks to your digital audio mixer compressed optimally, ducking for voice overs, de-essing, warming through "tape saturation" parameters and Hard Gate effects on individual tracks. Remember Tweakheadz rule of thumb: Any piece of gear can be used in any way as long as it enhances the quality of the final product.

What is a Multiband Compressor? Don't let the terms freak you. It's like the bass, treble and mid range controls on your stereo with a compressor on each, able to be tuned in fine adjustments. With experience and skill, you can dramatically transform your mix. You can also make it sound worse

Software Mastering and Post-Production

A good digital audio sequencer will let you master in the digital domain of your computer. Some softwares that are of particular merit for mastering are Nuendo, Logic, Cubase, Sound Forge and Vegas. Lets look at just one of them right now say Vegas. The main thing is to be able to draw a volume envelope over the whole waveform. Rather than botch a fade 20 times on an analog mixer, simply draw in the perfect fade with the mouse. Where the piece loses intensity, notch it up a tad, to restore your intended dynamism to your mix. Say you have the perfect mix except for one horrible "sp-p-p-lat" where your sequencer choked at bar 72. No prob. Just remix the offending bar again, cut out that piece in Vegas and drop in the new one and let the automatic crossfading give you the absolutely perfect, digitally calculated crossfaded splice. Works! Need to touch up the EQ and do your compression in software? Tweak it in. It's all undoable, so your not going to ruin anything. Decided the mix you did last year really sux? You need to cut out a chorus or fade 5 seconds earlier? Say you did a trance piece but the kick is so wimp that it makes you cringe? Just drag in a looped 808 kik and paint it on the next track, setting the volume and compression to make the whole song whupass. :) Vegas gives you the tools.

The Final Touch

You've worked hard on a song, maybe several weeks, maybe longer. It's now in final form, just a matter of the last transfer to DAT, Tape, Wave or CD. Here we enter into the subtlest, but arguably, most far reaching of your tweaks. Sometimes it makes sense to compare the quality of masters to metals. Do you want a mix of raw iron? Steel? Silver? Gold? Of course we choose gold for most things. Gold is firm, strong, yet soft, malleable, pleasing. This takes you right to the heart of the digital vs. analog controversy. And you no doubt have heard the advice "use your ears!". And perhaps you've heard of engineers said to have "golden ears", indeed a point of much pomp and egosity in the field. What does the golden-eared producer have that you don't? Listen close now, here's a secret, your reward for reading so far. What they have is an aural image in their minds of how things can sound beautiful, and they have the gear that allows them to get the audio to that place in their heads.

Think about that OK? It's like religion. The believers all see a truth that is obvious that no one else can. Is your audio like velvet? Silk? Or is it more like uncomfortable rayon, or dull like flannel or harsh like sandpaper.

The final touch is never easy. You are also fighting with "the audience in your head" on how something should sound. Finally, you have been working on it so long you might NOT be hearing what it really is as your brain is conditioned to filter what you want to hear. If you can't nail it by the 3rd or maybe 4th play in a session, can it for the rest of the day. Bring everything up to spec as close as you can and come back tomorrow. The most important factor in the final touch is not gear; it's the interaction between your ear and your mind. Yet having good gear at this stage helps your ear and mind "find" that doorway to quality, where you blow yourself away into sonic ecstasy, and your final master communicates that to everyone who hears it. This, my friends is the "holy grail" of audio. It's where a song becomes more than a song, it's an adventure, a voyage, a statement. I wish you happy journeys.

Summing Up:

Whether you are writing industrial hardcore or the darkest ambient, a 100 piece orchestra or a stark minimalist a capella mix, always keep your ears tuned to making an artistic statement, a work of unforgettable beauty. This is the bottom line. The more control your Mixer gives you, the better you can paint the overall image. Working with compressors and mastering processors gives you a shot a polishing that image much like we polish a stone to bring out its colors. Hope this article helped you get a handle on the concepts of the perfect Mix, mastering and post-production, and the Final Touch.
All the Best in your music making.

Notes :

SOUND CONCEPTS

The story of sound in film begins not, as many historians have presumed, with the introduction of the sound film, but with the invention of film itself. At no period in the history of films has it been customary to show them publicly without some sort of sound accompaniment. In other words, you could say, the silent film never existed.

As soon as films were invented, and long before there were such things as picture palaces, filmmakers and showmen began to employ devices to provide a sound accompaniment and complete the illusion. First they used the phonograph, to which Mr. Routledge refers in the extract quoted above. But not for long. Phonograph records are fragile, and synchronization of records has always been a chancy business. Furthermore, as films got longer and longer, they needed more records than it was convenient to make and maintain.

The Acoustic World

It is the business of the sound film to reveal for us our acoustic environment, the acoustic landscape in which we live, the speech of things and the intimate whisperings of nature; all that has speech beyond human speech, and speaks to us with the vast conversational powers of life and incessantly influences and directs our thoughts and emotions, from the muttering of the sea to the din of a great city, from the roar of machinery to the gentle patter of autumn rain on a windowpane. The meaning of a floorboard creaking in a deserted room, a bullet whistling past our ear, the deathwatch beetle ticking in old furniture, and the forest spring tinkling over the stones. Sensitive lyrical poets always could hear these significant sounds of life and describe them in words. It is for the sound film to let them speak to us more directly from the screen.

Discovery of Noise

The sounds of our day-to-day life we hitherto perceived merely as a confused noise, as a formless mass of din, rather as an unmusical person may listen to a symphony; at best he may be able to distinguish the leading melody, the rest will fuse into a chaotic clamor. The sound film will teach us to analyze even chaotic noise with our ear and read the score of life's symphony. Our ear will hear the different voices in the general babble and distinguish their character as manifestations of individual life. It is an old maxim that art saves us from chaos. The arts differ from each other in the specific kind of chaos which they fight against. The vocation of the sound film is to redeem us from the chaos of shapeless noise by accepting it as expression, as significance, as meaning. . . .

Only when the sound film will have resolved noise into its elements, segregated individual, intimate voices, and made them speak to us separately in vocal, acoustic close-ups; when these isolated detail sounds will be collated again in purposeful order by sound -montage, will the sound film have become a new art. When the director will be able to lead our ear as he could once already lead our eye in the silent film and by means of such guidance along a series of close-ups will be able to emphasize, separate, and bring into relation with each other the sounds of life as he has done with its sights, then the rattle and clatter of life will no longer overwhelm us in a lifeless chaos of sound. The sound camera will intervene in this chaos of sound, form it and interpret it, and then it will again be man himself who speaks to us from the sound screen.

The Picture Forms the Sound

In a sound film there is no need to explain the sounds. We see together with the word the glance, the smile, the gesture, the whole chord of expression, the exact nuance. Together with the sounds and voices of things we see their physiognomy (the art of judging human character from facial features). The noise of a machine has a different coloring for us if we see the whirling machinery at the same time. The sound of a wave is different if we see its movement. Just as the shade and value of a color changes according to what other colors are next to it in a painting, so the timbre of a sound changes in accordance with the physiognomy or gesture of the visible source of the sound seen together with the sound itself in a sound film in which acoustic and optical impressions are equivalently linked together into a single picture.

Silence

Silence, too, is an acoustic effect, but only where sounds can be heard. The presentation of silence is one of the most specific dramatic effects of the sound film. No other art can reproduce silence, neither painting nor sculpture, ‘neither literature nor the silent film could do so. Even on the stage silence appears only rarely as a dramatic effect and then only for short moments. Radio plays cannot make us feel the depths of silence at all, because when no sounds come from our set, the whole performance has ceased, as we cannot see any silent continuation of the action. The sole material of the wireless play being sound, the result of the cessation of sound is not silence but just nothing.

Silence and Space

Things that we see as being different from each other appear even more different when they emit sounds. They all sound different when they do this, but they are all silent in the same way. There are thousands of different sounds and voices, but the substance of silence appears one and the same for all. That is at first hearing. Sound differentiates visible things; silence brings them closer to each other and makes them less dissimilar. Every painting shows this happy harmony, the hidden common language of mute things conversing with each other, recognizing each other's shapes, and entering into relations with each other in a composition common to them all. This was a great advantage the silent film had over the sound film. For its silence was not mute; it was given a voice in the background music, and landscapes and men and the objects surrounding them were shown on the screen against this common musical background. This made them speak a common silent language and we could feel their irrational conversation in the music, which was common to them all.

But the silent film could reproduce silence only by roundabout means. On the theatrical stage cessation of the dialogue does not touch off the great emotional experience of silence, because the space of the stage is too small for that, and the experience of silence is essentially a space experience

How do we perceive silence? By hearing nothing? That is a mere negative. Yet man has few experiences more positive than the experience of silence. Deaf people do not know what it is. But if a morning breeze blows the sound of a cock crowing over to us from the neighboring village, if from the top of a high mountain we hear the tapping of a woodcutter's axe far below in the valley, if we can hear the crack of a whip a mile away -then we are hearing the silence around us. We feel the silence when we can hear the most distant sound or the slightest rustle near us. Silence is when the buzzing of a fly on the windowpane fills the whole room with sound and the ticking of a clock smashes time into fragments with sledgehammer blows. The silence is greatest when we can hear very distant sounds in a very large space. The widest space is our own if we can hear right across it and the noise of the alien world reaches us from beyond its boundaries. A completely soundless space on the contrary never appears quite concrete, and quite real to our perception; we feel it to be weightless and unsubstantial, for what we merely see is only a vision. We accept seen space as real only when it contains sounds as well, for these give it the dimension of depth.

On the stage, a silence which is the reverse of speech may have a dramaturgical function, as for instance if a noisy company suddenly falls silent when a new character appears; but such a silence cannot last longer than a few seconds, otherwise it curdles as it were and seems to stop the performance. On the stage, the effect of silence cannot be drawn out or made to last.

In the film, silence can be extremely vivid and varied, for although it has no voice, it has very many expressions and gestures. A silent glance can speak volumes; its soundlessness makes it more expressive because the facial movements of a silent figure may explain the reason for the silence, make us feel its weight, its menace, its tension. In the film, silence does not halt action even for an instant and such silent action gives even silence a living face.

The physiognomy of men is more intense when they are silent. More than that, in silence even things drop their masks and seem to look at you with wide- open eyes. If a sound film shows us any object surrounded by the noises of everyday life and then suddenly cuts out all sound and brings it up to us in isolated close-up, then the physiognomy of that object takes on a significance and tension that seems to provoke and invite the event which is to follow.

