SOUND AND SOUND SYSTEMS

By Bill Addison

This article is not intended as a set of rules for sound analysis. Rather, it is a rapid concentration of several volumes of material on sound systems and techniques for their use. It is an attempt to present in layman (caller) terms, sufficient information to help the caller obtain maximum effectiveness from his sound system.

BASICS.

There are a few basic rules which apply equally to the care and use of all sound systems. I call them "Cardinal Rules of System Care."

1. Keep it CLEAN. More damage is done to electronics equipment by lack of cleanliness than any other single factor. Dust and dirt cause damage to lubricated units which will cause their eventual failure; lack of cleanliness reduces effective cooling of the system which, in turn causes damage through overheating; dirty connections reduce the effectiveness of your system through loss of signal trying to drive through the dirt. This is especially true of cable connections between the system amplifier and the speakers.

2. Keep it DRY. Moisture causes rust damage to lubricated parts, deterioration of speaker cones, electrical shorts in wiring and printed circuit boards, and deterioration of equipment cases - to mention a few.

3. Treat it GENTLY. Severe shock can cause misalignment of mechanical connections and moveable parts, distortion of speaker cones, broken printed circuit boards and other electrical components, and case damage.

4. Give it RESPECT. Do not expect your system to perform properly if you misuse or abuse it. Turn the power off BEFORE you disconnect the speakers. Remove cable connectors by the connector body – not the cord. Use proper termination. IMPEDANCE IS IMPORTANT.

VARIABLES.

A great number of variables must be taken into account when setting up a sound system to produce listenable, danceable music and voice in any hall. We can eliminate a few of these variables by judicious use of our sound systems. Others can be resolved by your ability to overcome adversity. Let us consider some of the more serious variables.

HALLS.

First, there is the size of the hall. Table I provides the approximate size of the more common halls in use and the approximate power required to cover them. You will be able to interpolate these sizes and power requirements by using the floor space specified in the

table and then calculating the power required. Values of power recommended are based upon one set of dancers for every 120 square feet of floor space. When the population density is increased, power must be increased accordingly. Dancer density is calculated on average dance floor space while floor space figures are based on the overall (wall-to-wall) floor sizes.

The second variable to be considered is the structure of the hall. Most halls in this area have acoustical ceiling tiles and porous concrete walls. Floors are generally vinyl tile. Table I is calculated with these conditions assumed. Variations in floor type will not cause excessive changes in power requirements but excessively high ceilings or hard finished walls will require changes in power requirements. These variables will be addressed later in this discussion.

TABLE I

TYPE HALL	FLOOR SPACE (Length x Width)	DANCER DENSITY (Average)	VOLUME (Approx.)	POWER REQUIRED (Average)
Elem School M/P Room (Small Rec. Ctr)	2400 Sq. Ft. 40 x 60	10 Sets	29,000 cu. ft.	23 Watts
Jr. High School N/P Room (Large Rec. Ctr)	5100 Sq. Ft. 60 x 85	16 Sets	82,000 cu. ft.	28 Watts
Small Gym	8000 Sq. Ft 80 x 100	24 Sets	144,000 cu. ft.	34 Watts
Large Gym	18,000 Sq. Ft. 120 x 150	40 Sets	432,000 cu. ft.	65 Watts

The variables of dancer density and hall size and construction are with us for every event and must be reckoned with each time we call a dance. These variables can be overcome with use of good judgment, proper placement of speakers, and provision of adequate power from the amplifier. Although you do not need to be an acoustical engineer to get the most from your equipment, knowledge of some of the factors which an acoustical engineer would consider is most helpful in achieving "maximum sound success" at your dances. The following sections will give you a "working knowledge" of some of the more important elements which can lead you to "maximum sound success." We will give you a "crash course" in Power Output, Impedance, Speaker Construction, Microphone Construction, and Turntable Operation. Each of these areas will be reduced (as far as possible) to layman terminology; if you need more information than is presented here, several books on acoustical engineering and basic electronics are available at the local library. A clear understanding of amplifiers, microphones, speakers, and impedance can not be gained without understanding of some of the very basic principles of electronics and physics. A few definitions and principles are in order to allow us to gain an understanding.

SOUND PHYSICS

The first definition is that of sound. Sound is defined (for our purposes) as the compression and rarefraction air pressure applied to a conversion medium. Translated, this means that as we speak, we push and pull the air and that push/pulling of air works on the sensitive membrane of the ear to produce sound.

[*Rarefaction is the reduction of an item's density, the opposite of compression.*]

Next is the relationships of electric current and magnetic fields. When a current is passed through a conductor (wire), a magnetic field is produced around that conductor. When a conductor is passed through a magnetic field, the movement causes a current to flow through the conductor. These two facts are use to convert sound to electrical signals and to convert electrical signals into sound. These relationships are applied to both microphones and speakers. The functions of the microphone and the speaker are almost identical in their mechanical makeup.

We will not try to explain the functions of amplifiers other than to say that each amplifier takes a very small electrical signal and convert it into a large electrical signal. We will however delve into the mysteries of electronics in order to understand the functional relationships of the amplifier, the microphone, and speakers.

POWER AMPLIFIERS.

Let's begin with Power Output. The power output of an amplifier is usually stated by the manufacturer in Watts, Peak. Typical amplifiers used in our activity today have Peak Power outputs in the range of 40 watts to 300 watts depending on the model equipment you choose. Power is specified as peak power output from the unit. This does NOT mean that you can expect that level of power output for every sound emitted by the amplifier. The maximum continuous power output of a system is 63.7% of the peak power specified for the system. In explanation, peak power out of an amplifier is the power emitted when a single tone is passed through the amplifier with the gain control (volume) set at maximum and the amplifier is connected to a properly matched load (speaker system). A significant point is the fact that every amplifier is designed to deliver its maximum power output ONLY when the load impedance is equal to the impedance specified by the manufacturer for his equipment design.

IMPEDANCE (Z).

Impedance is next in line to be understood. The subject of impedance has either never been understood or, if understood, almost forgotten by most callers; their lack of knowledge or failure to adhere to the rules of impedance matching cause deterioration of sound quality, sound level loss, and sound system failure. Impedance is the opposition to current flow and determines the level of output actually delivered by the sound system to

the speaker or from the microphone to the amplifier. A clear understanding of impedance cannot be gained without the knowledge of speaker and microphone operation. We will use the speaker operation to gain this understanding. The typical speaker has a permanent magnet which establishes a reference field at the base of the speaker cone. A second magnetic field is established when current from the amplifier is passed through the speaker "Voice Coil." Since magnetic fields either attract or repel each other, variations in current caused by voice or music signals coming from the amplifier cause the voice coil (which is attached to the speaker cone) to move back and forth with reference to the fixed magnetic field at the base of the cone. This movement of the cone and attached speaker surface causes compression and rarefraction of the air on the surface of the speaker cone which is converted by the human ear into sound. At first glance, it would appear that lower impedance will allow more current to flow causing more sound. This logic is true within limits. There is a point in the design of each speaker coil at which a specific amount of current flow causes maximum transfer of energy that properly reproduces sound transferred to it. As we pass this point in either direction (more or less current flow), we will experience a reduction in energy transfer either in the form of lower sound output level or distortion of the sound output (overdriving the speaker). The actual mechanics of speakers are taken into account by the systems' design engineers and are of little concern to us except to understand the importance of load matching and power limitation.

To get a better understanding of impedance, we need to know one of the basic rules of energy transfer in electrical (electronic) circuits. That rule is "Maximum transfer of energy from a generator (source) to a load is accomplished when the internal impedance of the generator (source) and the impedance of the load are equal." Translated to out terms, this means that maximum sound power from the amplifier will be maximum sound energy will be produced when the speaker system has an impedance of 8 ohms. So, if our amplifier is designed to operate into an 8 ohm load and we connect a speaker system with an impedance of 16 ohms, the output of the speaker system will he reduced; if we go to the other extreme and connect a speaker system load of 4 ohms, not only will the sound output he reduced but we will cause excessive current to flow in the amplifier output circuit and this can be disastrous, especially in solid state (transistorized) amplifier systems. Although manufacturers design their systems with protection from high current damage in the output circuits, the use of speaker systems whose impedance is critically lower than that of the amplifier will disrupt the output of the amplifier (kick the circuit breaker on the equipment) and will ultimately cause damage or failure of the output circuit of the amplifier.

Generally, when you purchase your amplifier and speaker system from the same manufacturer, the sound system is "balanced", i.e., the speaker is of the same impedance as the output of the amplifier and is capable of handling the power output of the amplifier. However, many situations will arise when you will have to determine the mix of speakers and amplifier. A good rule of thumb is to ensure that the speaker system is capable of accommodating at least 50% more power than the output capability of the amplifier.

There will be times when a single speaker unit will not "cover" the hall and you will want to add speakers to provide a wider range of coverage. In a later section of this

document, we will show you how to get the most out of speaker placement but you MUST remember to match the speaker system impedance as a first consideration. Most speaker units are connected to the amplifier as a single unit. Typically, a single unit speaker assembly is housed in one case even though there may he several speakers in the single case such as those produced by Hilton, Caliphone, Newcombe, and Ashton. Each of these units have a nominal impedance of 8 ohms. The "Yak-Stak" and Director, produced by YakStak provide two individual units which can be used together or separately. Each of the units produced by YakStak have nominal impedance of 8 ohms and when used together can have an impedance at the amplifier output of either 4 ohms or 16 ohms depending on how the two units are connected. Individual; speaker units can be connected to each other and to the amplifier in either of two ways, series or parallel. Connection of speakers in series means that the voice coils are connected so that signal current flows from the "top" lead of the amplifier to the input of the "top" of first speaker, the "bottom" lead of that speaker is tied to the "top" lead of the second speaker, and the "bottom" lead of speaker two is tied to the "bottom" lead of the amplifier output. Parallel connection means that the "top" and "bottom" leads of all speakers are tied to the "top" and "bottom "leads of the amplifier respectively.