Sound- Explaining Pictures

Not only the micro dramatics expressed in the micro physiognomy of the face can be made intelligible by the sound which causes it. Such a close- up -plus sound can have the inverse effect. The close-up of a listener's face can explain the sound he hears. We might perhaps not have noticed the significance of some sound or noise if we had not seen its effect in the mirror of a human face. For instance we hear the screaming of a siren. Such a sound does not acquire a dramatic significance unless we can see from the expression on human faces that it is a danger signal, or a call to revolt. We may hear the sound of sobbing, but how deep its meaning is will become evident only from the expression of sympathy and understanding appearing on some human face. Further, the acoustic character of a sound we understand is different too. We hear the sound of a siren differently if we know that it is a warning of impending deadly peril.

The face of a man listening to music may also show two kinds of things. The reflected effect of the music may throw light into the human soul; it may also throw light on the music itself and suggest by means of the listener's facial expression some experience touched off by this musical effect. If the director shows us a close-up of the conductor while an invisible orchestra is playing, not only can the character of the music be made clear by the dumb show of the conductor, his facial expression may also give an interpretation of the sounds and convey it to us. And the emotion produced in a human being by music and demonstrated by a close -up of a face can enhance the power of a piece of music in our eyes far more than any added decibels.

Asynchronous Sound

In a close-up in which the surroundings are not visible, a sound that seeps into the shot sometimes impresses us as mysterious, simply because we cannot see its source. It produces the tension arising from curiosity and expectation. Sometimes the audience does not know what the sound is they hear, but the character in the film can hear it, turn his face toward the sound, and see its source before the audience does. This handling of picture and sound provides rich opportunities for effects of tension and surprise.

Asynchronous sound (that is, when there is discrepancy between the things heard and the things seen in the- film) can acquire considerable importance. If the sound or voice is not tied up with a picture of its source, it may grow beyond the dimensions of the latter. Then it is no longer the voice or sound of some chance thing, but appears as a pronouncement of universal validity. . . . The surest means by which a director can convey the pathos or symbolical significance of sound or voice is precisely to use it asynchronously.

Intimacy of Sound

Acoustic close- ups make us perceive sounds which are included in the accustomed noise of day -to-day life, but which we never hear as individual sounds because they are drowned in the general din. Possibly they even have an effect on us but this effect never becomes conscious. If a close- up picks out such a sound and thereby makes us aware of its effect, then at the same time its influence on the action will have been made manifest.

On the stage such things are impossible. If a theatrical producer wanted to direct the attention of the audience to a scarcely audible sigh, because that sigh expresses a turning -point in the action, then all the other actors in the same scene would have to be very quiet, or else the actor who is to breathe the sigh would have to be brought forward to the footlights. All this, however, would cause the sigh to lose its essential character, which is that it is shy and retiring and must remain scarcely audible. As in the silent film so in the sound film, scarcely perceptible, intimate things can be conveyed with all the secrecy of the unnoticed eavesdropper. Nothing need be silenced in order to demonstrate such sounds for all to hear -and they can yet be kept intimate. The general din can go on, it may even drown completely a sound like the soft piping of a mosquito, but we can get quite close to the source of the sound with the microphone and with our ear and hear it nevertheless.

Subtle associations and interrelations of thoughts and emotions can be conveyed by means of very low, soft sound effects. Such emotional or intellectual linkages can play a decisive dramaturgical part. They may be anything-the ticking of a clock in an empty room, a slow drip from a burst pipe, or the moaning of a little child in its sleep.

Sound cannot be Isolated

In such close -ups of sound we must be careful, however, to bear in mind the specific nature of sound which never permits sound to be isolated from its acoustic environment as a close -up shot can be isolated from its surroundings. For what is not within the film frame cannot be seen by us, even if it is immediately beside the things that are. Light or shadow can be thrown into the picture from outside and the outline of a shadow can betray to the spectator what is outside the frame but still in the same sector of space, although the picture will show only a shadow. In sound things are different. An acoustic environment inevitably encroaches on the close -up shot and what we hear in this case is not a shadow or a beam of light, but the sounds themselves, which can always be heard throughout the whole space of the picture, however small a section of that space is included in the close -up. Sounds cannot be blocked out.

Music played in a restaurant cannot be completely cut out if a special close- up of say two people softly talking together in a corner is to be shown. The band may not always be seen in the picture, but it will always be heard. Nor is there any need to silence the music altogether in order that we may hear the soft whispering of the two guests as if we were sitting in their immediate vicinity. The close -up will contain the whole acoustic atmosphere of the restaurant space. Thus we will hear not only the people talking, we will also hear in what relation their talking is to the sounds all round them. We will be able to place it in its acoustic environment.

Such sound- pictures are often used in the film for the purpose of creating an atmosphere. Just as the film can show visual landscapes, so it can show acoustic landscapes, a tonal milieu.

Educating the Ear

Our eye recognizes things even if it has seen them only once or twice. Sounds are much more difficult to recognize. We know far more visual forms than sound forms. We are used to finding our way about the world without the conscious assistance of our hearing. But without sight we are lost. Our ear, however, is not less sensitive, it is only less educated than our eye. Science tells us in fact that the ear can distinguish more delicate nuances than our eye. The number of sounds and noises a human ear can distinguish runs into many thousands-far more than the shades of color and degrees of light we can distinguish. There is however a considerable difference between perceiving a sound and identifying its source. We may be aware that we are hearing a different sound than before, without knowing to whom or what the sound belongs. We may have more difficulty in perceiving things visually, but we recognize them more easily once we have perceived them. Erdmann's experiments showed that the ear can distinguish innumerable shades and degrees in the noise of a large crowd, but at the same time it could not be stated with certainty whether the noise was that of a merry or an angry crowd.

There is a very considerable difference between our visual and acoustic education. One of the reasons for this is that we so often see without hearing. We see things from afar, through a windowpane, on pictures, on photographs. But we very rarely hear the sounds of nature and of life without seeing something. We are not accustomed therefore to draw conclusions about visual things from sounds we hear. This defective education of our hearing can be used for many surprising effects in the sound film. We hear a hiss in the darkness. A snake? A human face on the screen turns in terror toward the sound and the spectators tense in their seats. The camera, too, turns toward the sound. And behold the hiss is that of a kettle boiling on the gas-ring.

Such surprising disappointments may be tragic too. In such cases the slow approach and the slow recognition of the sound may cause a far more terrifying tension than the approach of something seen and therefore instantly recognized. The roar of an approaching flood or landslide, approaching cries of grief or terror which we discern and distinguish only gradually, impress us with the inevitability of an approaching catastrophe with almost irresistible intensity. These great possibilities of dramatic effect are due to the fact that such a slow and gradual process of recognition can symbolize the desperate resistance of the consciousness to understanding a reality which is already audible but which the consciousness is reluctant to accept.

Sounds Throw No Shadow

Auditive culture can be increased like any other and the sound film is very suitable to educate our ear. There are however definite limits to the possibilities of finding our way about the world purely by sound, without any visual impressions. The reason for this is that sounds throw no shadows-in other words that sounds cannot produce shapes in space. Things, which we see we must see side by side; if we do not, one of them, covers up the other so that it cannot be seen. Visual impressions do not blend with each other. Sounds are different; if several of them are present at the same time, they merge into one common composite sound. We can see the dimension of space and see a direction in it. But we cannot hear either dimension or direction. A quite unusual, rare sensitivity of ear, the so-called absolute-is required to distinguish the several sounds which make up a composite noise. But even a perfect

ear cannot discern their place in space, the direction of their source, if no visual impression is present to help.
It is one of the basic form -problems of the radio play that sound alone cannot represent space and hence cannot alone represent a stage.

Sounds have no Sides

It is difficult to localize sound and a film director must take this fact into account. If three people are talking together in a film and they are placed so that we cannot see the movements of their mouths and if they do not accompany their words by gestures, it is almost impossible to know which of them is talking, unless the voices are very different. For sounds cannot be beamed as precisely as light can be directed by a reflector. There are no such straight and concentrated sound beams as there are rays of light.

The shapes of visible things have several sides, right side and left side, front and back. Sound has no such aspects, a sound strip will not tell us from which side the shot was made.

Sound has a Space Coloring

Every natural sound reproduced by art on the stage or on the platform always takes on a false tone- coloring, for it always assumes the coloring of the space in which it is presented to the public and not of the space which it is supposed to reproduce. If we hear a storm, the howling of the wind, a clap of thunder, etc., on the stage we always hear in it the timbre proper to the stage not in the timbre proper to the forest, or ocean, or whatnot the scene is supposed to represent. If, say, a choir sings in a church on the stage, we cannot hear the unmistakable resonance of Gothic arches; for every sound bears the stamp of the space in which it is actually produced.

Every sound has a space -bound character of its own. The same sound sounds different in a small room, in a cellar, in a large empty hall, in a street, in a forest, or on the sea.

Every sound which is really produced somewhere must of necessity have some such space quality and this is a very important quality indeed if use is to be made of the sensual reproducing power of sound! It is this timbre local of sound which is necessarily always falsified on the theatrical stage. One of the most valuable artistic faculties of the microphone is that sounds shot at the point of origin are perpetuated by it and retain their original tonal coloring. A sound recorded in a cellar remains a cellar sound even if it is played back in a picture theater, just as a film shot preserves the viewpoint of the camera, whatever the spectator's viewpoint in the cinema auditorium may be. If the picture was taken from above, the spectators will see the object from above, even if they have to look upwards to the screen and not downwards. Just as our eye is identified with the camera lens, so our ear is identified with the microphone and we hear the sounds as the microphone originally heard them, irrespective of where the sound being shown and the sound reproduced. In this way, in the sound film, the fixed, immutable, permanent distance between spectator and actor is eliminated not only visually . . . but acoustically as well. Not only as spectators, but as listeners, too, we are transferred from our seats to the space in which the events depicted on the screen are taking place.

Lets look into some terms relating to film sound. Some of them you might have already heard or read about in your previous semesters.

ADR

ADR stand for "Automated" or "Automatic" Dialog Replacement.

Dialog that cannot be salvaged from production tracks must be re-recorded in a process called **looping** or **ADR**. Looping originally involved recording an actor who spoke lines in sync to "loops" of the image, which were played over and over along with matching lengths of recording tape. ADR, though faster, is still painstaking work.

An actor watches the image repeatedly while listening to the original production track on headphones as a guide. The actor then re-performs each line to match the wording and lip movements. Actors vary in their ability to achieve sync and to recapture the emotional tone of their performance.

Marion Brando likes to loop because he doesn't like to freeze a performance until he knows its final context. (People have said that one reason he mumbles is to make the production sound unusable so that he can make adjustments in looping.)

ADR is usually considered a necessary evil but there are moments when looping can be used not just for technical reasons but to add new character or interpretation to a shot. Just by altering a few key words or phrases an actor can change the emotional bent on a scene.