The calculations for determining the combination of more than one speaker on an amplifier output are fairly simple. When speakers are connected in series, the impedance of speakers are additive, i.e., $ZT = Z1+Z2+Z3$. When speakers are connected in parallel, the impedance of the speakers can be calculated using the formula $1/ZT = 1/Z1 + 1/Z2 + 1/Z3$. A quick way to calculate the impedance of speakers in parallel is to divide the impedance of one speaker by the number of speakers connected in parallel (assuming that all speakers have the same individual impedance). Figure I provides some typical speaker combinations with regard to impedance matching. Individual voice coil groupings are shown as coils in the figure. You should also match polarities of speaker coils since reversing polarities of voice coils will cause a shift of 180 degrees between two (or more) speakers thus canceling out sound produced by the speakers. Most speaker manufacturers put polarized plugs and jacks on their equipment to prevent "flipping the phase" on the speakers. Still, you should check when connecting more than one speaker to the system. Amplifier and speaker specifications should be checked to ensure proper load matching.

SPEAKER LOCATION.

Another variable to be considered is speaker positioning. Everyone has, at one time or another encountered the problem of not being able to "cover the hall." Many times, this is due to less than maximum advantageous positioning of speakers. In order to understand and solve, or at least reduce to a minimum the problem of speaker positioning, we must first understand the assets available to us. One asset is the sound radiation characteristics of the speaker. These characteristics are generally provided by the manufacturer (and generally lost along with the packing materials in which the speaker was shipped to us). All is not lost....We can determine with reasonable certainty the radiation characteristics of a speaker with a few simple manipulations and a little understand of sound dynamics. Simply stated, the dynamics of speakers and their production of sound are that the speaker

cone compresses and rarefracts (creates a void) the air on the surface of the speaker cone. This compression and rarefraction of air is then "transmitted" through air until it reaches another similar device which can detect the compression and rarefraction of the air and convert them into intelligence. This device, in our world, is the human ear or more simply, the listener/dancer. Every speaker (with enclosure) will, when driven by a predetermined amount of sound power (from the amplifier), produce a predictable sound pattern. This pattern can he determined by measuring the two planes of radiation, the horizontal and the vertical. The envelope of sound in these planes is called the "major sound lobe." The major sound lobe is actually three dimensional, having width (the horizontal lobe), height (the vertical lobe), and depth (the amount of sound power radiated). The height and width of the sound lobe are determined by the position and depth of cone of the speaker within the enclosure; the depth (length) of the sound lobe is a function of the power applied to the speaker cone.

Figure II shows a method for determining the sides of the major lobe. The same method is used for determining both the horizontal and vertical lobe patterns of a given enclosure. This procedure is relatively simple and will give you a pretty good feel for coverage by the speaker(s) in use. First, draw a line from the speaker cone through one edge of the enclosure (Line A in Figure II). Next, draw a line parallel to the side of the enclosure (Line B in Figure II). Draw a line which bisects the angle formed by these two lines (Line C). The line which bisects the angle will be a relatively reliable line of the side of the major sound lobe provided by the speaker. Repeat the process for the opposite side of the speaker enclosure and you will show the approximate width of the major sound lobe. Use the same procedure for determining the vertical lobe. In the case of column speaker enclosures, the measurement of the vertical lobe should be made using the angle formed by the top speaker in the enclosure for the top lobe and the bottom speaker in the enclosure for the bottom lobe. The angle between the sides of the lobe pattern will be the radiation angle of the speaker and will determine the general coverage you can expect from your speaker. Ideal radiation angles are 23° to 38° for the vertical plane and 110° to 160° horizontally.

Now that we understand the width of the major sound lobe, let's take a look at how to determine the length of the lobe. A few basic "facts of physics" are in order here. Sound power, when radiated in air travels at the rate of 1,130 feet per second at an ambient temperature of 68°F. Sound power is measured in decibels. The decibel is a relative measurement of energy between two points in a system. Trust me! These are facts. When you enter your course in acoustical engineering, these facts will be used frequently. For our purposes, these facts will be translated to allow a better understanding of sound power and sound radiation. Sound Power is reduced by the square of the distance from the speaker. In other words, if the sound level heard at 10 feet from the speaker is X decibels (dB), the sound level 20 feet away from the speaker will be 0.5X dB, at 30 feet the sound level will be 0.25X dB, and at 40 feet the level will be 0.125X dB. All this is presented to get your confidence in Table I which takes some of the guess-work and calculations out of determining the length of the major sound lobe. By using the information in Table I and your own adaptation of sound lobe characteristics developed in Figure II, you will be able to determine the depth of coverage in a straight line from your speakers. Overlaying this

information on the diagram of the halls you use frequently will give you a pretty good idea of how many speakers and how much power you will require to cover the hall.

If you will diagram halls you frequently use and overlay the major sound lobe diagrams using the procedures which were outlined earlier, you can determine the most efficient placement of speakers to obtain the best hall coverage. Once you have diagrammed the hall try the results. Set up your speakers according to plan and listen to the coverage it produces. Put a record on (using the called side of the record) and set the gain control to approximate the level indicated by your calculations. Now get on the floor and "walk the hall" and LISTEN....If the sound level is comfortable to you in an empty hall, you will have to increase the sound level by as much as one third as the hall fills with dancers. Soft, porous surfaces (dancers and soft walls) absorb sound so you must compensate for the absorption.

The major sound lobe of your speaker will be fine for coverage of dancers within the coverage area. Often, the location of the caller platform (stage) may be such that some dancers will dance outside the major sound lobe pattern (such as very close to a high stage and on the side of the hall). These dancers are dancing in the secondary sound lobe area (the area between Line A and Line C in Figure II). In particularly wide halls, you may be able to adjust the speaker position (move it further hack on the stage) to increase the major sound lobe coverage. You can make the secondary lobe useful by increasing the power output to the speaker slightly.

Now let's touch on the problem speaker placement in "hard to sound" hall. One of the most common difficulties is the "live sound" hall. This hall typically has a high ceiling, hard wall surfaces (cement or painted cinder block) or a combination of these. In these halls, reverberation (echo) is predominant. There are several ways to improve coverage but very few will overcome poor acoustical coverage. One method of coping with these conditions is called "stacking" — that is mounting speakers so that the vertical lobe is slanted downward toward the back of the hall near the floor. This will reduce the amount of sound which is directed toward the flat, hard, reflective surface and will reduce the echo since sound reflected from the back wall will effectively "cancel" sound reflected from the floor. Next, use a little more bass boost (increase the amount of bass and reduce the amount of treble using the amplifier controls) for both microphone and music inputs to the amplifier. Reverberation is predominantly the high frequency components of sound so reduction in the high frequencies being amplified will reduce echo response. Third, use the minimum gain (volume) required to reach the back of the hall. Reduction in gain will reduce the amount of power to be reflected (echo) by the hard surfaces. Only in very rare instances will you be able to "power" your way over a highly reflective hall. The horizontal lobe requirements will not normally change significantly in a "live" hall so you will probably be able to use normal speaker positioning in them. It has been proven that column speakers work best in live halls when the speakers are raised above what you would consider "normal" height and then tilted downward slightly. Close examination of column speaker radiation patterns will explain this fact. You can get equally good coverage with "horn" type speakers using similar methods (raising and tilting). Finally, it should be apparent that you MUST use your BEST microphone techniques in difficult halls. While

your best microphone techniques are always recommended, they are especially necessary in "hard" and "line" sound situations.

Once you are familiar with the capabilities of your sound system, you will be able to tell at a glance the power settings and speaker placement needed for the best hall coverage. Until you fully develop your "sighting the hall" capabilities, you may fine it helpful to lightly indicate the angles of the major sound lobe on the top and sides of your speaker enclosures to be used in "sighting the coverage." I recommend placing small pieces of tape on the sides of the speaker with radiation angles drawn on them to aid in "sightings."

MICROPHONES

We have examined the output side of our sound system at some length. Now let's talk a little about the input side of the system. The mysterious, essential, unreliable, perplexing little monster called the MICROPHONE (better known as Mike) and microphone techniques have an essential role in the production of an evening of happy square dancing. In this section we will try to shed some light on several facets of microphones and their use.

One of the greatest variables which enter the scene is the microphone. There are crystal mikes, ceramic mikes, ribbon mikes, reluctance conversion mikes, soft-faced mikes, static transducer mikes, and dynamic mikes just to name a few types. Lavaliere, stand, hand, wireless, boom, and immersible are terms used to refer to the size, environment of use, and methods of positioning. Prices range from about $20.00 to several thousand dollars.

The Caller must choose the microphone which will most adequately serve his needs if he to produce the most from his sound system. In the morass of all the descriptive information, the Caller may soon become desperate and say to a friend, "Hey, Fabitz, what kind of mike are you using? It sound great and looks impressive" To which Fabitz (being somewhat of a show-off) says "Oh, it's a Fandango 49933XY. It has an output of -22 dB with a whale of a good cardioid input incidence, variable impedance, ceramic transducer, and is transformer coupled. Has a frequency response of 2 hertz to 23 kilohertz and a real steal at $549.98." Your first reaction is generally "Sorry I asked." Now let's examine some of these deep dark mysteries in terminology and try to get something which is meaningful to the Caller who does not happen to also be an electronic engineer.

First, the manufacturer's type and model number is for identification only. For instance, a very common microphone used by callers today is the Electro-Voice 635. Equally good (or maybe better) microphones are made by Shure, Astatic. Peavey, Turner, and several other companies. I make no recommendation as to which microphone is the best because each company makes microphones which are "best" for the individual choosing the unit. When you have examined the specifications for each microphone, you can then choose the one which meets your particular need and you will find several different types which will suit you. It then becomes a matter of price — which one can you afford. The information presented here is to provide you with some of the information needed for you to make a wise choice.