According to Randy Thom: The way ADR is treated and approached is symptomatic of how little respect sound gets in the movies. You march the actor into a cold sterile room and usually give them no time to get into the character or rehearse. They are expected to just start performing a few minutes after they walk into the room. The emphasis is almost always on getting the dialogue in sync instead of getting the right performance. Of course the majority of the actor's brain is then occupied with whether his lips are flapping at exactly the same rate as they were on the day that the camera was rolling. It is no wonder that most ADR is not very good. In the final mix, directors almost always prefer the production sound, even if it is noisy and distorted.

One of the things you do with ADR to make it sound more like production sound is to pitch it up. ADR is almost always delivered at a lower pitch because the actor doesn't have the energy he/she had on the set. In the excitement of the shooting set the actor tends to talk louder and higher. In an ADR session, the director typically has to push the actor to get them anywhere near the level of vocal performance that came from the set.

If the recording of ADR were treated more like shooting the movie it would almost certainly be better. Recording it in more authentic environments (instead of studios) tends to help the actors' performance enormously. Environmental noise is a problem whenever you record outside of a studio, but well worth the trade-off in my opinion

Ambience

Ambience pertains to the pervading atmosphere of a place. (Often more of a psychological, rather than technical description) Ambience is widely used as a synonym for ambient sound. Ambient sound consists of noises present i the environment.

In film and video sound production term Ambience usually means the background sound accompanying a scene. Ambience is used for background sounds..

1 Present in the original production recording (a better term for it is presence)

2 Deliberately added in sound-effects editing in order to provide an acoustic space around the rest of the dialog and sound effects.

In 'Silence of the lambs', when Agent Starling (Jodie Foster) is down with Lecter in the dungeon, there were animal screams and noises built into the ambience. (One element of the ambience is a guy screaming in pain. The screaming was processed, slowed down and played in reverse)

Subjective Ambience:

In the trial scene of 'Philadelphia' - instead of using reverb to a voice as the convention says for hallucinating - sound designer Ron Bochar used subjective ambience. He dropped the previous room tone and shifted the ambient sound. He also changed the spatial placement of the ambient sound - from left, right, and center speakers to surround speakers.

Ambience helps establish the scene and works editorially to support the picture editing by, for example, staying constant across a picture cut to indicate to the audience that no change of space has occurred, but rather only a simple picture edit. Conversely, if ambience changes abruptly at a picture cut, an indication is made to listener that the scene also has changed.

Characteristic sounds

A characteristic sound is what a sound should be according to someone's perception of the sound.

When a natural sound is manipulated to achieve a desired effect, it becomes characteristic.
To satisfy either personal dictates or production demands sound effects have to....

- be recognizable
- meet audience expectations

When directors are forced to choose between authenticity and dramatic impact, they will usually opt for the latter and use "dramatic license" as an excuse. (If the same dramatic results can be provided by natural sounds, so much the better)

Emotional realism

The same sound can serve both the physical and the emotional meaning. It is possible to superimpose the emotional realism over the physical of the scene. The sound track reflects the mood of the story and the characters feelings.

Peckinpah's Straw Dogs includes gunshots and shattering glass as well. Peckingpah even uses emotional realism to heighten character's primal feelings. At a church social off-frame sounds expresses a woman's emotions about to being in the same room as the men who raped her. The shrillness of children's noise making gives the church social a relentless disturbing quality.

In Francis Coppola's Tucker: The man and his dream, sound designer Richard Beggs was pressed to find rhythmic acoustic elements (tied to manufacturing and cars) that could be used dramatically.

The Air Compressor…

- starts up, makes a rhytmic pumping sound and goes for a while.
- shuts down with a very characteristic sound (when the tank reaches capacity)
- is quite (either until the pressure drops naturally or it is used with some tool)
- turns on with a characteristic sound
- starts up and goes for a while
- et cetera

Beggs used the sounds of the compressor in several scenes to create a sense of tension. Like in the scene where Tucker explodes after the cars designer Alex Tremulus is almost crushed by the car. The sounds of the compressor is used at the beginning of the scene. It stops at a very critical point. The fact that it stops helps to create ambience and mood that is dramatic in nature.

Emotional realism is a kind of "antromorphic mirror" placed in the diegesis. In the same manner as music does, the soundscape reflects the mood of the story and the characters feelings.

Establishing Sound

Sound that establishes, from the very beginning of a scene, the general character of the surroundings.

During the late and thirties, Hollywood typically used onscreen establishing sound (for example, traffic sounds accompanying a shot of Times Square), but regulary turned to offscreen establishing sound during the forties (for example, traffic sound accompanying a shot of a bedroom with shades pulled down)

Establishing sound is usally removed or regularly reduced in volume during dialogue, but may return in the form of reestablishing sound (sometimes but not sysytematically, accompanied by a reestablishing shot).

Hyper-Real Sound

Many sound recordings for film and television are over-emphatically stated, over-hyped, and exaggerated compared to sound in real life.

One reason for this is that there is typically so much competing sound at any given moment that each sound that must be heard has to be rather over-emphatically stated, just to read through the clutter. Heard in isolation, the recordings seem silly, over-hyped, but heard in context, the assume a more natural balance.

The elements that often best illustrate this effects recorded while watching a picture such as footsteps, and are often exaggerated from how they would be in reality, both in loudness and in intimacy.

While some of this exaggeration is due to the experience of practitioners finding that average sound playback systems obscure details, a good deal if the exaggerations still is desirable under the best playback conditions, simply because of the competition for other kinds of sound.

Natural Sounds

A natural sound is that of an actual source. Natural sounds are unadorned production sounds.

In film and television, many natural sounds do not meet everyone's expectations. When this happens, they are either replaced with more suitable sounds or the natural sound is layered (other sounds are added) to make it more desirable.

- The actual sound item for a .38 pistol will come off sounding like a "anemic" cap pistol.
- The "dramatic ricochet" with gradually decreasing pitch is extensively used in film and TV (instead of all other possible ricochet sounds)

Natural sounds was popular in the early days of radio and film. Our perception of natural sounds has become so influenced by ideology about sound that we are often disappointed with reality.

Point-of-Audition Sound

Sound identified by its physical characteristic (principally reduced volyme and increased reverb) as it might be heard by a character within the film.

Regualary used both to join spaces whose relationship cannot easily be presented in a single establishing shot, and to promote identification between the audience and carefully selected characters.

Unlike the ponit-of-view sequence, which often moves from the viewer to the object or character viewed, the point-of-audition squence typically begins with a shot of the sound source, introducing point-of-audition sound when we cut to a shot of the auditor

Room Tone

(other terms are Presence and Atmosphere)
A location´s "aural fingerprint" - nonspecific sounds on the upper end (somewhere between 2 000 and 8 000 Hz)
Each room has a distinct presence of subtle sounds created by the movement of air particles in a particular volume. A microphone placed in two different empty rooms will produce different room tone for each.

- Room tone is recorded during 'production sound recording'
- Room tone is used to match the production sound track so that it may be intercut with the track and provide a continuous-sounding background.
- Room ton may smooth out edit points and give a feeling of life in a sound-deadened studio. The soundtrack "going dead" would be perceived by the audience not as silence but as a failure of the sound system.

Sound Designer

Sound designer is a illusive term with different meaning:

1. **"Sound designer"** are an artist who are brought on staff during the planning stages of a film, along with the set and costume designers, and who do their own mixing. The sound designer works with the director to shape an overall, consistent soundtrack that exploits the expressive possibilities of the sound medium. The over all sound design is organically related to the narrative and thematic needs of the film, and has integrity not possible if sound is divided among an entire bureaucracy. In Jurassic Park, Gary Rydstrom first designed the sounds of the dinosaurs and then models were built to match those roars.
2. **"Sound designer"** can also refer to a person brought on to create just one kind of effect. Ron Bochar was hired late in the postproduction of Wolf just to create the effects that accompanied Nicholson turning into a beast

Soundscape

The characteristic types of sound commonly heard in a given period or location.

For example, the late nineteenth-century American soundscape was largely limited to unamplified, live sounds, while the soundscape of the mid twenties included radio, electrically recorded disks, and public address, as well as live music, theater, and an increasing number of unmuffled motors.

In much of the world, today soundscape is characterized by competition among multiple amplified sounds, along with attempts (like the Walkman and acoustic panels) to restore individual aural autonomy in sound micro-atmospheres. Rural, maritime, and Third World soundscape of course offer their own particularities, as do early morning and late evening soundscapes.

Sweeten

Audio sweetening is a "catchall" phrase for fine-tuning sound in postproduction.

Sweeten/sweetener refer to subtly mixing an additional sound to a pre existing sound to "sweeten" the pre-existing sound. (As in sweets, sugar.)

For example in the film "Backdraft", the sounds of wolves howling were faintly added to the sound of fire crackling to sweeten and give it a haunting, animal like quality to it.

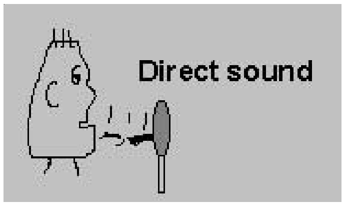
Sound Loop

A continuous loop of recorded sound, designed to prove background sound throughout a scene, often in semi-sync manner

Sound Hermeneutic

[Sound motif]

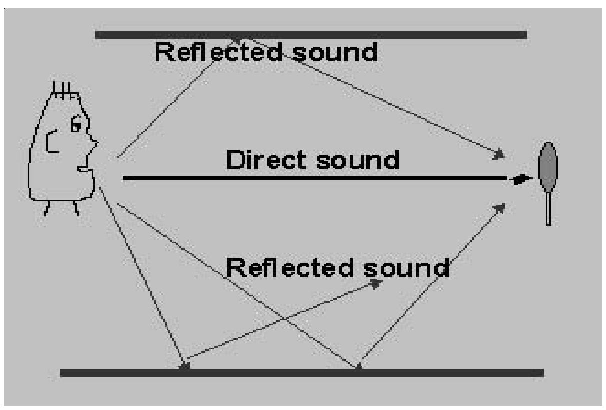
Cinema sound typically asks the question: "Where [does this sound come from]?" Visually identifying the source of the sound, the image usually responds: " Here!" The sound hermeneutic is full question and answer process, as followed by the film spectator/auditor



Sound Motif

A sound effect or combination of sound effects that are associated with a particular character, setting, situation or idea through the film.

The sound motifs condition the audience emotionally for the intervention, arrival, or actions of a particular character. The



sound motifs can be very useful in the rough cut, where they help clarify the narrative functions of the characters and provide a sound association for those characters as we move through the story.