In recent years, the dynamic microphone has more or less "taken over the field." The dynamic microphone is basically a very sensitive speaker operating in reverse. Generally, a moving coil in a fixed magnetic field is used to produce the conversion of sound pressure (your voice) to electrical signals. This is the same method that the speaker uses except that your voice now moves the diaphragm of the microphone which moves the "voice coil" in a magnetic field. The movement of the coil in the magnetic field produces a very small electrical current which is amplified by the amplifier. While the dynamic microphone generally has lower sensitivity than the ceramic microphone, this is a relatively minor factor to the caller since the microphone is usually very close to the sound source (the big open mouth). Frequency response from dynamic microphones normally is in the range of 50 hertz to 16000 hertz which means that the dynamic microphone is more "bassy" (producing more sounds in the lower voice range). This frequency response is more than adequate for most callers since the average male voice produces frequencies between 110 and 600 hertz. Lady callers' voice range generally extends to 800 hertz for the very "high-pitched" voice. The dynamic microphone is generally less expensive than other types with excellent quality microphones ranging in cost from $75.00 to about $250.00.

Two additional factors (beyond price and frequency response) which be considered in choosing a microphone. These factors are the input characteristics and the output capabilities of the microphone. The input characteristic to be considered is the "shape of the input." Microphone input is generally characterized by the area in which sound will be detected by the microphone. Microphones used in a studio environment will generally be less directional so that large areas can be "covered" by the microphone. A more desirable input pattern for the Caller is one which is highly directional; that is, one in which most of the sound pressure must be directly in front of the microphone. A cardioid pattern is the most desirable one to meet this requirement. Mr. Webster defines cardioid as "heart-shaped" and this definition holds well for microphones. The manufacturer designs many microphones to be most sensitive in the cardioid pattern for directional microphones and generally provides a brochure which shows the area around the microphone face which will provide the best results. Figure III shows the position of this area with respect to the axis of the microphone. It can be envisioned as a slightly inflated balloon tied to the end of the microphone. Sound produced inside the balloon will reproduced most reliable by microphone.

The second characteristic which must be considered is the output of the microphone. Output characteristics are expressed by the manufacturer in both level and impedance. When you examine the specifications for a microphone, you will find the measure of microphone sensitivity listed under "Output." A curious number usually is listed under Output such as -52 dB. The term "dB" is an abbreviation for decibel which was discussed slightly earlier in this document. As noted, the decibel is a relative term used to express the relationship between two energy points (such as the input and output of a microphone. I will not bore you with the lengthy explanation of how the value is derived but for the "physics-minded", a trip to the library will provide you volumes on the derivation. Here, it is sufficient to say that the ratio expressed is the ratio of input sound pressure (calculated at 0.002 dynes/sqcm) to the output voltage (0 dB = 1.73 volts). Good

quality microphones should have outputs rated between -36 dB and -52 dB. Microphones whose rated outputs are rated between -44 dB and -58 dB are excellent for use in most popular power amplifiers.

Microphone outputs are also rated according the output impedance (Z) and are generally specified as high impedance or low impedance units. Some of the better microphones offer a choice of high or low impedance through a switch mounted on the microphone body or through wiring connections at cable connection. Just as in the requirement to match the impedance of the speaker to the output of the amplifier, the Caller must match the impedance of the microphone to the input of the amplifier. Most amplifiers in use today require a high impedance input, typically about 30,000 to 50,000 ohms (an ohm is value of impedance). Should you happen to connect a low impedance microphone to an amplifier which requires a high impedance input, you will immediately notice that the microphone gain control becomes extremely sensitive to change and difficult to control. You should use either a high impedance microphone or a microphone adapter which contains a low-to-high impedance converter (usually a small audio transformer) such as those units sold by Hilton. Continued use of a low impedance microphone into an amplifier designed for a high impedance input will ultimately cause damage to the input circuit of the amplifier.

In summary, choosing a microphone should take into consideration the cost of the unit and its input and output characteristics. Excellent microphones are available for about $100,00 but even the most expensive microphone will not correct poor microphone technique. Good microphone technique is essential in the use of any microphone you choose. In fact, the novice caller might want to use a less sensitive microphone in order to require better diction habits to get the most of the system.

RECORD PLAYERS.

Where would the caller be without the music??? It is obvious that the record player (turntable and tone arm) is a major factor in the production of good square dancing. Most manufacturers of equipment for callers and cuers adapt a specific turntable and tone arm to suit their system; we are therefore limited in variations. You should still understand the variables which are encountered in this part of the sound system. Generally, the turntable and tone arm are mounted on the amplifier housing. In order to get the best results from the record player, you should therefore ensure that the amplifier housing is placed on a stable, level surface when in use. Most amplifier housings provide "shock mounts" as the feet of the housing. If your system does not provide shock mounting, you should "float" the unit on a piece of foam rubber to minimize external mechanical shock to the unit. Most turntables are friction driven. The friction drive system uses the weight of the platen (turntable platter) and the record to produce the friction necessary to turn the platen. accordingly, you should not stop the platen when placing or removing a record on the turntable. Stopping the platen causes excessive wear on the friction drive wheel and will eventually cause unstable operating speed of the turntable.