The use of sound motifs can help shape a story that requires many characters and many locations and help unify the film and sustain its narrative and thematic development.

Direct Sound and Reflected Sound

[Sound Perspective]

Direct sound issues from the source itself, such as those frequencies coming from an actors mouth. When a person is close to us, we hear essentially direct sound including low-frequency chest tones. As the person moves farther away, we hear more of the reflected sound. Unfortunatly "direct sound" is also synonym for the original production sound

Reflected sound is produced by the direct sound bouncing off the walls, floor etc. Reflected sound s much more complex in character than direct sound because the surfaces are at different distances from the source and have widely varying reflective properties. Interiors that contains a lot of hard surfaces - glass, stone, metal, etc. - are said to be "live" because their high reflectivity. Soft or porous materials, like carpeting, draperies and upholstered furniture, are sound deadening. As furniture is moved into a empty room, the acoustics became "dead".

Sound Perspective

Sound perspective refers to the apparent distance of a sound. Clues to the distance of the source include the volume of the sound, the balance with other sounds, the frequency range (high frequencies may be lost at a distance), and the amount of echo and reverberation.

A closer sound perspective may sometimes be simulated by recording with a directional microphone which rejects sound from other directions. A more distant perspective may sometimes be simulated in post-production by processing the sound and mixing in other sounds.

In recording sound for film, you usually select a sound perspective to match the picture with which it will be used. Compare these examples:

Close perspective sound contains a high ratio of direct sound to reflected sound

Distant perspective sound contains a high ratio of **reflected sound** to direct sound

Notes :

GLOSSARY

A

acoustical phase: The time relationship between two or more sound waves at a given point in their cycles.
acoustics: The science that deals with the behavior of sound and sound control. The properties of a room that affect quality of sound.
active combining network (ACN): An amplifier at which the outputs of two or more signal paths are mixed together before being routed to their destination.
ADAT: One of two formats used in modular digital multitrack tape recorders. It uses SVHS videocassette tape. ADAT stands for Alesis Digital Audio Tape recorder. See also **digital tape recording system.**
additive ambience: When the ambience of each track becomes cumulative in mixing a multitrack recording.
adhesion: One layer of audiotape sticking to another.
ADR: See automated dialogue replacement.
ADSR :See sound envelope.
AES/EBU interface: Internationally accepted professional digital audio interface transmitted via a balancedline connection using XLR connectors, specified jointly by the Audio Engineering Society and the European Broadcast Union. See also **SPDIF.**
ambience : Sounds such as reverberation, noise, and atmosphere that form a background to the main sound. Also called room tone and presence, and atmos in Great Britain.
amplifier: A device that increases the amplitude of an electric signal.
amplitude: The magnitude of a sound wave or electrical signal, measured in decibels.
amplitude processor: A signal processor that affects a signal's loudness.
analog recording: A method of recording in which the waveform of the recorded signal resembles the waveform of the original signal.
anechoic chamber : room that prevents all reflected sound through the dissipation or the absorption of sound waves.
assemble editing: Dubbing segments from one tape or tapes to another tape in sequential order.
atmos : Short for atmosphere, the British term for ambience. See **ambience.**
attack: (1) The way a sound begins—that is, by plucking, bowing, striking, blowing, and so on. (2) The first part of the sound envelope.
attack time : The length of time it takes a limiter or compressor to respond to the input signal.
audio leading video: When the sound of the incoming scene starts before the corresponding picture appears. See also **videoleadingaudio.**
automated dialogue replacement (ADR): A technique used to rerecord dialogue in synchronization with picture in postproduction. The picture is automatically replayed in short "loops" again and again so that the performers can synchronize their lip movements with the lip movements in the picture and then record the dialogue. Also known as automatic dialog recording and looping See also **dialogue recording studio.**
azimuth: Alignment of the record and playback heads so that their centerlines are parallel to each other and at right angles to the direction of the tape motion passing across the heads.

B

backtiming: Method of subtracting the time of a program segment from the total time of a program so that the segment and the program end simultaneously.
balanced line: A pair of ungrounded conductors whose voltages are opposite in polarity but equal in magnitude.
bandpass filter: A filter that attenuates above and below a selected bandwidth, allowing the frequencies between to pass.
bandwidth curve: The curve shaped by the number of frequencies in a bandwidth aud their relative increase or decrease in level. A bandwidth of 100 to 150 Hz with 125 Hz boosted 15 dB forms a sharp, narrow bandwidth curve; a bandwidth of 100 to 6,400 Hz with a 15dB boost at 1,200 Hz forms a more sloping, wider bandwidth curve.
bandwidth: The difference between the upper and lower frequency limits of an audio component. The upper and lower frequency limits of AM radio are 535 and 1,605 kHz; therefore, the bandwidth of AM radio is 1,070 kHz.
bass: The low range of the audible frequency spectrum; usually from 20 to 320 Hz.
bass rolloff : Attenuating bass frequencies. The control- for example, on a microphone-used to roll off bass frequencies.
bass tipup: See proximity effect.
bias: The inaudible DC or AC signal added to an audio signal to overcome nonlinearities of amplification or of the medium. In magnetictape recording, ultrasonic AC bias is used to linearize the tape medium, which would otherwise be highly distorted.
bias current : An extremely highfrequency AC current, far beyond audibility, added during a tape recording to linearize the magnetic information.
bidirectional microphone: A microphone that picks up sound to its front and back and has minimal pickup at its sides.
binaural hearing : Hearing with two ears attached to and separated by the head.
binaural microphone head : Two omnidirectional capacitor microphones set into the ear cavities of an art)ficial head, complete with pinnas. This arrangement preserves binaural localization cues during recording and reproduces sound as humans hear it, threedimensionallY Also called artificial head or dummy head stereo.

blast filter: See pop **filter.**
blocking: Plotting performer, camera, and microphone placements and movements in a production. **board** Audio mixing console.
boundary microphone: A microphone whose capsule is mounted flush with or dose to, but a precise distance from, a reflective surface so that there is no phase cancellation of reflected sound at audible frequencies.
bulk eraser :A demagnetizer used to erase an entire roll of magnetic tape without removing it from its reel. Also known as a degausser.
bus :A mixing network that combines the outputs of other channels.

C

calibration: Adjusting equipment components-for example, a console and a tape recorder-according to a standard so that their measurements are similar. See also **electronic alignment.**
camcorder: A handheld video camera with a builtin or dockable videotape recorder.
capacitor microphone :A microphone that transduces acoustic energy into electric energy electrostatically.
capstan: The shaft that rotates against the tape, pulling it across the heads at a constant speed.
cardioid microphone :A unidirectional microphone with a heartshaped pickup pattern.
CDR: See **recordable compact disc.**
CDrewritable (CDRW): A CD format that, as with tape, can be recorded on, erased, and used again for another recording.
CDRW :See **CDrewritable.**
center frequency: In peak/dip equalizing, the frequency at which maximum boost or attenuation occurs.
chorus effect: Recirculating the doubling effect to make one sound source sound like several. See also **doubling.**
cinching: Slippage between the tape layers due to loose packing. Also known as windowing.
clapslate: A slate used in synchronizing sound and picture during filming and editing. The slate carries information such as scene and take number, production title, location of shot-e.g., indoors or outdoors-and time code. A pair of hinged boards on top of the slate-called clapsticks- clap together, producing the sound that is used to synchronize picture and sound.
clipping: Audible distortion that occurs when a signal's level exceeds the limits of a particular device or circuit.
close miking: Placing a microphone close to a sound source to pick up mostly direct sound and reduce ambience and leakage. See also **distant miking.**
coercivity: The magnetic force field necessary to reduce a tape from saturation to full erasure. This value is expressed in oersteds.
coincident miking: Employing two matched microphones, usually unidirectional, crossed one above the other on a vertical axis with their diaphragms. See also XY miking.
combfilter effect; The effect produced when a signal is timedelayed and added to itself, reinforcing some frequencies and canceling others, giving sound an unnatural, hollow coloration.
commentative sound: Descriptive sound that makes a comment or interpretation. See also **descriptive sound and narrative sound.**
compander: A contraction of the words compressor and expander that refers to the devices that compress an input signal and expand an output signal to reduce noise. Also known as a noise reducer.
omplementary equalization : Equalizing sounds that share similar frequency ranges so that they complement, rather than interfere with, one another.
compression: (1) Reducing a signal's output level in relation to its input level to reduce dynamic range. (2) The drawing together of vibrating molecules, thu'sproducing a highpressure area. See also **rarefaction. compression ratio** The ratio of input and output signals in a compressor.
compression threshold: The level at which a compressor acts on an input signal and the compression ratio takes effect.
compressor: A signal processor with an output level that decreases as its input level increases.
condenser microphon:e See capacitor microphone.
console: An electronic device that amplifies, processes, and combines input signals and routes them to broadcast or recording.
constructive interference: When sound waves are partially **out of phase** and partially additive, increasing amplitude where compression and rarefaction occur at the same time.
contact microphone: A microphone that attaches to a sound source and transduces the vibrations that pass through it. Also called acousfic pickup mic.
contextual sound: Sound that emanates from and duplicates a sound source as it is. See also **diegetic sound.**
contrapuntal narration: Juxtaposes narration and action to make a statement not carried by either element alone.
coverage angle: The offaxis angle or point at which loud-speaker level is down 6 dB compared with the onaxis output level.
cps: See hertz.
crossfade: Fading in one sound source as another sound source fades out. At some point the sounds cross at an equal level of loudness.
crossover frequency: The frequency at which the high frequencies are routed to the tweeter(s) and the low frequencies are routed to the woofer(s).
crossover network: An electronic device that divides the audio spectrum into individual frequency ranges (low, high, and/or mid) before sending them to specialized loudspeakers such as the woofer(s) and tweeter(s).
crosstalk: Unwanted signal leakage ffom one signal path to another.
crystal synchronization: Synchronizing the operating speeds of a film camera and an audiotape recorder by using a crystal oscillator in both camera and recorder. The oscillator generates a sync pulse tone. See also **doublesystem** recording**.**
cupping: Deformation of the backing of audiotape due to expansion of the magnetic coating and base.
curling: Twisting of audiotape when it hangs due to a problem in the binding between the plastic and magnetic coatings.

cut: (1) An instantaneous transition from one sound or picture to another. (2) To make a disc recording. (3) A decrease in level.
cut and splice editing: Editing tape or film by physically cutting the material and joining the cut ends with splicing tape.
cycles per second (cps) :See hertz.

D

DASH format :See Digital Audio Stationary Head format.
DAT: Digital audiotape.
DAW: See **digital audio worktation.**
dB: See decibel.
dBm: An electrical measurement of power referenced to 1 milliwatt as dissipated across a 600ohm load.
dBSPL A measure of the pressure of a sound wave, expressed in decibels (dB).
dBu: A unit of measurement for expressing the relationship of decibels to voltage—0.775 volt.
dBv: See dBu.
dBV: A measure of voltage with decibels referenced to 1 volt.
DCA: See **digitally controlled amplifler.**
deadpotting: Starting a recording with the fader turned down all the way. Also known as dead rolling.
decay time: See reverberation time.
decibel (dB): A relative and dimensionless unit to **measure** the ratio of two quantities.
deeaser: A compressor that reduces sibilance.
degausser: See bulk eraser.
delay: The time interval between a sound or signal and each of its repeats.
descriptive sound: Describes sonic aspects of a scene not connected to the main action. See also **commentative sound and narrative sound.**
destructive editing : Permanently alters the original sound or soundfile. See also **nondestructive editing.**
destructive interference: When sound waves are partially out of phase and partially subtractive, decreasing amplitude where compression and rarefaction occur at different times.
dialogue recording studio: A studio in which dialogue is recorded and synchronized to picture. See also **automated dialogue replacement.**
diaphragmatic absorber: A flexible panel mounted over an air space that resonates at a frequency (or frequencies) determined by the stiffness of the panel and the size of the air space. Also called bass trap.
diegetic sound: Sound that comes from within the story space, such as dialogue and sound ‘effects. See also **nondiegetic sound.** diffraction The spreading or bending around of sound waves as they pass an object.
diffusion: The scattering of sound waves.
Digital Audio Stationary Head (DASH) format :A format agreed to by Sony, Studer, and TASCAM to standardize digital recording.
digital audio workstation (DAW): A multifunctional harddisk production system, controlled from a central location, that is integrated with and capable of being networked to other devices, such as audio, video, and MIDI sources, within or among facilities.
digital cartridge disk system: An audio recorder and/or playback system that uses compact, magnetooptical disc, mini disc, floppy disk, or hard disk as the recording medium.
digital delay: An electronic device designed to delay an audio signal.
digital news gathering (DNG): Reporting and gathering news from the field using digital equipment.
digital recording: A method of recording in which samples of the original analog signal are encoded on tape as pulses and then decoded during playback.
digital signal processing (DSP): A software program that provides various manipulations of sound in digital format using complex algorithms.
digital tape recording system (DTRS): One of two formats used in modular digital multitrack tape recorders. It uses Hi8 videocassette tape. **See also ADAT.**
digital versatile disc (DVD): A compact disc providing massive data storage of digitalquality audio, video, and text.
digitally controlled amplifier (DCA): An amplifier whose gain is remotely controlled by a digital control signal.
directional microphone: Any microphone that picks up sound from one direction. Also’called unidirectional microphone.
direct narration: Describes what is being seen or heard.
direct sound: Sound waves that reach the listener before reflecting off any surface. See also **early reflections.**
distant miking : Placing a microphone(s) far enough from the sound source to pick up most or all of an ensemble’s blended sound including room reflections. See also **close miking.**
distortion: The appearance of a signal in the reproduced sound that was not in the original sound. See also **harmonic distortion, intermodulation distortion, loudness distortion, and transient distortion.**
diversity reception: Multiple antenna receiving system for use with wireless microphones. See also **nondiversity receiver.**
DNG: See digital news gathering.
donut: An announcement in which music is established faded under the announcer, and reestablished after the announcer finishes reading the the copy.
Doppler effect: The perceived increase or decrease in frequency as a sound source moves closer to or farther from the listener.
doublesystem recording: Filming sound and picture simultaneously but separately with a camera and a recorder. See also **crystal synchronization.**
doubling: Mixing slightly delayed signals (15 to 35 ms) with the original signal to create a fuller, stronger, more ambient sound. See also **chorus effect.**
dropout :(1) A sudden attenuation of sound or loss of picture due to an imperfection in the magnetic coating. (2) Sudden attenuation in a wireless microphone signal due to an obstruction or some other interference.
dry sound: A sound devoid of reverberation. See also **wet sound.**
DSP: See digital signal processing.
DTRS: See digital tape recording system.
dub:Transferring sound from tape or disk to another tape or disk.
DVD: See **digital versatile disc.**

dynamic microphone: A microphone that transduces energy electromagnetically. Movingcoil and ribbon microphones are dynamic.
dynamic range: The range between the quietest and loudest sounds a sound source can produce without distortion.

E

early reflections : Reflections of the original sound that reach the listener within about 40 to 50 ms of the direct sound. Also called early sound. See also **direct sound.**
early sound:. See **early reflections.**
echo: Sound reflections delayed by 35 ms or more that are perceived as discrete repetitions of the direct sound.
edit decision list (EDL) :A list of edits, computergenerated or handwritten, used to assemble a production.
EDL :See **edit decision list.**
EFP: See **electronic field production.**
eigentones : The resonance of sound at particular frequencies in an acoustic space. May add unwanted coloration to sound. More commonly known as room modes.
elasticity: The capacity to return to the original shape or place after deflection or displacement.
electret microphone :A capacitor microphone which, instead of requiring an external highvoltage power source, uses a permanently charged element and requires only a lowvoltage power supply for the internal preamp.
electroacoustics : The electrical manipulation of acoustics.
electronic alignment: The adjustment of electronic and mechanical characteristics of a tape recorder to a defined standard specified by the manufacturer or by international industry bodies such as the Audio Engineering Society (AES), the National Association of Broadcasters (NAB), or the International Electrotechnical Commission (IEC). See also **calibration.**
electronic editing: Using one tape recorder and inserting-punching in-material, or transferring material from one tape recorder (the master) to another (the slave).
electronic field production (EFP): Video production done on location, involving program materials that take some time to produce.and reamplified. Electronic feedback is created in digital delay devices by feeding a proportion of the delayed signal back into the delay line. Also called regeneration.
electronic news gathering (ENG): News production done on location, sometimes taped and sometimes live, but usually with an imminent deadline.
ENG: See **electronic news gathering**
enharmonic: In music, two different notes that sound the same, for example, C#and Db G# and Ab.
EQ Equalization:. See **equalizer.**
equalizer : A signalprocessing device that can boost, attenuate, or shelve frequencies in a sound source or sound system.
equal loudness principle : The principle that confirms the human ear's nonlinear sensitivity to all audible frequencies: that midrange frequencies are perceived with greatest intensity and that bass and treble frequencies are perceived with lesser intensity.
erase head: Electromagnetic transducer on a tape recorder that automatically demagnetizes a tape before it reaches the record head when the recorder is in the record mode.
ergonomics: Designing an engineering system with human comfort and convenience in mind.
expander: An amplifier in which the output signal's dynamic

F

fadein: Gradually increasing the loudness of a signal level from silence (or hom "black" in video).
fadeout: Gradually decreasing the loudness of a signal level to silence (or to "black" in video).
fadeout/fadein: A transition usually indicating a marked change in time, locale, continuity of action, and other features.
fader: A device containing a resistor that is used to vary the output voltage of a circuit or component. Also known as an attenuator, a gain or volume control, or a pot or potentiom-eter.
feedback: When part or all of a system's output signal is resumed into its own input. Feedback can be acoustic or electronic. A commonly encountered example of acoustic feedback is the loud squeal or howl caused when the sound from a loudspeaker is picked up by a nearby microphone
fill leader: Old picture film or unrecorded magnetic film used to fill silences on a magneticfilm recording thereby reducing the noise of recorded magnetic film. Also called spacer.
filter: A device that removes unwanted frequencies or noise from a signal.
fixedfrequency equalizer: An equalizer with several fixed frequencies usually grouped in two (high and low) or three (high, middle, and low) ranges of the frequency spectrum.
flanging: Combining a direct signal and the same signal slightly delayed, and continuously varying their time
flat: Frequency response in an audio system that reproduces a signal between 20 and 20,000 Hz (or between any two specified frequencies) that varies no more than + 0r 3 dB.
flutter: Frequency changes in an analog tape recording resulting from faster variations in the speed of the tape transport. See also wow.
flutter echoes: Echoes between parallel walls that occur in rapid series.
FM microphone: Wireless microphone.
foldback : The system in a multichannel console that permits the routing of sound through a headphone monitor feed to performers in the studio.
Foley recording: Producing and recording sound effects in the studio in synchronization with picture.
formant: The resonance band in a vibrating body that mildly increases the level of specific steadystate frequencies in that band.
fourway system loudspeaker: A loudspeaker that uses three crossover frequencies to divide the bass, midrange, and treble ranges.
frame rate: The number of film frames that pass in one second of real time-frames per second (fps).
freewheel: A mode in a synchronizer that allows stretches of poorly encoded time code to be passed over without altering the speed of the slave tape recorder's ransport.
frequency: The number of times per second that a sound source vibrates. Now expressed in hertz (Hz); formerly expressed in cycles per second (cps).

frequency response: A measure of an audio system's ability to reproduce a range of frequencies with the same relative loudness; usually represented by a graph.
full coat: Magnetic film in which the oxide coating covers most or all of the film width. See also stripe **coat.**
fundamental: The lowest frequency a sound source can produce. Also called primary frequency and first harmonic.

G

gauss: A unit of magnetic density.
graphic equalizer: An equalizer with sliding controls that gives a graphic representation of the response curve chosen.
guard band: The space between tracks on an audiotape recorder head to reduce crosstalk.

H

Haas effect: See precedence effect.
harddisk recording: Using a harddisk computer system as the recording medium, which is more versatile than tape because data storage and retrieval is random, quick, and nonlinear; storage capacity is far greater; and data is nondestructive.
hardwired: Description of pieces of equipment wired to each other. See also **patch bay.**
harmonic distortion: Nonlinear distortion caused when an audio system introduces harmonics to a signal at the output that were not present at the input.
harmonics: Frequencies that are multiples of the fundamental.
headroom: The amount of increase in loudness level that a tape, amplifier, or other piece of equipment can take, above working level, before overload distortion.
headset microphone: A microphone attached to a pair of headphones; one headphone channel feeds the program and the other headphone channel feeds the director's cues.
headstack: A multitrack tape head.
height: One of the adjustments made when aligning the heads on an audiotape recorder. This adjustment aligns the height of the heads with the recording tape.
helical scanning: Using one or more rotating heads that engage the tape wrapped at least partially around the head drum.
Helmholtz absorber: A resonator designed to absorb specific frequencies depending on size, shape, and enclosed volume of air. The enclosed volume of air is connected to the air in the room by a narrow opening or neck. When resonant frequencies reach the neck of the enclosure, the air inside cancels those frequencies. Also called Helmholtz resonator.
humbuck circuit :A circuit built into a microphone to reduce hum pickup.
hertz (Hz): Unit of measurement of frequency; numerically equal to cycles per second (cps).
high end: The treble range of the frequency spectrum.
highoutput/ape: Highsensitivity tape.
highpass (lowcut) filter: A filter that attenuates frequencies below a selected frequency and allows those above that point to pass.
Hz: See hertz.