The tone arm is a more complex situation than the turntable. The tone arm contains the sensor head (cartridge and stylus), commonly called the cartridge, which will detect the minute variations in the record groove and the mechanical balance which will determine how hard the stylus presses on the record grove. The tone arm should be treated gently to ensure that it will both reproduce the music and will not skip or scratch the records. Two common cartridges are available, the ceramic cartridge and the reluctance pickup. The reluctance pickup is popular in fixed base stereo systems but is more expensive and less rugged than the ceramic cartridge. It works on the same principal as the dynamic microphone, i.e., a fixed and variable magnetic field in which the signal is produced by the stylus moving a permanent magnet in a coil. The most common cartridge in use in "square dance rigs" is the ceramic cartridge. The ceramic cartridge uses the principle known as the piezo-electric effect to produce signals which can be amplified. Piezo-electricity is the result of properties of certain crystalline materials which will when distorted (bent) will produce a small electrical signal that can be amplified. The piezo-electric effect is present naturally in some minerals and it can he produced in some man-made crystals called ceramics. These cartridges are rugged enough to withstand the handling given most portable equipments and have a frequency response which produces excellent sound. Since the manufacturer designs his system with a particular cartridge in mind, should you ever have to replace the cartridge, you should use the same cartridge type as originally designed into the system.

Two significant functions of the tone arm are worthy of note. The first of these is the tracking pressure (tracking weight).Tracking weight is the weight, expressed in grams, exerted by the tone arm on the stylus against the record groove. Tracking weight is generally about 1 gram and should not exceed 5 grams for normal operation. Tracking weight of more than 5 grams will produce excessive record wear, impede turntable speed, and can cause distortion of the output. Manufacturer set tracking weight of the tone arm and it will usually remain constant unless the tone arm is damaged. Some callers will increase track weight by putting a coin on top of the cartridge when poor tracking caused by poor turntable balance or stylus wear is encountered. If you begin to experience poor tracking (tone arm skipping over the record or marking of records by the stylus), you should correct the cause rather than try to overcome it with make-shift measures.

The second significant point with regard to the tome arm is the type of stylus to be used. There are three basic stylus types available for most cartridges. These are the osmium, sapphire, and the diamond styli. The osmium stylus will produce about 1,000 to 3,000 plays without significant loss of signal quality. The sapphire stylus is rates at about 10,000 plays while the diamond stylus will yield in excess of 15,000 plays. Unfortunately, you can not expect the maximum number of rated plays from your stylus since you will generally reset the needle (stylus) many times more than if you were playing records in a stable environment. Record conditions, rickety tables, and generally less than optimum environment can cause stylus wear. The Caller should check his stylus regularly to ensure its condition. You can see the condition of the stylus through a magnifying glass. It should be compared to a new stylus for signs of wear. Since the average caller drops the needle 34 times in a 2 1/2 hour dance, and since about 1 in 30 callers ensure proper tracking weight on a regular basis, we can safely assume that the average caller will

get about 50% of the rated life of the stylus. Moral....CARRY AT LEAST ONE SPARE STYLUS.

RECORD CARE.

Now that we have discussed the "hardware" used to produce the sound, we need to talk a little about the "silent partner" to the Caller in producing the dance — the records. Much has been published over the years about the care of records. Here is my condensation of the common rules to be applied.

1. KEEP RECORDS CLEAN. Damage to both the records and the stylus can occur when dust and dirt get in the groove of the record. Many excellent record cleaning cloths are on the market. You may also want to attach a record brush to the tone arm to "sweep" the record each time you play it. Just be careful that adding the brush does not alter the track weight of the tone arm.

2. STORE RECORDS IN RECORD SLEEVES. Record sleeves are available from most record suppliers and should be used to protect records from dust, dirt, and abrasive surfaces (like other records rubbing against each other).

3. RETURN RECORDS TO THEIR SLEEVES AFTER EACH TIP. It only takes a second to slip the record back into the sleeve and it will improve record life and reduce the possibility of scratching.

4. REPLACE WORN STYLUS IMMEDIATELY. This has been thoroughly explained earlier.

5. SET THE STYLUS DOWN GENTLY. This is obvious. If you drop the stylus, you are asking for compound damage to records, stylus and cartridge.

MONITORS.

Monitoring of your calling is a choice each caller makes. You should always be aware of the sound that is being produced for the dancers. Many manufacturers provide an output on the amplifier for a monitor speaker as well as main speakers. There are times when you may be in places where you are not able to be in front of the hall speakers or you may also want to just hear your voice. The monitor speaker is intended to allow the caller to hear himself on stage; it is NOT intended to add sound to the dance floor. If you feel the need to use the monitor, be sure that the sound system you are using is equipped with a monitor speaker output. Again, characteristic impedance of the monitor output must be matched to the monitor speaker. Refer to the manufacturer's specifications for information concerning the type of monitor to be connected and the method of connection.

TAPING.

Taping of square dance programs is the subject of many discussions and borders on problems of copyrights. I will not discuss the issue of whether to allow taping but rather, will provide some advice on measures to be considered if you decide to allow taping of your dances. Unless your amplifier is specifically equipped with a tape recorder output jack, you should allow taping only in the "passive mode" such as allowing tape recorder microphones to be placed near or against your speakers or to use "telephone pickups." DO NOT jury rig tape recorder connections to your sound system just to please the tape addict. The tape recorder microphone or telephone pickup will get a better blend of sounds from your system and you will not subject your sound system to damage due to mismatched connections.

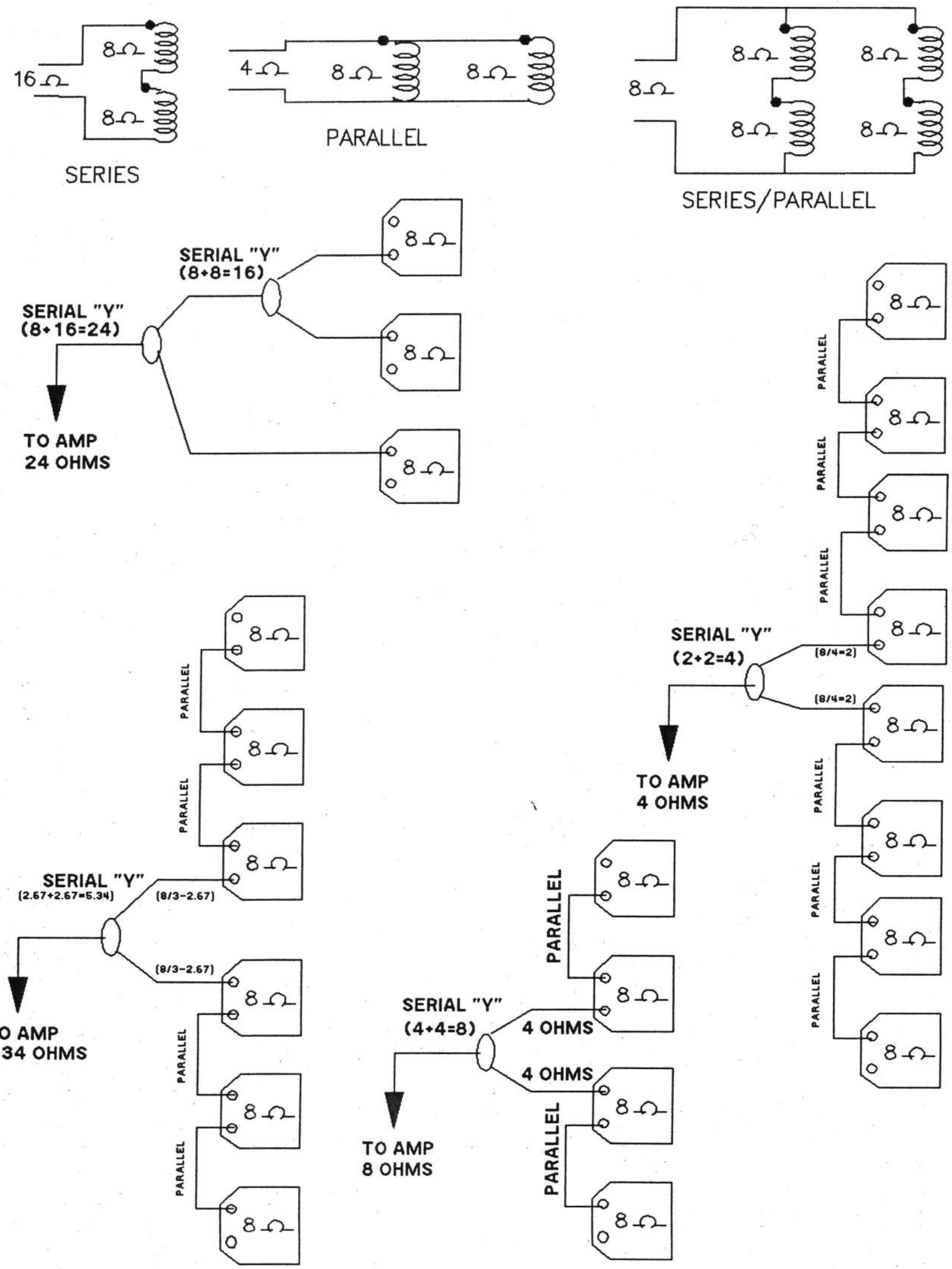
FEEDBACK.

One of the least understood problems encountered in sound systems is the problem of feedback. Feedback is that whine, whistle, or yowl you get when you pass the microphone too close to the speaker. Feedback is caused when the sound system output is reintroduced to the input of that system either through the microphone or through failure of the system cabling (wiring) to isolate the input and output of the system. Most manufacturers have designed their systems to reduce susceptibility to feedback but occasionally, we get feedback when we bring the microphone too close to the speaker while the microphone gain control is set too high. Feedback can be eliminated by reducing the microphone gain control, increasing the bass control slightly or a combination of these actions. And of course, by moving the microphone from in front of the speaker. Occasional feedback is inevitable but repeated incidence of feedback is a system is an indication of some sort of failure in the sound system. These indications should not be ignored. If you consistently get feedback, have your system checked by a technician.

SUMMARY.