I

IEC standard: The time code standard for RDAT recording, established by the International Electrotechnical Commission.
IFB : See interruptible foldback system.
IM :See intermoduiation distortion.
impedance: The measure of the total resistance to the current flow in an AC circuit; expressed in ohms.
indirect narration: Describes something other than what is being seen or heard.
indirect sound : Sound waves that reflect hom one or more surfaces before reaching the listener.
infrasonic: The range below the frequencies audible to human hearing.
inharmonic overtones: Pitches that are not exact multiples of the fundamental.
initial decay: In the sound envelope, the point at which the attack begins to lose amplitude.
inline console: A console in which a channel's input, output, and monitor functions are placed inline and located in a single input/output (1/0) module. See also **splitsection console and input/output (1/0) module.**
inner ear: The part of the ear that contains the auditory nerve, which transmits sound waves to the brain.
input/output (1/0) module: On an inline console, a module containing input, output, and monitor controls for a single channel
input section: On a console, the section into which signals from a sound source, such as a microphone, feed and are then routed to the output section.
insert editing: In electronic editing, inserting a segment between two previously dubbed segments. Also, electronic editing segments out of sequential order.
Integrated Services Digital Network (ISDN): A public telephone service that allows inexpensive use of a flexible, widearea, alldigital network for, among other things, recording simultaneously from various locations.
intermodulation distortion (IM): Nonlinear distortion that occurs when different frequencies pass through an amplifier at the same time and interact to create combinations of tones unrelated to the original sounds.
interruptible foldback (IFB) system :A communications system that allows communication from the producer or director and selected production personnel with the on air talent.
intheear monitoring: Using small headphones to feed the sound blend to onstage performers instead of stage monitors.
in the mud: Sound level so quiet that it barely "kicks" the VU meter.
in the red : Sound level so loud that the VU meter "rides" over 100 percent of modulation.
inverse square law: The acoustic situation in which the sound level changes in inverse proportion to the square of the distance from the sound source.
1/0 module: See input/output module.
ISDN :See Integrated Services Digital Network.

J

jack: Receptacle or plug connector leading to the input or output circuit of a patch bay, tape recorder, or other electronic component.

jam sync: A mode in a synchronizer that produces new time code during dubbing either to match the original time code or to regenerate new address data.

L

lavalier microphone: Microphone that used to be worn around the neck but is now wom attached to the clothing.
layback: Dubbing the composite audio track from the multitrack tape to the edited master videotape, or the dialogue, sound effects, and music tracks to separate reels of magnetic film. See also **layover and prelay.**
laydown: Seelayover.
layering: When many sounds occur at once, layering involves making sure that they remain balanced, in perspective, and intelligible in the mix.
layover: Dubbing the audio from the edited master videotape or audiotape, or both, to a multitrack recorder for premixing. Also called laydown. See also **layback and prelay.**
leader tape: Nonmagnetic tape spliced to the beginning and end of a tape and between segments to indicate visually when recorded material begins and ends.
limiter: A compressor with an output level that does not exceed a preset ceiling regardless of the input level.
linear editing: Nonrandom editing. See also **nonlinear editing.**
Iinearity: Having an output that varies in direct proportion to the input.
listening fatigue: A pronounced dulling of the auditory senses inhibiting perceptual judgment.
localization: (1) Placement of a sound source in the stereo or surroundsound frame. (2) The direction from which a sound source seems to emanate in a stereo or surroundsound field. (3) The ability to tell the direction from which a sound is coming.
longitudinal time code (LTC): A highfrequency signal consisting of a stream of pulses produced by a time code generator used to code tape to facilitate editing and synchronization. Also known as SMPTE t me code.
loudness distortion: Distortion that occurs when the loudness of a signal is greater than the sound system can handle. Also called ove load distortion.
low bass: Frequency range between roughly 20 and 80 Hz, the lowest two octaves in the audible hequency spectrum.
low end: The bass range of the frequency spectrum.
lowoutput tape: Lowsensitivity tape.
lowpass (highcut) filter: A filter that attenuates frequencies above a selected frequency and allows those below that point to pass.
LTC: See longitudinal time code.

M

magnetic film: Sprocketed film containing sound only and no picture. See also **full coat and stripe coat.**
magnetooptical (MO) recording: Discbased, optical recording medium that uses tiny magnetic particles heated to extremely high temperatures.
masking: The hiding of some sounds by other sounds when each is a different frequency and they are presented together.
master: (1) The original recording. (2) The final tape or disc recording that is sent to the CD mastering house or to distribution.
master fader: The fader that controls the combined signal level of the individual input channels on a console.
master section: In a multichannel production console, the section that routes the final mix to its recording destination.lt usually houses, at least, the master controls for the mixingbus outputs, reverb send and reverb return, and master fader.
maximum soundpressure level The level at which a microphone's output signal begins to distort, that is, produces a 3 percent total harmonic distortion THD).
MD: See mini disc.
MDM: See modular digital multitrack recorder.
microphone: A transducer that converts acoustic energy into electric energy. Also called mic.
middle ear: The part of the ear that transfers sound waves from the eardrum to the inner ear.
middleside (MS) microphone: Consists of two mic capsules housed in single casing. One capsule, usually cardioid, is the midposition microphone. The other capsule, usually bidirectional, has each lobe oriented 90 degrees laterally.
MIDI: See Musical Instrument Digital Interface.
MIDI time code (MTC): Translates SMPTE time code into MIDI messages that allow MlDIbased devices to operate on the SMPTE timing reference.
midrange: The part of the frequency spectrum to which humans are most sensitive; the frequencies between 320 and 5,120 Hz.
mil :Onethousandth of an inch.
milking the audience: Boosting the level of an audience's sound during laughter or applause and/or reinforcing it with recorded laughter or applause or applause.
mini disc ™ (MD): Magnetooptical disc 2.5inches wide that can store more than an hour of digitalquality audio.
minimic: Short for miniature microphone. Any extremely small lavalier microphone designed to be unobtrusive oncamera and which can be easily hidden in or under clothing or on a set.
mixminus: A program feed through an interruptible foldback (IFB) circuit minus the announcer's voice. See also **interruptible foldback system.**
mixdown : The point, usually in postproduction, when all the separately recorded audio tracks are sweetened, positioned, and combined into stereo or surround sound.
mixer: A small, highly portable device that mixes various elements of sound, typically coming from multiple microphones, and performs limited processing functions.
MO: See **magnetooptical recording.**
mobile unit: A car, van, or tractortrailer equipped to produce program material on location.
modular digital multitrack (MDM) **recorder:** An audiotape recorder that uses a videocassette transport with videocassette tape. It can record up to eight channels and, linked to multiple MDMs, can expand track capability in eightchannel increments.
monitor section: The section in a console that enables the signals to be heard. The monitor section in multichannel production consoles, among other things, allows monitoring of the line or recorder input, selects various inputs to the control room and studio monitors, and controls their levels.
movingcoil loudspeaker: A loudspeaker with a movingcoil element.

MS microphone: See middleside microphone.
MTC: See MIDI **time code**
mult: See multiple.
multidirectional microphone: Microphone with more than one pickup pattern. Also called polydirectional microphone.
multipath: In wireless microphones, when more than one radio frequency (RF) signal from the same source arrives at the receiver's front end, creating phase mismatching.
multiple: (1) On a patch bay, jacks interconnected to each other and to no other circuit. They can be used to feed signals to and from sound sources. Also called mults. (2) An amplifier with several mic level outputs to provide individual feeds, thereby eliminating the need for many. Also called a press bridge or a presidenhal patch.
multipleentryport microphone: A microphone that has more than one opening for sound waves to reach the transducer. Most of these openings are used to reject sound from the sides or back of the microphone through phase cancellation. Each port returns a different frequency range to the mic capsule out of phase with sounds reaching the front of the mic.
Musical Instrument Digital Interface (MIDI:) A protocol that allows synthesizers, drum machines, sequencers, and other signalprocessing devices to communicate with or control one another, or bot

N

NC: See **noise criteria.**
nearcoincident miking: A stereo microphone array in which the mics are separated horizontally but the angle or space between their capsules is not more than several inches. See also XY miking.
nearfield monitoring : Monitoring with loudspeakers placed close to the operator, usually on or just behind the console's meter bridge, to reduce interference from control room acoustics at the monitoring position.
noise: Any unwanted sound or signal.
noise criteria (NC): Contours of the levels of background noise that can be tolerated within an audio studio.
noise gate: An expander with a threshold that can be set to reduce or eliminate unwanted lowlevel sounds, such as room ambience, rumble, and leakage, without affecting the wanted sounds.
noise processor: A signal processor that reduces tape noise.
noisecanceling microphone: A microphone designed for use close to the mouth and with excellent rejection of ambient sound.
nondestructive editing: Editing that does not alter the orlgnal sound or soundfile, regardless of what edlting or.signal processing is effected. See also **destructive editing.**
nondiegetic sound: Sound that is outside the story space, such as music underscoring. See also **diegetic sound.**
nondirectional microphone: See omnidirectional microphone.
nondiversity receiver: Singleantenna receiving system used with wireless microphones. See also **diversity reception.**
nonlinear: The property of not being linear-not having an output that varies in direce proportion to the input.
nonlinear editing: Instant random access to and easy rearrangement of recorded material. See also **linear editing.**
notch filter: A filter capable of attenuating an extremely narrow bandwidth of frequencies.

O

octave: The interval between two sounds that have a frequency ratio of 2 to 1. **oersted** A unit of magnetic force.
Offmic: Not being within the optimal pickup pattern of a microphone; offaxis.
offmiking: Miking technique that employs microphon farther from the sound source than the close mics to add moreambient, airier sound to the overall recording.
ohm:A unit of resistance to current flow.
Omnidirectional microphone: Microphone that picks up sound from all directions. Also called a nondirectional microphone.
onmic: Being within the optimal pickup pattern of a microphone; onaxis.
openreel audiotape recorder: A tape recorder with the feed reel and takeup reel not enclosed in a cartridge, requiring that they be mounted manually
oscillator: A device that generates pure tones or sine waves.
Outer ear: The portion of the ear that picks up and directs sound waves through the auditory canal to the middle ear.
Output section: In a mixer and console, the section that routes the signals to a recorder or broadcast, or both.
overroll: Recording ambience after recording narration or dialogue by letting the recorder continue to run.
overdubbing: Recording new material on a separate tape track(s) while listening to the replay of a previously recorded tape track(s) in order to synchronize the old and new material.
overload Feeding: a component or system more amplitude than it can handle and thereby causing overload distortion.
overload distortion: See loudness distortion.
Overload indicator: On a console, a lightemitting diode (LED) that flashes when the input signal is approaching or has reached overload and is clipping.
overtones: Harmonics that mayor may notbe multiples of the fundamental. Subjective response of the ear to harmonics.pad An attenuator inserted into a component or system to reduce level.