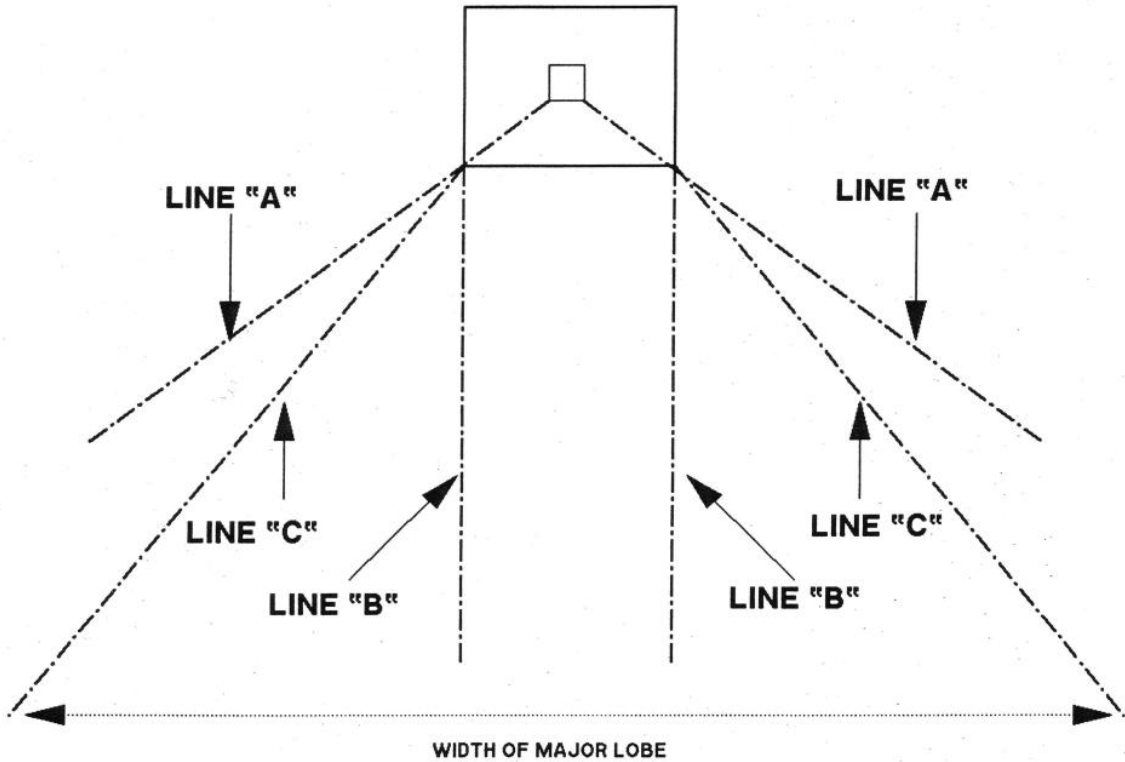
In summary, I hope that you will become completely familiar with your sound system's capabilities and limitations. If you do not already have them, you should obtain the manufacturer's specifications for your system. Each manufacturer will gladly supply specifications. They will also be happy to refer you to a representative in your area who will provide expert services. You should examine your equipment handling habits to be sure that you treat the equipment with the respect it deserves. Last, but by no means least, you should develop habits which provide the best results from your system. Good microphone technique, attention to proper speaker placement, and use of proper sound levels will yield the most enjoyment for the dancers who share your activity.

FIGURE I



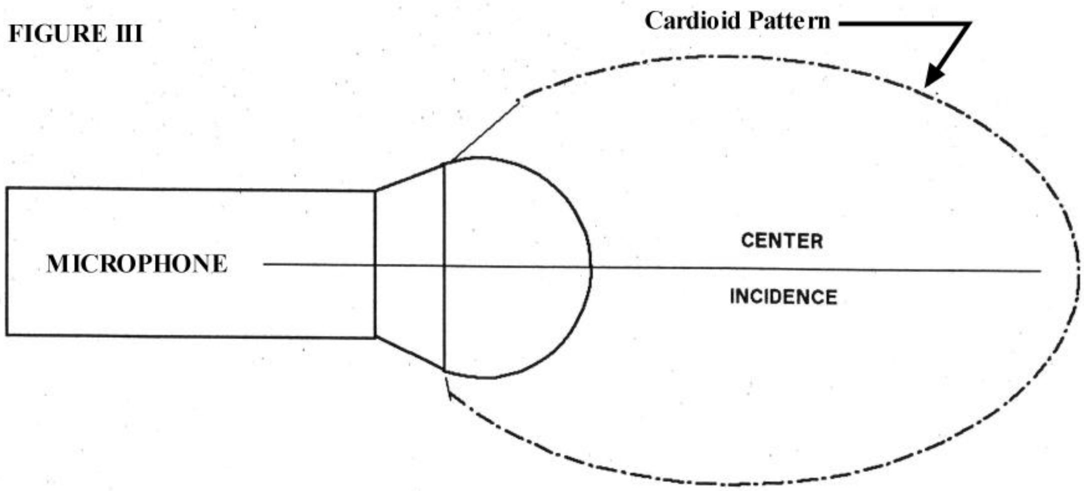
VOICE COIL AND SPEAKER COMBINATIONS TO ACHIEVE IMPEDANCE MATCHING

FIGURE II



SPEAKER LOBE DIAGRAM

FIGURE III



MICROPHONE CARDIOID PATTERN