P

pan pot: A volume control that shifts the proportion of sound to any point from left to right between two output buses and, hence, between the two loudspeakers necessary for reproducing a stereo image. Pan pot is short for panoramic potentiometer.
parabolic microphone system: A system that uses a concave dish to focus reflected sound into a microphone pointed at the center of the dish.
paragraphic equalizer: An equalizer that combines the features of a parametric and a graphic equalizer.
parametric equalizer: An equalizer in which the bandwidth of a selected frequency is continuously variable.
patch bay: An assembly of jacks to which are wired the inputs and outputs of the audio components in a console and/or sound studio. Also called patch panel. See also **hardwired.**
patch cord: A short cord or cable with a plug at each end, used to route signals in a patch bay.

peak program meter (ppm): A meter designed to indicate transient peaks in the level of a signal.
percentage of modulation : The percentage of an applied signal in relation to the maximum signal a sound system can handle.
perspective miking: Establishing through mictosource distance the audio viewpoint in relation to the performers and their environment in screen space.
phantom power: Operating voltage supplied to a capacitor microphone by an external power source or mixer, thereby eliminating the need for batteries.
phase: The time relationship between two or more sounds reaching a microphone or signals in a circuit. When this time relationship is coincident, the sounds or signals are in phase and their amplitudes are additive. When this time relationship is not coincident, the sounds or signals are out of phase and their amplitudes are subtractive.
phase shift: A change in the phase relationship of two signals at a given time, or the phase change of one signal over an interval of time.
phasing: An effect created by splitting a signal in two and timedelaying one of the signal portions.
phon: A dimensionless unit of loudness level related to the ear's subjective impression of signal strength.
phone line (PL) system: See private line system.
pickup pattern: See polar response pattern.
pin: When the needle of the VU meter hits against the peg at the righthand corner of the red. Pinning is to be avoided because it indicates too high a loudness level and it could damage the meter.
pinch roller: On a tape recorder, the springloaded, freespinning rubber wheel that holds the tape against the capstan. Also called capstan idler and pressure roller.
pink noise: Wideband noise that maintains constant energy per octave. See also **white noise.**
pitch: The subjective perception of frequency. **pitch shifter** A signal processor that varies the pitch of a signal.
PL system: See private line system.
plant microphone: A stationary mic positioned on the set to cover action that cannot easily be picked up with a boom or a body mic or to provide fill sound. Also referred to as a fixed mic.
playback head: Electromagnetic transducer on a tape recorder that converts magnetic energy into electric energy.
polarity: The relative position of two signal leads—the high (+) and the low ()—in the same circuit.
polarity reversal : The control on a console that inverts the polarity of an input signal 180 degrees. Sometimes called phase reversal.
polar response pattern: The graph of a microphone's directional characteristics as seen from above. The graph indicates response over a 360degree circumference in a series of concentric circles, each representing a 5dB loss in level as the circles move inward toward the center. Also called pickup pattern.
polydirectional microphone : See multidirectional microphone.
pop filter: Foam rubber windscreen placed inside the microphone head. Particularly effective in reducing sound from plosives and blowing. Also called blast filter. See also **windscreen**
porous absorber: A sound absorber made up of porous material whose tiny air spaces are most effective at absorbing high ffequencies.
pot: Short for potenfiometer. See also fader.
potentiometer: See fader.
ppm: See peak program meter.
precedence effect: The tendency to perceive direct and immediate repetitions of a sound as coming from the same position or direction even if immediate repetitions coming from another direction are louder. Also known as the Haas effect.
prefader listen (PFL): See solo.
prelay :Recording audio elements, other than those ffom the edit master videotape, onto a multitrack recorder. See also **layback and layover.**
premix: The stage in postproduction sweetening when dialogue, sound effects, and music are prepared for final mixing.
presence: Perception of a sound as being close and realistic. See also **ambience and room tone.**
press bridge: See multiple (2).
printthrough: Unwanted transfer of a magnetic signal from one tape layer to an adjacent tape layer.
private line (PL) system:An intercom system consisting of a headset with an earpiece and a small microphone used during production to connect production and technical personnel. Also called phone line system.
production source music: Music that emanates from an onscreen singer or ensemble and is produced live during shooting or in postproduction.
proximity effect: Increase in the bass response of some mics as the distance between the mic and its sound source is decreased. Also known as bass tapup.
psychoacoustic processor: Signal processor that adds clarity, definition, overall presence, and life, or "sizzle," to recorded sound.
psychoacoustics: Study of the perception of sound stimuli.
pure tone: See **sine wave.**

Q

quantization : Converting a waveform that is infinitely variable into a finite series of discrete levels.

R

radio microphone: Wireless microphone.
rarefaction: Temporary drawing apart of vibrating molecules, causing a partial vacuum to occur. See also **compression** (2).
RDAT: See **rotaryhead digital audiotape recorder.**
read mode: Mode of operation in an automated mixdown when the console controls are operated automatically by the data previously encoded in the computer. Also called safe mode. See also **update mode and write mode.**
realtime analyzer: A device that shows the total energy present at all audible frequencies on an instantaneous basis.
record head: Electromagnetic transducer on a tape recorder that converts electric energy into magnetic energy.

recordable compact disc (CDR): A CD fommat allowing users to record one time but to play back the recorded information repeatedly.
reflected sound: Reflections of the direct sound that bounce off one or more surfaces before reaching the listener.
release: The time and manner in which a sound diminishes to inaudibility.
release time: The length of time it takes a limiter or compressor to retum to its normal level after the signal has been attenuated or withdrawn. Also known as recovery fime.
remnance: The residual magnetization ffom a previous recording after erasure.
remote: Any broadcast done away from the studio.
rerecording: The process of combining individual dialogue, sound effects, and music tracks into their final form- stereo or surround sound.
resonance: Transmitting a vibration from one body to another when the frequency of the first body is exactly, or almost exactly, the natural ffequency of the second body **retentivity** Measure of a tape's ability to retain magnetization after the force field has been removed. Retentivity is measured in gauss-a unit of magnetic energy.
reverberation: Multiple blended, random reflections of a sound wave after the sound source has ceased vibrating. Also called reverb and reverberant sound.
reverberation time: The length of time it takes a sound to die away. By definition: the time it takes a sound to decrease to onemillionth of its original intensity, or 60 dBSPL. Also called decay fime.
ribbon microphone: A microphone with a ribbon diaphragm suspended in a magnetic field.
ride the gain : Continually adjusting controls on a console
ridging :A bulge or depression, seen after winding, caused by deformed layer(s) of tape.
ring off: When a dialogue line ends with the ambient ring of a room and another line begins with that ring decaying under it.
room modes: See eigentones.
room tone: Another term for ambience. Also called presence.
rotaryhead digital audiotape (RDAT) **recorder:** Specifically, a digital cassette audiotape recorder with rotary heads. See also **stationaryhead digital audiotape recorder.**

S

safe mode: See read mode.
SA system: See studioaddress system**.**
SDAT: See **stationaryhead digital audiotape recorder.**
sampler: An audio device that records a short sound event-such as a note or a musical phrase-into computer memory. The samples can be played by triggering them with a MIDI signal from a MIDI controller or a MIDI sequencer
sampling: (1) Examining an analog signal at regular intervals defined by the sampling frequency (or rate). (2) A process whereby a section of digital audio representing a sonic event, acoustic or electroacoustic, is stored on disk or into a memory.
sampling frequeny: The hequency (or rate) at which an analog signal is sampled. Also called sampling rate.
scrape flutter filter: A cylindrical, lowfriction metal surface installed between the heads to reduce the amount of unsupported tape, thereby restricting the degree of tape movement as it passes across the heads. It reduces flutter.
scrubbing: In harddisk editing, moving the playbar cursor through the defined region at any speed to listen to a sound being readied for editing. Scrubbing is similar to rocking a tape in cutandsplice editing.
SCSI (Small Computer Systems Interface): The standard for hardware and software command language that allows twoway communication between, primarily, hard disk and CDROM drives. Pronounced "scuzzy."
segue: (1) Cutting hom one effect to another with nothing in between. (2) Playing two recordings one after the other, with no live announcement in between.
Sel Sync™ :Changing the record head into a playback head to synchronize the playback of previously recorded material with the recording of new material.
selective synchronization: See Sel Sync.
selfnoise: The electrical noise, or hiss, an electronic device produces.
sensitivity: (1)Measurement of at ape's output level capability relative to a standard reference tape. (2) Measurement of the voltage (dBV) a microphone produces, which indicates its efficiency. (3) The soundpressure level directly in front of the loudspeaker, onaxis, at a given distance and produced by a given amount of power.
sequencer: An electronic device that can be programmed to store and automatically play back a repeating series of notes on an electronic musical instrument such as a synthesizer.
shelving: Maximum boost or cut at a particular frequency that remains constant at all points beyond that hequency so the response curve resembles a shelf.
shock mount: A device that isolates a microphone from mechanical vibrations. It can be attached externally or built into a microphone.
shotgun microphone: A highly directional microphone with a tube that resembles the barrel of a rifle.
signaltonoise ratio (S/N): The ratio, expressed in decibels (dB), of an electronic device's nominal output to its noise floor. The wider the signaltonoise ratio, the better.
silent film: Film carrying picture only.
sine wave: A pure tone or fundamental frequency with no harmonics or overtones.
singleD™: microphone See singleentryport microphone.
singleentryport microphone: A directional microphone that uses a single port to bring sounds hom the rear of the mic to the capsule. Because these sounds hom the rear reach the capsule out of phase with those that reach the hont of the capsule, they are canceled.
singlesystem recording: Recording picture and sound in a film or video camera simultaneously.
slap back echo: The effect created when an original signal repeats as distinct echoes that decrease in level with each repetition.
slate: The part of a talkback system that feeds sound to tape. It is used to record verbal identification of the material being taped, the take number, and other information just before each recording.

slave: The tape or disk to which the material on a master recording is transferred.
SMPTE time code: A highhequency signal consisting of a stream of pulses produced by a time code generator used to code tape to facilitate editing and synchronization. Also known as longitudinal fime code.
S/N :See signaltonoise ratio.
solo: A control on a multitrack console that automatically cuts off all signals feeding the monitor system except those feeding through the channel that the solo control activates. Sometimes called prefacer listen (PFL).
sound absorption coefficient: A measure of the soundabsorbing ability of a surface. This coefficient is defined as the haction of incident sound absorbed by a surface. Values range hom 0.01 for marble to 1.00 for the materials used in an almost acoustically dead enclosure. Also known as noise reducfion coefficient (NRC).
sound chain: The audio components that carry a signal hom its sound source to its destination.
sound design: The process of creating the overall sonic character of a production (usually in relation to picture).
sound designer: The individual responsible for a production's overall sonic complexion.
sound envelope : Changes in the loudness of a sound over time, described as occurring in four stages: attack, initial decay, sustain, and release (ADSR).
soundfile A sound stored in the memory of a harddisk recorder/editor.
sound film: Film carrying both picture and sound.
sound frequeny spectrum: The range of frequencies audible to human hearing: about 20 to 20,000 Hz.
soundpressure level: See dBSPL
sound transmission class (STC): A rating that evaluates the effectiveness of barriers in isolating sound.
source music: Background music hom an onscreen source, such as a stereo, radio, or juke box.
spaced miking: Two, sometimes three, microphones spaced hom several inches to several feet apart, depending on the width of the sound source and the acoustics, for stereo recording.
spacer: See fill leader.
SPDIF (Sony/Philips Digital Interface): The consumer version of the AES/EBU interface calling for an unbalanced line using phono connectors. See also **AES/EBU interface.**
spectrum processor: A signal processor that affects a sound's spectral range.
splicing tape: A specially made adhesive tape that does not ooze, is nonmagnetic and pressure sensitive, and is used to join cut ends of audiotape and magnetic film.
split editing: (1) Editing the same sound into two or more separate tracks to facilitate control of its length and in editing transitions. In dialogue, for example, this makes it easier to extend lines that may have been cut too short during picture editing, to overlap voices, and to carry over lines hom one scene to the next. (2) A type of transition where the audio or video leads or lags a portion of the previous edit.
splitsection console: Multichannel production console in which the input, output, master, and monitor sections are separate. See also **inline console.**
splittrack recording: Recording two separate sound sources on two separate tracks of a stereo recorder or VCR with two audio tracks.
spotting: Going through a script or work print and deciding on the placement of sound effects and music.
spotting sheet: Indicates the sound effect, or music, cue and whether it is synchronous or nonsynchronous, its in and outtimes, and its description.
stationaryhead digital audiotape (SDAT) recorder: A fixedhead digital audiotape recorder. See also **rotaryhead digital audiotape recorder.**
STC: See sound transmission class.
stereotomono compatibility: Ensuring that a recording made in stereo is reproducible in mono without spatial or spectral distortion.
stereophonic microphone: Two directional microphone capsules, one above the other, with separate outputs, encased in one housing.
stripe coat: Magnetic film that contains two stripes of oxide coating, a wide stripe for recording singletrack mono and a narrow balance stripe to ensure that the film wind on reels is smooth. See **also full coat.**
studioaddress (SA) system: An intercom system used like a publicaddress system to communicate with people in the studio not connected to the privateline system, such as the performers, and for general instructions to all studio personnel. Also called a tallback.
subtractive equalization: Attenuating, rather than boosting, hequencies to achieve equalization.
sustain: In the sound envelope, the period during which the sound's relative dynamics are maintained after its initial deCaY
sweet spot: In control room monitoring, the designated listening position that is the optimal distance away hom and between the loudspeakers.
sweetening: Enhancing the sound of a recording through the procedures of layover, prelay, premixing, and layback.
sync beep: See sync pop. synchronization The ability to lock two or more devices that have microprocessor intelligence so that they operate at precisely the same rate.
synchronizer :(1) Device with sprocketed, ganged wheels that locks in the film reels of picture and sound so they can be wound in synchronization during editing. (2) Device that regulates the operating speeds of two or more recorders so they run in sync.
sync pop A single frame of magnetic film cut across hom the Academy 2 with a 1,000Hz tone that creates a beep. Also called sync beep.
sync tone: The tone or pulse that synchronizes tape recorder speed and film camera speed in doublesystem recording.
system microphone: Interchangeable microphone capsules of various directional patterns that attach to a common base. The base contains a power supply and a preamplifier.
system noise: The inherent noise an electronic device or system generates.

T

tails out: Having the end of the material on a tape or film at the head of the reel.

talkback: Studioaddress intercom system that permits communication hom a control room microphone to a loudspeaker or headphones in the studio.
tangency: One of the adjustments made when aligning the heads of an audiotape recorder. This adjustment aligns the forwardness of the heads so that the tape meets them at the correct pressure.
tape transport system: The mechanical portion of the tape recorder, mounted with motors, reel spindles, heads, and controls, that carries the tape at the constant speed from the feed reel to the takeup reel.
temporal fusion: When reflected sound reaches the ear within 10 to 20 ms of the original sound, the direct and reflected sound are perceived as a single sound. This effect gradually disappears as the time interval between direct and reflected sound increases hom roughly 30 to 50 ms.
threetoone rule: A guideline used to reduce the phasing problems caused when a sound reaches two microphones at slightly different times. It states that no two microphones should be closer to each other than three times the distance between one of them and its sound source.
threeway system loudspeaker: A loudspeaker that uses two crossover hequencies to divide the bass, midrange, and treble hequencies.
threshold of hearing: The lowest soundpressure level (SPL) at which sound becomes audible to the human ear. It is the zero reference of 0 dBSPL.
threshold of pain: The soundpressure level at which the ear begins to feel pain, about 140 dBSPL, although levels of around 120 dBSPL cause discomfort.
tie line: Facilitates the interconnecting ot outboard devices and patch bays in a control room or between studios. **timbre** The unique tone quality or color of a sound.
time code address: The unique SMPTE time code number that identifies each 1/30 of a second of a recording.
time compression: Altering the time of material without changing its pitch.
time processor: A signal processor that affects the time interval between a signal and its repetition.
tinnitus: After prolonged exposure to loud sounds, the ringing, whistling, or buzzing in the ears, even though no loud sounds are present.
TL: See **transmission loss.**
transducer: A device that converts one form of energy into another.
transient: A sound that begins with a sharp attack followed by a quick decay.
transient distortion: Distortion that occurs when a sound system cannot reproduce sounds that begin with sudden, explosive attacks.
transmission loss (TL): The amount of sound reduction provided by a barrier such as a wall, floor, or ceiling. **transmitter microphone** Wireless microphone.
treble: Frequency range between roughly 5,000 and 20,000 Hz, the highest two octaves in the audible hequency spectrum.
trim: (1) To attenuate the loudness level in a component or circuit. (2) The device on a console that attenuates the loudness level at the microphone/line input.
tube microphone: A capacitor microphone using a tube circuit in the preamp.
tweeter: The informal name of a loudspeaker that reproduces high hequencies. See also **woofer.**
twoway system loudspeaker: A loudspeaker that uses one crossover frequency to divide the highs from the lows.

U

ultrasonic: Frequencies above the range of human hearing.
unbalanced line: A line (or circuit) with two conductors of unequal voltage.
underscore music: Nondiegetic music added to enhance the informational or emotional content of a scene.
unidirectional microphone: A microphone that picks up sound from one direction. Also called directional microphone.
update mode: Mode of operation in an automated mixdown when an encoded control can be recorded without affecting the coding of the other controls. See also read mode and write mode.
upper bass: Frequency range between roughly 80 and 320 Hz.
upper midrange: Frequency range between roughly 2,560 and 5,120 Hz.

V

variableD™ microphone: See multipleentryport microphone.
variablespeed control: Device on an audiotape recorder that alters the playing speed to various rates of the recorder's set speeds.
VCA: See voltage controlled amplifier.
velocity The speed of a sound wave: 1,130 feet per second at sea level and 70 degrees Fahrenheit.
vertical interval time code (VITC): Time code that is recorded vertically on videotape and within the video signal but outside the picture area.
videoleadingaudio: When the picture of a new scene starts before the sound of the old scene has finished. See also **audioleadingvideo.**
virtual track: In harddisk recording, a track that provides all the functionality of an actual track but cannot be played simultaneously with another virtual track.
VITC: See vertical interval time code.
voltage controlled amplifier (VCA): An amplifier used to decrease level. The amount of amplification is controlled by external DC voltage.
volume unit (VU) meter: A meter that responds to the average voltage on the line, not true volume levels. It is calibrated in volume units and percentage of modulation.
VU: See **volume unit meter.**

W

walla : A nonsense word that used to be spoken by film extras to create ambient crowd sound, without anything discernable actually being said.
waveform :A graphical representation of a sound's characteristic shape displayed, for example, on test equipment and harddisk editing systems.
wavelength: The length of one cycle of a sound wave. Wavelength is inversely proportional to the hequency of a sound; the higher the frequency, the shorter the wavelength.

weighting network: A filter used for weighting a hequency response before measurement.
wet sound: A sound with reverberation or signal processing. See also **dry sound.**
white noise: A wideband noise that contains equal energy at each hequency. See also **pink noise.**
windscreen : Foam rubber covering specially designed to fit over the outside of a microphone head. Used to reduce plosive and blowing sounds. See also pop **filter.**
wireless microphone system : System consisting of a transmitter that sends a microphone signal to a receiver connected to a console. Also called radio, FM, transmitter, or cordless microphone.
woofer: Informal name for a loudspeaker that produces the bass hequencies. See also **tweeter.**
worldizing: Recording room sound to add to a dry recording or to use to enhance or smooth ambient backgrounds that are already part of the dialogue track.
wow: (1) Starting a recorded sound before it reaches full speed. (2) Frequency changes in an analog tape recording resulting hom slower variations in the speed of the tape transport. See **also flutter.**
wrap: One of the adjustments made when aligning the heads of an audiotape recorder. This adjustment aligns the head so that it is in full physical contact with the tape.
write mode: The mode of operation in an automated mixdown during which controls are adjusted conventionally and the adjustments are encoded in the computer for retrieval in the safe mode. See also **read mode and update mode.**

X,Y,Z

XLR connector: Commonly used male and female microphone plugs with a threepin connector.
XY **miking:** Coincident or nearcoincident miking that places the microphones' diaphragms over or horizontal to one another. See also **coincident miking and nearcoincident miking.**
zenith: One of the adjustments made when aligning the heads of an audiotape recorder. This adjustment aligns the vertical angle of the heads so they are perpendicular to the tape.

Notes :

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